DETERMINATION OF VIDEO-TELEPHONY CONNECTION QUALITY UNDER THE INFLUENCE OF PACKET LOSS

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Abstract
In this paper we present a new method for video-telephony connection quality assessment. This method is based on application of 1 - cumulative distribution function (1-CDF) of video connection quality. This method provides much better insight into the connection quality depending on the mutual influence of packet loss, implemented codec type, bit rate and frame rate than it is in usual way, just based on the mean value of quality. The method is demonstrated by simulation program for quality distribution assessment on the base of random packet loss generation.

Keywords: video-telephony, connection quality distribution, cumulative distribution function, simulation program.

INTRODUCTION

In recent years, great efforts are directed towards finding a reliable method for determining the video connections quality. The investigations are carried out in two directions. The aim of the first direction is to come up to objective measure of the video connection quality assessment, while the other direction is intended to provide a numerical assessment of subjective video connection quality. The first group of models determines the measure of the video signal distortion on the connection receiving side in relation to the originally transmitted signal. The problem is that, although these models provide an objective assessment how much of the video signal is poorly transmitted, it does not necessarily mean that users surely estimate the video signal transmitted with less degradation as better. Another problem with objective models for determining the quality of video connections is that they are relatively complex and require a lot of processing time, so they are not easy to apply in practice, i.e. to determine the quality of video connections in real time. The resulting connection quality assessment is not an easily understandable measure to users, who are not specialists in the field of video signal processing. As opposite, subjective estimations of the video connection quality are based on the characteristics of human image perception. The numeric quality assessment is easy to understand. However, the assessment is significantly connected to the image content (its dynamics, object types, etc.), so it is difficult to implement an overall subjective model, applicable in different conditions.

Models for determining video connections quality can be divided into three groups: full reference (FR), reduced reference (RR) and no reference models (NR). When FR model is implemented, originally generated video sequence and the video sequence at the receive connection side are directly compared and their mutual differences are expressed on the basis of a predefined criteria. Although this estimation is the most accurate, it is not often implemented in practice, because, on one hand, it requires a lot of calculation, and, on the other hand, in practice, originally generated video sequence is not known at the receive side, where video signal quality is determined. On the contrary, when RR is implemented limited number of data related to the generated video signal is sent to the connection receive side, and they are used as parameters for the calculation of video connection quality, while in the case of NR model video signal quality is determined only on the basis of the video signal at the receive side. An overview of different methods for the video connection quality determination is presented in [1], [2].

CALCULATION OF VIDEO CONNECTION QUALITY

One important characteristic of the digitized video signal is that each image, which forms a video signal, is not transmitted complete but after one complete image follow few images, which present only the difference to the original, fully encoded image, or the difference compared to the previous and the next image, [3]. Images that do not carry complete information on the video signal require less bandwidth for their coding than images, which are completely coded.

Video connection quality is impaired by errors which occur on a link between transmit and receive side of the connection. Video-telephony connection quality is displayed by the parameter, called video quality (V_q). It is a number between 1 and 5.

Recommendations regulate method of video connections quality calculation in a case of random packet loss, [4]. For a case of burst packet loss recommendations are still not defined. That's why significant efforts have been made in order to find a method to determine video connection quality, if there is burst packet loss on a link. Some of the developed methods are presented in [5-8]. The most convenient for use are formulas presented in [8], because they give a subjective, easily understandable assessment in the form of numerical values from a range between 1 and 5 and because the calculation procedure is relatively simple.

Video-telephony connection quality is calculated by the formula, [4]:

$$V_q = 1 + I_{coding} \cdot e^{-\frac{P_{plV}}{D_{PplV}}} \tag{1}$$

where it is:

P_{plV} - packet-loss rate;

D_{PplV} - packet-loss robustness factor: degree of video quality robustness against packet loss;

I_{coding} – coding distortion.

The value of parameter I_{coding} is determined in [4] as:

$$I_{coding} = I_{Ofr} \cdot e^{-\frac{(\ln Fr_V - \ln O_{fr})^2}{2 \cdot D_{FrV}^2}}, 0 \le I_{coding} \le 4 \tag{2}$$

where it is

$$O_{fr} = v_1 + v_2 \cdot Br_V, 400 kbit/s \le Br_V \le 2000 kbit/s \tag{3}$$

$$I_{Ofr} = v_3 - \frac{v_3}{1 + (\frac{Br_V}{v_4})^{v_5}}, 0 \le I_{Ofr} \le 4 \tag{4}$$

$$D_{FrV} = v_6 + v_7 \cdot Br_V, 0 < D_{FrV} \tag{5}$$

In these formulas it is:

Br_V - video bit rate;

O_{fr} – optimal frame rate (i.e. frame rate, which maximizes video quality at each Br_V);

I_{Ofr} – a value of video connection quality at optimal frame rate;

Fr_V - video frame rate;

D_{FrV} – degree of video quality robustness due to frame rate.

Packet-loss robustness factor from equation (1) may be now expressed by:

$$D_{PplV} = v_{10} + v_{11} \cdot e^{-\frac{Fr_V}{v_8}} + v_{12} \cdot e^{-\frac{Br_V}{v_9}} \tag{6}$$

The values of coefficients v_1 to v_{12} in presented equations depend on video codec type, video format, key frame interval and video display size and for some combinations of these parameters are presented in Table I.2 in Appendix I from [4], while the method for determination of these parameters in a general case is presented in Annex A, also from [4].

CDF AS ESTIMATION OF VIDEO-TELEPHONY CONNECTION QUALITY – SIMULATION METHOD

Calculation on the basis of equations (1) to (6) gives a mean value of quality assessment of video-telephony connection. The starting point is the mean value of packet loss and then, as a final result, the mean quality value is calculated. However, this estimation may in some cases provide not enough accurate measure of quality. That's why it is suitable to introduce the cumulative distribution function (CDF) as a measure which gives a more comprehensive assessment. CDF shows what percentage of time video connection quality falls within certain limits, which, for example, could correspond to the unsatisfactory quality value, while in the same time obtained mean value may correspond to a satisfactory quality.

CDF application for assessing the connection quality is already presented in a case of voice connections in [9]. On the basis of the mean value and the probability distribution of packet loss for real, recorded trace on a link has been established (in the analyzed case) that the mean value indicates unsatisfactory voice call quality, while the analysis based on CDF indicates that the connection was satisfactory even for 44% of time.

Method of connection quality analysis (whether it is voice or video connection) based on the CDF implementation starts from the fact that the entire connection may be divided on a number of segments of exact pre-defined duration (for example 5s or 10s). Each segment comprises a number of packets, which transmit a part of voice (or video) signal content. Then, packet loss statistics is determined for each such segment by collecting data whether each individual packet within the segment is lost or successfully transmitted over the network. Overall packet loss probability within the segment and distribution of packet loss (on random loss and burst loss, if burst loss exists) are determined at the end of segment duration. This is followed by calculation of (voice or video) connection quality for the considered segment. The resulting value of quality contributes to the value of the probability density function (PDF), for a total distribution of connection quality. Finally, based on the total accumulated PDF value, CDF of connection quality is determined. Statistics of lost packets may be established by the real recorded trace or by process simulation.

In this paper, the results are based on simulation. We have chosen to present (instead of the value of the CDF distribution) the value of the 1-CDF distribution, i.e. the probability that the video connection quality is higher than some specific numerical value. ($V_q>x$, where it is $1\leq x\leq 5$). The flow-chart of such a program is presented in Fig. 1.

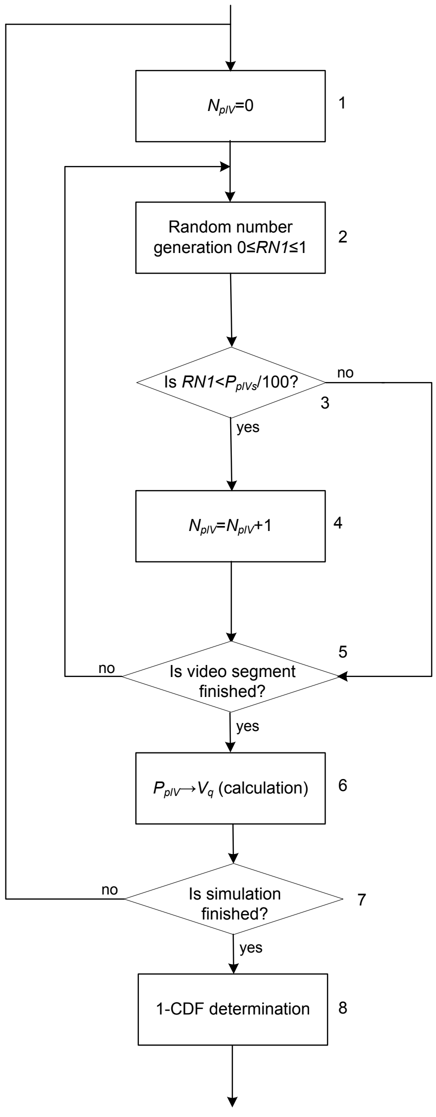

Fig. 1. *Flow-chart of simulation program*

At the program start the counter of lost video packets is initialized on zero (N_{plV}) (block 1). After that, simulation process starts from the random number generator with uniform distribution within the limits (0,1) (block 2). In the case that the generated random number is smaller than $P_{plVs}/100$ (where P_{plVs} is desired percentage of packet loss for the simulation) (block 3), the considered packet is lost in the transmission, so the value of N_{plV} is increased (block 4). A simulation process (blocks 2, 3 and 4) is continued until it is not satisfied a number of simulation steps, which corresponds to the total length of the simulated segment (test in the block 5). When simulation on a considered segment is completed, packet loss probability (P_{plV}) is calculated on the basis of N_{plV} value and duration of the simulated segment. Then a V_q value in the considered segment is determined by the formulas (1) to (6) (block 6).

Simulation steps 1 to 6 are repeated until approaching the desired value of the number of simulated segments (block 7). When a desired number of simulated segments is approached, the collected V_q values for all segments form a 1-CDF distribution of V_q.

SIMULATION RESULTS

Simulation results in this paper present the impact of three factors on the video-telephony connection quality: packet loss probability, applied type of codec and video bit rate.

1-CDF distribution of video-telephony connection quality (V_q) as a function of a numerical assessment of connection quality is presented in Fig. 2. The graphs are given for several values of the probability of video packets loss (P_{plV}) in the range between 0.5% and 4% in the case of MPEG-4 QVGA codec implementation for B_{rV}=1024kb/s and F_{rV}=30. Importance of applying the 1-CDF function may be assessed, for example, on the basis of the curve for P_{plV}=0.5%. The mean assessment of V_q, that would be obtained on the basis of the procedure described in [4], or in other appropriate references, would be ~3.9. However, in such an assessment one does not see the fact that the video-telephony connection quality is worse than 3.9 even during ~50% of time. Therefore, it is better to choose the criteria for the assessment in such a way that we want to find the value of assessment, which will be overcome during, for example, 90% of time ($V_{q90\%}$), or worse will be just during 10% of time. Thus adopted assessment gives value V_q~3.5.

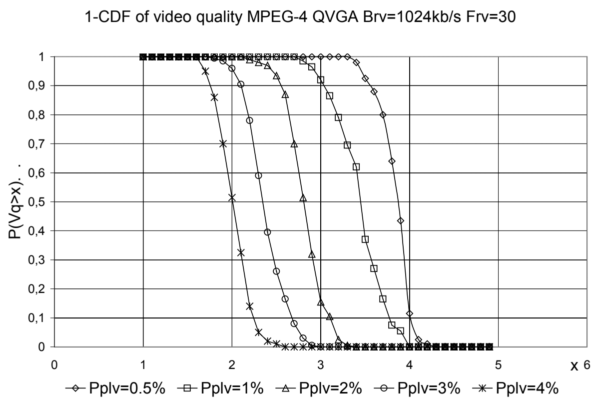

Fig. 2. *1-CDF distribution of V_q for different values of P_{plV} when MPEG-4 QVGA codec is implemented*

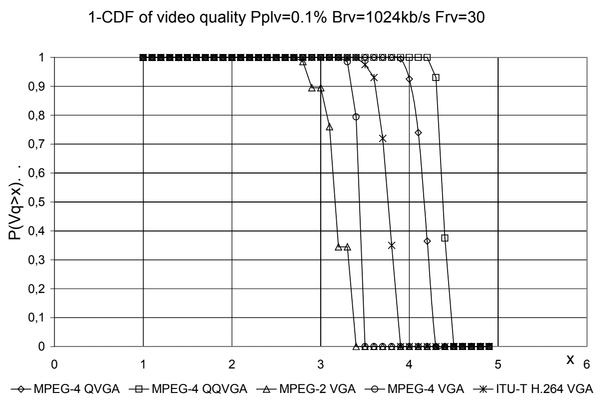

Fig. 3. *1-CDF distribution of V_q for different implemented codec types when it is P_{plV}=0.1%*

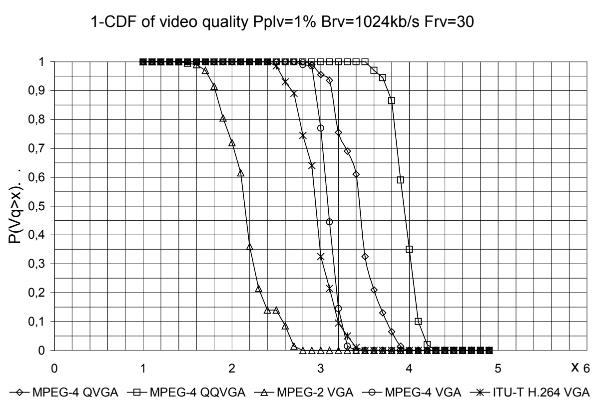

Fig. 4. *1-CDF distribution of V_q for different implemented codec types when it is P_{plV}=1%*

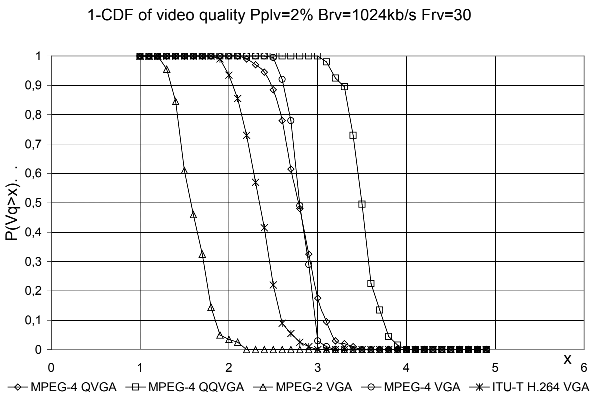

Fig. 5. *1-CDF distribution of V_q for different implemented codec types when it is P_{plV}=2%*

Table 1: *$V_{q90\%}$ values for MPEG-4 VGA and ITU-T H.264 VGA codec for different P_{plV} values*

P_{plV}	$V_{q90\%}$ for MPEG-4 VGA	$V_{q90\%}$ for ITU-T H.264 VGA
0.1%	3.3	3.6
1%	2.9	2.7
2%	2.6	2.1

Figs. 3, 4 and 5 present 1-CDF distribution of V_q for different implemented codec types in the case of B_{rV}=1024kb/s and F_{rV}=30. The graph in Fig. 3 is for P_{plV}=0.1%, in Fig. 4 for P_{plV}=1%, and in Fig. 5 for P_{plV}=2%. When comparing these graphics, it can be noticed significantly lower sensitivity of MPEG-4 VGA codec to variations of packet loss probability, compared to other codec types. This is most easily illustrated by comparing the values of $V_{q90\%}$ for MPEG-4 VGA codec and ITU-T H.264 VGA codec, which is summarized in Table 1. It can be also seen that the dispersion of V_q around its mean value increases as the value of P_{plV} increases, resulting from a decrease of a slope of 1-CDF codec characteristics when P_{plV} increases. Again, this dispersion of characteristics is least noticeable in the case of MPEG-4 VGA codec implementation. This codec retains a greater slope of the characteristics in relation to the other codec types for P_{plV}=1% and P_{plV}=2%.

The influence of B_{rV} variation is illustrated by the graphs in Figs. 6 and 7. These graphs present 1-CDF distribution of V_q for B_{rV} values in the range 400kbit/s – 2Mb/s in the case of ITU-T H.264 VGA codec implementation, for values P_{plV}=0.1% (Fig. 6) and P_{plV}=1% (Fig. 7). The value of F_{rV} is fixed to 30. Although the idea of the equations (1) to (6) is to change F_{rV} values when B_{rV} is changed in order to achieve maximum V_q, the same value of F_{rV} is retained in Figs. 6 and 7 just for illustration. It can be noticed in Fig. 6 that for low packet loss (0.1%) video connection quality is constantly improving when the video bit rate is increased. With the increase of packet loss (on 1%), Fig. 7, video connection quality is first improved when B_{rV} increases, and then begins to decline (the best is at 1Mb/s). Also, dispersion of V_q values increases when increasing B_{rV}.

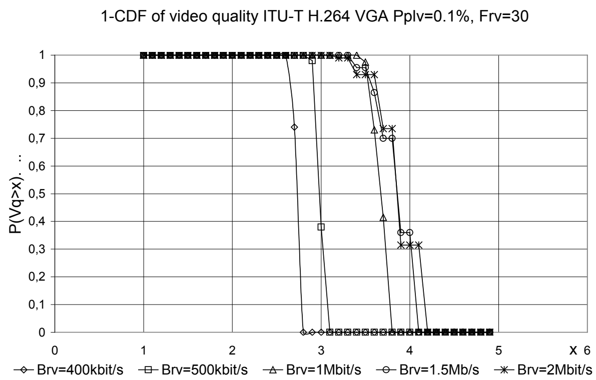

Fig. 6. *1-CDF distribution of V_q for different bit rate values when ITU-T H.264 VGA codec is implemented at P_{plV}=0.1%*

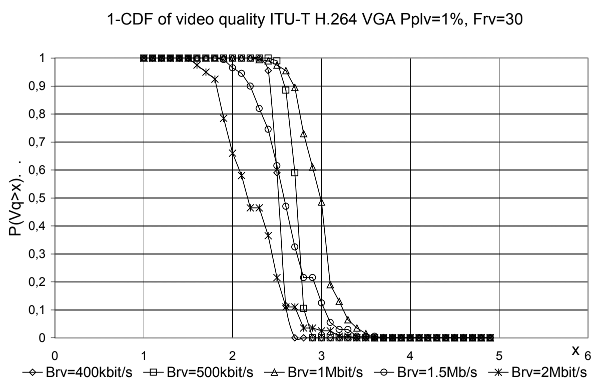

Fig. 7. *1-CDF distribution of V_q for different bit rate values when ITU-T H.264 VGA codec is implemented at P_{plV}=1%*

CONCLUSION

This paper analyzes the impact of several factors (packet loss probability, applied codec type and video bit rate) on the video-telephony connection quality. In this analysis, our primary concern is for (direct and indirect) impact of packet loss. This analysis is generally based on the formulas presented in ITU-T Recommendation [4]. The designation of this less known reference (G.1070) is suggestive, because it associates on a far better known, more often applied and more often updated recommendation G.107, [10], intended for the analysis of packet-telephony connection quality.

A simulation method for quality assessment based on the generation of random events (packet loss) is applied in this paper. 1-CDF characteristics are introduced as more comprehensive assessment of video-telephony connection quality than it is the assessment only on the basis of calculating the mean quality value. Such an assessment makes it easier to perceive simultaneous influence of several factors on the video connection quality. Also, the use of these new characteristics makes it possible to estimate how the loss distribution affects the connection quality distribution. At the same time, some codec characteristics (as, for example, greater MPEG-4 VGA codec immunity against variation of the packet loss probability and its smaller variation of connection quality around the mean value when loss probability is altered, comparing to the other codec types) can best be perceived using 1-CDF characteristics.

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