Chairman's Summary of the First SIP Forum SIP Interoperability Workshop

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Summary

The first SIP Forum SIP Interoperability Workshop (SFSIW-1) was held on 3 December 2007 in Vancouver, BC, Canada. The venue was co-located with IETF 70. The SIP Forum worked with the RAI area directors to ensure the Workshop did not conflict with SIP-related sessions. We did this as many of the principals are involved with SIP, SIP applications, and SIP infrastructure. The authors, Eric Burger and Robert Sparks, were the General Chair and Workshop Chair, respectively.

The essential results were there are too many RFCs, making it hard to know what to implement. There is a combinational explosion of protocol options, making inter-implementation interoperability very hard to achieve. Picking a path to address these issues warrants a further workshop, as each comes with controversy and hard problems to solve. Paths include promoting SIP to Draft Standard, creating a basic SIP profile, creating a SIP certification program, or publishing a guide to SIP implementation that includes a roadmap of all of the SIP documents.

There were eight formal papers and three presentations accepted by the Workshop Program Committee. There were ten presentations made at the workshop. The Program Committee members were Robert Sparks, Anwar Siddiqui, Bob Penfield, and Eric Burger.

Paper Overviews

Interoperability Experiences

Markus Isomaki from Nokia presented on Interoperability Experiences[1]. They found that it is rare for anyone to select a random selection of proxies, registrars, SBCs and ALGs and get them to have all of their features interoperate is very low. Service providers end up writing their own detailed requirements, picking and choosing between the various SIP options, effectively creating new profiles of SIP. Many of these requirements are from obsolete drafts and are sometimes contrary to IETF recommendations, BCPs, and common knowledge. This also puts an undue burden on user agent vendors. User agent vendors have to make configuration profiles to meet these needs, many of which require proprietary extensions or interworking with proprietary implementations that are "almost SIP."

The paper identified areas needing examination. There are too many RFCs, which makes it hard to know what to read. Moreover, it is very easy to miss corrections. There are too many ways of doing the same thing. Worse, it is difficult if not impossible to implement all of the various combinations. The hardest problem is SDP offer-answer negotiation. There are many combinations of options that are literally too numerous to write code to cover them all. Likewise, there is no standard way for user agent configuration and outbound proxy discovery.

One particular concern is 3GPP is diverging from IETF on a number of issues, including registration logic and security. Moreover, 3GPP is creating SIP profiles without proper feature negotiation.

Some suggestions made to the SIP Forum for moving forward include publishing meaningful groupings of the many SIP RFCs, publish more BCPs on how to configure elements to make them work, providing or cultivating open source SIP reference implementations, and create a certification program for SIP compliance. A suggestion for the IETF is to revisit the offer-answer protocol and to significantly reduce the number of options available.

Profiles

Christer Holmberg from Ericsson described what profiling is and how the 3GPP has used profiling as an example of improving interoperability.[2] In particular, profiles create convention amongst User Agents that to offer a higher-level service built upon SIP, that they would use only a subset, and hopefully one-and-only-one procedure, from the set of SIP procedures for building that service. It does not change SIP or SIP feature negotiation. However, it does dramatically improve the higher-level feature interoperability than if implementations could use any of the combinations of SIP building blocks to create the feature.

Basic Call Flow Testing

Archana Rao and Henning Schulzrinne from Columbia University described the Columbia VoIP Testbed and some results from running the SIP Basic Call Flows Examples, SIP PSTN Call Flows, and the SDP Offer/Answer Examples RFCs against a suite of 5 servers and over 20 clients.[3] Their results show that even against published profiles, very few of the SIP components interoperated. They found that interoperability issues came from a lack of clarity in the specification; implementation of an older specification; incomplete implementation of the specification; incorrect implementation of the specification; and failure against robustness tests.

One idea proposed by the paper is to establish a designated SIP interoperability liaison at interested vendors. When one discovers an interoperability issue in a laboratory or at a test event such as SIPit, the normal customer service path for reporting issues structurally cannot handle the request.

SIP Interoperability Issues

Hadriel Kaplan from Acme Packets enumerated sources of SIP interoperability problems he found in deployment.[4] A principle source of interoperability problems is there are many parameters that require exact, correct, and compatible configuration. This problem affects proxies, application servers, and other middle-boxes, in addition to the expected client configuration problem of locating servers, certificates, and so on.

Another source of interoperability problems are attempts to coexist with legacy SIP infrastructure. This creates yet more configuration complexity. However, the market demands such interoperability.

Some response codes are ambiguous, such as 404 and 503. The error result itself may be unambiguous, but the cause may be. Different implementations take different corrective action, resulting in interoperability failures.

There are so many ways of exchanging user stimulus, particularly for DTMF, that non-interoperability is virtually guaranteed. This is in spite of there being one and only one IETF-approved and documented way of exchanging DTMF. Likewise, there are incompatible methods of firewall traversal. While the SIP Outbound work is almost complete, it is not done yet. Moreover, as with KPML for DTMF, it will take years for SIP Outbound deployment.

The paper presented an excellent overview of why there is confusion between the tel: URI scheme and the sip: URI scheme that indicates a E.164 number. This occurs due to ambiguities created from the limited, 12-key user interface, where the UAC has no idea if the number entered has local scope (e.g., voice VPN within an enterprise or service provider) or global scope (i.e., a real E.164 number). Moreover, many SIP UAS reject non-sip URI scheme URLs, making even proper use of the tel URI scheme not interoperate.

Not unexpectedly, with some deployment experience, we have discovered that some RFCs are either broken or result in configurations that are very difficult to operate. The example given in the paper is the Service-Route header.

SDP Offer/Answer Details

John Elwell from Siemens enumerated many of the SDP negotiation problems found in the field.[5] Specific examples included call hold, SDP-less re-INVITE, dynamic DTMF RTP (RFC 4733) payload types, video negotiation, unrecognized extensions, failure to recover if the other side does not support an extension, ptime issues, and codec changes.

SIPit Experiences

Alan Johnston et al. from Avaya described some issues they have come across at recent SIPit events.[6] Most of the errors were triggered by SDP offer/answer problems, particularly where there were many, correct ways of implementing a handshake, but the end points did not implement all of the ways of doing the handshake. They also found a problem with a pair of legacy emulation features, call transfer and line appearance number that gave erroneous user interface results.

IMTC Experiences

Anatoli Levine from RADvision, in his capacity as President of the International Multimedia Telecommunications Consortium (IMTC), described how the IMTC approaches interoperability testing.[7] In their experience the sheer number of variations allowed in SIP implementation drives many interoperability issues. Examples include message parsing issues such as case sensitivity; order of protocol elements, such as m-lines in SDP; DNS support; unexpected, but legal sequences of messages, such as multiple 183 responses; and implementers not implementing all of the specifications.

Comparison of H.323 and SIP for Video Conferencing

Norbert Oertel from Siemens described why most high-end video conferencing systems still use H.323, even though most support SIP.[8] The paper documented many situations where H.323 has one and only one method for implementing a feature, whereas SIP has many. For example, while there is one way for requesting an I-Frame in H.323, there are at least three ways of doing it with SIP. Unless the server implements all of the methods, there will be interoperability problems.

User Agent Experiences

R. Pathasarathi from Aricent described interoperability issues found from examining existing SIP deployments. The paper described to prominent models for User Agent construction. One model is the thin client, or telco model, where there is a server, such as an IP PBX that does most of the work on behalf of the client. The other model is the rich client, or Internet model, where the endpoint implements and executes the necessary service logic for the user.

The paper goes over a number of call flows demonstrating totally legal and non-interoperable ways of implementing different call services.

The major conclusion is that if a User Agent vendor wishes to gain a foothold in the market, they need to implement all of the models and call flows. This might have issues as a dual-mode thin/thick client is more complex than a thick client.

Suggestions

Workshop participants made the following suggestions. The appearance of these suggestions does not imply the SIP Forum or the IETF agrees with or will follow any of the suggestions. We present them here to continue the discussion.

Documentation

- Highlight that UAs must support INVITEs and re-INVITEs without offers.
- Create a number of small BCPs that highlight the right way to implement SIP features, and that interoperability requires the implementation of these features.
- Create a few, big BCPs that describe which features a UA must support for interoperability and the right way to implement those features.
- Publish a BCP indicating minimum field lengths a UA needs to accept in a SIP header.
- Create a (set of) profile(s) that guarantee interoperability.
- Create a "SIP HD Video" profile and certification program.
- Write test cases around basic, interoperable features.
- Identify reference endpoints that have full feature implementation and have them available for face-to-face (e.g., at SIPit) and remote testing.
- Encourage vendors to publish interoperability reports in a standard format

Protocol

- Dramatically reduce number of options for SDP offer/answer.

- Create a protocol for User Agent configuration.
- Drop RFC 2543 compatibility.
- Drop UDP support.
- Fix G.722 codec sampling rate error in RFC 3551.

Other

- Expand certification programs – possibly certify the basic call flows.
- Establish a designated interoperability liaison at each vendor.
- Provide self-certification tests – possibly a "SIP Good Citizen" certification mark.
- Provide remote access for development testing and self-certification testing.

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[1] Isomaki, M., SIP Interoperability Experiences and Recommendations, http://www.sipforum.org/component/option,com_docman/task,doc_download/gid,121/Itemid,75/

[2] Holmberg, C., Profiling, http://www.sipforum.org/component/option,com_docman/task,doc_download/gid,133/Itemid,75/

[3] Rao, A. and Schulzrinne, H., Real-world SIP Interoperability: Still an Elusive Quest, http://www.sipforum.org/component/option,com_docman/task,doc_download/gid,124/Itemid,75/

[4] Kaplan, H., A Brief List of SIP Interoperability Issues, http://www.sipforum.org/component/option,com_docman/task,doc_download/gid,122/Itemid,75/

[5] Elwell, J., Common Interoperability Problems with SIP, http://www.sipforum.org/component/option,com_docman/task,doc_download/gid,119/Itemid,75/

[6] Johnston, A. et al., Recent SIP Interoperability Experience at SIPit, http://www.sipforum.org/component/option,com_docman/task,doc_download/gid,117/Itemid,75/

[7] Levine, A., IMTC Approach to Interoperability Testing, http://www.sipforum.org/component/option,com_docman/task,doc_download/gid,120/Itemid,75/

[8] Oertel, N., Comparison of H.323 and SIP Video-Conferencing Support, http://www.sipforum.org/component/option,com_docman/task,doc_download/gid,123/Itemid,75/