Overview

- IETF Multimedia Conferencing
- Key Features
- SIP Evolution
- SIPit: Interoperability Events
- Working Group(s) and Tasks
- Conclusion
IETF Conferencing

- Packet multimedia experiments since 1980s
  - Audio/video tools + protocols for A/V over IP
  - Conference announcement and control protocols
- First IETF Audiocast (1992)
- Since then: IETF sessions on the Mbone
  - Audio + video (+ sometimes slides)
- Other uses of Mbone conferencing
  - Lectures, seminars, project meetings, …
  - Broadcasting NASA missions, concerts, …

IETF Conferencing Architecture

- Conference Control
- Audio Video
- Media Streaming
- Shared Apps
- Session Direct.
  - SDP
- RSVP
- RTP / RTCP
- RTSP
- SAP
- SIP
- HTTP
- SMTP
- UDP
- TCP
- IP / IP Multicast
- Integrated / Differentiated Services Forwarding
IETF Conferencing Model

Session Description

Workshop 1. Create

2a. Disseminate
   SAP
   NNTP
   HTTP

2b. Invite
   SMTP
   SIP

3. Join

4. Media Streams

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

The latter is not SIP — but it is the way SIP is looked at today in many cases.
SIP Key Features...

Learning from and using other Internet protocols
- Clean, flexible, distributed architecture
- URL addressing scheme
- Well-defined logical functions
- User location, forking, redirection
- Media and session independent
- Text-based protocol syntax

Openness: support from a broad community…

Sample SIP System Architecture

[SIP System Architecture Diagram]

Provider X SIP domain
SIP Server
SIP signalling for initial call routing and setup
SIP in-call signaling
RTP / RTCP

Local SIP domain
SIP backbone network
Provider Y SIP domain
SIP Endpoint
SIP Endpoint
SIP Endpoint
SIP Endpoint
A Simple SIP Call

INVITE sip:bob@example.com

UA Ann  SIP Server(s)  UAs Bob

INVITE

100 Trying

ACK

200 OK

Call media stream

ACK

200 OK

BYE

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 -04
CONNECTED ➔ ACK
UNREGISTER
Sequence: ➔ CSeq:
Call-Disposition:
Require:

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

Timeline: 1998

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

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

IETF Action:
Last Call for Proposed

SIP -06
SIP -07
SIP -08
SIP -09
Call Hold SDP
“Weight” of SIP Base Spec

SIP Functionality and Services

- Set up, modify, and terminate media sessions
- Registration, user location, mobility
- Hooks for policing (and user control/preferences)
- Web and media integration

- Flexible service creation platform — end-to-end
- Telephony, PBX functions, …

Has been / will be covered in other talks!
What comprises SIP today?

RFC 2327: Session Description Protocol (SDP)
RFC 2543: Session Initiation Protocol + bis-03
RFC 2824: Call Processing Language
RFC 2976: The SIP INFO Method
RFC 3050: SIP CGI
RFC 3087: SIP Request-URIs for Service Control

SIP Guidelines
SIP Call Flows
Session Timer
Reliable provisional responses
Third Party Call Control
State Cookies

-term Banned because of lawsuit

-definition of a SiPit event:
- Informal gathering of engineers
- To test interoperability and robustness
- Of products and prototypes
- With on-site debugging

Goals: improve implementations and specs
SIP Interoperability Tests

- Tests
  - Peer to peer and in groups
- Test classification
  - Basic and Intermediate SIP capabilities for UAs
  - Advanced SIP capabilities for Proxies
  - Test scenarios
- Preparation
  - Documentation: Torture tests, call flows
  - Tools: Free / cheap implementations, servers, ...

Interop Events

1. Apr 1999  Columbia University  16
2. Aug 1999  Pulver.com  15
3. Dec 1999  Ericsson  26
4. Apr 2000  3Com  36
5. Aug 2000  Pulver.com  44
6. Dec 2000  Sylanstro/Sun  57
7. Mar 2001  ETSI  58
8. Aug 2001  Ubiquity
9. Dec 2001  Nuera
“Newer” Aspects for SIP

- (SIP for Telephony Interworking: SIP-T)
- SIP for Mobility (3G)
- SIP for QoS and Billing
  - Tough in the end-to-end world (“what to bill for?”)
- SIP and Firewalls
- SIP and Conferencing?
- SIP for Instant Messaging, Personal Presence
- Proposal: SIP for Appliances?

SIP for Conferencing…?

- Multimedia Capability Signaling
  - SDPng development

- Multiparty Conferencing
  - Invitation to Mbone sessions
  - Initiation, media control, termination
  - Distribution of membership information
  - And other state?

- SIP for Conference Management?
  - LIMITED INTEREST
SIP Working Group(s)

- MMUSIC
  - Developed SIP from Feb 1996 to Feb 1999
  - Still takes care of SDP and SDPng
- SIP
  - Initiated in Oslo (Sep 1999) for “load balancing”
  - Look after the base spec + core protocol extensions
- SIPPING
  - About to be approved by the IESG
  - Work on applications of SIP
  - and possibly feed requirements to SIP WG

SIP WG

- RFC 2543 bis
- SIP Call Control
- Caller preferences, server features
- Reliable provisional responses
- Session timers
- SIP MIB
- State Cookies
- Security and Privacy
- SIP Events
- NAT-friendly SIP
SIPPING WG (Draft Stage)

- SIP for Telephony (SIP-T)
- SIP Support for Hearing Impaired Users
- H.323 Interworking
- 3G Networks
- SIP Usage Guidelines
- Multiparty Conferencing
- SIP Application Components
- Call Control, Message Waiting, …
- Living w/ MIME, DNS, DHCP, ENUM, …
- …

SIP-related Groups

- PINT: origin of SUBSCRIBE/NOTIFY
- IPTEL: CPL and TRIP
- SIMPLE: SIP for Presence
- SPIRITS: SIP as “transport” mechanism
- PacketCable DCS
- 3GPP, 3GPP2
  - Using SIP for the next generation wireless networks
- ETSI Tiphon, IMTC: H.323 Interworking, Tests
- SIP Forum
Important: Keeping SIP “Clean”

- “Trendy” standards attract many contributors
  - well, sometimes too many contributors...
- Difficult to maintain architectural integrity
  - explosion of functions, fields, uses, interpretations, ...
- Sheer volume of contributions hard to co-ordinate
- When SIP is no longer used as SIP...
  - “We use SIP - but with the following changes…”
  - “SIP for everything - just because it’s there…”
- Risks for durability and future evolution

Conclusion

- SIP has changed and expanded quite a bit
- Managed to keep its architectural integrity

Key components:
- Internet-style protocol for the Internet
- Developed and reviewed in an open process

But:
- Don’t SIP everything!
Further Information

www.ietf.org/html.charters/sip-charter.html

www.softarmor.com/sipwg

www.cs.columbia.edu/~hgs/sip

www.cs.columbia.edu/~hgs/sip/sipit