Wireless Video Transport Using Conditional Retransmission and Low-Delay Interleaving

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ABSTRACT

We consider the scenario of using Automatic Repeat reQuest (ARQ) retransmission for two-way low bit-rate video communications over wireless Rayleigh fading channels. Low-delay constraint may limit the number of retransmissions, thus there will be packet-errors at the decoder which results in video quality degradation. In this paper, we propose a scheme to improve the video quality. First, we propose a low-delay interleaving scheme that uses video encoder buffer as a part of interleaving memory. Second, we propose a conditional retransmission strategy that reduces the number of retransmissions. Simulation results indicate our proposed scheme can reduce the number of packet errors and improve the channel utilization. As a result, we obtain PSNR (Peak Signal to Noise Ratio) improvement up to about 4 dB compared to H.263 TMN8.

1. INTRODUCTION

For two-way video communications over a narrow-band wireless Rayleigh fading channel with burst-errors nature, the video is usually encoded with a low-bit-rate low-delay video coding standard such as H.263 [1]. H.263 applies motion-compensated prediction and variable-length coding to reduce the temporal and statistical redundancy between the video frames. This scheme increases the compression ratio but also makes the signal susceptible to transmission errors.

Since a feedback channel is available in two-way communications applications and the communication channels of our consideration have short and constant end-to-end delays, ARQ schemes can be used to obtain reliable communication [2]. A block diagram of an ARQ-based wireless video system is shown in Fig. 1. In Fig. 1, during the retransmissions of error packets when the channel is in deep fades, the video data in the video encoder buffer are not transmitted. Due to the reduced channel throughput, the video encoder buffer fills up quickly as shown in [3,4]. Also, in two-way interactive applications, low-delay constraint may limit the number of retransmissions. Thus, there will be packet-errors due to the corrupted retransmitted packets. Several error concealment schemes in [5] can be used to minimize the visible distortion of the video frame. However those schemes may not be as effective when the packet error rate in the wireless channel is high during fading. We have investigated the effects of retransmission and packet-errors at the video decoder when delay constraint allows only one retransmission. The results show a significant drop in PSNR due to the packet errors. In this work we propose a scheme to significantly improve the video quality.

To reduce the packet errors, interleaving can be used with Forward Error Correction (FEC) to spread out burst errors to random errors so that they can be more easily corrected. However, in two-way video communications, the end-to-end delay constraint places a limitation on the use of interleaving. In this paper, we propose a low-delay interleaving scheme which uses the video encoder buffer as a part of the interleaving memory so that the interleaving does not increase the delay and the memory in the video encoder.

In addition to the interleaving, the retransmission strategy itself can be improved. Several refinements of ARQ schemes with video have been proposed in the literature. In [6], it has been proposed that for delay-constrained continuous media applications in which retransmission should be aborted if the time left before presentation is less than the roundtrip delay time. The combination of layered coding with prioritized retransmission, by placing the base-layer packets in the front of the transmission queue to increase the number of retransmission trials for the base layer, has been proposed in [12]. Multiple copies of an error packet can also be sent in each single retransmission trial to increase the probability of receiving a correct packet [13]. However, such approaches are more applicable in less delay critical applications such as one-way real-time video applications (Internet video streaming and broadcast). Using this scheme may worsen the problem in the low-bandwidth wireless transmission scenario since it will reduce effective channel throughput and increase delay [3]. In this paper, we provide an alternative approach to improve the effective channel throughput. We propose a conditional retransmission scheme to reduce the number of retransmissions. We use the concealment error and the channel condition to determine whether a packet is worthwhile to retransmit. We also investigate the trade-off between the saved-bits due to the reduced retransmission and the distortion resulting from the concealment error due to the not-retransmitted packets, and provide an analysis based on the rate-distortion framework.

The organization of this paper is as follows: In Section 2, we discuss our low-delay interleaving scheme. In Section 3, we describe the proposed conditional retransmission scheme and a theoretical analysis based on the rate-distortion framework. Section 4 presents our simulation results and demonstrates that our proposed scheme is effective in reducing the Packet Error Rate (PER), which results in significant PSNR improvements compared to TMN8. The conclusion is presented in Section 5.

2. LOW-DELAY INTERLEAVING

An interleaving scheme with a BCH error-correction code has been shown to provide good performance in improving BER [8]. However, applying interleaving has two negative aspects: increasing end-to-end delay and increasing the required memory at the encoder and the decoder.

To alleviate the negative aspects of interleaving, we propose to use the encoder buffer as a part of the interleaving memory. The block diagram of incorporating the interleaving scheme into the encoder buffer is shown in Fig. 3. If the encoder buffer-size is M bits (corresponds to M/R_c ms delay where R_c is the channel rate in kbits/s), the interleaving memory size is set to be X% of the video encoder buffer size M. The interleaving is performed only when the video encoder buffer fullness is greater than X% (to not introduce interleaving delay into the system, since the data are already in the buffer) and when the channel is in the bad state (to save overhead when the channel condition is in the good state). If the encoder buffer fullness is low or the channel condition is good, we will rely only on retransmissions without the interleaving. The algorithm can be summarized as follows:

If (Buffer_fullness_level ≥ X% of the encoder_buffer_size) and (Current_State = S_b) {
Mark the boundary of data and Perform Interleaving;
}
else {
Retransmissions only;
}
where S_b is the bad channel state which can be determined as described in [3,4] (e.g, by the average ratio of retransmitted bits to the average number of transmitted bits in the past N coded frame-intervals).

We define $n\lambda$ as the interleaving block where n is the FEC codeword length (in bits) and λ is the interleaving depth (in bits). We use a BCH(n,k) code where for every k bits of actual data, n-k redundancy bits are added to the codeword [9]. The interleaving degree should be sufficiently large to spread out burst errors in time. At the receiver side, the interleaved packets will be read into the deinterleaving memory in vertical direction and read out in horizontal direction. Error correction is performed after this process. If a packet has uncorrectable errors, it will be requested for retransmission.

From our proposed algorithm, we could save the encoder interleaver delay of ~ $k\lambda/R_c$ ms, and the encoder interleaving memory of ~ $k\lambda$ bits.

In the simulations, $k\lambda$ was set to 50% of the video encoder buffer size. Since in TMN8, the encoder buffer size is M = 3200 bits (corresponds to 100 ms delay), the interleaving memory is set to be 1600 bits. Due to the constraint from this interleaving memory and the need to have low-overheads, we investigated several choices of BCH(n,k) which can correct one-bit error. Based on our simulation results, we choose BCH(25,20) code with a block interleaving depth λ = 80 bits which represents a reasonable tradeoff between the overhead incurred from the BCH codes and the resulting PER (= 0.05) after applying the FEC code with interleaving and retransmission.

3. CONDITIONAL RETRANSMISSION BASED ON CONCEALMENT ERROR

To further improve the channel bandwidth utilization, we propose a conditional retransmission strategy based on the concealment error. The motivation is from the observation that some packets may not be worth retransmitting if the concealment at the decoder can do a good job. To calculate the concealment error when a packet is lost, the same concealment mechanism used at the decoder is implemented at the encoder so that there is no mismatch between the encoder and the decoder. The following analysis based on the rate-distortion (R-D) framework in [10] gives some insights.

3.1. Quality Penalty Due to Error Concealment

If an error packet is not resent, the concealment error based on mean squared error (MSE) caused by repeating the previous frame's content is:

$$D_{CE} = \frac{1}{N_L} \sum_{(x,y)\in L} (\hat{f}_k(x,y) - \hat{f}_{k-1}(x,y))^2 \qquad (1)$$

where D_{CE} is the concealment error of the damaged area, frames k and k-1 are the current frame and the previous frame respectively, $\hat{f}(x,y)$ is the reconstructed pixel value at the coordinate (x,y), L is the damaged area due to the error packet, and N_L is the number of pixels in the damaged area. This can also be calculated at the encoder when a NAK is received and after completing the encoding of the corresponding GOB.

3.2. Quality Penalty Due to Resending Lost Packets

Resending the error packets may avoid the above concealment error provided that if they are not corrupted again. However it will reduce the effective throughput. According to the TMN-8 rate distortion model [10], we can derive the relationship between the mean squared error and the bit-allocation at frame-level if the optimum quantization scheme in [10] is adopted.

$$\begin{aligned} D^* &= \frac{1}{N}\sum_{n=1}^{N}\alpha_n^2 \frac{Q_n^2}{12} \\ &= \frac{1}{N}\sum_{n=1}^{N}\left(\frac{\alpha_n^2}{12}\frac{AK}{(B_{frame}-ANC)}\frac{\sigma_n}{\alpha_n}\sum_{m=1}^{N}\alpha_m\sigma_m\right) \\ &= \frac{AK}{12N}\left(\sum_{n=1}^{N}\alpha_n\sigma_n\right)^2 R^{-1} \end{aligned} \tag{2}$$

where D^* is the mean square error of the coded frame, $R = B_{frame} - ANC$, N is the number of macroblocks in a frame, A is the number of pixels in a macroblock, C is the average rate to encode the motion vectors and the bit-stream header for the frame, B_{frame} is the bit-allocation to the frame, Q_n is the quantizaion step-size of the nth macroblock, and σ_n^2 and α_n are the variance and the distortion weighting factor of the nth motion-compensated residual macroblock, respectively.

Therefore, the quality penalty caused by resending the lost packets in the ith GOB is

$$D^i = \frac{AK}{12N}\left(\sum_{n=1}^{N}\alpha_n\sigma_n\right)^2\left(\frac{1}{B_{\text{frame}} - \sum\limits_{\text{the lost packet } j\in \text{GOB}_i} B_{\text{packet}_j} - ANC} - \frac{1}{B_{\text{frame}} - ANC}\right) \tag{3}$$

where B_{packet} is the number of bits in a packet.

Note that, we should also take into account the possibility of the retransmitted packets being corrupted again. If the packet error probability is β, then

$$D_{\text{retx}} = \frac{AK}{12N}\left(\sum_{n=1}^{N}\alpha_n\sigma_n\right)^2\left(\frac{1}{B_{\text{frame}} - \sum\limits_{\text{the lost packet } j\in \text{GOB}_i} B_{\text{packet}_j} - ANC} - \frac{1}{B_{\text{frame}} - ANC}\right) + \beta D_{\text{CE}} \tag{4}$$

where the second term at the right-hand side is the concealment error when the retransmitted packets is lost.

Base on this analysis, we can use D_{CE} and D_{retx} to decide whether to resend the lost packet. However D_{retx} will require significantly more computation. Instead of comparing D_{CE} and D_{retx}, we found from simulations that comparing D_{CE} with a constant threshold can also give satisfactory results. Effectively, when the concealment error is smaller than the threshold, it indicates that the concealment can do a good job, and the packet does not need to be retransmitted. The algorithm can be summarized as follows:

```
If (NAKPacket_j ∈ GOB_i) and (NAKPacket_{j-1} ∈ GOB_i) {
        Decision_as_previous_packet;
}
else {
        If ( D_CE < T) and ( Current_State = S_b){
                do not retransmit;
        }
        else {
                retransmit;
        }
},
```

where S_b is the bad channel state, and T is a threshold. The same threshold is used in all of the simulations. From this decision rule, if a packet is not retransmitted, the succeeding packets in the same GOB will also not be retransmitted since the concealment will be used for the area of that GOB. Also, when D_{CE} is less than T and the channel condition is bad, we will not retransmit that packet. Under these conditions, the packets are not worth retransmitting because the concealment can do a good job and there is a high probability that the retransmitted packets will be corrupted again due to the bad channel condition.

4. SIMULATION RESULTS

In the simulations, a wireless channel simulator simulating Rayleigh fading channels is used to produce various wireless channel conditions through the setting of several parameters. The bit-error pattern is generated to represent a particular channel characteristic. Based on the packet size, a packet-error pattern and its statistics such as the average packet-error-burst-length and packet-error-rate can be obtained. More details of the simulator can be found in [3,4],[11]. In our simulations, channel parameters corresponding to a BER of 0.01, a packet-size of 80 bits with corresponding packet error-rate of 0.15, and an average burst-length of 19 packets, which corresponds to a slow-fading environment, are used.

We consider the case where the low-delay constraint allows only one retransmission which results in packet errors at the video decoder. These packet errors will cause errors to propagate. The Group-of-block (GOB) is the synchronization unit in H.263. For a QCIF video, each picture is subdivided into 9 GOBs, indexed 0 to 8 from top to bottom. Each GOB includes 11 macroblocks, where the column of macroblocks is indexed from left to right [1]. At the receiving end, if the H.263 decoder detects a packet error, the decoder will give up decoding the corresponding macroblock and the following macroblocks in that GOB, and seek the next GOB sync-word. The corrupted macroblocks in the GOB will be discarded and replaced by the macroblocks at the same location in the previous decoded-frame [14]. Selective-repeat ARQ with a wireless channel round-trip delay of 30 ms is assumed.

Test video sequences including “Claire”, “Car phone”, "Miss America", and "Suzie" in the QCIF format (176x144 pixels/frame) were encoded at 32 kb/s with a target frame-rate of 10 frames/s using TMN8 and our proposed scheme.

Figs. 3 and 4 show the simulation results for conditional retransmission of "Miss America" sequence with the PSNR comparisons among TMN8 with clean channel, TMN8 with packet-errors and concealment, and our proposed scheme. It has been shown that TMN8 under channel errors has a PSNR drop of around 7 dB compared to the clean channel. Fig. 3 shows the results of our proposed interleaving scheme of “Miss America” sequence. Our scheme shows an improvement of about 2 dB compared to TMN8 (with error concealment) under the same condition. Fig. 4 shows the simulation results for the “Miss America” sequence with the proposed interleaving and conditional retransmission . Our overall scheme shows an improvement of about 4 dB compared to TMN8. Table 1 shows the average channel throughput and PSNR comparison for all video sequences tested. Fig. 5 gives a subjective evaluation of the video quality for "Miss America" sequence of our proposed scheme compared to TMN8 where frames with significant PSNR improvement in video sequences are shown. It shows the effects of packet errors after error concealment. The improvement is due to the reduced packet error.

5. CONCLUSIONS

In this paper, we proposed a low-delay interleaving and conditional retransmission scheme to improve the video quality for wireless video. We also analyzed the tradeoff between the saved bits (from the conditional retransmission) and the concealment error. Simulation results show improvement in PSNR of up to about 4 dB for our scheme compared to H.263 TMN8. Subjective evaluations also confirm the significant video quality improvement.

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Table 1 Comparison of the average throughput and PSNR for TMN8 under clean channel, channel errors, and using our proposed scheme

Video Sequence	Total Frames	Clean Channel		TMN8 with packet loss and concealment		Proposed Interleaving Scheme		Interleaving and Conditional Retransmission	
		Average Throughput (kbps)	PSNR (dB)	Average Throughput (kbps)	PSNR (dB)	Average Throughput (kbps)	PSNR (dB)	Average Throughput (kbps)	PSNR (dB)
“Claire”	162	32.0	39.51	28.1	32.68	29.0	34.58	29.9	36.48
“Car phone”	124	32.0	30.81	27.3	24.41	28.5	26.09	29.3	27.12
"Miss America"	49	32.0	39.86	25.5	32.00	28.0	34.31	30.2	36.46
"Suzie"	49	32.0	34.16	25.5	27.09	29.1	29.74	30.9	32.65

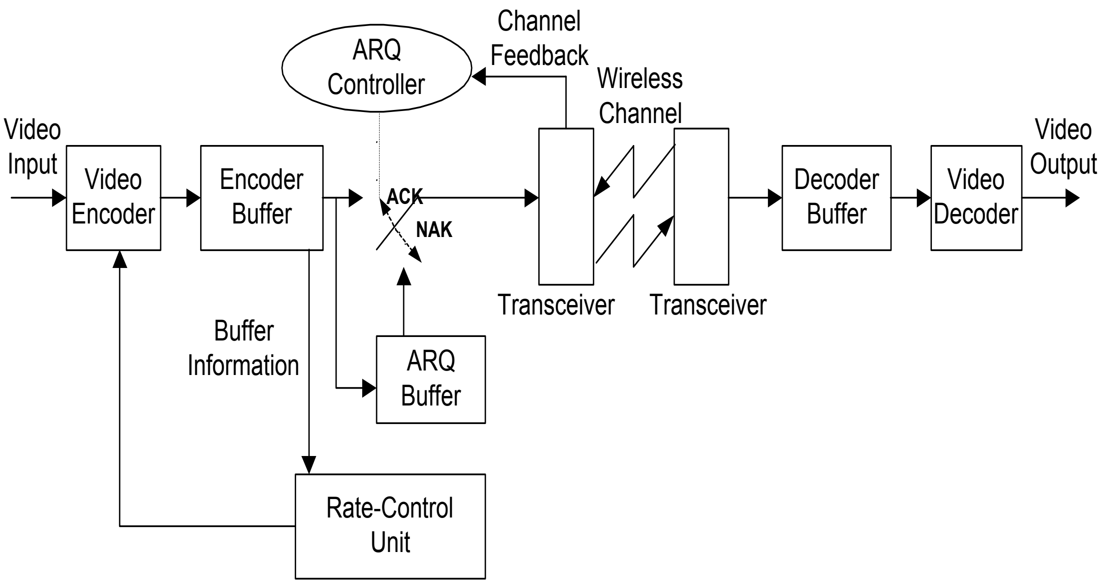


Figure 1 A block diagram of a retransmission-based wireless video system

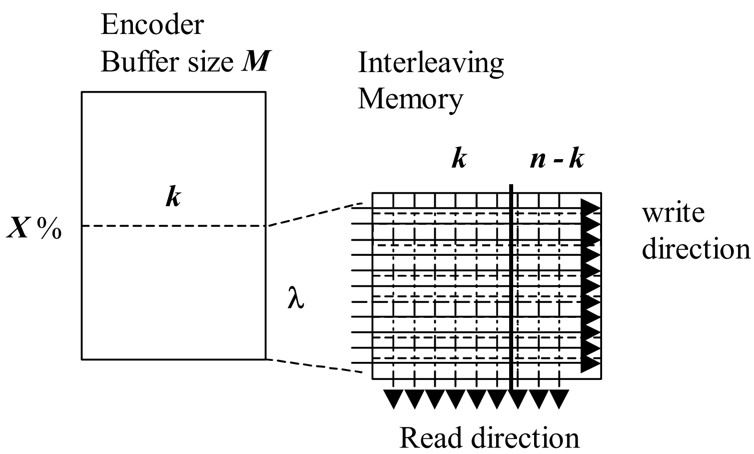


Figure 2 Block Diagram for the combined encoder buffer and the interleaver

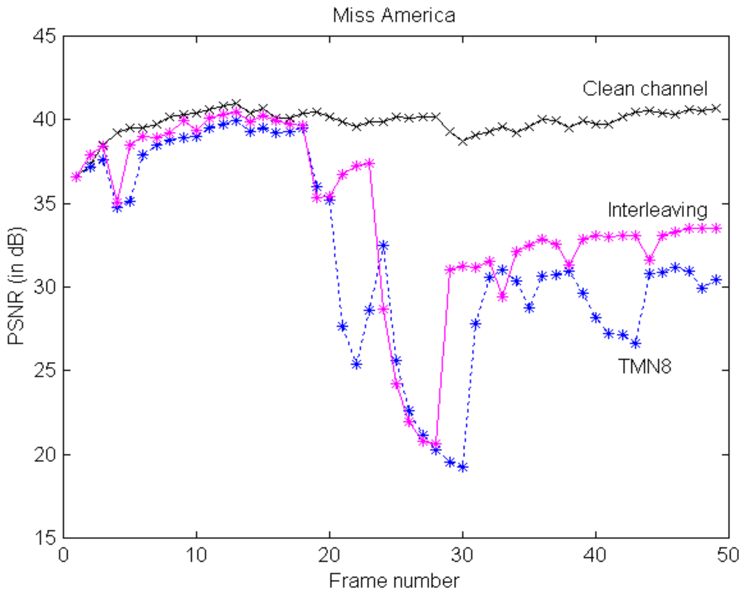


Figure 3 PSNR comparison for the "Miss America" sequence between TMN8 in clean channel (-x- line), TMN8 with packet loss and concealment (dashed -*- line), and our proposed interleaving scheme (solid -*-line). TMN8 results in 7 frames skipped while our proposed scheme has no frames skipped.

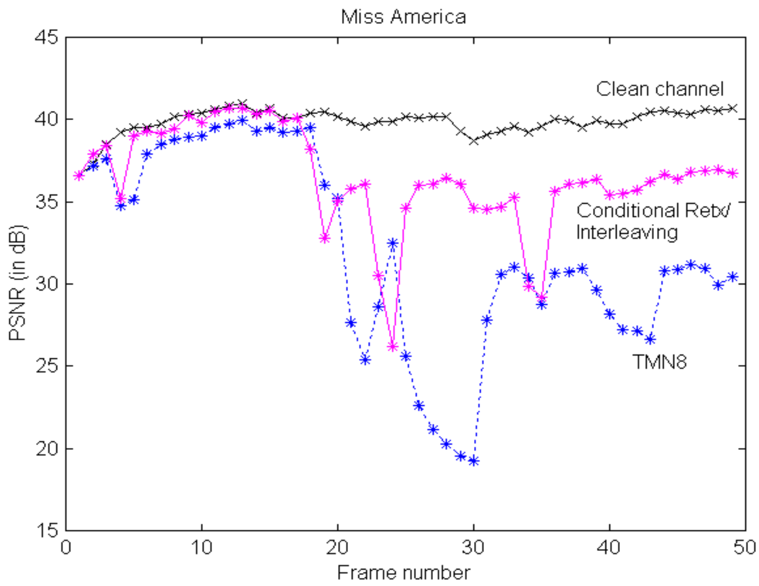
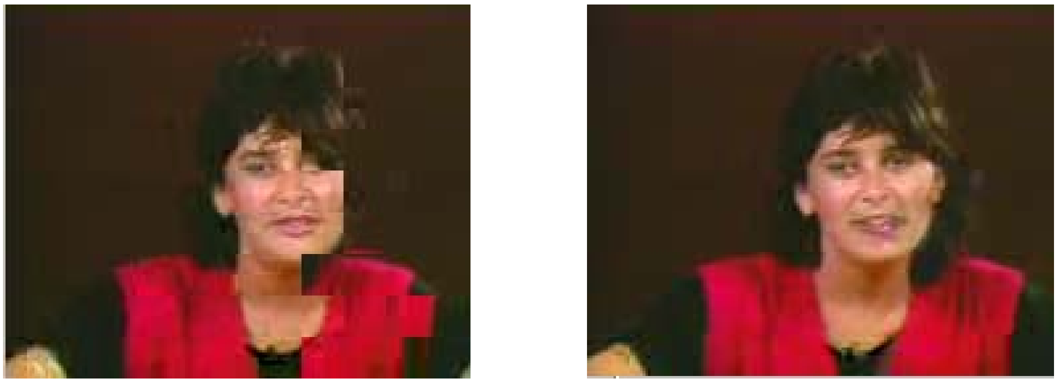


Figure 4 PSNR comparison for the "Miss America" sequence between TMN8 in clean channel (-x- line), TMN8 with packet loss and concealment (dashed -*- line), and our proposed conditional retransmission and interleaving scheme (solid -*-line). TMN8 results in 7 frames skipped while our proposed scheme has no frames skipped.

TMN8 **Proposed Method**



(a) **(b)**

Figure 5 Subjective evaluation for 32 kb/s Miss America sequence. Reduce number of packet errors from interleaving with PSNR improvement of 7.5 dB for frame number 31. TMN8 shown in (a) and proposed conditional retransmission and interleaving scheme shown in (b).