Modern Performance Testing with Open-Source Tools

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- Are you making VoIP software that needs to run at high loads?
- Are you running a VoIP service that might suffer bursts of traffic?
- Are you concerned about quality of service call setup times and media jitter?

What this talk covers

- SIP performance testing with SIPp
- Diameter/MEGACO performance testing with Seagull
- Automation and integration into test suites
- JSIPp my recent attempt to fix some of the problems with SIPp

What is this talk not about?

- Functional testing
- IMS performance benchmarking (ETSI TS 186 008)

SIPp

- Describe your call scenario in XML
- Run over 5,000 calls a second (18M calls/hour) per core
- Get success rates, response times, failure rates back out

Sample SIPp scenario

Activi	es 🕞 Emacs 🗸 Sat 29 Mar, 13:15:54 📭 📢 🗋 🕇
	uac.xml - emacs@localhost.localdomain *
21	
22	scenario name="Basic Sipstone UAC">
23	In client mode (sipp placing calls), the Call-ID MUST be
24	generated by sipp. To do so, use [call_id] keyword
25	<send retrans="500"></send>
26	<![CDATA[</td>
27	
28	INVITE sip:[service]@[remote_ip]:[remote_port] SIP/2.0
29	Via: SIP/2.0/[transport] [local_ip]:[local_port];branch=[branch]
30	From: sipp <sip:sipp@[locat_ip]:[locat_port]>;tag=[pid]sippragoo[catt_number]</sip:sipp@[locat_ip]:[locat_port]>
32	Call_TD+ [call_id]
33	CSea: 1 INVITE
34	Contact: sip:sipp@[local ip]:[local port]
35	Max-Forwards: 70
36	Subject: Performance Test
37	Content-Type: application/sdp
38	Content-Length: [len]
39	
40	v=0
41	o=user1 53655765 2353687637 IN IP[local_ip_type] [local_ip]
42	
43	c=IN IP[media_ip_type] [media_ip]
44	t=0 0
45	
40	
48	
49	
50	
51	<recv <="" response="100" td=""></recv>
52	optional="true">
53	
54	
55	<pre><recv optional="true" response="180"></recv></pre>
56	
57	
58	<pre>/<recv optional="true" response="183"></recv></pre>
1:	uac.xml 31% L40 (nxmL valid ws yas Fill)

Sample SIPp command-line output

Call-rate(ler 5000.0(0 ms)/1	So ngth) Port Tota .000s 5061	cenario So al-time 1 10.01 s	creen Total-calls 59188	[1-9]: Remote-ho 127.0.0.1	Change Screen ost 1:5060(UDP)	
5947 new calls during 1.001 s period 0 ms scheduler resolution 24 calls (limit 18000) Peak was 150 calls, after 7 s 0 Running, 59174 Paused, 6429 Woken up 15 dead call msg (discarded) 0 out-of-call msg (discarded)					tion fter 7 s scarded)	
3 open socket	ts					
		Messages	Retrans	Timeout	Unexpected-Msg	
INVITE	>	59176	65	Θ	1 0	
100 <-		Θ	Θ	Θ	0	
180 <-		59168	Θ	Θ	0	
183 <-		Θ	Θ	Θ	0	
200 <-	E-RTD1	59168	Θ	Θ	0	
ACK	>	59168	Θ			
Pause [0ms]	59168			0	
BYE	>	59164	83	Θ		
200 <-		59164	Θ	Θ	Θ	
[+ - * /]: Adjust rate [q]: Soft exit [p]: Pause traffic Last Error: Dead call 44798-12562@127.0.0.1 (successful), re <u>ceived 'SIP/</u>						

Sample SIPp log files

ecrease Indent

C	D	E	F	G	Н	
	ElapsedTime(P)	ElapsedTime(C)	TargetRate	CallRate(P)	CallRate(C)	Inc
32791396098517.543279	00:00:00	00:00:00	6000	0	0	
48881396098518.544888	00:00:01	00:00:01	6000	5923.08	5917.17	
54531396098519.546453	00:00:01	00:00:02	6000	5911.09	5911.18	
34321396098520.548432	00:00:01	00:00:03	6000	5865.13	5893.88	
95911396098521.549591	00:00:01	00:00:04	6000	5932	5901.92	
99341396098522.549934	00:00:01	00:00:05	6000	5883	5898.14	
12461396098523.551246	00:00:01	00:00:06	6000	5906.09	5898.49	
36841396098524.553684	00:00:01	00:00:07	6000	5925.15	5902.3	
48771396098525.554877	00:00:01	80:00:00	6000	5961.04	5909.64	
50631396098526.556063	00:00:01	00:00:09	6000	5887.11	5906.48	
59251396098527.556925	00:00:01	00:00:10	6000	5947	5910.53	
38671396098528.558867	00:00:01	00:00:11	6000	5954.05	5913.94	
00001396098529.560000	00:00:01	00:00:12	6000	5994.01	5920.12	
77881396098529.817788	00:00:00	00:00:12	6000	2124.51	5840.65	
32421396098529.818242	00:00:00	00:00:12	6000	0	5840.18	

Sample SIPp log files

	BI	BJ	BK	BL	BM	BN	
(C)	ResponseTime1(P)	ResponseTime1(C)	ResponseTime1StDev(P)	ResponseTime1StDev(C)	CallLength(P)	CallLength(C)	CallLeng
C	00:00:00:00:00000	00:00:00:00:00000	00:00:00:00:00000	00:00:00:00:00000	00:00:00:00:00000	00:00:00:00:00000	00:00:00:
C	00:00:00:00:00000	00:00:00:00:00000	00:00:00:00:00000	00:00:00:00:00000	00:00:00:002000	00:00:00:002000	00:00:00:
C	00:00:00:00:00000	00:00:00:00:00000	00:00:00:00:00:00	00:00:00:00:00000	00:00:00:002000	00:00:00:002000	00:00:00:
C	00:00:00:00:00000	00:00:00:00:00000	00:00:00:00:00:00	00:00:00:00:00000	00:00:00:002000	00:00:00:002000	00:00:00:
C	00:00:00:001000	00:00:00:00:00000	00:00:00:026000	00:00:00:013000	00:00:00:006000	00:00:00:003000	00:00:00:
C	00:00:00:001000	00:00:00:00:00000	00:00:00:00:00000	00:00:00:012000	00:00:00:002000	00:00:00:003000	00:00:00:
C	00:00:00:00:00000	00:00:00:00:00000	00:00:00:00:00000	00:00:00:011000	00:00:00:002000	00:00:00:003000	00:00:00:
C	00:00:00:00:00000	00:00:00:00:00000	00:00:00:00:00:00	00:00:00:010000	00:00:00:002000	00:00:00:003000	00:00:00:
C	00:00:00:00:00000	00:00:00:00:00000	00:00:00:001000	00:00:00:009000	00:00:00:002000	00:00:00:003000	00:00:00:
C	00:00:00:004000	00:00:00:001000	00:00:00:045000	00:00:00:017000	00:00:00:012000	00:00:00:004000	00:00:00:
C	00:00:00:001000	00:00:00:001000	00:00:00:00:00000	00:00:00:016000	00:00:00:002000	00:00:00:004000	00:00:00:
C	00:00:00:001000	00:00:00:001000	00:00:00:00:00000	00:00:00:015000	00:00:00:003000	00:00:00:004000	00:00:00:
0	00:00:00:001000	00:00:00:001000	00:00:00:001000	00:00:00:015000	00:00:00:003000	00:00:00:004000	00:00:00:
0	00:00:00:048000	00:00:00:001000	00:00:00:146000	00:00:00:020000	00:00:00:084000	00:00:00:004000	00:00:00:
0	00:00:00:00:00000	00:00:00:001000	00:00:00:00:00000	00:00:00:020000	00:00:00:00:000	00:00:00:004000	00:00:00:
_							
_							

Limitations

- RTP support
- Scenario model can be inflexible
- Log file parsing can be tricky (and requires spare disk space)

Seagull

- Multi-protocol test tool from same team as SIPp
- (I don't maintain this)
- Similar principles scenario defined in an XML file
- Diameter and H.248 are probably the most interesting

<send channel="trans-ip-v4">

<command name="CER">

<avp name="Origin-Host" value="seagull.ims.hpintelco.org"> </avp>

<avp name="Origin-Realm" value="ims.hpintelco.org"> </avp>

<avp name="Host-IP-Address" value="0x00010a03fc5e"> </avp>

<avp name="Vendor-Id" value="11"> </avp>

<avp name="Product-Name" value="HP Cx Interface"> </avp> <receive channel="trans-ip-v4">
<action>
<stop-timer></stop-timer>
</action>
<command name="SAA">
</command>

</receive>

</init>

```
<send channel="channel-1">
```

<action>

```
<inc-counter name="transaction-counter"></inc-counter>
```

<set-value name="transaction-id"

```
format="$(transaction-counter)"></set-value>
```

</action>

```
<message>
```

```
<!-- header -->
```

```
<![CDATA[!/1 [16.16.88.188\]:55554
```

T=18571]] >

```
<!-- body -->
```

```
<![CDATA[C=${A=${M{TS{SI=iv,BF=off}},
```

ST=1{O{MO=sr,RV=off,RG=off},

R{m=audio 49152 RTP/AVP 3 97 98 8 0 101

c=IN IP4 16.16.214.175

a=rtpmap:3 GSM/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-11,16

}}}}}]]] >

</message>

Integration

- CSV injection
- Control sockets:
 - http://host:port/seagull/command/ramp&value=n&duration=d
 - echo '*' >/dev/udp/127.0.0.1/8888
- Log file parsing
 - Success rates
 - Response times
 - SIP error codes
- Running commands
 - SIPp's <exec> action



jSIPp

- Recent rewrite of SIPp in Java
- Use of Java and building on OSS libraries mean 90% reduction in codebase size
- More flexible, easier to add features
- Uses the same XML files and gets similar performance

jSIPp – the major change so far

- Stats are now published over ZeroMQ, a lightweight messaging protocol
- Every successful call, unexpected message, every timeout – all with timestamps
- A platform for writing test infrastructure

Activities 🕹 Firefox 🗸 😤 💘 🔒 🗸
SIPp Web Dashboard - Mozilla Firefox
Eile Edit Yiew History Bookmarks Tools Help If Facebook X If Clojure w X M Tech mail X If Bug 4710 X Java: How X If Clojure Do X <td< td=""></td<>
y Twitter www bbC News ™ Gmail Maracebook ✓ Light keading Wvebcomics ✓ Se nabiter's Your Life
SIPp Dashboard Settings Profile Help
Per-message details

Туре	In/out	Message count	Timeout	Unexpected
MESSAGE	<	719	0	0
200	>	717	0	0



Why Java?

The existing C++ code isn't going away

- Manual memory management = yikes!
- Speed
- Good SIP/RTP parsers already available
- The future: easy JRuby/Jython/Groovy scripting based on the SIPp core

jSIPp - coming up

- Better RTP testing including getting RTCP stats out for jitter analysis
- More flexible scenarios including registering then receiving a call
- Support for SIP-over-WebSocket performance testing
- I'm open to suggestions!



SIPp

- Docs: http://sipp.sourceforge.net/doc/reference.html
- Mailing list: sipp-users@lists.sourceforge.net
- Seagull
 - Docs: http://gull.sourceforge.net/doc/
 - Mailing list: gull-users@lists.sourceforge.net
- jSIPp
 - Github page: https://github.com/rkday/jsipp