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The new, parametrised VS Model for Determining the Quality of Video Streams in the Video-telephony Service

This paper describes a new measuring method for determining the quality of video streams in the video-telephony service in IP environments (VToIP). The method uses the so-called VS Model and belongs to the group of parameter-based measurement techniques (offline operation, i.e. without intrusion). The method yields results that come very close the curves gained by the corresponding "Perceptual Evaluation of Video Quality (PEVQ)" It is easy to implement and operates quickly and efficiently. All in all, there a many advantages to using this new measuring method.

1. Introduction

Quality of Service (QoS) and Quality of Experience (QoE) play a very important role in modern digital networks. The term is becoming a household word and can be found among other things in the definition of Next Generation Networks according to the ITU-T Standard Y.2001 [1]. In 2009 the European Parliament and European Council published directives for the standardisation of networks and services [2-3], placing great priority on quality of service.

The QoS respectively QoE in modern networks should be measured continuously - preferably automatically. This makes specialised measurement systems and methods indispensable. There are, however, hardly any standardised QoE and QoS measuring methods for video-communications applications such as video-telephony. There are at present only two standards: ITU-T Rec. J.247 [4] and ITU-T G.1070 [5] to resort to. A third method for measuring the QoE of video services, the PEVQ (Perceptual Evaluation of Video Quality) algorithm, has yet to be standardised [6]. It is one of the signal-based QoE measuring methods. According to the German license holders, the company Opticom [7], the PEVQ algorithm is in line with Recommendation J.247. So it seems it would make sense to work with this method to measure QoE in the VToIP service. However, the QoE measuring methods mentioned so far are very complex and the licences expensive. There simply are no simple, parametrised QoS measuring techniques that are quick and easy to use. The VS Model (VS = video streaming) described in this paper aims to close that gap.

The VToIP service operates according to Recommendation H.323 [8]. This Recommendation defines the encoding of audio and video signals. Codecs H.263, H.263+ and H.263++ are provided to video streaming. A VToIP connection can be established, controlled and terminated using a range of signalling protocols. In practice, however, the SIP [9] protocol is by far the most widely used. Real-life measurements of the VToIP service have revealed that a refresh rate of 25 frames per second is widely used. Common formats for the service are: CIF, QCIF, QVGA and QQVGA. From a practical point of view, this paper takes all these observations into consideration.

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To begin with, the new model will be formulated and illustrated using an actual video codec as an example. Following that, there is an analysis of the model's practicability, i.e. its suitability for everyday use will be put to the test in a comparison study. The results of the investigation will then be presented graphically, and interpreted. The paper concludes with a summary and an outlook on future work.

2. The new, parametrised VS Model for determining QoS in Video Streaming

Paper [10] describes in detail a large-scale investigation that, among other things, demonstrated that packet loss is one of the major factors of impairment in a VToIP environment. The QoS curves display a distinctly exponential character. Other parameters that affect QoS values include: image format and size, encoding rate and burst size (average number of consecutively lost packets in case of random loss). The size of the jitter buffers in the terminal equipment have a significant influence on QoS as well. Minor jitters and some out-of-order packet sequences can be smoothed out in the jitter buffer. If these impairments are too great in relation to the size of the jitter buffer, additional losses will occur, and the model will register them. All of these factors must be considered when creating a new, parametrised model for determining QoS in VToIP. Fig. 1 shows the newly established parametrised VS Model for determining the quality of video streams in the VToIP service. The concept for the VS Model is based on the A-Model [11], that was created for the VoIP service.

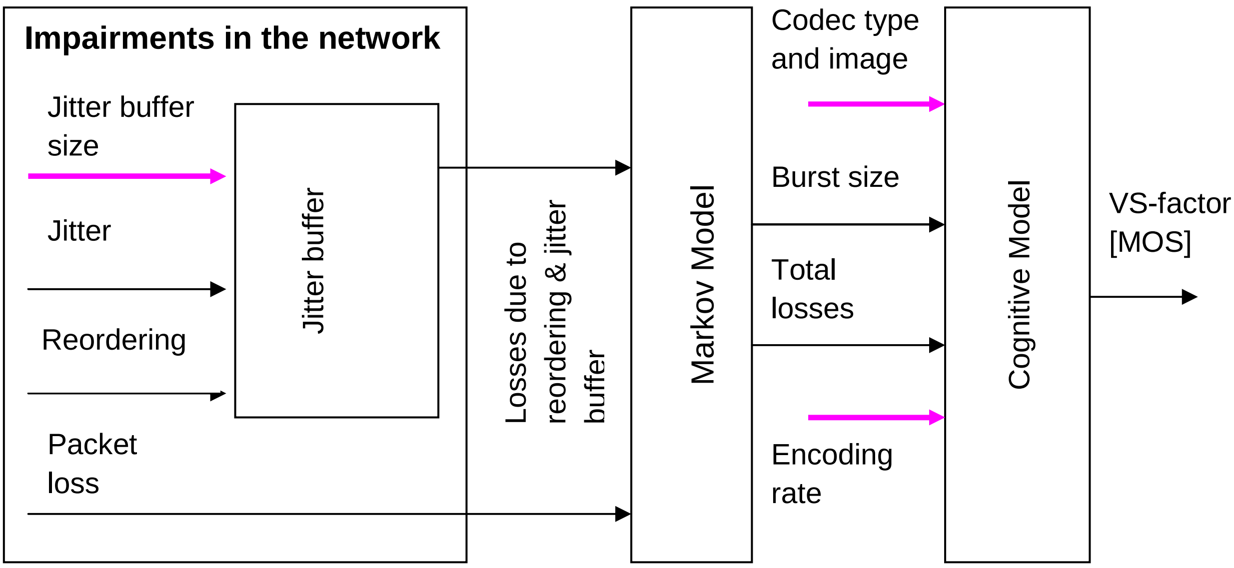

Fig. 1. Block diagram for the VS Model

In practice, it is assumed that when the QoS is determined, the packet streams from an RTP session are collected by a protocol analyser and then passed on to a suitable evaluation tool. The new VS Model is such a tool. It works on the following principle: All network impairments are collected and processed in the first block of the diagram. The effects of jitter and out-of-order packet delivery are converted into losses, not forgetting that some errors can be compensated with the aid of the jitter buffer. The values attained in this block and the packet losses from the network are passed on to the second block where total losses and burst size are determined. For the VS Model we assume the Markov property "memorylessness", that is widely used in analyses of networks. The ensuing recalculated parameters are passed on to the third and final block. Further inputs for the third block include information about the image size, the codec type and the encoding rate that are being used. These data are gained from measuring the RTP streams. The last block, called "Cognitive Model" calculates and outputs the VS-factor as a value on the MOS scale [12].

The mathematical dependencies needed to do this are stored in the block in the form of a table. Its contents are calculated through the following steps:

Step 1 Using a suitable too such as the software tool from paper [10] for instance, determine the PEVQ curves as a function of packet losses, burst size and encoding rate for various codecs and image formats. The curves serve as a basis for further calculations. This approach replaces analyses in special studios in which groups of people are needed to voice a subjective opinion. This solution saves time and money when it comes to developing parametrised QoS/QoE models.

Step 2 Approximate each of the PEVQ curves as a function of packet losses at burst sizes of "1" to "5" and selected encoding rates using formula (1). The approximation method "least squares" was used.

$$VS - factor = P \cdot e^{\frac{a \cdot packetloss}{burstsize}} + Q \cdot e^{\frac{b \cdot packetloss}{burstsize}} \tag{1}$$

The constants a and b are selected so that they display values equal to or smaller than 0, with significantly smaller values being chosen for b. The result of this is that the two summands are responsible for the steepness of the curves when packet loss is small (2nd summand) or large (1st summand). All constants (P, Q, a, and b) are now calculated iteratively as best possible values for each encoding rate.

Step 3 Determine the formulas for the constants P, Q, a and b. Once calculated the constants are recorded as functions of the encoding rate. Their corresponding formulas can then be calculated by means of polynomial approximation. The degree of the polynomial is caused by the complexity of the curve. Formulas (2) to (5) show the simple relationship:

$$P = w_n \cdot Bitrate^n + w_{n-1} \cdot Bitrate^{n-1} + ... + w_1 \cdot Bitrate^1 + w_0 \tag{2}$$

$$Q = z_n \cdot Bitrate^n + z_{n-1} \cdot Bitrate^{n-1} + ... + z_1 \cdot Bitrate^1 + z_0 \tag{3}$$

$$a = x_n \cdot Bitrate^n + x_{n-1} \cdot Bitrate^{n-1} + ... + x_1 \cdot Bitrate^1 + x_0 \tag{4}$$

$$b = y_n \cdot Bitrate^n + y_{n-1} \cdot Bitrate^{n-1} + ... + y_1 \cdot Bitrate^1 + y_0 \tag{5}$$

The steps described here will now be demonstrated using as an example the video codec H.263, the CIF image format and the numerical tool described in paper [10]. In this investigation the following values will be assumed for encoding rates: 305 kbps, 1702 kbps, 3809 kbps, 4978 kbps and 7413 kbps. The burst size displays an exponential distribution with mean values of 1 to 5. The packet losses are subject to a binomial distribution with probability P. 31 calculations were performed per packet loss value. This means that at a probability of error of 5 %, confidence intervals can be achieved that are less than 10 % of the mean values under investigation. The results obtained for 3 selected encoding rates are presented in Figs. 2 to 4.

Figs. 2 to 4 show that all QoE curves develop exponentially. The curves also show that the encoding rate has a great influence on the QoE values. The small encoding rate delivers very poor QoE values which are not acceptable in the praxis. Burst size also has a significant influence on QoE. A burst size of 1 delivers the worst quality of experience in all three cases and improves with increasing burst size. When small groups of packet losses occur frequently, the synchronisation of I/P/B images breaks off more often than when infrequent large groups of packet losses occur. The more often synchronization breaks off, the more frequently a static image will occur, which is reflected in a deterioration of the QoS/QoE.

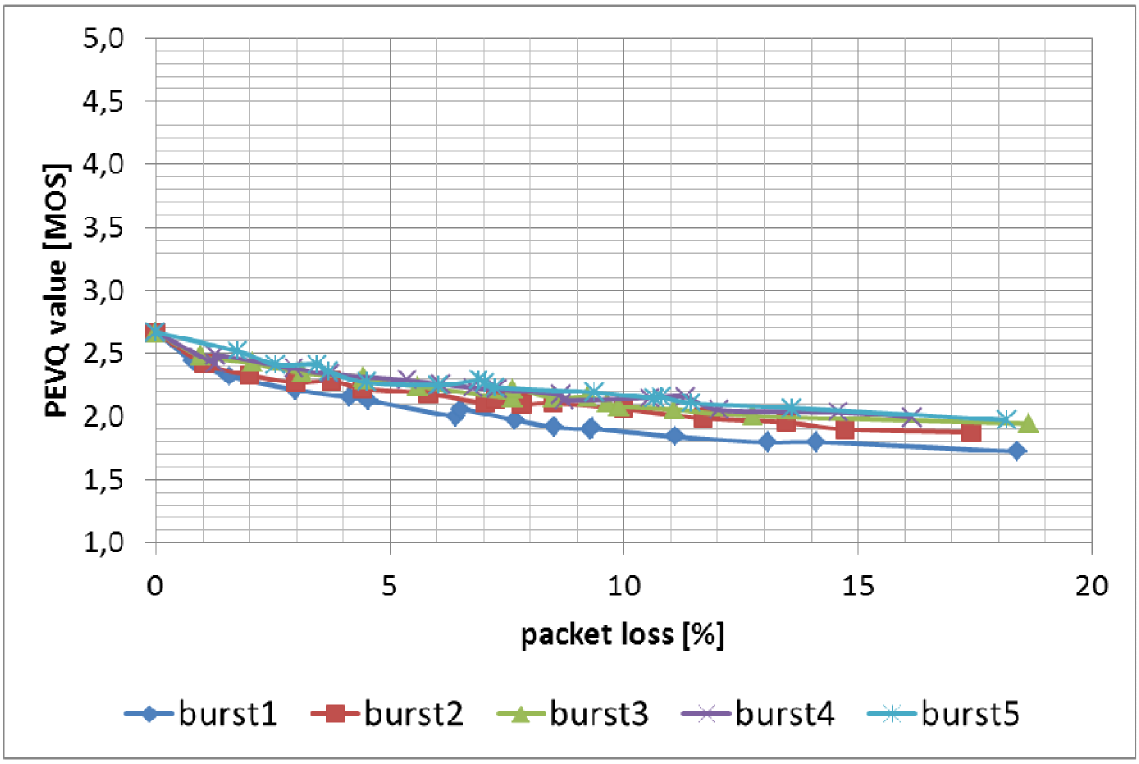

Fig. 2. PEVQ values as a function of packet losses and burst sizes at an encoding rate of 305 kbps for the Codec H.263 and the image format CIF

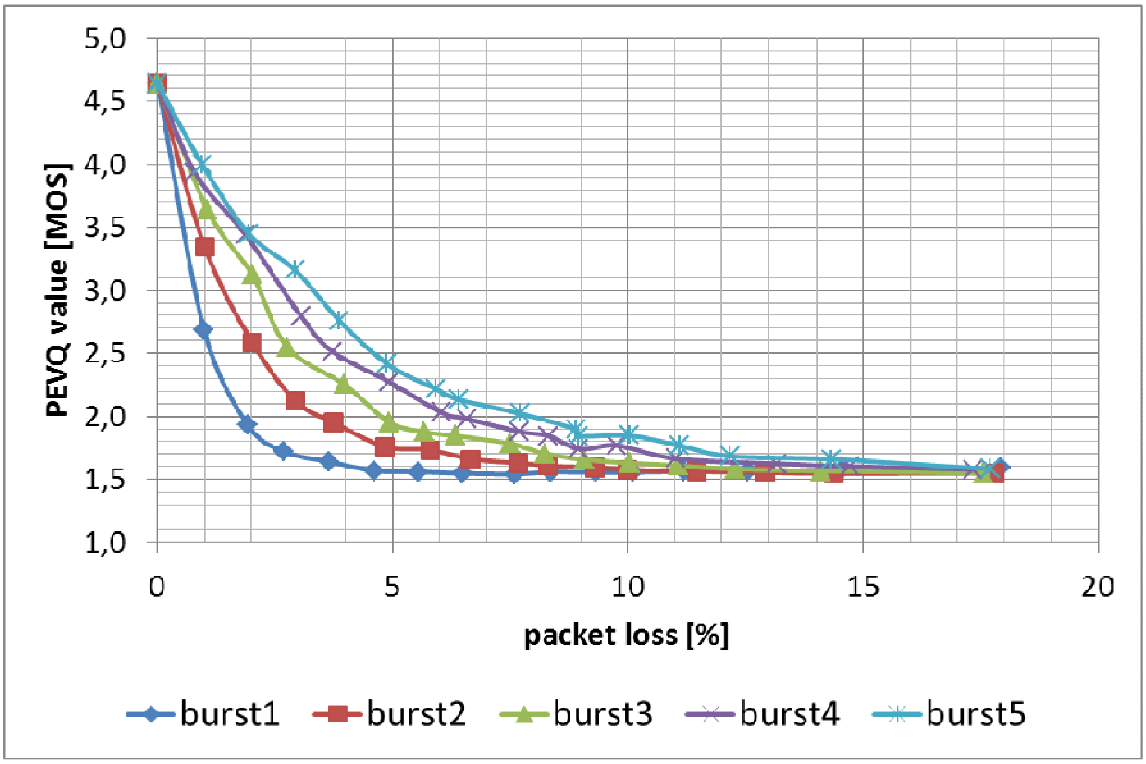

Fig. 3. PEVQ values as a function of packet losses and burst sizes at an encoding rate of 3809 kbps for the Codec H.263 and the image format CIF

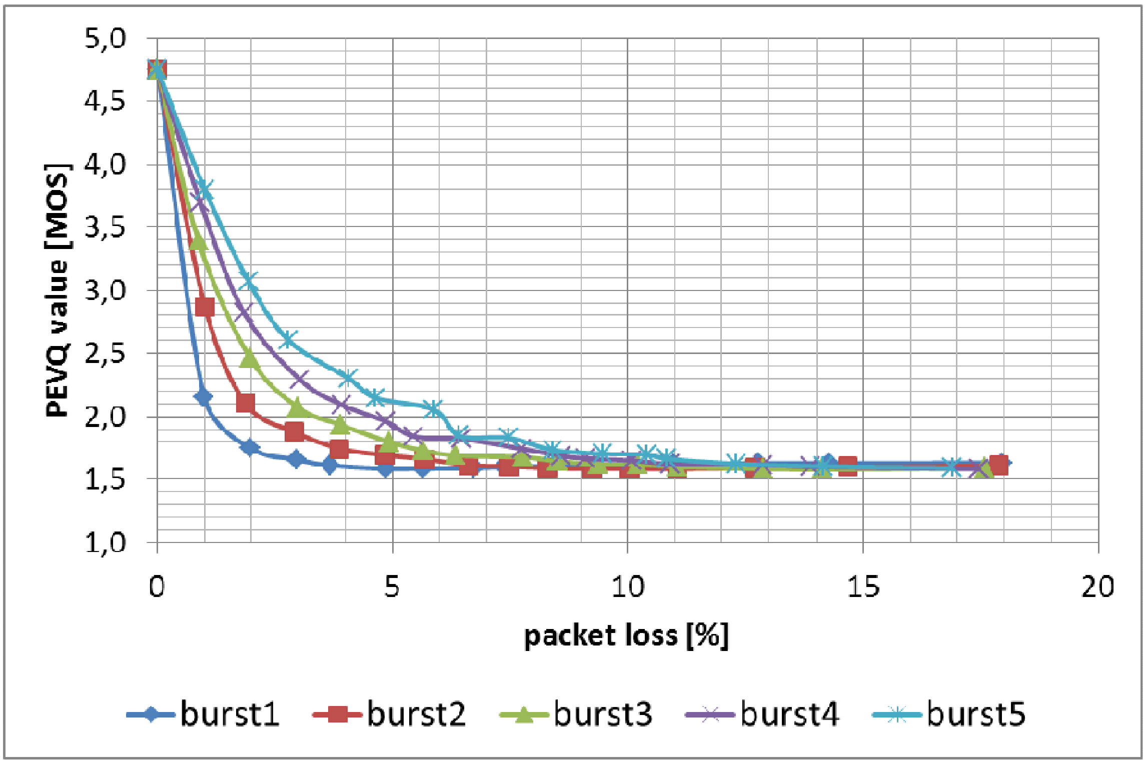

Fig. 4. PEVQ values as a function of packet losses and burst sizes at an encoding rate of 7413 kbps for the Codec H.263 and the image format CIF

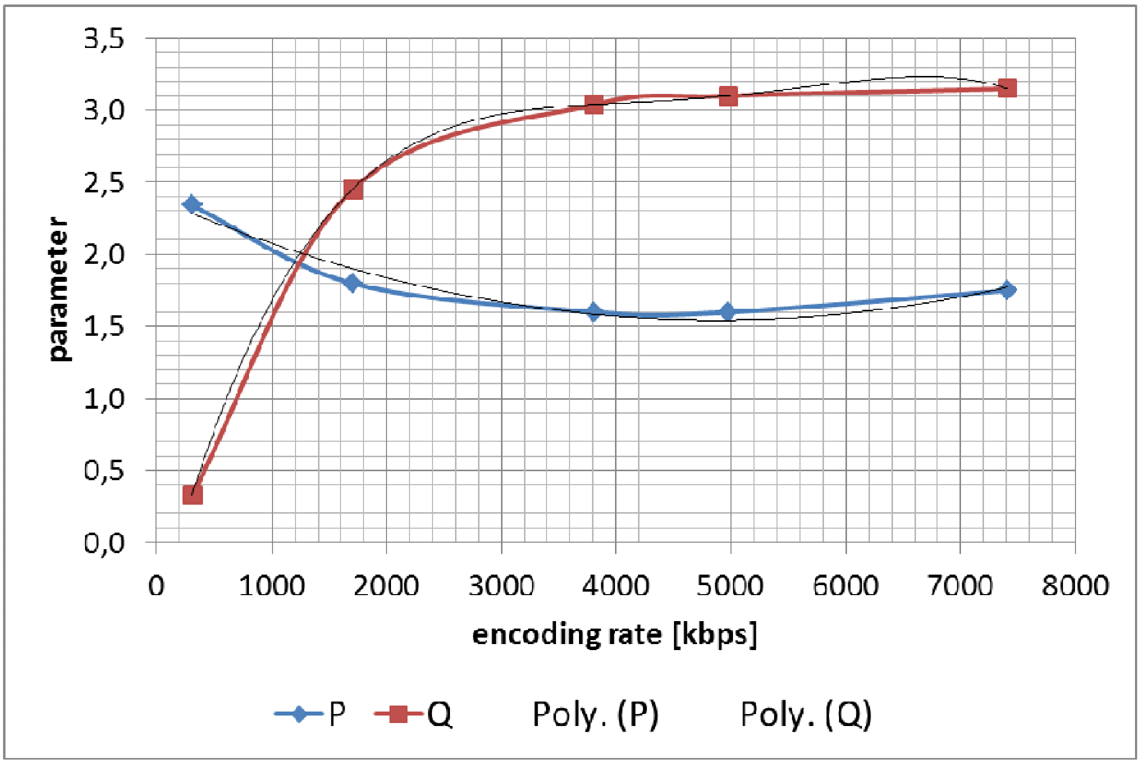

Fig. 5. Approximation of parameters P and Q as functions of the encoding rate for the Codec H.263 and the image format CIF

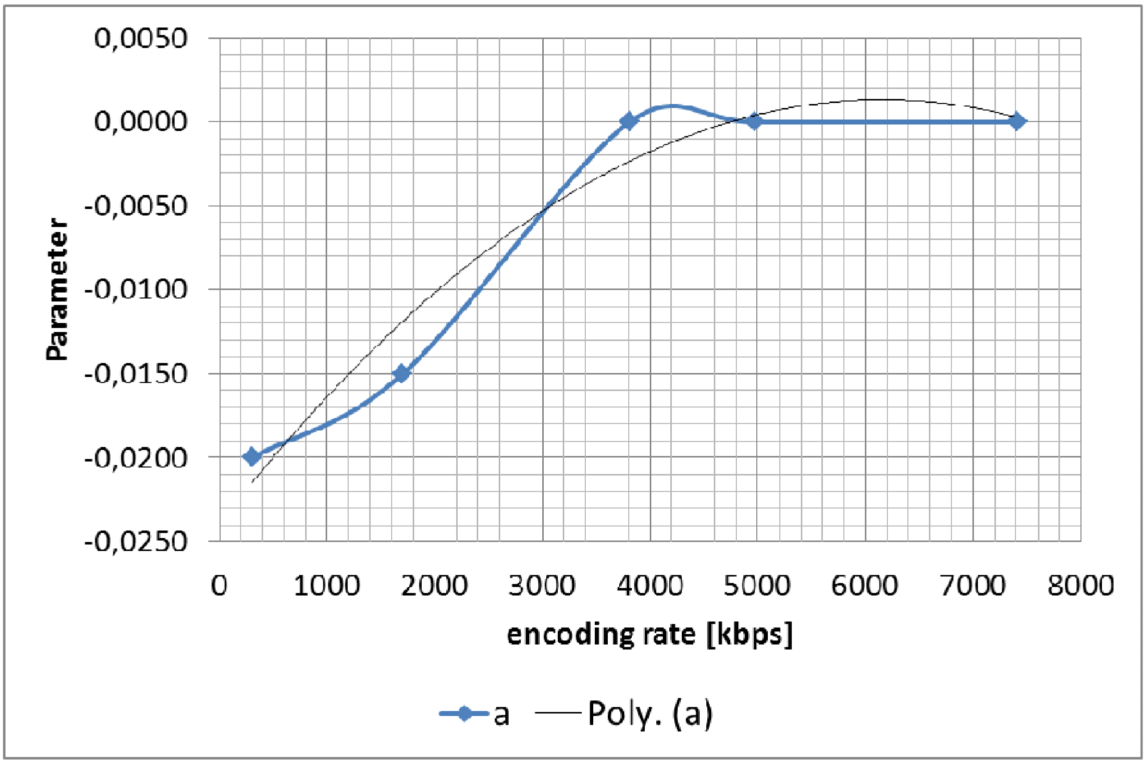

Fig. 6. Approximation of parameter a as functions of the encoding rate for the Codec H.263 and the image format CIF

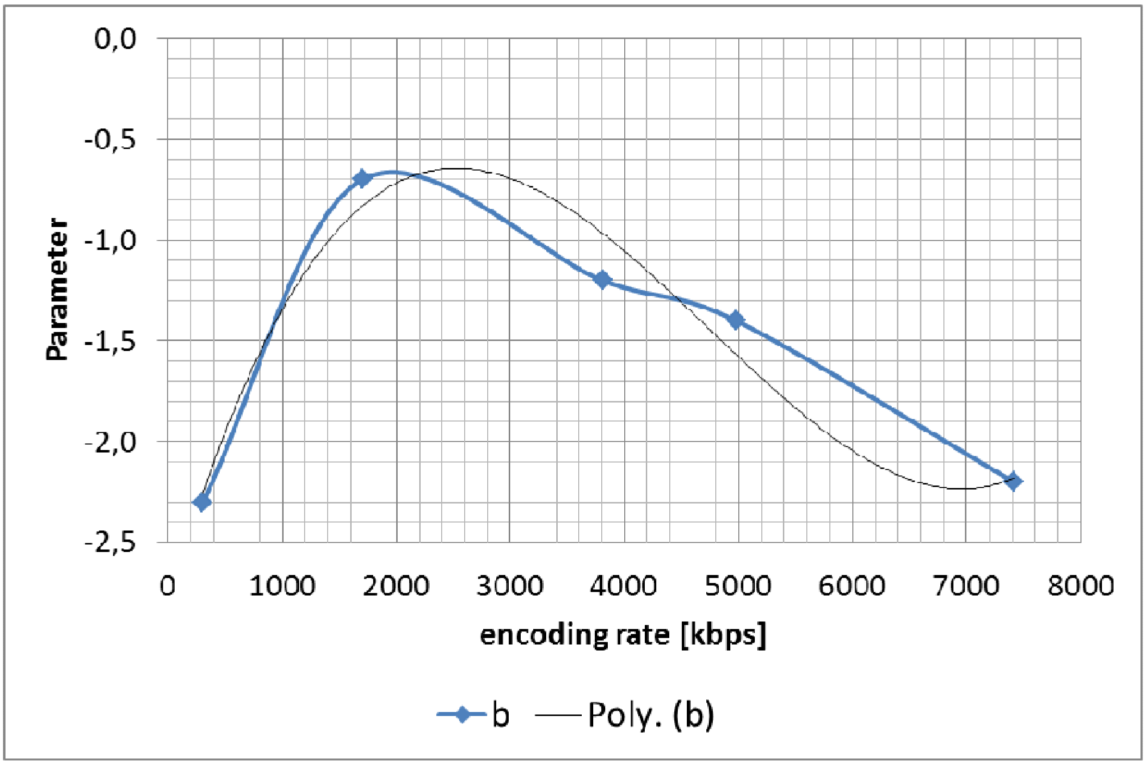

Fig. 7. Approximation of parameter b as functions of the encoding rate for the Codec H.263 and the image format CIF

Figs. 5 to 7 show the parameters superimposed on the encoding rate and the results obtained from the approximations that were conducted using Equations (2) to (5). The actual formulas for the Codec H.263 and the image format CIF are therefore:

$$P = 3{,}54 \cdot 10^{-8} \cdot Bitrate^{2} - 3{,}45 \cdot 10^{-4} \cdot Bitrate + 2{,}39 \qquad (6)$$

$$Q = -7{,}02 \cdot 10^{-15} \cdot Bitrate^{4} + 1{,}36 \cdot 10^{-10} \cdot Bitrate^{3} - 9{,}66 \cdot 10^{-7} \cdot Bitrate^{2} + 3{,}02 \cdot 10^{-3} \cdot Bitrate - 0{,}51 \qquad (7)$$

$$a = -7{,}00 \cdot 10^{-10} \cdot Bitrate^{2} + 8{,}00 \cdot 10^{-6} \cdot Bitrate - 2{,}39 \cdot 10^{-2} \qquad (8)$$

$$b = 3{,}68 \cdot 10^{-11} \cdot Bitrate^{3} - 5{,}23 \cdot 10^{-7} \cdot Bitrate^{2} + 1{,}94 \cdot 10^{-3} \cdot Bitrate - 2{,}80 \qquad (9)$$

3. Comparison Study

3.1. Numerical Analyses

The software tool presented in paper [10] was used here, too. The following parameters were assumed for the numerical comparison study:

- # Nondeterministic distributed packet losses of 0 to 20 % and constant burst size of 1 at an encoding rate of 1702 kbps
- # Nondeterministic distributed packet losses of 0 to 20 % and nondeterministic burst size of 2 at an encoding rate of 1702 kbps
- # Nondeterministic distributed packet losses of 0 to 20 % and constant burst size of 1 at an encoding rate of 4978 kbps
- # Nondeterministic distributed packet losses of 0 to 20 % and nondeterministic burst size of 2 at an encoding rate of 4978 kbps
- # Video codec H.263
- # Image format CIF
- # Image refresh rate of 25 images/s
- # 31 measurements per value of each of the variables (here: packet losses). This ensures that confidence intervals are achieved that are less than 10 % of the mean values under analysis (with a probability of error of 5 %).
- # PEVQ and VS Models as the QoS measuring techniques
- # As the reference video an AVI file from the company Opticom [7] was chosen. The file has duration of 8 seconds and a resolution of 352 x 288. Figure 8 shows a screenshot of the reference video.

Fig. 8. Screenshot of the reference video

The results of the comparison study are presented graphically in Figs. 9 to 12.

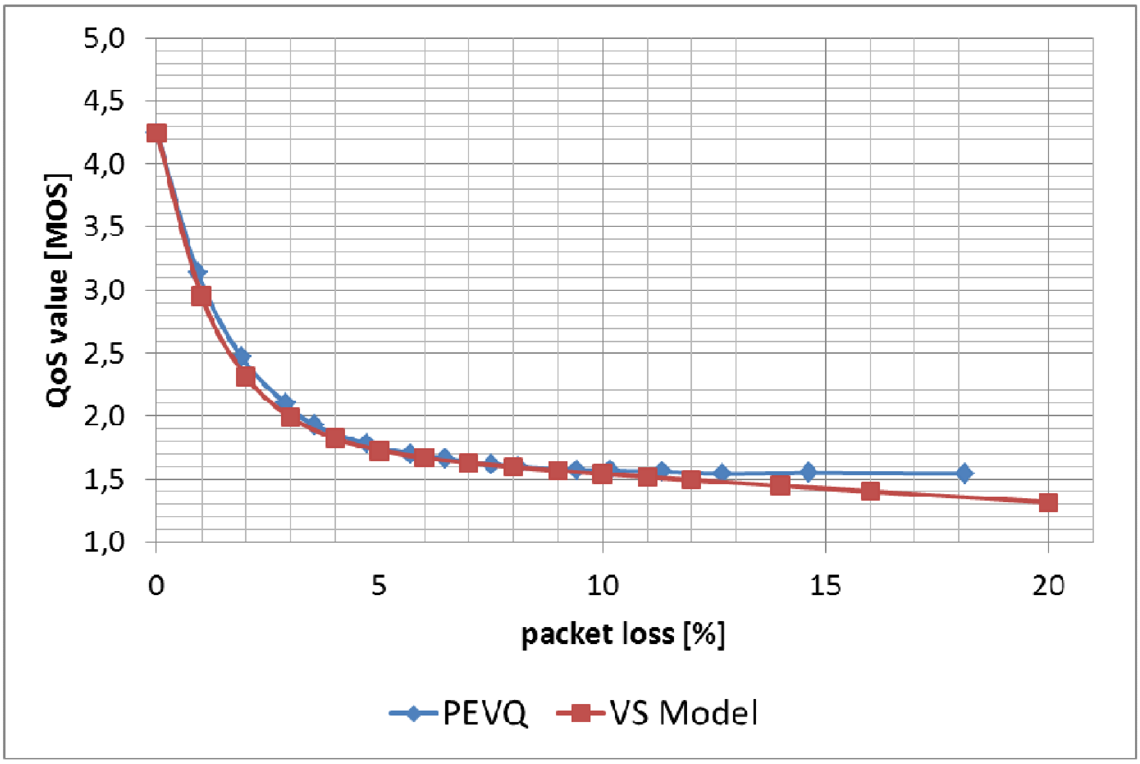

Fig. 9. QoS values as a function of packet losses gained from different measuring methods for the Codec H.263, the image format CIF, burst size 1 and an encoding rate of 1702 kbps (numerical analysis)

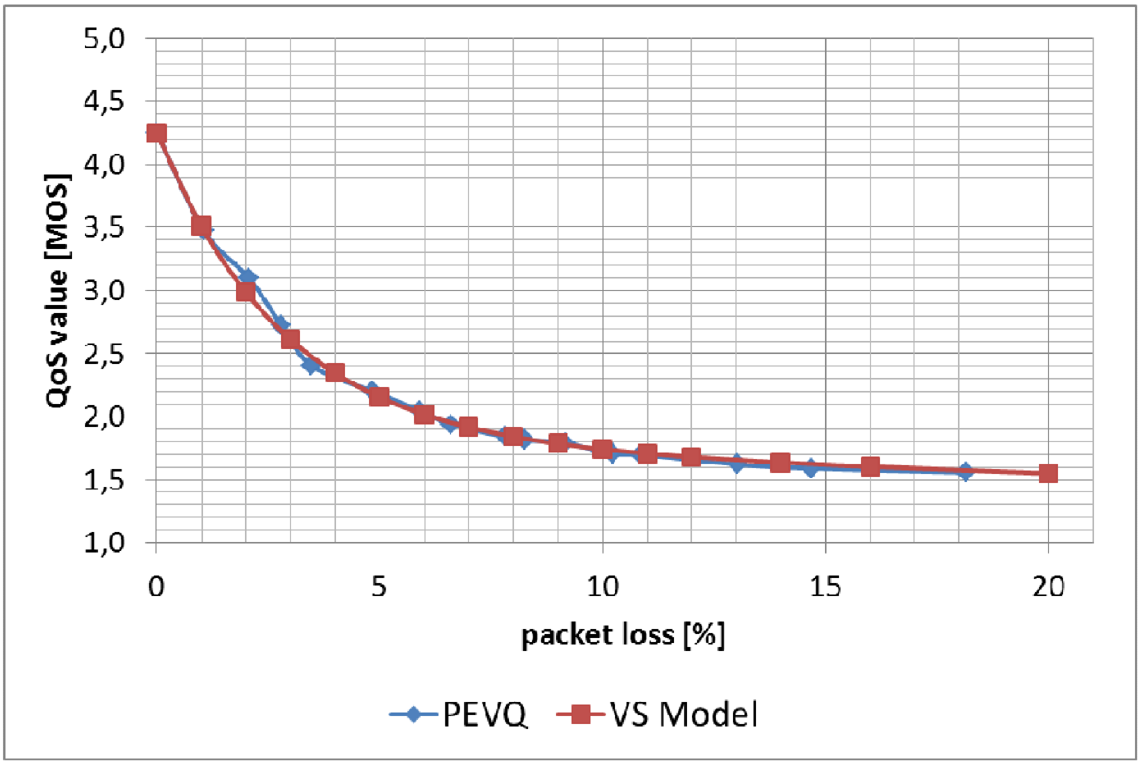

Fig. 10. QoS values as a function of packet losses gained from different measuring methods for the Codec H.263, the image format CIF, burst size 2 and an encoding rate of 1702 kbps (numerical analysis)

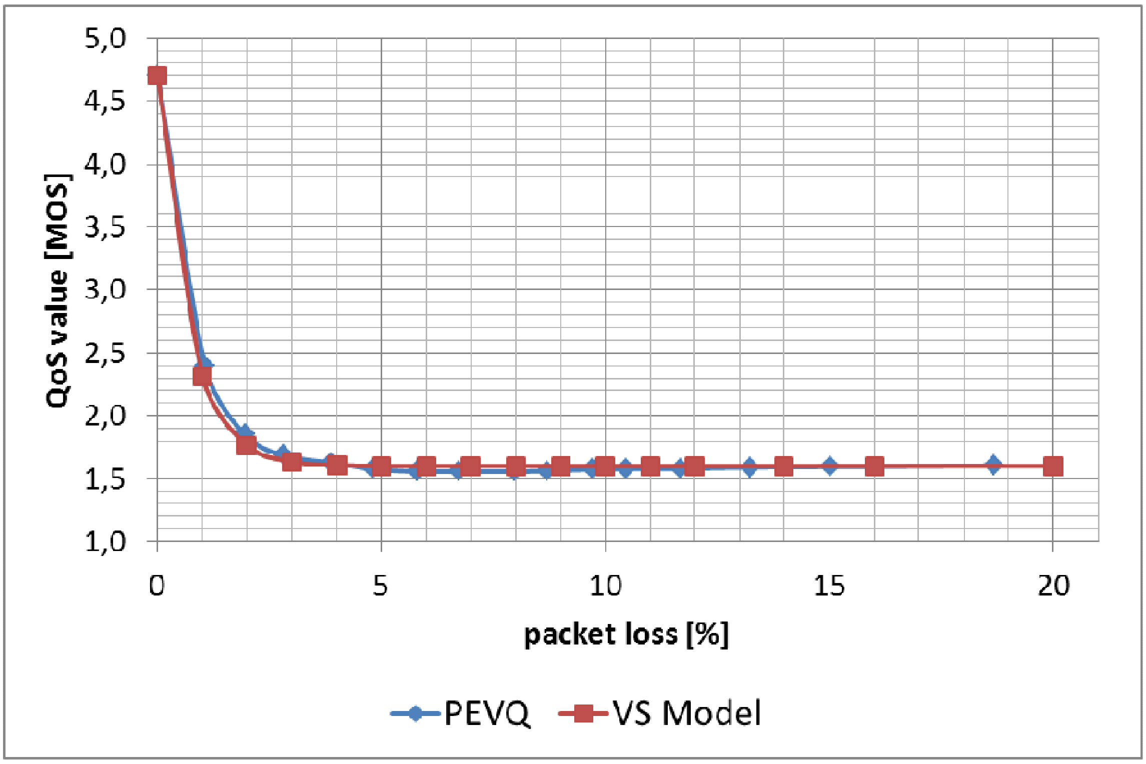

Fig. 11. QoS values as a function of packet losses gained from different measuring methods for the Codec H.263, the image format CIF, burst size 1 and an encoding rate of 4978 kbps (numerical analysis)

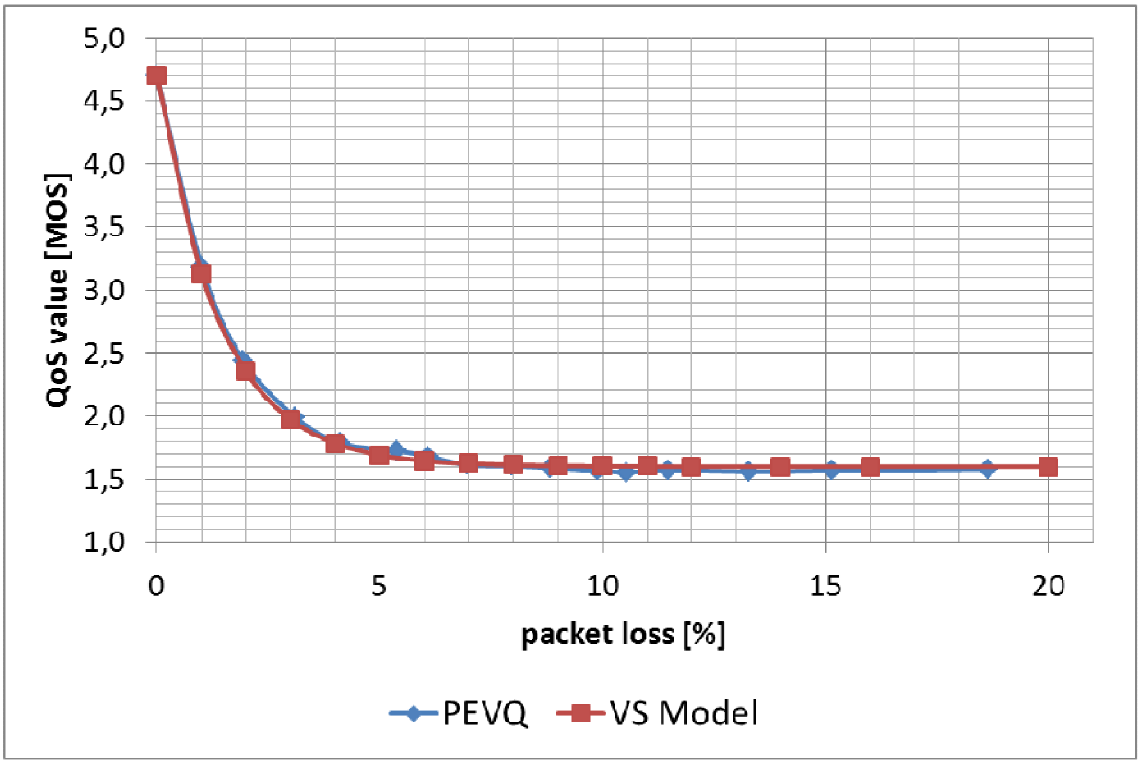

Fig. 12. QoS values as a function of packet losses gained from different measuring methods for the Codec H.263, the image format CIF, burst size 2 and an encoding rate of 4978 kbps (numerical analysis)

Figs. 9 to 12 show that the quality of service deteriorates exponentially as packet losses increase. This is the case for both QoS measuring methods used here. Furthermore, the curves fall less steeply as burst size increases. The upshot of this is: it is far better for the service if fewer, larger bundles of packets are lost than lots and lots of smaller bundles. The reason for this lies in the properties of human vision. The PEVQ and the VS Model curves proceed very close to each other, meaning that the numerical comparison study has proved that the VS Model is quite suitable for practical use.

3.2. Analyses in the real Environment

Fig. 13 shows the real environment used in the analysis. The Video Streaming Quality Measurement System (VSQMS), that was described in detail in paper [13], is the mainstay of the environment. The measurement system is capable of establishing an RTP connection between two VoIP stations and using it to transmit video streams encoded according to H.263. The VSQMS client functions here as a "mirror" that sends the incoming RTP packets back to the VSQMS agent. The agent comprises both the PEVQ Algorithm and the new VS Model. Both are methods to determine values for QoE respectively QoS.

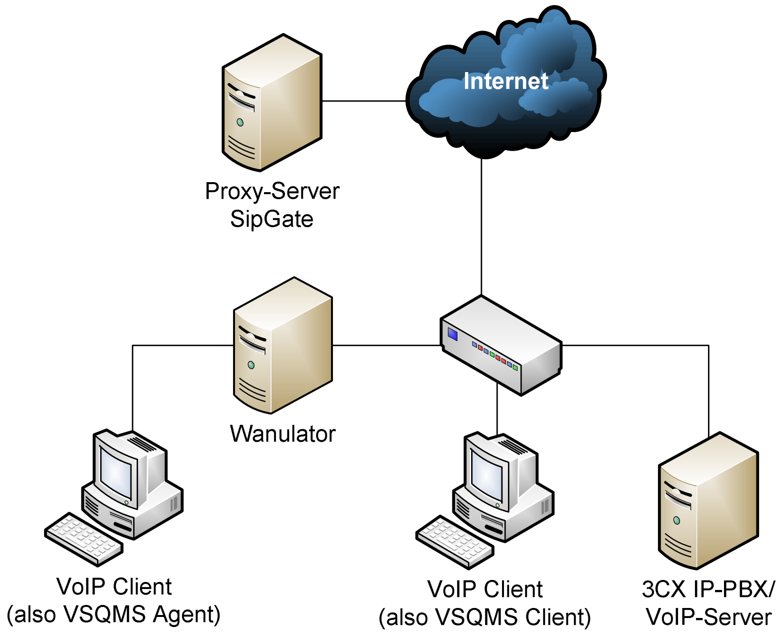

Fig. 13. The real environment

Figs. 14 and 15 show examples of the results obtained from the real analysis environment. The curves from Fig. 14 show a remarkable agreement with the corresponding curves from the numerical analyses (cf. Fig. 9). Comparing Figs. 10 and 15 leads to similar conclusions, i.e. increases in burst size lead to higher QoS values. Here it becomes evident that the two measuring methods yield comparable QoS values in a real environment. Consequently, the usability of the new VS Model in real-life situations has been proved here as well.

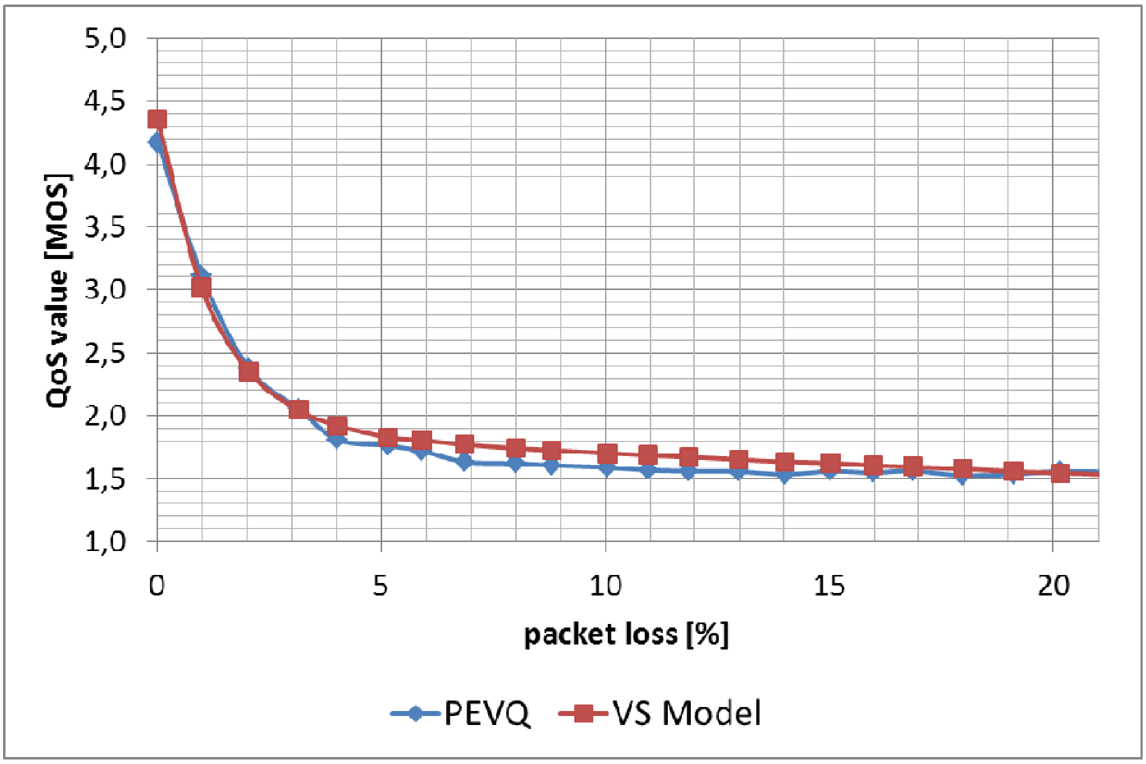

Fig. 14. QoS values as a function of packet losses obtained from different measuring methods for the Codec H.263, image format CIF, burst size "1" and an encoding rate von 1702 kbps (real environment)

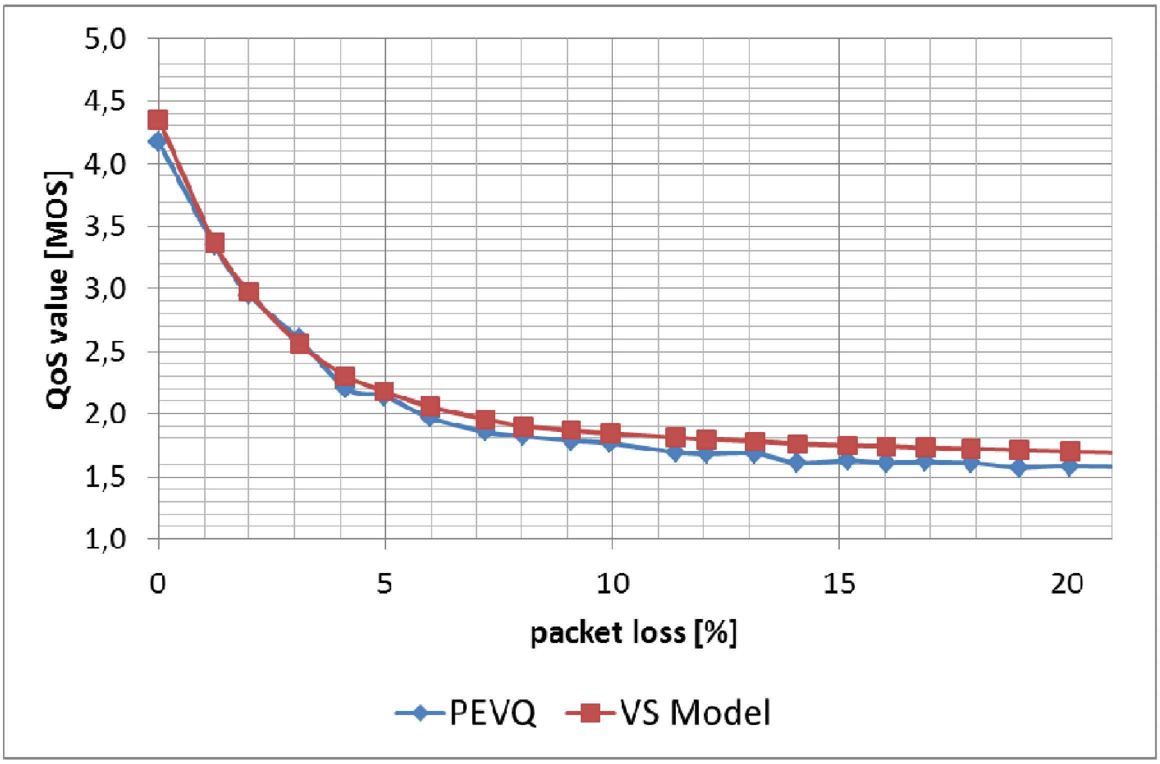

Fig. 15. QoS values as a function of packet losses obtained from different measuring methods for the Codec H.263, image format CIF, burst size "2" and an encoding rate von 1702 kbps (real environment)

4. Summary and Outlook

In the course of the work described in this paper a new parametrised QoS measuring model was developed to evaluate the quality of video streams in the video-telephony service and its functionality was proved in a real-environment analysis. When the new model was defined, all due consideration was paid to the typical parameters for the VToIP service. The new VS Model is based on the PEVQ curves. This is of especial practical significance since the PEVQ is considered to be the most objective QoS measuring method for video. The comparison study has proved the practicability of the new QoS model beyond any shadow of doubt. The new, quick and easy-to-use and inexpensive VS Model can definitely be seen as a superior alternative to the laboriously slow PEVQ method with its expensive licences.

In order to be able to assess the QoS of the VToIP service in its entirety, it is necessary in practice to consider the quality of the audio streams as well. The ITU-T Recommendation G.1070 [5] contains suggestions as to how this might be achieved. Paper [14] introduced the new E(IP) Model. It is very a very effective tool for determining the quality of audio streams in a VoIP environment. It is an obvious next step to combine the two models to create a joint VT Model (VT = Video-telephony) to analyse the complete VToIP service. Research work towards that end has already begun at the Flensburg University of Applied Sciences.

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