



AN AUTOMATED DIALING SOLUTION TO SIMULATE CALLS TO AVAYA MIDMARKET ARCHITECTURE

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19/06/2017

THE CHALLENGE

- ▶ Challenge 1: Providing meaningful data in our demo systems for customer and business partner demonstrations, particularly IPOCC.
- ▶ Challenge 2: Replication of reported issues that are suspected to occur at a particular threshold of traffic or after a period of time, such as buffer overflow or out of memory.
- ▶ Replication has generally involved making manual test calls to the demonstration/ test system.
- ▶ This is time consuming and labour intensive.

A SOLUTION

- ▶ Star Trinity have a free SIP tester that can automate inbound traffic.
- ▶ The product registers to the IP Office as a SIP UAC (SIP Endpoint) and initiates internal calls, for example to IPOCC topics.
- ▶ The product consumes 1x IP Office Third Party IP Endpoint License.
- ▶ The unlicensed version allows 150 calls before the software must be restarted.
- ▶ Automated calls can disconnect at random intervals.
- ▶ The product can play a .wav file to simulate speech for Media manager.
- ▶ The product can be downloaded here:
<http://startrinity.com/VoIP/SipTester/SipTester.aspx>

WHAT IT LOOKS LIKE

► Making automated calls:

StarTrinity SIP Tester

Simulation: Registration (UAC) ? Reg. (UAS) **Outgoing calls simulation ?** Incoming calls handling ? Manual tests Impairments generation Stepwise testing ?

Create calls on timer: ? **Start** **Stop** **Create single call** with fixed interval between calls interval: 368.31 ms CPS: max = 2.72 actual = 0.07

☐ start timer on schedule ☐ stop timer on schedule Limit number of concurrent calls (incoming+outgoing): 1

☒ Limit total number of attempted calls 150 attempted calls: 2 / 150 [Reset counters](#) **Pause** Burst mode: **create** 1 call(s) per burst

Make random calls without registration to list of destinations [Add new destination](#)

Destination Number	Destination Host	Port	Transport	Auth. User	Password	Limit number of calls	Proxy Host	Port
7001	mattova.sip	5060	UDP	870		<input checked="" type="checkbox"/> 1 Delete	mattova.sip	5060

☒ Send SDP in INVITE ? forced codec: [allow G.711, G.729, G.723] custom SDP attributes:

☒ Terminate call if not answered within ☐ random ☒ fixed interval: 10,000 ms

☒ Terminate call after answering within ☒ random ☐ fixed interval: 10,876 ms Play RTP audio from file: speech.wav repeat count: 10000

☐ Record mix of RX and TX audio streams [Show folder with recordings](#) ☐ Record RX audio streams [Show folder with RX recordings](#) [customize script](#) [send fax on answer](#) [simulate DTMF events](#)

GUI XML XML (visual) Changes are saved and applied automatically drag to move the splitter

Reports: Current calls - SIP info ? Current calls - RTP info ? **Calls history (CDR) ?** Lowest quality calls ? Reports/Statistics ? Performance chart ? Log ? Stepwise testing

Keep 2000 calls in memory [clear memory \(2 calls\)](#) Display 35 latest calls from memory ☒ Save CDRs to CSV files fields delimiter: , [Show in folder](#) [Delete files](#) [Load from file\(s\)](#) ☐ Save CDRs to database

Filter: OK fields.. ☐ Auto-scroll to end [select columns](#)

Created ?	Answered ?	Destroyed ?	Direction ?/CallerId ?/CalledId ?	Disconnection status ?	Released by ?	SIP Call-ID ?	CallerIP ?	CalledIP ?	Answer delay ?	100 delay ?	180 delay ?
17-06-19 15:52:12.053	15:52:12.175		out/870/7001	trc man pcap pcap+RTP		3350a0f6676b41159ad26292e5709446	135.27.70.97	mattova.sip	122.00ms	25.00ms	
17-06-19 15:52:02.215	15:52:05.496	15:52:12.012	out/870/7001	200 OK trc man pcap pcap+RTP	caller	054e35c2d2fe42958fc20fc3fbb8168c	135.27.70.97	mattova.sip	3,281.00ms	26.00ms	953.00ms

Current calls: 1 **Abort All** Δ: 2 R: 0 AD: 3281.00/3281.00/3281.00 I: 0.34/0.34/0.34 CPU: 26% Version: 2017-06-01 14:25 UTC update Licensed to: [no license key] license information Help/support Settings Web interface misc.

LICENSING

- ▶ An unlimited license can be purchased.
- ▶ Prices shown are in US. Most local currencies are available.

License type

- ☒ 1 server, 10 concurrent calls, unlimited attempted and received calls, 3 calls per second \$100.00
- ☐ 2 servers, 10 concurrent calls, unlimited attempted and received calls, 3 calls per second \$165.00
- ☐ 1 server, 50 concurrent calls, unlimited attempted and received calls, 15 calls per second \$200.00
- ☐ 1 server, 100 concurrent calls, unlimited attempted and received calls, 30 calls per second \$350.00
- ☐ 1 server, 200 concurrent calls, unlimited attempted and received calls, 60 calls per second \$420.00
- ☐ 1 server, 400 concurrent calls, unlimited attempted and received calls, 120 calls per second \$560.00
- ☐ 1 server, unlimited calls \$700.00
- ☐ [extended support +1 year] \$600.00
- ☐ development of new feature for StarTrinity SIP Tester \$1,000.00
- ☐ unlimited servers, 50 concurrent calls, unlimited attempted and received calls \$3,000.00
- ☐ unlimited servers, 100 concurrent calls, unlimited attempted and received calls \$3,900.00
- ☐ unlimited servers, 200 concurrent calls, unlimited attempted and received calls \$4,900.00
- ☐ unlimited servers, unlimited calls \$19,000.00

AVAYA