

Welcome to SIPp

by Richard GAYRAUD [Initial code], Olivier JACQUES [code/documentation], Many contributors

SIPp is a free Open Source test tool / traffic generator for the SIP protocol. It includes a few basic [SipStone](http://www.sipstone.org) (http://www.sipstone.org) user agent scenarios (UAC and UAS) and establishes and releases multiple calls with the INVITE and BYE methods. It can also read [custom XML](#) (doc/uac.xml.html) scenario files describing from very simple to [complex](#) (doc/reference.html#3PCC) call flows. It features the [dynamic display](#) (doc/reference.html#stat_screen) of statistics about running tests (call rate, round trip delay, and message statistics), periodic CSV [statistics](#) (doc/reference.html#Statistics) dumps, TCP and UDP over multiple sockets or multiplexed with retransmission management and [dynamically adjustable](#) (doc/reference.html#traffic_control) call rates.

Other advanced features include support of [IPv6](#) (doc/reference.html#ipv6) , [TLS](#) (doc/reference.html#tls) , SIP [authentication](#) (doc/reference.html#authentication) , [conditional scenarios](#) (doc/reference.html#branching) , UDP retransmissions, [error robustness](#) (doc/reference.html#Error+handling) (call timeout, protocol defense), call specific variable, Posix [regular expression](#) (doc/reference.html#action_regexp) to extract and re-inject any protocol fields, [custom actions](#) (doc/reference.html#actions) (log, system command exec, call stop) on message receive, field injection from [external CSV](#) (doc/reference.html#infile) file to emulate live users.

SIPp can also send media (RTP) traffic through [RTP echo](#) (doc/reference.html#RTP+echo) and [RTP / pcap](#) (doc/reference.html#PCAP+Play) replay. Media can be audio or audio+video.

While optimized for traffic, stress and performance testing, SIPp can be used to run one single call and exit, providing a [passed/failed](#) (doc/reference.html#Exit+codes) verdict.

Last, but not least, SIPp has a [comprehensive documentation](#) (doc/reference.html) available both in HTML and PDF format.

SIPp can be used to test many real SIP equipments like SIP proxies, B2BUAs, SIP media servers, SIP/x gateways, SIP PBX, ... It is also very useful to emulate thousands of user agents calling your SIP system.

Here is a screenshot:

```

ocadmin@vista:~/sipp
----- Scenario Screen ----- [1-4]: Change Screen --
Call-rate(length)  Port  Total-time  Total-calls  Remote-host
    10 cps(0 ms)   5061      4.01 s       40  127.0.0.1:5060(UDP)

10 new calls during 1.000 s period      16 ms scheduler resolution
0 concurrent calls (limit 30)           Peak was 1 calls, after 0 s
0 out-of-call msg (discarded)
1 open sockets

          Messages  Retrans  Timeout  Unexpected-Msg
INVITE  ----->      40      0        0
    100 <-----      0      0
    180 <-----      40      0
    200 <----- E-RTD  40      0
    ACK  ----->      40      0
          [    0 ms]
    BYE  ----->      40      0        0
    200 <-----      40      0

----- [+|-|*|/]: Adjust rate ---- [q]: Soft exit ---- [p]: Pause traffic -----

```

And here is a video of SIPp in action (Windows Media Player 9 codec or above required):

[sipp-01.wmv](#) (images/sipp-01.wmv)

Want to know more? Please jump to the [documentation section](#) (doc/index.html) .