Investigate common work of software phone systems in virtual environments and real switching systems

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Abstract - **This paper discusses the common work of Linux - based IP phone systems Trixbox and Elastix, build in virtual environments. Their interconnection with real switching systems is applied. Advantages of using virtualization software in communications are shown using VoIP software analyzators. Experiments are made, revealing the advantages of using VoIP telephony and virtualization, leading to much easier maintenance and more flexible services.**

Keywords – **Voice-over-IP, Virtualization, Phone systems**

I. Introduction

Transmission of data and voice via VoIP is extremely attractive to business users, service operators and for home users because it allows Internet and data networks, already established in offices[1], enterprises and administrative areas, to transmit voice calls, video conferences to support and other real-time applications.

VoIP telephone applications such as Asterisk-based Trixbox can be uploaded to a virtual environment, which means reducing the number of physical machines and leads to less power consumption, simplifying IT infrastructure, a much easier maintenance and much greater service flexibility [2].
Installation of multiple virtual environments (operating systems and applications) in one physical server (homogeneous hardware) is more economically reasonable than provision and maintenance of each physical server for each application [3].
Figure 1.1. shows the qualitative leap made in the period 2007 to 2009, when the total number of installed physical servers stops to increase and even decreases at the expense of the rapid growth of virtual machines. In the U.S. over the next 2-3 years the number of installed physical servers will be reduced by 20%.

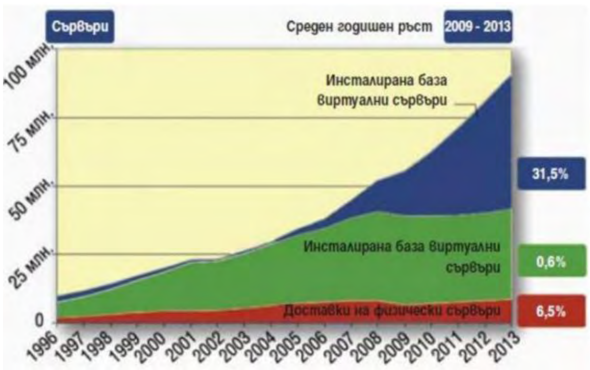


Fig.1.1. Increasing use of virtual servers

In practice, only 10% of servers capacity in a continuous mode is used. A large data center could save electricity equivalent to the consumption of 200,000 households under loading up to 50% of its capacity. A serious problem for the ecological balance is large carbon dioxide emissions. According to calculations, covering 50.000 data centers, emissions will exceed 10 million tons by 2013. These figures leave no doubt that information technology bear their share of responsibility for environmental protection [4].
Optimized use and better distribution of work is achieved By application virtualization, which in turn leads to fewer servers and to shorten the prolonged periods in which they operate without load. Integrated approach in the management of information technology is beneficial not only for climate but also for budget of the companies. It enables them to reduce hardware investments and make large cost savings for electricity [5].

Two experiments are made to examine the virtualization. Telephone software systems Trixbox and Elastix are used as well as softphones and VoIP analizators.

II. Experimental results

Virtual TRIXBOX and ELASTIX are loaded in the virtual machine.

Different software phone systems TRIXBOX and ELASTIX are used when conducting the experiments. Some of them are in virtual environment, while the rest are real working (not in virtual environment). For the purposes of the study some other phone systems can be used such as 3CX and different hardware switching systems. Test computers have 1GB RAM operating memory and processor Intel Pentium 4. Emulating virtual programs can be VMWare products (Workstation, VMWare Player), Microsoft products, Oracle (Virtual Box) etc.

2.1. Virtual Trixbox - Virtual Elastix

At the start of the call as well as at its termination, SIP signaling protocol is used.

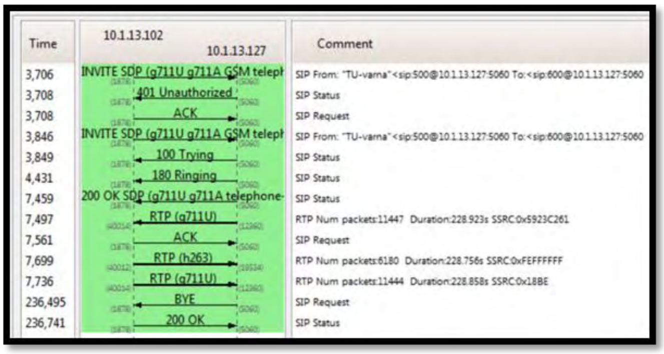


Fig.2.1. Signalling on call

Figure 2.2. shows the number of packets (axis Y) per unit time for 120 seconds. The number of RTP packets vary over time because the length of each packet is not the same and for one unit of time different number of packets can be transmitted.

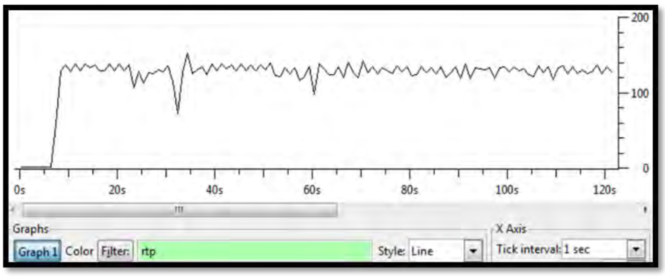


Fig.2.2. Distribution of RTP packets per unit time - IO Graphs

In fig. 2.3. with black the values of jitter are shown, the average is about 3ms. In red delays between packets are marked.

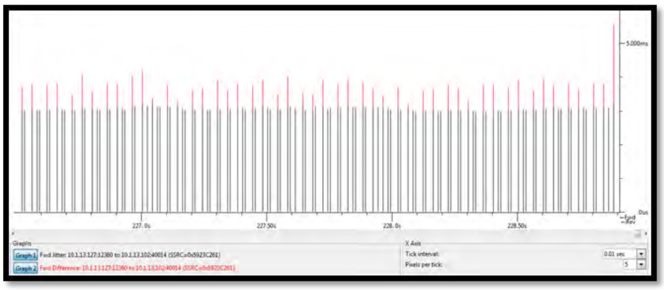


Fig.2.3. Jitter and delay of packets in graphical form

Fig. 2.4 shows service quality of the audio signal (Audio QoS). Reports for 30 seconds back have been noted. R factor (R Factor) is within the allowable (ranging from 0 to 120). Packet loss in unusual circumstances (Burst Packet Loss Rate%) is 0.102% at source and destination at 0.055%. Rejected packs (Discarded Packets) are respectively 2.105 and 26.

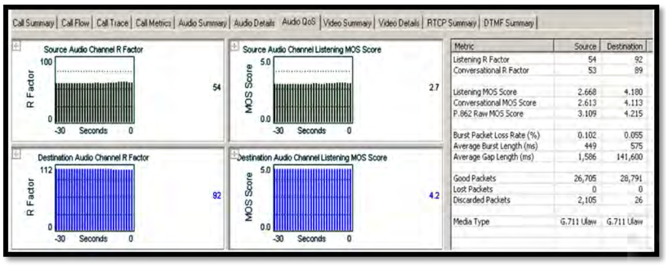


Fig.2.4. Quality of Service audio

2.2 Virtual Trixbox - real Trixbox

In this experiment, TRIXBOX is loaded in the virtual machine and real built PBX server TRIXBOX is used.

The graph in Fig. 2.5 shows the number of RTP packets per unit time. Again some hesitation is noticed, because the length of each packet is not the same and therefore requires much time to transmit if more bytes are used.

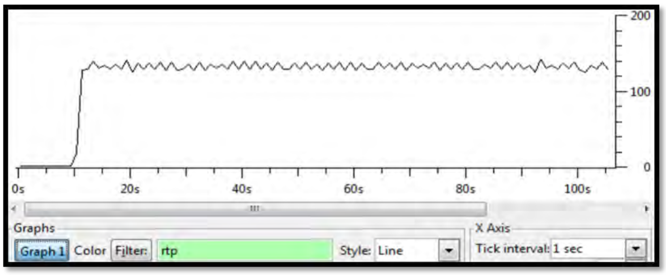


Fig.2.5. Distribution of RTP packets per unit time

The values of jitter (black) and the delay between packets (in red) have improved by about 2ms (Fig. 2.6) compared to the first investigation.

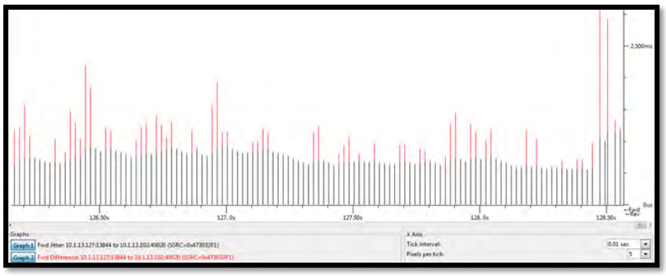


Fig.2.6. Jitter and delay of packets in graphical form

Below is shown the total activity of the network - network activity, VoIP activity, conducted conversations, provided packet activity, protocol activity. Everything is within normal limits.

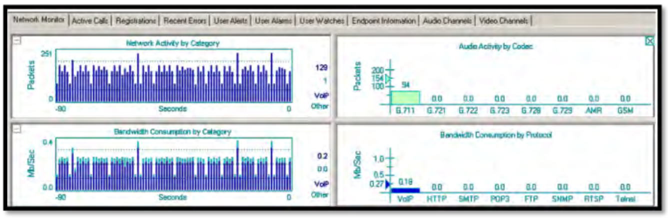


Fig.2.7. Network Monitor

The main parameters that are subject to change in a better direction are shown in fig.2.8 and fig.2.9.

Source R Factor is improved by 10 units and MOS score by 0.5 as the result become almost 3.1. Rejected packets and the loss possibility are less than previous experiment. The average jitter value is significantly reduced in the destination as it becomes 0,415 ms, and in sourse - 3,5 ms. The average delay between packets is again about 20ms.

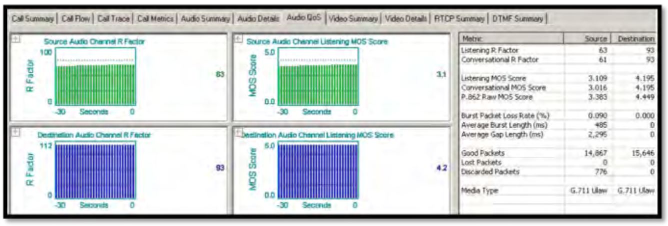
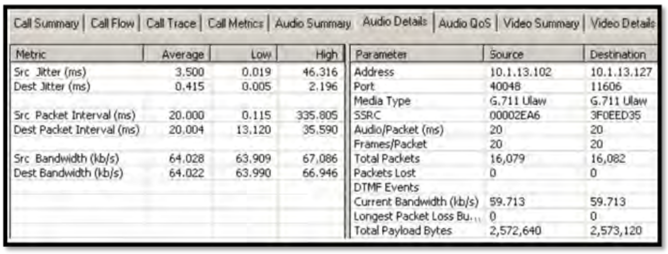


Fig.2.8. Quality of Service audio



Call Summary | Call Flow | Call Trace | Call Metrics | Audio Summary | Audio Details | Audio QoS | Video Summary | Video Details

Metric	Average	Low	High
Src Jitter (ms)	3.500	0.019	46.316
Dest Jitter (ms)	0.415	0.005	2.196
Src Packet Interval (ms)	20.000	0.115	335.805
Dest Packet Interval (ms)	20.004	13.120	35.590
Src Bandwidth (kb/s)	64.028	63.909	67.086
Dest Bandwidth (kb/s)	64.022	63.990	66.946

Parameter	Source	Destination
Address	10.1.13.102	10.1.13.127
Port	40048	11606
Media Type	G.711 Ulaw	G.711 Ulaw
SSRC	00002EA6	3F0EED35
Audio/Packet (ms)	20	20
Frames/Packet	20	20
Total Packets	16,079	16,082
Packets Lost	0	0
DTMF Events		
Current Bandwidth (kb/s)	59.713	59.713
Longest Packet Loss Bu...	0	0
Total Payload Bytes	2,572,640	2,573,120

Fig.2.9. Audio Details

III. CONCLUSION

The advantages of virtualization are undisputed. It goes faster, both in business and in everyday life. In common work with communication systems and applications it has abilities that will undoubtedly make its use more widely.

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