

SIP WG Status

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Overview

- SIP Working Group(s)
- SIP WG Rules
- SIP Work Items
- SIP Today and Tomorrow
- Related Work in the IETF
- SIP Standardization Issues
- Conclusion

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
 - Initiated in Minneapolis (Mar 2001) – same reason
 - About to be approved by the IESG
 - Work on applications of SIP

SIP WG Status

Kind of busy...

- ~25 Active Drafts
- 13 items on Last Call Calendar
- 2 day interim meeting in February
- 3 meetings at last IETF + several Bar BOFs
- 700+ mails over last two months
- 2 meetings at next IETF + several Bar BOFs

SIPPING WG Status

- Split decided at last IEFT
- New WG close to approval by IESG
- Specify uses and applications of SIP
- Derive and elaborate requirements on SIP
- Feed new requirements to SIP WG
 - to consider appropriate SIP extensions
- First meeting(s) at 51st IETF
- ~40 Internet Drafts to look after

SIP-related Groups

- PINT: origin of SUBSCRIBE/NOTIFY
- IPTEL: CPL and TRIP
- SIMPLE: SIP for Presence (+ IMPP to define payload)
- SPIRITS: SIP as “transport” mechanism

- PacketCable DCS
- SoftSwitch Consortium
- 3GPP, 3GPP2
 - Using SIP for the next generation wireless networks
- ETSI Tiphon, IMTC: H.323 Interworking, Tests
- SIP Forum, SIP Center



What are we doing...?

SIP Work Items

- RFC 2543 bis
- **SIP Call Control**
- Caller preferences, server features
- Reliable provisional responses
- Session timers
- **SIP MIB**
- State Cookies
- **Security and Privacy**
- **Packet Cable DCS Convergence**
- **SIP Events**
- NAT-/Firewall-friendly SIP

SIPPING Work Items

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



How are we doing it...?

SIP Process Demystified

- “Why does it take so long...?”
- Process to move documents ahead...
 - Tracking documents and nagging people
 - Rakesh Shah from dynamicsoft volunteered
 - Helps to keep the overview of what is going on
 - WG web pages updates (together w/ Dean Willis)
- Information at our supplemental web site

Remember...

- We are trying to make standards.
 - Aiming for quality – so this takes a while.
- Not every RFC is a [{{proposed,draft}}] standard.
 - Informational and Experimental RFCs
 - (Those may become de-facto standards though.)
- An Internet-Draft has no standing whatsoever!
- Many Internet-Draft will silently disappear.
 - Wait for a stable spec to implement against...

SIP Last Call Process (1)

1. Proposal to go to WG Last Call
 - Create tracking page (so we know what happens)
2. Initial Consensus
 - Chairs review, inquire list, determine consensus
 - Hand-over to “Last Call Coordinator”
3. Pre-screening
 - NITS review: 1 reviewer
 - Make the draft “formally” IESG-proof
4. Prioritization & Scheduling
 - Detailed review: 3 reviewers
 - WG Last Call

SIP Last Call Process (2)

5. WG Discussion

- List discussion of issues, suggestions, solutions
- Modify and re-submit draft as needed
- Re-issue WG Last Call (if needed)

6. Determine WG Consensus

- May incur further work (and may start over again)

7. Hand-over to IESG

8. IESG Decision Process



When will it be done...?

SIP Today

RFC 1889: Real-time Transport Protocol (RTP)

RFC 1890: RTP Profile for Conferencing

RFC 2198: Redundancy for RTP

RFC 2327: Session Description Protocol (SDP)

RFC 2543: Session Initiation Protocol (+ bis-03)

RFC 2824: Call Processing Language (CPL)

RFC 2833: Tones over RTP (“DTMF”)

RFC 2976: The SIP INFO Method

RFC 3050: SIP CGI

RFC 3087: SIP Request-URIs for Service Control

SIP Tomorrow

- Autoconfiguration
 - DHCP option for SIP
 - SIP server location
 - (phone control – no SIP WG activity)
- SIP Server Features
 - Supported: Unsupported: Proxy-Require:
- SIP ISUP MIME
- Reliable Provisional Responses
 - PRACK method

SIP: The Day After Tomorrow

- Session Timer
- SIP Call Flows

- Call Control Framework
- Call Transfer

- SIP-T
 - SIP ISUP interworking
 - SIP overlap sending

SIP Next Week

- SIP Guidelines
- Application Components Outline
- SIP Caller Preferences
- SIP Security Requirements
- SIP Privacy
- SIP Session State
- SIP Resource Condition Met (COMET method)
- SIP MIB
- SIP Events (SUBSCRIBE / NOTIFY)
- H.323 Interworking Requirements

SIP further down the road...

- SIP for Mobility (3G)
- SIP with QoS and Billing
 - Tough in the end-to-end world (“what to bill for?”)
- SIP and Conferencing
- Others...
- Proposal: SIP for Appliances?

SIP for Draft Standard...

- Plans
 - WG Last Call beginning of October 2001
 - Completion in December 2001
- Prerequisites:
 - Stable spec (only minor changes from Proposed)
 - ≥ 2 interoperable implementations for each feature
 - We are not worried about this part
 - SIP MIB!



What else is done...?

Reminder: SIP is Multimedia

- 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.

MMUSIC WG: SDP

SDP (RFC 2327) being revised

- Bug fixes and clarifications
- Minor extensions / changes

Limited extensions being finalized

- Simple Capability Negotiation
 - Status: Passed WG Last Call, now for IESG
- Flow IDs
 - Status: Discussion in WG Last Call

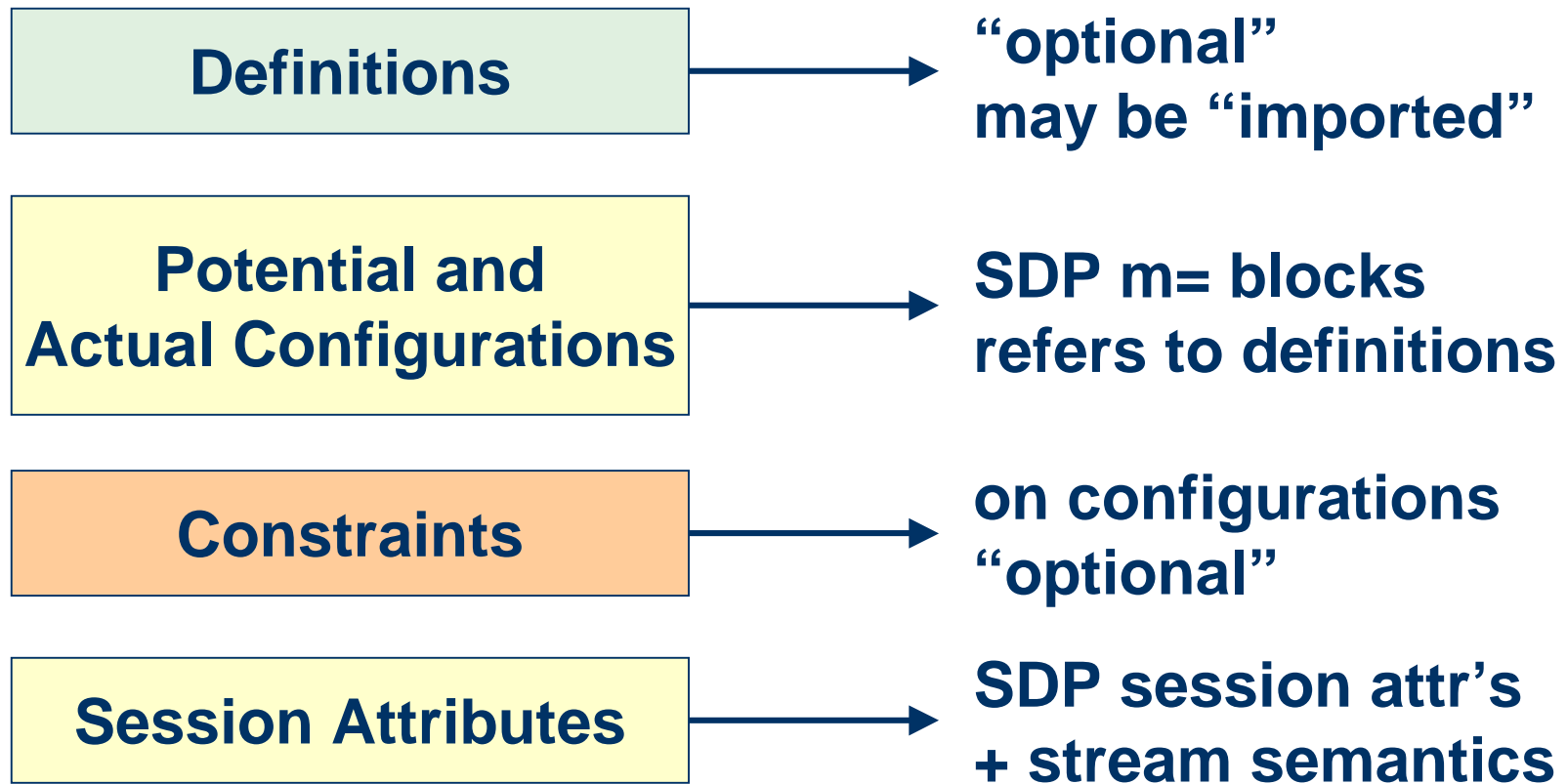
From SDP to SDPng

- SDP has enabled SIP + streaming applications
 - works fine for many cases
 - makes many implicit assumptions
- BUT: Designed for Session Announcements
 - rather than for interactive “negotiations”
 - has exceeded its limit
- Many recent extensions
 - to better support SIP, MEGACO in the short-term
 - General solution being worked out

SDP Next Generation (SDPng)

- Being designed to address SDP's flaws...
 - Limited expressiveness
 - For individual media and their combination
 - Often only very basic media descriptions available
 - No real negotiation functionality
 - Limited extensibility (clumsy, hard to coordinate)
 - No semantics for media sessions (only implicit)
- Also: Avoid second system syndrome!
 - Simple, easy to parse, extensible, limited scope

SDPng Structure



SDPng Status

- Requirements agreed upon in MMUSIC
 - Also input from SIP, MEGACO
- Basic structure agreed upon
- XML-based syntax chosen
- Strawman proposal available
- Draft spec expected for 51st IETF
- Next steps: definitions (media, transport, ...)

IPTTEL: CPL & TRIP

- Call Processing Language (CPL)
 - Done: RFC 2824
- Telephony Routing over IP (TRIP)
 - RFC 2871: Framework for Telephony Routing
 - TRIP Protocol: With IESG for Proposed Standard
 - “TRIP light” for Gateways

Finally: Keep SIP SIP!

- “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

Summary

- Interest in and use of SIP grows tremendously
- A lot of work done – and still a lot to do

- SIP: Core protocol and architecture
- SIPPING: Applications and their requirements
- MMUSIC: Session description

- Further groups are picking up on SIP
- BUT: Don't SIP everything!

Further Information

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

www.greycouncil.com/sipwg

www.greycouncil.com/sippingwg

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

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