Abstract—Real-time rate control techniques are necessary to applications such as video streaming in order to adapt video sending rate to network conditions so that the perceived video quality is optimised. In this paper two real-time rate control schemes, are presented. The first one uses the currently available bandwidth in order to perform rate control by selecting the optimum QP (Quantization Parameter). The second rate control scheme is based on measurements of the current packet loss in order to select the optimum QP parameter. Through extensive simulations on a test-bed environment both schemes are studied under diverse network conditions and are compared against the commonly used Rate-Distortion optimization scheme, which is embedded on a video encoder.

Keywords—real time video rate control, perceived video QoS optimization, PSNR

I. INTRODUCTION

Advances in video coding standards, along with rapid developments and improvements of network infrastructures, storage capacity, and computing power are enabling an increasing number of video applications. Real time video networking over IP based wired and wireless networks are constantly becoming more popular and more demanding. Unlike previous coding standards (i.e. MPEG-1/2/4), H.264/AVC (standard jointly by ITU-T and ISO [1]) offers advanced features such as variable block sizes, multi-reference frames, improved coding efficient and better quality. Video Rate control is an important element in order to adapt video to the varying network conditions across heterogeneous wireless and wire-line networks.

Rate control within the H.264 codec has been studied extensively in the literature [2], [3]. Rate control algorithms can be grouped in two major categories. The first category deals with the optimisation of Rate-Distortion at the encoder [4]-[6]. The second category deals with the determination of an optimum QP by taking into account rate-distortion and frame complexity [7], [8]. Many of these algorithms do not take into account the variation of network conditions.

The objective of this paper is to compare the performance two new real time video rate control schemes that achieve maximum perceived video quality based on the currently available bandwidth (BW) and the measured packet loss, respectively. Unlike the previous methods, this paper proposes a rate control method in real-time where QP is determined by taking into account the network conditions. Two algorithms are examined. In particularly, the first rate control scheme considers current available BW that is estimated based on RTCP feedback from the user terminal. An entity named Rate Control Module (RCM) resided in the video encoder is responsible for selecting a new quantization parameter (QP) that can be used by the encoder and result in a higher perceived video quality (PSNR), as compared to the current one. The second rate-control scheme takes into account the current packet loss that is measured at the user terminal and is sent as a feedback to RCM via RTCP messages, periodically. This scheme maximizes perceived video quality by selecting a new QP that reduces the sending bit rate and at the same time minimizes the current packet loss. The performance of both schemes is compared against the standard method of rate-distortion optimisation (The encoder decides by itself about the QP parameter by taking as an input the available bandwidth).

The rest of the paper is organized as follows. Section II includes an analytical description of the proposed adaptation schemes. In Section III the test-bed environment is presented and the simulation results are discussed. Finally, Section IV concludes the paper.

II. REAL TIME VIDEO ENCODER RATE CONTROL

The proposed real time video rate control framework requires that pre-encoded video sequences have been tested over different network condition. This is required in order to extract important statistical information. Such statistical information regards the relationship between encoding distortion and QP, sending bit rate and QP, as well as, the relationship between perceived video QoS and QP for different network loads and packet losses. The above relationships are obtained through experimentation.

A Rate Control Module (RCM) is defined as an entity within the video encoder that stores the aforementioned statistical information, collects real-time information from the...
network and sends feedback to the encoder in order to control the sending rate. Without loss of generality, we assume that this control is applied to each video frame. RCM receives periodic feedback with information regarding current available BW and packet losses, and decides upon the optimum QP value that maximizes the perceived video QoS (PSNR at the decoder). RCM’s decision is forwarded to the video encoder, which select the optimum QP parameter, as shown in Fig. 1.

**A. Available Bandwidth based rate control algorithm**

In this scheme, the perceived video QoS in terms of PSNR is maximized by selecting the optimum value of the QP parameter, according to the current available bandwidth (BW). Without loss of generality, the term available bandwidth, refers to the capacity of the access network that becomes available to the user. That is the bottleneck between the video encoder and the end-user. This network capacity is periodically monitored by probing the network with a predefined stream of dummy RTP packets. The user is informed for the available capacity (or available BW) by periodic RTCP Extended Report (RTCP-XR) messages that carry this information to the message header. In order to determine the optimum QP, the RCM is based on pre-stored PSNR versus QP data for different BW conditions. This relationship is illustrated in Fig. 2. In order to create this relationship a video sequence has been encoded at different QP values and is transmitted multiple times over a network with varying network load conditions. It is evident that for low available bandwidth, perceived PSNR increases with QP, due to the fact that larger QP results in lower video rate. The PSNR reaches a certain peak value which is the maximum perceived quality that a video user can have for specific network conditions. This is the point where both coding distortion and packet loss have the least impact on the perceived video quality for a given available bandwidth. Any further increase to the QP value will result in higher coding distortion and smaller video transmission rates (packet loss increases) that deteriorate the perceived video QoS. As the available BW becomes higher, PSNR reaches its peak value earlier (i.e. at smaller QP), shifting this point towards the left. Furthermore, Fig. 3 illustrates the surface fitted model of Fig. 2. This curve is true for one particular video sequence (NTIA gold fish pond [14]), however, the algorithm can be extended to other video sequences as well, without loss of generality.

![Fig. 1 Video rate control framework](image1)

![Fig. 2 PSNR vs. QP under different network conditions](image2)

![Fig. 3 Surface fitted model of perceived PSNR vs. QP under different network conditions](image3)

For the surface fitting model a number of different polynomials and rational equations was used, which resulted in the following polynomial equation with a reasonable fitting goodness.

$$PSNR = 605 - 1.2*BW - 57.6*QP +4*10^{-3}*BW^2 + 0.12*BW*QP + 1.26*QP^2 - 3.7*10^{-3}*BW*QP^2 + 0.02*QP^3 - 6*10^{-4}*QP^5$$  \( (1) \)

According to the BW based rate control algorithm, both the currently available BW and the PSNR versus QP relationship over different available BWs (Fig. 3) are regarded as input to the algorithm. BW conditions are collected from RCM via RTCP reports [10]. The algorithm optimizes the perceived PSNR by selecting the optimum QP value according to the network conditions. Depending on network conditions optimum QP can either be higher or lower than the current QP. As the network load increases (decreases), QP should increase (decrease) so that the sending rate is adapted according to the available bandwidth. The solution of the partial derivative of the three-dimensional function of Fig. 3 (Eq. 1), with respect to the QP given that the available BW information is collected by RTCP, returns the critical points (QP values) that maximize the perceived PSNR. Fig. 4 outlines the proposed BW based rate control algorithm.
B. Packet loss based rate control algorithm

This algorithm maximizes the perceived video QoS by selecting the optimum QP value for each video frame that in return will reduce the current packet loss measured at the user terminal. This means that as packet loss increases (decreases), RCM should select a higher (lower) QP, so that the combination of coding distortion and packet loss has the least impact at the decoder. This means that in case where packet loss increases, the algorithm should select a higher QP, so that both the sending rate and the packet loss decrease proportionally. In the context of this algorithm the RCM collects and stores measured RTP losses via the RTCP receiver report, periodically. In this study we only consider network packet losses, which is an indication of a congested network.

The perceived video quality (perceived PSNR) versus packet loss relationship is derived by encoding the same video sequence as previously, with different QPs and sending it over a network multiple times under different packet loss rate conditions. From Fig. 5 it is clear that the perceived by the user video QoS (PSNR) deteriorates as the network becomes progressively more congested. Moreover, Fig. 6 illustrates the surface fitted model of Fig. 5.

In a similar way as in the previous algorithm, the RCM determines the optimum QP values based on the relationship between PSNR and packet loss. An increase (decrease) to QP by ΔQP, will lead to a decrease (increase) to the generated bit rate and to a modification, in the same time, of the encoding distortion. Furthermore, we consider a maximum variation in terms of QP in order to avoid drift among subsequent frames. The default value of ΔQP has been set to 2, which is within the constraints defined in the H.264/AVC reference software JM [9]. Without loss of generality, a reduction to the generated bit rate will proportionally lead to a reduction to the packet loss, ΔPL. Therefore, it is foreseen that when this bit rate adaptation occurs, the new packet loss will be: PL_{new} = PL_{old} - ΔPL. Using Fig. 6, a new QP will be selected with higher value than the current one. In the case where the received RTCP reports indicate packet loss rate lower than PL_{new}, then the QP could be lowered in order to improve the quality of the sending video since there is more available bandwidth within the network. This is performed by the algorithm in a more conservative way, by lowering QP by 1. It has been found that by lowering QP by 1, there is an increase to the sending bit rate by 12.5% [8]. Fig. 7 includes the overview of the proposed Packet loss based rate control algorithm.
III. SIMULATION RESULTS

A. Experimental test-bed

In order to compare the performance of the proposed rate control scheme and study their impact on the perceived video quality, an experimental test-bed was implemented as in Fig. 8 Test-bed platform.

The experimental test-bed includes the VSOFT H.264/AVC video streamer [11], specifically parameterized in order to be capable of exchanging RTCP messages through a client server application with the terminal. During the experiments, the network is stressed with background traffic based on a statistical video traffic model, which regards a number of multiplexed homogeneous and mutually independent video sources that transmit simultaneously. This model can accurately simulate the effect of aggregate video traffic from multiple video sources [12]. Moreover, Dummynet [13] is implemented in order to emulate packet losses and network load.

In addition, the testing video sequence used during the experiments is YUV 4:2:0 color CIF (352x288 pixels) “NTIA gold fish pond” [14], [15] at 25fps. The length of the sequence is 1080 frames and it was encoded with a GOP size of 12 frames (GOP has the form of IPP...I) with QP values: QP=18, 20, 22, 24, 26, 28, 30. We have used a single NAL unit packetization (one RTP packet – one NAL unit) with RTP packet size of 1024 bytes (payload). The generated video packets are delivered through the network at the form of RTP/UDP/IP protocol stack.

B. Simulation results and discussion

The performance of the two proposed rate control schemes is measured in terms of perceived video quality (PSNR). In order to evaluate the produced results, we compare them against the Rate-Distortion (R-D) optimization scheme which, is an inherent process of the encoder and is based on the adaptation of the bit rate to the current network conditions feedback to the encoder periodically [11]. During this procedure the encoder decides the values of QP on its own, in order to transmit the video at a target bit rate, according to the current network conditions.

![Fig. 7 Proposed Packet loss based rate control scheme](image)

**Input**

- \( P_{lr} \rightarrow \) Application Layer (RTP) packet loss rate
- \( Q_{P_{current}} \rightarrow \) Current Quantization Parameter
- \( B(QP) \rightarrow \) Function describing the average bit rate
- \( F(QP, PL) \rightarrow \) Function describing the surface of the Packet loss based rate control

**Output**

- \( Q_{P_{new}} \rightarrow \) Best Quantization Parameter for current packet loss rate

\[
\begin{align*}
\text{if } prl &= 0 \& Q_{P_{current}} > Q_{P_{min}} \\
\Delta QP &= 1 \\
Q_{P_{new}} &= Q_{P_{current}} - \Delta QP \\
\text{else if } prl > \text{threshold} \& Q_{P_{current}} < Q_{P_{max}} \\
\Delta QP &= 2 \\
Q_{P_{new}} &= Q_{P_{current}} + \Delta QP \\
\text{else} \\
\text{ bitrateNow} &= B(Q_{P_{current}}) \\
\text{PSNR}_{current} &= F(prl, Q_{P_{current}}) \\
\text{Find } QP \in (Q_{P_{current}} \ldots Q_{P_{max}}) \text{ } \\
\Delta \text{Bitrate} &= \Delta QP \\
\text{ipacketLoss} &= \frac{plr * \text{MaxNetThroughput} - \text{bitrate}}{\text{MaxNetThroughput}} \\
\text{PSNR}_{new} &= F(ipacketLoss, xQP) \\
\text{if } PSNR_{new} > \text{PSNR}_{current} \\
Q_{P_{new}} &= Q_{P_{current}} + \Delta QP
\end{align*}
\]

Fig. 7 Proposed Packet loss based rate control scheme

![Fig. 8 Test-bed platform](image)

Fig. 8 Test-bed platform

![Fig. 9 Average PSNR achieved by each of the three schemes during the simulation test](image)
The simulation tests have been repeated 10 times for every rate control scheme, in order to average out unexpected behaviors of the network. Fig. 10 Comparison of PSNR per frame under the three studied rate control schemes illustrates the PSNR per frame resulted by each rate control scheme during one of the performed tests. The performance of the proposed schemes is shown clearly in Fig. 9, which compares the average PSNR of the two proposed schemes with the PSNR achieved by the R-D optimization in the encoder, during each simulation run.

Table 1 Minimum and maximum average PSNR values

<table>
<thead>
<tr>
<th>Rate Control</th>
<th>Minimum Average PSNR (dB)</th>
<th>Maximum Average PSNR (dB)</th>
</tr>
</thead>
<tbody>
<tr>
<td>BW based rate control</td>
<td>37.73</td>
<td>38.32</td>
</tr>
<tr>
<td>PL based rate control</td>
<td>32.64</td>
<td>37.21</td>
</tr>
<tr>
<td>Encoder R-D optimization</td>
<td>35.60</td>
<td>37.79</td>
</tr>
</tbody>
</table>

Table 2 Average Statistics (30 simulations per algorithm)

<table>
<thead>
<tr>
<th>Rate Control</th>
<th>BW based rate control</th>
<th>PL based rate control</th>
<th>Encoder R-D optimization</th>
</tr>
</thead>
<tbody>
<tr>
<td>PSNR (dB)</td>
<td>38.15</td>
<td>34.40</td>
<td>36.65</td>
</tr>
<tr>
<td>Packet loss</td>
<td>0.003</td>
<td>0.058</td>
<td>0.033</td>
</tr>
<tr>
<td>Throughput (kbps)</td>
<td>727.21</td>
<td>747.87</td>
<td>966.91</td>
</tr>
<tr>
<td>PSNR variance (dB)</td>
<td>4.44</td>
<td>18.40</td>
<td>11.25</td>
</tr>
</tbody>
</table>

It is evident that the BW based rate control scheme achieves the highest PSNR as it selects optimum QP parameters according to the currently available transmission BW that will minimize the packet loss. In comparison, the PL based rate control decides whether to increase or decrease the QP based on the current packet loss measured at the user terminal. This rate control scheme causes many variations to the perceived video quality, since it cannot capture very well the network variations, due to the fact that packet loss depends on the interval between consecutive RTCP packets (i.e. typical values are in the range of 5-8 ms). Apparently, the PSNR that is achieved by the encoder’s inherent R-D optimization scheme is lower than that achieved by BW based rate control but higher than that of the PL based algorithms. This is due to the fact that the encoder selects QP values such that adapt the transmission bit rate to match exactly the currently available bandwidth. The random behavior of the network due to the background utilization, combined with the bit rate adaptation, instead of the QP adaptation, reduces the performance of the encoder’s inherent R-D optimization.

Moreover, Table 1 summarizes the minimum and maximum average PSNR values that were collected during the simulations runs. In Table 2 the average values and the variance of the PSNR, packet loss and achieved throughput, are shown. The information in both tables further supports the above analysis. It can be seen that by applying BW based rate control we can achieve better PSNR, although the average throughput achieved is the smallest of the three schemes. This is due to the fact that the algorithm utilizes the PSNR versus QP relationship of Fig. 3, which indicates QP values that will maximize PSNR based on the available BW.

IV. CONCLUSIONS

This paper compares real-time video rate control algorithms for video transmission across IP-based networks that are based on network feedback. The first algorithm collects statistics in order to model the perceived video quality (PSNR) versus available network bandwidth and QP. The network bandwidth is monitored periodically through RTCP messages. The algorithm tries to optimise PSNR=f(QP, BW) by solving the first derivative with respect to QP. The second algorithm collects statistics in order to model the perceived video quality (PSNR) versus packet loss and QP. Packet loss is monitored through RTCP messages. The algorithm estimates the maximum possible reduction of packet loss by modifying the QP parameter, accordingly. It then tries to optimise PSNR=f(QP, PL) by solving the first derivative with respect to QP and the new estimated packet loss. Through, test-bed simulation results it has been found that the first algorithm

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**Fig. 10 Comparison of PSNR per frame under the three studied rate control schemes**
gives better average PSNR and leads to less variations of the perceived video quality.

REFERENCES


