Transactions Letters

Wireless Video Transport Using Conditional Retransmission and Low-Delay Interleaving

Supavadee Aramvith, Chia-Wen Lin, Sumit Roy, and Ming-Ting Sun

***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 require that a corrupted retransmitted packet not be retransmitted again, and 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 the video encoder buffer as a part of interleaving memory. Second, we propose a conditional retransmission strategy that reduces the number of retransmissions. Simulation results show that our proposed scheme can effectively reduce the number of packet errors and improve the channel utilization. As a result, we reduce the number of skipped frames and obtain a peak signal-to-noise ratio) improvement up to about 4 dB compared to H.263 TMN-8.**

***Index Terms*—Channel feedback, conditional retransmission, interleaving, wireless channels, wireless video.**

I. Introduction

TWO-WAY video communications over a low bit-rate channel is suitable for support of several applications such as videophone and video conferencing. The H.263 standard [1] developed for this purpose achieves good compression ratios but also makes the signal susceptible to transmission errors. Even a single-bit error may cause the error to propagate to many frames because of the motion compensated prediction and the variable-length coding used. In this paper, we investigate the impact of bursty errors typical of a wireless (fading) channel, and propose a hybrid forward error correction (FEC) and retransmission (ARQ) scheme for robust video transport. The presence of a feedback channel and constant end-to-end delay makes delay-constrained ARQ approaches suitable [2].

A block diagram of the retransmission-based system is shown in Fig. 1. The encoder buffer is used to smooth out the video bit-rate to prevent the bits from being discarded when the instantaneous video bit-rate exceeds the channel bandwidth. The transmitted packets are kept in the ARQ buffer until they are acknowledged being received correctly at the decoder. In Selective Repeat ARQ, the receiver sends a positive ACKnowledgement (ACK) or a Negative AcKnowledgement (NAK) to the transmitter, depending on whether the packet is received correctly or not. The transmitter retransmits the corresponding packet in the ARQ buffer when it receives a NAK from the receiver. From the video transmission point of view, the wireless channel has time-varying capacity due to the retransmissions.

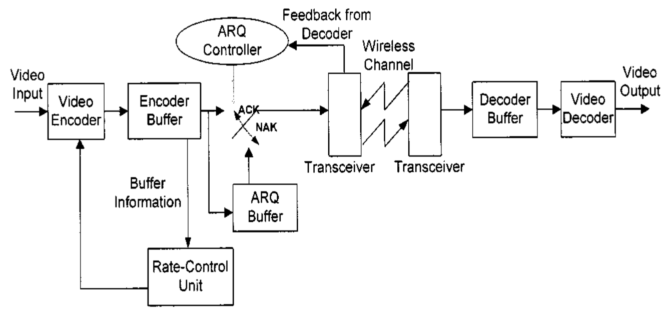

Fig. 1. Block diagram of a retransmission-based wireless video system.

During times of reduced channel throughput when the channel is in a deep fade and there are lots of retransmissions, the video encoder buffer may fill up quickly, causing the rate-control algorithm to significantly reduce the number of bits allocated to each frame and to skip video frames. In our earlier work [3], a rate-control scheme that exploits an *a priori* two-state Markov model to estimate the future channel throughput to better allocate the target numbers of bits than TMN-8 for each frame and each macroblock was proposed. It effectively improves the PSNR and reduces the number of skipped frames. The results reported there were based on the assumption that multiple retransmissions are allowed to ensure reliable packet delivery which will eventually result in error-free packets, i.e., no delay constraint was imposed. In this paper, we consider a more practical scenario for two-way interactive applications where the number of retransmissions is limited; thus, there will be packet errors.

A traditional approach to mitigating the effect of burst errors is by use of interleaving in conjunction with FEC and ARQ [6], [7]. Interleaving spreads burst errors into multiple bit-errors, so that they can be corrected using simple error-correction coding. For uncorrectable errors, it will be requested for retransmission. Unfortunately, interleaving adds a significant additional delay.

Manuscript received July 15, 2001; revised April 15, 2002.

S. Aramvith is with the Department of Electrical Engineering, Faculty of Engineering, Chulalongkorn University, Bangkok 10330, Thailand (e-mail: supave@ee.eng.chula.ac.th).

C.-W. Lin is with the Department of Computer Science and Information Engineering, National Chung Cheng University, Chiayi 621, Taiwan, R.O.C. (e-mail: cwlin@cs.ccu.edu.tw).

S. Roy and M.-T. Sun are with the Department of Electrical Engineering, Box 352500, University of Washington, Seattle, WA 98195-2500 USA (e-mail: roy@ee.washington.edu; sun@ee.washington.edu).

Publisher Item Identifier 10.1109/TCSVT.2002.800326.

In this paper, we propose a low-delay interleaving scheme which uses the video encoder buffer as a part of interleaving memory so that the interleaving does not increase significantly the delay and the memory requirements beyond that imposed by 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, such as delay-constrained retransmission [7], prioritized retransmission of multilayer video [16], and dynamic retransmission of multiple copies of an erroneous packet [18]. However, the above methods did not address the problem of reduced channel bandwidth due to the retransmissions. In this work, we propose a conditional retransmission scheme to reduce the number of retransmissions so as to improve the effective channel throughput. We use the concealment error and the channel condition to determine whether a packet is worthwhile to retransmit. We also provide a rate-distortion analysis of the trade-off between the saved-bits due to the reduced retransmission and the increased distortion resulting from the concealment error due to the not-retransmitted packets.

The organization of this paper is as follows. In Section II, we first illustrate the problem by showing the effects of packet errors on the video quality. We then describe our proposed low-delay interleaving and conditional retransmission schemes. We also provide a theoretical analysis based on the rate-distortion framework. Section III 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 TMN-8. Conclusions are presented in Section IV.

II. Proposed Low-Delay Interleaving and Conditional Retransmission Schemes

A. Effects of Packet Errors on the Video Quality

We perform simulations to show the effects of packet errors on the video quality in the retransmission-based system. 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. 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 will be replaced by the macroblocks at the same location in the previous decoded-frame [20]. Through the NAK from the decoder, the encoder knows the damaged area and can perform the same error-concealment as in the decoder so that there is no drift-errors. Selective-repeat ARQ with a wireless channel round-trip delay of ~30 ms is assumed, since the duration between retransmission attempts is determined by the round-trip delay (RTD) [15]. The ITU-T G.114 standard recommends a maximum delay of 150 ms for international telephone conversations [14] and is also relevant for interactive video conferencing applications. Detailed studies of where the delays typically happen in wireless communications can be found in [30], [31]. In this paper, to be conservative, we impose a delay constraint to allow only one retransmission; if a packet arrives at the decoder too late to meet the delay constraint, it is considered a lost packet.

TABLE I
WIRELESS CHANNEL AND AIR INTERFACE PARAMETERS USED IN THIS STUDY

Multiple Access	TDMA
Modulation	QPSK
Channel rate	32 Kbps
Maximum Doppler Frequency	1 Hz
Signal-to-Noise Ratio	15 dB
Time delay spread	¼ of symbol period
Antenna Diversity	1
Frame Size	80 bits

A wireless channel simulator simulating Rayleigh fading channels with the personal access communication services (PACS) air interface [4] as described in [3], [11] is used in our study. This simulator was designed based on the techniques described by Jakes [5] and has proven to be effective [11]. The channel parameters used in our experiments are listed in Table I. The corresponding bit-error-rate (BER), PER, and average-burst-length are 10^{-2}, 0.15, and 20, respectively. To show the effects of the packet errors on the video quality, several video sequences were encoded at 32 kbits/s using TMN-8 [10]. The coded video sequences are corrupted using the error patterns generated from the wireless channel simulator representing various wireless channel conditions. The 494-frame "Claire" video sequence has an original frame-rate of 30 frames/s. It is encoded with a target frame-rate of 10 frames/s. The result encoded by TMN-8 without channel-error contains 162 coded video frames. With channel errors, TMN-8 results in 12 frames skipped and a PSNR drop of around 7 dB, compared to the clean channel. With the improved rate-control scheme in [3], it results in no frame skipped and a PSNR drop of around 6.4 dB compared to the clean channel, as shown in Fig. 2. These are typical of many other video sequences that we experimented with. In the following, we propose new schemes to improve the result.

B. Low-Delay Interleaving

To reduce the number of error packets, a hybrid ARQ/FEC scheme can be used. Interleaving can be used with an FEC code to spread out burst errors to random errors. 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: 1) increasing end-to-end delay and 2) increasing the required memory at the encoder and the decoder.

To alleviate the negative aspects of interleaving, we propose to use 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. The 50% of the encoder buffer also served as the interleaving memory. When the video encoder

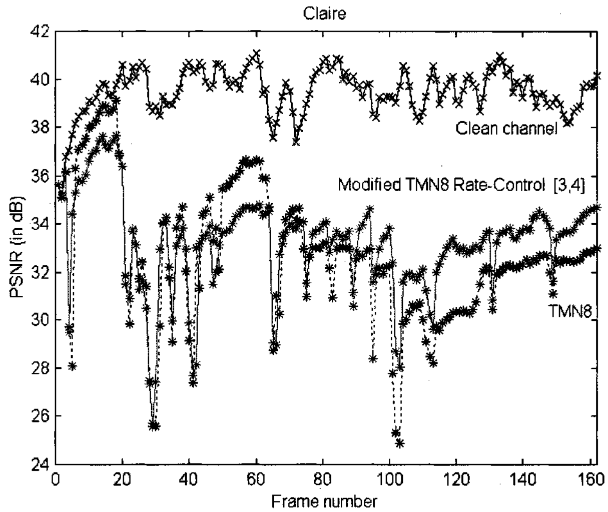

Fig. 2. PSNR comparison for the "Claire" sequence between TMN-8 in clean channel (-x-line), TMN-8 with packet loss and concealment (dashed -*-line), and the scheme proposed in [3] (solid -*-line). TMN-8 results in 12 frames skipped at the encoder while the proposed scheme in [3] has no frames skipped.

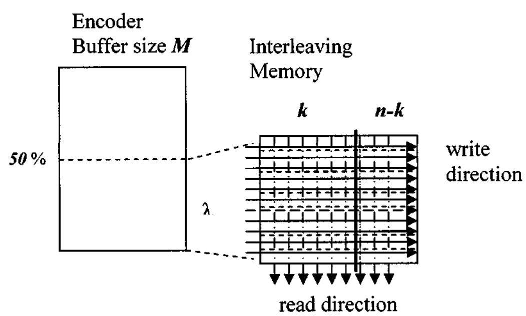

Fig. 3. Block diagram for the combined encoder buffer and the interleaver.

buffer fullness is greater than 50% and the channel is in the bad state, the interleaving is performed on the data in the interleaving memory in the encoder buffer (this does not introduce extra interleaving delay, since the data are already in the encoder buffer). If the encoder buffer fullness is lower than 50% or the channel condition is good, we rely only on retransmissions without the interleaving. The algorithm can be summarized as follows:

If (*Buffer_fullness_level* $>$ 50% *of the encoder_buffer_size*)
and (*Current_State* $= S_b$) $\{$
Mark the boundary of data and Perform Interleaving;$\}$
else $\{$*Rely on retransmissions only*;$\}$

where S_b represents the bad channel state which can be determined as described in [3].

The interleaving memory is organized as a $n\lambda$ 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. Conceptually, at the encoder side, the data are written into the interleaving memory in the horizontal direction and read out in the vertical direction. At the receiver side, the interleaved packets are written into the deinterleaving memory in the vertical direction and read out in the horizontal direction. Error correction is performed after this process. If a packet has uncorrectable errors, it will be requested for retransmission.

TABLE II
COMPARISON OF VARIOUS BCH CODES WITH 1-BIT ERROR CORRECTION CAPABILITY, INTERLEAVING DEGREE, AND TRADE-OFF BETWEEN THE OVERHEAD AND PER

BCH(n,k)	Interleaving degree	Overhead (%)	PER (After Retx)
NO BCH	-	-	0.11
(14,10)	160	30	0.025
(21,16)	100	24	0.04
(25,20)	80	20	0.05
(30,25)	64	17	0.06
(46,40)	40	13	0.07

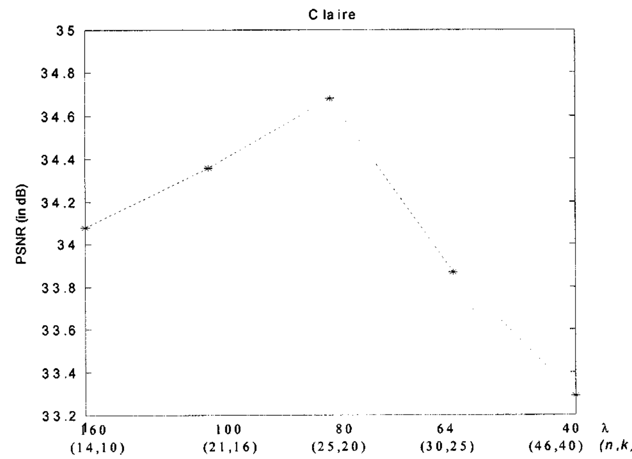

Fig. 4. Effects of different interleaving degree (λ) and BCH code (n, k) on video quality. Test sequence is "Claire."

Traditionally, the encoder buffer and the interleaving memory were implemented separately. The data were read out from the buffer at the channel rate. From our proposed algorithm, we could save the encoder interleaver delay of $\sim k\lambda/R$ ms and the encoder interleaving memory of $\sim k\lambda$ bits, where R is the rate (in kb/s) of data being read out from the encoder buffer. Since in TMN-8, the encoder buffer size is $M = 3200$ bits (corresponds to 100 ms delay), $k\lambda$ is set to be 1600 bits. Several choices of BCH(n, k) which can correct one-bit errors were investigated. The overhead incurred from the BCH codes and the resulting PER after applying the FEC code with interleaving and retransmission are shown in Table II. The simulation results show that, for a fixed $k\lambda$, with each longer choice of BCH codeword and shorter interleaving degree λ, the overhead and delay are decreased but the PER is increased. Fig. 4 shows the simulation results of different combinations of BCH(n, k) codeword and interleaving degree λ on video quality for the test sequence "Claire." When a shorter codeword with a larger interleaving degree is used, the larger overhead reduces the effective channel throughput but the PER is improved, and vice versa. The optimal point is near $k = 20$ and $\lambda = 80$. After the point, the PSNR drops because a smaller interleaving degree cannot effectively reduce PER even with smaller overhead which increases the effective channel throughput. The simulation results suggest that the effect of reduced PER on the video quality is stronger than the effect of reduced the channel throughput caused by the overhead. Note that the optimal λ value is related to the burst-error-length statistics since the interleaver is to spread out the burst-errors into bit-errors [12], [13]. Based on the results in Fig. 4, in this study, we choose the BCH (25, 20) code with a block interleaving depth $\lambda = 80$ bits.

C. Conditional Retransmission Based on Concealment Error and Channel Condition

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. Since the same error concealment is implemented in the encoder to prevent the mismatch between the encoder and the decoder, the concealment error when a packet is lost can be calculated at the encoder. The following analysis based on a rate-distortion (R-D) framework gives some insights.

If an error packet is not resent, the concealment error based on the mean squared error (MSE) caused by replacing with the content from the previous frame is

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

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, $f(x,\,y)$ 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.

Resending the error packet may avoid the above concealment error provided that the retransmitted packets 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 coding error and the bit-allocation at the frame-level if the optimum quantization scheme in [10] is adopted

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

where K is a model parameter, which can be approximated by $e/ln2$ if the DCT coefficients are Laplacian distributed and independent [10], D^* is the mean squared error of the coded frame, $R = B_{\mathrm{frame}} - ANC$, 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 weight of the nth motion-compensated residual macroblock, respectively.

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

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

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

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

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

Based on this analysis, we can use D_{CE} and D_{retx} to decide whether to resend the lost packet. However, evaluating D_{retx} will require significantly more computation and the estimation of the packet loss probability, which makes the method complicated. 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*$_j \in$ *GOB*$_i$) and (*NAKPacket*$_{j-1} \in$ *GOB*$_i$) {
Decision_as_previous_packet;}
else {If ($D_{\mathrm{CE}} < T$) 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

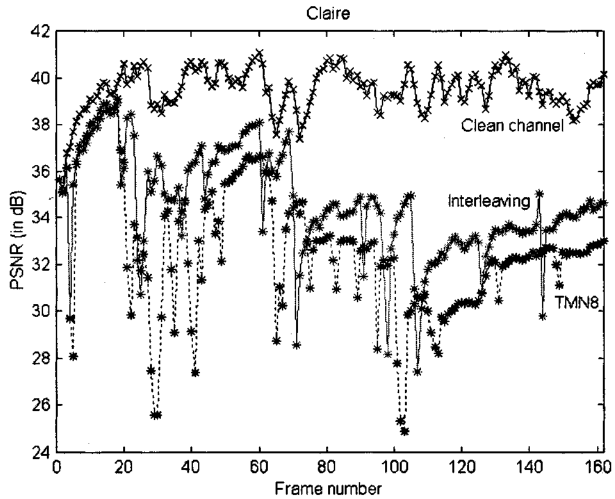

Fig. 5. PSNR comparison for the "Claire" sequence between TMN-8 in clean channel (-x-line), TMN-8 with packet loss and concealment (dashed -*-line), and our proposed interleaving scheme (solid -*-line). TMN-8 results in 12 frames skipped, while our proposed scheme has no frames skipped.

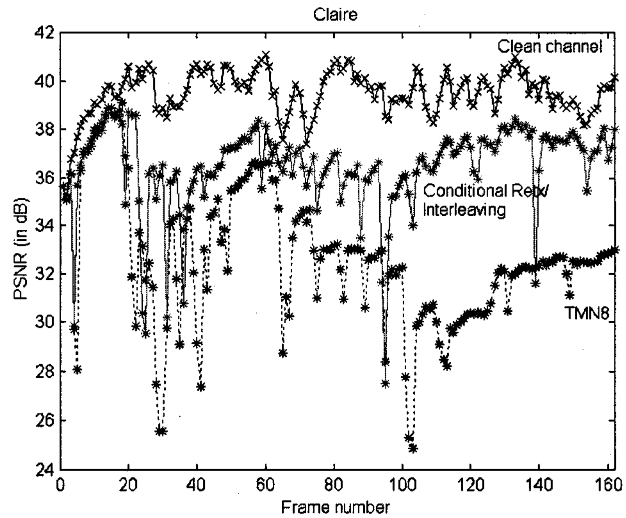

Fig. 7. PSNR comparison for the "Claire" sequence between TMN-8 in clean channel (-x-line), TMN-8 with packet loss and concealment (dashed -*-line), and our proposed conditional retransmission and interleaving scheme (solid -*-line). TMN-8 results in 12 frames skipped, while our proposed scheme has no frames skipped.

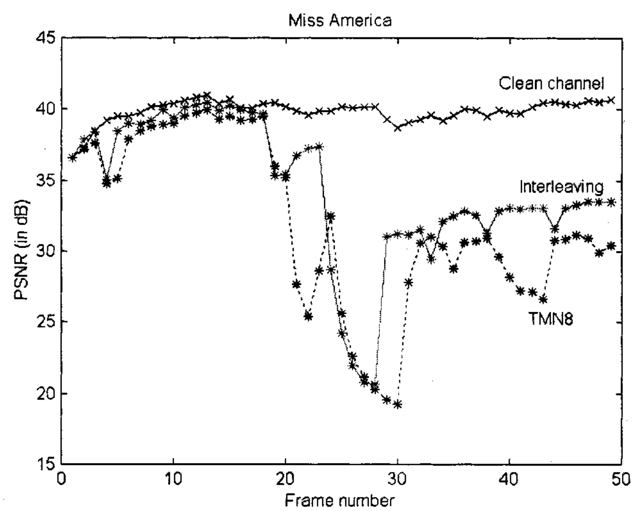

Fig. 6. PSNR comparison for the "Miss America" sequence between TMN-8 in clean channel (-x-line), TMN-8 with packet loss and concealment (dashed -*-line), and our proposed interleaving scheme (solid -*-line). TMN-8 results in 7 frames skipped, while our proposed scheme has no frames skipped.

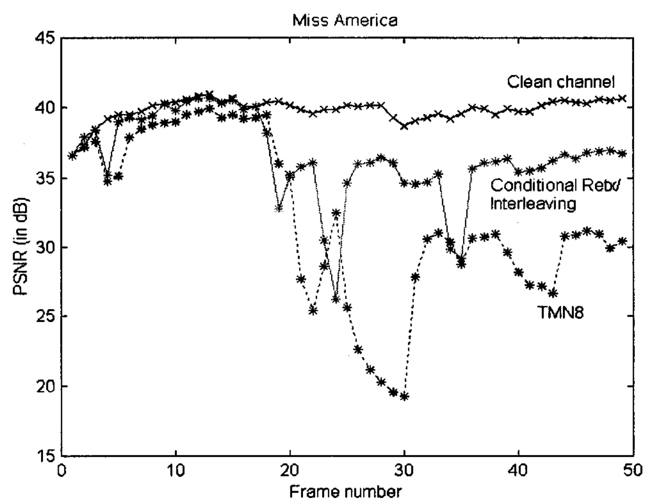

Fig. 8. PSNR comparison for the "Miss America" sequence between TMN-8 in clean channel (-x-line), TMN-8 with packet loss and concealment (dashed -*-line), and our proposed conditional retransmission and interleaving scheme (solid -*-line). TMN-8 results in 7 frames skipped, while our proposed scheme has no frames skipped.

$D_{\rm CE}$ 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. However, the error concealment will inevitably lead to poorer video quality when compared to error-free case. Furthermore, using the error-concealed video to predict the following frames will also reduce the coding efficiency, thereby leading to further quality penalty. Therefore, in the case where $D_{\rm CE}$ is less than T and the channel condition is good, the packets are still worth retransmitting because there is a high probability that the retransmitted packets will be received correctly. Our experimental results show that the retransmission of these packets may result in up to 3-dB improvement in average PSNR compared to performing error concealment on the lost packets without retransmission.

III. Simulation Results

Test video sequences including "Claire," "Carphone," "Miss America," and "Suzie" in the QCIF format (176 $\times$ 144 pixels/frame) were encoded at 32 kb/s with a target frame-rate of 10 frames/s using TMN-8 and our proposed scheme.

Figs. 5 and 6 show the results of our proposed interleaving scheme for "Claire" and "Miss America," respectively. Our scheme shows an improvement of about 2 dB compared to TMN-8 under the same condition. Figs. 7 and 8 show the simulation results for "Claire" and "Miss America" sequences, respectively, with the PSNR comparisons among TMN-8 with

(a) (b)

Fig. 9. Subjective evaluation for 32 kbits/s “Miss America” sequence. Reduce number of packet errors from interleaving with PSNR improvement of 7.5 dB for: (a) frame #31. (a) TMN-8 and (b) proposed conditional retransmission and interleaving scheme.

(a) (b)

Fig. 10. Subjective evaluation for 32 kb/s “Carphone” sequence. Reduce number of packet errors from interleaving with PSNR improvement of 4.5 dB for frame #21. (a) TMN-8 and (b) proposed conditional retransmission and interleaving scheme.

(a) (b)

Fig. 11. Subjective evaluation for 32 kb/s “Miss America” sequence. Saved bits from conditional retransmission helps with better bit allocation with PSNR improvement of 6 dB for frame #44. (a) TMN-8 shown in and proposed conditional retransmission and (b) interleaving scheme.

clean channel, TMN-8 with packet-errors and concealment, and our proposed low-delay interleaving and conditional retransmission scheme. Our overall scheme shows an improvement of about 4 dB for both sequences compared to TMN-8. Table III shows the average channel throughput and PSNR comparison for all video sequences tested. The throughput is the number of data bits actually transmitted which does not include the BCH overhead. The PSNR improvement is up to about 3 dB for our proposed interleaving scheme and up to about 5.5 dB for our conditional retransmission and interleaving schemes. Figs. 9–12 give some subjective evaluations of the video quality for “Miss America” and “Carphone” sequences of our

(a) (b)

Fig. 12. Subjective evaluation for 32 kb/s "Carphone" sequence. Saved bits from conditional retransmission helps with better bit allocation with PSNR improvement of 4 dB for frame #47. (a) TMN-8 and (b) with proposed conditional retransmission and interleaving scheme.

TABLE III
COMPARISON OF THE AVERAGE THROUGHPUT AND PSNR FOR TMN-8 UNDER CLEAN CHANNEL, CHANNEL ERRORS, AND USING OUR PROPOSED SCHEME

Video Sequence	Total Frames	Clean Channel		TMN-8 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

proposed scheme compared to TMN-8, where frames with significant PSNR improvement in video sequences are shown. Figs. 9 and 10 show the effects of packet errors after error concealment. The improvement is mainly due to the reduced packet error. Figs. 11 and 12 show the better video quality with our proposed scheme for a frame without packet errors. The improvement is due to the saved bits from the conditional retransmission scheme that can help improve the coding quality of video frames.

IV. CONCLUSION

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 5 dB for our scheme compared to H.263 TMN-8. Subjective evaluations also confirm the significant video quality improvement. For future development, several channel- and source-coding techniques can be applied to further improve the performance of our proposed scheme. The error packets received at the decoder can be combined to increase the probability of successfully recovery [19]. An ARQ system with incremental redundancy where the transmitter sends more parity when requested by the receiver can be used [21], [22]. Some feedback-based drift control techniques, such as error tracking [23], can be combined with our method to alleviate the drift due to unsuccessful retransmission attempts when the delay of NAK from the decoder is long. More sophisticated error-concealment techniques, such as those described in [24]–[27], could be used as well to improve the concealment error. Content-based conditional retransmission scheme can also be applied such that if an error packet belongs to the foreground object, it will be retransmitted, while if an error packet belongs to the background object, we will rely on error concealment. Through the investigation of channel model, joint source and channel coding, and improved retransmission strategy, we could design a better wireless video transport system based upon the states of the channel.

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