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Joint Adoption of QoS Schemes for MPEG Streams

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Abstract. Indiscriminated packet discards strongly degrade the quality perceived by end users of MPEG video transmissions. This paper investigates different Quality of Service (QoS) schemes and the tradeoffs of jointly adopting such schemes to improve the delivery quality of an MPEG stream. From an analytical model, we evaluate the impact of frame losses on the quality of MPEG streams and on the waste of network resources. Our assessment considers issues such as the use of redundancy by applying a Forward Error Correction (FEC) scheme to tolerate losses, the changing of the compression factor in MPEG encoding, the unequal protection of MPEG frames in a Differentiated Services environment, and how to evaluate the impact of network losses onto application quality. Results provide predicted bounds on the quality to be expected by end users as well as guidelines on how to take the best advantage from the joint adoption of the investigated QoS schemes.

Keywords: Quality of Service, video communications, MPEG, differentiated services, forward error correction

1. Introduction

The best-effort model of the traditional Internet has become inadequate to deal with the very diverse requirements on network Quality of Service (QoS) of an ever-increasing range of traffic types [34]. A key point for the success of the new multimedia applications is the network QoS provided to video and audio streams [25, 33, 35, 36]. To address the issue of providing a QoS support for the transmission of flows from applications with different QoS requirements, different approaches are currently being developed, such as adaptive applications [11, 12, 18], QoS-based routing [7, 8], and the Differentiated Services (DiffServ) architecture [4]. In the case of a video transmission over the Internet, both the encoding and the transmission processes affect the QoS [31].

The wide adoption of the MPEG encoding turns it into an attractive way for the distribution of audio and video over the Internet. The MPEG encoding is pointed out as a standard for future networked interactive video applications [22]. Nevertheless, the hierarchical structure of MPEG encoding with a possible error propagation through its frames imposes a great difficulty on sending MPEG video streams over lossy networks. Small packet loss rates may translate into much higher frame error rates. For example, a 3% packet loss percentage could translate into a 30% frame error probability [5]. This situation may seriously degrade the perceived quality by a user at the video reproduction. Moreover, network resources may be wasted with the transmission of information that becomes useless to the receiver. Some portion of the received data may become useless to the decoder as insufficient frame data are available for decoding the frame because of losses in the network or because a frame, in which the current frame depends on, is undecodable. Thus, the transmission of video streams with a certain level of QoS is a challenging problem [23], possibly requiring the combined adoption of multiple strategies [38].

The transmission of an MPEG video stream adopting a combination of different QoS strategies over a lossy network like the Internet or an intranet is of significant importance because of the large installed base of the IP networks. Quality video delivery over such widely deployed IP-based networks is required by several applications such as distance learning and collaboration, video distribution, video conferencing, and interactive virtual environments.

This paper evaluates different possibilities to improve the delivery quality of an MPEG video stream crossing a lossy network like the Internet or an intranet. We evaluate the joint adoption of such QoS strategies as well. Results cover issues as how to:

- apply redundancy, either by using Forward Error Correction (FEC) to tolerate some losses or by using a different compression factor in the MPEG encoding. FEC schemes protect video streams against packet losses up to a certain level at the expense of data redundancy. Adopting different frame patterns allows a MPEG stream to better adapt itself to the available transmission conditions;
- adequately protect the different types of MPEG frames adopting a DiffServ architecture [4]. Unequal protection based on frame type avoids quality degradation due to the loss of one particularly important frame and the possible propagation effect of such loss throughout the hierarchical structure of MPEG video streams;
- analytically do the mapping of network QoS (in terms of packet loss probabilities) onto application QoS (in terms of decodable MPEG frames).

Furthermore, we evaluate how to best combine the investigated techniques to improve the delivery quality of the MPEG streams.

The remainder of the paper is organized as follows. Section 2 presents an overview of the related work. Section 3 briefly reviews the MPEG structure and defines our evaluation metrics. Section 4 introduces our analytical model to measure the impact of losses in the video stream considering the defined evaluation metrics. In Section 5, we discuss the obtained results for the different QoS schemes and their joint adoption. Section 6 presents our concluding remarks.

2. Related work

In the MPEG video literature, common strategies for QoS provisioning include the adoption of different GOP structures, different frame dropping priority mechanisms, or FEC schemes.

Changing the GOP pattern is a strategy to better adapt the MPEG stream to the available transmission conditions. Adjusting the adopted GOP structure to the image contents taking into account scene changes is studied in [14]. Detection of scene changes is used to optimize the GOP pattern for the actual video sequence. The impact of using different GOP structures on video quality is investigated in [9]. The work focuses on the tradeoff between storage benefits due to more compression and picture quality improvements. Turaga et al. [30] propose models for variable bit rate (VBR) video traffic that allow for different frame types and variable GOP patterns.

Indiscriminated packet discards may seriously degrade an MPEG video stream because of its hierarchical structure. Due to this feature of MPEG systems, different frame dropping priority schemes are proposed to protect different types of frames against such indiscriminated discards. In [10], a method to provide deterministic service guarantees to MPEG video streams is proposed. Alam et al. [1] study the effects of a traffic shaper at the video source on the transmission characteristics of MPEG video streams while providing delay and bandwidth guarantees. Hemy et al. [21] propose the use of filters in some nodes within the network to selectively drop frames of an MPEG stream, which must be partially decoded at each filter node.

A content-based packet video forwarding mechanism based on a relative priority score that is mapped onto a proportional loss-rate differentiation mechanism is proposed in [27]. Unequal protection of different types of frames using mechanisms based on the DiffServ architecture [4] are also analyzed [3, 37]. Shin et al. [26, 28] propose a framework to map categorized video packets (in terms of loss and delay) onto relative DiffServ levels, while taking into account a given pricing model.

Video delivery can significantly be improved by the adoption of FEC mechanisms. Basically, redundancy is added to the data so a receiver may recover from losses or errors in the transmission without further intervention from the sender. A media-independent FEC scheme is the focus of recent efforts [17]. Another alternative is the development of FEC mechanisms that take into account the characteristics of the data being protected. The PET scheme [2] has been developed based on the particular features of MPEG video streams and allows for FEC including the support of priorities.

Although QoS schemes for MPEG video streams have been widely investigated, commonly such QoS strategies are only studied individually. This paper analyzes the joint adoption of different QoS approaches to enhance the quality of MPEG streams. We investigate the mutual influence such approaches exert on each other when jointly adopted. Moreover, we evaluate how to best combine the analyzed techniques to improve the delivery quality of an MPEG video stream.

3. Evaluation metrics for MPEG

MPEG encodes video as a sequence of three types of compressed frames (I, P, or B) [24]. The whole video sequence is decomposed into smaller units which are coded together and called GOPs (Group of Pictures). The GOP pattern specifies the number and temporal order of P and B frames between two successive I frames. Such a GOP pattern is characterized by two parameters: the I-to-I frame distance (N), and the I-to-P frame distance (M). This

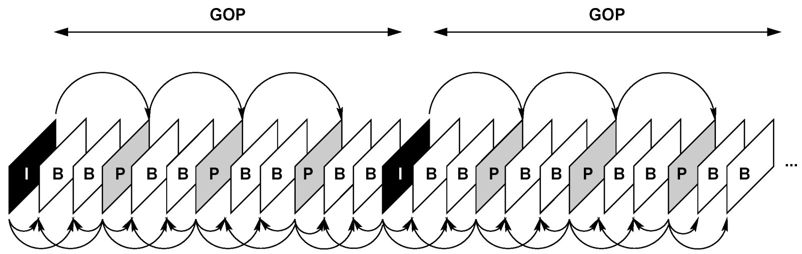


Figure 1. GOP structure ($N = 12$ and $M = 3$).

hierarchical structure with dependencies for the decoding of some frames in each GOP is illustrated in figure 1. For a given pattern (N, M), the number of frames from each type ($N_{\{I,P,B\}}$) in a single GOP are $N_I = 1$, $N_P = \frac{N}{M} - 1$, and $N_B = \frac{N}{M}(M - 1)$.

In order for a P or B frame to be considered decodable, all the frames they depend on must also be considered decodable. Thus, an incorrect I or P frame propagates problems to all dependent frames and corrupts these as well. In the worst case, a whole GOP may be considered undecodable due to an incorrect I frame, as all other frames in the GOP depend directly or indirectly on the I frame. For example, the effects of an undecodable I frame of a GOP with $N = 12$ will persist for 12 frames, or almost 500 ms at a typical frame rate of 25 frames per second, which is quite noticeable to a user.

To evaluate the transmission of an MPEG video stream, we consider the following metrics: *Fraction of decodable frames* and *fraction of useless data received*. The former reports the number of decodable frames over the total number of frames sent by a video source. The latter reports the fraction of the data received that is useless to the decoder. The decoder may have useless data from partially received frames or from frames that can not be decoded because they depend on other undecodable frames.

The adopted criterion to consider a frame decodable defines that at least a fixed fraction dt (decodable threshold) of the data in each frame must be received to consider the frame decodable. Further, a frame is considered decodable if and only if all the frames it depends on are also considered decodable. Therefore, when $dt = 1.0$, the decoder is completely intolerant to losses, that is, one lost packet is enough to render a frame undecodable. Similarly, with $dt = 0.75$, 25% of the data from a frame may be absent due to losses in the network and the frame is still considered decodable. If the decoder can tolerate a certain level of losses ($dt < 1.0$), then we assume the use of a FEC scheme in the video transmission. The tolerance to losses is obtained at the expense of additional data introduced into the video stream (FEC redundancy).

4. Analytical model

We adopt the notation described in Table 1 to model the impact on quality considering the defined evaluation metrics. From the hierarchical structure of MPEG encoding, a frame may be lost (considered undecodable) directly or indirectly. Direct loss of a frame indicates that

Table 1. Adopted notation.

Variable	Definition
ξ_I, ξ_P, ξ_B	Probability for a frame to be directly lost, but not indirectly lost
$\alpha_I, \alpha_P, \alpha_B$	Packet loss probability of packets containing data from each type of frame
C_I, C_P, C_B	Mean number of packets to transport the data of each type of frame

there are not enough frame data to decode it. We assume direct frame losses to be mutually independent. Indirect loss of a frame happens when a frame is considered undecodable because some frame it depends on is directly undecodable. We observe also that consecutive B frames have the same dependency throughout the hierarchical structure and may be considered as a B group. Each GOP pattern has $(M-1)$ B frames in each B group. Thus, the expected number of correctly decodable frames in a GOP (cf. Appendix A) is

$$N_{\text{dec}} = (1-\xi_I) + (1-\xi_I)\sum_{j=1}^{N_P}(1-\xi_P)^j + (M-1)(1-\xi_I)(1-\xi_B) \times \left((1-\xi_I)(1-\xi_P)^{N_P} + \sum_{j=1}^{N_P}(1-\xi_P)^j\right). \tag{1}$$

To evaluate the quality of a video stream, we adopt as a criterion the fraction of decodable frames per GOP. Therefore, we define the picture quality (Q) of an MPEG video stream as

$$Q = \frac{N_{\text{dec}}}{N}, \quad 0 \le Q \le 1. \tag{2}$$

The measure Q is an objective metric to assess video quality. Evidently, in video communications (such as in communications in general) we would rather be interested in evaluating QoS as observed by (human) end users, taking into account measures such as MOS (mean opinion scores) [19]. As our study is based on a modeling approach, contents of video streams (i.e. the semantics of video communications) are not taken into account. In earlier studies [19, 32], we observed that a quality measure such as Q is slightly pessimistic as compared to the quality observed by humans, because MPEG streams tend to recover from partial losses of frames thanks to their temporal and spacial redundancy. Nevertheless, measure Q tends to be a sufficiently good approximation to subjective video quality assessments, more so, as in network optimizations there is some inherent approximation in boundary conditions (such as overall system load) anyway.

At the receiver of a video transmission, the mean number β of correctly received packets for each frame type are $\beta_I = (1-\alpha_I)C_I$, $\beta_P = (1-\alpha_P)C_P$, and $\beta_B = (1-\alpha_B)C_B$. Thus, for an entire GOP the expected number of correctly transmitted packets is

$$P_{\text{corr}} = \beta_I N_I + \beta_P N_P + \beta_B N_B. \tag{3}$$

From the total number of correctly received packets, not all of them are useful to decode a frame. Useful packets are those which belong to a decodable frame. Using Eq. (1) and the mean number β of correctly received packets for each frame type, the expected number of useful packets in a GOP is

$$P_{\text{useful}} = (1-\xi_I)\beta_I + (1-\xi_I)\beta_P \sum_{j=1}^{N_P}(1-\xi_P)^j + (M-1)(1-\xi_I)(1-\xi_B)\beta_B \times \left((1-\xi_I)(1-\xi_P)^{N_P} + \sum_{j=1}^{N_P}(1-\xi_P)^j\right). \quad (4)$$

A fraction of the correctly received packets is not useful to the decoder. This fraction (U) of useless data received is defined as

$$U = \frac{P_{\text{corr}} - P_{\text{useful}}}{P_{\text{corr}}}, \quad 0 \le U \le 1. \quad (5)$$

From Q and U we can evaluate the delivery quality of an MPEG video stream and the waste of resources due to some amount of useless data received.

To validate the described analytical model, we carried out simulations in the ns-2 simulator [13] using the network scenario depicted in figure 2. A video source generates packets based on a public available MPEG frame trace file (see [15] and references therein for details of the generation of the frame trace file). We used the Starwars-IV movie sequence, encoded with high quality, which has a mean bit rate of 0.28 Mbps and peak rate of 1.9 Mbps. Each frame is fragmented into packets of 200 octets. The video stream is transmitted from node V0 to node V1 over UDP. The TCP background sources use the TCP-Reno implementation. The TCP cross traffic is generated by FTP applications, which are active during the whole simulation, with packets of 1500 octets. A number of TCP sources are uniformly distributed among the nodes T0, T1, and T2, and send packets to nodes D0, D1, and D2, respectively.

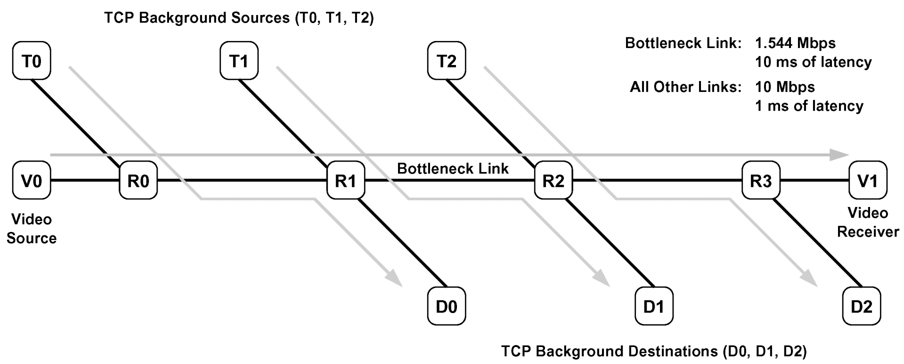
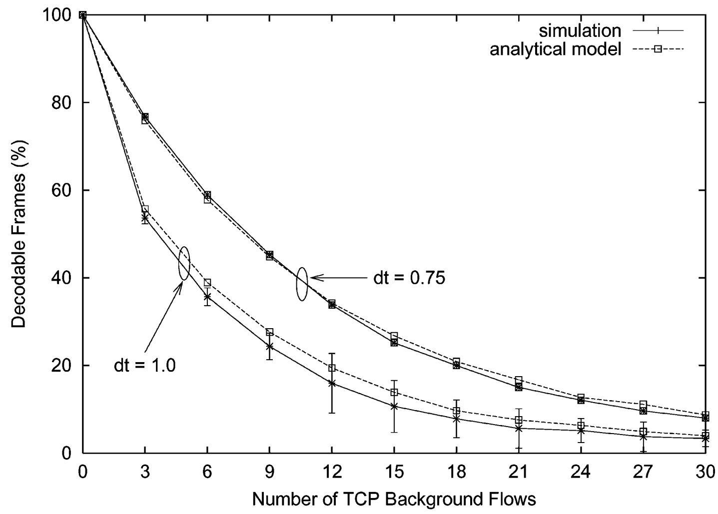
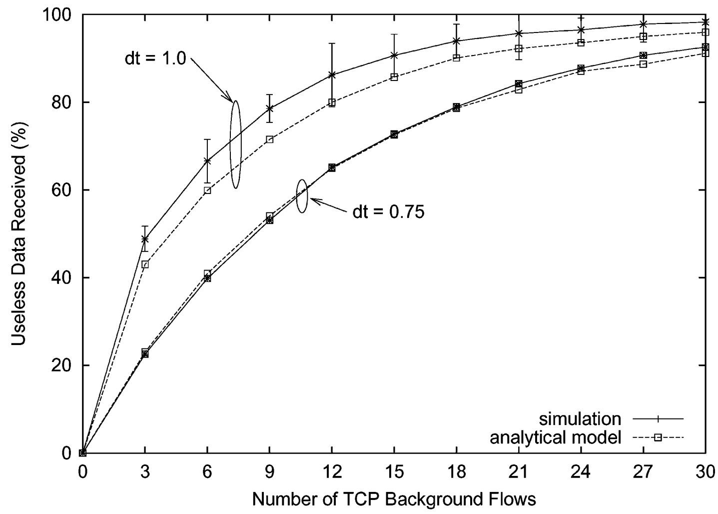


Figure 2. Network scenario.



(a) Decodable frames



(b) Useless data received

Figure 3. MPEG video stream through a best-effort network.

Doing so, independent TCP background traffic interferes with the video stream at each hop. The TCP sources start transmitting before the video sources. Thereby, video streams find intermediate routers already loaded with background traffic, avoiding thus initial transient periods. Such actions provide a conservative and pessimistic scenario that reflects realistic traffic conditions.

Figure 3 shows the fraction of decodable frames (Q) and fraction of useless data received (U) in face of an increasing load in the network. In the simulation results, the error bars show the 99% confidence interval. We compare simulation results to the results obtained from our analytical model that has been fed with the parameters ξ and α extracted from the simulations. Besides validating the model, the results make evident how strongly an MPEG stream may be degraded in a best-effort network. Although figure 3 indicates that a certain level of loss tolerance ($dt < 1.0$) may significantly improve the quality of the video transmission, we investigate hereafter the effects of the compression factor, the tolerance to losses, and the introduction of drop precedence levels.

5. QoS and video communications

Our objective is to evaluate the video transmission of an MPEG stream considering the compression factor of different GOP patterns, the level of tolerance to losses, and the mapping of different types of frames onto different drop precedence levels. Furthermore, our investigation covers the effects of simultaneously adopting such QoS strategies. To perform the evaluation, we consider two classes of GOP patterns:

$$\begin{aligned} \text{Class 1: } & I(BP)^k; \quad k \in \{1, 2, \ldots, 7\} \\ \text{Class 2: } & I(BBP)^k BB; \quad k \in \{1, 2, 3\} \end{aligned}$$

GOPs from Class 1 have less B frames than GOPs from Class 2 for the same GOP size (N). Therefore, GOPs from Class 2 are expected to have on average a higher compression rate than comparable GOPs from Class 1 when applied to the same video sequence using a constant quantizer. By comparable GOPs, we mean two GOP patterns that despite belonging each one to one of the defined classes have the same GOP size (N).

To illustrate the quality degradation because of the hierarchical structure of an MPEG stream and the effects on quality of the compression factor on different GOP patterns, we consider the following situation. Suppose the probability ξ of directly losing a frame being the same whatever the type of the frame is ($\xi = \xi_{\{I,P,B\}}$). In such an environment, the quality offered by GOPs belonging to Class 1 degrades slightly stronger when facing an increasing frame loss probability ξ than the quality offered by comparable GOPs from Class 2, as shown in figure 4. As opposed to intuition, this happens even if GOPs from Class 2 are expected to be more compressed on average than comparable GOPs from Class 1. The explanation to this seemingly contradictory finding is that GOPs from Class 1 have more P frames for the same GOP size (N). As the different frames share the same probability ξ for direct frame loss, GOP patterns belonging to Class 1 are expected to have a greater probability of directly losing a P frame and thereby indirectly losing the frames that depend on the directly lost P frame as well. The effects on quality of the compression factor of different GOP patterns are further discussed in Section 5.1.

For a given frame type $F \in \{I, P, B\}$ and considering no loss tolerance ($dt = 1.0$), the probability ξ_F of directly losing a frame F is mapped from the packet loss probability α_F

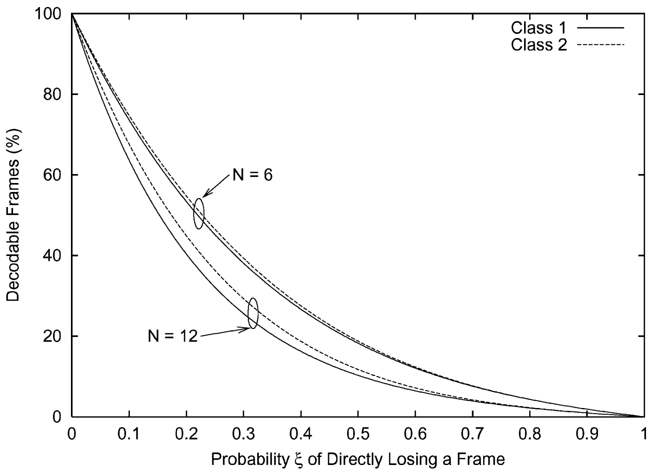


Figure 4. Effects of the probability ξ on comparable GOPs from Classes 1 and 2.

by

$$\begin{aligned}\xi_F &= \sum_{k=1}^{C_F} \binom{C_F}{k} \alpha_F^k (1-\alpha_F)^{C_F-k} \\ &= 1-(1-\alpha_F)^{C_F}.\end{aligned} \tag{6}$$

In a non-differentiated environment, like a best-effort network, the probability α of losing a packet is the same whatever the frame content of the packet is ($\alpha = \alpha_{\{I,P,B\}}$). In such an environment, Eq. (6) may be simplified to $\xi_F = 1-(1-\alpha)^{C_F}$. We assume packet losses to be mutually independent. Figure 5 illustrates the effect of the packet loss probability α on comparable GOPs from Class 1 and 2 for GOP size $N = 12$. In the case the number of packets containing data from each type of frame is the same whatever the type of frame is ($C_I = C_P = C_B = 1$ in figure 5), we have the same probability ξ of directly losing the frame for all types of frames. Since less compressed frames take up more data and thus may need more transmitted packets than the more compressed frames ($C_I \geq C_P \geq C_B$), if the same packet loss probability α applies to all packets, then $\xi_I \geq \xi_P \geq \xi_B$. In figure 5, two particular cases ($C_I = 4, C_P = 2, C_B = 1$ to consider larger packets and $C_I = 20, C_P = 10, C_B = 5$ to consider smaller packets) are shown to depict the quality degradation of an MPEG stream in a non-differentiated environment with no loss tolerance. The most important frames may suffer higher direct frame loss rates because they have more packets and the packet loss rate is the same for all packets independent of the frame information they carry. Such a higher probability for direct frame losses of the most important frames implies indirect losses of dependent frames, strongly compromising the delivery quality in a non-differentiated environment.

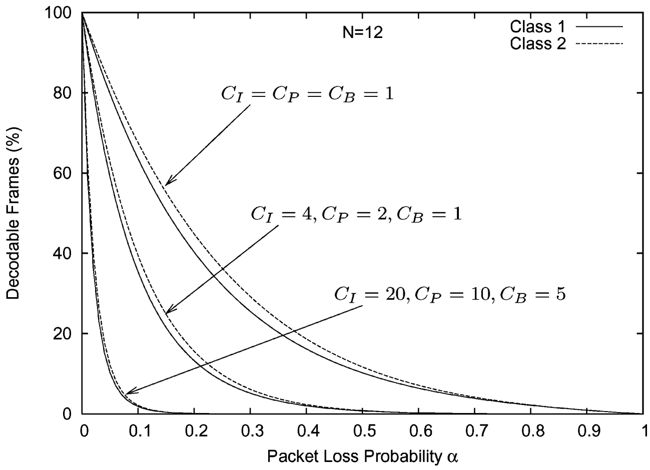


Figure 5. Effects of the probability α on comparable GOPs from Classes 1 and 2.

Differentiation among packets from distinct frames may be achieved by mapping those packets onto different drop precedences within an Assured Forwarding (AF) class [20]. The mapping of frames onto network priorities P_h (high priority) and P_l (low priority) covers three possibilities as shown in Table 2. Background traffic is always mapped onto P_l. Network priorities P_h and P_l are implemented as two different virtual queues managed in a RIO (RED with IN and OUT) scheme [6], reflecting two different levels of drop precedence within an AF class. The RIO parameters are min, max, and $P_{\max}$ (see [6, 16] for further details about the meaning of each parameter). Issues as how to best configure the RIO parameters are outside the scope of this paper. Our goal when adopting the RIO mechanism is to provide a certain level of discrimination between the P_l and P_h packets that would be consistently effective, but not necessarily optimal, in the general case for any MPEG video stream.

In the cases a simulation applying the mapping variants of Table 2 is needed, routers R0, R1, and R2 (see figure 2) adopt a RIO mechanism for active queue management instead of the traditional Drop Tail policy. The threshold parameters of the RIO mechanism for the two drop precedence levels are configured as percentages of the total queue length ($qlen$), which is kept equivalent to 50 packets. The RIO parameters (min, max, $P_{\max}$) we use

Table 2. Mapping of frames onto network priorities.

Mapping	I	P	B
M1	P_h	P_h	P_h
M2	P_h	P_h	P_l
M3	P_h	P_l	P_l

are ($0.2 * qlen$, $0.5 * qlen$, 0.1) and ($0.5 * qlen$, $0.8 * qlen$, 0.02) for P_l and P_h packets, respectively. This set discriminates P_l packets, progressively avoiding the discard of P_h packets. We further discuss the effects of different drop precedence levels in Section 5.3.

For the experiments, we first take two GOP patterns, named GOP1 and GOP2:

GOP1: *I B P B*
GOP2: *I B B P B B P B B P B B*

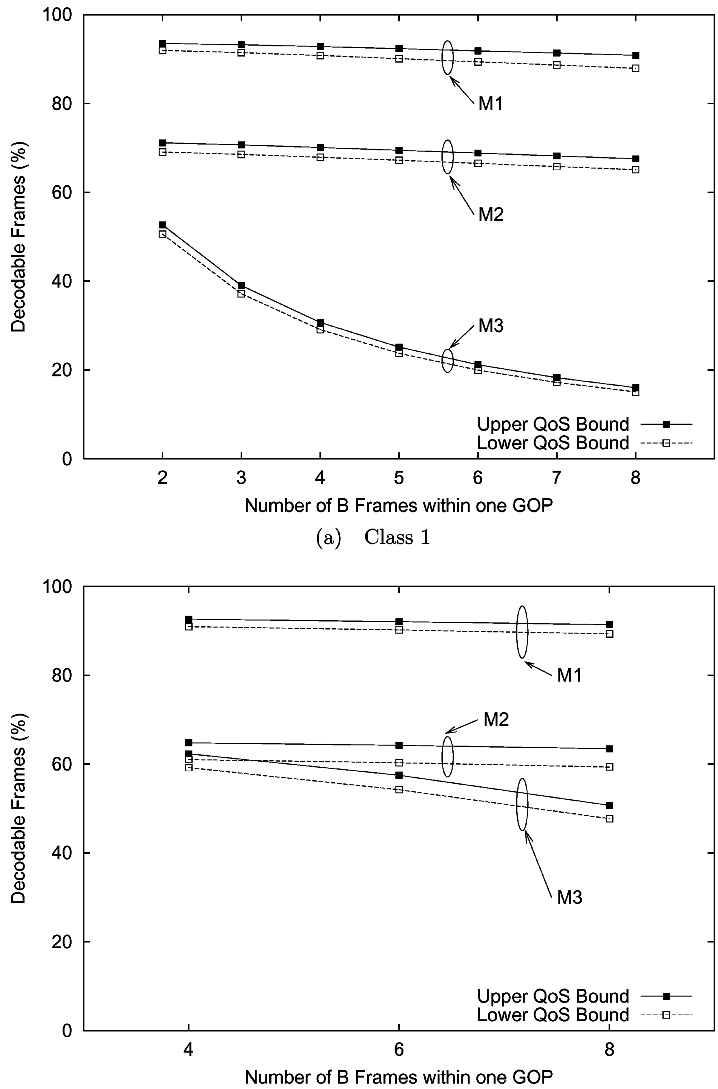
GOP1 and GOP2 correspond respectively to the lowest and highest compression for GOP patterns belonging to Class 1 and to Class 2. For each proposed experiment, we extract through simulation over the scenario of figure 2 the parameters ξ and α for GOP1 and GOP2 when facing 30 TCP background flows, the case where quality is already heavily degraded in figure 3. Then, these parameters are used to feed our analytical model and obtain the results for the large variety of remaining patterns. It should be noted that these analytical QoS predictions are done without any support by additional simulation experiments. Results are shown as a range between the highest and lowest expected picture quality Q (as defined in Eq. (2)) for the MPEG video stream under the given conditions, i.e. the analytical model provides an upper and a lower bound for the video quality to be expected by the receiving end user. The level of quality predicted analytically in this way is valid (i.e. very close to actual system behavior) indeed as we could successfully verify thanks to additional simulation experiments.

5.1. *Effects of compression*

Experiment I compares the performance achieved by GOP patterns that have different compression factors. We consider all the GOP patterns that belong to either Class 1 or Class 2, as shown in Table 3. There are $k + 1$ and $2(k + 1)$ B frames at each GOP in Class 1 or Class 2, respectively. Figure 6 presents the fraction of decodable frames of patterns from each class handled by different mappings onto network drop precedences. The results presented here consider no loss tolerance ($dt = 1.0$).

Table 3. GOP patterns for Experiments I and II.

GOP pattern	B frames
I B P B	$2 \times B$
I B P B P B	$3 \times B$
⋮	
I B P B P B P B P B P B P B P B	$8 \times B$
I B B P B B	$4 \times B$
I B B P B B P B B	$6 \times B$
I B B P B B P B B P B B	$8 \times B$



(a) Class 1

(b) Class 2

Figure 6. Effects of compression with $dt = 1.0$.

The higher the compression is within each class (more *B* frames), the lower is the expected quality. For both GOP classes the decrease in quality due to an increasing compression is relatively small on either M1 or M2. However, a more significant degradation in quality is observed for M3. This degradation in quality because of an increasing compression is stronger in GOPs from Class 1 than in GOPs from Class 2. This points out the importance of the *P* frames. In Class 1, for the same number of *B* frames we have more *P* frames.

Moreover, we observe how schemes that protect the data from P frames can significantly improve the expected quality of a video stream.

5.2. *Effects of tolerance to losses*

In experiment II, we evaluate the QoS improvement if the decoder can tolerate a number of losses in a frame and thus allows to still consider such frames as being decodable ($dt < 1.0$). The adoption of a FEC scheme provides this feature at the expense of data redundancy. The intended FEC scheme [17] adds redundant information in a fashion that k video packets are protected by FEC in such a way that n packets ($n > k$) are transmitted, which are all of the same importance to the receiver. If at least k packets out of the n packets composing a frame are correctly received, the video information can be correctly decoded and the frame is considered decodable. Such result is independent of which packets are lost within the frame.

We denote by c_0 the needed data rate to send the video stream when $dt = 1.0$. When $dt = 1.0$, the system is completely intolerant to losses and there is no redundant data in the video stream as no FEC scheme is in use. Given a certain decodable threshold (dt) value q ($0 \leq q \leq 1$), the needed capacity to send the redundant video stream is $c(q) = \frac{c_0}{q}$. Thus, considering $c(q) = c_0(1 + v(q))$, the overhead $v(q)$ imposed by the adoption of a certain level of tolerance to losses is given by

$$v(q) = \frac{c(q) - c_0}{c_0} = \frac{1 - q}{q}. \tag{7}$$

Similar relation between the FEC overhead and the level of loss tolerance has been found in [29] when modeling the usage of Priority Encoding Transmission (PET) [2] to transfer MPEG video streams. Therefore, to take into account the data redundancy due to the adoption of a FEC scheme, we use Eq. (7) to consider an overhead of 33% for $dt = 0.75$ and of 100% for $dt = 0.50$ in the size of each frame. The GOP patterns are the same as for experiment I (Table 3). Under the M1 mapping variant, all packets from video frames are mapped onto P_h and thus the background traffic has a higher drop precedence when compared with video packets from any frame. As a consequence of not having many discards of video packets, the results for different values of dt almost overlap, diminishing the effects of some tolerance to losses. Under mapping variants where certain video packets and the background traffic are mapped onto the same drop precedence, the benefits of some tolerance to losses become clear as in figures 7 and 8, which present different values of dt handled by the M2 and M3 mapping variants, respectively.

Considering the M2 mapping variant (figure 7) and the same level of tolerance to losses, an increasing compression within each class of GOPs does not significantly influence the perceived quality of the delivered stream. Nevertheless, the application of a FEC scheme can strongly improve the delivery quality of the MPEG video stream (similar observation as in [32]). GOPs from Class 1 take a better advantage of a certain level of loss tolerance. This can be seen when comparing GOP patterns from the two classes with the same number of frames, for example, comparing $N_B = 6$ in figure 7(a) and $N_B = 8$ in figure 7(b).

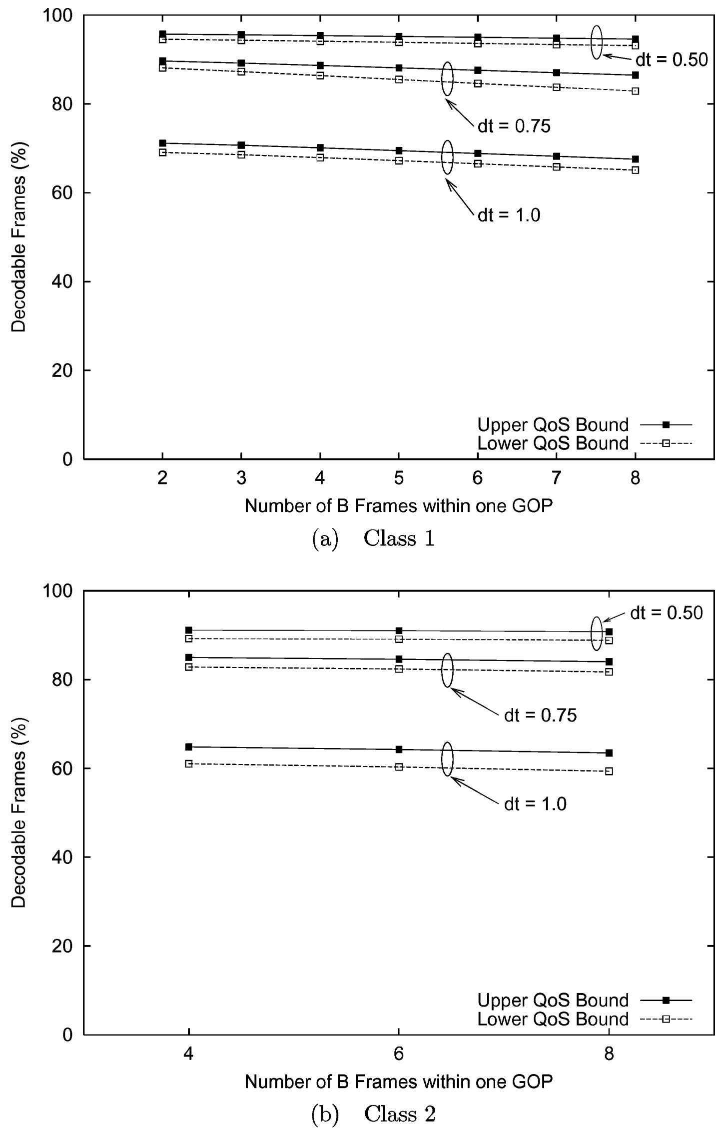


Figure 7. Effects of tolerance to losses for the M2 mapping variant.

For the M3 mapping variant (figure 8), however, we observe that the quality offered by GOPs from Class 1 degrades more quickly with an increasing compression than the quality provided by GOPs from Class 2 (cf. figure 8(a) in comparison with figure 8(b)). Such effect is due to the presence of more *P* frames in GOPs from Class 1. As *P* frames are not discriminated from *B* frames, contrasting figures 7 and 8 shows, as expected, that the quality degrades faster under the M3 mapping variant than under M2, specially for GOPs

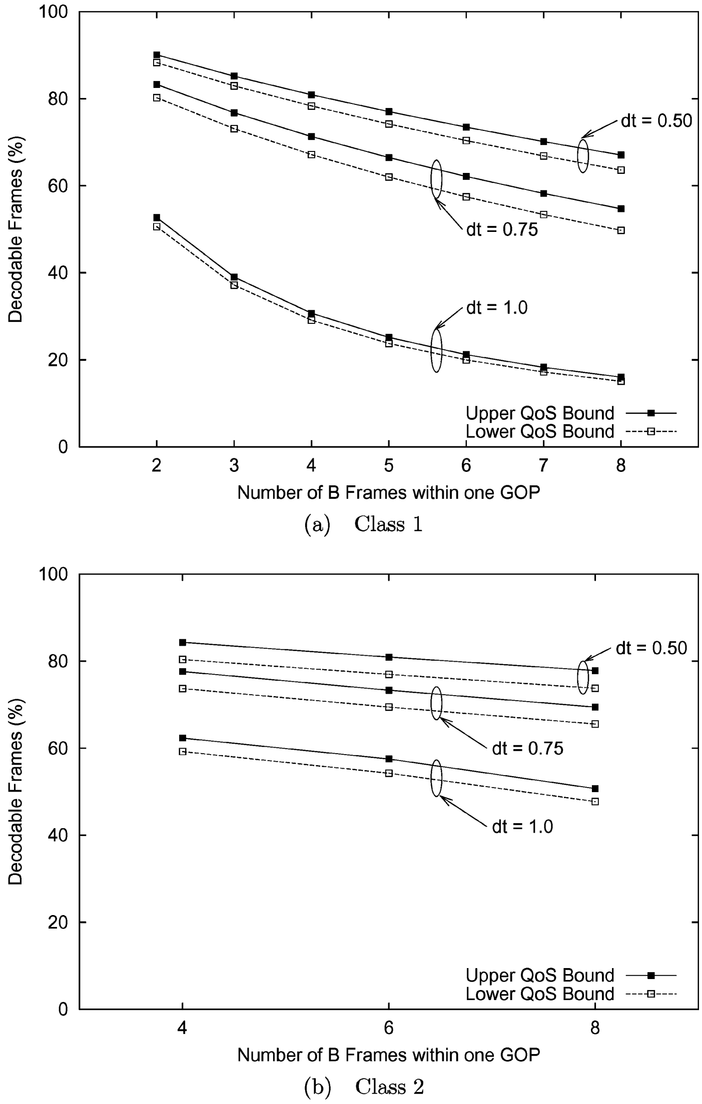


Figure 8. Effects of tolerance to losses for the M3 mapping variant.

belonging to Class 1. Once again, a larger number of P frames in GOPs from Class 1 contributes for such stronger degradation.

5.3. *Effects of drop precedence levels*

Experiment III compares the appliance of different drop precedence levels or priority mappings (Table 2) to five representative GOP patterns presented in Table 4. Some of these GOP

Table 4. GOP patterns for Experiment III.

	GOP pattern	B Frames
1	$IBPB$	$2 \times B$
2	$IBPBPBPB$	$4 \times B$
3	$IBBPBB$	$4 \times B$
4	$IBPBPBPBPBPBPBPB$	$8 \times B$
5	$IBBPBBPBBPBB$	$8 \times B$

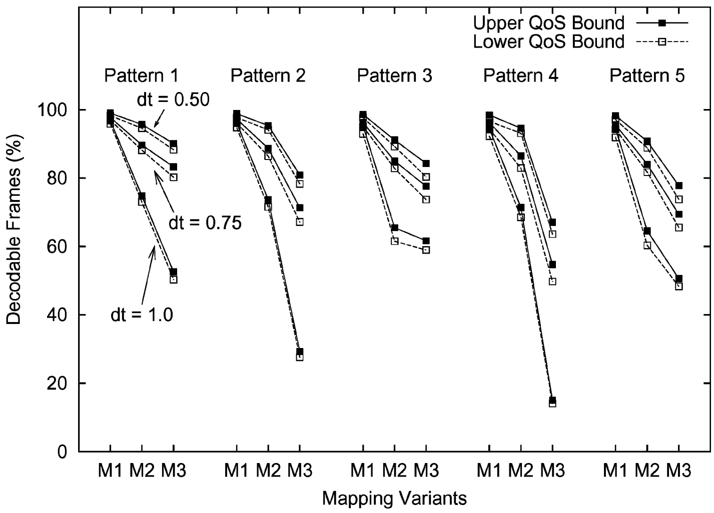


Figure 9. Drop precedence mappings for different GOP patterns.

patterns present the same number of B frames, although distributed in a different manner depending on which class they belong to. Figure 9 shows the fraction of decodable frames for each pattern handled by a different drop precedence mapping and for different levels of loss tolerance.

For the same number of B frames, quality degrades more on GOPs from Class 1 than on GOPs from Class 2 as frames get a lower priority mapping. Note that for the same number of B frames, GOPs from Class 1 have more P frames. Again, this shows the importance of P frames in the hierarchical structure of MPEG. GOPs from Class 2 take a better advantage of a certain tolerance to losses provided by a FEC scheme as frames get a lower priority mapping. As opposed to intuition, more compressed GOP patterns may have better quality than less compressed ones under specific circumstances (as an example, cf. Pattern 1 versus Pattern 3, for dt = 1.0 and M3 mapping in figure 9). This quite astonishing result may become more evident to the reader by considering that under a mapping as M3, a pattern like *IBBB* is expected to get on average a better quality than a less compressed pattern like *IBPB*.

6. Conclusion

In this paper, we have identified a variety of techniques that allow the improvement of the delivery quality of MPEG streams to end users. From an analytical model, we have evaluated the impact of frame losses in the quality of MPEG streams and in the waste of network resources. Our assessment has considered issues such as the use of redundancy either by applying a FEC scheme to tolerate losses or by changing the compression factor in a GOP pattern generation. We have also covered how to best use a differentiation scheme among different types of MPEG frames and how to adequately protect them from losses. Furthermore, we have evaluated the effects of simultaneously adopting different QoS strategies to improve the delivery quality of MPEG streams.

The obtained results pointed out the importance of P frames in the GOP structure. Any scheme that intends to differentially handle data from different types of frames must strongly consider the protection of P frames to achieve a significant quality improvement. This is true, specially if the adopted GOP pattern belongs to Class 1. The difference between the highest and the lowest quality as predicted by means of our analytical model (in order to get bounds for performance as observed on Application Layer) is always small and therefore negligible, which suggests the good predictability quality that may be achieved by the adopted model. Another key point is the need of adopting FEC mechanisms, as they can strongly improve the delivery quality over lossy networks. The combination of protecting more important frames (from the point of view of the hierarchical structure of MPEG encoding) with an additional tolerance to losses for the frames that are more susceptible to losses has been shown as an efficient strategy to improve the delivery quality of a video transmission. From our findings, GOP patterns belonging to Class 2 can typically achieve a stronger quality improvement by the different analyzed techniques. Hence, for most boundary conditions, we recommend the adoption of GOP patterns from Class 2 to take a better advantage of the analyzed QoS schemes.

Appendix A: Correctly decodable frames in a GOP

The expected number of correctly decodable frames in a GOP is calculated based on the probability for a frame (I, P, or B) to be directly lost and the impact of such a loss in the subsequent frames due to the hierarchical structure of MPEG encoding. The model of Eq. (1) takes into account all the consequences of direct frame losses which impose indirect frame losses within a GOP, though the direct frame losses themselves are assumed to be mutually independent. We denote by $S(F)$ the probability that a frame F is considered decodable. Each term of Eq. (1) refers to the effects of the direct loss of one particular type of frame within a given GOP_x based on the probability of correctly decoding the frame and by considering its dependencies:

(a) The probability $S(I)$ that the I frame in GOP_x is decodable is

$$S(I) = 1 - \xi_I.$$

Thus, the expected number of correctly decodable I frames in GOP_x is

$$N_{\mathrm{dec}}^{I} = 1 - \xi_I.$$

(b) The probabilities $S(P_1)$, $S(P_2)$, $\ldots$, $S(P_{N_P})$ that each of the N_P frames of type P in GOP_x is decodable are

$$\begin{aligned} S(P_1) &= (1-\xi_I)(1-\xi_P),\\ S(P_2) &= (1-\xi_I)(1-\xi_P)^2,\\ &\vdots\\ S(P_{N_P}) &= (1-\xi_I)(1-\xi_P)^{N_P}. \end{aligned}$$

The expected number of correctly decodable P frames in GOP_x is thus

$$N_{\mathrm{dec}}^{P} = \sum_{j=1}^{N_P} S(P_j) = (1-\xi_I)\sum_{j=1}^{N_P}(1-\xi_P)^j.$$

(c) As consecutive B frames have the same dependency throughout the hierarchical structure, we consider consecutive B frames as composing a B group. Each GOP pattern has $(M-1)$ B frames in each B group. We represent by B_n each of the B frames that compose the nth B group. The probabilities $S(B_1)$, $S(B_2)$, $\ldots$, $S(B_{\frac{N}{M}-1})$ that each B frame in each of the $\frac{N}{M}-1$ initial B groups is considered decodable are

$$\begin{aligned} S(B_1) &= (1-\xi_I)(1-\xi_P)(1-\xi_B),\\ S(B_2) &= (1-\xi_I)(1-\xi_P)^2(1-\xi_B),\\ &\vdots\\ S\big(B_{\frac{N}{M}-1}\big) &= (1-\xi_I)(1-\xi_P)^{\frac{N}{M}-1}(1-\xi_B). \end{aligned}$$

The last B group is also affected by the I frame of GOP_{x+1}. Thus,

$$S\big(B_{\frac{N}{M}}\big) = (1-\xi_I)^2(1-\xi_P)^{\frac{N}{M}-1}(1-\xi_B).$$

As each B group has $(M-1)$ B frames and $N_P = \frac{N}{M} - 1$, the expected number of correctly decodable B frames in GOP_x is

$$\begin{aligned} N_{\mathrm{dec}}^{B} &= (M-1)\sum_{j=1}^{\frac{N}{M}} S(B_j)\\ &= (M-1)(1-\xi_I)\sum_{j=1}^{N_P}(1-\xi_P)^j(1-\xi_B) \end{aligned}$$

$$
\begin{aligned}
&\quad + (M-1)(1-\xi_I)^2(1-\xi_P)^{N_P}(1-\xi_B) \\
&= (M-1)(1-\xi_I)(1-\xi_B) \\
&\quad \times \left((1-\xi_I)(1-\xi_P)^{N_P} + \sum_{j=1}^{N_P} (1-\xi_P)^j \right).
\end{aligned}
$$

Therefore, the expected number of correctly decodable frames in a GOP is $N_{\text{dec}} = N_{\text{dec}}^I + N_{\text{dec}}^P + N_{\text{dec}}^B$, that is equivalent to Eq. (1).

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