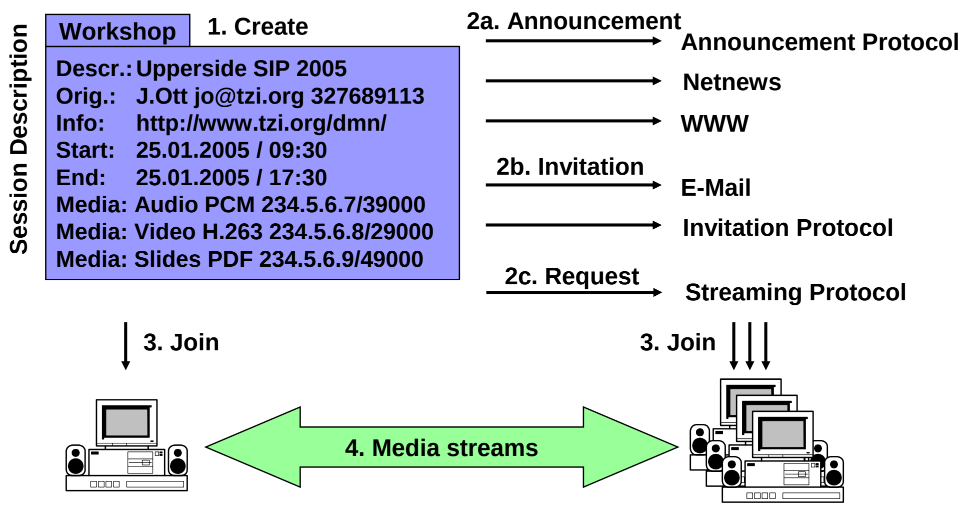
SIP: Session Initiation Protocol

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

Conference Establishment & Control



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***

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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

22 Feb 1996

SIP: Setup + Caps Negotiation

SCIP: Setup + Caps Modify + Terminate

4-8 Mar 1996

Presentations at 35th IETF, Los Angeles

2 Dec 1996

Merged Draft: SIP -01

Main Features set:
TCP/UDP, Forking, Redirection, addrs
INVITE,CAPABILITY
From: To: Path:

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:

Draft SIP -04

CONNECTED ➔ ACK
UNREGISTER
Sequence: ➔CSeq:
Call-Disposition:
Require:

27 Mar 97 | 31 Jul 97 | 11 Nov 97 | Dec 97

Timeline: 1998

SIP -05

CANCEL
UNREGISTER ➔ ∅
URL sip://jo ➔ sip:jo
Record-Route:
IANA assignments
Security Cons. Sect.

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

SIP -06

SIP -07

SIP -08

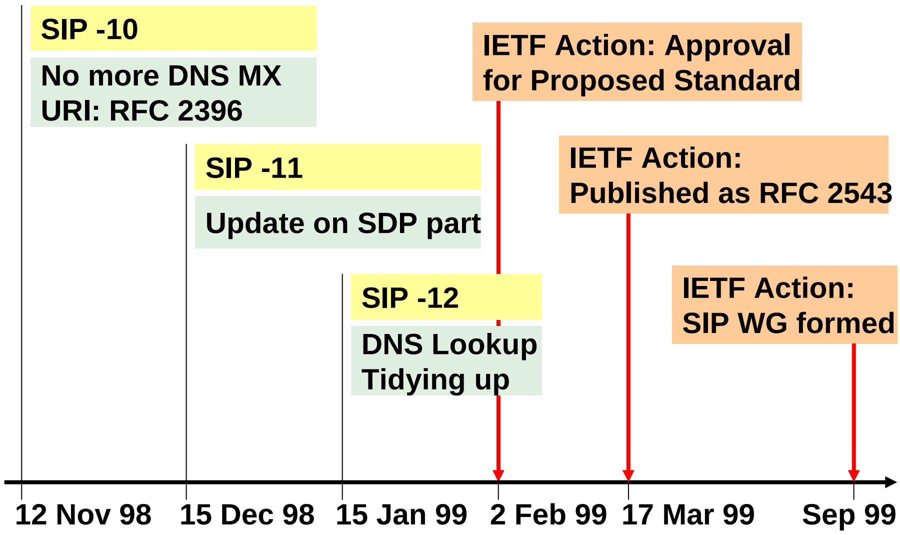
IETF Action: Last Call for Proposed

SIP -09

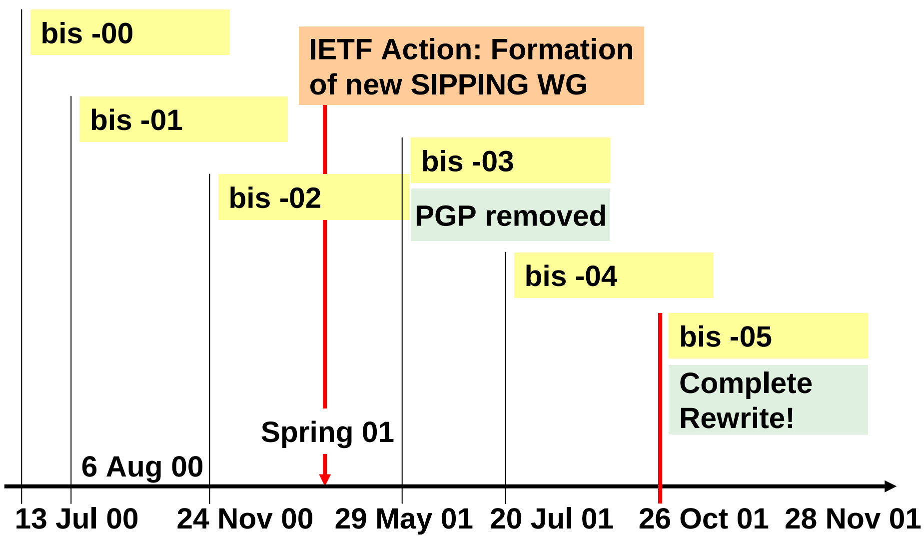
Call Hold SDP

14 May | 17 Jun | 16 Jul | 8 Aug | 18 Sep | 28 Sep

Timeline: 1998/99



Timeline: RFC2543bis (2000/2001)



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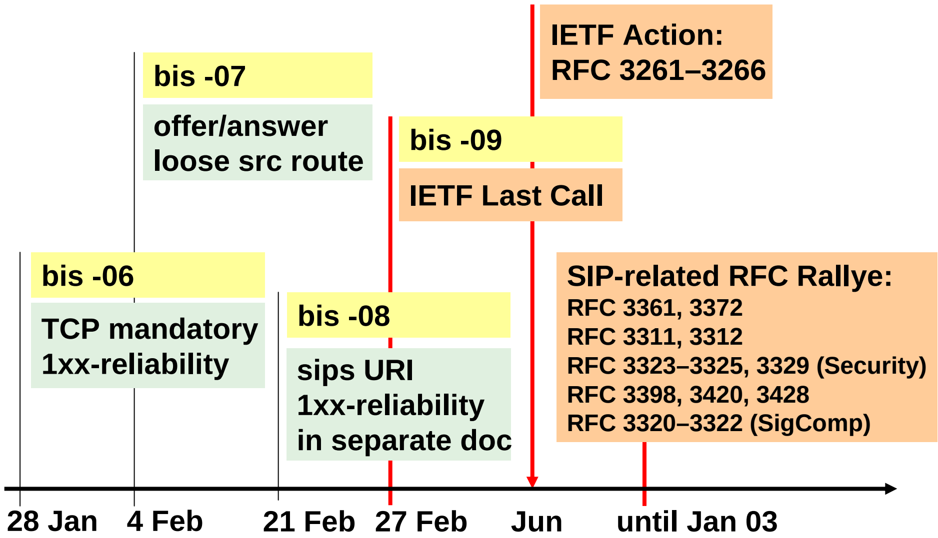
Timeline: RFC2543bis, RFC3261 (2002)



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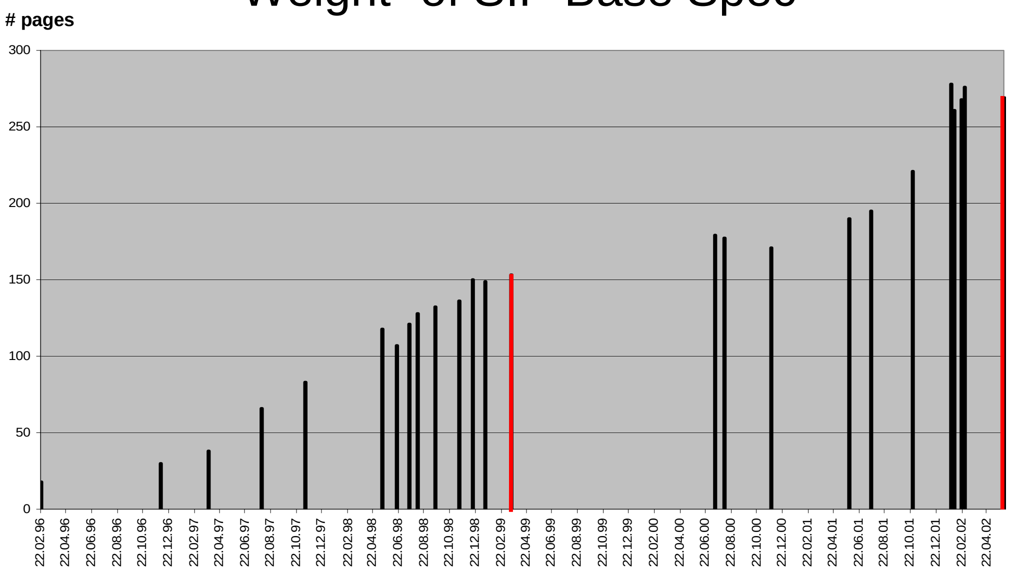
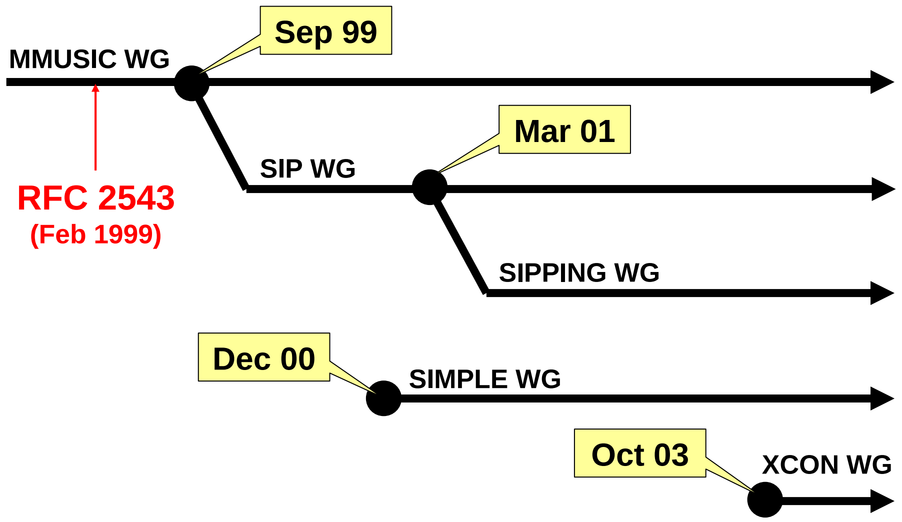
“Weight” of SIP Base Spec



IETF SIP-related Working Groups (1)



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

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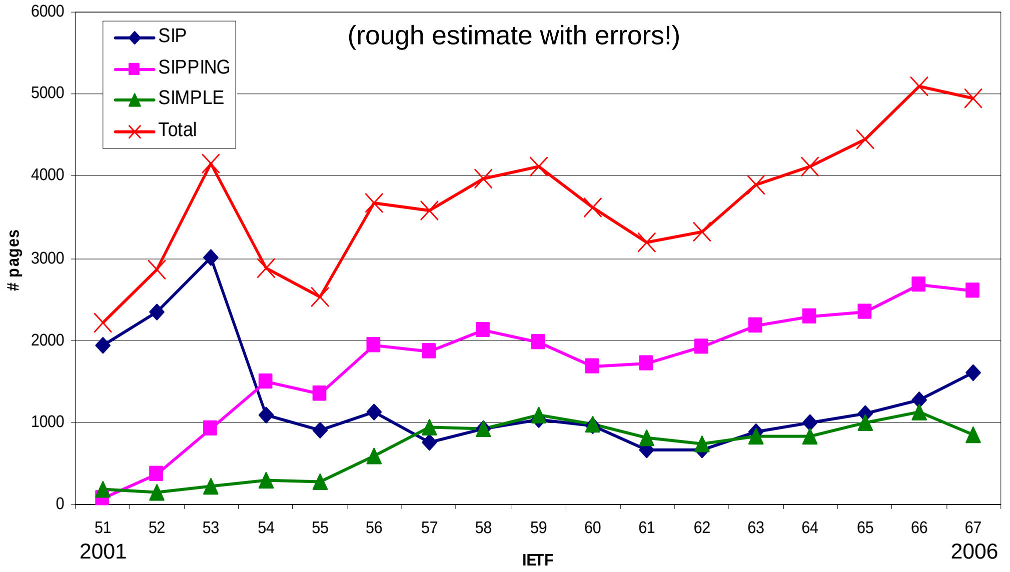
"Productivity" (1): Internet Draft Pages



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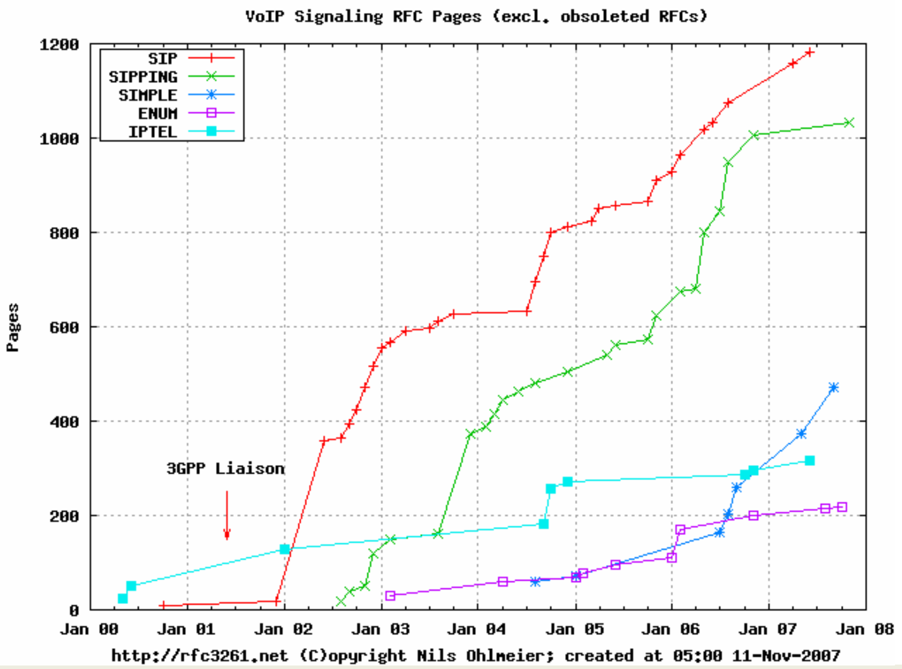
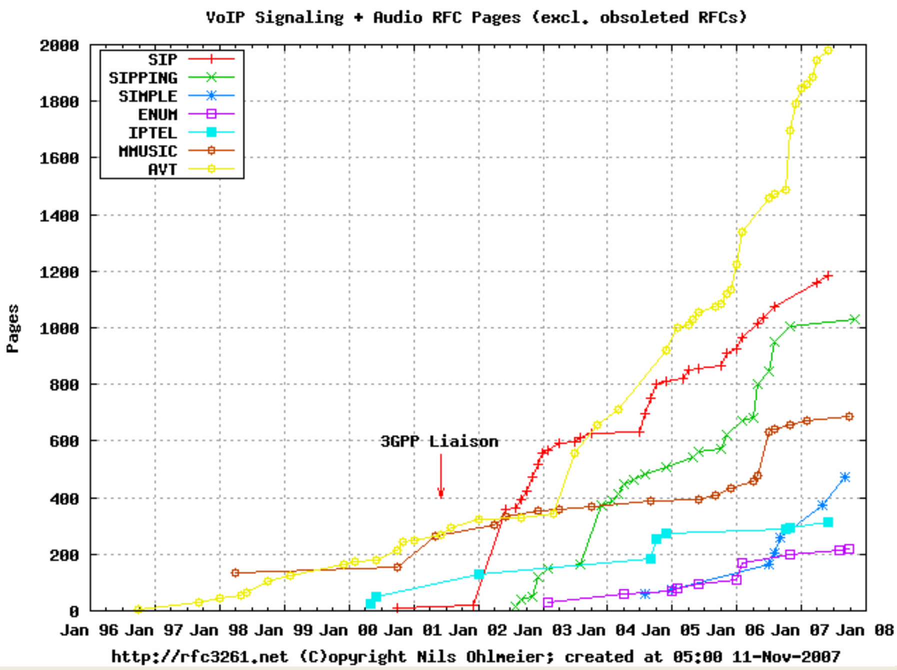
"Productivity" (2): RFC Pages



“Productivity” (3): RFC Pages



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 2976: 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 3680: 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 4321: Problems 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 4245: Requirements for SIP conferencing
  - RFC 4353: 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 4579: 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

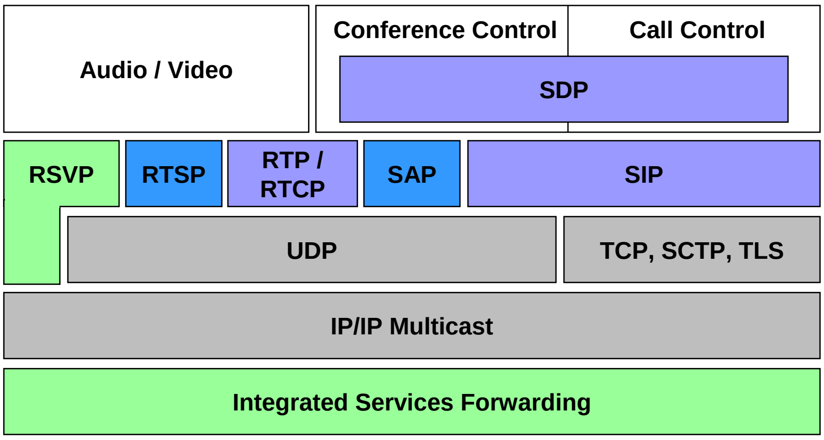
Plus some 100+ Internet Drafts

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



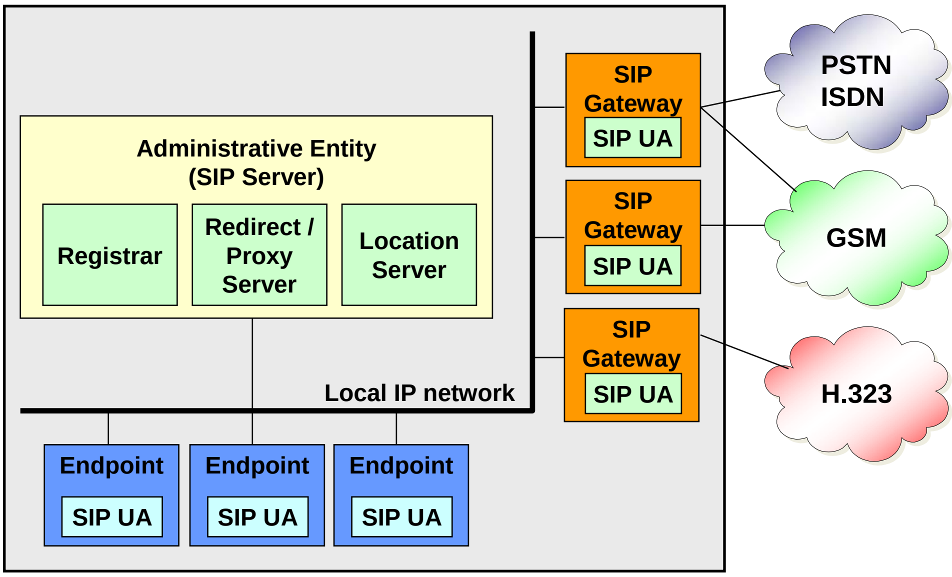
Base Terminology

- User Agent Client (UAC):
  - Endpoint, initiates SIP transactions
- User Agent Server (UAS):
  - Handles incoming SIP requests

User Agent

- 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



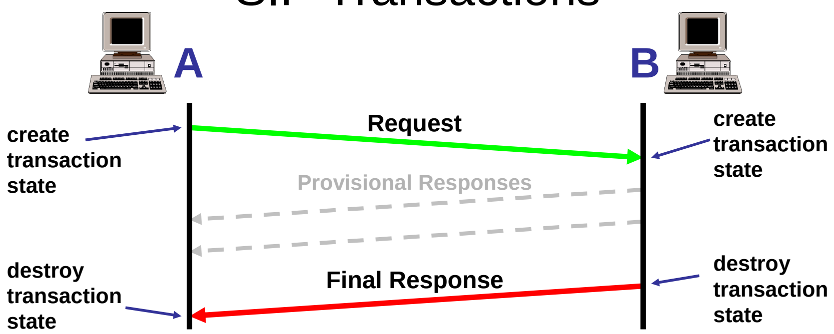
Protocol Characteristics

- 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 → anycast 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

Layer	Function
Transaction User	session creation, application-specific processing
Transaction	Transaction handling request retransmission
Transport	send/receive SIP messages
Syntax / Encoding	Message parsing
UDP / TCP / SCTP (TLS)	Transport Protocol

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

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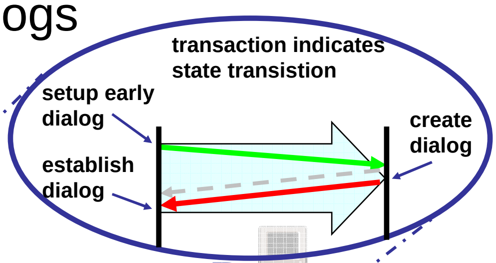
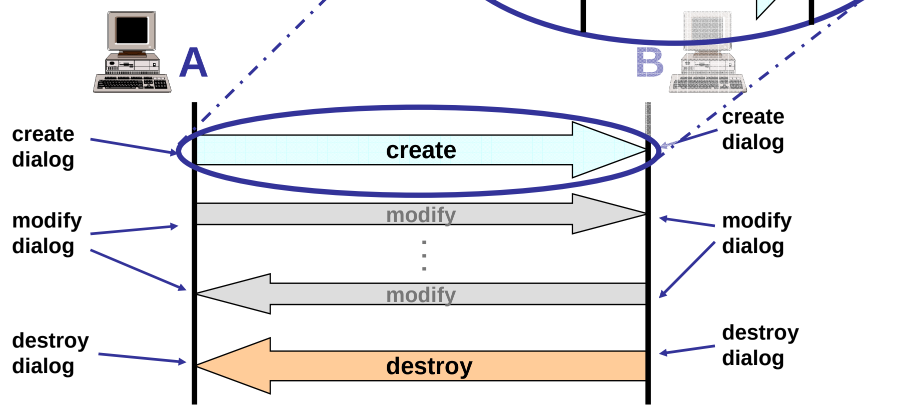
Dialogs

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





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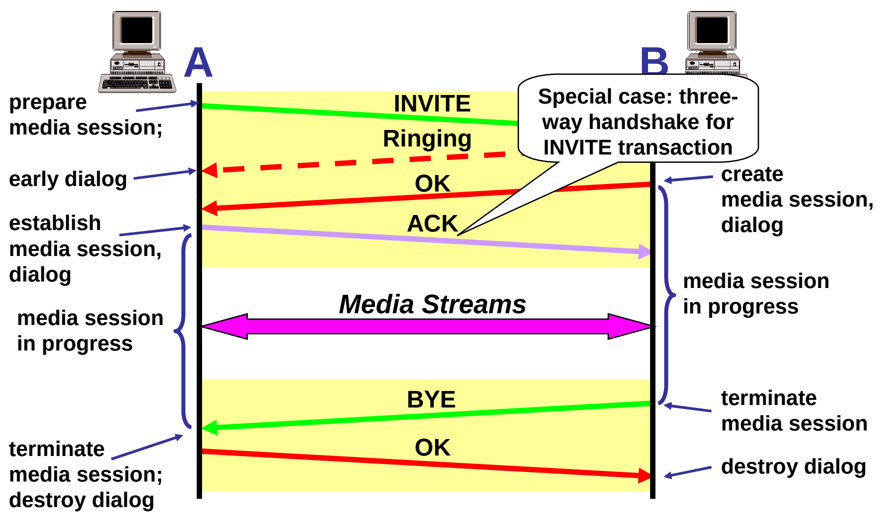
Dialog Example: Media Sessions



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: applicaton/sdp
Call-ID: 2342344233@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
```

Message body (SDP content)

```
v=0
o=jo 75638353 98543585 IN IP4 134.102.218.1
s=SIP call
t=0 0
c=IN IP4 134.102.224.152
m=audio 47654 RTP/AVP 0 1 4
```


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: applicaton/sdp
Call-ID: 2342344233@134.102.218.1
CSeq: 49581 INVITE
Contact: sip:jdoe@somehost.domain
Via: SIP/2.0/UDP 134.102.218.1;
      branch=z9hG4bK776asdhds
```

Message body (SDP content)

```
v=0
o=jdoe 28342 98543601 IN IP4 134.102.20.22
s=SIP call
t=0 0
c=IN IP4 134.102.20.38
m=audio 61002 RTP/AVP 0 4
```

SIP URI Addressing Scheme

sip: / sips:

- 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` `sip:192.168.42.1`	**Registration domain or IP address**
`sip:john@example.com`	**SIP URI to call (Address of Record)**
`sip:john@host1.example.com` `sip:john@192.168.42.9:9950`	**SIP Contact Address (actual user location)**

`sip:voicemail@service.com`
`sip:conf-1234@confserv.com`
`sip:user34@anonymizer.org`
	Service identifier; semantics opaque to the user

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%3Ajo%40192.168.42.5%3Bmaddr=134.102.3.99@example.com
```

Need to encode reserved characters

Service indication example

```
sip:voicemail.replay=ab1x817m@media-engine;msgid=78
```

Additional header fields (line breaks inserted for readability)

```
sip:sales@warehouse.com;method=INVITE \
?Subject=gw%20c2651&Call-ID=c239xa2-as921b%40warehouse.com

sip:jo@example.com?Replaces=abcd@example.com%3B  \
from-tag%3D28%3Bto-tag%3D234ab1&Accept-Contact=  \
%3Csip%3Ajo%40134.102.218.1%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:

URI parameter

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

Header parameter

✓

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
```