Dynamic Rate Shaping on MPEG-2 Video Streams for Bandwidth Saving on a Faded Satellite Channel*

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Abstract. In a previous work, we studied how to send a video stream over a noisy satellite channel using the scalability features of the MPEG-2 standard. The base and the enhancement flows were sent with different levels of protection (bit and coding rates) in order to deal with the channel degradation. In this paper, we add Dynamic Rate Shaping (DRS), thus obtaining a greater flexibility in the choice of transmission parameters. In fact, it is possible to trim the required bandwidth in continuous mode in the working range, thus saving some bandwidth when keeping the video quality constant during changing fade conditions. The performances of DRS applied to both the non-scalable and the scalable MPEG-2 coded video streams are shown separately. An application of the techniques presented is discussed, and some criteria are given for choosing the most suitable parameters, according to the service quality requirements.

1 INTRODUCTION

In [1] we analysed the relationships between bandwidth occupancy, satellite link degradation and the quality of a video stream coded as the Motion Pictures Expert Group-2 (MPEG-2) standard [9], and sent over a satellite channel. We observed that it is possible to decode a corrupted video stream when the bit error rate (BER) is in the range $10^{-7} \div 10^{-3}$ depending on the video quality required and the type of channel coding applied to the data sent on satellite. We proposed a method for maintaining the video quality not lower than a required level while the signal fading changes, by varying the bit and coding rates of the data stream. We exploited the signal to noise ratio (SNR) scalability of MPEG-2 in order to enhance the video stream transmission's ability to deal with the channel degradation. A scalable approach was preferable to a simulcast technique [6], where a set of independent streams of the same video sequence were transmitted simultaneously at different bit rates. The scalability was also useful because it allowed a lower protection of the enhancement, in order to better protect the base flow, up to the limit case of totally discarding the enhancement, in any point of the transmission chain, when severe conditions of signal fading require so. We found that it is feasible to trade video quality for bandwidth in the degradation range considered. The main drawbacks are the increased complexity of the encoder and the fact that the intermediate nodes cannot graduate the flow.

The logical continuation of that work was to study how to trim the flow more gradually so that a wide variety of channel conditions can be addressed. This operation can generally be performed by any intermediate node that knows the real channel conditions. Moreover, for practical reasons, the computational effort must be very low. With these requirements in mind we have designed a rate shaper that is able to trim the MPEG-2 bitstream. This rate shaper should be considered as a parser because it does not perform a decoding and requantization process (this would be too expensive) but simply parses the bitstream and selectively discards an amount of information proportional to the percentage of the flow reduction required. The technique used is called dynamic rate shaping (DRS).

Unlike other works in which optimality conditions were imposed on the rate shaper [8] our solution is very simple. Since it is driven only by the flow rate, it can trim the bitstream very finely in order to dynamically adapt it to the bandwidth requirements.

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The present work investigates on the best compromise between the bandwidth occupancy and the average video quality achieved, provided that a stream of MPEG-2 coded data is available in non-scalable and/or scalable form. It may be possible to apply a DRS compression technique on the variable bit rate (VBR) part of the flows as well. In other words, given a satellite channel, shared in TDMA among a variety of users, the present work outlines some adaptive techniques to send a video coded data stream, in order to obtain a certain quality of service in different states of the link which may vary with the weather (rain attenuation) and/or traffic conditions. We assume that the channel is affected by additive white Gaussian noise (AWGN).

We have improved the results obtained in [1] by trimming the bitstream so that it can be better protected, while still occupying the same bandwidth, and we have tried to answer the question about how to put a VBR transmission over a channel limited in band.

In the following three sections the DRS technique, its implementation and performance are presented. Section 5 outlines the experiment we set up to produce the measurement results about the performance of the various video data transmission techniques, and in Sections 6 and 7 DRS on non-scalable and scalable video coded streams are analysed, respectively. In section 8 a possible application of DRS is discussed, and conclusions are drawn in section 9.

2 DYNAMIC RATE SHAPING

Rate shaping is a technique to adapt the rate of compressed video bitstreams to variable rate constraints. It provides an interface between the encoder and the network, which allows a perfect match of the encoder's output with the network's quality of service characteristics.

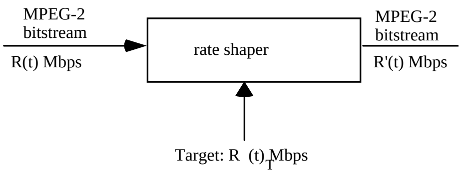

Figure 1: General scheme of DRS.

The approach does not require any interaction with the encoder, so it can be applied in all precoded and stored video services (i.e. video-on-demand video systems) where a wide variety of video programs may be available. In these applications a multiresolution coding with too many layers would be undesirable, due to the loss in channel efficiency. A general rate shaper scheme is illustrated in figure 1. An input MPEG-2 stream with instant bit-rate of R(t) Mbps is squeezed in order to achieve a constrained instant bit-rate of $R_T(t)$. The instant output bit-rate, $R'(t)$, is defined as:

$$R'(t) \cong R_T(t) \ \text{ if } \ R_T(t) < R(t)$$

$$R'(t) = R(t) \ \text{ if } \ R_T(t) \geq R(t)$$

In a general situation, R(t) and $R_T(t)$ can either be constant or variable, and are associated with the characteristics of the source and channel.

2.1 BIT-RATE CONVERSION

Assuming a general hybrid motion-compensated block-based transform coding technique, there are two fundamental ways to reduce the rate:

- adopt a transcoder, composed of a cascaded decoder and encoder;
- selectively discard coded information: transform coefficients, macroblocks, slices, etc.

Although different levels of complexity are allowed in a transcoder [4], the first approach is significantly more complex than the second one. The reason is that the former needs a coefficient requantization that includes Huffman decoding, run-length decoding, dequantization, requantization, run-length encoding, and Huffman encoding. Despite this, an additional reconstruction error will occur, caused by a drift between the decoder's prediction signal and the original (not requantized) prediction signal. One solution is to adopt a drift correction which entails an additional inverse and direct Discrete Cosine Transform (DCT) and motion compensating prediction.

Since the scheme adopted must be as uncomplicated and as inexpensive as possible in order to install several cheap modules in suitable points of the distribution network, in this paper we consider a method based on the second approach. To further justify our choice, note that we refer to an unbalanced coding scheme (the MPEG-2 encoding process is more complex than the decoding one) and thus we generally assume pre-recorded video traces.

3 DRS IMPLEMENTATION

In our implementation, depicted in figure 2, we use a rate shaping algorithm based on a selective information discard approach, applied to MPEG-2 Variable Bit Rate (VBR) sources, which tests several output constrained bit-rates in order to match the channel bandwidth requirements.

A very important feature of this method is that the resulting bitstream syntax is faithful to the MPEG-2 standard. Rate reduction is obtained by eliminating a contiguous string of DCT coefficients at the end of each block. When, with a small target bit-rate, the rate-control is not able to track the bit-rate reduction, a selective slice discarding is performed, for bidirectionally-predicted pictures only.

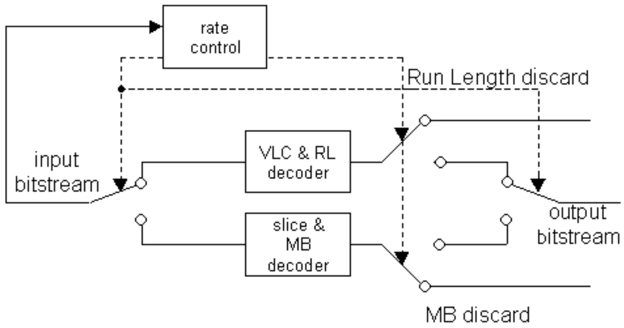

Figure 2: Rate shaping module scheme.

3.1 RATE CONTROL

The underlying idea is to use a rate control in a similar way to the one used in the Test Model 5 [3] of MPEG-2, using the parser in order to select the information to discard. A picture target bit allocation is thus calculated, based on the picture type and the target bit-rate. The rate control mechanism sets the bits available to each macroblock, by comparing the overall spent bits with the target bits by means of a virtual buffer associated with each picture type. In MPEG-2 three picture types are defined: intra-pictures (I), predicted pictures (P), and bidirectionally-predicted pictures (B). I pictures are self-contained, since they only use transform coding with a moderate compression factor. P pictures are temporally predicted and generally used for further prediction. B pictures, which provide the highest degree of compression, are coded by using a bi-directional temporal prediction. Moreover, they are never used as a temporal reference for other frames. Following this scheme, the DRS rate control calculates the target bit-rate for the GOP and, consequently, assigns target rates to I/P/B pictures. Therefore, it adapts output bitstream rates to target ones discarding some of the DCT coefficients instead of using a coarser quantization step. If this is not sufficient, the DRS rate control may discard entire slices. This is depicted in figure 2 where, in accordance with the instantaneous rate control buffering, two different parsing levels associated to blocks (variable length code parsing and run length discard) and to slices/macroblocks can be selected. The rate control feedback is performed by using as overall bits of information spent the effective number of bits kept from the DRS module. Within each picture type, the macroblock bits are distributed to the various blocks (luminance and chrominance), by assigning them the number of bits that can be kept.

3.2 DCT COEFFICIENT DISCARD

In normal conditions the rate shaper parses and counts the input bitstream, by copying it in output. The discarding of the DCT coefficients starts when, at a block level, the number of input bits exceeds the target one determined by the rate control. In this situation, the last run-length code encountered is followed by an end-of-block code by discarding the remaining coefficients associated with the higher spectral frequencies. The process is very similar to MPEG-2 data partitioning, though here the breakpoint value (the number of DCT run-length codes to be kept in each block) is not a fixed value within the same picture, since it is generally different for each block. We require that at least one DCT coefficient remains in each block. This is necessary in order to avoid re-calculating and re-coding some syntax fields, such as the coded block patterns (which indicate which blocks in each macroblock are included in the bitstream), DC prediction, where DC is the DCT coefficient associated with frequency zero (mean value), and the macroblock address within the current slice. In this way, breakpoint values range from 1 to 64. The fundamental elements of this method are a bitstream parser and a rate control module. The DCT coefficients' discard is applied on I, P and B pictures.

3.3 SLICE DISCARD

Sometimes discarding DCT coefficients is not sufficient to track the shaping requirement. This is especially true when the source input bitstream has a high compression ratio, as in our case. In this situation the source coder applies a coarse quantization and only a small number of coefficients are coded in the original bitstream. In other words, DCT coefficients are a relatively small percentage of the coded information, which includes overhead information such as header fields and motion vector. Thus, when the rate shaping compression ratio exceeds the DCT coefficient rate, the rate control mechanism fails. To compensate for this situation a slice discard mechanism is used for B pictures. The basic idea is that the slice is completely discarded when the picture target rate is lower than a threshold depending on the local sequence activity, and the number of bits used for all slices up to the current one exceed the target value by a given percentage. In this situation only the first and the last macroblocks are maintained.

4 PERFORMANCE OF THE PROPOSED RATE SHAPING SCHEME

The selective discarding of coded information has a predictable impact on the resulting video quality. Discarding the higher frequencies corresponds to a low-

Figure 3: MPEG-2 I-picture detail shaped at different DRS rate levels.

pass filtering process. Since it is applied to I and P pictures as well, a drift error associated with the wrong temporal prediction is added to the error caused by the coefficient discard. At high compression rates, the shaping reduces the coded signal to the side information fields and the lower frequency DCT coefficients, thus causing an annoying block effect artefact. This can be observed in figure 3, where a detail of an I picture is shown, shaped at different rates.

The slice discard in B pictures, on the other hand, has a more evident effect on the decoded pictures, since the decoder retrieves the information of the missed macroblocks as if they would be skipped (i.e. the motion vector associated with the first macroblock in the slice is used to perform the motion-compensation of the missed macroblocks).

To measure the loss of video quality of the proposed method, it is possible to compare the decoded video with the original sequence at different levels of compression. The quality of the video may be computed either by using the peak signal to noise ratio (PSNR), or by using an objective video quality assessment method [7] that emulates the HVS (Human Visual System). With this method, a linear combination is calculated of three complementary video quality measurements, based on spatial and temporal distortion. The resulting quality factor (QF) is strictly correlated with the subjective mean opinion score: imperceptible (5), perceptible but not annoying (4), slightly annoying (3), annoying (2), and very annoying (1). All the simulations are performed using a 60 second video sample taken from the movie “The sheltering sky”[1]. Figure 4 depicts the instant bit-rate and the corresponding mean errors expressed in PSNR for different levels of rate shaping (from 5% to 50%) applied to the video sample coded in VBR mode using a fixed quantization step of 6. All the other results will be expressed by using the quality factor assessment method. To understand how the rate control performs the rate reduction, figure 5 plots the rate of block and slice cutting for the above specified rate-shaping range.

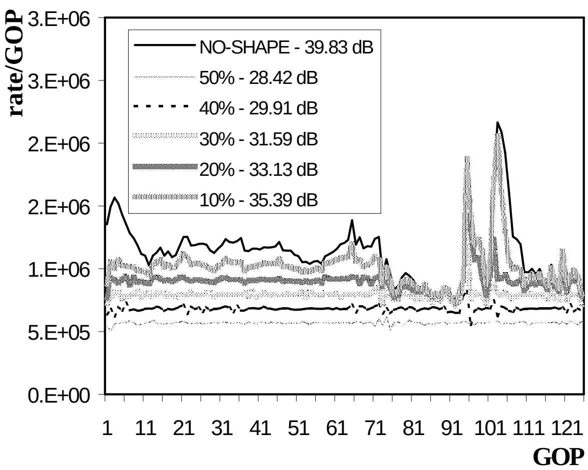

Figure 4: Instant bit-rates and mean PSNRs at different DRS compression rates.

[1] The authors would like to thank Videotime s.p.a. for providing the movie used in the simulations

To understand how the rate control performs the rate reduction, figure 5 plots the rate of block and slice cutting for the above specified rate-shaping range. As can be seen, for high percentages of rate shaping, a high B-slice cutting is the only way to track the rate control target. For this reason the B-blocks and the P-blocks cutting rate is rather low. This rate increases for medium shaping rates (20%-40%), when the rate-reduction initially leans on block cutting.

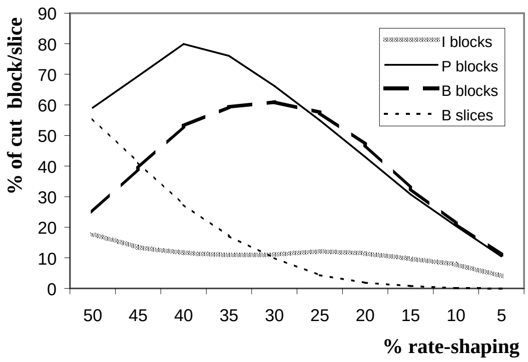

Figure 5: Cut macroblocks/slices versus DRS compression rate.

Another significant fact is the matching of video shaped at a rate of R'(t) with an unshaped trace coded at the same medium rate. Figure 6 compares the video quality between the sequence decoded from bitstreams obtained from the original at different levels of shape (5% to 50%) and the corresponding standard simulation coded at the same bit-rate.

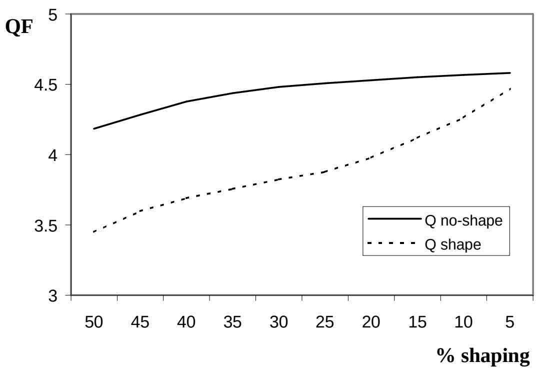

Figure 6: QF comparison between the sequence decoded at different levels of shape (5% to 50%) and the corresponding standard simulation coded at the same bit-rate.

5 THE EXPERIMENT ENVIRONMENT

On satellite channels the transmitting signal may be attenuated by the spreading factor (scintillation), atmospheric absorption (due to uncondensed water vapour, rain, fog and clouds, snow and hail, free electrons in the atmosphere, and molecular oxygen), and other losses. The signal *attenuation* is defined as the dB difference between the currently received power and the power received under clear sky conditions. Unfavourable propagation conditions (*fades*) are a major problem in transmissions above 10 GHz in that they attenuate the signal and increase the noise level thus worsening the channel signal-to-noise ratio (CSNR), which is the indicator of the channel quality. We express CSNR in terms of E_b / N_o (bit energy to one-sided noise spectral density ratio). Hereafter *link degradation* will denote the difference between the reference and the current value of E_b / N_o.

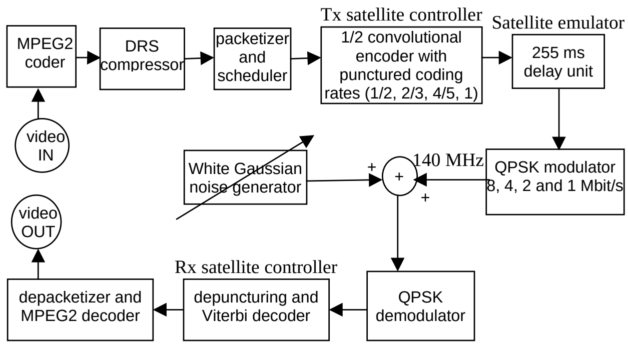

Figure 7: The experiment setup.

The reference value (12 dB), relative to clear sky conditions, allows a BER of 10^{-8}. The signal attenuation due to rain means that fade countermeasures have to be adopted to ensure an acceptable level of link availability with a reasonable channel quality. The fade countermeasure used in our experiment is based on the dynamic combination of the data coding and bit rates variation, a technique which is allowed by the hardware equipment at our disposal. The TDMA controller, burst mode modem, and codec (which constitute an *earth station*) were used in past experiments on the Olympus and Italsat geostationary satellites [5]. For our transmission experiment we used an emulated geostationary satellite channel affected by white Gaussian noise. The satellite emulator introduces the correct round trip time and the amount of noise needed to produce various E_b / N_o ratios, to emulate fading situations. Figure 7 shows the experiment set-up.

The traces generated by the MPEG-2 encoder are rate shaped; subsequently the packet scheduler generates packets at given time instants so as to emulate the real time data processing. The transmission satellite controller applies the channel coding to the base band data stream, and adjusts the packet sending times according to the TDMA channel access scheme. The 1/2 convolutional encoder with the puncturing feature allows 1/2, 2/3 and 4/5 coding rates on the satellite channel. A base band delay unit introduces the satellite round trip time of 255 ms, then the QPSK (quaternary phase shift key) modulation of the 140 MHz intermediate frequency (IF) carrier allows bit rates of 8, 4, 2 and 1 Mbps. The bit and coding rates can be set for individual packets, thus allowing different protections for different receiving paths. Additive white Gaussian noise is introduced at IF level.

All the tests in our experiment were made in steady state conditions, as far as the link characteristics are concerned, that is, the QF of the movie was estimated while maintaining a constant satellite link quality. This procedure prevents biased results due to video scenes having specific characteristics, because the quality factor measured is averaged over a video sequence of a significant length. Table 1 reports the mean QFs and mean bit rates for the three different video coding versions we used. We applied DRS on the enhancement (VBR) part of the scalable2 version only, because the average bandwidth of the enhancement part of the scalable1 version is rather small, so a compression would not give any considerable reduction in the total average bandwidth.

Table 1: QF and mean bit rate of the three video coding simulations.

MPEG 2 video data	Mean QF	Mean bit rate
non-scalable	4.66	2.37 Mbps
scalable1 (base only)	4.36	1.5 Mbps
scalable1(base+enhancement)	4.68	1.5+1.17 Mbps
scalable2 (base only)	4.09	1.066 Mbps
scalable2(base+enhancement)	4.66	1.066+1.482 Mbps

We used the simulation environment previously described to study the quality of the video bit stream corrupted by white Gaussian noise, and we neglected transmission impairments, such as the jitter and the packet loss due to carrier and bit timing acquisition by the modem, which operates in burst mode. We measured the video quality using the QF parameter, and we analysed its relationship with the bandwidth needed by the video bit stream, averaged over the video sample duration, while varying the data coding and bit rates as well as the rate shaping. Hereafter *bandwidth of the video stream* will be used to mean the average share of the link bandwidth which would be used by an uncoded sequence sent at the maximum bit rate on the satellite link, without any DRS. Thus, for example, we will say that an MPEG-2 video stream which has an average information bit rate of 3 Mbps uses a 3 Mbps link bandwidth when it is sent uncoded at the maximum bit rate without DRS, and uses a $2 \cdot 5/4 \cdot 3 \cdot 0.9 = 6.75$ Mbps link bandwidth when it is sent at half the maximum bit rate using a 4/5 coding rate after reducing its information rate by 10% by using DRS.

6 DRS APPLIED TO THE NON-SCALABLE VIDEO STREAM

In figure 8 QF versus DRS is reported for various data sequences in different channel degradation and coding conditions.

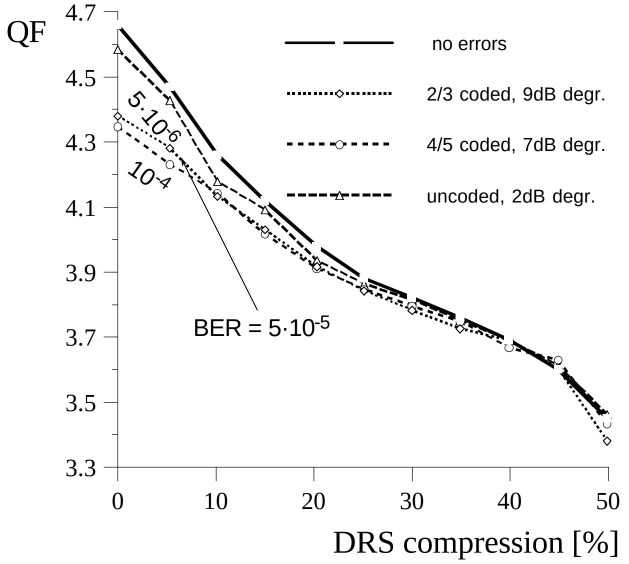

Figure 8: QF versus DRS compression levels for various coding rates and channel degradation conditions. Non-scalable case.

The relevant BERs, ranging from 10^{-4} up to zero (no errors case) are shown as well. The figure highlights that all the curves, whose QF is spread over the interval 4.35-4.66 when data are not compressed (DRS=0), exhibit practically the same QF for DRS from 35% and above. Using high data protection, for a strong BER reduction, when we need a high rate shaping on the video data stream. is thus not very convenient because it is bandwidth wasteful.

The result described above is confirmed if we consider the effect of DRS from an energy point of view, i.e. considering the QF versus link degradation with and without DRS. Figure 9 gives the QF versus the link degradation at different data coding rates, without any DRS. In all cases, we can observe a quasi-constant QF up to a certain degradation, then a knee follows after which QF degrades steeply. A DRS compression applied in the leftmost part of each curve causes a small loss, in terms of signal power, while, if it is applied after the knee, some gains are made. As an example we consider the uncoded curve when no link degradation is present (E_b/N_o = 12 dB, 8Mbps). A 50% of flow compression with DRS is energetically equivalent to 3 dB of channel degradation, because by halving the bandwidth a reduction of 50% of the signal power does not lead to any change in E_b / N_o. Such a compression by means of DRS makes QF fall to 3.45 (see figure 8), while a channel degradation of 3.65 dB is needed to fall to the same value (figure 9), with however a loss of 0.65 dB.

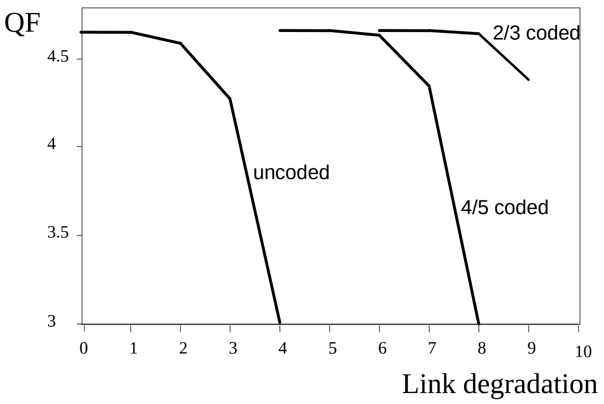

Figure 9: QF versus link degradation for different data coding rates at the bit rate of 8 Mbps. Non-scalable case. No DRS is applied.

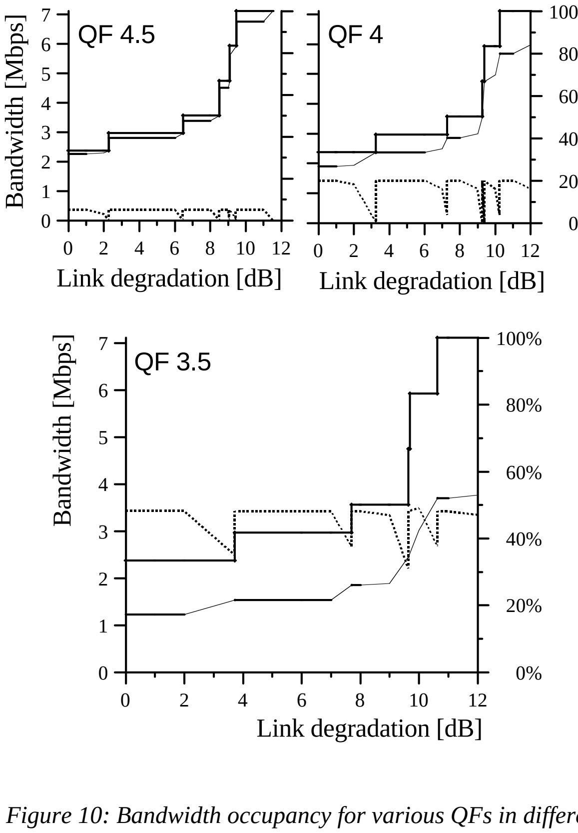

Figure 10: Bandwidth occupancy for various QFs in different fade conditions, with and without (bold lines) DRS. Non-scalable case. The compression percentage is represented by the dotted lines, and the relevant scale is the rightmost one in each figure.

On the other hand, if we start from 2 dB of degradation on the same curve, the compression allows a 1.35 dB gain, and starting from 7 dB of degradation with a flow 4/5 coded, a further 0.7 dB of degradation is all that is needed to make QF go to 3.45, with a gain in the compression case of 2.3 dB. A problem with the method proposed in [1], which does not use any rate shaping of the video stream, is that the number of available bit and coding rates is usually small for a given satellite modem. Even with the ample choice of our modem, there are attenuation regions where a trade-off between video quality and bandwidth occupancy is not possible. In each of the graphs in figure 10 the three bold (upper) lines show the bandwidth occupancy for QFs not lower than 4.5, 4, and 3.5, respectively. For each attenuation value, the curves are obtained by using the combination of coding and bit rates which yields the smallest bandwidth. There is no trade-off possible in the ranges 0 ÷ 2.3 dB and 3.7 ÷ 6.5 dB, for example, because whatever the desired QF is, the channel coding rate to be used is 1/1 in the first range, and 4/5 in the second one, with the result that a higher QF than required is obtained in many cases.

If, for example, we want to closely match a target QF, then it would be more efficient to have a method of varying in continuous mode some parameters in order to get a lower QF, and a corresponding bandwidth gain, for any given satellite link attenuation. By using DRS such a continuous variation can be obtained for very wide bandwidth ranges.

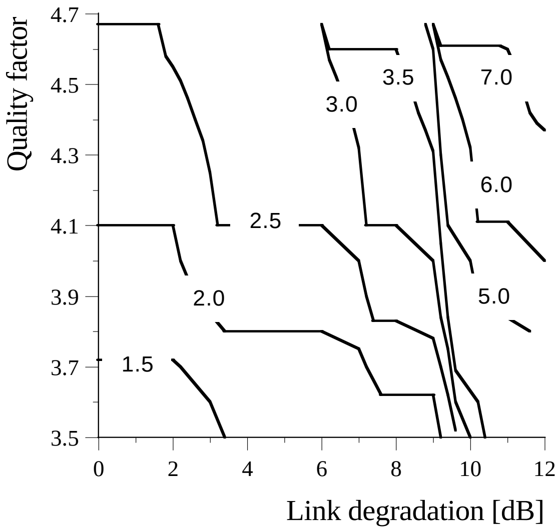

Figure 11: QF versus link degradation for various values of the bandwidth occupied.

Figure 10 highlights the bandwidth gain allowed by using DRS in the range 0 ÷ 12 dB of degradation, for some given values of QF. The gain is small for target QFs close to the maximum (4.5) as the compression is by necessity small, but becomes noticeable for lower QFs. In each of the three pictures, the stepwise curve is obtained without DRS, while the lower bandwidth one is obtained by using the combination of bit rate, coding rate and DRS compression percentage that yields the smallest bandwidth for each degradation level.

Figure 11 shows the best QF that can be obtained by using a constant bandwidth versus link degradation, for values of the bandwidth occupied ranging from 1.5 to 7 Mbps.

Figure 12 shows a more detailed scenario of DRS application for QFs in the range from 3.5 to 4.6. It is now possible to notice that the use of DRS, besides giving a bandwidth saving (figure 10), offers more flexibility as well. In fact, in practically all regions of figure 12 it is possible to trade bandwidth for QF for each link degradation.

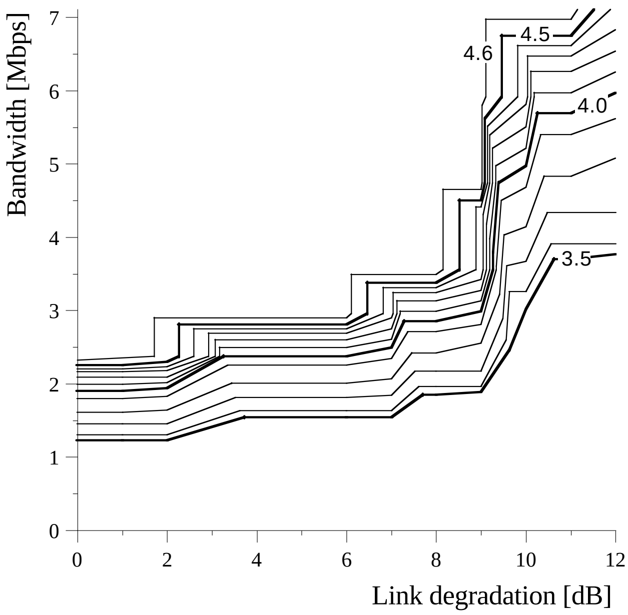

Figure 12: Bandwidth occupied by the video stream versus link degradation for various QFs.

7 DRS APPLIED TO SCALABLE VIDEO STREAMS

We applied the same rate shaping technique used on the non-scalable video stream on the enhanced (VBR) flow of the scalable2 stream (see table 1), in order to investigate the best compromise between bandwidth occupancy and QF achieved in a wider scenario.

Figure 13 represents the QF versus DRS applied to the enhancement flow for different channel codings and degradation conditions. Figure 14 shows the QF for a variety of coding rates versus channel degradation. The performance of the base flow alone is shown as well. The figure highlights once again the convenience of compressing data streams which will be more corrupted by noise when transmitted over a faded channel. For example, observe in figure 13 the curve with no errors and the curve labelled “base2/3+enh.4/5” at 8 dB of degradation, which has BERs of $1.5 \cdot 10^{-6}$ (base flow) + 10^{-3} (enhanced flow). In the uncompressed case, the relative QFs are 4.66 and 4.5, respectively, while the resulting QF is practically the same for the two streams and is close to 4.25 when a 60% compression is applied on the VBR part.

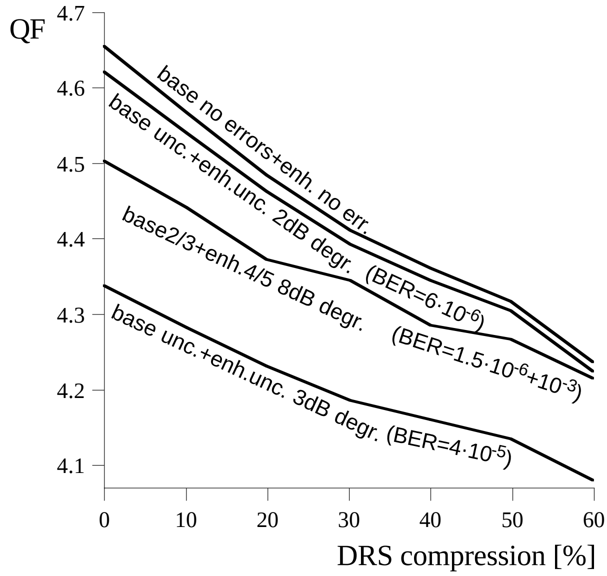

Figure 13: QF versus DRS compression levels for various coding rates and in different conditions of channel degradation. Scalable2 case.

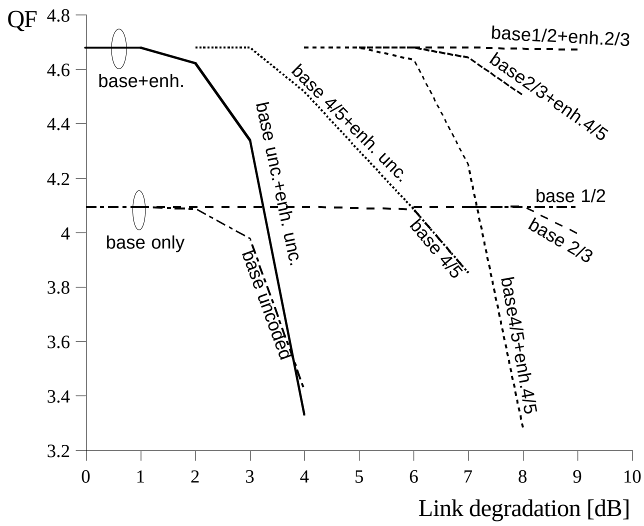

Figure 14: QF versus link degradation for different data coding rates at the bit rate of 8 Mbps. Scalable2 case. No DRS is applied.

In order to see the above noted convenience from an energy point of view, let us consider (figure 13) the base and enhancement flows both uncoded, when the channel degradation passes from 2 to 3 dB. The 2 dB sequence, when the enhancement flow is compressed at 41%, gives the same QF as the 3 dB sequence, but the former has a bandwidth reduced by 1.2 dB, thus only gaining 0.2 dB. However, if we start from a degradation of 3 dB, a further degradation of 0.3 dB is sufficient to make the QF fall to 4.07 (figure 14), a value which is reached with a 60% compression of the enhancement flow with a reduction in bandwidth of 1.86 dB. The resulting gain, in this case, is then 1.56 dB.

8 APPLICATIONS OF THE DRS TECHNIQUE

The problem is now how to assess the convenience of using the above presented techniques for sending an MPEG-2 encoded bit stream over a satellite link, for different levels of link degradation and/or traffic conditions. Note that by using a TDMA technique with variable data coding and bit rates, the time occupancy of a data packet depends on the data redundancy applied. A major link degradation, which imposes a higher data redundancy, and thus a higher time occupancy, requires an increase in resource allocation as if there were an increase in the amount of traffic.

Let us assume that we both dynamically trim the video flow and adapt the data protection (coding and bit rates) in the earth station. This procedure is more flexible than the use of an end-to-end feed-back technique, where the end user evaluates the quality perceived and sends that information back to the source. The source then operates the video flow trimming in view of the suitable data protection which the earth station will operate to guarantee the video quality required. The major drawback of end-to-end feed-back is that the video regulation must be operated at the source (which is generally separated from the satellite earth station). Instead it is much more convenient to play with flow trimming and data protection at the earth station level, where all the information about the overall link quality and the amount of traffic directed towards the satellite is available. Once the video source has been coded, in non-scalable or in scalable mode, and with a given quantization step, the sending earth station first operates any DRS on the VBR part of the flow, then operates the data protection needed, according to each individual link condition. In the scalable case, the protection on the base flow may be different from the one applied on the enhancement flow.

Let us consider the flows in table 1, already studied in [1]. Recall that, depending on the target QF and the channel conditions, in [1] (where DRS was not applied) we showed that the transmission of the base alone of the scalable1 flow was in some cases the most suitable choice. We will see now that this situation may also happen when DRS is applied. Figure 15 shows the required average bandwidth versus the link degradation for various target QFs (4.66, 4.3, 4.0, and 3.5), obtained with and without DRS. These graphs clearly show that using DRS is always advantageous in both non-scalable and scalable2 cases and it allows a bandwidth saving which increases as the QF decreases. If the goal is to keep the QF as close as possible to a certain value, we could conclude that the highest QF (4.66) privileges the non-scalable uncompressed solution. In the QF=4.3 case it is slightly more convenient to send only the base flow of the scalable1 video, but the scalable2 with compression offers almost the same performance, while the non-scalable is definitely worse.

In the QF=4 case the DRS adoption allows a considerable bandwidth saving in the non-scalable case, but the transmission of the base only of the scalable2 sequences turns out to be more advantageous.

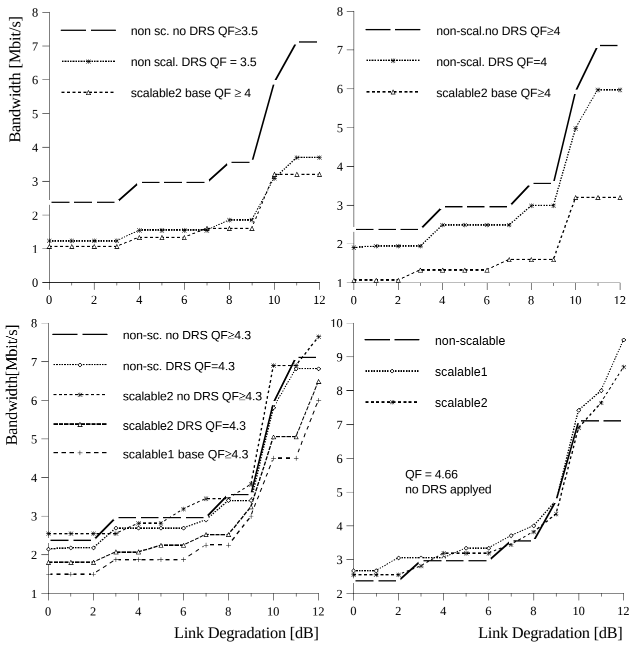

Figure 15: Average bandwidth versus the link degradation for various target QFs obtained with and without DRS compression.

In the low quality (QF=3.5) case the only base flow of the scalable2 video allows the greatest band saving. Here, however, the gain in bandwidth given by the compression of the non-scalable sequence is particularly evident. Note the utility of using the curves which give the QF and the bandwidth needed as functions of the link degradation. This in fact allows us to employ a link quality estimator, which is much cheaper and quicker in response than a QF meter. The QF values shown are averaged over the entire duration of the movie, whereas quasi-instant measurements of QF, as required by dynamic adaptive techniques, may accidentally give spurious values, which are undesirable in a control system. This happens, for example, when some bits in the header fields of a video sequence are corrupted. The link quality estimation, like the BER, is on the other hand much more stable and can be quick and accurate [2].

Several factors, in any case, contribute to the choice of the most suitable technique. Other than the average bandwidth occupancy, we must consider also the bandwidth variability, which is reduced by the DRS application (see figure 4). This factor is important both if a traffic statistic multiplexing is performed on the channel, and if dynamic techniques for demand assignment of the bandwidth are employed. Furthermore we must consider that a satellite link is generally in a non-fade state most of the time, and also that the non-scalable solution requires a less sophisticated coder, and that the DRS technique is easily implementable.

9 CONCLUSIONS

First of all, we must stress that, at least when the rate of corrupted bits is very low, the optimisation of the average satellite bandwidth occupied by the video stream is reached with a non-scalable VBR sequence obtained with a quantization step that is suitable for achieving the target QF. This is only applicable to end-to-end feedback systems, which have the drawbacks we have pointed out. In systems like the one we have presented, where the data protection is chosen at the earth station level for each individual link, significant advantages may be obtained by handling scalable video sequences such as a double flow of data and/or the adoption of DRS techniques. The reduction in bandwidth, obtained by using DRS, may have practical advantages, because it introduces more flexibility in the bandwidth occupation, but it does not produce any actual gain, in the sense that the QF resulting from a DRS application is lower than the one obtained with a quantization step that allows the same bandwidth occupation of the compressed stream. However, the adoption of DRS, used in addition to the channel coding techniques, leads to more flexibility in the way a video data stream can be sent, because it allows the calibration of the bandwidth, in a certain range, in continuous mode. If the goal of using adaptive techniques is to keep the QF as close as possible to a desired value, a certain amount of bandwidth is certainly saved, because the bandwidth versus link degradation curve is smoother than if only the code and bit rate step variations on the flows are adopted, as in [1]. In any case, other criteria can contribute to choose the way to send the video, such as, the maximum instantaneous bandwidth, the maximum average bandwidth, a constant quality, a mix between bandwidth saving and video quality, and simplicity of the system implementation. Furthermore, we have seen that when DRS is used on video sequences, which are subsequently heavily corrupted by noise, the video bandwidth reduction may allow a better signal protection, which gives a better QF as a result.

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