Low-latency video transmission over high-speed WPANs based on low-power compression

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***Abstract*—This paper disscuses latency-constrained video transmission over high-speed wireless personal area networks. Low-power single-layer video compression is proposed as an alternative to others video processing approaches. End-to-end distortion and end-to end latency in video transmission system are analyzed. A near-optimal video source rate control based on MINMAX quality criteria is introduced. Practical results for video encoder based on H.264/AVC standard are also given.**

I. Introduction

In the last few years a number of high throughput wireless personal area networks (WPANs) have appeared, such as the IEEE 802.15.3c standard [1]. These networks have low power data transmitters and throughput up to 6 Gigabit per second. Therefore, they can provide a transmission of high definition video from mobile device to display over wireless channel instead wired cable. These networks are lacking of the best method for the choice of video processing so this task still remains actual.

To choose the best video processing approach for such type of networks the following restrictions and requirements should be taken into account. To provide low power consumption at mobile transmitter one-pass low-complexity and low-memory approaches without motion compensation or temporal filtering (intra processing only) have to be used. At the same time these solutions should provide very low transmission latency, continuous video playback at the receiver and acceptable visual quality for all variety of video sources: sequences of computer graphics, snapshots, natural and mixed images.

From our point of view there are four potential approaches that can satisfy these restrictions and requirements:

- uncompressed video transmission;
- intra single-layer video compression;
- intra scalable video coding;
- distributive video coding.

Let us consider the list stated above. The straightforward solution in these networks can be based on uncompressed video transmission [1]. This approach does not require any compression algorithm and provides low processing latency. On the other hand, several disadvantages arise. Firstly, the throughput of a wireless channel is time-varying where, in addition, other traffic such as audio or IP data can be transmitted along with video data. Therefore, it could not be guaranteed that the channel rate is high enough for continuous video playback. Secondly, this approach does not use the channel in an efficient way in the sense of throughput and energy consumption, because it does not take into account the video source redundancy. In the third place, this solution requires technical change at the network layers like combination of video data unequal error protection and special automatic repeat request methods [1]. Fourthly, the pixel partitioning technique which is used for unequal error protection is not efficient in rate-distortion sense especially for desktop snapshot type of the video that contains a lot of details commensurable with pixel size.

The second solution can be based on Scalable Video Coding (SVC) and unequal error protection of different quality layers [2]. In this case latency-constrained video transmission over variable wireless channels is achieved due to dropping the higher enhancement layers of the scalable video [3]. However, known SVC algorithms have higher computation complexity and lower compression efficiency than single-layer video compression [4].

The third solution can be based on Distributive Video Coding (DVC) [5] which became very popular in the last few years. Practical results show that DVC can provide compression efficiency better than single-layer video compression in intra-mode [6]. Many papers present DVC as very low-power approach for video compression. But at this time there are no DVC implementations which verify it. In addition DVC encoder has to contain two compression cores: Winer-Ziv encoder and traditional intra-encoder, and has very high power consumption on the decoder side that is not efficient for consumer products.

And final solution can be based on single-layer video compression which is most extended in video processing devices. This approach has low encoder and decoder complexity, it has good compression efficiency in rate-distortion sense and do not require any changes at link and physical network layers.

The particular properties of the video processing methods are stated in Table I. Complex comparison shows that intra single-layer video compression is more preferable for video data transmission over high-speed wireless personal area networks.

Note, that transmission system based on solutions, which were described above, has a set of parameters (like quantization step, macroblock type, transport packet length, modulation and code scheme and so on) which have to change in real time depending on video source and wireless channel

Table I
VIDEO PROCESSING APPROACHES COMPARISON

Video processing approach	Encoder / Decoder complexity	Compr. efficiency	Is network layers modification needed?
Uncompressed video transmission	very low / very low	very low	yes
Intra single-layer video compression	low / low	medium	no
Intra scalable video coding	medium / medium	low	partly
Distributive video coding	low / high	high	no

states. In common case, algorithms, which are controlling these parameters, should minimize video distortion based on objective or subjective quality criteria taking into account power consumption and transmission latency restrictions. In common case it is very difficult task and an open problem at the current time.

This work is a continuation of our research in this way. In papers [11], [12] we propose latency-constrained video source rate control algorithm based on MINMAX quality criteria for transmission of video data over constant throughput channels. In this paper we extend our approach taking into account power consumption restrictions and variable throughput channel and adopt it to intra single-layer video compression based on H.264/AVC standard [7]. We introduce several assumptions, formulate optimization task and propose corresponding one-pass video source rate control algorithm.

This paper is organized as follows. Section II describes low-power implementation of H.264/AVC Encoder. In Sections III–IV end-to-end distortion and end-to-end latency in video compression and transmission systems are discussed. In Section V video source rate control based on MINMAX quality criteria is introduced. Finally, the practical results for different test video sequences are shown.

II. LOW-POWER VIDEO COMPRESSION BASED ON H.264/AVC STANDARD

H.264/AVC compression standard [7] is based on exploiting the spatial and temporal redundancy of video sources. This is achieved by using motion estimation and compensation, intra-frame prediction, discrete cosine transform, quantization, entropy coding and others methods.

To achieve low-power compression, low computation complexity and memory consumption is needed. To decrease the memory size it is proposed to eliminate motion estimation and to use intra-coding only. In this case the encoder can be implemented by using internal memory which is needed to store 32 pixel lines of the input video only. For example, for the resolution size of 1920×1080 180 Kbytes is needed instead of more than 6 Mbytes in the motion compensation case.

In addition to the proposed scheme, even for this small memory size the simple case of the temporal redundancy removal could be used. It is often the case that many regions in the current and the previous frames are the same in computer graphics and desktop snapshots (static regions). Therefore, it is possible to detect "static" macroblocks at the encoder side by calculating hash function value. If hash function value for the current macroblock is equal to the corresponding hash function value for the previous frame, it can be encoded in SKIP mode and the decoder shows the corresponding macroblock which was transmitted earlier.

The further decrease of the computational complexity can be achieved by using DC intra-prediction, 4×4 DCT and CAVLC (Context-adaptive variable-length coding) compression modes only. For improving the encoding performance and achieving absolutely RGB-lossless compression it is proposed to use reversible YCoCg 4:4:4 color space transform.

III. END-TO-END DISTORTION IN TRANSMISSION SYSTEM

Let us assume that each video frame is separated into non-overlapping *units* that include several macroblocks. The end-to-end distortion d_t for unit t in wireless video communication systems consists of two main components [13]:

$$d_t = d(q_t) + d_c, \tag{1}$$

where $d(q_t)$ is distortion caused by quantization at the encoder side and d_c is distortion caused by channel errors and error concealment algorithm at the decoder side. In this paper we describe video transmission based on MINMAX quality criteria [14] that can be interpreted as follows. For each unit t the distortion d_t should be provided, so that

$$\textit{minimize} \max_t d_t. \tag{2}$$

Usually the automatic repeat request (ARQ) method is used to achieve reliable data transmission over an unreliable channel. For each packet the receiver sends to transmitter special message (acknowledgement) that indicates if the packet is received correctly or not. If the packet is not received correctly then the transmitter sends it again. The probability of this situation can be defined as $p_t = 1-(1-p_b)^l$, where p_b is bit error rate (BER) and l is transport packet length assuming independent bit errors. If the packet is not received after n retransmissions then it is dropped at the transmitter side with the probability p_t^n and the decoder shows the corresponding co-located macroblocks which were transmitted earlier.

Assume that *channel rate controller* chooses modulation and coding scheme (MCS) and transport packet size to maximize the throughput depending on channel feedback. In this case the transmission scheme that provides the BER $< 10^{-4}$ is chosen. Then the optimal transport packet size for this BER values can be chosen [16] to guarantee that packet loss probability is $p_t^n < 10^{-10}$. Therefore, for further optimization we can disregard packet losses. It means that for end-to-end distortion minimization it is enough to minimize quantization distortion $d(q_t)$ only.

IV. END-TO-END LATENCY IN TRANSMISSION SYSTEM

A. Video transmission system description

The video transmission system model discussed in this paper is shown in the Figure 1. Consider the system timing

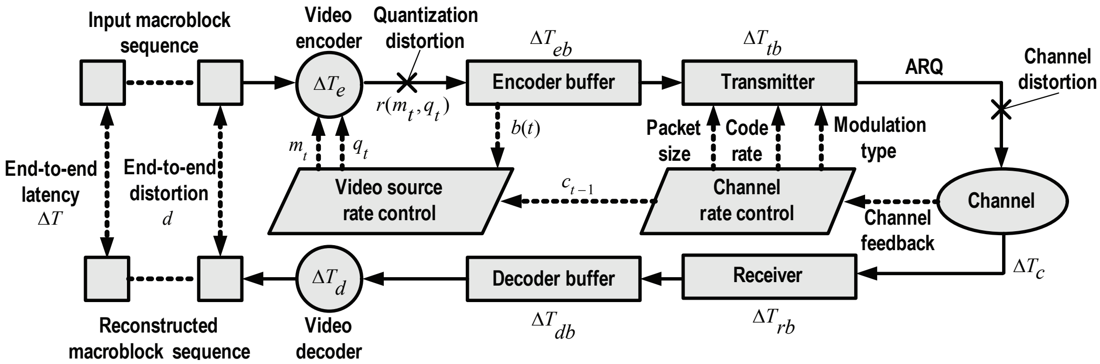

Figure 1. End-to-end distortion and end-to-end latency in video transmission system

is discrete and slotted. The slot time is a part of the system time $[t, t+1)$ and time moment t refers to the end of this slot. *Channel rate controller* chooses the transmission scheme that maximizes the channel throughput. Taking into account transport packet headers, ARQ and time division between different types of traffic, let us define channel throughput for video data c_t as the number of bits that are transmitted during time slot t. The video source gives encoder a *unit* that contains M macroblocks of the encoded frame. After compressing unit t into $r(q_t)$ bits, where q_t is a quantization step, encoder places it into the *encoder buffer*. Depending on the number of bits in the encoder buffer and channel state, *video source rate controller* chooses the quantization step q_t and macroblocks type $m_t \in \{\textit{intra}, \textit{skip}\}$ for the next unit.

The number of bits in the encoder buffer $b^e(t)$ after placing there a new compressed unit t and transmitting over the channel with the throughput c_t, changes as follows:

$$b^e(t) = \max\{0, b^e(t-1) - c_t\} + r_t(q_t, m_t). \qquad (3)$$

Data on the receiver side is accumulated for some time L after which the decoding and playing starts.

B. Latency definition, necessary and sufficient conditions

Generally, latency ΔT between the time moment when some unit has been sent to the encoder and the time moment when this unit has been shown at the receiver device display consists of the following components:

$$\Delta T = \Delta T_e + \Delta T_{eb} + \Delta T_{tb} + \Delta T_c + \Delta T_{rb} + \Delta T_{db} + \Delta T_d, \qquad (4)$$

where ΔT_e and ΔT_d are the encoding and decoding processing latency, ΔT_{eb} is the encoder buffer latency, ΔT_{tb} is transmitter buffer latency, ΔT_{rb} is receiver buffer latency, ΔT_{db} is the decoder buffer latency, ΔT_c is the channel transmission latency.

Let us suppose that the encoder and the decoder work real-time and values ΔT_e, ΔT_d and ΔT_{tb}, ΔT_{rb} are significantly less than L. In [8] it was shown that

$$\Delta T_{eb} + \Delta T_c + \Delta T_{db} = L, \qquad (5)$$

if the number of bits in the encoder buffer is

$$b^e(t) \le b_{eff}(t) = \sum_{i=t+1}^{t+L\cdot f\cdot N} c_i, \qquad (6)$$

where $b_{eff}(t)$ is the *effective buffer size* [15], N is a number of units in the frame and f is a frame rate.

For time-varying wireless channel $b_{eff}(t)$ is equal to the sum of the future channel rates in time interval $[t, ..., t+L]$ and it can not be calculated at the time moment t, because future channel rates are not known yet. Therefore, effective buffer size is usually estimated at the encoder side by using channel model [9]. However, for time-varying wireless channel it is not possible to guarantee that estimated value $\hat{b}_{eff}(t) \le b_{eff}(t)$ for any time moment t. Then the situation when $b^e(t) > b_{eff}(t)$ is possible and latency requirements (5) do not hold.

C. Required latency restoration approach

If required L value is low (e.g. 1ms) then we can use the following approach to restore the required latency. At the time moment $t+L$ we can calculate effective buffer size $b_{eff}(t)$. If at the time moment $t+L$ inequality (6) does not hold then at the time moment t latency requirements (5) do not hold. It means that at the time moment $t+L$ number of bits in decoder buffer is $b^d(t+L) = 0$ and decoder can not start the reconstruction process.

Assume that decoder works in real-time, therefore decoding time for INTRA unit is less than $\Delta T_d^{intra} \le \frac{1}{f\cdot N}$ and decoding time for SKIP unit is close to zero $\Delta T_d^{skip} \approx 0$. Then, to restore equation (5) we propose the following algorithm.

Step 1. Compress all units in SKIP mode until at the time moment t^* encoder buffer will be emptied $b^e(t^*) = 0$.

Step 2. Compress all units in SKIP mode at the time interval $[t^*, ..., t^* + 2\cdot L + \frac{1}{f\cdot N}]$.

At the time moment $t+L$ encoder buffer contains not more than $n = 2\cdot L\cdot f\cdot N + 1$ units. At the time moment t^* all these units will be available at the decoder side together with SKIP units that were formed in the time interval $[t+L, t^*]$. To decompress it decoder spends time

$$\Delta T_d(n) \le (2\cdot L\cdot f\cdot N + 1)\cdot \Delta T_d^{intra} \le 2\cdot L + \Delta T_d^{intra}. \qquad (7)$$

It means that at the time moment $t^* + 2 \cdot L + \frac{1}{f \cdot N}$ encoder buffer will contain SKIP units and decoder buffer will be empty. This event is equivalent to the system starting state.

V. Video source rate control algorithm

A. MINMAX optimization task description

Note that for high-speed video transmission we can use high-resolution quantization hypothesis [10] that defines distortion as $d(q) = q^2/12$, therefore MINMAX criteria (2) corresponds to

$$\textit{minimize} \max_t q_t. \tag{8}$$

Suppose that despite statistical properties of the units in the frame may be quite different from each other, statistical properties of all frames vary insignificantly. It means that there is only one scene in the input video sequence. This assumption does not hold true generally, because video sequence usually consists of subsequences (scenes) with different statistical properties. To make understanding of the algorithm with several scenes easier, let us initially take into account the case when video sequence has only one scene.

Let us formulate rate control optimization task according to the latency requirements (6) and the MINMAX quality criteria (8). For each unit t it is necessary to choose the quantization step q_t, so that

$$\begin{cases} minimize \max_t q_t \\ b(t) \le b_{eff}(t). \end{cases} \tag{9}$$

B. Solution of MINMAX task by consecutive search algorithm

Solution of the task (9) can be found by the following hypothetic algorithm which consists of the following two steps:

Step 0. (*Initialization*)
0.1 Set $\{q_i\} = \{0, 1, ..., q_{max}\}$, $i \leftarrow 0$.
0.2 Go to Step 1.1

Step 1.
1.1 $\tilde{q} \leftarrow q_i$, $\tilde{b}(0) \leftarrow 0$.
1.2 For units $t = 0, 1, ...$ calculate
$\tilde{b}(t) \leftarrow \max\{0, \tilde{b}(t-1) - c_t\} + r_t(\tilde{q})$.
If $\tilde{b}(t) > b_{eff}(t)$ then $i \leftarrow i + 1$ and go to Step 1.1

The algorithm described above is called the *consecutive search algorithm.*

Theorem 1. *Consider $\tilde{q}$ the solution found by the consecutive search algorithm. There is no sequence of quantization steps $y_1, y_2, \ldots$ for which $\max_t y_t < \tilde{q}$ that does not lead to effective buffer size exceeding.*

Proof. Suppose that *consecutive search algorithm* has stopped at the step i. Then for each step $j < i$ for every unit t quantization step $x_t = q_j$ was chosen. From consecutive search algorithm description follows that after encoding unit τ number of bits in encoder buffer $\tilde{b}(\tau) > b_{eff}(\tau)$.

Let us choose any sequence of quantization steps $y_1, y_2, \ldots$, where $y_t \le q_j$, and $b(t)$ is the number of bits in encoder buffer, when unit t is encoded with y_t value. Then $y_t < x_t$, consequently,

$$r(y_t) \ge r(x_t) \tag{10}$$

So if $\tilde{b}(0) = b(0) = b_0$, then from (3) and (10) follows that $\tilde{b}(t) \le b(t)$. It means that exists such $\tau' \le \tau$ that

$$b(\tau') > b_{eff}(\tau'). \blacksquare \tag{11}$$

C. Single-scene MINMAX rate control algorithm

Consecutive search algorithm is a hypothetic one that shows the solution of (9), but can not be implemented in a real-time system, because it is impossible to rerun data transmission after effective buffer size exceeding.

Therefore, this paper proposes an algorithm that allows to find the estimation of $\tilde{q}$ for the consecutive search algorithm. Consider $\hat{q}_t$ to be the estimation of $\tilde{q}$ value. It is supposed to estimate $\tilde{q}$ value as follows. All units are compressed with quantization step $\hat{q}_t$ until the number of bits in the buffer $b^e(t)$ will not exceed effective buffer size $b_{eff}(t)$. This exceeding means that it is impossible to hold the $\hat{q}_t$ value for the given channel throughput for fixed end-to-end latency without increasing it. So, the end-to-end latency exceeds its initial value L and, consequently, the required latency restoration approach is used and the estimation of $\hat{q}_t$ increases. The algorithm consists of the following three steps.

Step 0. (*Initialization*)
0.1 Set $\hat{q}_0 \leftarrow q_0$, $t \leftarrow 0$, $b^e(0) \leftarrow 0$.
0.2 Go to Step 1.1

Step 1. (*Buffer accumulation*)
1.1 $t \leftarrow t + 1$, $\hat{q}_t \leftarrow \hat{q}_{t-1}$.
1.2 $b^e(t) \leftarrow \max\{0, b^e(t-1) - c_t\}$.
1.3 Compress unit t with quantization step $\hat{q}_t$.
1.4 If $b^e(t) > b_{eff}(t)$ go to Step 2.1
1.5 $b^e(t) \leftarrow b^e(t) + r(\hat{q}_t)$ and go to Step 1.1

Step 2. (*Latency restoration*)
2.1 Compress all units in SKIP mode until at the time moment t^* encoder buffer size $b^e(t^*) = 0$.
2.2 Compress all units in SKIP mode at the time interval $[t^*, ..., t^* + 2 \cdot L \cdot f \cdot N + 1]$.
2.3 $t \leftarrow t^* + 2 \cdot L \cdot f \cdot N + 1$, $\hat{q}_t \leftarrow \hat{q}_t + \Delta q^+$ and go to Step 1.1

Theorem 2. *Consider that consecutive search algorithm finds the quantization step value $\tilde{q}$. Then for the proposed algorithm with initial value $\hat{q}_0 \le \tilde{q}$, the inequality $\hat{q}_t < \tilde{q} + \Delta q^+$ holds true for any time moment t.*

Proof. Let $\tilde{b}(t)$ be the buffer size for the consecutive search algorithm. From its description

$$\tilde{b}(t) \le b_{eff}(t). \tag{12}$$

Let us suppose that $\hat{q}_0 \le \tilde{q}$ and at the time moment τ this inequality holds true firstly:

$$\tilde{q} \le \hat{q}_\tau < \tilde{q} + \Delta q^+. \tag{13}$$

So at this moment the number of bits in the encoder buffer (see Step 2.1) is:

$$b^e(\tau) = 0. \tag{14}$$

From (13) for $t \geq \tau$ the following inequality holds true:

$$r(\hat{q}_t) \leq r_t(\tilde{q}), \quad (15)$$

so that from (3), (12), (14) and (15) follows that at the time moment $t \geq \tau$ the number of bits in the buffer is:

$$b^e(t) \leq \tilde{b}(t) \leq b_{eff}(t). \quad (16)$$

Thereby, from the time moment τ the statement of Step 1.4 of this algorithm fails. Consequently, the algorithm will not reach Step 2.3 and parameter $\hat{q}_t$ will not be increased.■

From Theorem 2 follows that after algorithm adaptation for the source and channel properties the quantization step for each unit will not exceed the quantization step for the solution of the task (9) and the value not more than Δq^+. The time of the adaptation depends on the source and channel properties, and starting value q_0 and Δq^+. With Δq^+ increase the time of the adaptation decrease, and vice versa.

D. Scene change and virtual buffer concept

Now let us take a look at the video sequences that consist of several scenes $s_0, s_1, \ldots, s_n$. Then MINMAX optimization task (9) should be applied for each scene. Let $\tilde{q}(s_i)$ be a solution provided by consecutive search for scene s_i. If $\tilde{q}(s_{i+1}) \geq \tilde{q}(s_i)$, then algorithm proposed above will adapt to a new scene. However, if $\tilde{q}(s_{i+1}) < \tilde{q}(s_i)$, then algorithm will not decrease $\hat{q}_t$, that means that the quality will not be improved.

Therefore, to overcome this problem we introduce an heuristic approach based on a *virtual buffer* concept. For each unit t the following value is calculated:

$$b_v^e(t) \leftarrow \begin{cases} b^e(t), \ if\ t = t^*, \\ \max\{0, b_v^e(t-1) - c_v(t)\} + r(\hat{q}_t - \Delta q^-), \\ if\ t \neq t^*, \end{cases} \quad (17)$$

where t^* is a number of the first unit in the current frame, $c_v(t)$ is a virtual channel rate that is calculated as follows:

$$c_v(t) = \frac{b_{eff}^{min}(w,t)}{L \cdot f \cdot N}, \quad (18)$$

where $b_{eff}^{min}(w,t)$ is a minimum of the effective buffer size for the previous w frames

$$b_{eff}^{min}(w,t) = \min_i b_{eff}(i), i \in \{t^* - w \cdot N, \ldots, t^* - 1\}. \quad (19)$$

In addition, the difference between the number of bits that is placed into the buffer and maximum number of bits that could be transmitted is accumulated:

$$\Delta r_v \leftarrow \sum_{i=t^*}^{t^*+N-1} r(\hat{q}_t - \Delta q^-) - c_v(t). \quad (20)$$

Let us take a look on the virtual buffer concept. If $\Delta r_v > 0$, the number of bits sent to the transmission buffer is more then the number of bits sent to the channel and this can lead to the effect of latency exceeding the limit during the transmission of the next frames. On the other side, the bit size distribution for units in each frame may be so, that this can happen even if $\Delta r_v \leq 0$. Therefore, in addition $b_v^e(t)$ is calculated. Thus, if before the encoding of the unit t^* the following statements are fulfilled:

$$\begin{cases} \max_i b_v^e(i) \leq b_{eff}^{min}(w, t^* - 1), \ i \in \{t^* - N, \ldots, t^* - 1\}, \\ \Delta r_v \leq 0, \end{cases} \quad (21)$$

and rate control was not in the latency restoration mode during coding of previous frame, the quantization step value is modified as follows:

$$\hat{q}_t \leftarrow \max\{0, \hat{q}_t - \Delta q^-\}. \quad (22)$$

E. Using static units detection in rate control

For improving video quality for the low channel throughput case, static units can be transmitted repeatedly in lossless mode. However, we have to take into account that the types of the units (static or non-static) in the future are unknown. Therefore we should keep a significant part of the encoder buffer free for non-static units. Thus, if rate control works in buffer accumulation mode then lossless mode is used for static units, if they were not transmitted as lossless earlier and number of bits in encoder buffer $b^e(t) \leq \alpha \cdot b_{eff}(t)$, $\alpha \in [0, ..., 1]$.

VI. Practical results

To obtain practical results the suggested rate control algorithm was embedded into the low-power H.264/AVC encoder that was shortly described in Section II. In the rate control algorithm the following parameters were used: $L = 1$ ms, $M = 2$, $w = 5$, $\Delta q^+ = 3$, $\Delta q^- = 2$, $\alpha = 0.15$.

Channel throughput c_t simulation is executed as following. At first, propagation measurements in the presence of human activity for a 60 GHz channel [17] were used for obtaining of the temporal variations of the channel SNR(t). Secondly, SNR(t) vs. p_t dependencies for each MCS were calculated based on transport packet length $l = 4092$, number of retransmissions $n = 10$ and SNR/BER curves from IEEE 802.15.3c standard proposals documents [18]. Finally, for each SNR(t) value one of the MCS was chosen that provide BER$< 10^{-4}$.

The performance of the discussed algorithm was tested on two video sequences with 1920×1080 frame resolution, frame rate $f = 60$. The first test video sequence ("Breeze") is a typical movie which contains natural images. The second test video sequence ("Desktop") corresponds to computer desktop snapshots: running office applications and dragging windows. Figure 2 shows video source rate and peak signal-to-noise ratio (PSNR) for the given channel throughput. For the convenience of graphic expression PSNR= 70 dB corresponds to the absolutely lossless compression.

Practical results show that in good channel condition case the low-power encoder provides lossless video source rate equal to 1.5 Gbps for natural video sequences that allows to economize channel throughput or use it for other data traffic.

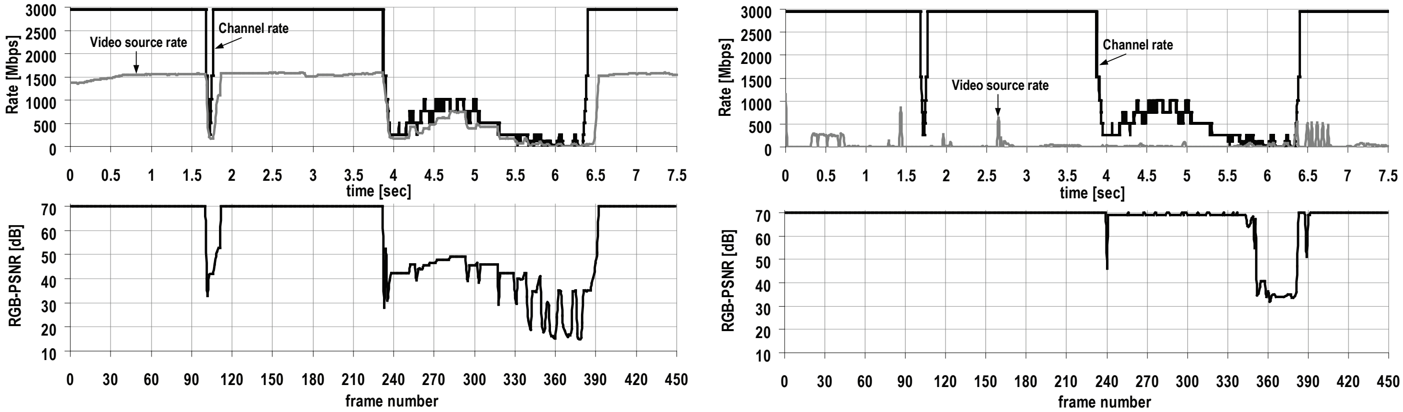

Figure 2. Practical results for “Breeze“ (on the left side) and “Desktop“ (on the right side) video sequences

In bad channel state the proposed rate control algorithm provides adaptation to varying channel conditions and guarantees acceptable video quality for the given channel throughput. Greater effect is obtained for desktop video sequences that contain a lot of static regions.

VII. Conclusions

In this paper the latency-constrained video transmission over high-speed wireless personal area networks was discussed. In the complex comparison it leads to the conclusion that single-layer video compression suits at most for the described situations. The video source rate control algorithm based on MINMAX quality criteria was proposed and practical results for the real channel are shown. The proposed rate control was constructed in one-pass mode: it does not need recompression and it does not necessary need the channel model constuction. By this, the number of operations that is needed for macroblocks types and quantization steps selection in the unit is tiny in comparison to the encoder operations. Therefore, the proposed algorithm does not contribute much to the general encoder power consumtion.

The future works will be devoted to the rate-distortion performance comparison of single-layer video compression with other approaches like uncompressed video, scalable video coding and distributed video coding for high-speed wireless personal area networks taking into account transmission latency and power consumtion restrictions.

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