

# 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

```
Activities Emacs v Sat 29 Mar, 13:15:54 uac.xml - emacs@localhost.localdomain x
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">
26     <![CDATA[
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@[local_ip]:[local_port]>;tag=[pid]SIPpTag00[call_number]
31     To: [service] <sip:[service]@[remote_ip]:[remote_port]>
32     Call-ID: [call_id]
33     CSeq: 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     s=-
43     c=IN IP[media_ip_type] [media_ip]
44     t=0 0
45     m=audio [media_port] RTP/AVP 0
46     a=rtpmap:0 PCMU/8000
47
48     ]]>
49   </send>
50
51   <recv response="100"
52     optional="true">
53   </recv>
54
55   <recv response="180" optional="true">
56   </recv>
57
58   <recv response="183" optional="true">
59
60 1:--- uac.xml 31% L40 (nXML Valid WS yas Fill)
```

# Sample SIPp command-line output

```
----- Scenario Screen ----- [1-9]: Change Screen --
  Call-rate(length)  Port  Total-time  Total-calls  Remote-host
5000.0(0 ms)/1.000s  5061   10.01 s     59188  127.0.0.1:5060(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)
3 open sockets

          Messages  Retrans  Timeout  Unexpected-Msg
INVITE ----->      59176    65       0         0
  100 <-----      0         0       0         0
  180 <-----      59168    0       0         0
  183 <-----      0         0       0         0
  200 <----- E-RTD1 59168    0       0         0
  ACK ----->      59168    0       0         0
Pause [    0ms]      59168    0       0         0
  BYE ----->      59164    83      0         0
  200 <-----      59164    0       0         0

----- [ + | - | * | / ] : Adjust rate ---- [ q ] : Soft exit ---- [ p ] : Pause traffic -----
Last Error: Dead call 44798-12562@127.0.0.1 (successful), received 'SIP/...'

```







# 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

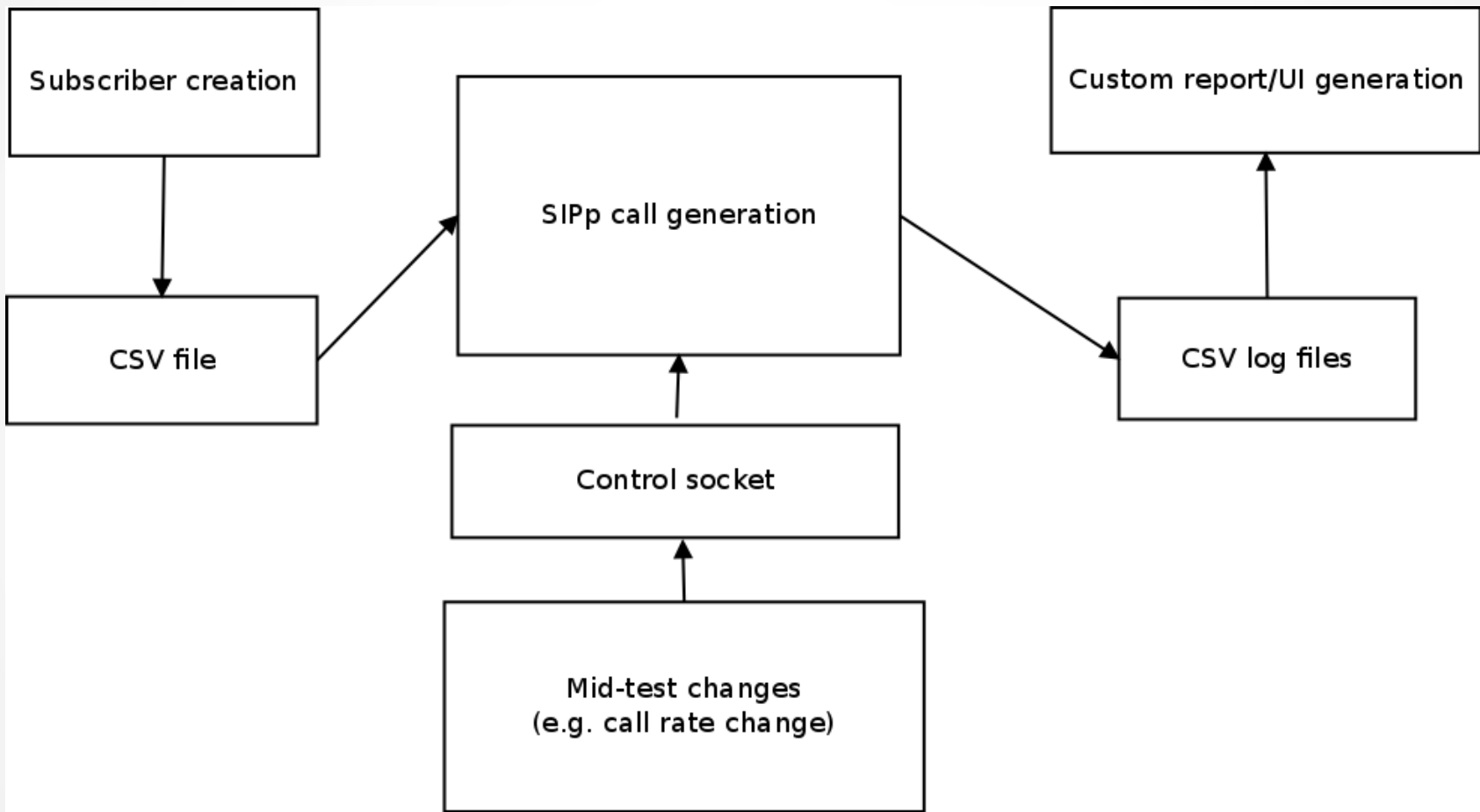
```
<send channel="trans-ip-v4">
  <command name="CER">
    <avp name="Origin-Host"
value="seagull.ims.hpintelco.org"> </avp>
    <avp name="Origin-Realm"
value="ims.hpintelco.org"> </avp>
    <avp name="Host-IP-Address"
value="0x00010a03fc5e"> </avp>
    <avp name="Vendor-Id" value="11"> </avp>
    <avp name="Product-Name" value="HP Cx
Interface"> </avp>
</init>
```

```
<receive channel="trans-ip-v4">
  <action>
    <stop-timer></stop-timer>
  </action>
  <command name="SAA">
  </command>
</receive>
```

```
<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



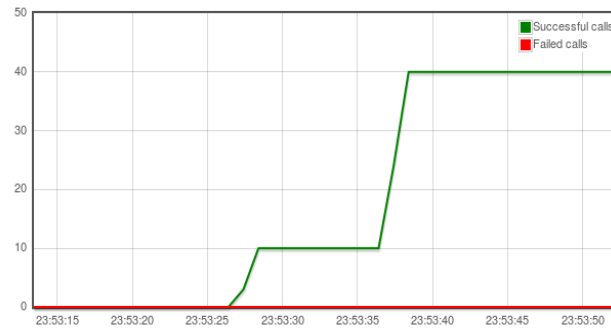
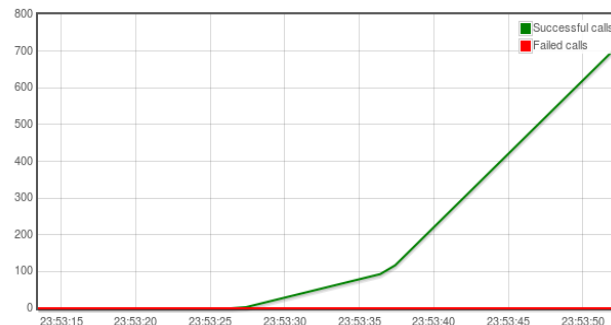
# jSIPp

- 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



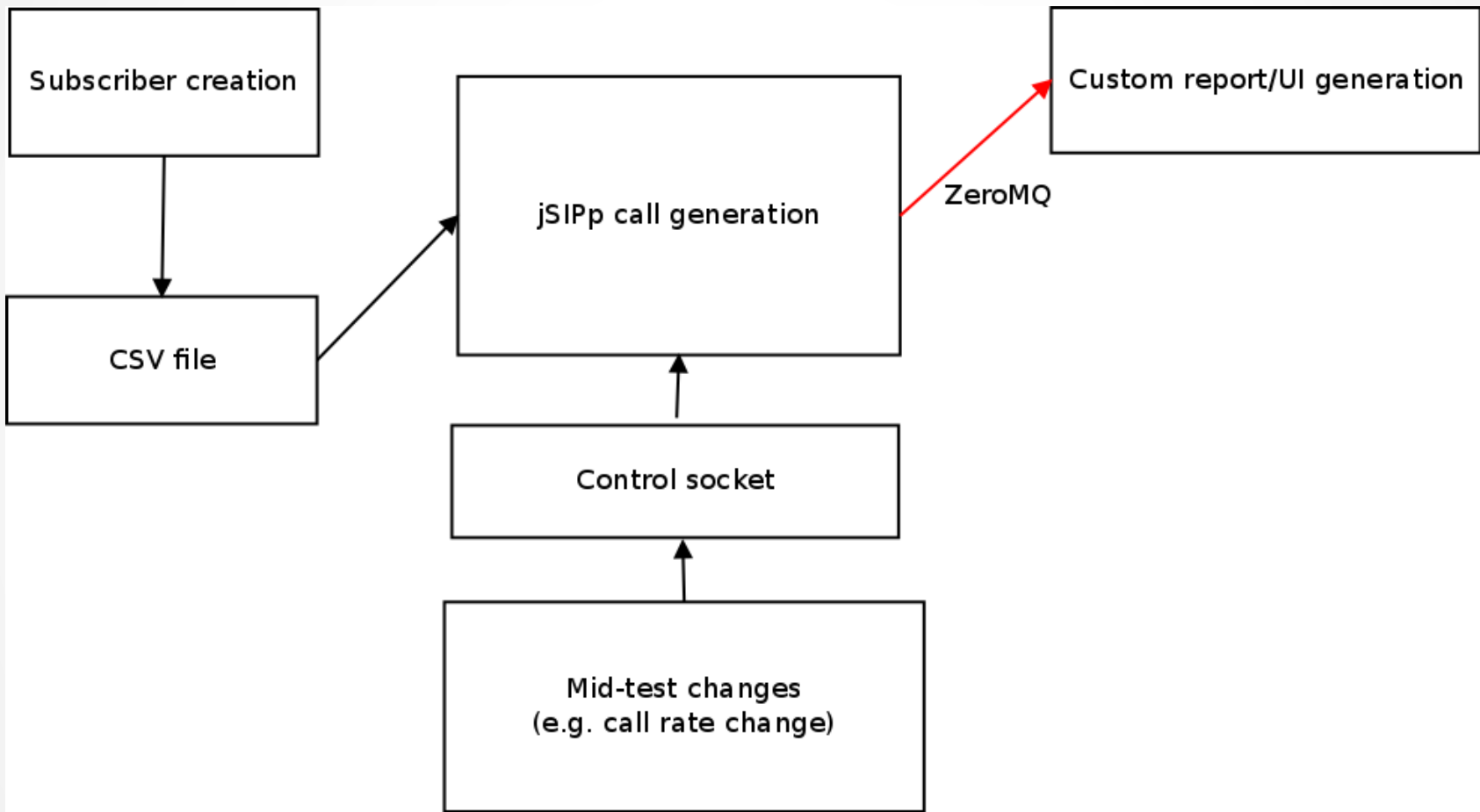
# jSIPp – the major change so far

- 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



### Per-message details

Type	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!

# References

- SIPp
  - Docs: <http://sipp.sourceforge.net/doc/reference.html>
  - Mailing list: [sipp-users@lists.sourceforge.net](mailto:sipp-users@lists.sourceforge.net)
- Seagull
  - Docs: <http://gull.sourceforge.net/doc/>
  - Mailing list: [gull-users@lists.sourceforge.net](mailto:gull-users@lists.sourceforge.net)
- jSIPp
  - Github page: <https://github.com/rkday/jsipp>