



Open Source Test Tools for telephony

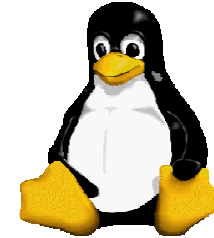
Linux Tag 2006

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Agenda

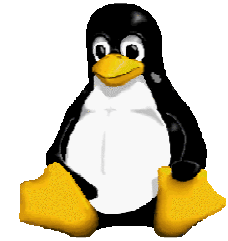


- Challenges
- Testing telephony with Open Source test tools
- The tools
 - List of tools relevant to telephony, per category
 - Main features
 - Health level
 - Features that are missing: **get involved!**
- Demos

Challenges

- Testing is critical in telephony
 - The famous 5 nines (99.999% availability – a little more than 5 minutes down time per year)
 - A phone call could save a life!
- Security threats on IP telephony: telephony is now more accessible than ever
- Media quality – voice and video
- New technologies, new services deployed very quickly: how to cope with testing?

Testing telephony with Open Source test tools



- Advantages of using Open Source test tools for telephony:
 - **Up-to-date**: the feature list follows the state of the art
 - **No usage limit**: easy to get: more tests, less bugs
 - **Customizable**: add your own features and get feedback
 - Some **de-facto standard** test tools are already Open Source
 - Some tools are **unique**, not available in a commercial form
- Drawbacks
 - No committed **support**: be careful to choose the right, active tools.
 - Tool's **quality** could be so so: you can enhance it yourself
 - Difficult to implement **licensed** algorithms (eg. PESQ voice quality)

Soft phones

Name	Features	Missing	Licen.	Health
Linphone	SIP/audio only, G711-a/ulaw, LPC10-15, GSM, SPEEX and iLBC, DTMF 2833, sipomatic	Win32 port, video, scripting	GPL	Moderate
WengoPhone	SIP/audio + video, SMS, chat/presence (via libgaim), STUN, G711, iLBC, GSM, AMR, SPEEX, H263, Python binding	SIP traces, SDP negotiation, media stats, session record	GPL	Full speed
Ekiga	SIP, H323/audio (iLBC, GSM-06.10, MS-GSM, G.711-a/ulaw, G.726, G.721 and Speex Audio Codecs) + video (H.261)	Same as WengoPhone except SIP traces	GPL	Full speed

Protocol

Name	Features	Missing	Licen.	Health
SIPsak	Small comand line tool to send/receive SIP messages. Authentication (MD5, SHA1), flooding tests.		GPL	Moderate
SFTF	Test tool for common SIP errors. Includes a User Agent test suite		GPL	Moderate
SIPUnit	Environment to unit test SIP applications, extending JUnit framework.		Apache	Full speed

Monitoring



Name	Features	Missing	Licen.	Health
Ethereal	Capture and decode most protocols, support all telephony protocols over IP and SS7 (E1/T1), RTP capture and listening (G711), scripting language (Lua)	More audio decoding, Video	GPL	Full speed
ntop	ntop is a network traffic probe that shows the network usage, similar to what the top command does.	More VoIP and Video support (introduced in 3.2)	GPL	Moderate
ngrep	grep for network		BSD Like	Moderate
VoIPong	Detection and recording of RTP streams as wav files (control plane independent) – lightweight.	SIP filters, multi-streams (video+audio)	GPL	Moderate
Oreka	Recording and retrieval of VoIP streams (RTP using SIP or Skinny). Aimed at call centers.	Multi-streams (video+audio)	GPL	Full speed

Security



Name	Features	Missing	Licen.	Health
SIPbomber	Create SIP messages from BNF, analyze RFC compliance, CERT tests (PROTOS) – UAS only testing.	UAC support	GPL	Dead
Nastysip	Generate bogus SIP-messages, torture tests.		GPL	Dead
PROTOS SIP suite	Test-suite to evaluate implementation level security and robustness of SIP implementation	More tests, fuzz testing	GPL	Moderate
SIP-scan	Scan network for SIP clients		GPL	Moderate
SIP-kill	Scan and kill SIP calls		GPL	Moderate
SIP-redirecttp	Intercept RTP by manipulating SIP SDPs		GPL	Moderate
SIPp	See "Traffic/load generators" category. Security features: can send any (mis-constructed) SIP message as well as any (mis-constructed) RTP packet.	Security test suites, fuzz testing, SIP + RTP test suites, test sequencing + report	GPL	Full speed

Traffic / load generator

Name	Features	Missing	Licen.	Health
SIPp	SIP + RTP (capture replay), XML scenarios, external database, call variables, regular expressions evaluation in scenarios, dynamic scenarios (branching), UDP/TCP/TLS, IPv4/IPv6.	Media quality metrics (MOS/E-factor), AKA authentication.	GPL	Full speed
Grinder	Java load testing framework. Supports http, https, soap, XML/RPC, pop3, smtp, ldap. Plugin architecture.	Telephony protocols	BSD Like	Full speed
Seagull	Multi-protocol traffic generator. Diameter (base and all applications (Cx, Dx, Ro, Rf, Sh, ...)), radius, TCAP over SS7 or Sigtran. Transports: UDP/TCP/SCTP/SS7. To come: http, xcap, h248 ascii, SIP, Sigtran SUA.	Media, more protocols, traffic shaping	GPL	Just GPL'ed

Network impairment

Name	Features	Missing	Licen.	Health
NISTNet	Packet delay, jitter, bandwidth limitation (asymmetric), packet re-ordering, congestion, packet lost and duplication. Linux Kernel module. Feedback loop.	Dynamic (change over time) scenarios	NIST	Low
Dummynet	Same as NISTNet, for freeBSD		BSD	Dead

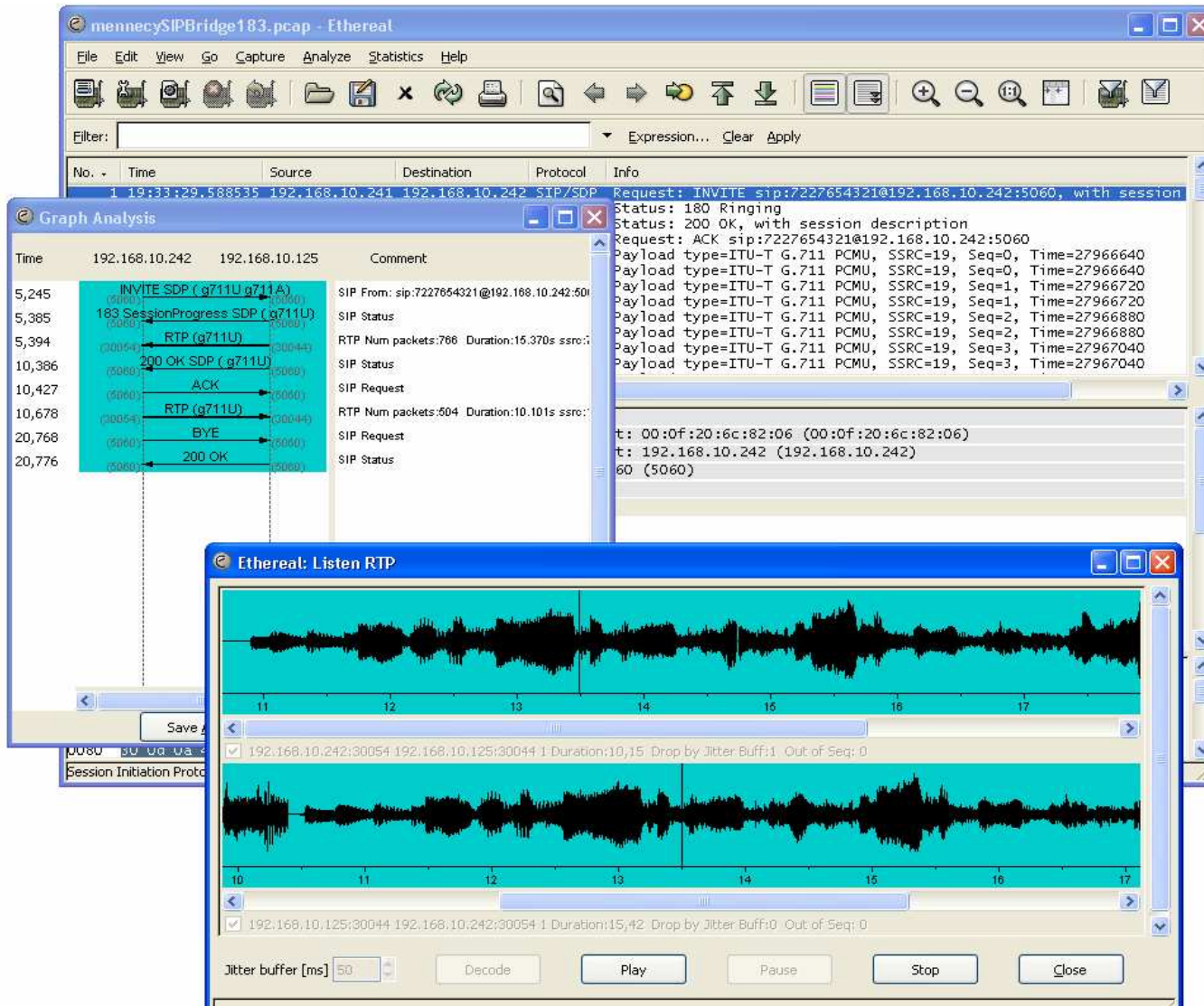
Media



Name	Features	Missing	Licen.	Health
Ffmpeg	Encoding, transcoding and streaming of audio and video files. Supports 3gp, amr, h263, h264, mpeg4, ...		GPL	Full speed
VLC	Multi-media player, encoder, streaming client & server		GPL	Full speed
GStreamer	Multimedia framework to handle media. Flexible "graphs" to create complex combinations between functions.	Audio and video quality metrics	LGPL	Full speed
H323 Beacon	H323 video conference monitoring (frame rate, throughput, jitter, mos). H323 generator.	SIP	Mozilla	Moderate

Demo 1: SIP+RTP monitoring (Ethereal)

Launch demo



The screenshot shows the Ethereal network protocol analyzer interface. The main window displays a packet list with the following entries:

No.	Time	Source	Destination	Protocol	Info
1	19:33:29.588535	192.168.10.241	192.168.10.242	SIP/SDP	Request: INVITE sip:7227654321@192.168.10.242:5060, with session

The 'Graph Analysis' window shows a sequence of messages between 192.168.10.242 and 192.168.10.125:

Time	Source	Destination	Comment
5,245	192.168.10.242	192.168.10.125	INVITE SDP (g711U g711A)
5,385	192.168.10.125	192.168.10.242	183 SessionProgress SDP (g711U)
5,394	192.168.10.242	192.168.10.125	RTP (g711U)
10,386	192.168.10.125	192.168.10.242	200 OK SDP (g711U)
10,427	192.168.10.242	192.168.10.125	ACK
10,678	192.168.10.242	192.168.10.125	RTP (g711U)
20,768	192.168.10.125	192.168.10.242	BYE
20,776	192.168.10.242	192.168.10.125	200 OK

The 'Ethereal: Listen RTP' window shows two audio waveforms for RTP packets. The top waveform is for packet 1 (Duration: 10,15) and the bottom waveform is for packet 2 (Duration: 15,42). Both waveforms show a clear audio signal.

Demo 2: SIP + RTP traffic (SIPp)

Launch demo

```

root@shasta:/home/testvas/vas/programs/sipp.cumulus.2006-...
----- Scenario Screen ----- [1-4]: Change Screen --
Call-rate(length)  Port  Total-time  Total-calls  Remote-host
135.0(0 ms)/1,000s  5060  88.67 s    11860  192,168,10,182;5061(UDP)

136 new calls during 1,010 s period  5 ms scheduler resolution
1240 concurrent calls (limit 5000)   Peak was 1257 calls, after 62 s
0 out-of-call msg (discarded)
1 open sockets
2769319 Total RTP pkts                8019,026 last period RTP rate (kB/s)

  INVOKE ----->      Messages  Retrans  Timeout  Unexpected-Msg
    100 <-----      11860    0        0         0
    180 <-----      11860    0        0         0
    200 <----- E-RTD  11860    0        0         0

  ACK ----->         11860    0
  [ NOP ]
  Pause [ 8000ms]      11860    0
  [ NOP ]
  Pause [ 1000ms]     10761    0
  BYE ----->         10620    0        0         0
  200 <-----         10620    0        0         0

----- [+|-|*|/]: Adjust rate ---- [q]: Soft exit ---- [p]: Pause traffic ----

```

```

root@voip1:/home/testvas/vas/programs/sipp.cumulus.2006...
----- Scenario Screen ----- [1-4]: Change Screen --
Port  Total-time  Total-calls  Transport
5061  116,82 s    13026  UDP

144 new calls during 1,070 s period  34 ms scheduler resolution
2973 concurrent calls                Peak was 3004 calls, after 116 s
1 open sockets

-----> INVITE      Messages  Retrans  Timeout  Unexpected-Msg
      13026    0         0         0
<----- 180      13026    0         0         0
<----- 200      13026    0         0         0
-----> ACK      E-RTD    13026    0         0

-----> BYE      10591    0         0         0
<----- 200      10591    0         0         0
[ 4000ms] Pause  10591    0         0         0

----- Sipp Server Mode -----

```

```

root@shasta:/home/testvas/vas/programs
bwm-ng v0.5 (probing every 1,000s), press 'h' for help
input: /proc/net/dev type: rate
-   iface      Rx              Tx              Total
-----
  lo:          0,00 KB/s      0,00 KB/s      0,00 KB/s
  eth0:        11,47 KB/s     1,76 KB/s     13,24 KB/s
  eth1:       173,84 KB/s  9259,06 KB/s  9432,91 KB/s
-----
 total:       185,32 KB/s  9260,83 KB/s  9446,15 KB/s

```

```

root@voip1:~
15:02:34 up 7 days, 20:57, 6 users, load average: 0.02, 0.02, 0.13
219 processes: 215 sleeping, 4 running, 0 zombie, 0 stopped
CPU states:  cpu  user  nice  system  irq  softirq  iowait  idle
total       3.4%  0.0%  2.8%   0.0%  10.6%   2.2%   80.6%
cpu00       0.1%  0.0%  0.5%   0.0%   0.0%   2.7%   96.4%
cpu01       6.7%  0.0%  5.1%   0.0%  21.3%   1.7%   64.8%
Mem: 3599008k av, 2536548k used, 1062460k free, 0k shrd, 261628k buff
774460k active, 1212212k inactive
Swap: 2044056k av, 0k used, 2044056k free 1777412k cached

PID USER  PRI  NI  SIZE  RSS  SHARE  STAT %CPU %MEM  TIME CPU COMMAND
23009 root   15  0 16112 15M  9532 S   3.8  0.4  0:24 1 ethereal
23099 root   15  0 12524 12M  1276 S   2.1  0.3  0:03 1 sipp
23103 root   15  0 1440 1440  884 R   0.1  0.0  0:00 0 top
1 root   15  0 500 500  432 S   0.0  0.0  0:08 1 init
2 root   RT  0  0  0  0 SW   0.0  0.0  0:00 0 migration/0
3 root   RT  0  0  0  0 SW   0.0  0.0  0:00 1 migration/1
4 root   15  0  0  0  0 SW   0.0  0.0  0:00 1 keventd
5 root   34 19  0  0  0 SWN  0.0  0.0  0:00 0 ksoftirqd/0
6 root   34 19  0  0  0 SWN  0.0  0.0  0:00 1 ksoftirqd/1

```