ON VOIP QUALITY ASSESSMENT

Janusz Klink, Bogdan Miazga, Paweł Sieradzki, Marcin Marcisz

Abstract- The article presents the universal test-bed for Voice over IP transmission measurement. An experimental network has been built and some software tools for traffic generation, modification and evaluation has been used and described. Authors discuss different traffic parameters and their influence on voice quality. Some measurement scenarios and tests has been presented. Measurement environment presented in the paper can be used as a tool for VoIP QoS evaluation and can serve as a stand in students’ laboratory. Authors present first results from subjective method of voice quality assessment in laboratory IP network. Presented environment can be spread on any public network and used for VoIP QoS evaluation.

Index Terms – Quality of Service, VoIP.

I. INTRODUCTION

Real-time services offered in IP networks, especially voice over IP (VoIP), require specific QoS mechanisms, which should be implemented in the network environments. On the other hand, we should be able to supervise the VoIP network and collect the data needed to carry out the evaluation of the quality of service. We also should specify basic transmission parameters (and their proper ranges) which have an influence on service quality. The test-bed, presented in the paper, was built to measure basic transmission parameters of IP network carrying different traffic streams, and to evaluate the influence of these factors on quality of voice transmission.

II. QOS METRICS

One of the most significant classifications of traffic carried in packet-based networks takes into consideration two features of applications used in these networks, i.e.:

− sensitivity to the packet delays,
− packet loss sensitivity.

Usually high sensitivity to the packet delays characterizes real-time applications like telephony, video-telephony, videoconference, which require two-way data transfer to ensure conversational character of the meeting. In these cases exceeding of acceptable limit of packet delay makes this piece of information useless and it must be discarded, otherwise it would cause disturbances during the voice or video signal reception. Applications of this kind are more resistant to loss of data, especially as they often use voice or video codecs based on loss compression algorithms.

On the other hand, delay-tolerant applications are usually very sensitive to the packet loss (e.g. data transmission or data processing applications). Of course the level of delays or percentage of packet loss depends on many things, like the kind of application, retransmission mechanisms, QoS level acceptable by the user, etc.

The traffic issues concerning VoIP transmission can be considered on two levels: the control level and the user level.

The control level is responsible for setting-up, holding up and releasing all the connections while the user level carries data streams during VoIP connections. Therefore QoS metrics for IP-based networks can be divided into two classes [1]: the call control parameters and the information transfer parameters.

The first one plays significant role in all connection-oriented types of communication (not only in packet-switched but in circuit-switched networks as well), while the second group of parameters is important in both connection-oriented and connectionless types of communication.

A set of call control parameters in the connection-oriented networks describes the call set-up, information transfer and release phases. Additionally, the factors describing accessibility of network connections, in relation to all call attempts, are also valuable. This article presents a proposal of scenarios and test-bed for evaluation of VoIP QoS during the information transfer phase of connection.

Therefore we should control a group of parameters, which can be divided into following groups:

− the first one consists of parameters describing the network and the data transfer, irrespective of the application used,
− the second group includes some QoS metrics in strong connection with specific applications.

Most of the parameters can be the same but interpretation and evaluation of them is often different. For example: the information transfer delay value may be quite low and may suit the requirements of data transfer applications, while it may be unacceptable for some real-time applications (e.g. telephony, videoconferencing, etc.). Therefore classification and assessment of these parameters is not easy. The most important information transfer parameters, used for assessment of the network quality, irrespective of the applications, are described below.
A. Bit error ratio

BER is defined by the number of erroneous bits divided by the total number of bits transmitted, received, or processed over some stipulated period of time.

B. Loss ratio

The loss ratio is a parameter which tells how many data units have been lost in the network in relation to all units sent by the source.

C. Insertion ratio

The number of all data received, divided by the sum of data sent during the connection is called insertion ratio.

D. Transfer delay

Transfer delay is a period of time between sending the first bit of data unit by the originating point and receiving the last bit of data in the receiving side. The total transfer delay consists of several delay components inserted by different network elements and depending on the specific procedures functioning in the network.

E. Jitter

In general the total time between sending a data unit from the source and receiving it by the destination point can be not a constant value, so jitter can be described as an end-to-end delay variation.

III. QoS EVALUATION FOR VOIP

There are different methods for voice quality assessment - the most reliable are subjective ones [2, 3, 4, 5]. In these methods a group of listeners listen to hundreds of short speech utterances processed by the system under test and rate its performance as e.g. five-point scale [6].

Subjective methods are expensive [7] both in time and cost, and obviously difficult to reproduce. Thus it is very desirable to introduce an objective method which can reflect subjective ratings on speech signal in reliable manner. Description of such kind of method one can find in the ITU-T Recommendation G.107 [8]. This document presents E-Model which is a transmission-planning tool for estimating the user satisfaction of narrowband, handset conversations - as perceived by the listener - it is an objective method describing QoS in the VoIP transmissions (Fig. 1). A selected LAN/WAN parameters emulation - we can predefine, among other things, traffic delay and jitter, packet loss, bandwidth limitation, packet reordering and so on. It allows to shape traffic parameters very freely, according to different measurement scenarios.

The presented test-bed enables measurement of selected IP network parameters and their influence on VoIP quality of service. A set of basic measurement scenarios has been presented below:

- testing of bandwidth demand for VoIP, as a function of codec and its parameters,
- testing the impact of available bandwidth on packet loss ratio,
of synchronization between transmitter and receiver recalculated. This measurement method eliminates the necessity of synchronization between transmitter and receiver (left and right channel) and the transmission delay was calculated. Then these signals were compared by computer with a sound card connection between transmitter and receiver. Then the signals were conducted by computer with a sound card (left and right channel) and the transmission delay was calculated. This measurement method eliminates the necessity of synchronization between transmitter and receiver required in other methods. Packet delay is only one example parameter that may influence on voice quality in the network. To make a complex evaluation of voice transmission we should examine also the other parameters (mentioned in the Chapter II). If we have all the parameters measured, the problem is: how to translate the values of these parameters into VoIP QoS estimation and how to make this evaluation reliable?

Authors proposed to use an automatic voice recognition method, developed by AIPSA Group at Wroclaw University of Technology, as a way of QoS estimation. The experiment consists in transmission of prepared sets of logatoms (nonsense words) via real isolated IP network, which parameters (delay, jitter, packet loss etc.) can be modified according to our needs, and introducing them into PC with voice recognition tool installed. For generating, capturing and voice recognition activities authors used the personal computer (Fig. 3) with appropriate software: Logtrans (generation) and QE-ARM (capturing and automatic voice recognition). QE-ARM presents the results of voice recognition, derived from comparing transmitted voice samples (in *.wav format) with original ones, as a percentage of properly recognised logatoms. To compare the voice samples, the FFT parametrization with different windows (Bartlet, Blackman, Blackman-Harris, Hamming and Hanning) and Euclides metrics has been used (Fig. 3).

V. FIRST MEASUREMENTS

Measurement network allows to insert and measure not only one-way end-to-end transmission time of voice signal (mouth to ear) [15] but packet loss and jitter as well. The delay measurement concept consists in generating a test signal (using Audacity application [16]) and simultaneous transmission of the signal between two endpoints (transmitter and receiver) using two transmission paths: through the IP network (consisting of transmitter, router, IP network emulator – NetDisturb - and receiver) and through the direct connection between transmitter and receiver. Then these signals were compared by computer with a sound card (left and right channel) and the transmission delay was calculated. This measurement method eliminates the necessity of synchronization between transmitter and receiver required in other methods. Packet delay is only one example parameter that may influence on voice quality in the network. To make a complex evaluation of voice transmission we should examine also the other parameters (mentioned in the Chapter II). If we have all the parameters measured, the problem is: how to translate the values of these parameters into VoIP QoS estimation and how to make this evaluation reliable?

The next problem is: what is the relationship between objective method of voice recognition and subjective voice quality assessed by user? An answer to this question will be given in the next stage of the research, when authors will examine the subjective methods of VoIP QoS evaluation and will try to discuss it, comparing with earlier measurements.

VI. SUBJECTIVE METHOD OF VOICE QUALITY EVALUATION

Some subjective measurements of speech quality in the examined IP network have also been done. However quality of speech is a complex concept, intelligibility is still viewed as a basic and most important aspect of the quality [17].

Polish standards, published by Polish Committee for Standardization [18, 19], which describe requirements and methods of logatom intelligibility measurement, formulate 5 classes of speech quality).

Some subjective measurements of speech quality in the examined IP network have also been done. However quality of speech is a complex concept, intelligibility is still viewed as a basic and most important aspect of the quality [17].

According to the standards, to measure the logatom intelligibility one should conduct subjective tests with a number (minimum 5) of trained listeners. In our experiment the listeners heard specially prepared sequences of lagatoms, transmitted by the telecommunication channel and try write down (in orthographic form) received utterances on a sheet of paper. Some logatoms have not been recognized properly by the listeners because of distortion in the examined network, e.g. packet loss. Then the results have been revised and the average intelligibility has been calculated. Next the speech quality for this telecommunication channel has been estimated, as shown in the Table 1 [18].

Table 1. Speech quality classes.

<table>
<thead>
<tr>
<th>Quality Class</th>
<th>Class Profile</th>
<th>Logatom Intelligibility [%]</th>
</tr>
</thead>
<tbody>
<tr>
<td>I</td>
<td>Good understanding of the speech without any special attention and without subjective noticeable contamination of the speech.</td>
<td>Over 75</td>
</tr>
<tr>
<td>II</td>
<td>Good understanding of the speech without difficulties; with subjective noticeable contamination of the speech.</td>
<td>60-75</td>
</tr>
<tr>
<td>III</td>
<td>Understanding of the speech with attention, without repetitions and questions.</td>
<td>48-60</td>
</tr>
<tr>
<td>IV</td>
<td>Understanding of the speech with a big attention, with repetitions and questions.</td>
<td>25-48</td>
</tr>
<tr>
<td>V</td>
<td>No possibility of satisfactory understanding of the speech (communication break off).</td>
<td>Below 25</td>
</tr>
</tbody>
</table>

The main drawback is that this method is lengthy (it's a time-consuming operation) and relatively expensive (one should prepare appropriate group of listeners).

Thus, this method have been used here rather not as a basic one in a speech quality assessment process, but in order to find a reference for earlier objective tests.

G.711 codec has been examined. Results of the observations have been presented in Table 2.

Comparing earlier results (using objective method) and results of the subjective measurements, we can see, that FFT parametrization seems to be the close to the subjective authors’ evaluation of voice quality in the examined network.
Table 2. Evaluation of quality classes for G.711u codec.

<table>
<thead>
<tr>
<th>Packet loss [%]</th>
<th>Listener No.</th>
<th>Properly recognized logatoms [%]</th>
<th>Quality class for the individual listener</th>
<th>Quality class for the codec</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>60</td>
<td>II</td>
<td></td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>66</td>
<td>II</td>
<td></td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>69</td>
<td>II</td>
<td>II</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>51</td>
<td>III</td>
<td></td>
<td></td>
</tr>
<tr>
<td>5</td>
<td>55</td>
<td>III</td>
<td></td>
<td></td>
</tr>
<tr>
<td>1</td>
<td>60</td>
<td>II</td>
<td></td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>55</td>
<td>III</td>
<td></td>
<td></td>
</tr>
<tr>
<td>9</td>
<td>62</td>
<td>II</td>
<td>II</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>61</td>
<td>II</td>
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<td>5</td>
<td>63</td>
<td>II</td>
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<tr>
<td>1</td>
<td>62</td>
<td>II</td>
<td></td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>49</td>
<td>III</td>
<td></td>
<td></td>
</tr>
<tr>
<td>15</td>
<td>70</td>
<td>II</td>
<td>II</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>66</td>
<td>II</td>
<td></td>
<td></td>
</tr>
<tr>
<td>5</td>
<td>69</td>
<td>II</td>
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<td></td>
</tr>
</tbody>
</table>

All logatoms were revised according to standard PN-V-90002 [19] and quality classes were estimated according to PN-T-05100 [18].

VII. CONCLUSION

Packet loss and delays observed in the experimental network depend on voice application parameters and traffic conditions in the network as well. Authors wanted to show an example of possible measurement scenarios showing the influence of available bandwidth and packet loss in the IP network on VoIP QoS parameters. The worst conditions have been chosen, i.e. the biggest packet size of background traffic, which introduces big delays and the lack of any transmission control mechanisms (only UDP traffic). In the paper, only one of parameters, mentioned above, has been discussed in more detail – it is packet loss. Its influence on voice quality has been examined, using objective and subjective methods, respectively. Comparison of the results shows that it is possible to find a good relationship between both methods. Next, authors are going to make an effort to make this relationship more precise.

This test-bed makes possible full control of VoIP connection establishment and observation of transmission parameters. Thanks to isolating this experimental network from the external traffic (e.g. from other uncontrolled traffic in a real LAN or Internet) all network parameters and traffic streams are under our control. All measurements can be repeated under identical environment conditions. The presented method of automatic voice recognition will be used as an objective method of VoIP QoS evaluation. In the next step the authors are going to perform more experiments, using the presented test-bed, introducing into examined network more complex transmission disturbances and they to evaluate influence of these impairments on voice quality. The investigation will be performed in different environments – especially in wireless IP network.

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