

Test Report of Atom CPU with asterisk G729-G711 codec transcoding

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Date: 06/11/2008

Some people buy the Intel CPU (Atom 230) to build an asterisk server. I did a simple test for codec transcoding. The purpose of test case is only for reference when you build a Atom CPU based asterisk server, maybe the test environment is not really completed due to some limitations such as test tools, bandwidth of LAN, Network card, version of g729 and the duration of timing, but I try to give you a picture for asterisk server with transcoding. In this paper, I will cover installation of G729, testing tools, result of testing and some screens.

1) Installation of Open Source G729

Before installing g729 codec, make sure the asterisk server can run properly, then go to the official website to get the binary files and copy those two files into the default path. The two figure show the modules as below:

```

app_read.so          chan_phone.so
app_realtime.so     chan_sip.so
app_record.so       chan_skinny.so
app_sayunixtime.so  chan_zap.so
app_senddtmf.so     codec_adpcm.so
app_sendtext.so     codec_alaw.so
app_setcallerid.so  codec_g723.so
app_setcdruserfield.so  codec_g723-ast14-icc-glibc-pentium4-sse3.so
app_settransfercapability.so  codec_g726.so
app_sms.so          codec_g729-ast14-gcc4-glibc-pentium4-sse3.so
app_softhangup.so   codec_gsm.so

```

**under
/usr/lib/asterisk**

```

bogon*CLI> show translation
      Translation times between formats (in milliseconds) for one second of data
      Source Format (Rows) Destination Format (Columns)

```

	g723	gsm	ulaw	alaw	g726aal2	adpcm	slin	lpc10	g729	speex	ilbc	g726	g722
g723	-	6	2	2	6	2	1	9	14	-	-	6	-
gsm	19	-	3	3	7	3	2	10	15	-	-	7	-
ulaw	18	6	-	1	6	2	1	9	14	-	-	6	-
alaw	18	6	1	-	6	2	1	9	14	-	-	6	-
g726aal2	21	9	5	5	-	5	4	12	17	-	-	1	-
adpcm	18	6	2	2	6	-	1	9	14	-	-	6	-
slin	17	5	1	1	5	1	-	8	13	-	-	5	-
lpc10	21	9	5	5	9	5	4	-	17	-	-	9	-
g729	21	9	5	5	9	5	4	12	-	-	-	9	-
speex	-	-	-	-	-	-	-	-	-	-	-	-	-
ilbc	-	-	-	-	-	-	-	-	-	-	-	-	-
g726	21	9	5	5	1	5	4	12	17	-	-	-	-
g722	-	-	-	-	-	-	-	-	-	-	-	-	-

Figure 1

2) Set testing tools

Here, three tools are used: Sipp, tcpdump and wireshark. Please go to those official websites to get those tools. You must use tcpdump or wireshark to get a G729 code pcap file. The easy way to get G729 file is that, using Xlite-Pro version to call other SIP phone and record down the file with G729 codec by this:

```
tcpdump -T rtp -vvv dst 192.168.2.108 -w g729.pcap
```

This should capture the RTP stream from asterisk server and save it as g729.pcap file. You must make sure the Xlite-pro solely use G729 codec.

You also can use Wireshark to capture G729 codec and save as G729.pcap. Capturing the G729 RTP stream by Wireshark filter:

(ip.dst == 192.168.2.108) && (rtp.p_type == 18)

this will filter the G729 codec from 192.168.2.108. Once you get the G729 codec file, you put the file under pcap folder under

Sipp:

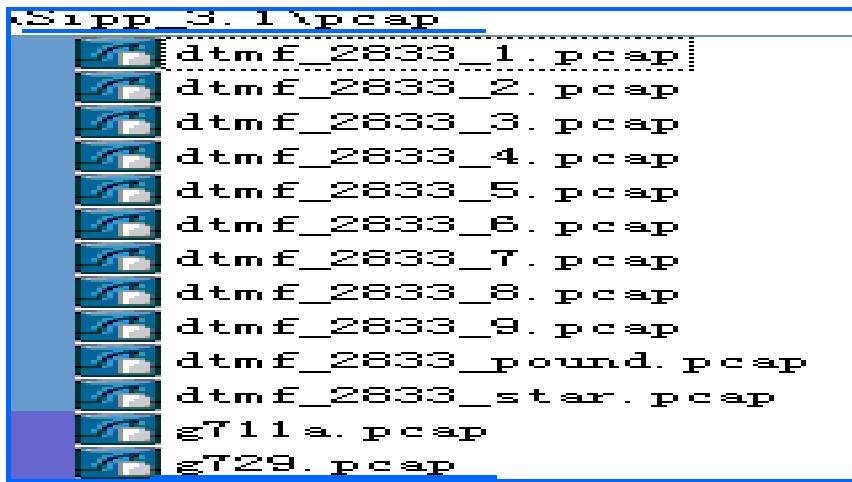


Figure 2

After that, you have to edit the uac_pcap.xml to make sure Sipp will play with RTP stream. You have to edit the uac_pcap.xml like this:

```

Subject: Performance test
Content-Type: application/sdp
Content-Length: [len]
v=0
o=user1 53655765 2353687637 IN IP[local_ip_type] [local_ip]
s=-
c=IN IP[local_ip_type] [local_ip]
t=0 0
m=audio [auto_media_port] RTP/AVP 8
a=rtpmap:8 G729/8000 // change to g729 if test g729
]]>
</send>
<recv response="100" optional="true">
</recv>
<recv response="180" optional="true">
</recv>
<send>
<![CDATA[
ACK sip:[service]@[remote_ip]:[remote_port] SIP/2.0
Via: SIP/2.0/[transport] [local_ip]:[local_port];branch=[branch]
From: sipp <sip:sipp@[local_ip]:[local_port]>;tag=[call_number]
To: sut <sip:[service]@[remote_ip]:[remote_port]>[peer_tag_param]
Call-ID: [call_id]
CSeq: 1 ACK
Contact: sip:sipp@[local_ip]:[local_port]
Max-Forwards: 70
Subject: Performance Test
Content-Length: 0
]]>
</send>
<!-- Play a pre-recorded PCAP file (RTP stream) -->
<nop>
<action>
<exec play_pcap_audio="pcap/g729.pcap"/> // change to g729, if test g729
</action>

```

Figure 3

Once the Sipp side is done, you have to add a sip account in asterisk server 1. The sip is named sipp. Please add an account in asterisk sip.conf. the SIP account information should like this:

```

[sipp]
type=friend
context=internal
host=192.168.2.111
port=6000
user=sipp
canreinvite=no
disallow=all
allow=g729
;allow=alaw
;allow=ulaw

```

Figure 4

And you add other sip (for example 1000) account with codec allow=ulaw or alaw only. SIP 1000 will forward the sip call from Spp to asterisk 2, in asterisk 2, some sound files will be played for certain periods. The dialplan in asterisk 1 likes this:

```
[internal]
; dummy extension just for Server Asterisk 2 IP
exten => 2005,1,Answer
exten => 2005,n,DIAL(SIP/1000@192.168.2.127,80,r)
exten => 2005,n,Hangup
```

Figure 5

In this scenario, transcoding will be done from G729 to G711. If you do not set it properly, asterisk server will report codec compatibility error. The Sipp test can not be made, please double check that. Until this step, you can execute the Sipp command to test:

```
sipp -sf uac_pcap.xml -s 2005 192.168.2.108 -r 20 -rp 10000
```

sipp will call uac_pcap.xml file first, and go to asterisk dialplan, the context “internal” will be called with asterisk server 1. It will generate 20 calls in 10 seconds. You can test it with different time variables. You also can press =-*/ to increase the calls or decrease calls. You can monitor the calls during call connection time by running *sip show channels* under asterisk console, you will see the sipp using g729 and 1000 using ulaw. The figure shows this:

```

-- Executing [2005@internal:1] Answer("SIP/sipp-b7d0da18", "") in new stack
-- Executing [2005@internal:2] Dial("SIP/sipp-b7d0da18", "SIP/1000@192.168.2.127|80|r") in new
-- Called 1000@192.168.2.127
-- SIP/192.168.2.127-09d110f0 answered SIP/sipp-b7d0da18
mogon*CLI> sip show channels
Peer          User/ANR      Call ID      Seq (Tx/Rx)  Format          Hold    Last Message
192.168.2.127 1000         40251edf272 00102/00000  Dx80004 (ulaw|h No      Tx: ACK
192.168.2.111 sipp        2-608@192.1 00101/00001  Dx100 (g729)  No      Rx: ACK
192.168.2.127 1000         398774836a5 00102/00000  Dx80004 (ulaw|h No      Tx: ACK
192.168.2.111 sipp        1-608@192.1 00101/00001  Dx100 (g729)  No      Rx: ACK
192.168.2.127 (None)      07c6bb5568e 00101/00102  Dx0 (nothing) No      Rx: OPTIONS
192.168.2.127 1000         39d177673b3 00102/00000  Dx80004 (ulaw|h No      Tx: ACK
192.168.2.111 sipp        453-2420@19 00101/00001  Dx100 (g729)  No      Rx: ACK
192.168.2.127 1000         3b2b9d1a43f 00102/00000  Dx80004 (ulaw|h No      Tx: ACK
192.168.2.111 sipp        452-2420@19 00101/00001  Dx100 (g729)  No      Rx: ACK
192.168.2.127 1000         330091333a1 00102/00000  Dx80004 (ulaw|h No      Tx: ACK
192.168.2.111 sipp        451-2420@19 00101/00001  Dx100 (g729)  No      Rx: ACK
192.168.2.127 1000         08ce67ad11a 00102/00000  Dx80004 (ulaw|h No      Tx: ACK
192.168.2.111 sipp        450-2420@19 00101/00001  Dx100 (g729)  No      Rx: ACK
192.168.2.127 1000         73edbf397a6 00102/00000  Dx80004 (ulaw|h No      Tx: ACK
192.168.2.111 sipp        449-2420@19 00101/00001  Dx100 (g729)  No      Rx: ACK
15 active SIP channels
-- Executing [2005@internal:1] Answer("SIP/sipp-b7d25538", "") in new stack
-- Executing [2005@internal:2] Dial("SIP/sipp-b7d25538", "SIP/1000@192.168.2.127|80|r") in new

```

Figure 6

3) Result of Testing

The results are summarized to give users some statistical data. The scenarios are:

The scenario one:

Sipp(g711)->asterisk-1 with Atom CPU (g711)->asterisk-2(g711)

The scenario two:

Sipp(g729)->asterisk-1 with Atom CPU

(g729->g711)->asterisk-2(g711)

After testing, the results are show as below:

G729->ulaw

	Usage of CPU	Current calls	Mem	CPS
10c-10s	40%	18	3.5	2
20c-10s	53%	24	3.5	2
25c-10s	70%	30	3.5	3
30c-10s	93%	36	3.5	3
Ulaw->ulaw				
	Usage of CPU	Current calls	Mem	CPS
10c-10s	9%	12	3.5	1
20c-10s	10%	24	3.5	1
25c-10s	17%	30	3.5	2
30c-10s	17%	37	3.5	3

Table 1

Measurement: calls in 10 seconds, for example: 10c-10s means sipp will generate 10 calls in 10 second.

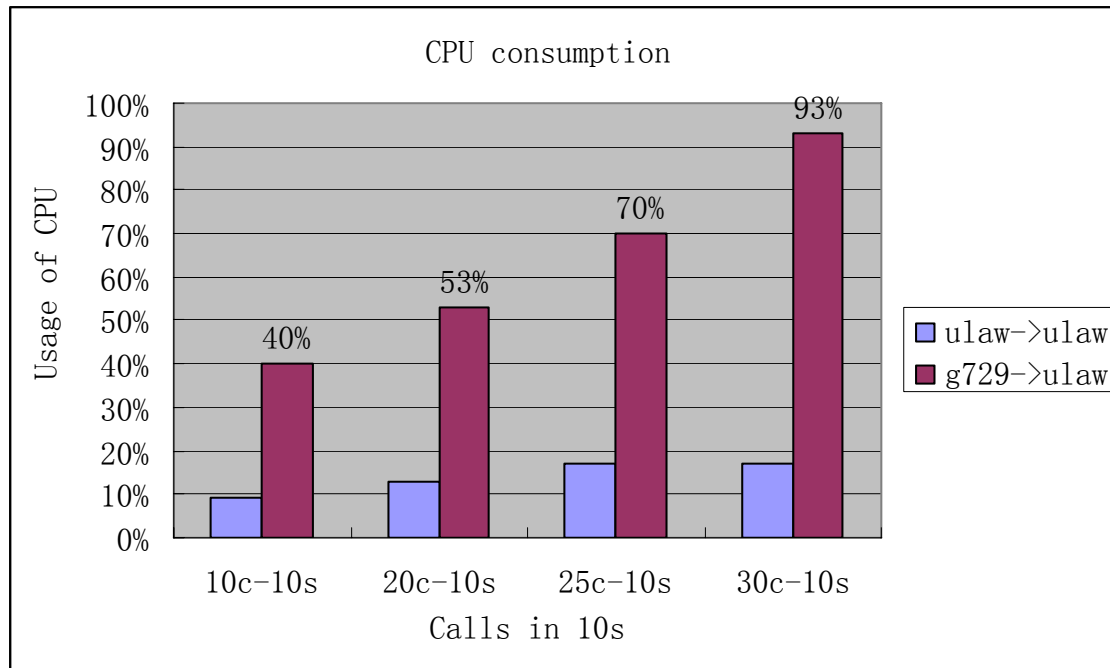


Figure 7

In conclusion, codec contanscoding will consume much CPU resource. During the test, some factors must be considered. They are duration of each events, codecs, length of RTP streams, condition of Lan transmission, Network cards of asterisk servers. For Intel Atom CPU, the current calls should be limited less 30 calls. When the peak time reaches, the SIP calls will generate some warning. For further test improvement, it is very necessary to make a further investigation with g729 codec under Sipp RTP test for more accurate result.

Some screens captured for reference:

The scenario one:

Sipp(ulaw)=>Asterisk1(ulaw)=>asterisk2(ulaw)

```
top - 19:45:07 up 5:55, 2 users, load average: 0.00, 0.03, 0.00
Tasks: 76 total, 1 running, 75 sleeping, 0 stopped, 0 zombie
Cpu(s): 0.3%us, 0.7%sy, 0.0%ni, 97.2%id, 0.0%wa, 1.7%hi, 0.2%si, 0.0%st
Mem: 505656k total, 287456k used, 218200k free, 40284k buffers
Swap: 1015800k total, 0k used, 1015800k free, 165764k cached
```

PID	USER	PR	NI	VIRT	RES	SHR	S	%CPU	%MEM	TIME+	COMMAND
2404	root	15	0	75764	30m	5548	S	2	6.3	15:44.55	asterisk
1	root	15	0	2040	632	544	S	0	0.1	0:00.64	init
2	root	RT	0	0	0	0	S	0	0.0	0:00.00	migration/0
3	root	34	19	0	0	0	S	0	0.0	0:00.00	ksoftirqd/0
4	root	RT	0	0	0	0	S	0	0.0	0:00.00	watchdog/0

```

CA start sipp - sipp -sf uac_pcap.xml -s 2005 192.168.2.108 -r 10 -r...
Call-rate<length> Port Total-time Total-calls Remote-host
10.0<0 ms>/20.000s 5060 21.00 s 10 192.168.2.108:5060<UDP>

0 new calls during 1.000 s period 1 ms scheduler resolution
5 calls <limit 13> Peak was 5 calls, after 10 s
0 Running, 10 Paused, 1 Woken up
0 dead call msg <discarded> 0 out-of-call msg <discarded>
3 open sockets
2012 Total RTP pkts sent 33.676 last period RTP rate <kB/s>

Messages Retrans Timeout Unexpected-Msg
INVOKE -----> 10 0 0
100 <----- 10 0 0
180 <----- 0 0 0
200 <----- E-RTD1 10 0 0

ACK -----> 10 0
[ NOP ]
Pause [ 8000ms ] 10
[ NOP ]
Pause [ 1000ms ] 6 0
BYE -----> 5 0 0
200 <----- 5 0 0
    
```

20s 10 Calls

40s, Cpu 30%, increased by 5 calls in 5 s

```

c:\ start sipp - sipp -sf uac_pcap.xml -s 2005 192.168.2.108 -r 1 -rp...
Start Time      | 2008-10-27 20:17:46:882 | 1225109866.882182
Last Reset Time | 2008-10-27 20:18:24:927 | 1225109904.927182
Current Time    | 2008-10-27 20:18:25:923 | 1225109905.923182
-----+-----+-----
Counter Name   | Periodic value         | Cumulative value
-----+-----+-----
Elapsed Time   | 00:00:00:996          | 00:00:39:041
Call Rate      | 4.016 cps              | 1.665 cps
-----+-----+-----
Incoming call created | 0                      | 0
OutGoing call created | 4                      | 65
Total Call created  |                        | 65
Current Call     | 29                     |
-----+-----+-----
Successful call  | 2                      | 35
Failed call     | 0                      | 1
-----+-----+-----
Response Time 1  | 00:00:00:001          | 00:00:00:024
Call Length     | 00:00:09:005          | 00:00:08:841
-----+-----+-----
[+!-!*!/: Adjust rate ---- [q]: Soft exit ---- [p]: Pause traffic ----

```

The scenario two:

Sipp(g729)=>Asterisk-1(g729->ulaw)=>asterisk-2(ulaw)

Case 1

Cpu 15% asterisk, 20s, 10 calls

```

new-host*CLI>
new-host*CLI> show translation
Translation times between formats (in milliseconds) for one second of data
Source Format (Rows) Destination Format (Columns)
g723 gsm ulaw alaw g726aal2 adpcm slin lpc10 g729 speex ilbc g726 g722
g723 - 8 2 2 6 2 1 9 14 - - 6 -
gsm 27 - 3 3 7 3 2 10 15 - - 7 -
ulaw 26 8 - 1 6 2 1 9 14 - - 6 -
alaw 26 8 1 - 6 2 1 9 14 - - 6 -
g726aal2 29 11 5 5 - 5 4 12 17 - - 1 -
adpcm 26 8 2 2 6 - 1 9 14 - - 6 -
slin 25 7 1 1 5 1 - 8 13 - - 5 -
lpc10 29 11 5 5 9 5 4 - 17 - - 9 -
g729 29 11 5 5 9 5 4 12 - - - 9 -
speex - - - - - - - - - - - -
ilbc - - - - - - - - - - - -
g726 29 11 5 5 1 5 4 12 17 - - -
g722 - - - - - - - - - - - -

```

```

10.0<0 ms>/20.000s 5060 21.01 s 10 192.168.2.108:5060<UDP>
0 new calls during 1.001 s period 1 ms scheduler resolution
4 calls <limit 13> Peak was 5 calls, after 10 s
0 Running, 10 Paused, 1 Woken up
0 dead call msg <discarded> 0 out-of-call msg <discarded>
3 open sockets
5036 Total RTP pkts sent 14.658 last period RTP rate <kB/s>

Messages Retrans Timeout Unexpected-Msg
INVITE -----> 10 0 0
100 <----- 10 0 0
180 <----- 0 0 0
200 <----- E-RTD1 10 0 0

ACK -----> 10 0
[ NOP ]
Pause [ 8000ms ] 10
[ NOP ]
Pause [ 1000ms ] 6
BYE -----> 6 0 0
200 <----- 6 0 0

```

CPU over 80%, increased by 5 calls in 5s

```

c:\ start sipp - sipp -sf uac_pcap.xml -s 2005 192.168.2.108 -r 1 -rp...
Start Time | 2008-10-27 20:37:42:054 | 1225111062.054057
Last Reset Time | 2008-10-27 20:39:07:191 | 1225111147.191057
Current Time | 2008-10-27 20:39:08:179 | 1225111148.179057
-----+-----+-----
Counter Name | Periodic value | Cumulative value
-----+-----+-----
Elapsed Time | 00:00:00:988 | 00:01:26:125
Call Rate | 8.097 cps | 3.994 cps
-----+-----+-----
Incoming call created | 0 | 0
OutGoing call created | 8 | 344
Total Call created | | 344
Current Call | 70 |
-----+-----+-----
Successful call | 8 | 274
Failed call | 0 | 0
-----+-----+-----
Response Time 1 | 00:00:00:007 | 00:00:00:006
Call Length | 00:00:09:015 | 00:00:09:012
----- [ +!-!*!/: Adjust rate ---- [q]: Soft exit ---- [p]: Pause traffic ----

谷歌拼音 半:

```

```

top - 20:29:32 up 6:39, 2 users, load average: 6.60, 1.86, 0.69
Tasks: 76 total, 3 running, 73 sleeping, 0 stopped, 0 zombie
Cpu(s): 79.4%us, 9.5%sy, 0.0%ni, 2.0%id, 0.0%wa, 3.6%hi, 5.5%si, 0.0%st
Mem: 505656k total, 301636k used, 204020k free, 44272k buffers
Swap: 1015800k total, 0k used, 1015800k free, 167132k cached

```

PID	USER	PR	NI	VIRT	RES	SHR	S	%CPU	%MEM	TIME+	COMMAND
2404	root	15	0	85712	38m	5556	S	196	7.8	24:32.29	asterisk
1	root	15	0	2040	632	544	S	0	0.1	0:00.64	init
2	root	RT	0	0	0	0	S	0	0.0	0:00.00	migration/0
3	root	34	19	0	0	0	S	0	0.0	0:00.00	ksoftirqd/0
4	root	RT	0	0	0	0	S	0	0.0	0:00.00	watchdog/0
5	root	RT	0	0	0	0	S	0	0.0	0:00.00	migration/1
6	root	34	19	0	0	0	S	0	0.0	0:00.00	ksoftirqd/1
7	root	RT	0	0	0	0	S	0	0.0	0:00.00	watchdog/1
8	root	10	-5	0	0	0	S	0	0.0	0:00.00	events/0
9	root	10	-5	0	0	0	S	0	0.0	0:00.00	events/1
10	root	10	-5	0	0	0	S	0	0.0	0:00.01	khelper
11	root	10	-5	0	0	0	S	0	0.0	0:00.00	kthread
15	root	10	-5	0	0	0	S	0	0.0	0:00.00	kblockd/0
16	root	10	-5	0	0	0	S	0	0.0	0:00.00	kblockd/1
17	root	16	-5	0	0	0	S	0	0.0	0:00.00	kacpid
109	root	16	-5	0	0	0	S	0	0.0	0:00.00	cqueue/0
110	root	16	-5	0	0	0	S	0	0.0	0:00.00	cqueue/1
113	root	10	-5	0	0	0	S	0	0.0	0:00.00	khubd
115	root	10	-5	0	0	0	S	0	0.0	0:00.00	kseriod
182	root	21	0	0	0	0	S	0	0.0	0:00.00	pdflush
183	root	15	0	0	0	0	S	0	0.0	0:00.02	pdflush
184	root	17	-5	0	0	0	S	0	0.0	0:00.00	kswapd0
185	root	17	-5	0	0	0	S	0	0.0	0:00.00	aio/0
186	root	17	-5	0	0	0	S	0	0.0	0:00.00	aio/1
352	root	11	-5	0	0	0	S	0	0.0	0:00.00	kpsmouse

Case 2:

1 call in 5s, after 40 m, the sip calls failed.

```
top - 14:12:09 up 3:15, 2 users, load average: 0.79, 0.79, 0.52
Tasks: 77 total, 1 running, 75 sleeping, 1 stopped, 0 zombie
Cpu(s): 30.7%us, 4.1%sy, 0.0%ni, 58.9%id, 0.0%wa, 2.2%hi, 4.1%si, 0.0%st
Mem: 505656k total, 268956k used, 236700k free, 30792k buffers
Swap: 1015800k total, 0k used, 1015800k free, 164544k cached
```

PID	USER	PR	NI	VIRT	RES	SHR	S	%CPU	%MEM	TIME+	COMMAND
2402	root	15	0	58520	23m	5548	S	76	4.7	21:16.11	asterisk
2579	root	15	0	9148	2732	2200	S	1	0.5	0:09.27	sshd
2614	root	15	0	9152	2732	2200	S	0	0.5	0:02.73	sshd
4552	root	15	0	2172	884	702	D	0	0.2	0:14.02	top

```
CA start sipp - sipp -sf uac_pcap.xml -s 2005 192.168.2.108 -r 1 -rp 15... - 5 X
```

Start Time	Last Reset Time	Current Time
2008-10-28 13:37:31:390	2008-10-28 14:22:51:686	2008-10-28 14:22:52:681
1225172251.390625	1225174971.686625	1225174972.681625

Counter Name	Periodic value	Cumulative value
Elapsed Time	00:00:00:995	00:45:21:291
Call Rate	4.020 cps	1.834 cps
Incoming call created	0	0
OutGoing call created	4	4990
Total Call created		4990
Current Call	33	
Successful call	4	4935
Failed call	0	22

Response Time 1
Call Length

15s one call, increased by 1 each 50s, duration 45M

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

Case 3:

5 calls in 10s

```
top - 20:49:03 up 6:59, 4 users, load average: 0.11, 0.82, 2.64
Tasks: 82 total, 1 running, 81 sleeping, 0 stopped, 0 zombie
Cpu(s): 8.5%us, 1.0%sy, 0.0%ni, 87.9%id, 0.0%wa, 2.2%hi, 0.5%si, 0.0%st
Mem: 505656k total, 307092k used, 198564k free, 46040k buffers
Swap: 1015800k total, 0k used, 1015800k free, 167304k cached

  PID USER      PR  NI  VIRT  RES  SHR  S  %CPU  %MEM    TIME+  COMMAND
 2404 root        15   0 86952  40m 5556  S   19   8.2  33:38.27 asterisk
25779 root        15   0 2172 1008  804  R    1   0.2   0:01.12 top
    1 root        15   0 2040  632  544  S    0   0.1   0:00.64 init

ca start sipp -sipp -sf uac_pcap.xml -s 2005 192.168.2.108 -r 5 -rp... - - x
Start Time           | 2008-10-27 20:57:47:429 | 1225112267.429000
Last Reset Time      | 2008-10-27 20:59:48:559 | 1225112388.559000
Current Time         | 2008-10-27 20:59:49:554 | 1225112389.554000
-----+-----+-----
Counter Name         | Periodic value          | Cumulative value
-----+-----+-----
Elapsed Time         | 00:00:00:995           | 00:02:02:125
Call Rate            | 1.005 cps               | 0.499 cps
-----+-----+-----
Incoming call created | 0                       | 0
OutGoing call created | 1                       | 61
Total Call created   |                         | 61
Current Call         | 5                       |
-----+-----+-----
Successful call       | 0                       | 56
Failed call           | 0                       | 0
-----+-----+-----
Response Time 1      | 00:00:00:001           | 00:00:00:016
Call Length          | 00:00:00:000           | 00:00:09:021
-----+-----+-----
[+|-!*|/]: Adjust rate  --- [q]: Soft exit  --- [p]: Pause traffic  -----
```

Case 4:

Realtek RTL8169/8110 Family Gigabit Ethernet NIC (Microsoft's Packet Scheduler) : Capturing

Filter: `rtp.p_type==18`

No.	Time	Source	Destination	Protocol	Info
451894	759.638726	192.168.2.111	192.168.2.108	RTP	Payload type=ITU-T G.729, SSRC=2718015020, Seq=2485, Time=2234
451895	759.638764	192.168.2.111	192.168.2.108	RTP	Payload type=ITU-T G.729, SSRC=2718015020, Seq=2486, Time=2234
451898	759.642421	192.168.2.111	192.168.2.108	RTP	Payload type=ITU-T G.729, SSRC=2718015020, Seq=1369, Time=1340
451902	759.643401	192.168.2.111	192.168.2.108	RTP	Payload type=ITU-T G.729, SSRC=305777981, Seq=1887, Time=1908
451906	759.644361	192.168.2.111	192.168.2.108	RTP	Payload type=ITU-T G.729, SSRC=2718015020, Seq=1370, Time=1341
451911	759.644502	192.168.2.111	192.168.2.108	RTP	Payload type=ITU-T G.729, SSRC=2718015020, Seq=1371, Time=1342
451912	759.644557	192.168.2.111	192.168.2.108	RTP	Payload type=ITU-T G.729, SSRC=2718015020, Seq=1372, Time=1343
451915	759.645486	192.168.2.111	192.168.2.108	RTP	Payload type=ITU-T G.729, SSRC=305777981, Seq=1888, Time=1909
451916	759.645542	192.168.2.111	192.168.2.108	RTP	Payload type=ITU-T G.729, SSRC=305777981, Seq=1889, Time=1910
451917	759.645583	192.168.2.111	192.168.2.108	RTP	Payload type=ITU-T G.729, SSRC=305777981, Seq=1890, Time=1911
451936	759.648282	192.168.2.111	192.168.2.108	RTP	Payload type=ITU-T G.729, SSRC=2718015020, Seq=1087, Time=1095
451938	759.648371	192.168.2.111	192.168.2.108	RTP	Payload type=ITU-T G.729, SSRC=2718015020, Seq=1068, Time=1100
451941	759.649374	192.168.2.111	192.168.2.108	RTP	Payload type=ITU-T G.729, SSRC=305777981, Seq=777, Time=10208
451942	759.649416	192.168.2.111	192.168.2.108	RTP	Payload type=ITU-T G.729, SSRC=305777981, Seq=778, Time=10216

Frame 3166935 (64 bytes on wire, 64 bytes captured)

- Ethernet II, Src: Asiarock.fid:9c:26 (00:13:8f:fd:9c:26), Dst: 00:1c:c0:6f:5d:5d (00:1c:c0:6f:5d:5d)
- Internet Protocol, Src: 192.168.2.111 (192.168.2.111), Dst: 192.168.2.108 (192.168.2.108)
- User Datagram Protocol, Src Port: 27600 (27600), Dst Port: 20536 (20536)
- Real-Time Transport Protocol
 - [Stream setup by SDP (frame 2227130)]
 - 10.. = Version: RFC 1889 version (2)
 - ..0. = Padding: False
 - ...0 = Extension: False
 - 0000 = Contributing source identifiers count: 0
 - 0... = Marker: False
 - Payload type: ITU-T G.729 (18)
 - Sequence number: 1634
 - Timestamp: 170640

```
top - 13:18:12 up 4:05, 2 users, load average: 0.09, 0.14, 0.09
Tasks: 76 total, 1 running, 75 sleeping, 0 stopped, 0 zombie
Cpu(s): 4.3%us, 0.5%sy, 0.0%ni, 92.0%id, 0.0%wa, 2.0%hi, 1.2%si, 0.0%st
Mem: 505656k total, 274048k used, 231608k free, 38328k buffers
Swap: 1015800k total, 0k used, 1015800k free, 167260k cached
```

PID	USER	PR	NI	VIRT	RES	SHR	S	%CPU	%MEM	TIME+	COMMAND
2413	root	16	0	25696	10m	5548	S	11	2.1	2:55.29	asterisk
2701	root	15	0	8996	2736	2212	S	0	0.5	0:01.34	sshd
3484	root	15	0	2168	988	792	R	0	0.2	0:17.26	top
1	root	15	0	2040	636	544	S	0	0.1	0:00.64	init
2	root	RT	0	0	0	0	S	0	0.0	0:00.00	migration/0

```
start sipp -sipp -sf uac_pcap.xml -s 2005 192.168.2.108 -r 10 -rp 2...
```

Counter Name	Periodic value	Cumulative value
Elapsed Time	00:00:00.996	00:15:00.083
Call Rate	1.004 cps	0.500 cps
Incoming call created	0	0
OutGoing call created	1	450
Total Call created	1	450
Current Call	5	
Successful call	1	445
Failed call	0	0
Response Time 1	00:00:00.002	
Call Length	00:00:10.005	

10 calls in 20 s traffic

```

-- Executing [2005@internal:1] Answer("SIP/sipp-b7d0da18", "") in new stack
-- Executing [2005@internal:2] Dial("SIP/sipp-b7d0da18", "SIP/1000@192.168.2.127|80|r") in new
-- Called 1000@192.168.2.127
-- SIP/192.168.2.127-09d110f0 answered SIP/sipp-b7d0da18
ogon*CLI> sip show channels
Peer          User/ANR      Call ID      Seq (Tx/Rx)  Format          Hold    Last Message
192.168.2.127 1000         40251edf272 00102/00000 0x80004 (ulaw|h) No      Tx: ACK
192.168.2.111 sipp        2-608@192.1 00101/00001 0x100 (g729)  No      Rx: ACK
192.168.2.127 1000         398774836a5 00102/00000 0x80004 (ulaw|h) No      Tx: ACK
192.168.2.111 sipp        1-608@192.1 00101/00001 0x100 (g729)  No      Rx: ACK
192.168.2.127 (None)      07c6bb5568e 00101/00102 0x0 (nothing) No      Rx: OPTIONS
192.168.2.127 1000         39d177673b3 00102/00000 0x80004 (ulaw|h) No      Tx: ACK
192.168.2.111 sipp        453-2420@19 00101/00001 0x100 (g729)  No      Rx: ACK
192.168.2.127 1000         3b2b9d1a43f 00102/00000 0x80004 (ulaw|h) No      Tx: ACK
192.168.2.111 sipp        452-2420@19 00101/00001 0x100 (g729)  No      Rx: ACK
192.168.2.127 1000         330091333a1 00102/00000 0x80004 (ulaw|h) No      Tx: ACK
192.168.2.111 sipp        451-2420@19 00101/00001 0x100 (g729)  No      Rx: ACK
192.168.2.127 1000         08ce67ad11a 00102/00000 0x80004 (ulaw|h) No      Tx: ACK
192.168.2.111 sipp        450-2420@19 00101/00001 0x100 (g729)  No      Rx: ACK
192.168.2.127 1000         73ed3f397a6 00102/00000 0x80004 (ulaw|h) No      Tx: ACK
192.168.2.111 sipp        449-2420@19 00101/00001 0x100 (g729)  No      Rx: ACK
15 active SIP channels
-- Executing [2005@internal:1] Answer("SIP/sipp-b7d25538", "") in new stack
-- Executing [2005@internal:2] Dial("SIP/sipp-b7d25538", "SIP/1000@192.168.2.127|80|r") in new

```

CPU information:

```

[root@new-host ~]# cat /proc/cpuinfo
processor       : 0
vendor_id     : GenuineIntel
cpu family    : 6
model         : 28
model name    : Intel(R) Atom(TM) CPU 230  @ 1.60GHz
stepping      : 2
cpu MHz      : 1596.208
cache size   : 512 KB
physical id   : 0
siblings     : 2
core id      : 0
cpu cores    : 1
fdiv_bug     : no
hlt_bug      : no
f00f_bug     : no
coma_bug     : no
fpu          : yes
fpu_exception : yes
cpuid level  : 10
wp           : yes
flags        : fpu vme de pse tsc msr pae mce cx8 apic mtrr
t_tsc pni monitor ds_cpl tm2 cx16 xtpr lahf_lm
bogomips     : 3257.81

processor       : 1
vendor_id     : GenuineIntel
cpu family    : 6
model         : 28
model name    : Intel(R) Atom(TM) CPU 230  @ 1.60GHz
stepping      : 2
cpu MHz      : 1596.208
cache size   : 512 KB
physical id   : 0
siblings     : 2
core id      : 0
cpu cores    : 1
fdiv_bug     : no
hlt_bug      : no
f00f_bug     : no
coma_bug     : no
fpu          : yes
fpu_exception : yes
cpuid level  : 10
wp           : yes
flags        : fpu vme de pse tsc msr pae mce cx8 apic mtrr
t_tsc pni monitor ds_cpl tm2 cx16 xtpr lahf_lm
bogomips     : 3191.77

[root@new-host ~]#

```


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Test environments:

Cnetos-5.0

Intel Atom 230 CPU

Tools: Sipp-3.1, tcpdump and Wireshark

Asterisk-1.4.21