Rate adaptation for wireless video streaming based on error statistics

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Abstract: This paper presents a new rate-control algorithm for live video streaming over wireless IP networks, which is based on selective frame discarding. In the proposed mechanism excess '*P*' frames are dropped from the output queue at the sender using a congestion estimate based on packet loss statistics obtained from RTCP feedback and from the Data Link (DL) layer. The performance of the algorithm is evaluated through computer simulation. This paper also presents a characterisation of packet losses owing to transmission errors and congestion, which can help in choosing appropriate strategies to maximise the video quality experienced by the end user.

Keywords: video streaming; rate-adaptation; IEEE 802.11; QoS.

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1 Introduction

The popularity of internet video streaming has grown tremendously in the past few years. Live video distribution is the most demanding application from the point of view of network requirements.

Since the video stream cannot be buffered prior to visualisation, delay jitter becomes a very important QoS parameter. Besides, internet communications are usually subject to the time-variability of available bandwidth, which depends on network load. The TCP protocol includes a flow control scheme that allows it to counter network congestion by decreasing the transmission window upon detection of missing acknowledgements. Nevertheless the ARQ-based reliability of TCP is not suitable to meet the delay and jitter requirements of live video, which is the reason why UDP is commonly used for this application. When TCP is transmitted concurrently with UDP, TCP is the only one to respond to congestion, which is unfair and does not mitigate congestion when its cause resides in the UDP traffic. The latter can be solved with rate-adaptation of UDP traffic, which is often based on the RTP/RTCP protocol pair (Schulzrinne et al., 1996). An example of such a rate-adaptation mechanism is proposed in Busse et al. (1996). In order to provide fairness between UDP and TCP traffic, TCP-friendly rate-adaptation of multimedia flows at the transport layer has become an important IETF requirement, leading to the specification of the Datagram Congestion Control Protocol (DCCP) (Kohler et al., 2003) as a replacement to UDP.

However, even if rate-adaptation works at the transport layer, there must be a way to adapt the video stream at the application layer accordingly in order to maximise the quality experienced by the receiver. Assuming that the available bandwidth is known, there are two main schemes to adapt the output rate of the video stream. Multirate stream switching is a widely used technique that allows adaptation to large-scale bandwidth fluctuations and consists of assigning the receiver to the stream whose rate best meets its bandwidth constraints. Nevertheless this cannot be made very often with most of the current codecs, whose switch rates are usually very poor, causing glitches in the received stream when the rate change is too frequent. In order to meet frequent bandwidth fluctuations, it is faster and more seamless to perform selective discarding of video frames, fine-tuning the stream rate to the transitory congestion rates experienced by the receiver while maximising the perceptual quality of the video.

While the effectiveness of rate-adaptation techniques is already demonstrated in wireline networks, wireless network present additional problems related to the fluctuations in terms of the physical packet error rate, specially when the Received Signal Strength (RSS) drops heavily due to propagation phenomena and interference. In wireline networks, the available bandwidth estimates are usually based on the Packet Loss Ratio (PLR), which can be completely attributed to congestion. On the other hand, in wireless networks the packet losses due to physical transmission errors are significant. If the latter are wrongly interpreted as a result of congestion, the PLR remains constant despite the decrease of the output bitrate enforced by the rate-adaptation algorithm. Transmission losses can only be effectively dealt with Forward Error Correction (FEQ) and Automatic Repeat Request (ARQ) techniques and thus must be distinguished from congestion losses.

This paper presents a different RTP/RTCP based rate-control scheme for MPEG-4 (ISO/IEC MPEG-4 std., 2001) based on selective frame discarding, where dependent frames (typically ‘*P*’ frames) are dropped first, thus maximising the quality of the video for the same PLR. The proposed framework was developed as an enhancement method to be applied in a major architecture for distribution and management of live content at the application layer, which was developed in the IST 2000-30046 (2004) OLYMPIC Project. The resulting architecture was designed in order to be able of distributing thousands of live streams over the internet to heterogeneous clients placed in several different access and local networks, dealing with multicast and QoS at the application layer, forming what is known as an overlay architecture.

This paper is organised as follows. Section 2 presents the most relevant related work. Section 3 presents the main differences between transmission packet losses and congestion packet losses. Section 4 presents the proposed rate-adaptation scheme, together with its performance evaluation. Finally, Section 5 concludes this paper.

2 Related work

At the transport and application layer, congestion management can be based on rate-adaptation of an encoded video stream according to a channel reported loss. In this solution the video source sends a stream rate close to the channel’s bandwidth. Examples of these strategies can be found in Busse et al. (1996), Wu et al. (2000) and Krasic et al. (2003).

Selective frame discard techniques can also be used to adapt to bandwidth constraints while maximising the stream quality perceived by the user. In Zhang et al. (2001), a number of optimal and heuristic selective frame discard algorithms are proposed. These algorithms try to find the minimum number of frames that must be discarded in order to comply with network bandwidth limitations, taking into account client buffer constraints. These algorithms can also take into account video scalability options, assigning different discard priorities to different types of frames. Scalability options are also taken into account in several proposals of congestion management at the network layer. These usually involve selective discard combining the scalability options available in video coding standards such as MPEG-4 with IP QoS architectures such as DiffServ. Examples of these strategies can be found in Ahmed et al. (2001a,b). None of these proposals evaluate the impact of selective frame discard of the end-to-end quality perceived by the users.

One of the advantages of functioning end-to-end at the application layer is that more complex and specific congestion control algorithms can be built capable of dealing with highly dependant video structures. Another advantage of this solution is that it works both with QoS capable networks and best effort ones. The selective frame discard algorithm proposed in this paper works at the application layer and takes into account the scalability options available in video coding.

Related work on video rate-adaptation in wireless networks has been mainly focused on the classification of packet losses as due to congestion or transmission errors (e.g. Pyun et al., 2003) and the selection of FEC coding and ARQ in order to optimise transmission over the air interface (e.g. Grilo and Nunes, 2003; Majumdar et al., 2002; Xu et al., 1999).

In Pyun et al. (2003) an RTP/RTCP based TCP-friendly rate-control algorithm is presented, which used information from a middleware wireless adaptation layer to distinguish packet losses due to transmission errors from packet losses due to congestion. The proposed scheme is then evaluated through computer simulation. Although the authors do not provide details concerning stream adaptation, the frame dropping seems to consist simply of dropping the queued frames whose maximum delays were exceeded. This kind of random frame dropping may lead to a decrease in video quality due to the interdependency between frames.

A QoS-Directed Error Control (QDEC) scheme is proposed in Xu et al. (1999) to increase the reliability of video multicast in wireless networks. QDEC differentiates between essential frames (e.g. MPEG '*I*' frames) and non-essential frames (e.g. MPEG '*P*' or '*B*' frames) and only applies the error control algorithms to the former. Error control is mostly based on FEC, which may be complemented with ARQ when FEC is not enough to completely cover the error bursts.

Another hybrid ARQ-FEC mechanism is proposed by Majumdar et al. (2002) to ensure graceful quality degradation in multiresolution scalable video streams. The FEC mechanism is based on the Multiple-Description-FEC (MD-FEC) algorithm, which partitions the video stream into a series of resolution layers, applying varying amounts of protection to each layer, depending on its importance. The MD-FEC is used alone without ARQ in multicast. A feedback protocol allows the receivers to notify the sender about their transmission profiles, so that the chosen MD-FEC encoding maximises the overall quality.

In Grilo and Nunes (2003) a layer-2 rate-adaptation algorithm is proposed that supports both unicast and multicast transmission in WLANs. In multicast, video frames are assigned a drop priority, which is lowest for the essential '*I*' or base layer frames. The latter are granted reception by all users while non-essential frames are assigned higher drop priority and are only received by a subset of users. In this way the algorithm manages the trade-off between WLAN utilisation and reception quality of video streams.

The selective frame discard algorithm proposed in this paper assumes the use of statistics on the wireless transmission losses obtained from the underlying Data Link (DL) layer. The latter allow the computation of effective congestion losses, based on which the selective frame discard procedures operate.

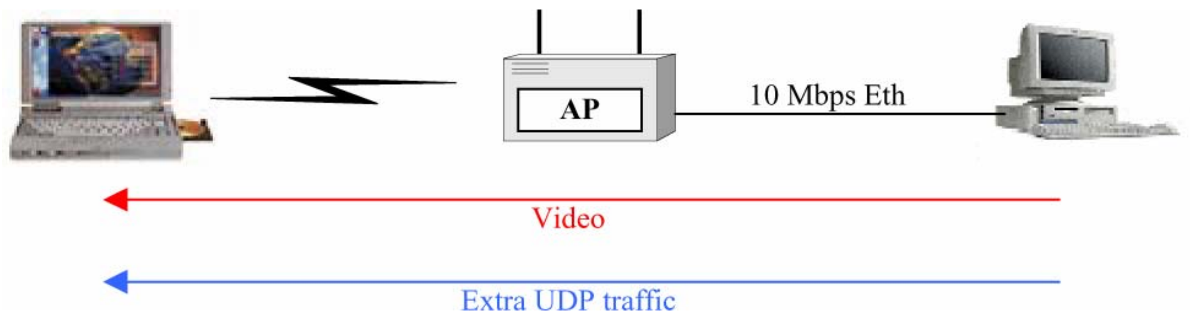
3 Characterisation of packet losses

The statistical characteristics of transmission packet losses are quite different from those of congestion packet losses. The following characterisation was obtained from experimental measurements. The used experimental setting is depicted in Figure 1.

A laptop running the LINUX operating system was connected to an IEEE 802.11b (IEEE Std. 802.11b, 1999) Access Point (AP), which in turn was connected to a desktop computer through a 10 Mbps Ethernet interface.

The laptop ran the video client application, receiving a video stream of 256 kbps and (average packet size of 395 octets) from the video server running at the desktop computer. When required, the desktop computer generated extra UDP traffic (constant bit-rate and packet size of 1500 octets) with the Iperf tool in order to test congestion conditions. The wireless network interface card used in the experiments was the ORiNOCO PC Card from Lucent Technologies, while the AP was the I-Gate 11M I/LAN from Siemens. RSS values were obtained from the laptop interface card drivers. Packet losses were obtained from traces created by Ethereal.

Figure 1 Experimental setting (for colours see online version)



In order to evaluate the effects of RSS on packet loss, the laptop was placed at different distances from the AP, receiving the video stream for 300 sec. RSS samples were taken with a period of 1 sec. Figure 2 depicts the PLR as a function of the average RSS. As can be seen, at RSS = −90.5 dBm the PLR is approximately 7.5%, decreasing to 0 at approximately RSS = −80.6 dBm. One must not forget that the IEEE 802.11 MAC protocol includes a stop-and-wait ARQ mechanism that is the reason the packet losses due to transmission errors are not higher.

Figure 2 PLR as a function of the RSS (for colors see online version)

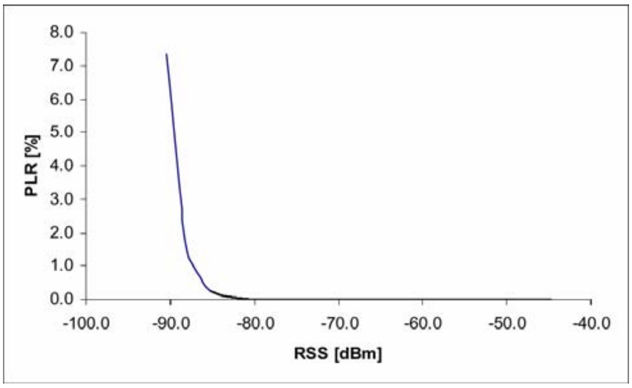
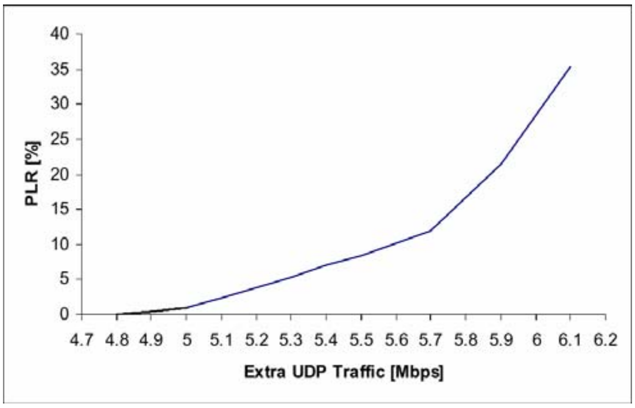


Figure 3 depicts the average PLR as a function of extra UDP traffic load when the RSS is good (RSS ≈ −40.0 dBm).

Figure 3 PLR as a function of extra UDP traffic load (for colors see online version)



The main statistical differences between transmission losses and congestion losses only become obvious when looking at the error distribution in time, considering a similar average PLR. Figure 4 depicts the PLR in time windows of 1 sec, when there is no congestion, with all the losses being owing to transmission errors (average PLR of 1.6% at an average RSS of −88.1 dBm). It can be seen that the PLR is quite variable, sometimes reaching values in the order of 45%, while in other time windows it falls to zero.

Figure 5 shows a similar plot for the situation where the average RSS is −40.0 dBm and all the losses (average PLR of 2.3%) are due to congestion (4.05 Mbps of extra UDP traffic). Although the values correspond to a higher average PLR, the maximum PLR is now around 9%. This lower variability of congestion losses is confirmed by the PLR standard variation, which is 5.9% in Figure 4 and 2.2% in Figure 5.

Figure 4 PLR without congestion for an average RSS of −88.1 dBm (for colors see online vesion)

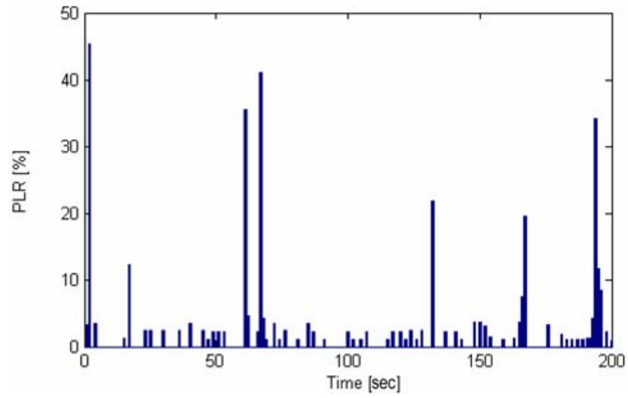
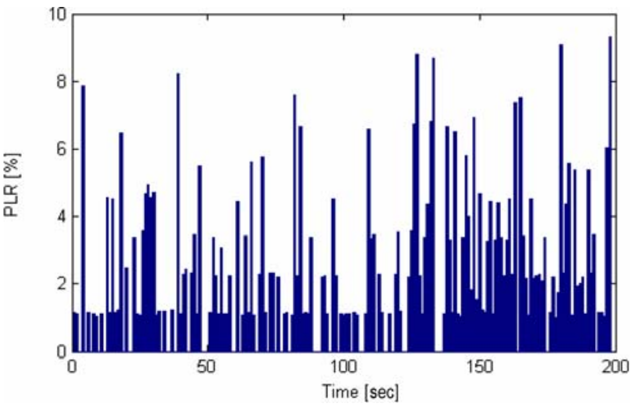


Figure 5 PLR due to congestion for an average RSS of −40.0 dBm (for colors see online version)



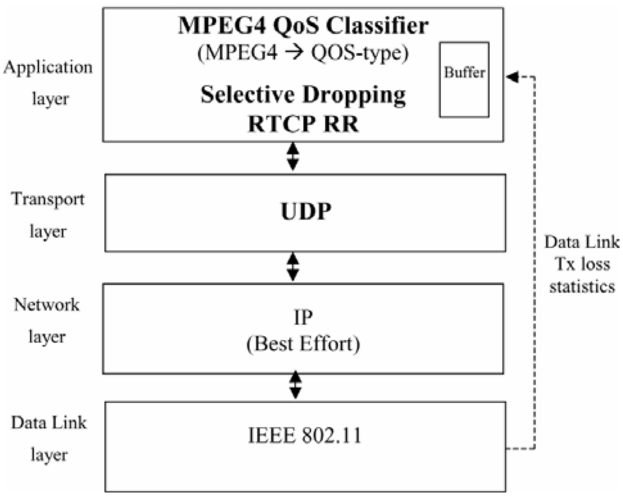
The lower variability of congestion PLR suggests that rate-adaptation techniques based on RTCP feedback are suitable to deal with congestion, forcing the decrease of excess traffic and hence packet losses. On the other hand, the high variability of the PLR due to transmission errors raises the issue of whether adaptive FEC mechanisms are really efficient for video streaming over wireless when compared with selective-repeat ARQ at the application layer. Due to the limited scope of this paper, this issue was left for future work.

4 Rate-adaptation algorithm

The rate-adaptation algorithm proposed in this paper aims for a quick reaction to congestion carried out at the source of the video stream. The solution is based on selective discarding performed at the application layer, using the RTCP RR reports as a feedback mechanism to report the overall packet losses.

The protocol stack that corresponds to this solution is depicted in Figure 6, where IEEE 802.11 constitutes the underlying wireless technology. It should be noted that the proposed scheme is generic and can be applied to other wireless technologies.

Figure 6 Protocol stack of the proposed scheme



4.1 MPEG-4 temporal scalability

In order to explore the MPEG-4 temporal scalability, it is important to understand its structure. An MPEG-4 elementary stream usually contains one or more Groups of Video Object Planes (GOVs), which, in turn, are composed of several '*I*', '*P*' or '*B*' Video Object Plane (VOP) sequences.

Although the MPEG-4 standard considers these three types of VOPs, currently almost all live encoders use only '*I*' and '*P*' VOPs, with large sequences of *P* VOPs between *I* VOPs, as shown in Figure 7.

This means that in order to properly decode a '*P*' VOP, the decoder needs to completely receive the previous '*I*' VOP and all the previous '*P*' VOPs of a GOV.

When delivering this highly dependant structure through IP networks, congestion losses can seriously degrade the received quality and, therefore, a solution must be found that selectively discards frames, taking into account this VOP sequence.

This means that when congestion occurs, frame discarding should start discarding '*P*' VOPs that have a higher index, that is, the last '*P*' VOPs of a GOV (P_i to P_n as represented in Figure 8).

This type of discard priorities can hardly be implemented using current QoS network level strategies, for example, DiffServ, simply because these models have limited discard levels (see Ahmed et al. (2001a) e.g. of a DiffServ-based frame discarding algorithm). For example, in the case of DiffServ Assured Forward AF PHB only three discard priorities are defined.

Nevertheless, since we are dealing with an overlay architecture, a discard algorithm can be implemented in the video server at the application layer. This means that, using a congestion estimate, the video server can perform a selective frame discarding, as described in the next section.

The application layer discard algorithm should adjust the ratio of discarded data to the congestion level previously computed. In order to do it, the discard index *i* (see Figure 8) from which the discard of VOPs starts, must be computed and adjusted dynamically according to the number of '*P*' VOPs of a GOV and comparing the real and target ratios of discarded data.

4.2 Selective frame discarding algorithm

The selective discarding algorithm dynamically counts all the data that is being transmitted, discarding all '*P*' frames that come after a certain discard index. The discard index is adjusted according to the congestion level computed through the RTCP RR reports sent from the client.

The selective discarding algorithm must distinguish between transmission and congestion losses, otherwise it

Figure 7 Typical GOV structure of a live encoded stream

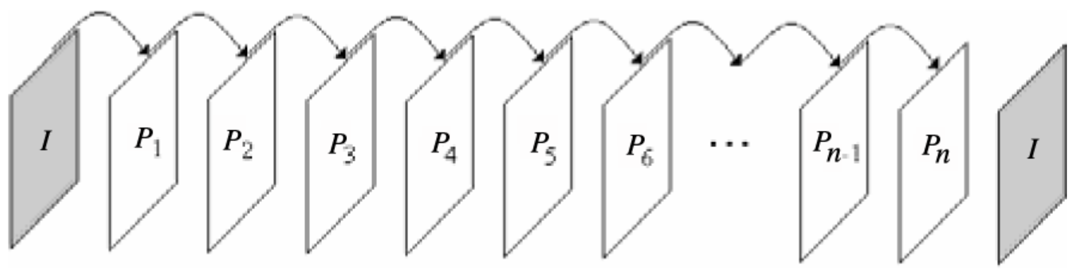
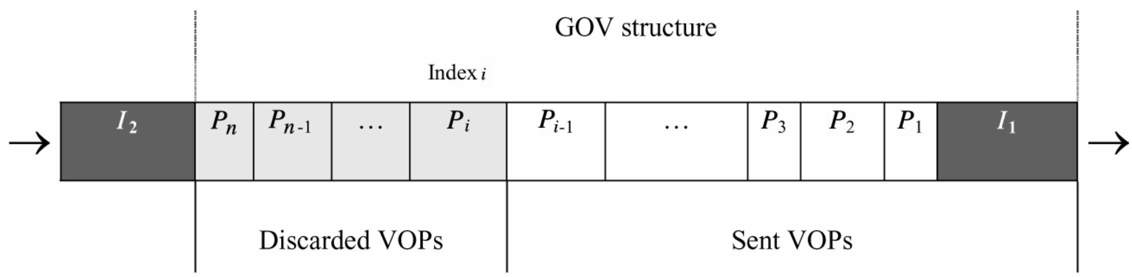


Figure 8 Sender queue showing the VOPs to be Discarded and Sent in a GOV



steadily decreases the output rate of the stream to no avail because frame discarding cannot eliminate transmission losses. Statistics on the wireless transmission losses are obtained directly from the underlying DL layer. For example, in the case of IEEE 802.11, loss statistics can be computed locally from the aTransmittedFragmentCount and aFailedCount variables of the IEEE 802.11 Management Information Base (MIB), which indicate the number of successful and failed frame transmissions, respectively.

The PLR due to transmission errors (PLR_{Tx}) can be calculated as follows:

$$PLR_{Tx} = \frac{aFailedCount}{aFailedCount + aTransmittedFragmentCount} \quad (1)$$

The difference between the overall PLR indicated in the RTCP RRs (PLR_{RTCP}) and the PLR_{Tx} provides the fraction of losses due to congestion (PLR_{Cg}), which is used to adjust the level of packet discard at the server:

$$\mathrm{PLR}_{\mathrm{Cg}} = \mathrm{PLR}_{\mathrm{RTCP}} - \mathrm{PLR}_{\mathrm{Tx}} \quad (2)$$

Since the number of '*P*' VOPs between '*I*' VOPs is not fixed, the selective discarding algorithm uses the number of '*P*' VOPs of the previous GOV to help it decide the discard index of next GOV. Using this discard index the selective discarding algorithm discards all frames that come after this index according to the PLR_{Cg} estimate.

Since the client's bandwidth changes with time, the discard ratio should follow it. Consequently, the algorithm should constantly monitor the PLR_{Cg}, comparing it to the discard ratio. If PLR_{Cg} is higher than the current discard ratio, it means that there are more losses due to congestion and, therefore, the discard ratio should be increased. When this is not the case, it may indicate that the discard ratio is greater than PLR_{Cg} and hence the algorithm should try to reduce the discard rate, increasing the discard index.

Figure 9 and 10 shows respectively the simplified pseudo-code and diagram of the selective frame discarding algorithm.

Figure 9 Simplified selective frame discarding algorithm

```
Compute PLR_Cg from Receiver Reports and DL layer statistics
If PLR_Cg > Discard Ratio
        Decrease Discard Index
   else
        Increase Discard Index
end
Do
   if (VOPIndex(frame) < Discard Index)
        Send frame
        Increase Transmitted Data with frame size
    else
        Discard frame
        Increase Discarded Data with frame size
Until next I frame
```

4.3 Performance evaluation

The following results aim to compare the proposed selective discarding algorithm with non-selective (experimental) discard. The results were obtained using a simulation framework developed in MATLAB for MPEG-4 live sport video sequences. The main target of these simulations was to quantify and compare the effect of the lost packets in the decoding process, which was computed through the Peak Signal-to-Noise Ratio (PSNR):

$$\mathrm{PSNR} = 20\log_{10}\left(\frac{255}{\sqrt{\mathrm{MSE}}}\right) \quad (3)$$

where MSE is calculated as follows:

$$\mathrm{MSE} = \frac{\sum[f(i,j) - F(i,j)]^2}{N^2} \quad (4)$$

where $f(i, j)$ is the source image, $F(i, j)$ is the reconstructed image and N^2 is the number of pixels of the source image

Figure 10 Selective frame discarding diagram

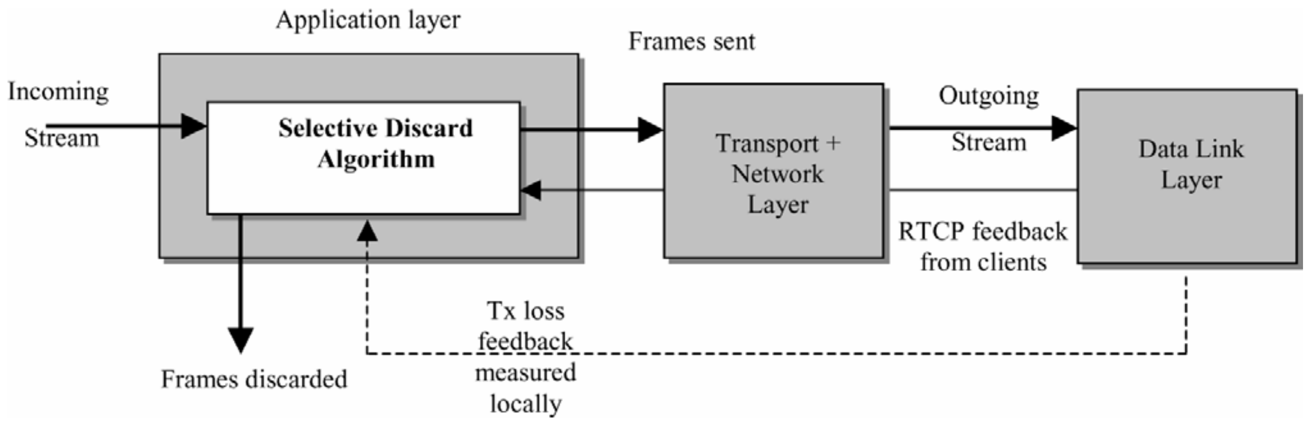


Figure 11 depicts the average PSNR as a function of the overall PLR (PLR_{RTCP}) for a scenario with no transmission losses. The advantages of the proposed selective frame discarding scheme are obvious, as the PSNR is kept between 25 and 22.5 dB for $PLR_{RTCP} < 23.9\%$. In contrast, when the discard is non-selective, the PSNR decreases below 20 dB (bad quality threshold) for $PLR_{RTCP} > 7.5\%$.

Figure 11 Average PSNR as a function of overall PLR, considering no transmission packet losses

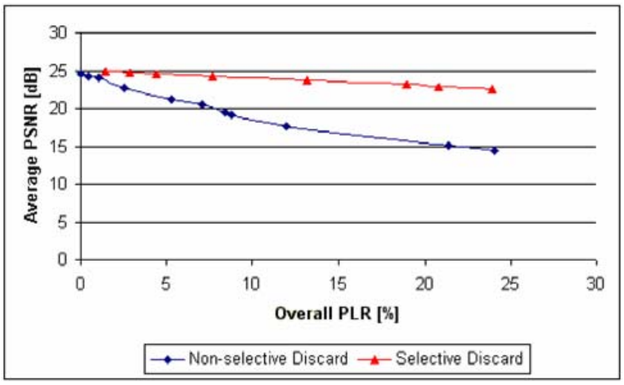
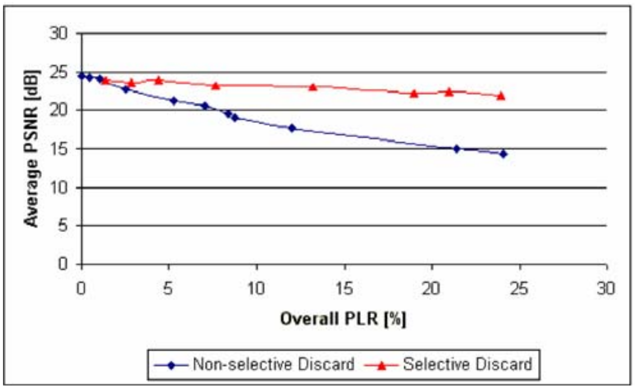


Figure 12 depicts the average PSNR as a function of the overall PLR but now the PLR_{Tx} is fixed at 1%. It is assumed that the selective discarding algorithm distinguishes between transmission losses and congestion losses.

Figure 12 Average PSNR as a function of overall PLR, considering 1% of transmission packet losses



The PSNR of the selective discarding curve decreases approximately 1 dB, remaining much better than non-selective discarding. Its PSNR is kept between 24 and 21 dB for $PLR_{RTCP} < 23.9\%$. The corresponding images for $PLR_{RTCP} = 19\%$ are depicted in Figure 13, where the differences in perceptual quality are clearly noticed.

In order to evaluate the responsiveness of the selective discard algorithm, the PLR_{Cg} level was made to vary in time (see the chart at the top of Figure 14). The bottom chart in Figure 13 depicts the corresponding evolution of the discard index, while the chart in the middle of the figure depicts the selective discard ratio. Although the default period of RTCP reports is 5 sec, it was set to 1 sec

Figure 13 Video sequence of sport event: non-selective dropping of 19% (left image) versus selective dropping of 18% plus 1% of transmission packet losses (right image)

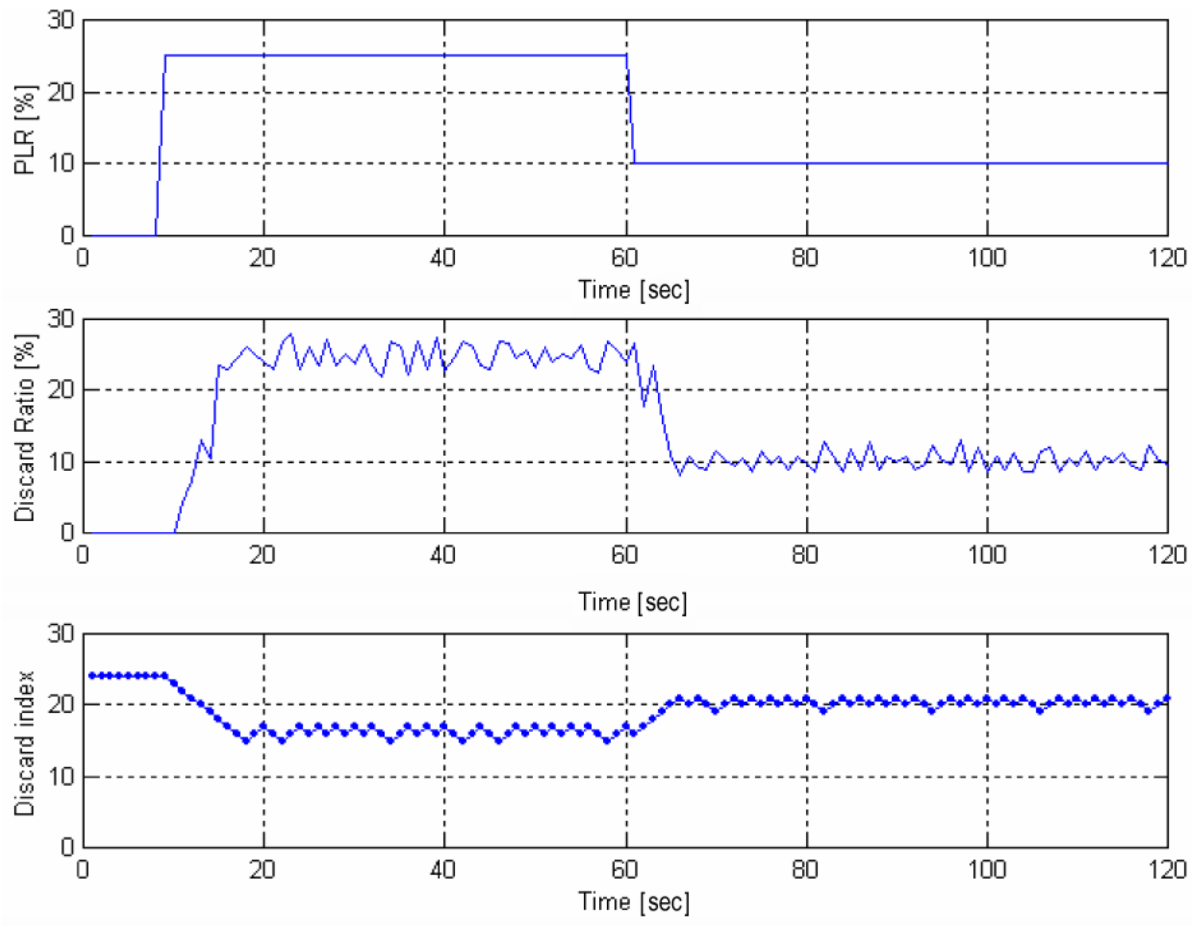


Figure 14 Transmitted P-VOPs versus PLR_{Cg} and discard ratio (for colours see online version)

in order to increase the responsiveness of the algorithm in the presence of the high variability of wireless network conditions.

As previously explained, the selective discard algorithm adjusts the selective discard rate to the PLR_{Cg} by either increasing or decreasing the discard index. In the initial phase, since the PLR_{Cg} is zero, all P-VOPs are transmitted and the discard ratio is also zero. When the reported PLR_{Cg} changed from 0 to 25%, the algorithm decreases the discard index until the discard ratio is close to the reported PLR_{Cg}. Finally, when the PLR_{Cg} change from 25 to 10%, the algorithm increases the discard index while the reported losses are approximately equal to the discard ratio.

5 Conclusion

This paper has presented a new rate-adaptation algorithm for MPEG-4 real-time video streaming, which aims to adapt the output rate to the fluctuations of the available data rate of the different links. The algorithm is based on the principle that the congestion losses match the excess traffic. The decrease of the output rate is achieved by dropping '*P*' VOPs in decreasing order of index within each GOV in order to avoid quality degradation due to inter-VOP dependency. The number of VOPs that are eliminated in each GOV depends on an index whose value depends on the comparison between the congestion PLR and the VOP discard ratio. In order to allow operation in error-prone wireless networks, the congestion PLR is obtained taking into account not only the RTCP feedback but also the the error statistics provided by the DL layer. The performance of the algorithm was evaluated through computer simulation and it was shown that it results in higher PSNR values than pure transmission packet losses.

A characterisation of congestion and transmission packet losses was also presented on the basis of experimental results. The high variability of the PLR due to transmission errors in wireless networks raised the issue of whether adaptive FEC mechanisms are less efficient for video streaming over wireless when compared with selective-repeat ARQ (or a combination of the two) at the application layer.

The results presented in this paper were based on the use of RTP/RTCP over UDP. In spite of the improvements of the perceptual quality obtained, there are still some problems related to the use of these protocols that need further study. In fact, one of the problems of this method is that RTCP Receiver and Sender reports (RR and SR) are relatively infrequent (1 sec period) and, therefore, rate adjustments may lack from fast feedback information, slowing the reaction to congestion. Additionally RTCP Receiver Reports and Sender Reports are also lost when congestion occurs. In future work a different solution, which uses DCCP as the transport layer, will be evaluated. DCCP offers several mechanisms of feedback that can be used to replace RTCP. Those mechanisms include the Ack Vector or TCP-Friendly Rate Control (TFRC) Congestion Control (CCID 3) Acknowledgements, which are sent once every Round Trip Time (RTT) interval. Using the Ack Vector a receiver can identify which packets were lost and which ones were marked with Explicit Congestion Notification (ECN), when available. In order to perform TFRC Congestion Control, the source uses a throughput equation to adjust its sending rate based on loss event rate and RTT obtained from Acknowledgements sent by clients. This ensures that the congestion control mechanisms react with little delay. The difference between the encoded stream rate and the computed throughput rate of a client gives the instant discard level of the source.

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