Modeling of a VoIP Server Based on the Number of Call

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ABSTRACT: *In this paper, we will discuss the experience of developing projects based upon a central real world example to introduce first year students to engineering. We will outline how the theme of two dimensional barcodes was used to introduce our students in the freshmen Foundations of Engineering I (ENGR111) course to the key skills of engineering design, problem solving, teamwork and computer programming. Results show that the students found the experience positive.*

Keywords: VoIP, SIP, RAM, CPU, Bandwidth

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1. Introduction

According to research done by Pablo Montoro and Eduardo casilari [1]. We will look at another aspect of their results and interpreted them in our own way for the simple reason that the Voice over Internet Protocol (VoIP) is the only technology that dominates Internet use those last decade. In the business world or in the computer world this technology shows us its future for its ease of use and simplicity in its deployment but especially for its cost. Migration to VoIP does not require additional infrastructure while for the traditional phone and mobile phone with a new technology requires new investments for its deployment. The great progress in research on the standard protocols of VoIP leads us to the migration to all IP. [2] [3] [4].

Many searches are conducted on many of the VoIP standard protocol [1], and for our case and for the considerable progress in its development but also for the license used for its operation, we chose the SIP [5] [6] [7] [8] [9]. The choice of the use of a Asterisk server is also the same as that of SIP is to say its evolution of its obvious license (LPG) [10] [11] [12].

The problem that then arises is what can we model an asterisk server usi ng the SIP protocol in order to predict its performance?

For this reason we decided to make a "*mathematical modeling of a VoIP server*"

This paper is structured as follows: Section 2 summarizes the results of previous research around VoIP technology and the results that we have taken as the basis of our research. Section 3 describes how we used to arrive at our model. The results and our model are discussed in Section 4 we then propose a mathematical model to conclude our work.

2. State of Art

In the article "*a comparative study of standard Voip with asterisk*", the authors do a lot of different research-based protocol used in VoIP. This experiment is as follows:

The block diagram of the system deployed for the study, shown in Figure 1, comprises three components: an Asterisk server (the unit under observation), a call generator (for calls to be sent to asterisk), and a Server Monitor (that will monitor the Asterisk server). Figure 2 depicts the interconnection of these three elements: the call generator and the Asterisk server are connected by two different Ethernet interfaces (eth0 and eth1). The calls will initially be transmitted via eth1 link, which supports the VoIP flow. The monitoring equipment is connected to the same switch as the eth0 link the other two units. For the Asterisk server, they installed a Debian GNU Linux system on a Pentium 4 (2'4 GHz) with 1GB of RAM memory. Asterisk 1.4 has been installed from the Debian packages, using a standard configuration [2].

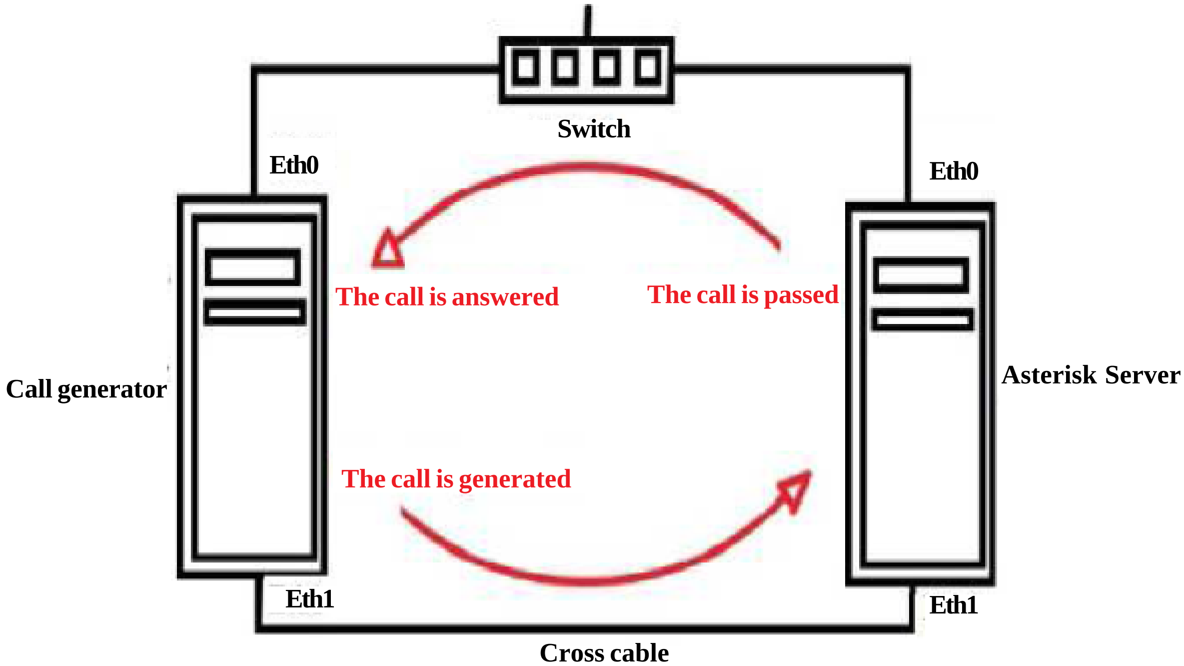

Figure 1. Call generator

For the result we took only the result that uses the G711 codec and SIP [13] and this result is in Table 1.

Nomber of calls	Processor Usage (%)	Memory usage (Mbytes)	Bandwidth usage (Kbps)
10	2,36	33,38	86,05
20	4,64	38,23	86,00
30	7,12	42,09	86,00

Table 1. Result of Experience

3. Description Calculates Correlation

Study the correlation between two or more random variables or numerical statistics, this study is the intensity of the connection that may exist between these variables. In our case we have three four variable ie the number of calls, CPU usage, memory usage and finally the use of bandwidth. The connection is an affine relation sought.

A measure of this correlation is obtained by calculating the linear correlation coefficient. This coefficient is the ratio of their covariance and nonzero product of their standard deviations. The correlation coefficient is between -1 and 1

$$R(x,y) = \frac{cov(x,y)}{ecartype\ x * ecartype\ y} \quad (1)$$

$$cov(x,y) = \frac{1}{N}\sum_{n=1}^{n} x_i y_i - \overline{x}\,\overline{y} \quad (2)$$

If $R(x, y) \sim \pm 1$ then x, y are correlated

If $R(x, y) \sim 0$ then x, y are not correlated

In our case we took as a variable

X = number of calls

Y = CPU usage

Z = use RAM

T = bandwidth use of

Thus we obtained:

$R(x,y) = 0.99970589 \simeq 1$

$R(x,z) = 1\ 099\ 785\ 374 \simeq 1$

$R(x,t) = -1\ -0.8660254 \simeq -1$

4. Comment on Results

We have found that there is really a relationship between the number of appeal and the variation of resource use Asterisk server hardware.

This is from cella we will derive a mathematical model based on the previous result that is to say, by defining an equation that can represented mathematically formulating the call that passes at the server according to these material resources.

5. Mathematical Model

To do so we took the method of least squares to find out if there is a correlation between two variable there will be an adjustment as a right and that right has the form y = ax + b with

$$a = \frac{\sum x\,y - n\,\overline{x} - \overline{y}}{\sum x^2 - n\,\overline{x}^{\,2}} \quad (3)$$

And

$$b = \overline{y} - a\,\overline{x} \quad (4)$$

According to (1), (2), (3), (4)

Thus we have obtained

$$Y = 0{,}238x - 0{,}05333 \quad (5)$$

$$Z = 0{,}4355x + 29{,}19 \quad (6)$$

$$T = -0{,}0025x + 86{,}066667 \quad (7)$$

As y, z and t are not summable so this equation is the number of calls based on a material parameter asterisk server or more precisely the ideal model to determine the number of customers possible if we fix y or z or t or yet the size of memory required if one has a number of customer or the clock frequency required for a processor in relation to the number of customer yet.

We will therefore seek the equation of our server for the number of call additionant (1) + (2) + (3)

Then

$$x = \frac{y + z + t - 115{,}2027}{0{,}671}$$

And after verification we obtained the following table

Nomber of calls	Processor Usage (%)	Memory usage (Mbytes)	Bandwidth usage (Kbps)
10	2,32667	33,545	86,04
20	4,706764	37,9	86,01
30	7,08	42,255	85,99

Figure 2. Result of equation (8)

From this table we could see that the result of our model resembles approximately the same reality this we have deduced that can calculate the size of the power of our memory and CPU utilization of bandwidth a given Asterisk server if you know the number of client and back that is to say, knowing the capacity of our server hardware we can deduce the number of possible customer you can assign to this server.

6. Conclusion

So to conclude we can say that we were able to extract a mathematical model calls in a VoIP server based on resource use materials that are the characteristics of a server.

We calculate the correlation we found that obviously there was a relationship between the number of call and CPU usage, there is a relationship between memory usage and the number of calls and between the latter and the bandwidth using a low hardware limits the number of users to an Asterisk server that is to say that an operator who intends to have the client limit the use of a powerful engine will be a ideal. However, we found that in terms of bandwidth variation of the value of the bandwidth is almost insignificant compared to the variation in number of calls that it proves that VoIP is one of the transiting the Internet without obstructing the other protocols.

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