

# DELAY-RATE-DISTORTION OPTIMIZED RATE CONTROL FOR WIRELESS VIDEO COMMUNICATION

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## ABSTRACT

In this paper, we extend the traditional rate-distortion optimization (RDO) for the end-to-end wireless video communication system and develop a novel delay-rate-distortion optimization (dRDO) based rate control (RC) algorithm, by investigating the allocation of end-to-end delay to different delay components. Tradeoffs regarding, respectively, source coding delay versus buffering delay and available source coding rate versus redundant rate incurred by channel coding, are coupled in the proposed dRDO algorithm. It targets at minimizing the average total end-to-end distortion under the transmission rate and end-to-end delay constraints, by joint selection of search range and quantization step size in source coding and channel code rate in channel coding. Experimental results demonstrate the superiority of the proposed dRDO algorithm over existing schemes.

*Index Terms*— Wireless video, delay-rate-distortion optimization, rate control, end-to-end distortion.

## 1. INTRODUCTION

An end-to-end wireless video communication system is conventionally designed based on the rate-distortion optimization (RDO), which aims at minimizing the end-to-end distortion subject to the transmission rate constraint, by jointly choosing both source coding and channel coding parameters [1] - [4]. Based on the statistical analysis of error propagation, error concealment, and channel decoding, these RDO based rate control schemes developed a theoretical framework to estimate the source coding bit-rate and the end-to-end distortion. However, they neglect the impact of the end-to-end delay constraint on the overall quality of service (QoS) performance. In order to achieve the optimal end-to-end QoS performance, the entire cross-layer wireless video communication system is expected to appropriately assign different delay components according to the total end-to-end delay bound. Specifically, for a practical real-time wireless video communication system, the end-to-end delay  $\Delta T$  experienced by each frame is composed of several delay components which, as illustrated in Fig. 1 respectively, are video encoding delay  $d_e$ , encoder buffer delay  $d_{eb}$ , channel transmission delay  $d_c$  (including channel coding, transmission, and channel decoding delay), decoder buffer delay  $d_{db}$ , and video decoding delay  $d_d$  [5] [6].

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For a given end-to-end delay bound, if the encoding time is increased to achieve better compression performance with higher bit rate, the allowed queuing delay at encoder and decoder buffers will decrease accordingly, which in turn reduces the delay constrained transmission throughput and increases the error incurred transmission distortion. Therefore, the overall system performance depends on the allocation of end-to-end delay to different delay components, and change of delay assignment in one component would affect the delay budget in other components, thereby impacting the overall system performance.

Some works have been done to extend the traditional R-D model for the video encoder by incorporating the statistical analysis of encoding delay and power [6] [7]. In these works, the analytic framework for delay-power-rate-distortion (d-P-R-D) modeling of the video encoder was proposed, with four dimensions (delay, power, rate and distortion) derived as functions of source coding parameters. The proposed d-P-R-D model can be applied to formulate the source rate control problem for the video encoder, i.e., minimizing the source coding distortion under the constraints of encoding delay, rate and power. However, such model is only investigated for the video encoder, and fails to achieve the optimal end-to-end QoS performance for the end-to-end wireless video communication system, which requires not only the allocation of encoding delay and source coding rate, but also the suitable assignment in other delay components and channel code rate according to the channel capacity. For example, from the perspective of the video encoder, larger source coding rate and longer encoding delay are preferred to achieve minimum source coding distortion. By doing so, however, more video packets are likely to be dropped during transmission with a larger transmission distortion incurred due to constrained channel capacity. In this case, the total end-to-end distortion including both source coding and transmission distortion may not be minimum.

To tackle the above issues of end-to-end wireless video communication systems, in this paper, we develop a novel delay-rate-distortion optimization (dRDO) based rate control (RC) algorithm, which targets at minimizing the average total end-to-end distortion under the transmission rate and end-to-end delay constraints, by joint selection of search range and quantization step size in source coding and channel code rate in channel coding. Tradeoffs with regard to source coding delay versus buffering delay and available source coding rate versus redundant rate incurred by channel coding, are also coupled in the proposed dRDO algorithm.

## 2. SYSTEM MODEL

### 2.1. End-to-end Distortion

The end-to-end distortion between the original video frame and the respective reconstructed frame at the decoder is commonly adopted as the performance measure for wireless video communication [1] [4]. Under the assumption that quantization error and transmission error are uncorrelated, the end-to-end distortion for the  $k$ -th frame is

$$D_{ete}^k = D_e^k + D_t^k \quad (1)$$

where  $D_e^k$  denotes the source coding distortion caused by quantization error during lossy video compression, and  $D_t^k$  is the transmission distortion caused by transmission error due to bandwidth fluctuation and packet losses.

According to [6] [7], the source coding distortion for IPPPP coding mode is given by

$$D_e^k = \frac{\Lambda^k Q^k e^{\gamma \Lambda^k Q^k} (2 + \Lambda^k Q^k - 2\gamma \Lambda^k Q^k) + 2 - 2e^{\Lambda^k Q^k}}{(\Lambda^k)^2 (1 - e^{\Lambda^k Q^k})} \quad (2)$$

where  $\Lambda^k = \sqrt{2}/\sigma^k$  is the Laplace parameter;  $\sigma^k$  denotes the standard deviation of the transformed residuals in motion estimation (ME); and  $\gamma Q^k$  represents the rounding offset and  $\gamma$  is a parameter between (0, 1), such as 1/6 for H.264/AVC inter-frame coding [8]. For a specific video sequence,  $\sigma^k$  can be well fitted by a closed form function of the search range  $\lambda^k$  in ME and the quantization step size  $Q^k$  [7], as

$$\sigma^k(\lambda^k, Q^k) = ae^{-b\lambda^k} + c + dQ^k \quad (3)$$

where  $a$ ,  $b$ ,  $c$  and  $d$  are fitting parameters dependent on the encoded video sequence as well as on the encoding structure. Therefore, integrating  $\Lambda^k = \sqrt{2}/\sigma^k$  and Eq. (3) into Eq. (2), the source coding distortion of the  $k$ -th frame can be further expressed as a function of  $\lambda^k$  and  $Q^k$ , i.e.,  $D_e^k(\lambda^k, Q^k)$ .

As derived in [9] [10], for single-reference motion estimation and no slice data partitioning, the transmission distortion of frame  $k$  is

$$D_t^k = \bar{P}^k (E[(\varepsilon^k)^2] + \rho^k E[(\xi^k)^2] + D_t^{k-1}) + (1 - \bar{P}^k) \alpha^k D_t^{k-1} \quad (4)$$

where  $\bar{P}^k$  is the average packet error probability (PEP) of all packets within the  $k$ -th frame;  $\varepsilon^k$  and  $\xi^k$  denote the residual concealment error and motion vector concealment error, respectively; the propagation factor  $\alpha^k$  and the correlation ratio  $\rho^k$  are system parameters that depend on the video content, channel condition and codec structure; and  $D_t^{k-1}$  is the transmission distortion of frame  $k - 1$ . According to [4], the transmission distortion is basically a function of PEP, while the other video frame statistics and system parameters could be simply estimated as discussed in [10].

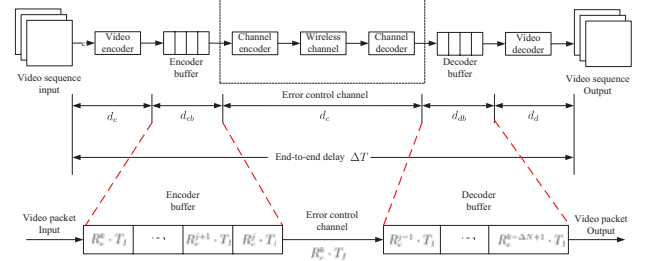


Fig. 1. Delay components of a video communication system.

### 2.2. End-to-end Delay Components and Constraints

Considering the constant video frame rate that is the same at both the encoder and decoder, the end-to-end delay per frame is required to be less than a maximum acceptable delay interval  $\Delta T_{max}$ , which is referred to as the end-to-end delay constraint [5].

$$\Delta T = d_e + d_{eb} + d_c + d_{db} + d_d \leq \Delta T_{max} \quad (5)$$

In other words, every single frame captured and entered the video encoder at time  $t$  has to be decoded and available for display before time  $t + \Delta T_{max}$ . Video packets arriving at the decoder too late to be decoded before their scheduled display time and thus violating the maximum end-to-end delay bound are useless and considered lost.

For a point-to-point wireless channel that connects the video encoding base station and the end user, the variation of channel transmission delay is relatively small. Thus it is reasonable to assume  $d_c$  to be constant. Furthermore, in [5] and [11], the video encoding time  $d_e$  and decoding time  $d_d$  are also both assumed to be constant. In fact, the video decoding time (delay) is much shorter than the encoding time (delay), and can be considered as a part of the video encoding time since the encoder has to decode the video sequence as well. Therefore, the video decoding delay is negligible compared to the encoding delay, and  $d_e + d_d$  can be approximated by  $d_e$ . As illustrated in [6] [12], however, both the distortion and bit rate of the compressed video that is transmitted over the channel is controlled by the video encoding time. On the other hand, given an end-to-end delay constraint, if the encoding time is increased to achieve better compression performance with higher bit rate, the allowed queuing delay at encoder and decoder buffers will decrease accordingly. In this case, more packets are likely to be dropped due to delay bound violation, which in turn reduces the delay constrained transmission throughput and thus increases the transmission distortion of the video.

Based on the above assumption and discussion, in this work, we focus on the assignment and tradeoff between the encoding delay  $d_e$  and the buffer delay  $d_{buffer}$  which is defined as the sum of both encoder and decoder buffer delay. Specifically, since  $d_c$  is assumed to be constant and  $d_d$  is neglected compared to  $d_e$ , the end-to-end delay constraint is reformulated as

$$d_e + d_{buffer} \leq \Delta T_{max} - d_c \quad (6)$$

Let  $T_f$  be the time duration of a frame interval. Then, the number of video frames stored in either the encoder or the decoder buffer is

$$\Delta N = \left\lfloor \frac{d_{buffer}}{T_f} \right\rfloor = \left\lfloor \frac{d_{eb} + d_{db}}{T_f} \right\rfloor \quad (7)$$

In addition, to ensure real time communication without introducing accumulated encoding delay for each frame, the maximum video encoding time at the encoder should not exceed the time duration of a frame interval, i.e.,

$$d_e \leq T_f \quad (8)$$

For the single-reference prediction case where only one reference frame is used for motion estimation of the current frame [6] [7], the encoding delay of the  $k$ -th frame can be expressed as a function of  $\lambda^k$  and  $Q^k$ , as follows

$$d^k(\lambda^k, Q^k) = \frac{N(2\lambda^k + 1)^2 \cdot v(Q^k) \cdot c_0}{f_{CLK}} \quad (9)$$

where  $N$  is the number of MBs in a frame;  $(2\lambda^k + 1)^2 \cdot v(Q^k)$  is the total number of SAD operations in the two dimensional search area for each MB and  $v(Q^k)$  denotes the ratio of the actual number of SAD operations in the JM codec to the theoretical total number of SAD operations;  $c_0$  is the number of clock cycles of one SAD operation over a given CPU;  $f_{CLK}$  is the constant clock frequency of the CPU.

### 2.3. Source Coding Rate Constraints

In accordance with the time-varying characteristics of wireless channels, we assume that time is slotted  $t \in \{1, 2, 3, \dots\}$  with each slot corresponding to each frame interval, and channels hold their states within the duration of a time slot [13]. To avoid packet drops incurred by delay bound violation, [5] has shown that for a given encoding delay  $d_e$ , the end-to-end delay constraint can be translated into several constraints on both the encoding rate and the transmission rate. Assume that at time  $t = k$ , frame  $k$  is the video frame currently being encoded, while the packets of frame  $j$  are being transmitted by the error control channel. Let  $R_e^i$  be the encoded source bit rate of the  $i$ -th frame,  $r$  represent the code rate of channel code, and  $R_c^i$  denote the maximum bit rate that is supported by the wireless channel capacity at time  $i$ . As shown in Fig. 1, at time  $t = k$  the encoder buffer contains the packets from frame  $j, j + 1, \dots, k$ , respectively. Therefore, the condition for these packets to arrive at the decoder before the due time is given by

$$\begin{aligned} R_e^j \cdot T_f &\leq \sum_{i=k}^{j+\Delta N-1} r \cdot R_c^i \cdot T_f \quad (10) \\ R_e^{j+1} \cdot T_f + R_e^j \cdot T_f &\leq \sum_{i=k}^{j+\Delta N} r \cdot R_c^i \cdot T_f \\ &\vdots \\ R_e^k \cdot T_f + \dots + R_e^{j+1} \cdot T_f + R_e^j \cdot T_f &\leq \sum_{i=k}^{k+\Delta N-1} r \cdot R_c^i \cdot T_f \end{aligned}$$

As derived in [6] [7], the source coding rate of the  $k$ -th frame is

$$R_e^k = -P_0 \log_2 P_0 + (1 - P_0) \left[ \frac{\Lambda^k Q^k \log_2 e}{1 - e^{-\Lambda^k Q^k}} - \log_2(1 - e^{-\Lambda^k Q^k}) - \Lambda^k Q^k \gamma \log_2 e + 1 \right] \quad (11)$$

where  $P_0 = 1 - e^{-\Lambda^k Q^k(1-\gamma)}$  is the probability of quantized transform coefficient being zero. The source coding rate can also be expressed as a function of  $\lambda^k$  and  $Q^k$ , i.e.,  $R_e^k(\lambda^k, Q^k)$ .

### 2.4. Error Control Channel

Through introducing forward error correction (FEC) into video communication systems, the reliability of transmission is improved with smaller PEP and thus lower transmission distortion. In order to enhance the error correction capability, however, redundant protection information is also introduced. As shown in the rate constraints (10), to maintain transmission data rate  $R_c$ , with channel coding the maximum achievable source data rate for video encoder has to be reduced to  $R_e \leq rR_c$ , where  $r$  is the channel code rate between the range of  $[0, 1]$ . For a given channel data rate  $R_c$ , the code rate  $r$  controls the bit rate allocation between source and channel coding, and thus the tradeoff between source coding and transmission incurred distortion. Specifically, a smaller  $r$  is preferred for the purpose of reducing the PEP as well as the transmission distortion. By lowering  $r$ , however, the available source coding rate is also decreased, which will result in higher source coding distortion. Since the optimization objective is minimizing the total distortion of both source coding and transmission distortion, it is required to find the best tradeoff between these two types of distortion by choosing a proper code rate  $r$ .

In this paper, we assume that  $(n, m)$  Reed-Solomon (RS) block code is used for FEC with a block of  $m$  information symbols and  $n - m$  parity symbols [1]. Here, we use the common choice of 8 bits per symbol, and thus one symbol corresponds to one byte. The code rate is therefore  $r = m/n$ , and the maximum block length is  $n_{max} = 2^8 - 1 = 255$ . In the proposed video communication system, we set the number of symbols within each encoded video packet less than  $n_{max}$ , and thus each video packet can be considered as the information symbols and be encoded into one coded block by RS coding. In this way, a video packet after source coding corresponds to a RS coded block in channel transmission. With  $(n, m)$  RS code, any received block with symbol errors less than  $t_c = \lfloor (n - m)/2 \rfloor$  can be successfully recovered. Thus, the probability of a block (video packet) being unable to correct is given by

$$P = \sum_{\kappa=t_c+1}^n P_b(n, \kappa) \quad (12)$$

where  $P_b(n, \kappa)$  is the probability that  $\kappa$  symbol errors occur within a block of  $n$  consecutively transmitted symbols. Further denoting  $P_s$  as the symbol error probability of the wireless channel,  $P_b(n, \kappa)$  is therefore given by

$$P_b(n, \kappa) = \binom{n}{\kappa} P_s^\kappa (1 - P_s)^{n-\kappa} \quad (13)$$

### 3. PROBLEM FORMULATION

Without loss of generality, a GOP level rate control problem will be proposed in this paper. Specifically, suppose that a GOP with IPPPP coding structure has one I-frame and  $K$  P-frames. The first I-frame within each GOP is indexed as frame 0, and assumed to be encoded with fixed coding parameters. Therefore, the delay-Rate-Distortion optimization based rate control problem can be formulated as

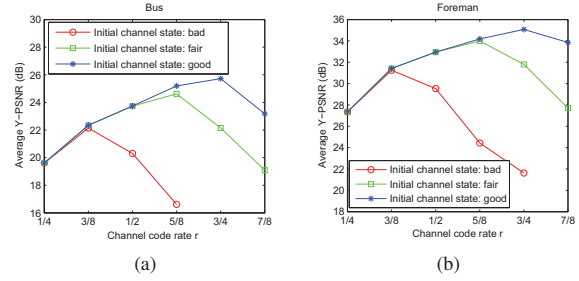
$$\begin{aligned} \min_{\lambda, Q, r} \quad & \frac{1}{K} \sum_{k=1}^K \left[ D_e^k(\lambda^k, Q^k) + D_t^k(\lambda^k, Q^k, r) \right] \quad (14) \\ \text{s.t.} \quad & d_e^k(\lambda^k, Q^k) + \Delta NT_f \leq \Delta T_{max} - d_c, \forall k = 1, 2, \dots, K \\ & \sum_{i=1}^k R_e^i(\lambda^i, Q^i) \leq \sum_{i=1}^{k+\Delta N-1} r R_c^i, \forall k = 1, 2, \dots, K \\ & d_e^k(\lambda^k, Q^k) \leq T_f, \forall k = 1, 2, \dots, K \end{aligned}$$

Two kinds of tradeoffs, regarding respectively source coding delay versus buffering delay and available source coding rate versus redundant rate incurred by channel coding, are coupled in the optimization problem (14) that targets at minimizing the average total end-to-end distortion. Theoretically, a larger search range  $\lambda^k$  as well as a smaller quantization step size  $Q^k$  is required to result in smaller source coding distortion with larger source coding bit rate. However, the source coding delay is also enlarged while the number of frames stored in the encoder and decoder buffers  $\Delta N$  is decreased, which in turn limits the available source coding bit rate supported by the transmission channel and thus affects the source coding distortion. Furthermore, from the perspective of channel coding and transmission, by lowering  $r$  the PEP as well as the transmission distortion can be reduced, while a larger  $r$  is preferred for source coding to promise sufficient source coding rate and thus smaller source coding distortion. Compared to the existing works on rate control problem in a wireless video communication system, the proposed optimization problem in Eq. (14) is constrained by one more condition of the end-to-end delay, in addition to the rate. To solve it, the Lagrange multiplier method and sequential quadratic programming method based algorithm similar to [7] can be applied, which is omitted here due to page limitation.

### 4. EXPERIMENTAL RESULTS

We implement the proposed algorithm in JM18.2 [14] codec, with test video sequences *Bus* (QCIF), *Foreman* (CIF), and *Mobile* (CIF), the IPPPP GOP structure, CABAC entropy coding, the maximum search range 16 of motion estimation, the dynamic range 0 to 51 of quantization parameter, and one reference frame. The wireless channel is modeled as a finite-state Markov channel with the same channel parameters as in [15]. The channel code rate can be selected from the discrete set  $\{1/4, 3/8, 1/2, 5/8, 3/4, 7/8\}$ .

Fig. 2 illustrates the impact of different channel coding rate on the average end-to-end distortion (measured in Y-PSNR) for *Bus* and *Foreman* video sequences, under different setup of the initial channel state. It can be seen that for a given initial channel state, there



**Fig. 2.** Impact of different channel code rate  $r$ , where (a)  $\Delta T_{max} - d_c = 0.6$  s,  $R_c = 100$  kbps, and  $T_f = 0.2$  s for *Bus* sequence, and (b)  $\Delta T_{max} - d_c = 1$  s,  $R_c = 100$  kbps, and  $T_f = 0.3$  s for *Foreman* sequence.

**Table 1.** Comparison of average Y-PSNR (dB) under different average signal-to-noise ratio (SNR) and channel capacity for *Mobile* sequence.

SNR (dB)	$R_c$ (kbps)	PA	JM			dPRD		
			$r^*$	$r=1/4$	$3/8$	$1/2$	$1/4$	$3/8$
2		$r^*$						
	100	20.3	18.5	19.8	18.3	19.0	19.8	18.0
	200	23.0	22.3	21.9	19.0	21.4	21.4	18.9
	500	26.0	24.7	24.3	20.6	24.9	24.3	20.3
5		$r^*$						
			$r=3/8$	$1/2$	$5/8$	$3/8$	$1/2$	$5/8$
	100	21.8	20.4	21.3	20.3	20.9	21.4	19.9
	200	24.2	23.0	23.8	21.5	22.6	23.7	21.2
10		$r^*$						
			$r=5/8$	$3/4$	$7/8$	$5/8$	$3/4$	$7/8$
	100	23.3	22.6	23.0	17.8	22.1	22.5	17.4
	200	25.8	24.7	25.4	18.2	24.9	25.3	17.9
	500	32.6	30.1	31.5	19.0	30.3	31.9	18.9

is an optimal channel code rate value  $r^*$  corresponding to the maximum average Y-PSNR. Though the PEP and the transmission distortion can be reduced when  $r < r^*$ , the available source coding rate is limited and thus the source coding distortion is increased, which will cause a larger end-to-end distortion. On the other hand, if we let  $r > r^*$  to promise sufficient source coding rate and thus smaller source coding distortion, the PEP as well as the transmission incurred distortion will be increased, which will also lead to a larger end-to-end distortion. Furthermore, when the initial channel state becomes worse, the optimal  $r^*$  value will accordingly decrease to introduce more parity symbols for forward error correction.

In Table 1, performance of the proposed algorithm (PA) is compared with two baseline schemes: 1) JM18.2 rate control algorithm [14]; and 2) dPRD optimization based rate control algorithm [7]. For PA, the optimal channel code rate  $r^*$  is solved by Eq. (14). Since the other two schemes can not determine such optimal value, we set  $r$  to three values around  $r^*$ . It can be seen that the average Y-PSNR of PA achieved by the optimal  $r^*$  is higher than those of the other two schemes under different selection of  $r$ .

### 5. CONCLUSION

We developed a dRD optimized rate control algorithm to minimize the end-to-end distortion subject to the transmission rate and delay constraints. Tradeoffs regarding source coding delay versus buffering delay and available source coding rate versus redundant rate incurred by channel coding were coupled in the proposed algorithm.

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