SIP: Session Initiation Protocol

From HTTP and Session Invitation to Setup and Control for Packet-based Multimedia Conferencing

Conference Establishment & Control

Workshop
1. Create
Descr.: Upperside SIP 2005
Orig.: J.Ott jo@tzi.org 327689113
Info: http://www.tzi.org/dmn/
Start: 25.01.2005 / 09:30
End: 25.01.2005 / 17:30
Media: Audio PCM 234.5.6.7/39000
Media: Video H.263 234.5.6.8/29000
Media: Slides PDF 234.5.6.9/49000

2a. Announcement
   → Announcement Protocol
   → Netnews
   → WWW

2b. Invitation
   → E-Mail
   → Invitation Protocol

2c. Request
   → Streaming Protocol

3. Join

4. Media streams
History of Mbone conference initiation

**Session Invitation Protocol**

(Handley/Schooler)

- Participant location
- Conference invitation
- Capability negotiation during setup

**Simple Conference Invitation Protocol**

(Schulzrinne)

- Participant location
- Conference invitation
- Capability negotiation during setup
- Changing conference parameters
- Terminate/leave conference

1996 Session Initiation Protocol

Session Initiation Protocol (SIP)

First draft in December 1996

- Joint effort to merge SIP and SCIP
- IETF WG MMUSIC
  (Multiparty Multimedia Session Control)

Application-layer call signaling protocol:

- Creation, modification, termination of teleconferences
- Negotiation of used media configuration
- Re-negotiation during session
- User location → personal mobility
- Security
- Supplementary services

RFC 3261
- June 2002
- obsoletes RFC 2543
SIP and Conferencing over Time…

- Origin: MMUSIC: Multiparty Multimedia Session Control
- From Invitation… to initiation, modification, and termination
- From Multiparty… to point-to-point-focused
- From Multimedia… to voice-centric

Now: Multiparty & multimedia rediscovered

But: Don’t believe in multicast (anymore)!

Timeline: 1996

Initial Internet Drafts:
- Session Invitation Protocol (SIP) – M. Handley, E. Schooler
- Simple Conference Invitation Protocol (SCIP) – H. Schulzrinne

SIP: Setup + Caps Negotiation
SCIP: Setup + Caps Modify + Terminate
Merged Draft: SIP -01
Main Features set: TCP/UDP, Forking, Redirection, addr
INVITE,CAPABILITY
From: To: Path:

Presentations at 35th IETF, Los Angeles

22 Feb 1996 4-8 Mar 1996 2 Dec 1996
Timeline: 1997

- **Draft SIP -02**: Formal syntax
  - CAPABILITY → OPTIONS
  - Path: → Via:
  - Ideas for Alternates:

- **IETF Action**: Split SIP into base spec and extensions

- **Draft SIP -03**: SIP URL: sip://jo@...
  - CONNECTED, BYE, REGISTER
  - Call-ID: Sequence: Allow: Expires:

- **Dec 97**:

Timeline: 1998

- **SIP -05**:
  - CANCEL
  - UNREGISTER → Ø
  - URL sip://jo → sip:jo
  - Record-Route:
  - IANA assignments
  - Security Cons. Sect.

- **IETF Action**: Last Call for Proposed

- **SIP -07**: Call Hold SDP

- **Clarifications & fixes**: Cleaning up the spec
  - Call-ID: MUST tag parameter

- **SIP -08**

- **SIP -09**

- **8 Aug 18 Sep 28 Sep**
Timeline: 1998/99

**SIP -10**
No more DNS MX
URI: RFC 2396

**SIP -11**
Update on SDP part

**SIP -12**
DNS Lookup
Tidying up

**IETF Action:** Approval for Proposed Standard

**IETF Action:** Published as RFC 2543

**IETF Action:** Approval for Proposed Standard

12 Nov 98 15 Dec 98 15 Jan 99 2 Feb 99 17 Mar 99 Sep 99

Timeline: RFC2543bis (2000/2001)

**bis -00**
IETF Action: Formation of new SIPPING WG

**bis -01**
PGP removed

**bis -02**
Spring 01

**bis -03**
PGP removed

**bis -04**
Complete Rewrite!

**bis -05**
Complete Rewrite!

13 Jul 00 24 Nov 00 29 May 01 20 Jul 01 26 Oct 01 28 Nov 01

- **bis -07**
  - Offer/answer
  - Loose src route

- **bis -06**
  - TCP mandatory
  - 1xx-reliability

- **bis -08**
  - Sips URI
  - 1xx-reliability
  - In separate doc

- **bis -09**
  - IETF Last Call

- **IETF Action:**
  - RFC 3261–3266

- **SIP-related RFC Rallye:**
  - RFC 3361, 3372
  - RFC 3311, 3312
  - RFC 3323–3325, 3329 (Security)
  - RFC 3398, 3420, 3428
  - RFC 3320–3322 (SigComp)

- Until Jan 03

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"Weight" of SIP Base Spec

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IETF SIP-related Working Groups (1)

- MMUSIC WG
  - RFC 2543 (Feb 1999)
  - Sep 99

- SIP WG
  - Mar 01

- SIPPING WG
  - Dec 00

- SIMPLE WG
  - Oct 03

- XCON WG

IETF SIP-related Working Groups (2)

- MMUSIC WG
  - SDP extensions
  - SDPng

- SIP WG
  - SIP core spec maintenance
  - SIP protocol extensions

- SIPPING WG
  - Requirements for SIP
  - Specific SIP application services

- SIMPLE WG
  - SIP for Presence and Instant Messaging

- XCON WG
  - Centralized Conferencing
"Productivity" (1): Internet Draft Pages

(rough estimate with errors)

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"Productivity" (2): RFC Pages

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For exclusive use with TKK Netlab course S-38.3150 Networked Multimedia Protocols and Services
RFCs related to SIP (1)

- Original base spec
  - RFC 3261: SIP: Session Initiation Protocol
  - RFC 3263: Locating SIP Servers
  - RFC 3264: An Offer/Answer Model with SDP

- Extended Features
  - RFC 2376: The SIP INFO Method
  - RFC 3262: Reliability of Provisional Responses in SIP
  - RFC 3265: SIP-specific Event Notification
  - RFC 3311: SIP UPDATE Method
  - RFC 3312, RFC 4032: Integration of Resource Management and SIP
  - RFC 3326: Reason Header
  - RFC 3327: Registering Non-Adjacent Contacts
  - RFC 3428: Instant Messaging
  - RFC 3487: Requirements for Resource Priority
  - RFC 3515: SIP REFER Method
  - RFC 3581: Symmetric Message Routing
  - RFC 3660: SIP event package for registrations
  - RFC 3725: Third-party Call Control (3PCC)
  - RFC 3840, 3841: Callee capabilities and caller preferences
  - RFC 3842: Message waiting indication / message summary
  - RFC 3857, 3958: Watcher Information event package + XML format
  - RFC 3891: Replaces: header
  - RFC 3892: Referred-By: header
  - RFC 3903: Event state publication (SIP PUBLISH method)
  - RFC 3911: Join: header
  - RFC 4028: Session timers
  - RFC 4168: SCTP as transport protocol
RFCs related to SIP (2)

- Extended features (continued)
  - RFC 4244: Request history
  - RFC 4320: Addressing issues with non-INVITE transactions
  - RFC 4412: Communications resource priority for SIP
  - RFC 4483: Content indirection in SIP
  - RFC 4488: Suppressing implicit subscriptions of REFER
  - RFC 4508: Conveying feature tags with REFER
  - RFC 4235: INVITE-initiated dialog event package
  - RFC 4425: Requirements for SIP conferencing
  - RFC 4376: SIP conferencing framework
  - RFC 4376: Floor control requirements
  - RFC 4411: SIP Reason header for preemption
  - RFC 4453: Requirements for consent-based communications
  - RFC 4475: SIP torture test messages
  - RFC 4479: A data model for presence
  - RFC 4480: RPID: rich presence
  - RFC 4481: Extensions for timed presence
  - RFC 4482: CPID: Contact information in presence
  - RFC 4575: SIP conference event package
  - RFC 4576: SIP call control: conferencing for user agents
  - RFC 4596: Caller preferences extensions
  - RFC 4597: Conferencing scenarios
  - RFC 4660: Functional description of event filtering
  - RFC 4661: XML for event filtering
  - RFC 4662: Event notifications for resource lists
  - RFC 4730: Key Press Stimulus Event Package (KPML)
  - RFC 4916: Connected identity

RFCs related to SIP (3)

- Extended features (continued)
  - RFC 4825: XCAP
  - RFC 4826: XCAP Processing Rules for Resource Lists
  - RFC 4827: XCAP For Manipulating Presence Contents
  - RFC 4975: MSRP
  - RFC 4976: MSRP Relays

- Security
  - RFC 3323: A Privacy Mechanism for SIP
  - RFC 3325: Private Extension for Asserted Identity in Trusted Networks
  - RFC 3329: Security-Mechanism Agreement for SIP
  - RFC 3603: Proxy-to-Proxy Extensions
  - RFC 3702: AAA requirements for SIP
  - RFC 3853: S/MIME AES
  - RFC 3893: Authenticated Identity Body
  - RFC 4189: Requirements for end-to-middle security
  - RFC 4474: Enhancements for authenticated identity management
  - RFC 4484: Trait-based authentication requirements
  - RFC 4538: Request authorization through dialog identification
RFCs related to SIP (4)

- Others
  - RFC 3665, 3666: SIP Call Flows
  - RFC 3361: DHCP Option for SIP Servers
  - RFC 3608: Service Route Discovery
  - RFC 3398, 3578: ISUP and SIP Mapping
  - RFC 3420: Internet Media Type message/sipfrag
  - RFC 3427: SIP Change Process
  - RFC 3455: Header Extensions for 3GPP
  - RFC 3485, 3486: SIP header compression
  - RFC 3764, 3824: Using ENUM with SIP
  - RFC 3959: Early Session disposition type (early-session, session)
  - RFC 3960: Early Media and Ringing Tone Generation
  - RFC 3968, 3969: IANA SIP header field and URI registry
  - RFC 3976: SIP – IN Interworking
  - RFC 4117: 3rd party call control invocation of transcoding services
  - RFC 4123: SIP – H.323 Interworking requirements
  - RFC 4485: Guidelines for authors of SIP extensions
  - RFC 4497: SIP – QSIG interworking
  - RFC 4569: IANA media feature tag registration
  - RFC 4780: SIP MIB

- Related: RTP, SDP, Security basics, 3GPP requirements and extensions

A Hitchhikers Guide to the Session Initiation Protocol (SIP)
draft-ietf-sip-hitchhikers-guide-04.txt

SIP is not …

- Intended for conference control by itself
  - No floor control
  - No participant lists
  - No policies, voting, …

- Designed for distribution of multimedia data
  - Some extensions allow for carrying images, audio files, etc.

- A generic transport protocol!
- Another RPC mechanism
  - SIP has no inherent support for distributed state information
- Something to put into every device on the planet
  - No general IP infrastructure part (yet?)
- Nevertheless: Application layer routing gets more and more important
- (but proposals for “misuse” show up again and again)
SIP and the Multimedia Conferencing Architecture

Audio / Video

Conference Control

SDP

Call Control

SIP

RSVP

RTSP

RTP / RTCP

SAP

UDP

IP/IP Multicast

TCP, SCTP, TLS

Integrated Services Forwarding

Base Terminology

- User Agent Client (UAC):
  - Endpoint, initiates SIP transactions
- User Agent Server (UAS):
  - Handles incoming SIP requests
- Redirect server:
  - Retrieves addresses for callee and returns them to caller
- Proxy (server):
  - Autonomously processes and routes requests
  - forward incoming messages (limited modifications only)
- Registrar:
  - Stores explicitly registered user addresses
- Location Service:
  - Provides information about a target user’s location
- Back-to-Back User Agent (B2BUA):
  - Keeps call state; more powerful intervention than proxy
Local SIP Architecture

- **Administrative Entity (SIP Server)**
  - Registrar
  - Redirect / Proxy Server
  - Location Server

- **Endpoint**
  - SIP UA

- **Local IP network**

Protocol Characteristics

- **Transaction oriented**
  - Request–response sequences

- **Independent from lower layer transport protocol**
  - Works with a number of unreliable and reliable transports
    - UDP, TCP, SCTP
    - Secure transport: TLS over TCP, IPSec
  - Retransmissions to achieve reliability over UDP
  - Optionally use IP multicast \( \rightarrow \) anicast service

- **Independent of the session to be (re-)configured**

- **Re-use syntax of HTTP 1.1**
  - Text-based protocol (UTF-8 encoding)

- **Enable servers maintaining minimal state info**
  - Stateless proxies
  - Transaction-stateful proxies
  - Dialog (call) state in endpoints (optional for proxies)
**Functional Layers**

- **Transaction User**
  - Session creation, application-specific processing

- **Transaction**
  - Transaction handling
  - Request retransmission

- **Transport**
  - Send/receive SIP messages

- **Syntax / Encoding**
  - Message parsing

**Transport Protocol**
- UDP
- TCP
- SCTP
- TLS

**SIP Transactions**

- RPC-like approach:
  - Initial request
  - Wait for final response

- Provisional responses:
  - Additional status information
  - May be unreliable

- Unique identifier (transaction id): (originator, recipient, unique token, sequence number, ...)

- Independent completion
Dialogs

- Signaling vs. media session
- Distributed state between endpoints
  - State change if transaction succeeds
  - No change on error
- Unique dialog identifier

Dialog Example: Media Sessions

Special case: three-way handshake for INVITE transaction

INVITE → Ringing → OK → ACK → INVITE

prepare media session;
establish media session, dialog
media session in progress

early dialog

INVITE → OK → BYE → ACK

create media session, dialog
media session in progress
terminate media session

Media Streams

A
B

create media session, dialog
create media session, dialog

create dialog
modify dialog
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SIP Message Syntax: Request

Start line

```
INVITE sip:user@example.com SIP/2.0
```

Message headers

```
To: John Doe <sip:user@example.com>
From: sip:jo@tzi.uni-bremen.de;tag=4711
Subject: Congratulations!
Content-Length: 117
Content-Type: application/sdp
Call-ID: 2342344333@134.102.218.1
CSeq: 49581 INVITE
Contact: sip:jo@134.102.224.152:5083
    ;transport=udp
Via: SIP/2.0/UDP 134.102.218.1;
    branch=z9hG4bK776asdhds
v=0
o=jo 75638353 98543585 IN IP4 134.102.218.1
s=SIP call
i=0 0
c=IN IP4 134.102.224.152
m=audio 47654 RTP/AVP 0 1 4
```

Message body (SDP content)

```
```

SIP Message Syntax: Response

Start line

```
SIP/2.0 200 OK
```

Message headers

```
To: John Doe <sip:user@example.com>;tag=428
From: sip:jo@tzi.uni-bremen.de;tag=4711
Subject: Congratulations!
Content-Length: 121
Content-Type: application/sdp
Call-ID: 2342344333@134.102.218.1
CSeq: 49581 INVITE
Contact: sip:jdoe@somehost.domain
Via: SIP/2.0/UDP 134.102.218.1;
    branch=z9hG4bK776asdhds
v=0
o=jdoe 28342 98543601 IN IP4 134.102.20.22
s=SIP call
i=0 0
c=IN IP4 134.102.20.38
m=audio 61002 RTP/AVP 0 1 4
```

Message body (SDP content)

```
```
SIP URI Addressing Scheme

- Follows basic URI syntax per RFC 2396
- Separating names (permanent) and addresses (temporary)
  - Basic mobility support
- Two roles reflected in SIP
  - Naming a user; typically sip:user@domain
  - Contact address of a user; typically contains
    host name or IP address, port, transport protocol, ...
- URIs may carry additional parameters

```
sip:[ user[ : passwd ] @ ] host[ : port ] params [ ? headers ]
```

```
params ::= ( ; name [ = value ] )*
headers ::= field = value? [ & headers ]
```

- URIs may also identify services

SIP URI Addressing Examples

- **sip:tzi.org**: Registration domain or IP address
- **sip:192.168.42.1**: SIP URI to call (Address of Record)
- **sip:john@example.com**: SIP Contact Address (actual user location)
- **sip:voicemail@service.com**: Service identifier; semantics opaque to the user
- **sip:conf-1234@confserv.com**
- **sip:user34@anonymizer.org**

Use URI scheme ‘sips’ to request secure communications.
SIP URI Addressing Examples (2)

URI parameters may carry detailed information on specific URI components:

```
sip:john@Example.COM;maddr=10.0.0.1
sip:+1555123456@tel-gw.myitsp.com;user=phone
```

Nested URI Encoding (e.g. for Service Description)

Encapsulation

```
sip:sip:3A1b%40.92.168.42.1%3Bmaddr=134.102.3.99@example.com
```

Need to encode reserved characters

Service indication example

```
sip:voicemail.replay=abl%X817m@media-engine;msgid=78
```

Additional header fields (line breaks inserted for readability)

```
sip:jo@example.com?Replaces=abcd@example.com%3B \
from=tag%3D234bto=tag%3D234abl&Accept-Contact= \
%3Ctip%3A5640134.102.218.1%3E%3Bonly%3Dtrue
```

Separator characters
URIs in Header Fields

URI-parameters vs. header parameters

Contact: sip:bob@p2.example.com:55060;methods="NOTIFY";expires=3600

→ angle brackets:

Contact: <sip:bob@p2.example.com:55060;methods="NOTIFY">;expires=3600

Required if
- URI contains comma, question mark or semicolon
- The header field contains a display name

Further Common URI Schemes

Telephony (RFC 2806)

tel:+1-555-12345678
tel:7595;phone-context=+49421218

ITU-T H.323 Protocol

h323:user@example.com

Instant Messaging

im:user@example.com

Presence

pres:user@example.com