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Quality of Service Analysis for Voice over IP Applications

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Abstract: **The paper presents research results of Voice over IP applications in Local Area Network (LAN) segment. VoIP calls creation, traffic data collection and Quality of Service (QoS) parameter have analyzed from the point of view of their objective and subjective evaluation. The research subjective parameters are R – factor and Mean Opinion Score (MOS) and the objective parameters jitter and packet loss have investigated. The research results are made through real network topology and appropriate traffic flow software.**

Keywords: **Voice over IP, Quality of Services parameters, QoS monitoring**

I. Introduction

Voice over Internet Protocol (VoIP) is one of the fastest growing Internet applications today. It has two fundamental benefits compared with voice over traditional telephone networks. First, by exploiting advanced voice compression techniques and bandwidth sharing in packet-switched networks, VoIP can dramatically improve bandwidth efficiency. Second, it facilitates the creation of new services that combine voice communication with other media and data applications, such as video, white boarding and file sharing [1].

VoIP uses a number of protocols which ensure that voice communication is appropriately established between parties, and that voice is transmitted with a quality close to that we are accustomed to in the Public Switched Telephone Network (PSTN) [2]. It uses signaling protocols such as the Session Initiation Protocol (SIP) [3] and H.323 [4].

II. Exposition

Referring to [5], QoS is defined from two points of view: QoS experienced by the end user and the QoS from the point of view of the network. From the end user's perspective, QoS is the end user's perception of the quality that he receives from the network provider for the particular service or application that he subscribes to, e.g., voice, video, and data. From the network's perspective, the term "QoS" refers to the network's capabilities to provide the QoS perceived by the end user as defined above.

As we said earlier the Quality of Services can be evaluated subjective from the point of view of the end user and objective from the point of view of the network parameters, which can be evaluated directly [6] (delays, jitter, packet loss, throughput and etc.).

The ITU-T E-Model [7] is an analytic model of voice quality that can be used for estimating the relative voice quality between two connections. The E-Model can be used to calculate the *R-factor* which is a simple measure of voice quality ranging from the best case of 100 to the worst case of 0. The R-factor uniquely determines the Mean Opinion Score which is the arithmetic average of opinions where 1 is unacceptable and 5 is .excellent. The R-factor is related in a non-linear fashion to the MOS through the following equation [8]:

$$MOS = 1 + 0.035 * R + 7 * 10-6 * R*(R - 60)*(100 - R) \qquad (1)$$

The authors in [8] presented the graph of the dependency of *MOS* on *R – factor*, which is shown in *fig.1*.

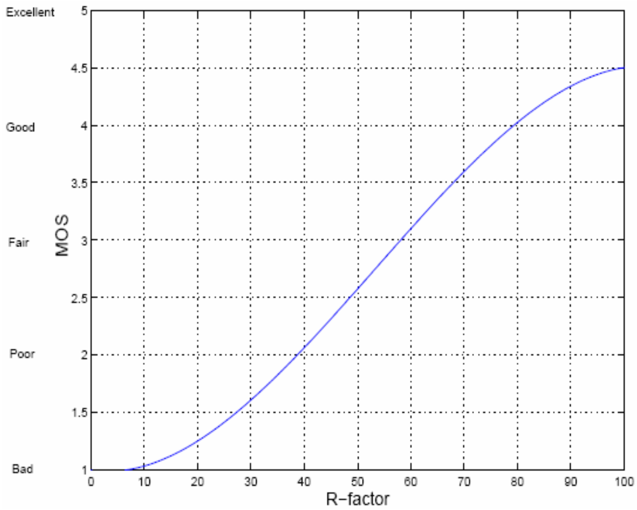

Fig.1. Dependency of MOS on R – factor

Table.1. Relationship of R-factor values to MOS and to the Quality of Voice Rating

R-factor	Quality of Voice Rating	MOS
0< R < 100	Best	4.34 – 4.50
80< R < 90	High	4.03 – 4.34
70< R < 80	Medium	3.60 – 4.03
60< R < 70	Low	3.10 – 3.60
50< R < 60	Poor	2.58 – 3.00

The relationship of *R-factor* values to *MOS* and the typical categorization of *R-factor* values are presented in *Table.1.* It can be seen that connections with *R-factors* of less than 60 are

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expected to provide poor quality, while *R-factors* of 80 and above provide high quality.

The main QoS parameters that quantify the quality degradation over a certain connection are the following: throughput, delay & jitter and packet loss [9]. Delay & jitter are probably the most important ones for VoIP as a real-time streaming application. Packets containing voice data must be delivered in a timely manner in order to ensure user satisfaction [10]. One-way delay influences interactivity: the larger the delay, the lower the perceived interactivity for the interlocutors. On the other hand, jitter (i.e. one-way delay variation) influences quality if it exceeds a maximum value. This maximum value is system dependent and is related to the size of the dejittering buffer used. A large buffer means that jitter has a smaller effect on the quality obtained but it decreases interactivity through the effect of delay. If the induced jitter value exceeds the size of the dejittering buffer, the VoIP packets do not arrive in time for playback, and the playback signal quality decreases. Hence, this distortion is the main effect that jitter has on user satisfaction. Packets that do not arrive in time for playback can be considered lost; therefore this effect is sometimes termed *jitter-loss*.

The jitter (delay variation) is an important factor that has a direct influence on the VoIP quality [9]. In order to counter the effects of the jitter, all VoIP applications use a jittering buffer to try to restore the initial distribution at the expense of adding a supplementary playback delay [10].

In the realization of VoIP applications, the availability of Quality of Services (QoS) is of great importance for the end users. The following parameter limits, the most significant ones for the IP telephony, can be found in the ITU-T recommendations:

- Jitter - less than 50 ms.
- One-way delay
  - up to 150 ms – acceptable;
  - from 150 ms to 400 ms – acceptable with a proviso
  - above 400 ms – unacceptable;

The presentation of empirical data of the research into the VoIP traffic in various LAN segments is a question of present interest. In [11] the authors consider the quality of VoIP calls provided by several VoIP applications running on PCs and VoIP phones connected to a simple LAN environment. Similar results are presented in [12] in which the authors consider an H.323 compatible setup and define a mapping between network layer measurements to VoIP quality. Testbed experiments in which VoIP artificial traffic is sent through a LAN and WAN environment are presented. In [13] a passive methodology for monitoring VoIP phone calls is described, but only simple measurements over an artificially loaded testbed are presented. Finally, the authors in [14] study VoIP quality mapping considering the ITU-T E-Model whose implementation is described in [15].

The aim of the present work is real-time Quality-of-Service (QoS) estimation of a VoIP trace built within a LAN segment. For investigations realization in VoIP network a method for QoS evaluation is need and it's transformation in MOS result. MOS is a method for subjective evaluation as a function of network parameters. Necessary test subjects should evaluate different test conditions. As a result MOS is the medium value of subject evaluations and estimate with the next formula:

$$MOS = \frac{(N_E \times 5) + (N_G \times 4) + (N_F \times 3) + (N_P \times 2) + (N_U \times 1)}{N} \quad (2)$$

where: N_E, N_G, N_F, N_P and N_U are subjects evaluated the test conditions and N is their number. These conditions we couldn't realize to test in our implementation. The software WinEyeQ have used for approximately evaluation of MOS result. The Quality-of-Service parameters in our research are the average jitter, the packet loss, the MOS (Mean Opinion Score) and the R-factor. On the basis of the achieved results conclusions for the performance of the built VoIP trace will be done.

III. EXPERIMENTAL RESEARCH

A. Experimental treatment

The experiments presented in the paper are related to the estimation of the VoIP trace in real time within a local segment. The communication between VoIP terminals in the local segment have been under observation. A hardware SIP server, with TrixBox management interface, has been used for configuring the connections. The topology of the built VoIP configuration is shown in fig.2. The devices have been examined at SIP signaling. A G.711 A-law codec, being an international standard for telephone audio signal encoding through a 64kbps channel, has been used for examining the connections.

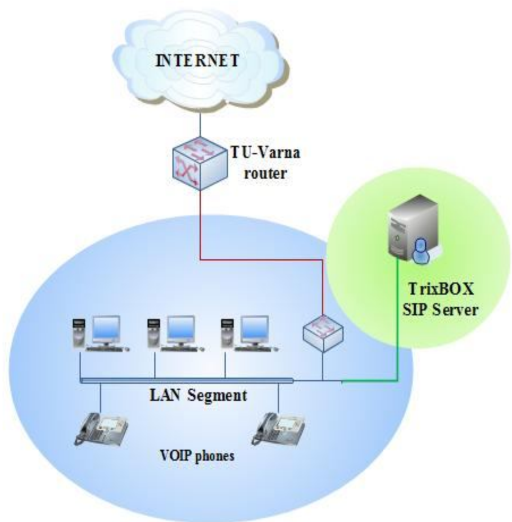

Fig. 2. Topology of the built VoIP configuration

The WinEyeQ software [16] has been used for control and analysis of the VoIP traffic under examination.

B. Research results

The Quality-of-Service parameters in our research were *the average jitter*, *the packet loss*, *the MOS* (Mean Opinion Score) and *the R-factor*. These parameters have been studied for communication between software end devices. The end device QoS options like *"best effort"*, *"guarantied QoS"*, *"qualitative"* and *"controlled load"* were used in the study.

The summary research results of the connections from the end user to the SIP server (Source VoIP phone / Destination SIP server) are presented in *Table.2* and this one from the SIP server to the end user (Source SIP server / Destination VoIP phone) are presented in *Table.3*.

TABLE.2
RESULTS FROM THE END USER TO THE SIP SERVER

QoS parameters	Best effort	Guarantied QoS	Qualitative	Controlled load
average Jitter [ms]	0,797	3,9	4,05	2,102
Packet interval [ms]	19,99	19,99	19,99	19,99
Lis R	88	88	88	88
Con R	87	87	87	87
Lis MOS	4,086	4,086	4,088	4,086
Con MOS	4,063	4,063	4,063	4,063
Packet loss	0	0	0	0

TABLE.3
RESULTS FROM THE SIP SERVER TO THE END USER

QoS parameters	Best effort	Guarantied QoS	Qualitative	Controlled load
average Jitter [ms]	2,98	5,1	3,1	4,391
Packet interval [ms]	19,99	19,99	19,99	19,99
Lis R	91	90	92	92
Con R	91	89	92	91
Lis MOS	4,156	4,137	4,18	4,18
Con MOS	4,156	4,113	4,156	4,156
Packet loss	0	0	0	0

The presented results in *Table.2* and *Table.3* show, that there are not packet loss and the packet interval is 20 ms.

The comparison results of the average jitter values of the connections from the end user to the SIP server and from the SIP server to the end user, in respect to the defined options for guarantied QoS, are shown graphically in *fig.3*.

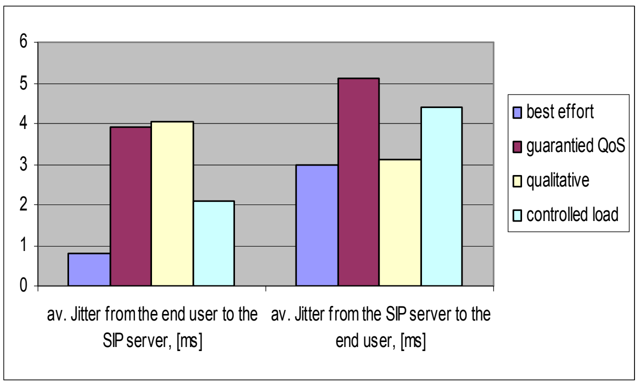

Fig.3. Comparison results of the average jitter values

Conclusion: The lowest jitter values of the connections from the end user to the SIP server and from the SIP server to the end user in the *"best effort"* option in comparison with *„guarantied QoS"*, where the jitter values are the highest. Although the jitter values in the four defined options are different in the both directions, they are less than 50 ms, according to the ITU-T recommendations.

The *R – factor* and *MOS* summary results of the connection from the end user to the SIP server are shown graphically in *fig.4* and *fig.5* and this one from the SIP server to the end user - *fig.6* and *fig.7*.

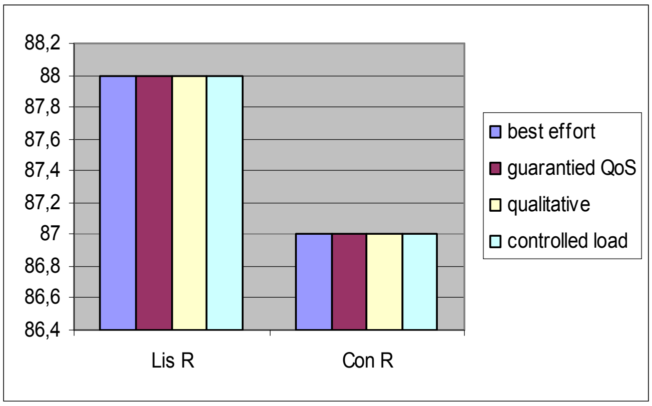

Fig.4. General results for R – factor straight

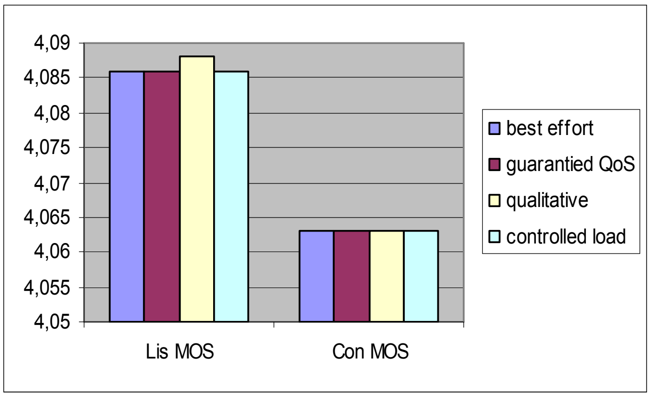

Fig.5. General results for MOS straight

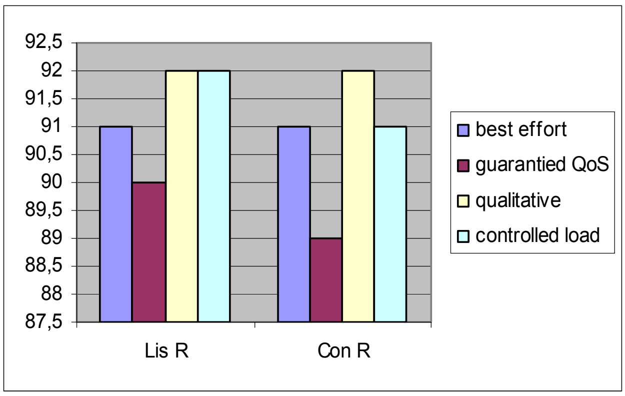

Fig.6. General results for R – factor in return direction

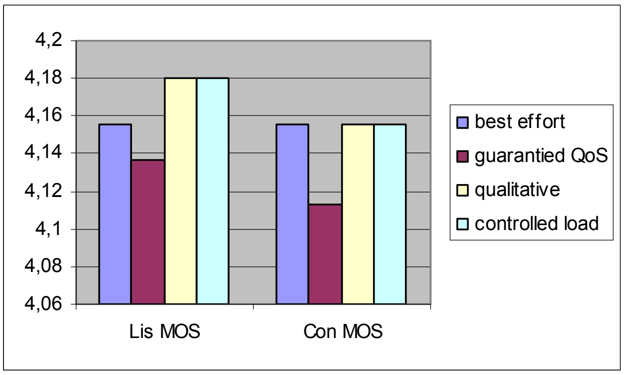

Fig.7. General results for MOS in return direction

These results present good parameters of voice service evaluation for R–factor and MOS straight and in return direction of end users.

III. Conclusion

The analysis of the study made of Quality of Service (QoS) in real time on real VoIP local segment and investigated parameters: *average jitter value*, *packet loss*, *MOS and R–factor,* presents the next conclusions about the quality of the built up VoIP connection:

- the results show that there is no packet loss and the Packet interval is 20 ms.
- The lowest values of jitter straight and in return direction we see at *"best effort"* in comparison with "*Guarantied QoS*", where values are highest. Although jitter values straight differ from values in return direction for the four defined opportunities, they are under upper limit of 50 ms, in accordance with ITU – T recommendations.
- Results for *R–factor* and *MOS* present good parameters of voice service for end users.

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