

Welcome to SIPp

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

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

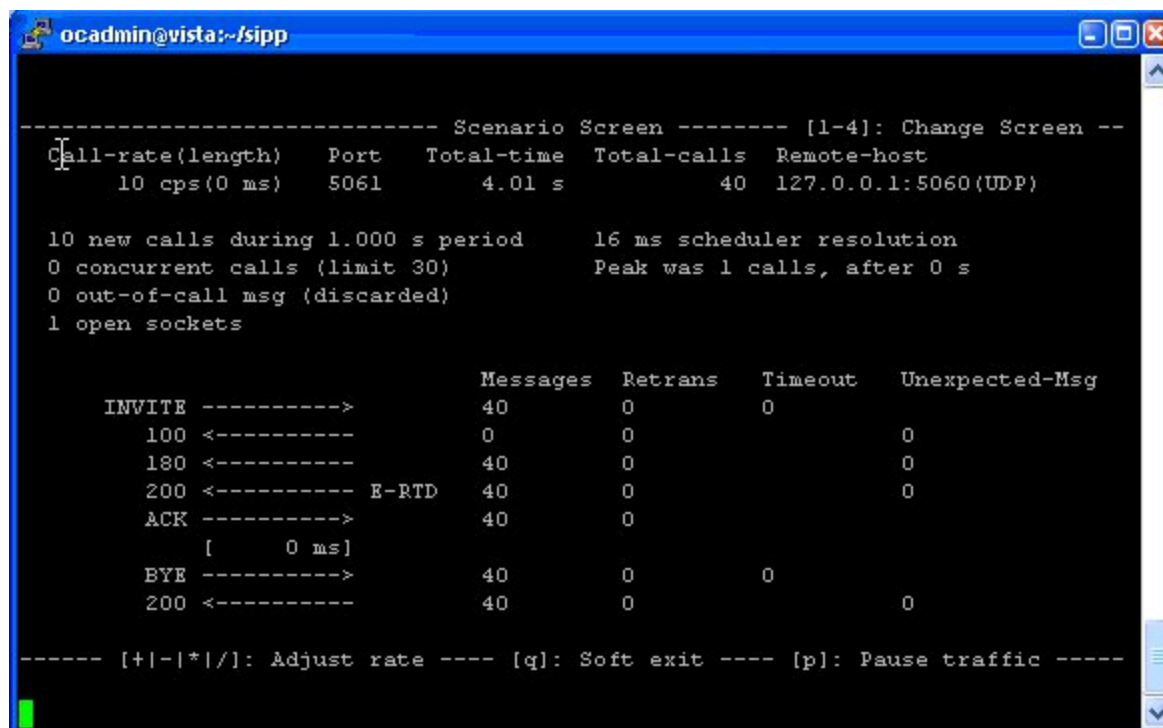
SIPp can also send media (RTP) traffic through [RTP echo](#) and [RTP / pcap](#) replay. Media can be audio or video.

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

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

SIPp can be used to test various real SIP equipment 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:



```

----- 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         0
 180  <-----      40        0         0
 200  <----- E-RTD  40        0         0
ACK     ----->      40        0
      [ 0 ms]
BYE     ----->      40        0         0
 200  <-----      40        0         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](#)

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

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