A Practical Analysis of Asterisk SIP Server Performance

Razvan Rughinis
Politehnica University of Bucharest
Romania
Email: razvan.rughinis@cs.pub.ro

Cristian Iconaru
RoEduNet
Romania
Email: cristian.iconaru@roedu.net

Abstract

This paper presents an analysis of the Asterisk server performance as a SIP server, as well as of bandwidth consumption in multiple scenarios. Server load has been tested using a certain type of codec, which requires a voice quality close to classic telephony (PSTN). Therefore, one can infer minimum VoIP requirements for an Asterisk SIP server by analyzing the maximum number of possible calls and the maximum bandwidth.

Keywords: IP telephony, Asterisk, SIP, SIPp, IAX.

1. Introduction

With the development of VoIP telephony, the Asterisk servers have become common in personal and organizational environments as well. Besides relying on the benefits of open source technology, Asterisk servers have also proven to be resilient and to function well over heterogeneous architectures [1]. Still, any fast expanding Internet technology becomes vulnerable to security threats [2, 3] and to increasing demands for quality of service. One of the main limits for addressing these concerns derives from the relatively high bandwidth consumption of VoIP, since bandwidth requirements as well as processor load increase significantly when measures for providing Confidentiality, Integrity and Availability are put into place [4].

Given the importance of low resource consumption for future improvements in the use of Asterisk servers, this paper attempts a performance evaluation of an Asterisk server, focusing on bandwidth consumption and processor load under several utilization scenarios.

2. Performance evaluation

We have analyzed the performance of an Asterisk server in multiple scenarios of use as an SIP server. The laboratory setup consisted of the following equipments: two PCs Pentium(R) D CPU 2.8GHz 1GB RAM, two NICs 1Gbps, and one switch 3560g. The following pieces of software were deployed: Asterisk PBX v1.4.18, Debian 4.0 / kernel 2.6.18-6-686, Microsoft Windows XP [Version 5.1.2600], SIPp v3.1, X-Lite v3.0.

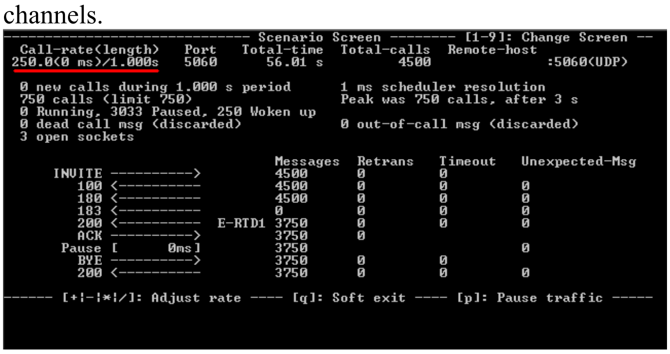
2.1. General bandwidth constraints

ITU-T G.711, referred to as PCM, is the most frequently used codec in present-day telephony system. There are two variants of this codification method: µ-law, used in the US and in Japan, and a-law in the other countries. Each of these methods transmits 8000 samples per second – which means that a total of 64000 bits may be transmitted per second.

Therefore, in order to digitally transmit voice, the G.711 codec uses 64 Kbps of the available bandwidth. In real usage there is an overhead load besides voice traffic, which leads to a real bandwidth usage of 87.2 Kbps. This bandwidth is used in both directions. This means that a minimum 128 Kbps canal is required for this codec to function normally.

2.2. External connection bandwidth constraints for the maximum call rate

An Asterisk server similar to the one used in the laboratory configuration may support up to 250 simultaneous calls, equivalent to 500 simultaneous channels.



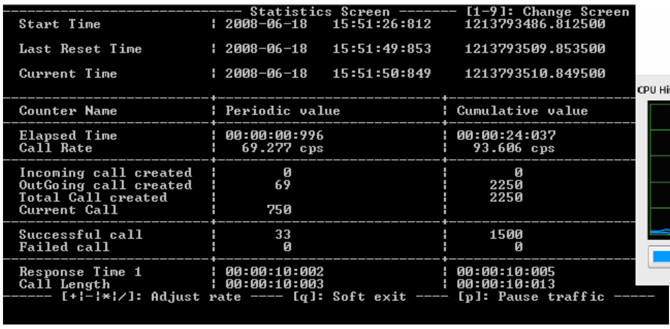


Figure 1. Details regarding the Asterisk server

A single call creates two channels, but one may use more, according to the Dial Plan logic. The above discussed scenario requires a 43,600 Kbps bandwidth, since each channel requires a 87.2 Kbps bandwidth.

If one uses a T3 connection (44,736 Kbps) this means that around 256 calls can be initiated, given the availability of 513 channels. Therefore, one Asterisk server is enough to saturate such a connection.

A significant processor load is noticeable in the IVR activation periods, for 250 calls per second, for the given Dial Plan.

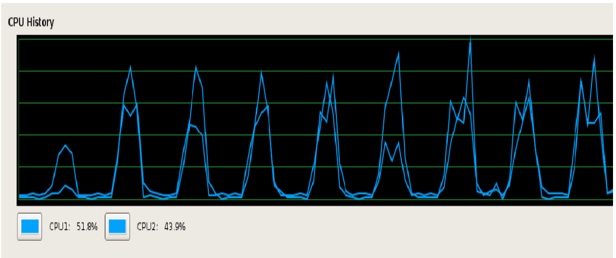


Figure 2. Processor load for 250 calls/second

The load is lower when calls are effectively initiated by the server.

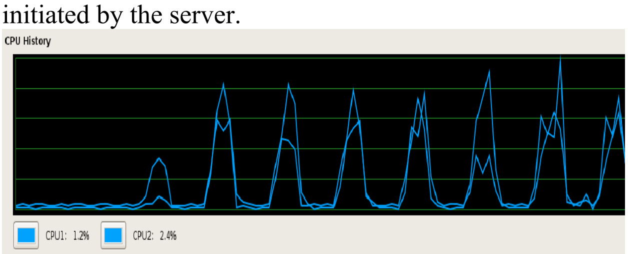


Figure 3. Processor load for 250 calls/second if calls are initiated by the server

The spikes in the figures below illustrate IVR activation (voice transmission), while lower segments represent the de facto call initiation.

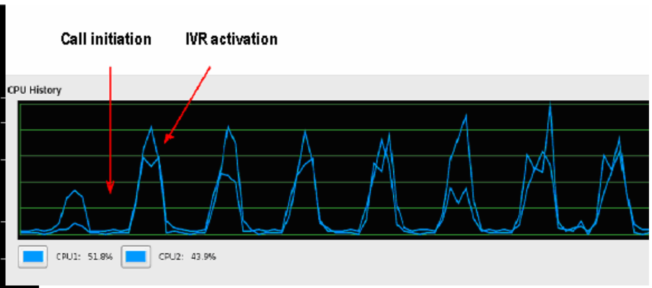


Figure 4. Spikes in processor load for 250 calls/second

Tests indicate a large number of large size SIP packets and an overall traffic of 16 Mbps.

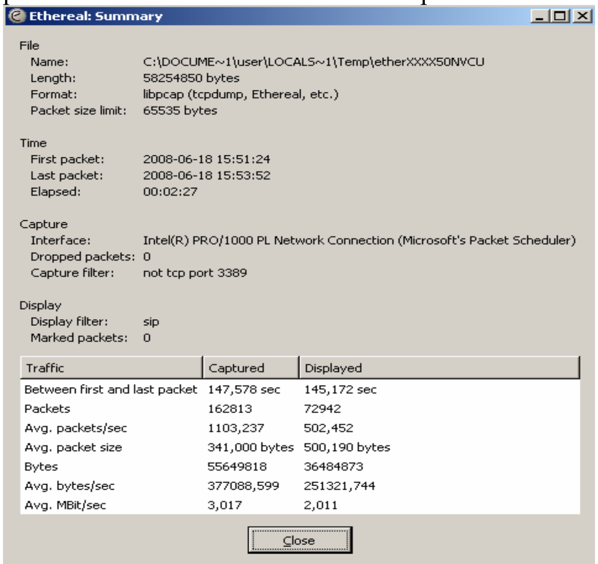


Figure 5. Test information for 250 calls / second

Figure 6 illustrates results for real voice traffic (RTP). One can see a large number of voice packets, with an overall traffic of around 8 Mbps.

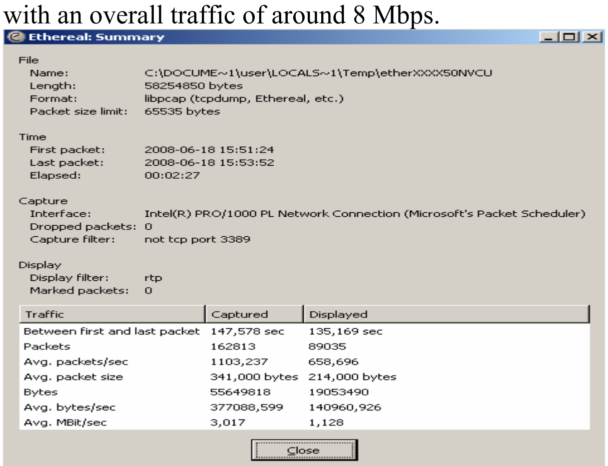


Figure 6. Test results for real voice traffic (RTP) for 250 packets/second

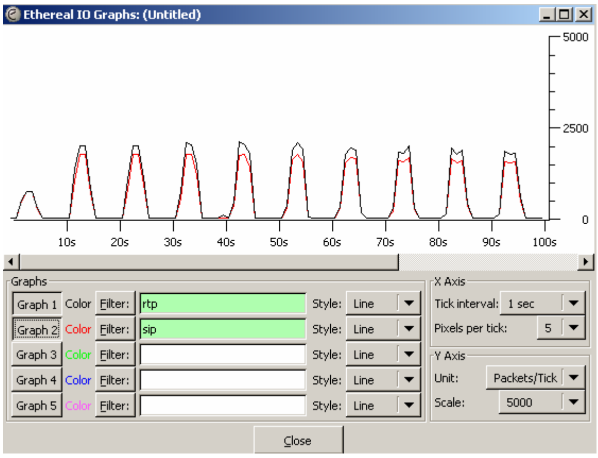


Figure 7. Spikes for IVR activation and call initiation

Figure 7 illustrates spikes for IVR activation and for call initiation, analogously to Figure 4 for processor load.

In order to obtain 250 calls per second we used two Pentium(R) systems D CPU 2.8GHz 1GB RAM, NIC 1Gbps. The Asterisk server used only the G.711 codec without echo supprimation.

The Asterisk configuration is detailed below:

```
Extensions.conf
[default]
exten => 2005,1,Ringing;          //ring
exten => 2005,2,Wait(10);         //wait 10 sec
exten => 2005,3,Playback(vm-nobodyavail);
        //after 10 sec IVR
exten => 2005,4,Playback(vm-goodbye);
exten => 2005,5,Hangup;

exten => 2006,1,Dial(SIP/user1,10);
exten => 2006,2,Playback(vm-nobodyavail);
exten => 2006,3,Playback(vm-goodbye);
exten => 2006,4,Hangup;

exten => 2007,1,Dial(SIP/user2,10);
exten => 2007,2,Playback(vm-nobodyavail);
exten => 2007,3,Playback(vm-goodbye);
exten => 2007,4,Hangup;

sip.conf
[user1]
username=user1
secret=1234
callerid=User1
type=friend
context=default
host=dynamic
disallow=all
allow=alaw

[user2]
username=user2
secret=1234
callerid=User2
type=friend
context=default
host=dynamic
disallow=all
allow=alaw
```

2.3. Performance for lower call rates

For relatively small numbers of simultaneous calls, the situation is completely different. For 10 calls / second the processor load is low, up to 3%.

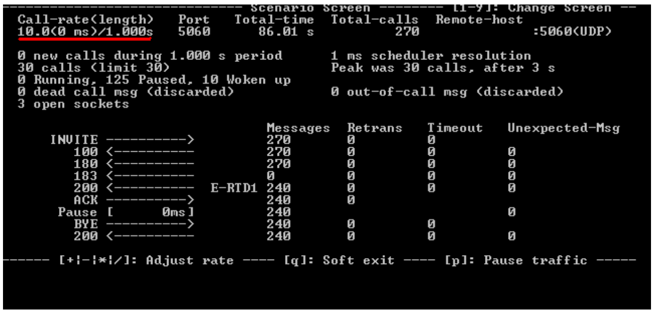


Figure 8. Server information for lower call rates

The two spikes are generated by recording the two software calls.

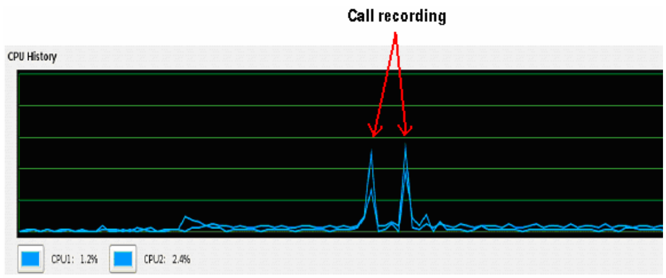


Figure 9. Spikes in processor load for lower call rates

The number of RTP packets is low, and they have the standard 214 byte size.

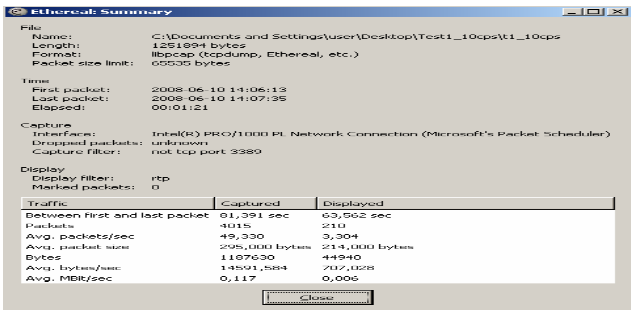


Figure 10. Test information for lower call rates

The system load does not increase significantly for a higher rate of calls per second, such as 50 or 100 calls/second. For 50 calls per second processor load reaches a maximum of 5%.

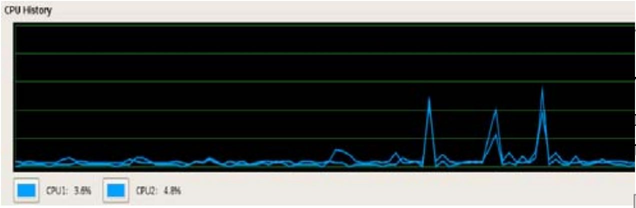


Figure 11. Processor load for 50 calls/second

For 100 calls / second the processor load reaches at most 7%.

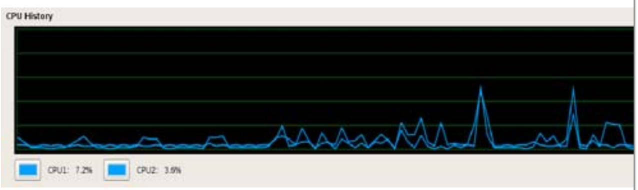


Figure 12. Processor load for 100 calls/second

2.4. Performance for excessive call rates

If the limit of 250 calls/sec is exceeded, such as attempting to initiate 300 calls/second, certain SIP requests cannot be supported and this leads to retransmissions, while the calls are lost. For call rates of 300 calls/second and above, retransmissions are initiated after 12 seconds.

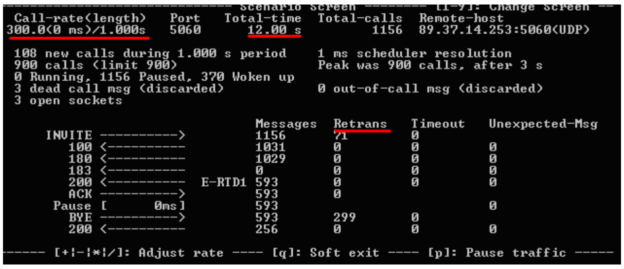


Figure 13. Server information for call rates of 300 calls/second and above

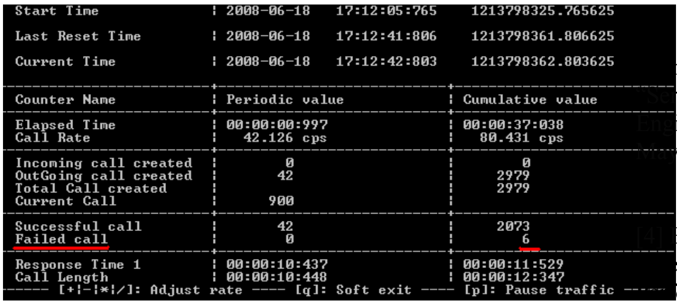


Figure 14. Call failure for high call rates

3. Conclusions

There is a non-linear relationship between the call rate and the processor load for the Asterisk server in the given configuration.

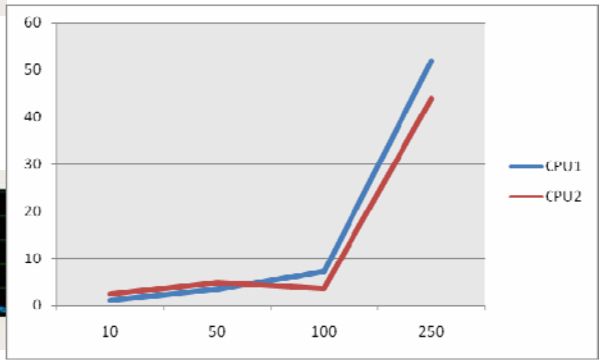


Figure 15. Relationship between call rate and CPU load

Therefore, it is possible that security policies that require significant resources may be accommodated by a relatively slight decline in the call rate, avoiding the maximum levels.

4. References

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