Hybrid Bitrate/PSNR Control for H.264 Video Streaming to Roaming Users

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Abstract

In wireless communications, the available throughput depends on several parameters, like physical layer, base station distance, fading and interference. Users experience changes in bandwidth within a cell and among same-technology cells, but also among different networks. Moreover, in case of video transmission, the user may specify a desired quality level. The source should encode the stream at a quality as close as possible to this value, without exceeding the available bitrate.

We propose a technique to decide whether to encode at constant quality, if resources are enough, or at constant bitrate, if the throughput is not sufficient. With negligible complexity, it proved to obtain better PSNR/bitrate ratios, with respect to only constant bitrate and only constant PSNR coding. We also show this algorithm working in a realistic scenario of a user roaming among heterogeneous networks (WLAN and UMTS). Also in this case, the algorithm proved to achieve high quality/bitrate ratios. [1]

1 Introduction

In present days, the interest in media delivery over data network is constantly increasing, due both to the improvement of bandwidth availability and to the growth of computational capabilities of devices. The access to multimedia contents is particularly challenging for mobile users, because of the wireless channel characteristics. The limited capacity and the high variability of the shared medium impose an accurate choice of scheduling policies, and a fast adaptivity of the coding algorithm to match the channel conditions. Especially in the case of video over wireless, adaptivity should not only care of network dynamics, but also of the signal distortion at the receiver, since the quality degrades noticeably when losses occur.

The available bandwidth depends on the local wireless channel (fading, interference, distance from the base station, etc.) and on the physical layer. For mobile users, the access to the data network can be performed in several ways, starting from reserving a GSM channel, or using the GPRS where available. Third generation cellular networks offer higher data rates but usually cover shorter distances from the base station and are still not widely available, as well as wireless LAN's which usually are accessible around buildings like offices, commercial areas or airports. Satellite communications are also a choice, at high data rates and global coverage, but also at high price and with long latencies.

Within each network, if several cells are present, an handover mechanism is provided to ensure continuous connectivity to moving users. More difficult is to provide continuity between cells implemented with different technology. In this paper, we suppose that this inter-technology handover is possible and immediate. Several approaches to this problem can be found in recent literature, on how to switch among wireless packet data networks [1–4].

The available bandwidth varies within the same cell according to the distance from the center, the number of users, their trajectory and speed, and the traffic they generate. In order to avoid losses, the video encoder should change its internal parameters to produce a stream at a bitrate lower or equal to the available bandwidth. For this work, we suppose the bandwidth to be correctly estimated and immediately transmitted to the coder side.

On the receiver side, the user can specify a level of PSNR desired for the video. The encoder will then decide, according to the feedback coming from both the network and the user, if it is possible to match the user

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requirement given the available bandwidth or not; as a consequence, in the first case it will produce a constant quality stream, in the second a constant bitrate sequence. In this paper, we will focus on this decision algorithm and we will show its capabilities applied to the H.264 encoder.

The paper is organized as follows. In Section 2 we provide some background information, and we describe the proposed approach to the problem in Section 3. Results to prove the effectiveness of the algorithm are presented in Section 4 both for sports video and movie samples. We show one possible scenario for the usage of the algorithm in the case of mobile users roaming among heterogeneous wireless networks in Section 5, along with some throughput estimation tools; results are in Section 6. Finally, we draw the conclusions in Section 7.

2 Background

At constant spatial and temporal resolution, per-frame bitrate and PSNR mainly depend on the quantizer parameter used (QP), the sequence content and the frame type.

To obtain constant bitrate, a rate control technique should be active, to vary the coding parameters and to get a nearly equal number of bits for each GOP. Different approaches are present in literature to bitrate adaptation to the network evolution [5–7]. In this work, we use the approach described in [8].

To encode at constant quality, several techniques have been proposed [9–11] as well. It is possible to adjust QP according to the PSNR obtained for the preceding frames, trying to match a given target; in this work we will use this approach, outlined in [12].

CBR techniques produce a fixed number of bits per GOP, but as a consequence the PSNR is subject to wide oscillations. This is the effect of how the rate control works, rising and lowering QP for each frame within the GOP.

On the other hand, at constant PSNR, the bitrate may vary widely, generating peaks. In this case, network overloading may occur and losses can be generated, so lowering the quality of the received stream.

A compromise between the quality and the rate is usually obtained by means of *rate-distortion optimization* (RDO) [13, 14], in which, for a given rate, coding modes and quantization parameters are chosen in order to maximize the quality. This operation can be computationally demanding.

The approach we propose here is not RDO, but only based on the choice of the quantization parameter to be used for each frame, in order to obtain a constant quality when enough bandwidth is available, or constant bitrate when an excessive amount of bits would be necessary to code at constant PSNR.

3 The mixed PSNR/bitrate control algorithm

Our hybrid control algorithm is built upon two independent components, one for constant bitrate control, the other for constant PSNR. Each one of these, when used alone, is able to commute between different target values; the constant-bitrate is obtained with an error in the order of 5% over the GOP, while the constant PSNR is achieved with a variance around 0.3. In this work, we propose an algorithm to switch among the two coding modes, which we will refer to in the following as *CBR* (constant bitrate) and *CPSNR* (constant quality).

To combine them, it is necessary to build a rule to decide when to code the stream as CBR or CPSNR, and this decision has to be driven not only by the video content, but also by two sequence-independent parameters, namely the available bandwidth and the user's desired quality.

The basic idea of the algorithm is as follows. At each beginning of a GOP, the encoder reads the bandwidth available and requested PSNR, then it computes the starting quantization parameter (QP_{br}) that would be used if coding at constant bitrate, and the one that would be selected if coding at constant quality (QP_{PSNR}). It compares the two parameters and selects the control algorithm as the one related to the highest QP:

$$algorithm = \begin{cases} CBR & \text{if} \quad QP_{br} \geq QP_{PSNR} \\ CPSNR & \text{if} \quad QP_{br} < QP_{PSNR}. \end{cases} \tag{1}$$

This approach could result in poor performance when the two QP's are close up, since the algorithm will not be able to decide exactly which is the best coding mode, being the derivation of the QP built over statistical observations, which are valid *on average*. To avoid this, further control needs to be performed.

At every switch in bitrate or requested PSNR, the algorithm starts from CBR. If the bitrate of the previous GOP results to be higher than the available bandwidth, then the algorithm switches automatically to constant bitrate to avoid coding several GOP's exceeding the maximum allowed rate, so avoiding losses. The algorithm will continue coding with this mode until a GOP is detected to have an average PSNR higher than the target quality, in which case it switches again to CPSNR.

Summarizing, the algorithm rules for coding mode switching results:

- change in br_T or $PSNR_T \Rightarrow algorithm = CBR$;
- for all the other cases:
  - select $algorithm$ according to rule (1)
  - if $\begin{cases} algorithm_{last} & = & CBR \\ PSNR_{last} & \geq & PSNR_T + 0.1 \end{cases}$
    $\Rightarrow\ algorithm = CPSNR$;
  - if $\begin{cases} algorithm_{last} & = & CPSNR \\ br_{last} & \geq & 1.05 * br_T \end{cases}$
    $\Rightarrow\ algorithm = CBR$;

It is also shown as a two-state diagram in Figure 1.

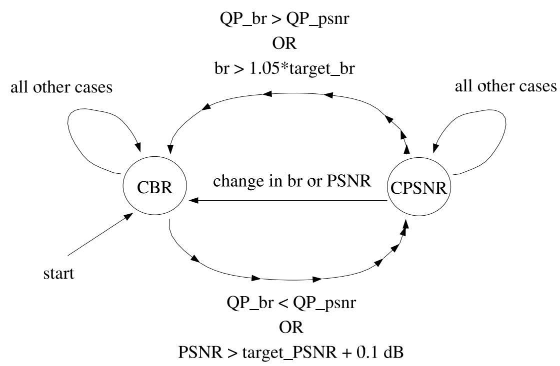

Figure 1. State diagram of the proposed algorithm.

4 Results in a test environment

To test the behavior of the above algorithm, we encoded two 90-second sequences, containing parts of a soccer match and of a movie.

For simplicity, the two sequences will be encoded using a constant target PSNR value; the available bitrate is variable and switches at the same time boundaries for both sequences, but the bitrate levels are different.

Both videos are QCIF format, the sport video is coded at 25 fps, while the movie is at 30 fps.

4.1 Sport videos

In sport videos, and especially in the team sports, the presence of many far-away scenes makes it possible to obtain low bitrates and high quality at the same time, at least for the majority of the frames. On the other hand, on close up views, which usually contain high motion of players, lower compression ratios are achieved if coding at constant quality, or lower and widely variable quality is obtained at fixed bitrate.

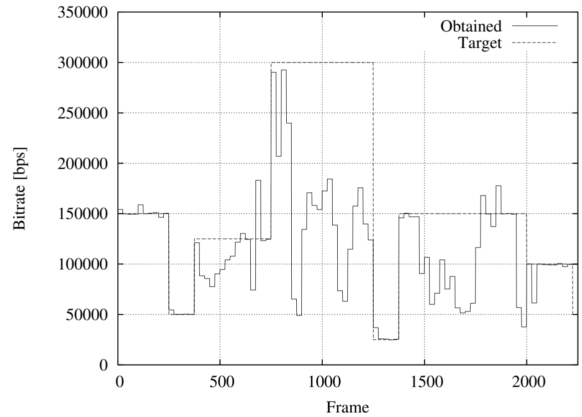

Figure 2. Bitrate obtained with the proposed algorithm and available bitrate, sport video.

Figure 2 shows the bitrate obtained with our approach, plotted together with the available bitrate pattern. In that plot, it is possible to notice the behavior of the switching algorithm. Each first GOP after a new region starts is encoded as CBR, then according to the outcome in PSNR of this first coding, the following GOP's either continue coding as CBR if it was not possible to reach the target PSNR, or the system will switch to CPSNR if better quality than the target was achieved.

The first two regions (frames from 1 to 400) are an example of regions coded entirely as CBR. Regions in which the algorithm works at CPSNR are the one around frame 500 and the one containing frame 1000. Finally, in the region from frame 1400 to 2000, the algorithm starts CBR for some GOPs, then switches to CPSNR and then again to CBR. In this region it is possible to notice that, whenever a bitrate higher than the target is obtained, the coding mode immediately switches to CBR.

In Figure 3, the PSNR obtained for the same setting as before is presented. Also here, regions with different behaviors can be individuated. These regions exactly correspond to the different runs of frames described for Figure 2. From frame 1 to 400, coding was CBR and the quality drops below the requested PSNR. Oscillations are also wide in this region, as well as in all the other GOPs coded at constant bitrate, like the last one or the ones between frames 1250 and 1500.

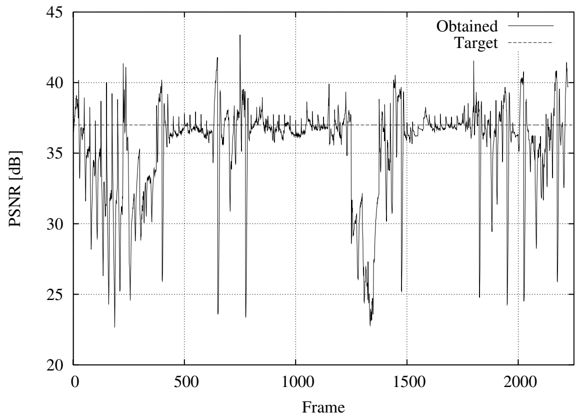

Figure 3. PSNR obtained with the proposed algorithm, sport video.

Table 1. Performance of the proposed algorithm, sport video.

Parameter	Control algorithm		
	Bitrate	PSNR	Mixed
% packets in excess	1.36	26.15	3.50
PSNR/kbps w/o losses	0.230	0.261	0.309
PSNR/kbps w. losses	0.221	0.222	0.300
Average PSNR w/o losses	37.01	36.90	35.43
Average PSNR w. losses	35.43	31.50	34.43
Overall bitrate [kbps]	161	142	115

On the other hand, regions coded at the requested PSNR show narrow oscillations and correspond to the portions of the graph in Figure 2 where the obtained bitrate is lower than the target.

To summarize what shown above, we present coding results for only-CBR and only-PSNR coding, together with our mixed approach for this sequence, in Table 1.

We present the results as follows. A first parameter is the percentage of packets in excess with respect to the available bitrate; this corresponds to the GOPs whose bitrate is higher than the dashed line in Figure 2. Those packets in excess can be lost in the network due to overload. This percentage is very low for CBR coding, and extremely high for CPSNR since it does not take care of the used bandwidth; for our hybrid approach, it is only slightly higher than the CBR case. It means that following this rule, the stream will experience few losses.

The second parameter is the PSNR units per kbps used. This gives an indication of how effectively the bandwidth was used to produce quality. According to this parameter, CPSNR works better than CBR; the proposed approach gains 0.4 dBs/kbps with respect to constant quality. Moreover, if we consider lost the packets in excess of bandwidth, this parameter is nearly equal for CBR and CPSNR, because of the high number of losses of the latter, which decreases in quality. In this situation, the proposed algorithm does not experience a drop in PSNR because of the small percentage of packets out of bandwidth. To perform this computation, we supposed that the lost packets were the ones which introduce the lowest distortion in the decoded stream.

Average PSNR is also presented, both considering and not considering the losses. The hybrid approach results to behave better in the case with losses. Last parameter, the average bitrate used is noticeably lower for the proposed algorithm, with a reduction from 20 to 25%.

Table 2. Performance of the proposed algorithm, movie.

Parameter	Control algorithm		
	Bitrate	PSNR	Mixed
% packets in excess	2.36	24.05	1.74
PSNR/kbps w/o losses	0.170	0.224	0.226
PSNR/kbps w. losses	0.154	0.203	0.219
Average PSNR w/o losses	38.36	34.90	34.97
Average PSNR w. losses	34.64	31.68	33.85
Overall bitrate [kbps]	225	156	155

4.2 Movies

The same observations made in Section 4.1 also apply to movies. The main difference in this case is given by the different coding efficiency of the compression algorithm.

The statistics for this sequence are shown in Table 2. The percentage of packets which exceed the per-GOP bandwidth is small for the mixed approach; the PSNR/kbps is higher than the CBR and CPSNR methods, both considering and not considering the losses. The drop in overall PSNR if the exceeding packets are lost is in the order of 1 dB for the proposed technique, compared to the more than 3 dBs of the other approaches. Finally, the bitrate obtained is the same as in the case of CPSNR, and nearly 30% lower than in the case of CBR.

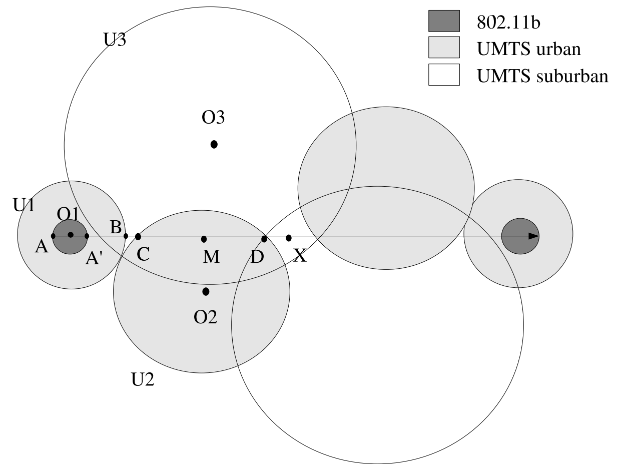

Figure 4. Path of the user and disposition of 802.11b and UMTS cells.

Table 3. Length of segments and speed of the user, for the first half of the path; in the second part, values are specular.

Segment	Length [m]	Speed [km/h]	Time [s]
A-A'	200	6	120.0
A'-B	400	30	48.0
B-C	240	60	14.4
C-D	1420	30	170.4
D-X	240	60	14.4
Total A-X	2500		344.4

5 Operative scenario: mobile user in heterogeneous networks

To better show the effects of the proposed algorithm, we will describe its behavior under a realistic available throughput pattern. To derive this, we suppose a user moving along the arrow of Figure 4.

The user moves at variable speed and passes through different cells; in particular, the travel starts from an 802.11b area and moves to different UMTS cells along the path, including urban and rural cells. The total travel length is 5 km.

Cells are intended as circular (omnidirectional antennas are supposed); the radius of the WLAN cells is set to 100 m, while the dimension of the UMTS cells varies according to the supposed density of population. Smaller UMTS cells have a radius of 500 m, medium sized ones have radius 1 km, while the widest are set to 2 km.

The different speeds, along with the length of the segments shown in Figure 4 are reported in Table 3. That table reports only the first half of the path, since the remaining part is specular both in segment length and user's speed.

The analysis of exact methods to estimate the available bandwidth in different wireless networks is not covered in this paper. In the following two subsections we will describe the simplified approaches we used to evaluate the available throughput in the IEEE 802.11b wireless LAN and ETSI UMTS (IMT-2000) standards. They are reported for completeness and to show how the throughput pattern was produced.

Complete studies can be found in [15] for WLAN and [16, 17] for UMTS.

5.1 Evaluation of available bitrate in WLAN

We suppose our WiFi spots use DCF. In the simple scenario where bit error rate is zero and no buffer overflow occurs, the time used for a packet transmission can be expressed as:

$$t(N, R) = t_{tr}(R) + t_{oh} + t_{cont}(N) \tag{2}$$

where t_{tr} is the frame transmission time, t_{oh} is the constant overhead and $t_{cont}(N)$ is the time spent in contention. R indicates the rate, N is the number of active stations within the network.

The constant overhead is defined as:

$$t_{oh} = DIFS + t_{pr} + SIFS + t_{pr} + t_{ack} \tag{3}$$

and is composed by the preamble header transmission time t_{pr}, the MAC acknowledgement transmission time t_{ack}, the two quantities $DIFS = 50\mu s$ and $SIFS = 10\mu s$. t_{pr} is showed twice because the header is sent twice. The transmission time t_{tr} can be computed as ratio between the size of an IP packet s_{MPDU} and the rate R. The contention time can be calculated as:

$$t_{cont}(N) = t_{slot} * \frac{1 + P_{err}(N)}{2N} * \frac{CW}{2} \tag{4}$$

where $t_{slot} = 20\mu s$ is the slot time, CW is the congestion window size which can vary from $CW_{min} = 31$ to $CW_{max} = 1023$, and P_{err} is the collision probability, defined as:

$$P_{err}(N) = 1 - \left(1 - \frac{1}{CW}\right)^{N-1}. \tag{5}$$

The access bitrate becomes:

$$r_{access} = \frac{s_{MSDU}}{t(N, R)} \tag{6}$$

where s_{MSDU} is the size of a MAC-level frame plus the CRC.

The time a station occupies the channel depends on the number of users N sharing the medium and the packet error rate P_{err}:

$$U = \frac{1}{N(1 + P_{err}(N))} \tag{7}$$

Then, the average throughput per station is:

$$T_{put} = U * r_{access} \tag{8}$$

In our scenario we suppose 19 stations fixed at 20 m from the Access Point, while our user is moving. In addition, we suppose the *user equipment* (UE) selects a rate $R = 11Mbps$ when its distance from the AP is less than $1/3$ of the cell radius, $R = 5.5Mbps$ when it is between $1/3$ and $2/3$, and $R = 2Mbps$ in the remaining part; this simplified behavior is introduced to emulate the reaction to higher bit error rates which can occur when the distance from the AP increases.

t_{pr} is equal to 192 ms for R=2, otherwise it is equal to 96 ms; t_{ack} is 10 ms for R=11, otherwise it is 20 ms.

5.2 Evaluation of available throughput in UMTS cells for our scenario

To estimate the throughput for our user, we follow a simple approach. At the receiver the carrier to interference ratio is:

$$\frac{C}{I} = \frac{P_r}{\alpha I_{intra} + I_{inter} + P_N} \tag{9}$$

where P_N is the thermal noise assumed equal to -99 dBm, I_{inter} is the sum of signal powers received from other cells, I_{intra} is the sum of the signal power received from other user within the same cell, and α is the loss of orthogonality factor. Table 4 shows the values for α and I_{intra} used in our scenario, where I_{inter} is considered on average equal for all points in a single cell. Only the first three cells are shown, the other ones can be obtained by symmetry.

The normalized energy per information bit is obtained as:

$$\frac{E_b}{N_0} = \frac{1}{2R_c} * SF * \frac{C}{I} \tag{10}$$

where R_c is the code rate. Now, using the Shannon theorem we can calculate the maximum information ($R_{c,max}$) that the channel is able to transport, in a second, to the receiver.

Table 4. α and I_{intra} in our scenario.

Cell	α	I_{intra} [dB]
U1	0.06	-66
U2	0.40	-200
U3	0.40	-120

Table 5. Parameters for different UMTS cells.

Cell	TTI [ms]	SF	R_c
U1	10	256	3/4
U2	20	64	1/2
U3	10	128	3/4

$$R_{c,max} = B * log_2 \left(1 + \frac{S}{N}\right) \tag{11}$$

where $B = 5\ MHz$, in an UMTS channel. The throughput for the user can be calculated as:

$$T_{put} = \frac{S_{IP}}{\frac{S_{RLC}}{R_{c,max}} + TTI} \tag{12}$$

where S_{IP} is the size of an IP packet and S_{RLC} is the size of a RLC (Radio Link Control) PDU.

In our scenario the DCH for data in downlink is used, RLC PDUs are of fixed size and able to contain a PDCP (Packet Data Convergence Protocol) PDU without fragmentation. Values of TTI and R_c for each cell are shown in Table 5, along with the spreading factor SF; again, only the first three cells are shown, the ones on the second part are specularly equal.

Finally, we can calculate the power received at the UE using the link budget as follow:

$$P_r = P_t G_r G_t L \tag{13}$$

Where P_t is the power transmitted, G_t and G_r are the antenna gains at transmitter and receiver respectively, and L is the path loss. The transmitted power is 200 W for U1, 300 for U2 and 400 for U3.

For all UMTS cell we assume a path loss L expressed as:

$$10log(L) = -(128.1 + 37.6log(d)) \tag{14}$$

where d is the distance between the UE and the base station.

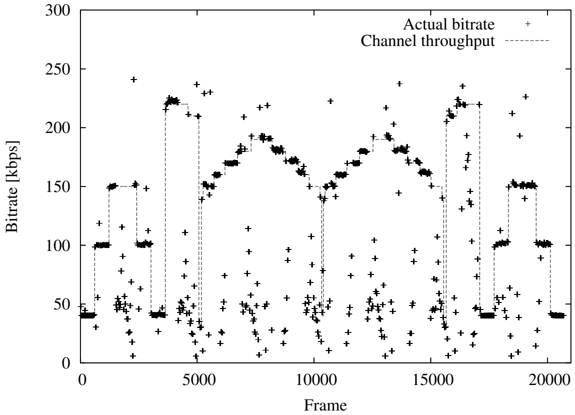

Figure 5. Actual bitrate and available throughput; the user requests a quality of 36 dB.

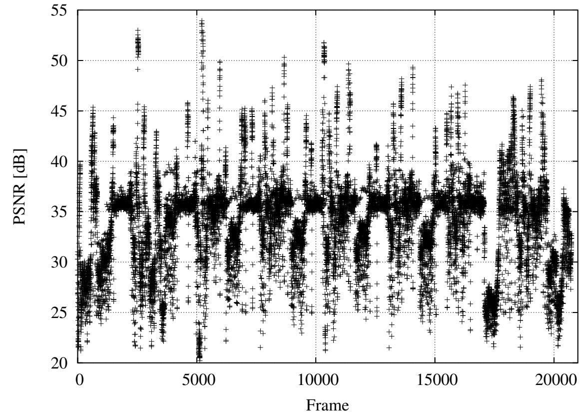

Figure 6. Obtained PSNR; the target is 36 dBs.

6 Results for heterogeneous networks

The approach presented in Section 5 has been used to estimate the available bandwidth for the roaming user along the path shown in Figure 4.

The user device prefers to use the WLAN connection whenever available, even in case of lower throughput with respect to the UMTS link, because of the lower cost. The user will then remain connected to the different cells for the times shown in Table 3.

The user is requesting the streaming of a movie during the travel; the choice of a movie is made because this kind of video contains several changes in motion content. Moreover, the user assigns a level of quality he desires for the video, and expresses this value as the coding PSNR. Two experiments have been carried out, using two different requested PSNR levels.

In the first experiment, the user requests a quality of 36 dBs. Using the coding approach described in Section 3, the source encodes the video CPSNR if there is enough bandwidth, or CBR in other cases. The coding bitrate profile is shown in Figure 5, along with the maximum bitrate pattern.

Points (one for each GOP) on the available throughput line indicate that GOP was coded as CBR, while points far from the line are at CPSNR. The number of points which exceed the available bitrate is limited and they are only isolated cases, since after the detection of a bitrate higher than the availability, the algorithm switches automatically to CBR. On the other hand, the number of points below the line is high and they represent GOP's where it was possible to achieve the desired quality without using all of the available bandwidth.

The behavior of the per-frame PSNR along the path is shown in Figure 6. The higher concentration of PSNR values is around the desired value of 36 dBs, except for regions coded at CBR, where the PSNR is lower.

Table 6. Coding results for a target quality of 36 dB.

Parameter	Maximum	Actual	Difference
Bitrate [kbps]	148.4	111.5	-24.86%
PSNR [dB]	36	33.97	-2.03 dB
PSNR/kbps	0.305 dB/kbps		

To better show the obtained results, Table 6 reports some statistics on the coded sequence.

The average bitrate for the sequence is nearly 25% lower than the maximum allowed average bitrate (the one defined by the dashed line of Figure 5); under the quality point of view, despite the high number of GOP's coded as CBR, the average PSNR is only 2 dB lower than the desired one. In this case, the ratio between the average PSNR and the average bitrate is in the order of 0.3, comparable with the results of Table 1.

A PSNR level of 36 dB represents a very high quality, which turns to require high bitrate; for this reason, several times the available throughput is not enough to guarantee a constant quality.

If we repeat the above experiment with a desired PSNR value of 33 dB, we get the statistics shown in Table 7. The bitrate is 40% lower than the maximum, and a quality loss of only 0.62 dB on average with respect to the desired PSNR is present. Moreover, the ratio quality/bitrate is relatively high, in the order of 0.36 dB/Kbps.

Table 7. Coding results for a target quality of 33 dB.

Parameter	Maximum	Actual	Difference
Bitrate [kbps]	148.4	89.1	-39.96%
PSNR [dB]	33	32.38	-0.62 dB
PSNR/kbps	0.363 dB/kbps		

7 Conclusions

In this paper, we presented an hybrid approach to video coding control. This technique takes as inputs the available bitrate of the channel and the user desired quality, and based on these parameters and video content, decides to code the current GOP using a constant quality approach, if the available bandwidth is enough, or to constant bitrate if the network resources are scarce. Video content is taken into account by means of observation of the algorithm behavior in the near past.

We described the algorithm as a two-state graph, and showed that it can code at noticeably lower bitrate with respect to a pure-CBR approach, and that it can achieve a quality level close to the one indicated by the user. This behavior is reflected in the high ratio between quality and bitrate. Experiments were performed both for sport videos and movies.

We also computed the available bitrate in a realistic scenario, with a user moving along a path and passing through different cells and different networks, Wireless LAN and UMTS. We used the available bitrate pattern obtained through estimation to encode a movie under two different user PSNR requirements, and showed the performance of the algorithm and in particular the high quality/bitrate ratio.

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