Video transmission on slowly fading channels using diversity

Slavche Pejoski and Venceslav Kafedziski

Abstract—In slowly fading channels, video frame transmission time is usually longer than channel coherence time, so that fading channel can be represented as a set of parallel channels providing transmission diversity. Using simulation, we evaluate bounds on the quality of the received video, obtained by utilizing the available channel diversity, with no additional delay incurred. We assume that transmission errors occur only when the channel is in outage, outage being defined in terms of outage capacity. Simulations show that utilizing the available channel diversity improves performance in cases when sufficient bandwidth and SNR are available. Performance gains of up to 1.83 dB are identified in the received video PSNR for diversity order 6 at symbol rate of 105000 symbols/sec.

Index Terms-video coding, diversity, outage capacity.

I. INTRODUCTION

THE wireless channel is a time varying medium which can be statistically characterized. According to [7] the time interval during which channel gains have close values is called coherence time and it depends on the velocity of moving. During times when channel gain sustains very small values all data will be lost. This problem is usually solved by retransmission or combination of FEC (Forward Error Correction) and interleaving. For real time video transmission, where the delay is limited to 250 ms, retransmission is not useful. The second option has been analyzed in [2], [4], [5], [8]. It is shown in [2] that diversity at physical layer brings greater benefit than diversity at application layer (multiple description video). In [4], a combination of RS coding and interleaving inside a single video frame has been used, so additional delay is not introduced in the system. Different RS codes are used for different video frames in accordance with the degree of motion of the video frame. This approach is suboptimal because of its inability to fully adapt to the diversity order that is offered by the channel (RS codes do not optimally trade bandwidth for diversity). Authors in [8] offer an algorithm that intelligently orders the video packets in a given time interval. The content of the video packets is taken into consideration and the goal is to minimize quality variation of the packet sequence. In [5] a method for adaptable packet interleaving is proposed. The adaptation is based on the transmitter buffer occupancy, display deadlines of packets and anticipated video input to the wireless channel. This last two methods take into consideration packet level interleaving and

S. Pejoski, Faculty of Electrical Engineering and Information Technologies, University Cyril and Methodius, Skopje, Republic of Macedonia (phone: +38923099191; fax: +38923064262; e-mail: slavchep@feit.ukim.edu.mk).

V. Kafedziski, Faculty of Electrical Engineering and Information Technologies, University Cyril and Methodius, Skopje, Republic of Macedonia (phone: +38923099120; fax: +38923064262; e-mail: kafedzi@feit.ukim.edu.mk). FEC across video frames. They show good results only if the tolerable delay is two or more video frames, so that additional delay is added to the system.

If channel conditions change quickly enough, so that video can be considered to be sent through fast fading channel, then the algorithm proposed in [9] can be used for the video transmission. If this is not the case, the algorithm described in [9] can be used at the price of increase of the video frame delay (in [9] perfect interleaving is considered).

Here we consider slowly fading channels. In our model, outage capacity is relevant quantity and outage probability defines the probability of error for those channels. We consider video transmission system behavior when partial or full available diversity of the wireless channel is utilized, and no additional delay is introduced. Using diversity decreases outage probability. We take this into consideration, adapt the video coding to the new outage probability by choosing mode for the video coding that minimizes the expected distortion of the received video. We evaluate received video PSNR (peak signal to noise ratio) gain in terms of number of symbols per video frame, and in terms of SNR.

The paper consists of four sections. In Section 2 video system is described. In section 3 simulation results are given. Conclusion is presented in section 4.

II. SYSTEM DESCRIPTION

The system used in this paper is shown in Figure 1. Here, wireless channel is modeled as block fading channel and its coherence time is T_c . Channel gain is kept constant during the block, and its value changes from block to block. If single video frame spans L coherence periods in time, then the channel can be represented by a set of L parallel channels with duration equal to the coherence time of the original channel.

In Figure 1, the element called controller unifies all the other elements. It is responsible for the cross layer communication and optimization. The controller receives information from the receiver about the statistics of the wireless channel. This statistics consists of coherence time and average channel SNR. Based on the coherence time, the controller decides about the maximum amount of diversity provided by the channel i.e. the number of parallel channels that can be used to represent the original channel.

In [10] symbol error probability is defined as:

$$P_e = P\{O\} + P(error, O^c) \tag{1}$$

The first term on the right-hand side of (1) is the outage probability, and the second term is symbol error probability



Fig. 1. Transmission system

when the channel is not in outage. The use of powerful channel coding brings second part close to zero, and makes first part a dominant cause of errors. Here, we assume that this is the case, so that we neglect the second term.

Main differences between systems utilizing the available diversity and systems not utilizing diversity is illustrated in Figure 2. In the figure, time flow starts from the left upper corner and ends at the right lower corner, it is always horizontal, and when it reaches the end of the block it continues from the left side of the next time block. Video frame, whose transmission is shown in the figure, consists of 6 video slices which are mapped into 6 video packets of equal length. In this example it is assumed that the wireless channel coherence time is one sixth of the frame duration, so that maximum available diversity order is 6. Figure 2 (a) describes video transmission without utilization of diversity, and Figure 2 (b) shows video transmission that utilizes diversity of order 3.

For communicating through system where probability of symbol error is dominated by the outage probability, approximately universal codes, defined in [6], can be used. In [6] a realization of approximately universal codes for high SNR regime is proposed. So, system that fully utilizes the available diversity is practically realizable at high SNR and requires completely new physical layer. Here, our goal is to find performance bounds for all SNR's in order to get an insight into the potential advantages of using diversity.

Among other functions, the controller swaps information with the source encoder. For every mode that can be used to encode a given video frame, the necessary bits are counted and sent to the controller. During the encoding process every frame is divided into slices. Every slice is mapped into a single packet. For every source coding mode that can be used on a particular slice, error probability is computed by the controller. The error probability in the proposed system is defined by the outage probability, i.e. the probability that the number of bits per transmitted symbol needed to encode that particular slice is higher then the channel capacity in bits per symbol. For a system that does not use diversity, the outage probability is given by:

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$$P_{out_a}(R) = P\left(\log_2(1+|h_i|^2SNR) < R\right) \\ = P\left(|h_i|^2 < \frac{2^R - 1}{SNR}\right)$$
(2)

and for a system that uses diversity of order L, it is given by:

$$P_{out_b}(R) = P\left(\frac{1}{L}\sum_{i=1}^{L}\log_2(1+|h_i|^2SNR) < R\right)$$
(3)

In the previous equations $|h_i|^2$ is random channel power gain (with mean equal to 1), SNR is average channel signal to noise ratio, and R is the ratio between the number of bits needed to send a given slice and the number of symbols available to send that slice.



Fig. 2. Video transmission (a) not using diversity (b) using diversity

Afterwards, the controller chooses a mode that minimizes the expected distortion. Expected distortion is calculated according to the following formula:

$$D = E(f_n^i - \tilde{f}_n^i)^2 \tag{4}$$

In (4), \tilde{f}_n^i stands for anticipated decoded value for the i^{th} pixel of the n^{th} video frame. The controller anticipates this value. f_n^i is the value for the i^{th} pixel of the n^{th} uncoded video frame. Several algorithms for expected distortion computation are known in the literature. Among them the algorithms from [10] and [11] are known to recursively compute the distortion. These algorithms take into consideration channel error probability and error concealment procedure.

After the decision making process, the controller feeds the information for the modes that will be used for the video packet transmission to the source encoder, channel encoder and transmitter.

Inverse operations to those in the transmitter are performed in the receiver. Moreover, the receiver can treat all the lost packets according to the error concealment policy. Information about average SNR and coherence time are sent by the receiver to the transmitter. Numbers, marking packets that were lost during the transmission, can be sent as additional information to the transmitter.

III. SIMULATION RESULTS

The outage probability $P_{out_a}(R)$ evaluated for Rayleigh fading channel is:

$$P_{out_a}(R) = 1 - e^{-\frac{2^{n}-1}{SNR}}$$
(5)

In [6] an algorithm for computation of $P_{out_b}(R)$ is proposed. This algorithm is based on numerical methods. First the pdf (probability density function) p_y of random variable

 $y = \log(1 + \rho)$ is found. Here, ρ is a random variable that follows exponential distribution and represents the channel SNR. Then, a random variable $z = \sum_{i=1}^{L} y_i$ is introduced. Its pdf p_z can be computed by taking convolution of the pdf's of the random variables y i.e.

$$p_z = \underbrace{p_y * \dots * p_y}_L$$

This can be done by computing the characteristic functions of y_i 's, multiplying them, and then, computing the inverse Fourier transform. In our simulations p_y was discretized using 8192 points in the range [-16,16]. Then, we introduce a random variable $z_1 = \frac{z}{L}$, whose pdf is evaluated as $p_{z_1}(x) = Lp_z(Lx)$. We compute $P_{out_b}(R)$ by integrating $p_{z_1}(x)$ in the region [0, R]. In figure 3 the pdf's of the random variables z_1 are shown for values of L equal to 1, 2, 3, and 6 and SNR = 16dB.



Fig. 3. Pdf's of random variable z_1 for (a) L = 1, (b) L = 2, (c) L = 3, (d) L = 6

From Figure 3 we conclude that increasing L, pdf of random variable z_1 becomes more peaky, as expected. Notice that, when $L \to \infty$, this pdf becomes a Dirac delta function, located at channel ergodic capacity. Figures 3(a)-3(d) give an insight in the amount of outage probabilities for arbitrary R when L changes from L = 1 to L = 6, as they are given by the areas between x = 0 and x = R below $p_{z_1}(x)$ respectively.

We encode every slice in a different slot, that consists of 1/c of the total number of symbols available for transmission of a single video frame, where c is the number of slices in the video frame. The symbols for transmission of each slice are sent on L different channel blocks (different coherence intervals), when diversity of order L is used. The number of symbols available for transmission of a single video frame is R_s/f_r , where R_s is the symbol rate and f_r is the number of frames transmitted in a single second. We assume that $f_r = 30$. Base version of h.263 available at [1] is used as source coder. The allowed quantization levels for slice encoding are from

the set $Q = \{10, 17, 26\}$. Every macro block in the slice can be encoded intra or inter.

For the computation of the expected distortion, we used the ROPE algorithm, first introduced in [10].

SQCIF video sequence Foreman was used in all the simulations. We used 299 video frames of the sequence and encoded them in IPPP... format i.e. the first video frame was coded intra and all the rest were coded inter. This sequence has maximum value for the PSNR (Peak Signal to Noise Ratio) of 32.97 dB, obtained with no channel errors and using the best quantizer. SQCIF video frame was divided into 6 slices. During the encoding process of every slice, resynchronization marker was used, in order to improve the error resilience of the transmitted video. Simple error concealment procedure, that fills in the pixels of the lost slices with copies of the pixels at the same locations in the previous video frame, was used. The information for packets lost during transmission is fed back without delay. In other words, during the encoding process of the current frame, the positions of packets lost during the transmission of the previous frame were assumed to be known.

In all simulations, the channel was modeled as block fading channel with uncorrelated channel gains on different blocks that follow Rayleigh distribution. The coherence time of the channel was set to $T_c = 1/(30 \times 6) = 5.55 \text{ ms}$ so that maximum available diversity order is 6.

In the simulations four different cases were compared. In the first case the available diversity was not utilized, so that the diversity order was equal to 1. Diversity orders of 2, 3 and 6 were used in order to compare performance with the non-diversity case. When L = 2 each slice was transmitted on two halves of two channel blocks, when L = 3 each slice was transmitted on three thirds of three channel blocks, and when L = 6 each slice was transmitted on six sixths of six channel blocks. All the simulations were performed for 20 different realizations of the wireless channel.

Simulation results are shown in figures 4 and 5. The dependence of PSNR on the number of available symbols R_s/f_r for transmission of a single video frame (or, equivalently, on the symbol rate R_s) is given in figure 4. The average signal to noise ratio was set to the value SNR = 16dB. Based on the figure it can be concluded that the quality of the received video, in case of sufficient symbol budget, is higher when using diversity i.e. coding over parallel channels results in performance gain, which increases when diversity order L increases from 2 to 6. Maximum PSNR gain of 1.29 dB is obtained from the curve corresponding to L = 6. In the region of low symbol rates the PSNR gain of the scheme with any diversity order compared to the scheme with diversity order 1 decreases, but in this region the quality of the received video is such that is of no substantial practical interest. In the region of high symbol rates, all four curves (diversity order equal to 1, 2, 3 and 6) tend to the upper limit (max PSNR value).

Received video PSNR in terms of the average channel SNR is presented in figure 5. During this simulation the number of available symbols for transmission of a single video frame was set to 3500 (symbol rate of 105000 symbols/sec).

Observation of figure 5 leads to a conclusion that gain of up



Fig. 4. PSNR in terms of number of available symbols in a single frame



Fig. 5. PSNR in terms of average channel SNR

to 1.83 dB in PSNR of the received video can be achieved by utilizing diversity order 6. Similar behavior as in Figure 4 was observed, i.e performance gain increases when diversity order increases. Also, similarly to what was observed in Figure 4, in the regions of very low and very high SNR, PSNR gain of the scheme with any diversity order compared to the scheme with diversity order 1 decreases. In the region of very high SNR all the curves (diversity order equal to 1, 2, 3 and 6) tend to the upper limit (max PSNR value).

Results presented here show that video encoding over parallel channels can achieve performance gain, if the number of available symbols in a video frame (or, equivalently, the symbol rate) and the average SNR are in the region of practical interest.

IV. CONCLUSION

In this paper real time video transmission on wireless channel was investigated. In our system, the transmitter has partial information about the channel. Based on this information it converts the wireless channel into a set of parallel channels, whose time duration is equal to the coherence time of the original wireless channel. The number of parallel channels is equal to the ratio of the time available for sending a single video frame and the coherence time of the wireless channel. It is assumed that the symbol error probability is dominated by the outage probability. Simulations are performed to determine the influence of utilization of available diversity in systems where outage probability is the dominant cause of transmission errors. Simulations show PSNR gain of up to 1.83 dB when PSNR is given as a function of SNR, at symbol rate of 105000 symbols/sec and when diversity of order 6 is utilized.

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