BULLETIN OF THE POLISH ACADEMY OF SCIENCES
TECHNICAL SCIENCES, Vol. 63, No. 2, 2015
DOI: 10.1515/bpasts-2015-0057

The new, parametrised IPTV Model for Determining the Quality in the IPTV Service

T. UHL[1]* and H.S. JÜRGENSEN[2]

[1] Maritime University of Szczecin, 11 Henryka Pobożnego St., 70-507 Szczecin, Poland
[2] Company "Nextragen" GmbH, Flensburg, 2 Lise-Meitner-St., D 24941 Flensburg, Germany

Abstract. This paper describes a new measurement method for determining the quality of the IPTV (Television over IP) service. This method uses the latest IPTV Model and is one of a number of parameter-based measuring techniques (offline operation, i.e. without intrusive measurement). It delivers results that come very close to those of the corresponding Perceptual Evaluation of Video Quality (PEVQ) curves. The method is quickly and easily implemented – one of the great advantages of using this method to measure QoS.

Key words: communication networks, communication services, IP transport platform, QoS measurement techniques.

1. Introduction

The notion of Quality of Service (QoS) plays a vital role in the newest digital networks. The term can be found everywhere, among other things in the definition of Next Generation Networks according to the ITU-T Standard Y.2001 [1]. 25th November 2009 the European Parliament and the European Council adopted the so-called Communications Package that includes: Directive 2009/140/EC [2] and Directive 2009/136/EC [3], in which high priority was placed on Quality of Service.

QoS and QoE (Quality of Experience) should be continuously monitored in modern networks, and preferably automatically. Specialized measuring systems and methods are obviously indispensable for that. When it comes to video communications (IPTV being one such service) there are already several methods available for measuring QoS and QoE. They are for the most part standardized, e.g. ITU-T Rec. J.144 [4], ITU-T J.147 [5], ITU-T J.247 [6]. All of these standards have evolved from signal-based QoE measuring methods with "Full Reference". That is what makes these measuring techniques so complicated, time-consuming and complicated to use. There is another method for measuring video services that is awaiting standardization: the PEVQ Algorithm (Perceptual Evaluation of Video Quality) [7]. It is also a signal-based measuring method. According to the German licence holders, Opticom [8], it already complies with Recommendation J.247. A look at its specifications [6] reveals this method to be one of the best, making it the first choice for the measuring QoS in the IPTV service. Another well-known QoS measuring method is Recommendation TR 101 290 [9], standardized by ETSI. It works using for the most part network impairment parameters. There are at present no other simple parametrised methods for measuring QoS which utilize service parameters in addition to using network parameters. The authors aim to satisfy that need with the IPTV Model described in this paper.

Transporting content via the IP transport platform presents an enormous challenge to the IPTV service. At present, there are two techniques designed to do this: MPEG-2 Transport Stream according to ISO/IEC 13818-1 [10] and Native RTP according to IETF RFC 3640 [11]. A previous paper [12] includes an interesting comparison of the two, and demonstrates just how sensitive content coded according to the MPEG-4/AVC Codec can be to the transport technique used. The paper [12] also suggests what optimizations must be made.

The present paper incorporates those insights. To begin with the new Model is formulated and illustrated within the context of an actual video codec. The Model is then analysed in various applications and its practicability is put to the test in a comparison study. The results gained from the analyses are presented graphically, and interpreted. The paper concludes with a summary and an outlook on future work.

2. The new, parametrised IPTV model

Paper [13] describes in detail a large-scale investigation which demonstrated that packet losses are some of the worst impairment parameters in a VToIP environment. This is true of both audio and video transmission. Other parameters that greatly affect QoS values include codec type, encoding rate and burst size. The size of the jitter buffers in the terminal equipment has a significant influence on QoS as well. Not only these factors must be considered when formulating a new, parametrised model for determining QoS in IPTV but also further parameters specific to the IPTV service, such as the type of the transport stream used. Figure 1 shows the newly established, parametrised IPTV Model for determining the quality of video streams in the IPTV service. The main impetus for its creation came from the VS Model [14], designed for the VToIP service.

*e-mail: t.uhl@am.szczecin.pl

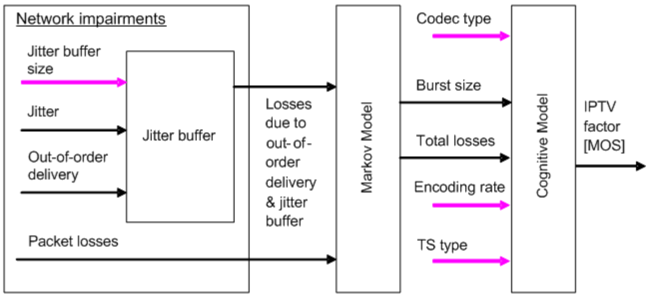

Fig. 1. Block diagram for the IPTV model

When determining QoS in practice, it is widely assumed that packet streams from a multicast session (and IPTV uses multicast sessions) are collected by a protocol analyser and then passed on to a suitable evaluation tool. The new IPTV 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, bearing in mind that this errors can be smoothed with the aid of the jitter buffer. The values calculated in this block are passed on, along with the packet losses from the network, to the second block, where total losses and burst size are determined. The IPTV Model implements the Markov property "memorylessness", which 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 codec type, and the encoding rate and the type of transport stream that are being used. These data are gained from measuring the multicast streams. In the block called "Cognitive Model" the IPTV factor is calculated and outputted as a value on the MOS scale [15]. 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 tool the PEVQ curves are determined as a function of packet losses, burst size and encoding rate for various codecs and types of transport streams. These curves serve as a basis for further calculations.

Step 2 Each PEVQ curve from Step 1 is described in terms of a polynomial. The degree of the polynomial is determined by the complexity of the curve. Now approximate each of the PEVQ curves as a function of packet losses for burst sizes of "1" to "5" and selected encoding rates using Eq. (1)

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

The constants a and b are selected so that they display values equal to or less than zero, 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 losses are few (2^{nd} summand) or many (1^{st} summand). The constants P and Q assume values from the interval *(0,* $QoS_{\max}$]. In addition, the sum of P and Q must always yield the value $QoS_{\max}$. All constants (P, Q, a, and b) are now calculated iteratively as best possible values for each encoding rate. The approximation method "least squares" will be used, and that is done as follows. An assumed initial set of parameters P, Q, a, b are inserted into Eq. (1), beginning with a burst size of "1". One finds the squared error between the corresponding curve from Step 1 and the curve just examined in Eq. (1). This is repeated for further burst sizes. Finally, the cumulative squared error is found and saved with the current set of parameters. The next set of parameters is then taken and the iteration described above is repeated. Of course, one variable may be different in the new parameter set. Once all the prepared parameter sets have been used up, it becomes a case of finding the parameter set for which the cumulative squared error is smallest. The optimum has been found! Now the next encoding rate is used and the entire procedure described above is repeated.

Step 3 Determine the formulas for the constants P, Q, a and b. That is done as follows. The optimum constants determined in Step 2 are recorded as functions of the encoding rate. The corresponding formulas can then be calculated by means of polynomial approximation. The degree of the polynomial is determined 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 with an example in which the video codec H.264/AVC has the preset "Medium", HD resolution (720p = 1280×720), and uses the MPEG-2-TS in conjunction with the tool from Paper [16]. In this examination the following values will be assumed for the encoding rates: 2125 kbps, 5175 kbps and 7000 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 the probability P. 31 calculations are performed for each packet loss value. This means that at 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 represented in Figs. 2 to 4.

Figures 2 to 4 show that all QoS curves progress exponentially. The encoding rate exerts a substantial influence on QoE values here, especially in a lossless environment. It is also evident that burst size has an equally large influence on QoE values, with a burst size of "1" producing the worst quality of service in all three cases and increases in burst size leading to improvements in service quality. This is in keeping with the psycho-visual model of the human visual perception system, which asserts that viewers will accept a few isolated major disruptions in reception far more readily than numerous, regularly occurring minor disturbances.

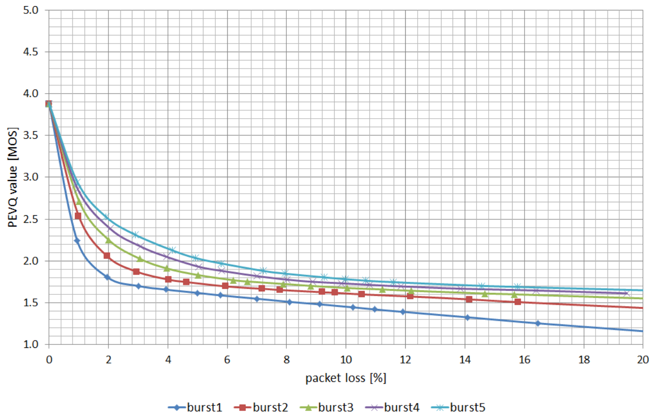

Fig. 2. PEVQ values as a function of packet losses and burst sizes at an encoding rate of 2125 kbps for the codec H.264/AVC and the MPEG-2-TS

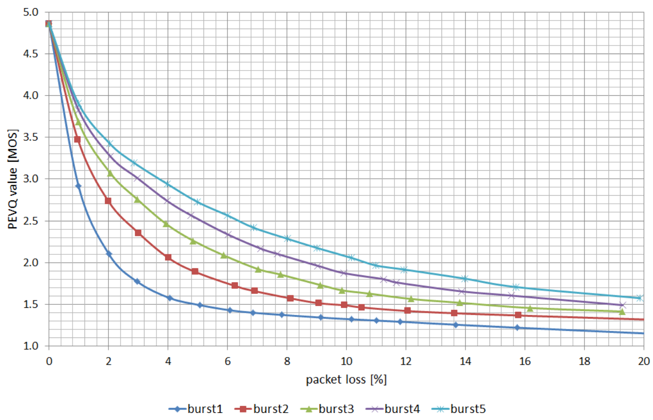

Fig. 3. PEVQ values as a function of packet losses and burst sizes at an encoding rate of 5175 kbps for the codec H.264/AVC and the MPEG-2-TS

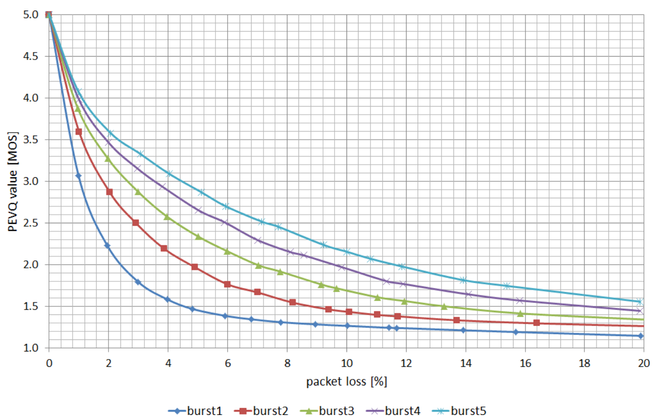

Fig. 4. PEVQ values as a function of packet losses and burst sizes at an encoding rate of 7000 kbps for the Codec H.264/AVC and the MPEG-2-TS

Figures 5 to 7 show the parameters as a function of the encoding rate and the results obtained from the approximations made in formulas (2) to (5).

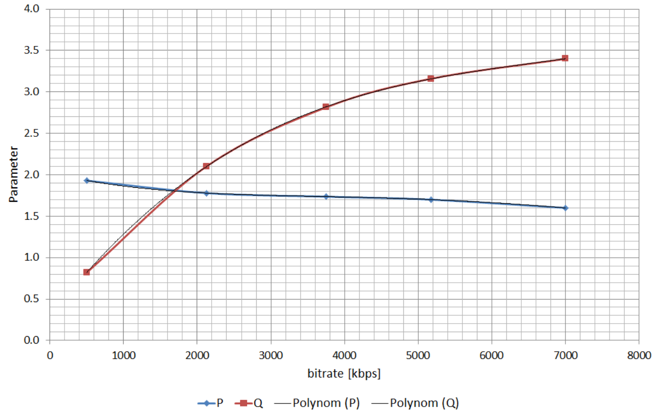

Fig. 5. Approximation of parameters P and Q as functions of the encoding rate for the codec H.264/AVC and the MPEG-2-TS

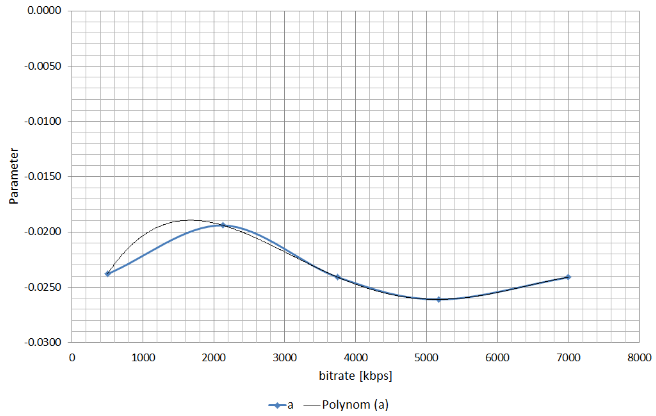

Fig. 6. Approximation of the parameter a as a function of the encoding rate for the Codec H.264/AVC and the MPEG-2-TS

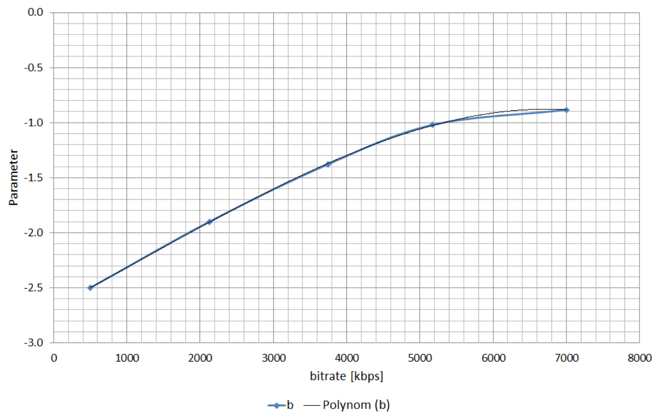

Fig. 7. Approximation of the parameter b as a function of the encoding rate for the codec H.264/AVC and the MPEG-2-TS

The actual equations for the codec H.264/AVC and the MPEG-2-TS are therefore:

$$P = 3.61 \cdot 10^{-16} \cdot bitrate^4 - 8.46 \cdot 10^{-12} \cdot bitrate^3 + 6.36 \cdot 10^{-8} \cdot bitrate^2 - 2.15 \cdot 10^{-4} \cdot bitrate + 2.02, \tag{6}$$

$$Q = 7.70 \cdot 10^{-12} \cdot bitrate^3 - 1.54 \cdot 10^{-7} \cdot bitrate^2 + 1.14 \cdot 10^{-3} \cdot bitrate + 0.29, \tag{7}$$

$$a = -6.39 \cdot 10^{-17} \cdot bitrate^4 + 1.21 \cdot 10^{-12} \cdot bitrate^3 - 7.54 \cdot 10^{-9} \cdot bitrate^2 + 1.63 \cdot 10^{-5} \cdot bitrate - 0.03, \tag{8}$$

$$b = -4.11 \cdot 10^{-12} \cdot bitrate^3 + 1.57 \cdot 10^{-8} \cdot bitrate^2 + 3.48 \cdot 10^{-4} \cdot bitrate - 2.68. \tag{9}$$

3. Comparison study

The software tool from paper [16] was used in the following analyses as well. The following parameters were assumed for the numerical comparison study:

Nondeterministic distributed packet losses of 0 to 20% and a constant burst size of "1" at an encoding rate of 5251 kbps.
Nondeterministic distributed packet losses of 0 to 20% and a nondeterministic burst size of "3" at an encoding rate of 5251 kbps.
Nondeterministic distributed packet losses of 0 to 20% and a constant burst size of "1" at an encoding rate of 10423 kbps.
Nondeterministic distributed packet losses of 0 to 20% and a nondeterministic burst size of "3" at an encoding rate of 10423 kbps.
Video codec H.264/AVC with preset "Medium".
HD image format 720p (see Fig. 8).
Image refresh frequency 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 IPTV Models as the QoE/QoS measuring techniques.

Fig. 8. Screenshot of the reference video (see source Ref. 8)

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

Figures 9 to 12 show that QoS deteriorates exponentially as packet losses increase. This is the case for both QoE/QoS measuring techniques used. Furthermore, the curves fall less steeply as burst size increases. The reason for this is that synchronization of I/P/B images fails more frequently when small groups of packet losses regularly occur than when large groups of packet losses occur infrequently. And the more numerous breakdowns in synchronization are, the more frequently the images will freeze. This is naturally reflected in a drop in QoE/QoS values.

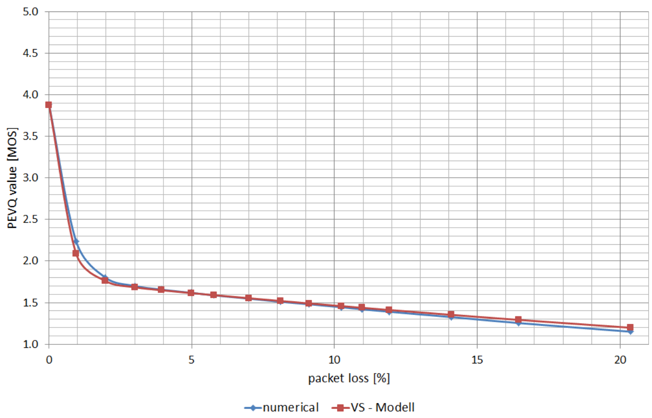

Fig. 9. QoS values as functions of packet losses gained from different measuring methods for the Codec H.264/AVC and the MPEG-2-TS, a burst size of "1" and an encoding rate of 2125 kbps

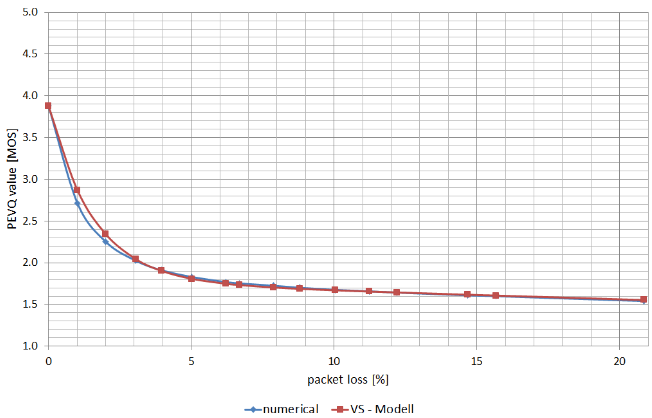

Fig. 10. QoS values as functions of packet losses gained from different measuring methods for then codec H.264/AVC and the MPEG-2-TS, a burst size of "3" and an encoding rate of 2125 kbps

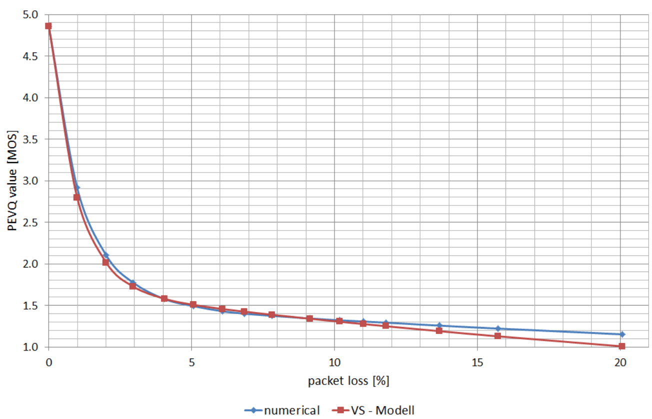

Fig. 11. QoS values as functions of packet losses gained from different measuring methods for the codec H.264/AVC and the MPEG-2-TS, a burst size of "1" and an encoding rate of 5175 kbps

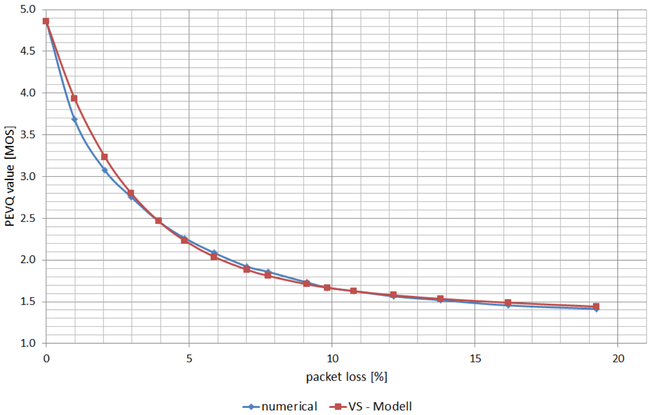

Fig. 12. QoS values as functions of packet losses gained from different measuring methods for the codec H.264/AVC and the MPEG-2-TS, a burst size of "3" and an encoding rate of 5175 kbps

Figures 9 to 12 also show that the curves of the PEVQ and IPTV Models progress very closely to each other. In other words: the numerical comparison study has delivered strong arguments for using IPTV Model in everyday practice.

4. Summary and outlook

In the course of the work described in this paper a new, parametrised QoS measuring model was developed to determine the quality of video streams in the IPTV service and its functionality was put to the test in a comparison study. When the new model was defined, all due consideration was paid to the characteristic parameters of the IPTV service. The new IPTV Model is based on the PEVQ curves. This is of especial practical importance since the PEVQ algorithm is considered to be the most objective QoE measuring method for video. The comparison study has proved the practicability of the new QoS model beyond a shadow of doubt. The new, inexpensive and quick and easy-to-use IPTV Model (off-line) is a more-than-adequate alternative to the laboriously slow PEVQ method (on-line; A measurement can often take several minutes.) with its expensive licences.

It would make sense to automate the production of further IPTV models defined according to Formula (1). One could save the PEVQ curves generated in the tool [14] in a format that is commonly used for the tool MATLAB [17] and then have the optimisation of the parameters needed for the IPTV Model executed automatically. Admittedly, that would require some additional programming, but such a modification is well worthwhile because there is a wide range of video codecs with different settings. In addition, these parametrised models are needed for new formats, such as HD 1080p, for instance. So such an approach would save considerable time when new IPTV models are created. Development work has already been started with this in view.

In order to be able to assess the QoS of the IPTV service in its entirety it is necessary, in practice, to consider the quality of the audio streams of the service as well. The ITU-T Recommendation G.1070 [18] and paper [19] contain suggestions as to how this might be achieved. This would inevitably begin with the creation of a parametrised model to evaluate IPTV audio streams (e.g. Codec MPEG-2 Layer3). Once such a QoS Model for audio existed, it would make sense to combine the two parametrised models and apply them to the IPTV service. As a result a joint QoS model (for both audio and video) would be born that evaluates the whole of the IPTV service. A new research work has already begun in this direction.

Another interesting endeavour would be to discover and document correlations between the ways in which the IPTV Model presented in this paper and the technique according to ETSI TR 101 290 [9] evaluated on the basis of network parameters for QoS of the IPTV service. In order to accomplish this, it would be necessary to load the datasets generated in the numeric tool [16] together with reduced test signals into a measurement system that implements the technique according to [9] and analyses them there. This could lead to the creation of a parametrised model for evaluating both the QoS and the QoE of the IPTV service. A most interesting endeavour indeed! Further work is envisaged in this direction too.

REFERENCES

[1] *Definition von NGN*, http://www.itu.int/rec/T-REC-Y.2001 (February 2015).

[2] *Directives 2009/140/WE for the Creation of Engineering Standards in Networks and Electronic Services*, http://eurlex.europa.eu/LexUriServ/LexUriServ.do?uri=OJ:L:2009:337:0037:0069:PL:PDF (February 2015).

[3] *Directives 2009/136/ WE for the Creation of Engineering Standards in Networks and Electronic Services*, http://eurlex.europa.eu/LexUriServ/LexUriServ.do?uri=OJ:L:2009:337:0011:0036:PL:PDF (February 2015).

[4] *ITU-T Recommendation J.144*, http://www.itu.int/rec/T-REC-J.144-200403-I/en (February 2015).

[5] *ITU-T Recommendation J.147*, http://www.itu.int/rec/T-REC-J.147-200207-I/en (February 2015).

[6] *ITU-T Recommendation J.247*, http://www.itu.int/rec/T-REC-J.247-200808-I (February 2015).

[7] *PEVQ algorithm and tool*, http://www.opticom.de/download/PEVQ-WP-v07-A4.pdf (February 2015).

[8] *The Company Opticom*, http://www.opticom.de (February 2015).

[9] *ETSI Recommendation TR 101 290*, http://www.etsi.org/deliver/etsi_tr/101200_101299/101290/01.02.01_60/tr_101290v010201p.pdf (February 2015).

[10] *ISO/IEC 13818-1, 2000 Generic Coding Of Moving Pictures and Associated Audio Information: Systems*, http://mumudvb.braice.net/mumudrupal/sites/default/files/iso13818-1.pdf (February 2015).

[11] IETF RFC 3640, 2003. *RTP Payload Format for Transport of MPEG-4 Elementary Streams*, http://www.faqs.org/rfcs/rfc3640.html (February 2015).

[12] S. Paulsen, T. Uhl and K. Nowicki, "MPEG-4/AVC versus MPEG-2 in IPTV", *Proc. Signal Processing and Multimedia Application Conf.* 1, 27–30 (2012).

[13] Ch. Hoppe, T. Uhl and K. Nowicki, "Determining QoS in the Video Telephone Service in an IP Environment", *Proc. Future Multimedia Networking Workshop* 3, 123–133 (2010).

[14] S. Paulsen and T. Uhl, "The new, parametrised VS model for determining the quality of video streams in the video-telephony service", *J. Telecom. Review* 8–9, 1155-1166 (2012).

[15] *Mean Opinion Score (MOS)*, http://www.itu.int/rec/T-REC-P.800/en (February 2015).

[16] T. Uhl and H. Jürgensen, "New tool for examining QoS in the IPTV service", *Proc. World Telecommunications Congress* 1, 1–3 (2014).

[17] *MATLAB Tool*, http://de.mathworks.com (February 2015).

[18] *ITU-T Recommendation G.1070*, http://www.itu.int/rec/T-REC-G.1070/en (February 2015).

[19] T. Uhl and S. Paulsen, "The new, parameterized VT model for determining quality in the video-telephone service", *Bull. Pol. Ac.: Tech.* 61 (3), 431–437 (2014).