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Unequal Loss Protection Using Layer-Based Recovery Rate (ULP-LRR) for Robust Scalable Video Streaming over Wireless Networks

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Abstract

Scalable video streaming over wireless networks has many challenges. The most significant challenge is related to packet loss. To overcome this problem, in this paper, we propose an unequal loss protection (ULP) method using a new forward error correction (FEC) mechanism for robust scalable video streaming over wireless networks. For an efficient FEC assignment considering video quality, we first introduce a simple and efficient performance metric, the layer-based recovery rate (LRR), for quantifying the unequal error propagation effects of the temporal and quality layers on the basis of packet losses. LRR is based on the unequal importance in both the temporal and the quality layers of a hierarchical scalable video coding structure. Then, the proposed ULP-LRR method assigns an appropriate number of FEC packets on the basis of the LRR to protect the video layers against packet lossy network environments. Compared with conventional ULP algorithms, the proposed ULP-LRR algorithm demonstrates a higher performance for various error-prone wireless channel statuses.

Index Terms: Forward error correction, Layer-based recovery rate, Packet loss, Scalable video, Unequal loss protection

I. INTRODUCTION

Recently, a variety of multimedia services like video streaming over wireless networks have emerged [1]. However, consecutive packet losses and rapid bandwidth changes in wireless environments cause serious video quality degradation.

The most promising mechanism for the wireless transmission of H.264 video is scalable video coding (SVC), which is a highly attractive solution to the problems posed by the properties of wireless video transmission. A scalable video can provide spatial, temporal, and quality scalabilities without any additional computational load. A scalable bitstream allows different video qualities and resolutions for different video receivers according to the characteristics of the transmission channel or the decoding capability of the receiver. It is possible to transmit and decode partial bit streams to support video services with lower temporal or spatial resolutions or reduced fidelity while retaining a reconstruction quality that is high relative to the rate of the partial bit streams. Hence, H.264/SVC provides functionalities such as graceful degradation in lossy transmission environments as well as the possibility for bit rate adaptations [2, 3]. However, due to the spatial and temporal dependencies among scalable video layers, consecutive packet losses may lead to a serious degradation

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of video quality during network communication. In general, unequal loss protection (ULP) algorithms are utilized for minimizing such video quality degradation. Meanwhile, the effect on the video quality degradation, according to the hierarchical coding structure and the dependencies among layers, is used as a weighting factor in various ULP algorithms.

There have been some studies on forward error correction (FEC) algorithms, focusing on the error propagation effect on the video quality degradation in both the base and the enhancement layers [4-12]. In [4], an FEC assignment algorithm using the property of consecutive packet losses is presented. Various ULP algorithms based on the rate distortion model have been proposed for reducing the video quality degradation due to packet losses [8-12]. Adaptive FEC assignment algorithms have been developed for video streaming over the best-effort networks [9-13]. To adapt to time-varying channel environments, optimal packet scheduling for rate distortion has been proposed [10-14]. In addition, a rate distortion-optimized scheduling algorithm has been developed to adapt the streaming video to time-varying channels for minimizing the total distortion under the transmission rate constraint [15, 16].

In this paper, we propose a new ULP algorithm based on the layer-based recovery rate (LRR) to minimize the video quality degradation from packet losses. The proposed algorithm consists of two stages. In the first stage, a new performance metric, the LRR, uses the compression structure of scalable video and the packet losses in the transmission channel. Hierarchical prediction structure-based scalable video coding utilizes the information of the base layer for encoding the enhancement layers. The LRR in an enhancement layer is dependent on that in the base layer. In the second stage, we propose an FEC assignment algorithm to maximize the LRR. The scheme proposed in this paper uses a low-complexity local hill-climbing search method. The simulation results show that the proposed ULP method improves the peak signal-to-noise ratio (PSNR) performance as compared to other previous schemes.

The remainder of this paper is organized as follows: Section II describes the system model. In Section III, we introduce the ULP scheme using the ULP-LRR for scalable video in a GOP in wireless networks. In Section IV, we evaluate the performance of the proposed algorithm. Finally, Section V concludes this paper.

II. SYSTEM MODEL

The overall structure of the proposed ULP algorithm is illustrated in Fig. 1. First, the scalable video encoder generates video packets on the basis of a hierarchical structure with spatial, temporal, and quality scalabilities.

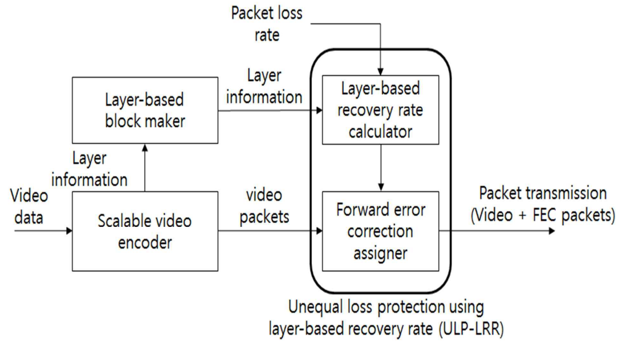


Fig. 1. Overall structure diagram of the proposed ULP algorithm.

The spatial layers provide the coarse grain scalability (CGS). Motion-compensated prediction is utilized for the base layer. Based on the motion and residual information from the base layer, inter-layer prediction is adopted to reduce redundancies between enhancement layers [17, 18].

In a layer-based block maker, all video packets in the encoded bit-stream are divided into several blocks on the basis of the layer information. The number of temporal layers is T, and each temporal layer is further divided into Q quality layers. If we denote t as the temporal quality level and q as the quality level, where $t = 0,1,\cdots T-1$ and $q = 0,1,2,\cdots Q-1$, we can define the layer-based scalable video data for unit (t,q) as $L(t,q)$.

Due to the dependency between the base and the enhancement layers, the effect of packet loss on the SVC bit stream can cause severe video quality degradation in error-prone channel environments. FEC has been utilized for protecting video packets against packet losses by using parity packets. If the number of parity packets is higher than the number of erased video packets, the source video packets can be retrieved. In this study, we utilized the well-known Reed–Solomon (RS) codes [4-8] for the FEC generation.

Based on the dependency in the quality and temporal layers, we propose an unequal loss protection mechanism by using ULP-LRR to provide the best video quality with the transmission rate constraint. As shown in Fig. 1, the ULP-LRR mechanism is composed of two major components, namely the LRR calculator and the adaptive FEC packet assigner. The LRR calculator determines the recovery rate of each layer, $L(t,q)$, in accordance with both the layer-based motion-compensated prediction model and the feedback packet loss rate using real-time transport control protocol (RTCP) [19]. Then, the FEC packet assigner allocates the optimal redundant rate for each layer by using the analytical model of obtaining the recovery rate from the packet loss, LRR.

III. ULP-LRR ALGORITHM

A. Performance Metric

As described above, the proposed ULP-LRR mechanism determines the appropriate degree of redundancy for each scalable layer, $L(t,q)$, utilizing an analytical model of the LRR. In the FEC recovery process, $k(t,q)$ source packets in $L(t,q)$ are encoded on the sender side into $N(t,q)$ packets. The original source packets can be successfully recovered at the receiver side if the $K(t,q)$ packets are received, including the source packets and the redundant packets. Therefore, the LRR can be expressed as follows:

$$\mathrm{LRR}_l(t,q) = \sum_{i=K(t,q)}^{N(t,q)} C_i^{N(t,q)} \cdot (1-p_e)^i \cdot {p_e}^{N(t,q)-i}, \quad (1)$$

where p_e and $C_i^{N(t,q)}$ represent the packet loss rate and all possible combinations of i packets successfully received in the whole block, respectively. For quantifying the recovery rate of each layer, $L(t,q)$, by using (1), we consider the hierarchical prediction structure in the temporal and quality layers.

Fig. 2 shows the prediction structure of different temporal and quality layers in the scalable video coding structure. In the temporal layer, $L(t,q)$ refers to both $L(t-1,q)$ and $L(t,q-1)$. In the same GOP, $L(t,q)$ is decodable if $L(t-1,q)$ and $L(t,q-1)$ are decodable or not. Accordingly, the LRR of $L(t,q)$ is directly determined by $L(t-1,q)$ and $L(t,q-1)$. For example, $LRR_l(2,1)$ is affected by both $LRR_l(1,1)$ of the lower temporal layer in the same quality layer and $LRR_l(2,0)$ in the lower quality layer of the current temporal layer. Due to the hierarchical prediction structure between the temporal and the quality layers, $L(t,q)$ is influenced by the lower scalable layers with the non-zero temporal layer index or quality layer index. Based on (1), the proposed LRR of $LRR_l(t,q)$ can be expressed as follows:

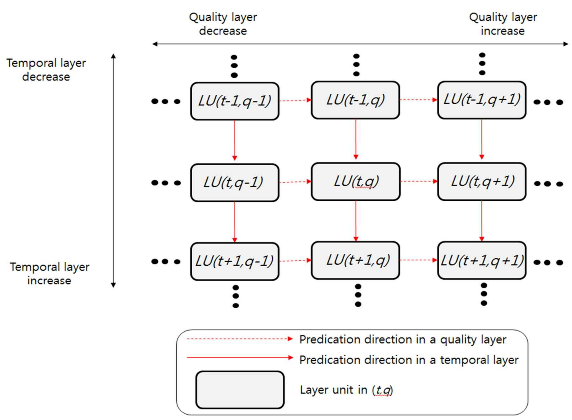


Fig. 2. Prediction structure of different temporal and quality layers in the scalable video coding structure.

$$\widehat{LRR}_l(t,q) = LRR_l(t-1,q) \cdot LRR_l(t,q-1). \quad (2)$$

Based on (2), the average LRR in a GOP is defined as LRR_{gop} and can be calculated as follows:

$$LRR_{gop} = \frac{\sum_{t=0}^{T-1}\sum_{q=0}^{Q-1}\widehat{LRR}_l(t,q)}{T \cdot Q}, \quad (3)$$

where T and Q represent the maximum number of temporal and quality layers, respectively.

B. FEC Assignment using ULP-LRR

Based on the performance metric LRR_{gop} in (3), an FEC assignment algorithm for minimizing the video quality degradation from packet losses is proposed by adjusting the number of FEC packets for each layer. When the FEC packets are allocated to the video packet $L(t,q)$, the expected amount of error propagation reduction is defined as $E(t,q)$ and is calculated using the number of FEC packets as follows:

$$E(\mathrm{t,q}) = LRR_{gop}(F(t,q)+1) - LRR_{gop}(F(t,q)), \quad (4)$$

where $F(t,q)$ represents the number of FEC packets in $L(t,q)$. Using (4), the largest error propagation reduction of $L(t,q)$ is selected to be $L(\hat{t},\hat{q})$. The number of FEC packets for the selected $L(\hat{t},\hat{q})$ is increased by 1. The accumulated $F(t,q)$ is represented as $S(t,q)$. If the total number of assigned FEC packets $S(t,q)$ exceeds the given total number of FEC packets (T_{tot}) under the limited channel resource, the proposed algorithm is terminated.

IV. SIMULATION AND RESULTS

In this section, we present the numerical results of the proposed FEC assignment algorithm for the performance analysis. We use QCIF video sequences of *Foreman* and *Mobile* with a frame rate of 15 fps. The number of frames is 81, and the GOP size is 16 frames. Video sequences are encoded using version 9 of the joint scalable video model [20]. There are five temporal layers with one quality base layer and two quality enhancement layers. The quantization parameters (QPs) for each layer are set as 40, 30, and 25. The two-state Markov channel model described in [21] is used for modeling the packet losses, assuming that the average burst length L_b is 2 and P_b is the average packet loss rate.

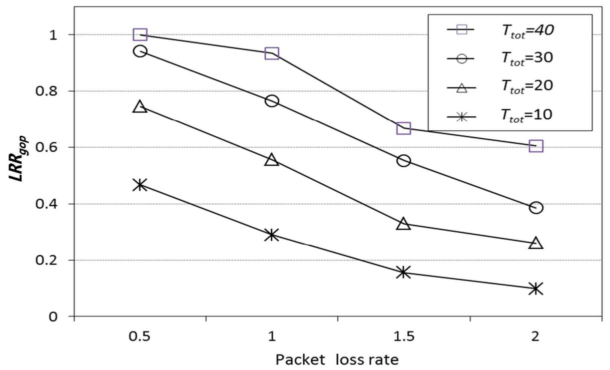


Fig. 3. Variation in the layer-based recovery rate (LRR_{gop}) according to the total number of FEC packets (T_{tot}) versus the packet loss rate.

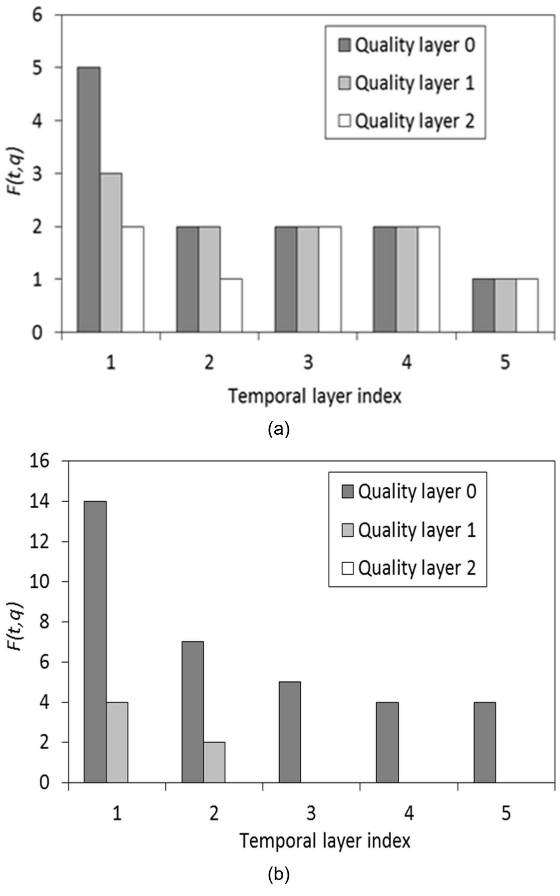


Fig. 4. Distribution of $F(t,q)$ for each temporal (t) and quality layer (q) for different packet loss rates: (a) $P_b = 5\%$ and (b) $P_b = 15\%$.

Fig. 3 shows that the variation of LRR_{gop} in (3) gradually increases with an increase in the total number of FEC packets (T_{tot}). With an increase in the packet loss rate, the average layer-based recovery rate gradually decreases. Further, with an increase in the number of FEC packets, the value of LRR_{gop} also increases.

Fig. 4 shows the distribution of $F(t,q)$ for each temporal layer t and quality layer q at different packet loss rates P_b= 5% and P_b= 15%, respectively. Redundant FEC packets are intensively assigned to the lower layer index. Due to the hierarchical prediction structure in the scalable video, packet losses in the lower layer have more serious error propagation effects than the packet losses in the higher layers. For different packet loss rates (P_b = 2%–20%), the proposed ULP-LRR method (3) is compared with another unequal loss protection scheme (ULP-RD) using the rate distortion information [11].

Fig. 5 shows the PSNR comparisons of ULP-LRR, ULP-RD, and Equal FEC. Equal FEC shows the lowest PSNR values for all packet loss rates. The ULP-LRR and ULP-RD methods have similar performance in the case of a low packet loss rate. Because the effect of a low packet loss rate does not significantly impact performance, these methods have similar PSNR values. On the other hand, the ULP-LRR algorithm shows better performance than ULP-RD in the case of a high packet loss rate. The proposed ULP-LRR is more effectively adapted for various channel statuses than the ULP-RD methods.

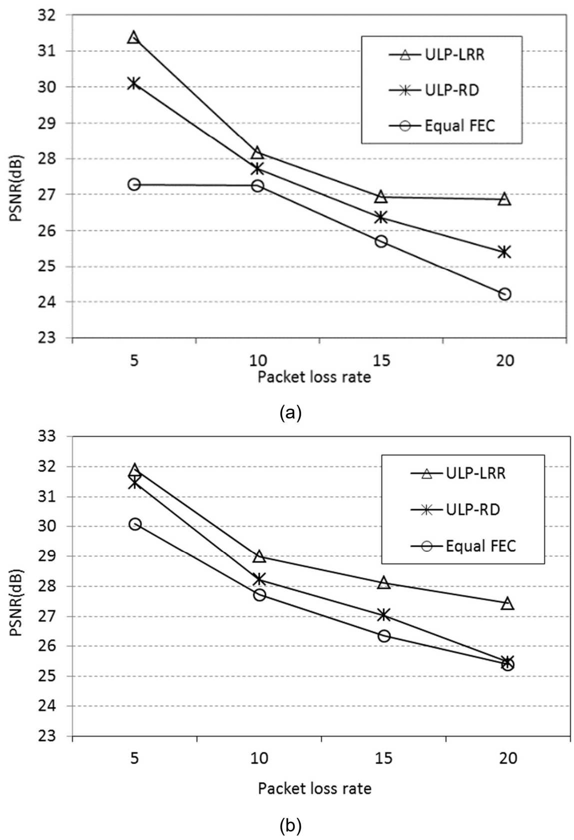
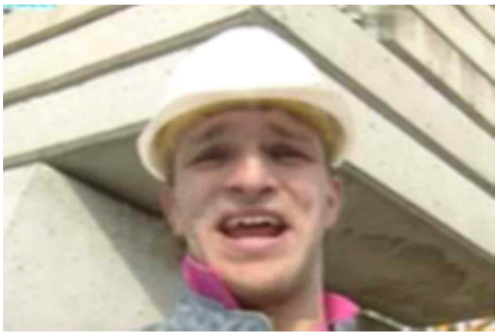
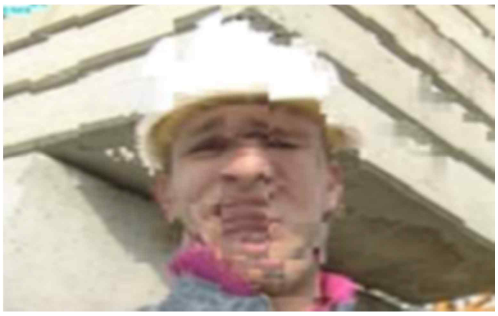


Fig. 5. Peak signal-to-noise ratio (PSNR) comparisons of unequal loss protection using layer based recovery rate (ULP-LRR), unequal loss protection using rate distortion (ULP-RD), and equal forward error correction (FEC) for various packet loss rates: (a) *Foreman* and (b) *Mobile*.



(a)

(b)



(c)

Fig. 6. Snapshots of the 28th reconstructed frame of the *Foreman* test sequence using different unequal loss protection (ULP) schemes: (a) original, (b) unequal loss protection using layer-based recovery rate (ULP-LRR), and (c) unequal loss protection using rate distortion (ULP-RD).

ULP-LRR results in higher PSNR by 1–3 dB under time-varying packet loss rate conditions than the ULP-RD method. These improvements are achieved through an efficient reduction of the error propagation effects from packet losses by adaptively adjusting the level of the LRR for each scalable layer in the time-varying wireless channel status.

Fig. 6 shows snapshots of the 28th reconstructed frame of the *Foreman* video sequence. The visual quality of the reconstructed frame for the proposed ULP-LRR is better than that obtained using the ULP-RD scheme. The improvement of visual quality is achieved by exploiting the LRR in the proposed ULP-LRR method considering unequal error propagation effects.

V. CONCLUSION

In this paper, we proposed an unequal loss protection algorithm based on a new FEC assignment method to improve the scalable video streaming quality in packet-lossy wireless environments. We first proposed a low-complexity performance metric, the LRR, by using the hierarchical coding structure of the scalable video and the packet loss rate. Then, we developed the FEC assignment algorithm to maximize the value of LRR. The simulation results showed that the proposed algorithm outperformed the previous ULP algorithms by 1–3 dB under variable channel conditions. Furthermore, the proposed FEC algorithm is expected to be useful for various video streaming services in wireless network environments.

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