Welcome to SIPp

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SIPp is a free Open Source test tool / traffic generator for the SIP protocol. It includes a few basic SipStone (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 reads 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) , SCTP, 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#inffile) file to emulate live users.

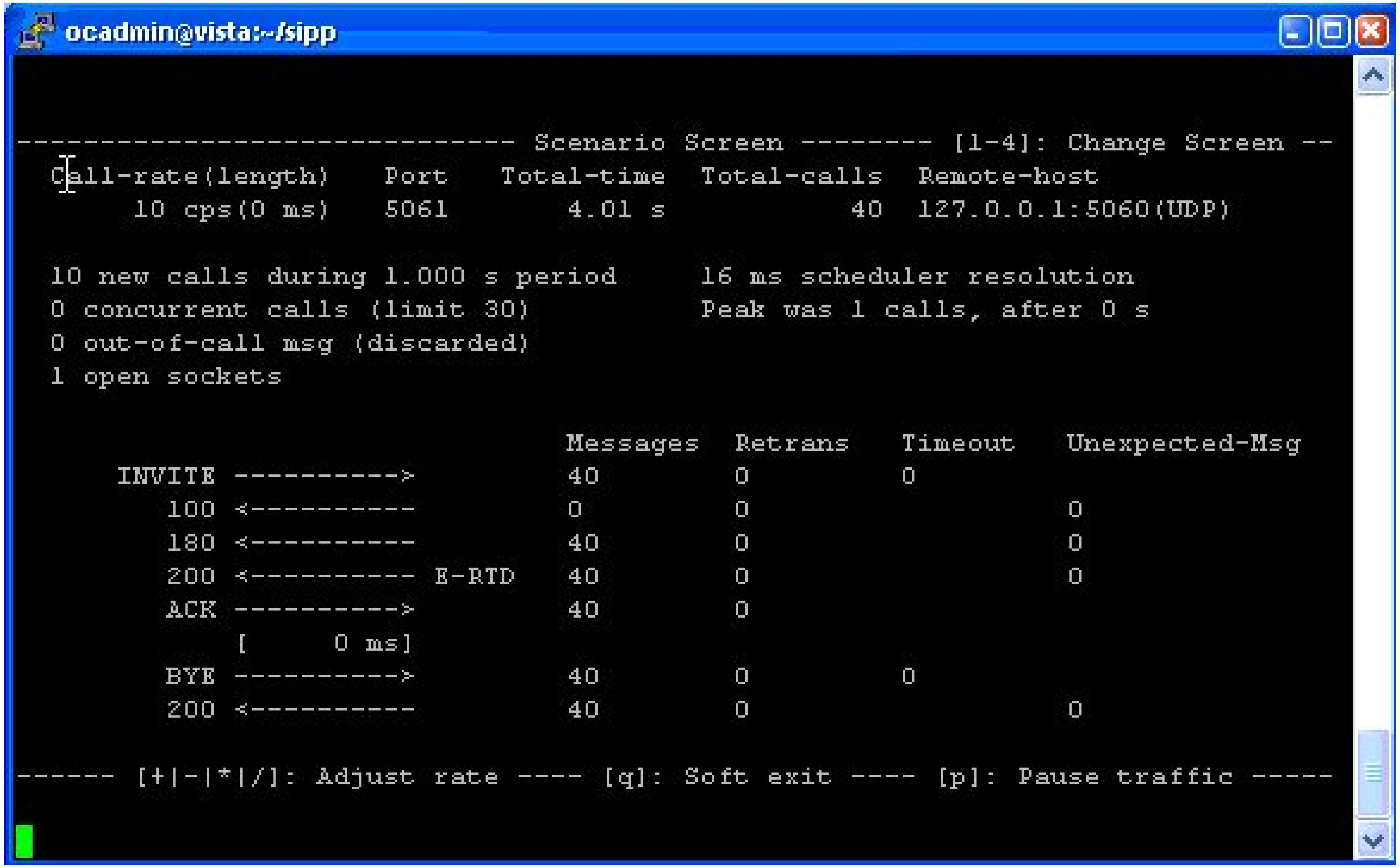
SIPp can also send media (RTP) traffic through RTP echo (doc/reference.html#RTP+echo) and RTP / pcap (doc/reference.html#PCAP+Play) replay. Media can be audio or video.

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 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:



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) .