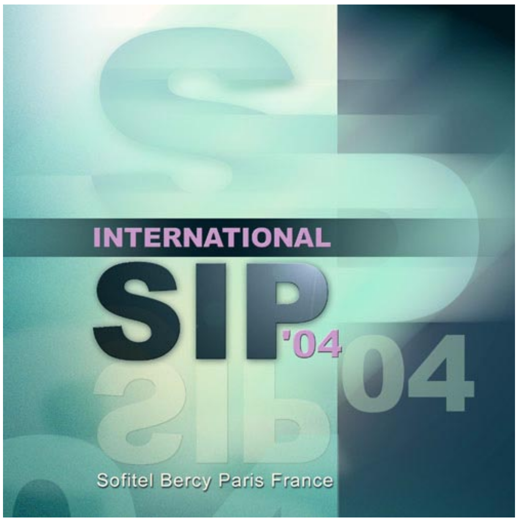
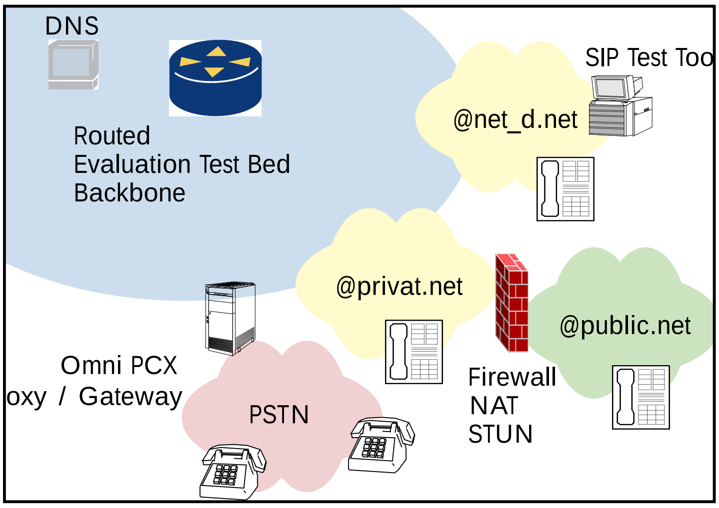


PUBLIC SIP INTEROPERABILITY EVENT

DURING »INTERNATIONAL SIP 2004«



TEST PLAN AND RESULTS

Organized by: EANTC

In cooperation with: multiservice switching forum, SIP FORUM

Hosted by Upperside
www.upperside.fr

Introduction

This interoperability event is organized by the European Advanced Networking Test Center (EANTC), in cooperation with the Multiservice Switching Forum (MSF) and the SIP Forum, and hosted by Upper Side.

A multi-vendor SIP (Session Initiation Protocol) telephony infrastructure with components of different SIP equipment manufacturers, complemented by emulators was used to show scalable interworking and reliability between legacy and next generation infrastructures. The show case scenarios evaluated different aspects of global SIP networks: infrastructure and scalability, services and applications as well as robustness and security. In addition, new interoperability test methodologies were verified and improved.

To ensure the success of the event, a 4 day private hot-staging event with all the participating vendors was conducted before the International SIP 2004 Conference. It took place at the EANTC (European Advanced Networking Test Center) in Berlin, Germany. The test plan was defined by EANTC in cooperation with the MSF, the SIP Forum and participating companies.

Participants and Devices

The following companies and devices demonstrated their interoperability in the test event:

Participant	Devices
Codenomicon Ltd.	• SIP Test Tool
Hotsip AB	• Active Contacts PC • SIP Application Server / Presence Engine
Navtel Communications Inc.	• InterWatch 95000 and SIP Test Suites
snom technology AG	• snom 200 / 105 phones • snom SIP 4S Proxy / Registrar Server
Spirent Communications	• Abacus 5000

Further equipment used to complement the evaluation test bed	
Company	**Devices**
Ahead Software AG	• SIPPS Softphone
Xten	• X-Lite Softphone
SJ Labs	• SJphone Softphone
Alcatel	• Alcatel Onni PCX, different Phones (analog, digital)

Showcase Goals

This interoperability showcase focused scalability and robustness of SIP infrastructure and services in a multi-vendor environment.

In contrast to private interoperability test activities carried out under non-disclosure agreements, this event publicly demonstrated the results at the International SIP 2004 conference — allowing visitors to get an insight picture of the interoperability status of current SIP implementations.

Two industry forums, the Multiservice Switching Forum (MSF) and the SIP Forum, ensured that the showcase demonstrated test areas relevant to carriers: Showcase scenarios were derived from two MSF SIP Implementation Agreements (IAs) addressing the SIP-T profile for media gateway controllers and the SIP profile for VoIP (MSF-IA-SIP-T.001 and MSF-IA-SIP.001). Participants also showed interoperability of their equipment with regard to a list of test scenarios created by the SIP Forum Certification Workgroup.

	Infrastructure and Scalability Tests	Services and Applications Tests	Security and Robustness Tests
Codenomicon Ltd.	●		●
Hotsip AB	●	●	●
Navtel Communications Inc.	●	●	●
snom technology AG	●	●	●
Spirent Communications	●	●	●

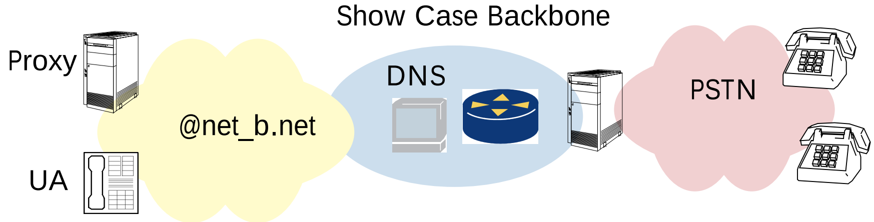
Test Areas And Participating Vendors

Infrastructure and Scalability Tests

SIP-to-SIP Calls: Direct, One Proxy and Proxy Chaining. This test group aimed at determining a basic level of interoperability between the participating vendors according to the MSF Implementation Agreement [MSF-IA], chapter 8.1. We focused the evaluation of multi-vendor scenarios instead of showing that these scenarios worked with the equipment of only one vendor. The basic setup verified interoperability for direct user agent calls. It applied to Codenomicon and all five user agents as well as to Navtel calling Spirent and vice versa. Later on, more complex scenarios involved call setups via one proxy server as well as via a chain of two proxies using DNS services.

Calls via a Firewall with NAT and STUN. This test area aimed to evaluate interoperability issues with call setups via a firewall using full cone NAT. As a solution to overcome the restrictions of the firewall, STUN (»Simple Traversal of UDP Through Network Address Translators«, see [IETF-STUN]) was chosen during the test plan design as the most widely deployed tool.

SIP-to-PSTN Calls. Emulating a very commonly used SIP scenario, this test group verified call setups from SIP user agents via a SIP proxy and gateway to standard PSTN phones.



Topology for the SIP-to-PSTN Call Setup Evaluation

Services and Applications Tests

Call Transfer and Call Forwarding. This group verified interoperability of unattended call transfer and unconditional call forwarding according to the IETF's Session Initiation Proposal Investigation, SIPPING, working group document [IETF-SE].

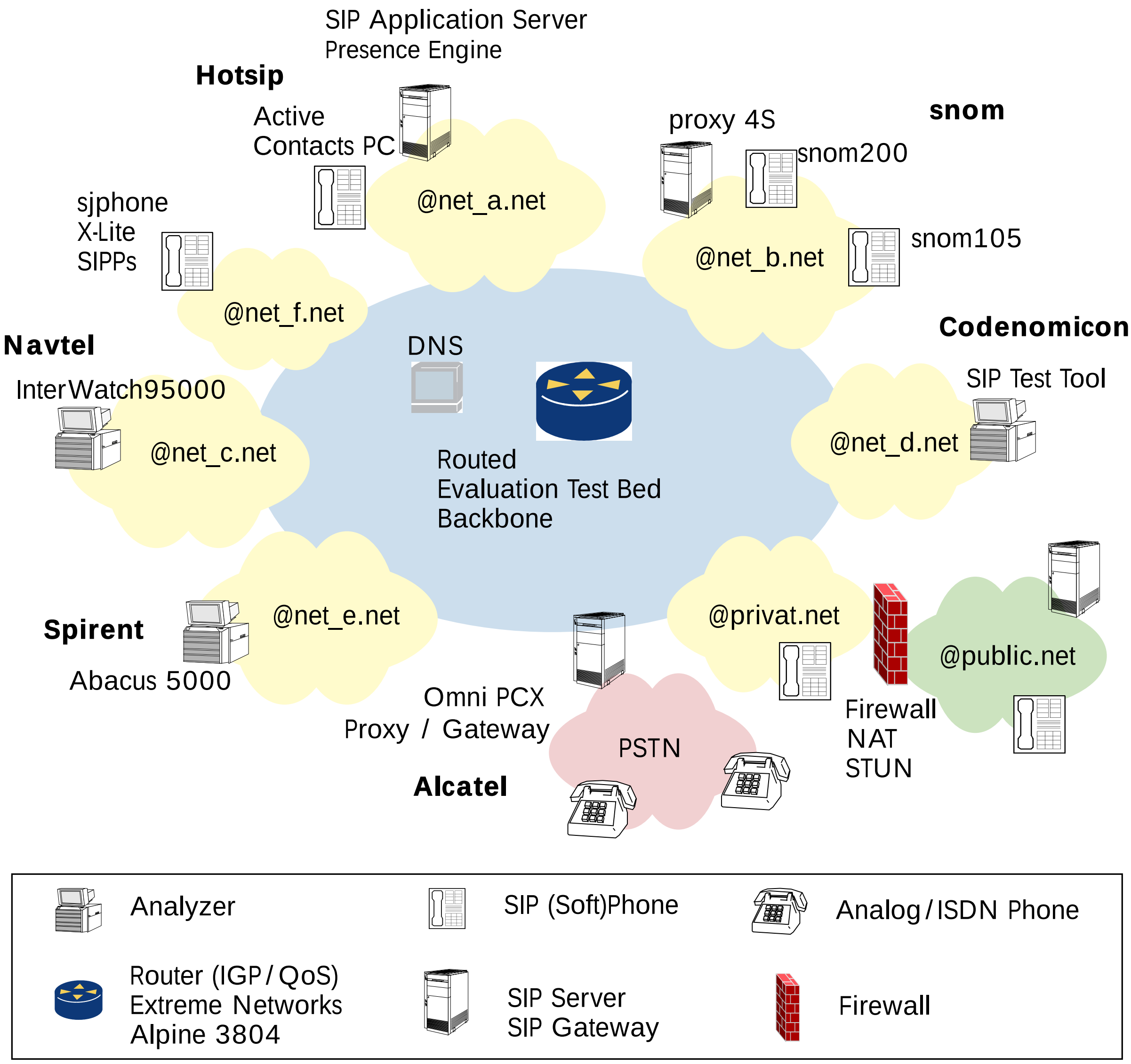
Codec Negotiation. This test evaluated the call flow in chapter 8.12 of the MSF Implementation Agreement [MSF-IA] where a user agent re-INVITEs with a different Media CODEC if the initial INVITE fails.

Security and Robustness Tests

Proxy Authentication. In this test case, we verified the general interoperability status of SIP clients and SIP server for a RFC 3261 based new registration using proxy authentication using a MD5 key, see 'Successful New Registration' in Session Initiation Protocol (SIP) Basic Call Flow Examples, RFC 3665.

Robustness Tests. Conducted with the Codenomicon SIP Test Tool, these tests identified security and reliability issues within SIP implementations by robustness testing the SIP nodes with a large number of torture messages.

Evaluation Test Bed



Results

In this section, we summarize the results and observations collected during the hotstaging for the public showcase. In general, scalability and robustness goals in the multi-vendor SIP environment were reached. Some interoperability issues and future test areas were identified.

Results: Infrastructure and Scalability Tests

SIP-to-SIP Calls: Direct, One Proxy and Proxy Chaining. The call generators successfully showed that the proxies of all participating vendors were able to handle traffic patterns of small to medium size enterprises. Between 500 and 1000 clients were simulated setting up calls with up to 20 calls per second without observing any error conditions. As the tests were conducted with laptop based systems, we would expect significant higher figures with a distributed system targeting large scale service provider deployments. Basic call functionality tests showed different results depending on the test combination, ranging from exact matches with the reference call flow up to crashed user agents. Tests involving user agents from the same manufacturer worked fine and independent of the signalling path, using either the vendor's own proxy, a third party proxy or even a proxy chain.

Calls via a Firewall with NAT and STUN. We discovered interoperability issues of an open source STUN server solution with participating vendors. Only after we switched to a STUN server solution of one vendor, a successful handshake between the STUN client and server was possible, followed by a basic SIP call setup. To improve interoperability in this area, explicit interoperability scenarios need to be designed, combining various STUN and SIP solutions.

SIP-to-PSTN Calls. The tests demonstrated robust and interoperable PSTN services between the multi-vendor evaluation SIP network and PSTN phones. The PSTN phones were connected to a local SIP to PSTN gateway. Test calls were set up between different SIP domains and a variation of PSTN phones. The integration of real PSTN carrier lines was identified as a future test group. Test methodologies in this area should be expanded to include speech quality measurements under different impairment conditions as well as data plane latency tests both with different codec configurations.

Results: Services and Applications Tests

Call Transfer and Call Forwarding. We proved interoperability of unattended call transfer and unconditional call forwarding for all participants. For a further evaluation, scalability and stress tests should be added to this test group to demonstrate implementation robustness e.g. for applications in the area of hotline or call center services.

Codec Negotiation. The verified implementations support codec negotiation in a different way than specified with the MSF IA call flow example, stating that the UAC sends INVITEs asking for one specific codec and reissues INVITEs if the codec was not supported. Instead the UAC tested for this event send a list of supported codecs with the INVITE message letting the UAS choose a supported codec. We showed with our tests that if the UAS supports more codecs and the UAC offers a list of codecs, the UAS will choose the first codec in the list.

Results: Security and Robustness Tests

Proxy Authentication. The tests showed the expected results according to RFC 3665 in all combinations: user agent to proxy as well as call generators or analyzer to proxy. Test scenarios with failed authentications were identified as a future evaluation option, as it was reported very recently, that this may lead to infinite authentication loops with some user agent implementations. Such a user agent with an incorrect user and password configuration would keep on sending new registration request after receiving a negative response from the server.

Robustness Tests. The event proved that the robustness tests were effective in testing both user agents and other pieces of the infrastructure individually, and the infrastructure as a whole. We noted that robustness testing is most effective when tested directly against the tested implementation. No intermediate devices should exist in the signalling path between tested implementation and the tester, since they may modify the requests and thus effect the test results.

The free PROTOS robustness test suite for SIP, released in early 2002, has given the vendors a good opportunity to upgrade their quality in voice over IP products. Robustness tests during the hotstaging ‘showed that the products of those software developers that have worked with the PROTOS tests and who have used robustness testing in the past are much more tolerant of exceptional inputs, and are thus quite secure against intrusion attempts. Even though these products crashed only in a few test cases, the torture messages used for this test were still effective in finding flaws. Products that had never been tested for reliability failed in almost all tests.

Results Summary

Key Features Tested		Results
Infrastructure and Scalability	SIP-to-SIP Calls: Direct, One Proxy and Proxy Chaining	expected results for most user agents, some interoperability issues, scalable for small to medium size environments
	Calls via a Firewall with NAT and STUN	was demonstrated a for single vendor solution, problems with public STUN server
	SIP-to-PSTN Calls	worked fine for all participants
Services and Applications	Call Transfer and Call Forwarding	worked fine for all participants
	Codec Negotiation	expected results when operating with a codec list and prioritizing scheme, different approach than in MSF IA
Security and Robustness	Proxy Authentication	worked fine for all participants
	Robustness Tests	different test results for participants, vendors who did torture tests in the past showed very stable system behavior, other vendors failed

Problem Summary

Problem Area	Description of the Problem	Temporary Resolution, if any	Recommendation
interoperability of user agents	no successful call setups for dedicated UA combination	vendor software patches	–
interoperability with STUN service	no successful STUN address binding	–	improve / design interoperability test scenarios or call flow examples within IETF or MSF
failure during torture tests	different severity levels up to system crash	vendor software patches	establish basic robustness with publicly available test tools, continue work on IETF torture test draft

Conclusion

This event was an important step to advance SIP multi-vendor interoperability demonstrations to the next level. Two industry fora, an independent test lab and a conference host collaborated to organize a showcase with advanced, real-life scenarios focusing carrier requirements. The benefits of a public event were combined with detailed private hotstaging scenarios prior to the public demonstration.

The showcase proved that SIP is a mature technology in many areas and is ready for deployment in a larger scale. As we have seen, the wealth of options in SIP standards, ongoing standardization work and vaguely defined operator requirements may lead to interoperability issues. Manufacturers should regularly verify interoperability and security issues. Future interoperability events will need to continue evaluate scalability issues, especially when carrier grade equipment is involved.

Both industry forums involved have proactively created implementation agreements for carrier environments (MSF) and manufacturer-oriented test suites (SIP Forum). The hotstaging showed that such efforts are reasonable and necessary to improve multi-vendor interoperability. While most of the deployed SIP networks are homogeneous right now — and thus not affected by interoperability issues —, the power of SIP technology in service provider environments will increase with the number of vendors committed to interoperability of common feature sets.

References

[MSF-IA] MSF-IA-SIP.001-FINAL, Implementation Agreement for SIP Profile, for Voice over IP, between a line-side Media Gateway Controller and a Trunks Media Gateway Controller.

[IETF-SE] draft-ietf-sipping-service-examples-05.txt

[IETF-STUN] STUN - Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs), RFC 3489

[IETF-CF] Session Initiation Protocol (SIP) Basic Call Flow Examples, RFC 3665

[IETF-TT] draft-ietf-sipping-torture-tests-02.txt

[PROT-07] PROTOS Test-Suite: c07-sip

[SIPVAL] http://developer.berlios.de/projects/sipvalidator/

EANTC AG

Einsteinufer 17
10587 Berlin
Germany

Tel: +49 30 3180595-0
Fax: +49 30 3180595-10

www.eantc.com
info@eantc.com

Multiservice Switching Forum

39355 California Street
Suite #307
Fremont, CA 94538

Phone: +1.510-608-5922
Fax: +1.510-608-5917

www.msforum.org
info@msforum.org

SIP Forum

c/o Matrisen AB
Box 22059
104 22 Stockholm

www.sipforum.org

www.upperside.fr

Upperside

54, rue du Fbg
Saint-Antoine
75012 Paris, France

Tel: +33.153.46.63.80
Fax: +33.153.46.63.85

www.upperside.fr
info@upperside.fr