Measurement QoS Parameters of VoIP Codecs as a Function of the Network Traffic Level

Jugoslav Jocić*, Zoran Veličković**
* Telekom Srbija AD, Prokuplje, Serbia
** College of Applied Technical Sciences, Niš, Serbia
jocic.jugoslav@gmail.com, zoran.velickovic@vtsnis.edu.rs

***Abstract*—This paper analyzes the effect of the level of network traffic to the implementation QoS parameters of VoIP service in the LAN. We analyzed the number of dropped packets and packet delay variation for ten simultaneous VoIP connections. Manage Connections are made using the Asterisk VoIP server and several typical codecs are analyzed. Basic QoS parameters of VoIP services are measured in terms of variable network traffic. Measuring results clearly indicate that the number of dropped packets increases with increasing intensity of network traffic, and that very little depends on the user's VoIP codecs. On the other hand, the variation of packet delay crucial role has a selection of VoIP codecs is shown. In the realized experiment, all QoS parameters were in the preserved required limits.**

I. INTRODUCTION

The rapid expansion and distribution of computer networks has made them very suitable medium for the transmission of different kinds of content. In addition to the exchange of documents in text format, computer networks have become the main medium for the transmission of all types of multimedia content. Computer networks are not designed for multimedia content in real time, but modern communication protocols ensures the implementation of appropriate application's QoS (Quality of Service) [1]. This fact has contributed to that the exchange of digital multimedia content becomes the dominant form of network traffic. The LAN networks that are based on TCP/IP protocols cannot guarantee QoS, but they provide only "best effort" QoS services. Thus, IP networks can provide to the multimedia applications only those resources that are at that moment available. It is evident that in the multi-services networks is not easy to achieve the required application QoS. Since different applications require different QoS, the distribution of available network resources is a major research challenge [2], [3]. For voice transmission over computer networks was developed VoIP (Voice over Internet Protocol) technology that is based on the communication protocol IP (Internet Protocol). Unlike traditional telephone services, which can be realized, by circuit-switched, VoIP service is implemented with packet switching. Each IP packet among other things in the header contains the source and destination IP address, while the rest of the pack carries the application data. Network routes for delivering the packets to the destination depend on many factors, and in this paper, we analyzed the dependence of the type and intensity of network traffic. Different routes packets through the computer network as a result of the routing asymmetry can cause side effects and the VoIP service [4]. Thus, in IP networks that can happen: package does not reach the destination, that packet arrives too late, that the package arrives damaged, you arrive at your destination two identical packets, or packets arrive at their destination in the order in which they were sent.

This paper analyzes effect of the network traffic intensity in LAN to the VoIP service performance. Unlike data transfer, voice service is real time application and sets stringent QoS requirements in connection with delay and delay variation (jitter) package. On the other hand, speech shows greater tolerance to packet loss in relation to the data transfer. Providing appropriate QoS for multimedia applications is the basic problem which to be solved in VoIP networks. The implementation of appropriate QoS in VoIP networks is extremely important because in this way provides a normal conversation participants. For the implementation of VoIP services in digital networks, the most important parameters are: a) packet loss, b) packet delay and c) packet delay variation [2]. In this paper we do not discuss methods for the implementation of QoS (they are embedded in the analyzed protocols), but the performances of VoIP services based on measurable technical parameters such as packet delay variation and packet delay estimated. For the provision of high quality services, computer network must achieve the aforementioned QoS parameters within the limits defined class of service. Acceptable delay packet of speech is 150 ms, while for international connections this parameter can be tolerated in the range of 150 ms to 400 ms. The values of packet delay of 400 ms are unacceptable. Packet loss up to 1% and the jitter value of 30 ms is acceptable [1], [2], [3], [4]. The paper also analyzed the impact of VoIP codecs used in consideration of QoS parameters. The effectiveness of the applied compression algorithm is very importance because it directly affects the packet delay.

Besides the objective QoS parameters that are used in this paper, can be used subjective methods of evaluating QoS. Subjective assessment of quality speech MOS (Mean Opinion Score) is implemented in a controlled environment with the participation of a large number of listeners [5], [6]. PESQ (Perceptual Evaluation of Speech Quality) is an objective measure of the quality of the speech signal based on a comparison of the signal at the entrance and exit of the VoIP system [7]. PESQ actually

TABLE I.
BASIC FEATURES OF VOIP CODECS

VoIP Codecs	Sampling (kHz)	Bandwith (kb/s)	RAM (ms)	CPU load (MIPS)	PESQ
G.711	8	64	10	0.5	4.3
G.722	16	48,56,64	10	14	4.1
G.729	8	8	10	22	3.8
GSM	8	13	20	5	3.4
iLBC	8	15.2, 13.3	30	15,18	3.8

gives a good objective assessment of the MOS estimates but requires specialized equipment. The structure of the paper is as follows. The second chapter provides an overview of the characteristics of the considered VoIP codecs. In the third chapter describes the network protocols that transport voice signals. Networktopology and objective way of measuring QoS parameters are presented in the fourth section, while the fifth section presents the results of measurements. The final section presents conclusions based on the obtained results.

II. BASIC FEATURES OF VOIP CODECS

This paper analyzes the following VoIP codecs: G.711, G.722, G.729, iLBC and GSM. Selected codecs are very different by coding techniques, the required minimum bandwidths and processor loads. Selected codecs on the server side (VoIP server) and client-side (softphone) VoIP applications are supported. Table I shows the basic characteristics of the analyzed VoIP codecs. G.711 codec is used in standard VoIP telephony for narrowband speech signal from 0.3 kHz to 3.4 kHz. This codec requires a large bandwidth of 64 Kbit/s and achieves a high quality speech signal from PESQ = 4.3. G.722 belongs to a group of broadband voice codecs on the frequency range of 50 Hz to 7 kHz. This codec is characterized by clarity and quality of the speech signal. The bandwidth of this codec is equal to or below that of the G.711 codec. G.729 codec achieves extremely high compression, which results in a very small bandwidth with tolerance to errors. This codec is designed to transmit narrowband speech signal and speech quality is very good. GSM codec is used in mobile telephony, but it can be used in VoIP in a low bandwidth required. The main disadvantage of this codec is relatively low quality of speech. Like GSM, iLBC codec is designed for narrowband voice range. iLBC codec is an open source solution, and requires little bandwidth with very good quality contracted through speech.

III. PROTOCOLS FOR VOIP PACKETS TRANSMISSION

VoIP applications use IP, UDP and RTP protocols for packet transmission. Considering the purpose of developing, in IP protocol is a not incorporated mechanism to control data flow, as well as procedure for the correction of received packets. In addition, the IP protocol has no mechanisms for retransmission, so that in case of transmission errors due to congestion or disconnection package may be lost. These are the reasons because the IP protocol is considering unreliable. The primary function of IP is to enable the routing of traffic through the network or to provide the best possible

IP (20 bytes)	UDP (8 bytes)	RTP (12 bytes)	Payload Useful part is variable and depends on the size of codecs

← IP/UDP/RTP header 40 bytes →

Figure 1. The structure of VoIP packets.

path from the source to the destination, using a single IP address of the participant concerned. Besides the already mentioned, a protocol enables IP fragmentation data that is necessary for various network specifics in terms of the length of the package [8]. UDP is a transport layer protocol suitable for real-time communication, so it is applied in applications for voice transmission. UDP is a connectionless protocol, which increases its efficiency due to the absence stages of establishing and closing the connection. However, UDP does not provide a mechanism for confirming delivery of data to the destination, as well as maintaining correct order of packets. Because of the displayed weakness, UDP in conjunction with RTP is often used. RTP solves problems for that UDP was not designed. RTP protocol provides the transfer function of end-to-end network necessary for the implementation of QoS for multimedia applications. Applications on the receiving side can detect packet loss, jitter or packet delay based on information provided by RTP. Each VoIP package consists of two components: a) headers and b) speech samples - the usable portion - payload. The structure of a VoIP packet is shown in Fig. 1. VoIP packet header is constant length of 40 bytes [9], whiles the useful part of the variable length and depends on the used codec. With increasing length packets, increases the efficiency of transmission, but reduces the bandwidth for other VoIP connection, which will result in packet delay. Calculating the optimal length of VoIP packets ($VoIP_{ps}$) can be determined by (1) [10]:

$$VoIP_{ps} = \left(f_s * bps * \frac{p_t}{8} \right) + S_H \tag{1}$$

where f_s is sampling frequency, *bps* indicates the number of bits per sample, p_t time packetization, and *SH* size of the header. On further increase VoIP packets can affect the type of media used and the use of security or tunnel protocol that adds its header information [11].

IV. TOPOLOGY SYSTEM AND METHOD OF MEASUREMENT

A. Computer Network

A simplified computer network topology in which the experiments were carried out is shown in Fig. 2. The foundation of presented computer network is LAN network in the *High Technical School of Applied Studies Niš* (VTŠ Niš), which owns a number of routers, switches and user computers. VoIP network consist of the VoIP server, twenty computers equipped with VoIP accessories and a separate PC for generating the network traffic. All devices are equipped with Fast Ethernet 100Mb/s network adapters and connection unleashed UTP cables. The network has the Asterisk VoIP server. The client part of VoIP communication is implemented with *CounterPath Eyebeam* softphone. Generating additional

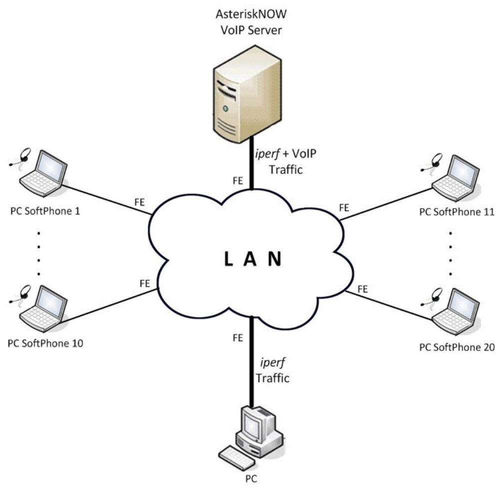

Figure 2. Block diagram of measuring topology.

traffic is carried by *iperf* program, while the measurement of QoS parameters is carried Wireshark program.

B. Asterisk VoIP server

As a VoIP server is used Asterisk server with specialized software *AsteriskNOW* that is installed on anon a standard PC. *AsteriskNOW* software is open source solution developed for the Linux operating system. Access to the Asterisk server is provided through the Web interface *FreePBX* and it was done by adjusting the *SIP* (Session Initiation Protocol) accounts as well as the activation of the codecs to be used in the experiment. To start, maintaining, and terminating sessions are also used SIP protocol [12].

C. CounterPath eyeBeam

At 20 computers that are running Microsoft Windows XP, as well as VoIP client is installed softphone CounterPath Eyebeam. SIP accounts of this program are set by the configuration of network parameters and SIP accounts AsteriskNOW server. In the present experiment was achieved ten simultaneous VoIP connection, so that all VoIP packets serve AsteriskNOW server [13].

D. Iperf

As already mentioned, to generate additional network traffic intensity in the VoIP network used is a software-tool called *iperf*. *Iperf* is also a tool for measuring network performance that is designed to run on various platforms such as Windows, Linux, UNIX or Android. *Iperf* program is installed on two computers designated as *AsteriskNOW* and *PC* (Fig. 2). On both computers are running the client and server application component, so that both computer generated and accept UDP traffic [14]. In order to achieve the desired level of network traffic should set the appropriate parameters via *iperf* command on the client as follows:

```
iperf -c -u -b 90m 192.168.1.10 -t 600 s 1,
```

while on the server side should set the appropriate parameters iperf command:

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iperf -s -u –i,
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first -c Parameter indicates the part of the client, while -s indicates the server part. IP address 192.168.1.10 is the address of the computer that accepts generated traffic, -u indicates the type of generated packets - UDP. Parameter -b 90m provides generating traffic from 90Mb/s, while the parameter -t 600 defines the total time interval of sending packets of 600s. Display interval is defined -i parameter in this case is 1s.

E. Wireshark

Wireshark is open source software and belongs to a class of network protocol analyzer. The main function of this software tool is to record the packets on a network interface as defined and view the captured packets for analysis [15]. For purposes of measuring QoS parameters in the local network, such as packet loss and jitter, using a software package Wireshark. Because the variation of packet delay only measured at the receiver side, it should be noted that Wireshark collects data based on information from the RTP packet. Wireshark is installed and running before the establishment of VoIP connections on all computers marked with the PC SoftPhone 11 to PC SoftPhone 20.

V. RESULTS OF MEASUREMENT

A. Measurement Procedure

An experiment in which the measured QoS consisted of 10 simultaneous VoIP connections in a time of 10 min. We implemented a series of seven measurements. First measurement was performed without additional network traffic, while the other six measurements are performed with additional network traffic. Software tool iperf generates additional network traffic from 20Mb/s, 50Mb/s, 70Mb/s, 80Mb/s, 90Mb/s and 95Mb/s between Asterisk VoIP server and PC. Generating an additional network traffic between iprf and VoIP servers is simulated desired level of network traffic. While this is not the typical VoIP situation in the LAN, this is a way to obtain a valid measurement data. The program Wireshark analyses the packet losses (lost or discarded) and jitter of all VoIP connections. At eight VoIP simultaneous connections are applied only voice codecs, while the other two besides VoIP codecs and video codecs implemented H.263 and H.264. We used the already mentioned VoIP codecs: G.711, G.722, G.729, GSM and iLBC.

B. Measured data

In the performed experiments, the number of dropped packets is measured, as well as average and maximum jitter value of all VoIP connections. In Fig. 3 shows the number of dropped packets, in Fig. 4 the average value, and Fig. 5 maximum jitter of VoIP packets. We analyzed the following VoIP codecs G.711, G.722, G.729, GSM, iLBC with variation of network load. Packet loss represents packets that have not arrived at their destination, and derive from errors in the network, i.e., damaged packages that usually occur because of LAN overloading. Fig. 3 shows the percentage of lost packets, which are estimated in relation to the total number of sent

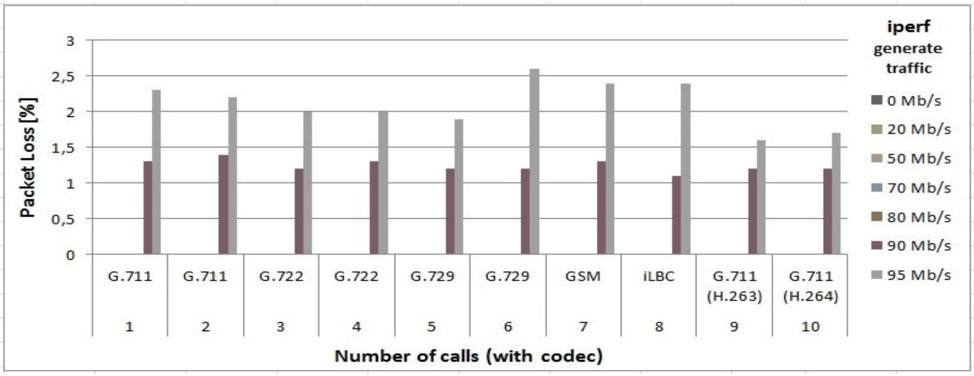

Figure 3. Packet loss for codecs G.711, G.722, G.729, GSM, iLBC

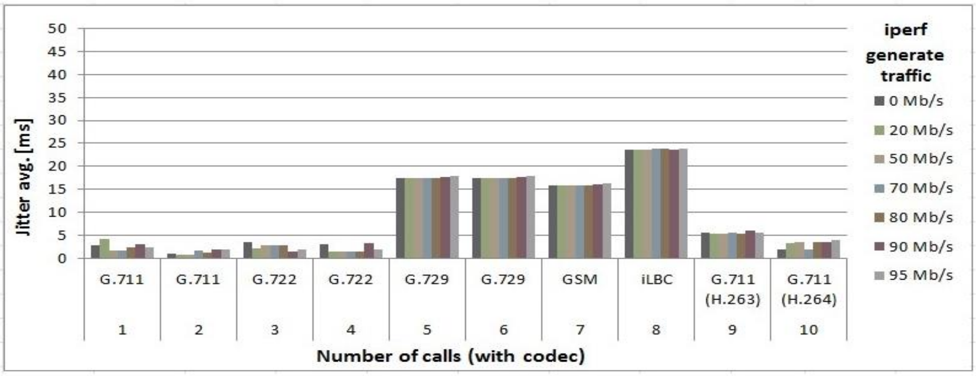

Figure 4. Average time jitter for codecs G.711, G.722, G.729, GSM, iLBC

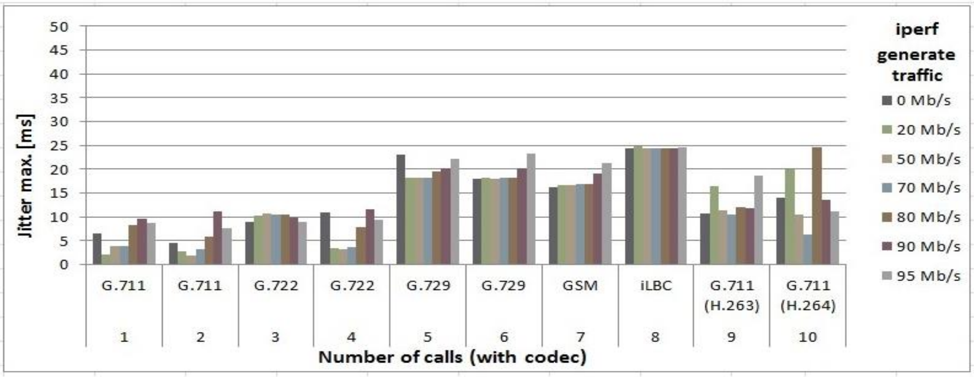

Figure 5. The maximum time jitter for codecs G.711, G.722, G.729, GSM, iLBC

packets. From the graph it can be clearly established that packet loss is not detected when the network load is less than 80Mb/s.

The loss of packets of all codecs can be observed when the network traffic is greater than 90Mb/s. The percentage of packet loss is in the range from 1.1% to 1.4%, while the network load of 95 Mb/s packet loss percentages was in the range from 1.6% to 2.6%. At this level of network traffic at all codecs, the percentage of packets dropped considerably increased. From Figure 3 it can be observed that for some VoIP connections percentage of packets dropped nearly doubled (connection no. 6).

Jitter is a variation of packet delay, i.e., the difference in time delay packets in the same stream. Average time jitter for codecs G.711 and G.722 when the experiment used video codec, i.e., without it, there was no greater than 6ms (Fig. 4). Significantly higher jitter is observed for codec G.729 (8ms), while for the GSM codec jitter was 16ms. The average value of jitter for iLBC codec was 24ms, which is the worst result. It is important to note that in all the codecs maximum jitter value did not exceed 25 ms can be seen from Fig. 4.

Based on these results, we can conclude that the average and maximum jitter values for all codecs are not greater than 30 ms, which means that they are within recommended limits.

However, it is evident that jitter values depend on the used VoIP codecs. Percentage of packet loss values are in the recommended values for the load in the network [16] and VoIP servers up to 80Mb/s. Packet loss is greater than the recommended 1% were recorded for loads over 90Mb/s. From the graph, it can be concluded that packet loss very little depends of the selected VoIP codecs.

VI. CONCLUSION

The paper presents the impact of the level of network load into a local network on technical QoS parameters of voice codecs. Based on experiments it was found that very little packet loss depends on the type of applied voice codec. Significant impact on packet loss had a level of network traffic. Measured data indicate that the packet loss occurred only when the level of network traffic approaching the designed capacity of the connection. Packet loss measurements at 6 and 7 are higher than 1%, which had a negative impact on the quality of the received speech. By measuring was determined that jitter does not depend on the network load and the VoIP server, it depends only of the type of applied voice codec. The values of the measured jitter were less than 25ms and it had no significant effect on the quality of the received speech.

The measurements from 1 to 5 have QoS parameters for VoIP codecs within recommended limits. In this paper, it is shown that it is possible to realize quality VoIP service in LAN using standard VoIP codecs: G.711, G.722, G.729, GSM and iLBC. However, when the level of network traffic approaches to the projected capacity, it may cause increased packet loss and jitter, which can have the effect of variable quality VoIP service. The experiment shows that in this limit loads jitter remains within specified limits. Given that the packet loss and jitter occur only when the network load greater than 80 Mb / s, it can be concluded that the implementation of VoIP service is not a problem in this way conceived LAN.

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