SIP Interoperability Test Events (formerly known as SIP Bake-Offs)

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SIPit
SIP interoperability test event

Overview

- Goals
- What it is (and isn't)
- History
- Traditions
- Capability levels
- Preparing for an interoperability event
- The future

What is it?

- gathering of small engineering teams
- bringing products and prototypes
- from companies, research labs and universities
- test interoperability and robustness of SIP-related software and hardware

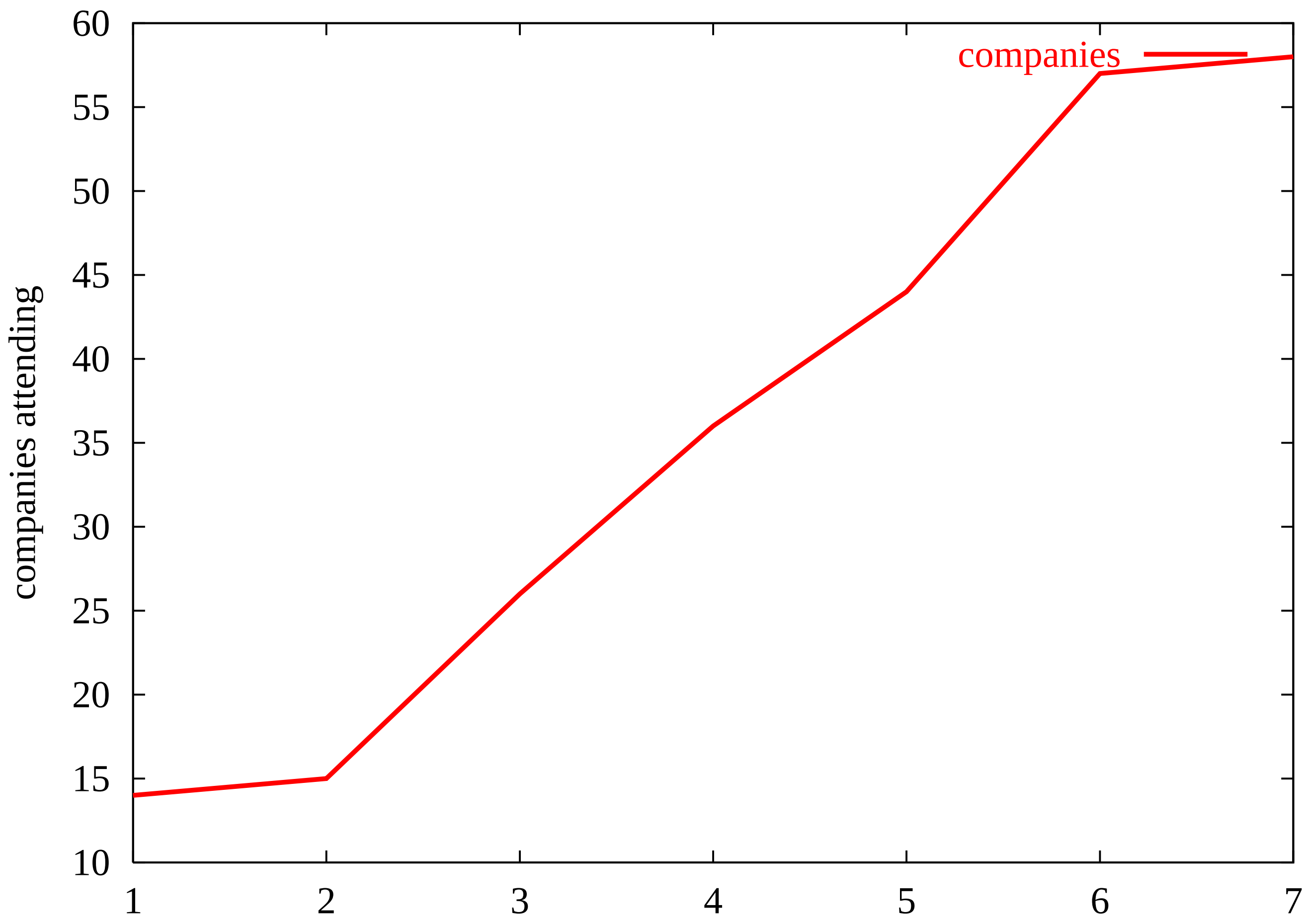
Goals

- diverse, interoperable, robust SIP implementations
- unambiguous base and extension specifications
- best current practices
- community development

What it is not. . .

- competitive – no winner
- recruiting event
- press conference
- demo
- benchmarking test
- VON, IEEE conference

History


companies attending
60
55
50
45
40
35
30
25
20
15
10
1
2
3
4
5
6
7
companies

Events

Every four months:

#	when	where/host	location
1	April 1999	Columbia University	New York, New York
2	August 1999	Pulver	Melville, New York
3	December 1999	Ericsson	Richardson, Texas
4	April 2000	3Com	Schaumburg (Chicago), Illinois
5	August 2000	Pulver	Melville, New York
6	December 2000	Sylantro/Sun	Santa Clara, California
7	March 2001	ETSI	Cannes, France
8	August 2001	Ubiquity	Cardiff, UK
9	December 2001	Nuera	San Diego, California

What can be (and has been) tested

Anything that speaks SIP:

- SIP phones, PC “soft clients”
- proxy servers
- test tools
- unified messaging servers
- conferencing servers
- RTP interoperability (mostly audio, some video)

What does the interoperability test event look like?

- lab ← conference room ← hotel ball room
- tables with Ethernet connectivity and power
- (SIP) phone or intercom system for communication
- separate rooms for "scenarios"

Traditions

- engineers and programmers, not laywers or marketing
- on-site bug fixing
- no written non-disclosure agreements or MoUs – mutual trust (and threat to be asked not to come again)
- detailed results remain confidential – no press releases touting individual achievements (“Our Acme SIP widgets were better than anybody else’s SIP widgets”)
- offered at or below cost, students free

Tests

- most tests are pair-wise, but also group scenarios
- scheduling semi-spontaneous, with scheduled scenarios
- based on capability self-assessment (basic, intermediate, advanced)

Typical test experience

First time (novice): make phone ring, avoid crashing on basic calls, test with relatively few (since lots of bugs to fix)

Second time (intermediate): test with more implementations, more features

Third time (veterans): participate in multi-group scenarios

Basic SIP capabilities for UAs

- send and receive INVITE over UDP
- generate ACK properly
- can accept or reject calls
- SDP with single m and c line, one codec
- To, From, Call-ID, CSeq, Via, Content-Length, Content-Type headers handled properly
- generate tags in To field
- send basic call termination with BYE via UDP
- receive BYE over UDP
- compact form for headers
- reject unknown request methods with 501 response
- send/receive RTP media, possibly without RTCP

Intermediate SIP capabilities for UAs

- support TCP for all messages
- Require, Proxy-Require
- handle packet loss for INVITE and BYE (with exponential backoff)
- pays attention to Contact header in INVITE and in 2xx response to INVITE (i.e., goes directly to peer for following requests)
- process CANCEL for INVITE
- Authentication for registrations: basic and digest
- allow redirection to web pages or email
- receive text or HTML in 3xx or 4xx responses
- Accept headers without SDP
- DNS SRV records
- register with periodic refresh to unicast address

Intermediate SIP capabilities for UAs

- understands redirection
- multiple codecs listed in SDP m line, finds common one with peer
- multiple SDP m= lines handled correctly
- unknown SDP m= media types handled correctly
- Domain name as well as IP address accepted in SDP c= line
- generate RTCP packets
- respond to OPTIONS request
- allows non-SIP URLs in REGISTER
- copy Record-Route from response into Route of request and route appropriately
- checks equality of action parameters on REGISTER
- can retrieve current registrations
- can clear registrations with Contact: * and Expires: 0

Advanced SIP capabilities for proxies

- forking proxies: sequential
- forking proxies with multiple 200 OK responses
- recursion on forking (fork response of 3xx triggers new branch)
- legal fork looping (with different request URIs)
- forking for non-INVITE
- digest and digest authentication for INVITE
- can detect loops
- can insert Record-Route
- drops request when Max-Forwards is zero
- obeys Expires in INVITE
- third party registration
- registration proxying
- process multicast REGISTER
- multicast INVITE
- received parameter in Via field
- can always redirect (configurable)
- IPsec support
- TLS support

Test scenarios

Typically require more than two participants:

- tel: URL translation
- multi-stage proxy
- media changes (re-INVITE
- multicast REGISTER
- loop detection

Preparing your implementation for a SIP test event

- liberal in what you receive, conservative in what you send
- pass SIP parser torture tests
- follow standard SIP call flows
- use one of the SIP phones
- use free/cheap SIP implementations
- on-line servers (Lucent, Nortel, Ubiquity, 3Com, Worldcom, Columbia University, . . .)

The future

- evolving event – balance between informality and structure
- “users of SIP”: 3GPP, CableForum specs, . . .
- presence and messaging
- services and features
- third-party call control
- CPL and sip-cgi
- interworking with H.323, soft switches

Summary

- cooperative effort of lots of vendors to improve reliability and interoperability
- help to make specifications clearer and less ambiguous
- create a community of SIP developers