

## Research Article

# Content-Adaptive Packetization and Streaming of Wavelet Video over IP Networks

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This paper presents a framework of content-adaptive packetization scheme for streaming of 3D wavelet-based video content over lossy IP networks. The tradeoff between rate and distortion is controlled by jointly adapting scalable source coding rate and level of forward error correction (FEC) protection. A content dependent packetization mechanism with data-interleaving and Reed-Solomon protection for wavelet-based video codecs is proposed to provide unequal error protection. This paper also tries to answer an important question for scalable video streaming systems: given extra bandwidth, should one increase the level of channel protection for the most important packets, or transmit more scalable source data? Experimental results show that the proposed framework achieves good balance between quality of the received video and level of error protection under bandwidth-varying lossy IP networks.

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## 1. INTRODUCTION

There is a growing demand for video transmission over heterogeneous networks for communication and entertainment applications. Scalable video coding (SVC) techniques are often proposed for such systems since, ideally, a video sequence can be encoded once and adapted on the fly to different frame rate, bitrate, and resolution for different applications. Although scalable video is an interesting concept, it takes complete end-to-end system design to show the advantage of SVC over single-layer coding techniques. With single-layer coding, techniques like bitstream switching and simulcasting can be used to achieve video adaptations. However, it is easier to achieve good rate versus source-and-channel distortion tradeoff with scalable coding techniques.

The mainstream video compression techniques are based on hybrid motion-compensated transform coding approach, where the transform algorithms are typically either discrete cosine transform (DCT) or 3D wavelet transform [1]. So far, DCT-based SVC approaches have demonstrated better coding efficiency than wavelet-based SVC techniques [2], especially for low bitrate applications. However, a wavelet-based SVC framework can provide fine-granularity bitrate (i.e., SNR) scalability with less system complexity than that of an FGS-based DCT framework. In addition, many ongoing

efforts show that wavelet-based SVC approaches still have room for improvement [3]. Therefore, in this paper, wavelet-based SVC is used as the core codec for the development of a scalable video streaming framework.

The most challenging problem for scalable video streaming over IP networks is about how to optimally adapt source data rate and degree of packet loss protection to real-time network conditions. Video packet packetization and scheduling algorithms are mostly responsible for mitigating the effects of bandwidth variation and packet losses in the network. The packetization and scheduling algorithms are mainly based on resource-versus-distortion optimization [4–7], where resource can be available computation power, rate, delay, and so forth. A general resource allocation treatment for streaming systems is presented in [5]. Some researchers try to apply the rate-distortion optimization (RDO) principle [8] of source coding theories to video streaming over lossy networks [4]. For a streaming system, the distortion is a result from both source coding and channel losses. A key issue in an RDO-based streaming system is that the distortion due to packet losses is much more difficult to quantify than the distortion due to lossy source coding.

Several frameworks for 3D wavelet based video streaming system have been proposed in the literature recently. Chu and Xiong [9] introduced a combined packetized wavelet video

coding and FEC approach for video streaming and multicast. The packetized wavelet video coder marks the truncation points of the bit stream at the nearest packet boundaries (instead of the end of each fractional bit plane). In the FEC-based error protection scheme, it applies Reed-Solomon (RS) coding to produce parity packets. And then the scheme broadcasts all source packets to one multicast group and parity packets to different multicast groups. Hence, for each client, the optimal number of layers and error protection to subscribe to can be determined by the packet loss ratio and the available channel bandwidth. However, data interleaving is not used in this work, which makes the system less robust to burst errors. Dong and Zheng [10] proposed a content-based retransmission framework for wavelet video streaming. The compression module adopts dynamical grouping and bounded coding scheme for improving compression efficiency and removing unnecessary dependency to each coefficient subband. In the transmission module, a video packet includes one or more subbands, and a content-based retransmission is used to provide robustness against transmission errors. The content-based retransmission scheme is based on the importance of packet content which is computed by the square sum of coefficients for each wavelet subband. Later, Zhao et al. [11] incorporated an error concealment scheme into this content-based retransmission framework to increase its error resilience capability. Nevertheless, retransmission-based error control requires longer jitter buffer and may consume too much extra bandwidth in high error rate channels [12, 13].

Chou and Miao [4] developed a framework for RDO streaming of packetized media. The RDO framework is flexible to extend the optimizing packet transmission scheduling to a wide range of receiver/sender/proxy driven streaming systems [14]. However, the scheme maps (probability of) packet losses into rate increment of redundant packet forward transmission (ARQ can be avoided in this approach). However, although redundant packet transmission makes the RDO system simpler for analysis, it is not cost-effective for practical systems. R-D performance can be greatly improved if FEC is used instead. Zhu et al. [6] proposed a congestion-distortion optimized scheme. Zhai et al. [7] presented an integrated joint source-channel coding framework for video streaming. Wang et al. [15] proposed a cost-distortion optimization framework. Chang et al. also proposed sender-based [16] and receiver-based [17] RDO frameworks for 3D wavelet video streaming, which basically follow the framework introduced by Chou and Miao. The proposed system uses source rate-distortion profiles to optimize for playout latency and bandwidth allocation among a group of data packets in a way that minimizes distortion in the reconstructed frames.

There are many error control schemes for video streaming, including forward error correction (FEC) [18–21], unequal error protection (UEP) [22–24], and automatic retransmission request (ARQ) [25]. Until recently, error control schemes for streaming systems are designed independently to rate control schemes. Joint design of error and rate control is important to a variable bandwidth lossy network.

For example, when the channel bandwidth increases during runtime, should more bits be allocated to send extra (enhancement) source data, or to increase the level of protection of crucial (also known as base layer) source data? Based on the RDO principle, one should pick whichever approach that reduces more distortion. However, this is not trivial since distortions from channel losses are nondeterministic. Another issue is that not all source data bits carry equal amount of information (i.e., entropy). Although some of the error control techniques try to put different degree of protection based on the degree of importance of the content, unequal error protection is done coarsely since the error control scheme is based on either single-layer video coding model or coarse-granularity layered scalable video coding mode.

In this paper, a content-adaptive packetization scheme for wavelet-based streaming video is proposed. The mechanism is based on detail analysis of the mainstream wavelet-based video codec [26]. Due to its fine-granularity SNR scalability feature, the proposed packetization scheme can apply various degrees of Reed-Solomon (RS) codes on interleaved video subband data so that the streaming video is very robust over IP networks. In addition, the paper proposes to map the distortion caused by packet loss to distortion caused by source data rate reduction due to extra FEC protection (for error-free transmission). Since measuring operational video distortion from packet loss is very difficult while measuring source coding distortion is much simpler, the proposed mechanism can be applied to practical systems. In summary, the main features of the proposed system are highlighted as follows.

- (1) The streaming algorithm searches along the R-D curve for an optimal operating point between the scalable source coding rate and the FEC protection level.
- (2) The FEC protection level is also influenced by runtime packet loss rate feedback from the client. Therefore, it is adaptive to both the video content entropy and the run-time packet loss rate.
- (3) The rate-distortion tradeoff of the system takes into account both distortion due to source data rate reduction and distortion due to packet losses (predicted by FEC protection bits required for error-free transmission).

The rest of this paper is organized as follows. Section 2 presents a detail analysis on the wavelet compressed video bit stream and its characteristics for content-adaptive protection. The detail of the proposed packetization scheme and streaming framework is described in Section 3. Some experimental results of the proposed system are shown in Section 4. Finally, some conclusions and discussions are given in Section 5.

## 2. INVESTIGATION OF WAVELET VIDEO BIT STREAMS WITH DATA LOSSES

For streaming applications, the quality of video is affected by packet losses. One of the most difficult problems for RDO streaming is about how to measure the distortion caused by

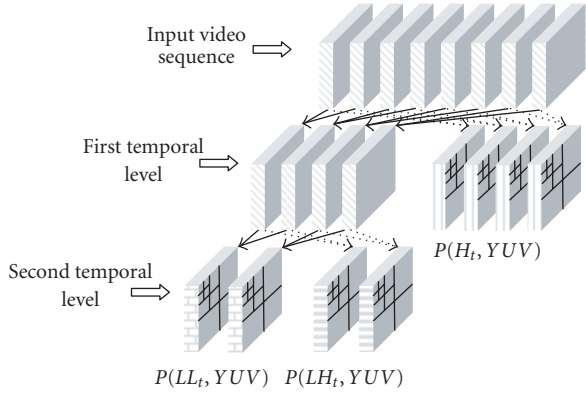


FIGURE 1: Wavelet video coding block diagram.

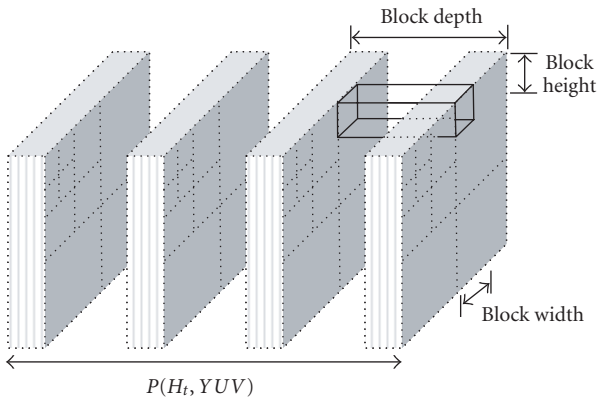


FIGURE 2: Examples of coding block in wavelet video coding.

packet losses. The distortion depends heavily on the source coding method. In this section, the wavelet video coding schemes presented in [26, 27] are investigated in detail. In particular, some experiments are conducted to exhibit the impact of different wavelet subband data losses on the reconstructed video quality.

The block diagram of a wavelet-based video coding system is shown in Figure 1. In a  $T + 2D$  wavelet coder, an input video sequence is temporally decomposed first using motion-compensated temporal filtering (MCTF) [1]. The output of MCTF is then further decomposed by a 2D spatial wavelet transform on a frame-by-frame basis. For example, two-level temporal decomposition results in three temporal subbands, namely,  $P(H_t, YUV)$ ,  $P(LH_t, YUV)$ , and  $P(LL_t, YUV)$ . When the group of pictures (GOPs) size is eight, a typical set of transformed subband data produced by the  $T + 2D$  wavelet coder has four  $P(H_t, YUV)$  frames, two  $P(LH_t, YUV)$  frames, and two  $P(LL_t, YUV)$  frames. Each frame contains one luminance component ( $Y$ ) and two chrominance components ( $U$  and  $V$ ). The coefficients of different subbands are logically segmented into coding blocks, based on the structure of Figure 2, and each coding block is independently coded by an entropy coder. For instance, a coding block size in Figure 2 has block depth

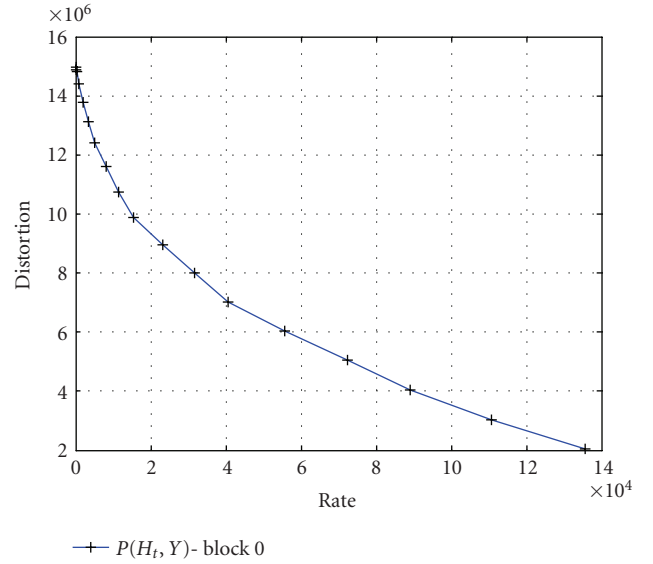


FIGURE 3: The R-D curve of coding block 0 of subband  $P(H_t, Y)$  of STEFAN.

2 (i.e., two frames), block height 36 ( $=288/2^3$ ), and block width 44 ( $=352/2^3$ ). Common entropy coding techniques for wavelet video are 3D embedded subband coding with optimized truncation (3D-ESCOT) [27] and 3D set partitioning in hierarchical trees (3D-SPIHT) [28]. The 3D-ESCOT algorithm has higher compression efficiency and better scalability than the 3D-SPIHT algorithm. Therefore, the proposed scheme is based on 3D-ESCOT coding technique.

During the 3D-ESCOT entropy coding process, the entropy coder (fractional bit plane coding and context-based arithmetic coding) operates one coding block at a time, and each coding block consists of  $N$  total bit planes, where  $N$  is the number of bits in the most significant coefficients. Three encoding operations of the context-based arithmetic coding (zero coding, sign coding, and magnitude refinement) are used to characterize the significance of coefficients in a bit plane. Following the 3D context modeling, fractional bit plane coding ensures that the bit stream is arranged with fine granularity of SNR scalability for each coding block. The fractional bit plane coding procedure consists of three distinct passes which are the significant propagation pass, the magnitude refinement pass, and the normalization pass. Since the first bit plane of a coding block can only be processed with the normalization pass, a coding block contains  $3N - 2$  coding passes. After entropy coding, candidate truncation points of a coding block are associated with rate-distortion slopes (R-D slopes). Any truncation points that are not on the convex hull are eliminated, and the R-D slopes are  $\lambda_0, \lambda_1, \dots, \lambda_{3N-2}$ , where  $|\lambda_0| > |\lambda_1| > \dots > |\lambda_{3N-2}|$ . All coding blocks have R-D curves similar to the example shown in Figure 3, and the top coding passes contain the most important video data. Therefore, higher level of protection is required for top bit plane coding passes.

In order to gain better insight into the significance of different bit stream segments across different temporal



FIGURE 4: Reconstructed video when a chunk of TSB data is lost. The loss occurs in coding block 0 of SSB 0 for the TSB in (a)–(d), and coding block 0 of SSB 18 for the TSB in (e)–(f).

subbands, some experiments are conducted. For example, using a four-level MCTF temporal decomposition, a group of frames is temporally decomposed into the  $LLLL$ ,  $LLLH$ ,  $LLH$ ,  $LH$ , and  $H$  subbands. In addition, each temporal subband may further be spatially decomposed. For an encoded video with four-level temporal and three-level spatial decompositions, each temporal subband (TSB) is split into nineteen spatial subbands (SSB) indexed from 0 to 18. The distortion impact of the first coding block within a higher spatio-temporal subband (e.g., Figures 4(b), 4(c), 4(d)) is indeed more sensitive than that of the last coding block within a lower spatio-temporal subband (e.g., Figure 4(e)).

In practice, given an estimated packet loss rate, different amount of error protection should be applied to different portions of a coding block based on their influence on visual quality. Therefore, further “rate” versus “channel-distortion” analyses of wavelet subband data are conducted as follows. Since the size of different coding blocks varies (see Figure 5), it is not suitable to use coding block as the data interleaving unit for FEC protection. A coding block should be split into several smaller units for data interleaving. Within each coding block, the bit stream size of the first coding pass is usually small (see Figure 6), but it has major impact on video quality (see Figure 7). To evaluate the effect of degradation from burst data loss, a 10% burst loss of bits is placed in different portions of a coding block (see Figure 8). When the burst data loss is located at the beginning of a coding block, it usually causes large degradation of visual quality. Hence, the error protection level for different portions of a coding block should be different.

Packet loss is the major cause of nondeterministic distortion for video streaming applications. For example, over fiber networks, bit errors rarely occur. The bit error rate of

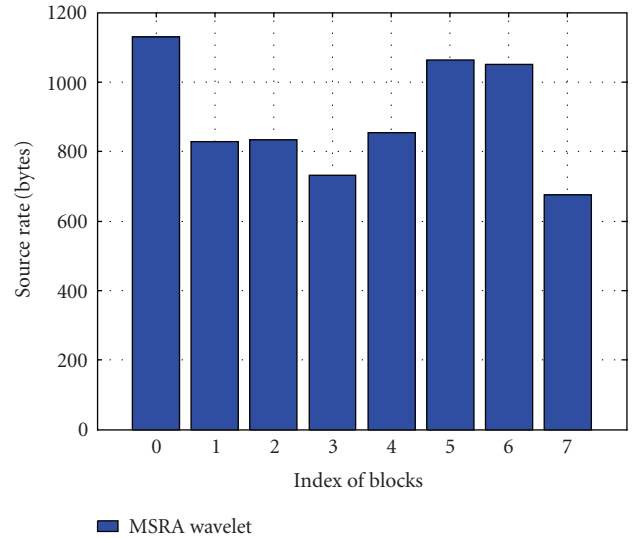


FIGURE 5: Source data rate in SSB 0 of subband  $P(H_t, Y)$  of STEFAN.

fiber networks is only  $10^{-9}$  [29]. The main reasons for packet losses are mostly because of network congestion, which causes packet losses in the network router queue buffer [30]. As Fang et al. [29] and Biersack [30] pointed out, FEC protection scheme is effective to recover packet loss with minimum transmission overhead for multimedia streaming. Hence, in this paper, a content-adaptive FEC protection scheme for scalable streaming systems is proposed based on previous investigation of channel distortion impact on wavelet video.

The basic concept of our context-adaptive FEC streaming scheme is to add different FEC protection level (subject

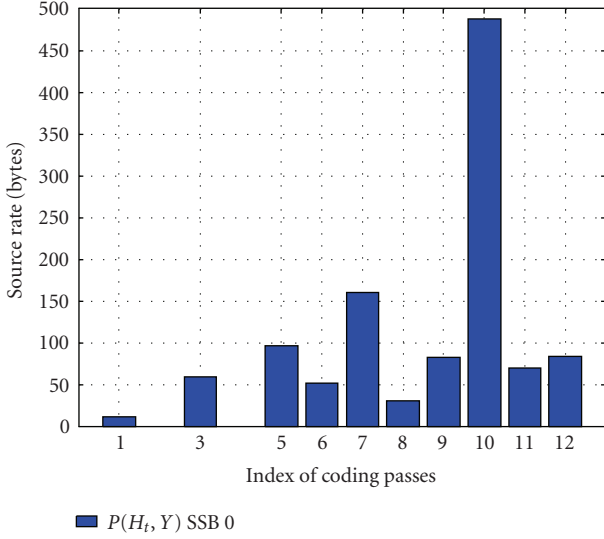


FIGURE 6: Source data rate of coding passes on the convex hull in the block 0 of STEFAN.

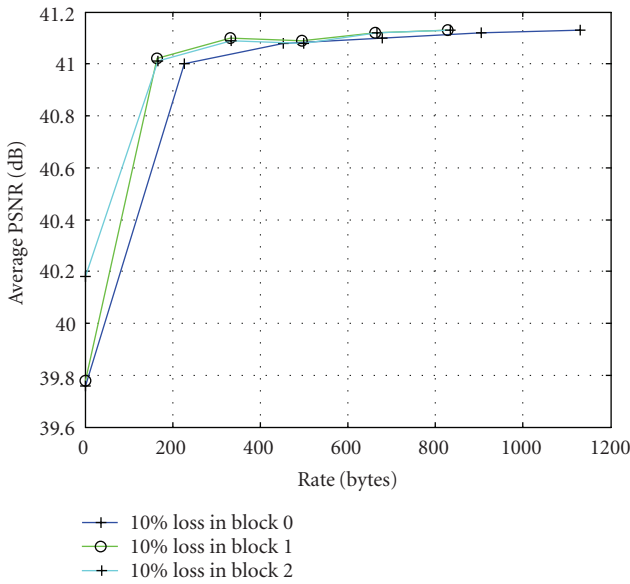


FIGURE 7: RD curves of STEFAN with 10% loss of coding passes in SSB 0 of the TSB  $P(H_t, Y)$ .

to predicted packet loss rate) to different wavelet subband data based on the data set’s R-D slope (or, equivalently, the distortion-reduction rate). Figure 9 illustrates this concept with some examples of real data. The content-adaptive FEC protection is applied to the coding block 0 of temporal subband  $P(H_t, Y)$  and spatial subband 0 of the STEFAN sequence. In this plot, the  $y$ -axis is the distortion reduction rate (i.e., the slopes of the conventional R-D curve as in Figure 3) and the  $x$ -axis is the bitrate (including source data bits and FEC protection bits). The dashed line is the original subband data without any protection, while the solid line with circle markers is the FEC protected data given 3% estimated packet

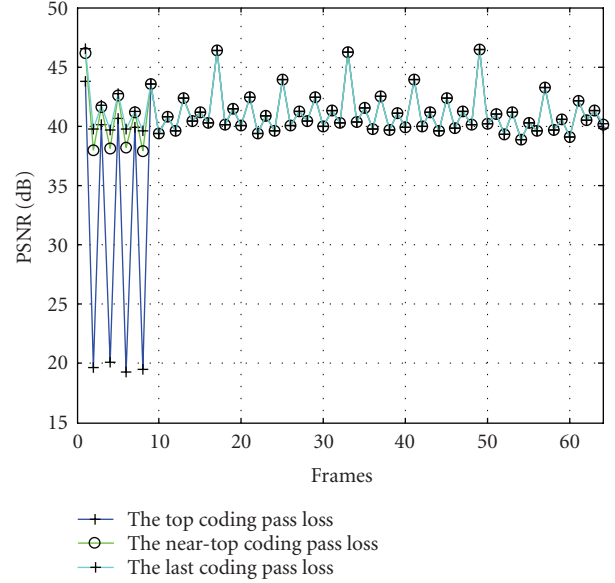


FIGURE 8: PSNR of STEFAN@202 kbps with 10% loss of coding passes in block 0 of SSB 0 of the TSB  $P(H_t, Y)$ .

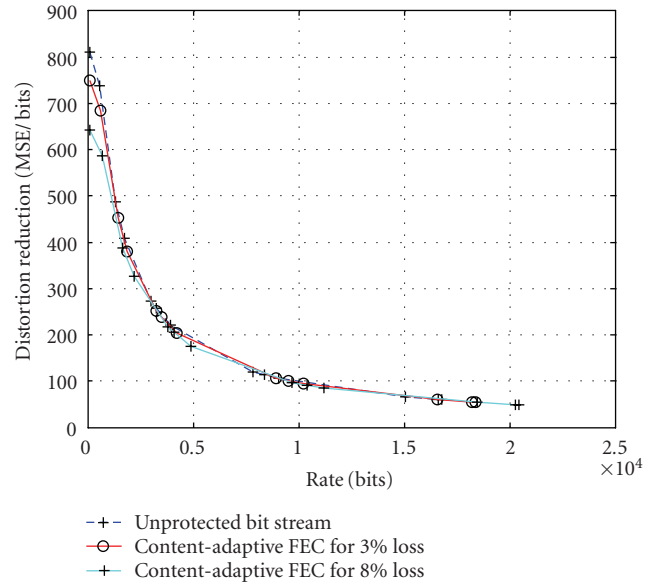


FIGURE 9: Example of overhead of content-adaptive FEC protection for different rate points (or equivalently, coding passes) within a coding block.

loss rate and the solid line with “plus” markers is the protected data given 8% estimated packet loss rate. The lower the rate point, the higher the protection level. The exact equation used to compute the protection level will be described in a moment. Note that the function in Figure 9 can be used for operational RDO streaming decision since it exhibits rate versus source-and-channel distortion tradeoff.

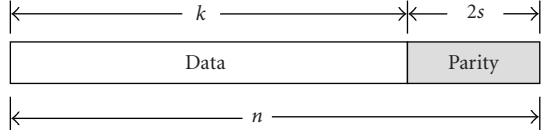


FIGURE 10: An  $(n, k)$  RS code word with  $k$  symbols of video data and  $2s$  symbols of parity.

In the proposed framework, for each group of video bit-streams, an  $(n, k)$  Reed-Solomon (RS) code-based FEC is applied to add resiliency to the data. In Figure 10,  $n$  is the code word length of the RS encoder,  $k$  is the number of video data symbols (8 bits of bit stream data in this case), and  $s$  is the number of correctable symbols. The number of parity symbols is  $2s$ , where  $2s = n - k$ . If burst errors occur during transmission, then the RS decoder can correct up to  $s$  errors and detect up to  $2s$  errors per code word.

For 3D-ESCOT, each coding block  $j$  has temporal level index  $\omega_j$ , component index  $\nu_j$ , and spatial subband index  $\tau_j$ . Assuming that the bit stream of a coding block is divided into  $l$  code words, the importance of a coding block can be expressed as in (1),

$$c_j(x, y) = \left[ \exp \left( \frac{\alpha}{y} \cdot \sum_{n=0}^x \left( \frac{(T - \omega_j) \cdot U_1}{T} + \frac{U_2}{(Y - \nu_j)} + \frac{1}{(B - \tau_j)} \right) \right) \right], \quad (1)$$

where  $x = 0, 1, \dots, l - 1$ ,  $y$  is the R-D slope of the first coding pass in block  $j$ ,  $\alpha$  is a scale factor,  $T$  is the maximal temporal level index,  $Y$  is the maximal component index,  $B$  is the maximal spatial subband index, and  $U_1$  and  $U_2$  are weighting factors. Note that the value of  $c_j(x, y)$  is defined to be  $0 \leq c_j(x, y) \leq n/2$ . The protection level of the content-adaptive FEC scheme is determined based on the characteristics of the coding block  $c_j(x, y)$  given by (1) subject to the network conditions. The bit stream of a coding block is composed of several coding passes. Since the coding passes of a coding block are roughly ordered based on their impact to visual quality, therefore, the protection level applied to different coding passes (indexed by  $x$ ) of block  $j$  is proposed to be  $s_{j,x}$ , which is defined in (2):

$$\hat{s}_{j,x} = \left[ \exp \left( \frac{|\lambda_{j,0}|}{\beta} \right) \cdot n_{pl} \right] - c_j(x, |\lambda_{j,0}|), \quad (2)$$

$$s_{j,x} = \hat{s}_{j,x} + o, \quad o = \begin{cases} 0, & \text{if } \hat{s}_{j,x} \text{ is even,} \\ 1, & \text{if } \hat{s}_{j,x} \text{ is odd,} \end{cases}$$

where  $0 \leq s_{j,x} \leq n/2$ ,  $\lambda_{j,0}$  is the R-D slope of the first coding pass in block  $j$ ,  $n_{pl}$  denotes the estimated packet losses given current bandwidth  $R_{BW}$ , average packet size  $P_s$ , and packet loss rate  $\varepsilon_{pl}$ , and  $\beta$  is a scale factor determined empirically. Equation (2) is designed so that  $s_{j,0} \geq s_{j,1} \geq \dots \geq s_{j,l-1}$ , that is, the level of protection decreases following coding passes order. Note that  $n_{pl} = \lfloor \varepsilon_{pl} \times R_{BW}/P_s \rfloor$ , where the operator  $\lfloor \cdot \rfloor$  returns the largest integer smaller than or equal to the operand.

### 3. THE PROPOSED PACKETIZATION SCHEME AND STREAMING FRAMEWORK

In the following discussions, we use the terminology “block bit stream segment” to describe a portion of bit stream bytes of a coding block across spatiotemporal subbands (see Figure 2). A block bit stream segment is composed of one or more coding passes. The packaging of the scalable bit streams into UDP packets is accomplished following both rate control and error control constraints. These constraints try to fulfill the following goals.

- (1) Error protection level of a block bit stream segment should depend on its entropy. The higher the entropy, the higher the protection level. Note that since a block bit stream segment is only a small chunk of data in a coding block, the granularity of content adaptation of the FEC protection is at a very fine scale.
- (2) The streaming packet rate of the system should stay as low as possible. UDP packet size should be smaller than the MTU (maximum transmission unit) allowed by the network links (typical size is around 1500 bytes for wired networks, and MTUs ranging from 250 to 750 bytes commonly have better throughput under no bit error rate circumstances for mobile ad hoc networks [31]). On the other hand, processing a lot of small packets causes very high overhead to the streaming system, especially on the client side. Therefore, a reasonable packet size is slightly smaller than the MTU.
- (3) Although interleaving with FEC works well for handling packet losses, it does introduce extra delay to the transmission of video data. Therefore, the selection of interleaving group size must take into account the end-to-end delay of the whole systems. In general, for broadcast video streaming, overall delay should be less than 20 seconds [32].

#### 3.1. Packetization of FEC-protected data

As mentioned in the previous section, a systematic Reed-Solomon (RS) code word comprising of data symbols and parity symbols is used for content-adaptive FEC protection. RS coding used for the protection of the block bit stream segment is depicted in Figure 11. Assume that the total number of coding block is  $L$ ,  $i = 0, \dots, L - 1$ , for each coding block  $i$ , bit stream can be divided into  $m$ -data symbol units, it begins with the first block bit stream segment  $C_{i,0}$  and continues through  $C_{i,1}, C_{i,2}, \dots$  to  $C_{i,m}$ . An  $(n, k_x)$ ,  $x = 0, \dots, m$ , RS code is then applied to add resiliency to the  $m$ -data symbol unit. Since the block bit stream segments have large variations in size, one must pack variable number of block-bitstream segments into a data unit to reduce packet overhead. In addition, different levels of protection are allocated to different portions of the coding block,  $k_m \geq k_{m-1} \geq \dots \geq k_0$ . Furthermore, the data symbols gathered at the front end of the data unit, and the parity symbols are located at the back end of the data unit. For each data unit, there is a header that describes the protection level of the data unit. The header is also protected by RS coding. Also note that if data unit is not

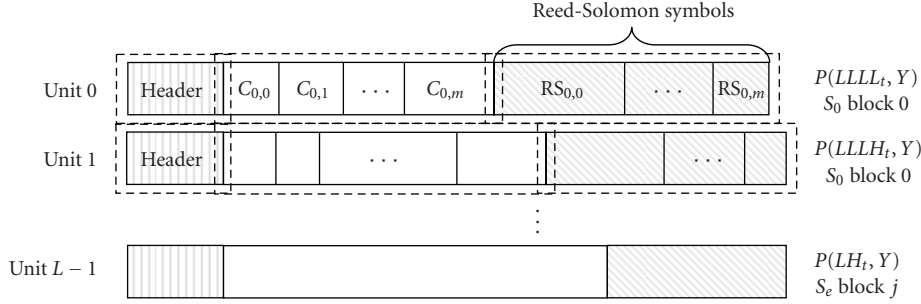


FIGURE 11: Packetization for one group of video data.

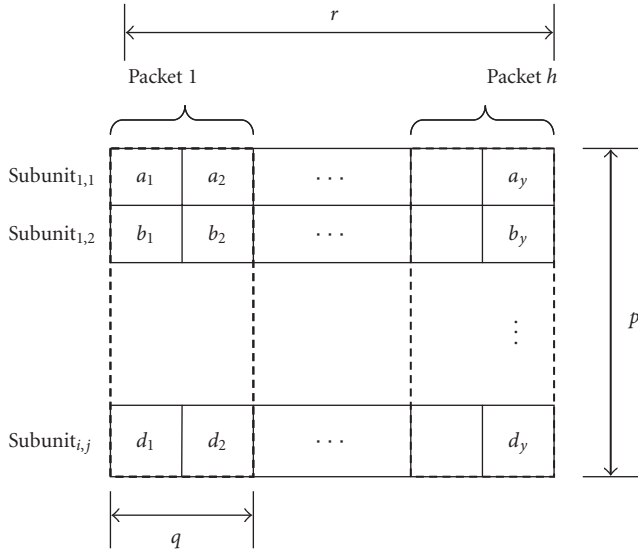


FIGURE 12: Data interleaving scheme for one group of video data.

a multiple of  $k$ , zero padding will be applied at the end of the data. These padding bytes do not have to be transmitted though.

Since we are dealing with a packet loss channel, not a bit error channel, a byte-wise data-interleaving scheme is used to shuffle the RS coded data among several data packets before transmission. As illustrated in Figure 12, a block bit stream segment is spread across many packets (each packet is composed of the group of data in dashed lines in Figure 12). For each packet, in addition to video data payload, we also have to transmit the highest protection level, temporal subband index, component index, spatial subband index, and block index in order to properly deinterleave the data. When interleaving is used, the interleaving depth must match the worst case of channel conditions against burst errors. In addition, a large interleaving depth will have impact on the packet buffer size of the client and the end-to-end delay of packet transmissions. The interleaving depth should be appropriately chosen to handle the worst-case error bursts of the networks. As mentioned in Section 2, the number of parity symbols is  $2s$ , where  $s$  means the number of correctable errors by an RS

decoder. A data unit can be split into several  $r$  equal length sub-units and each interleaved packet is composed of  $q$  data symbols from each subunits. Hence,  $q$  is limited by the number of parity symbols  $s$ , and  $p$  is limited by the maximum end-to-end delay.

### 3.2. Streaming policy

The proposed framework will adapt to the fast varying channel conditions by using the real-time network statistics feed-backs from the client side. Through standard RTCP receiver reports, the server can obtain the statistics such as round-trip time (RTT), jitter, short-term packet losses, and accumulative packet losses. The packet loss rate is used to compute the content-adaptive FEC-protected data rate-distortion trade-off information as described in Section 2. In addition, the server can compute the effective channel bandwidth through the last packet sequence number received by the client and loss rate. Based on the estimated channel bandwidth and the rate-distortion information, the system performs a dynamic rate allocation at discrete transmission time to enhance the perceived quality whenever the network bandwidth is good enough for perceptible quality improvement.

For the correction of errors, parity packets are employed to recover from lost data packets. But some of parity packets may be lost or corrupted when transmitting packets over the networks based on the UDP protocol. For enhancing the system performance, error recovery mechanisms such as retransmission or error correction can be applied to handle uncorrectable errors. Instead of using retransmission scheme to all parity packets, the proposed system delivers more redundancy parity packets to those packets carrying important portion of blocks and fewer to other packets. As seen in Figure 13, all of the blocks are arranged according to the degree of importance of each spatial-temporal subband. In addition, the higher protection-level parity symbols are gathered together into one packet for the maximum efficiency of the error recovery scheme.

## 4. EXPERIMENTS

This section presents the experimental results of the proposed video streaming system. The block diagram of the proposed streaming system is shown in Figure 14. The system is

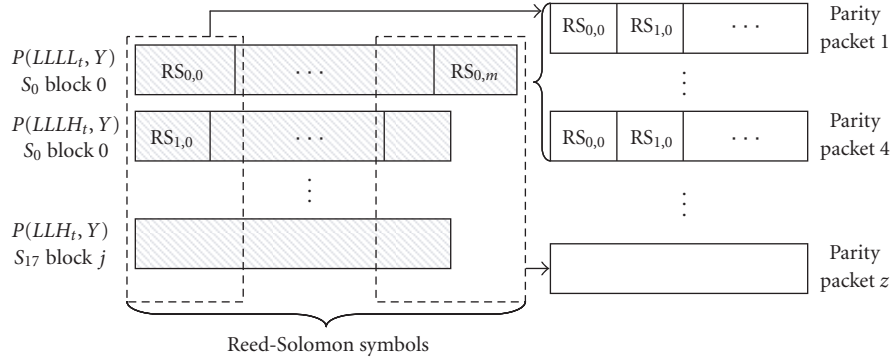


FIGURE 13: Duplication of some parity packets for enhanced protection of important video data.

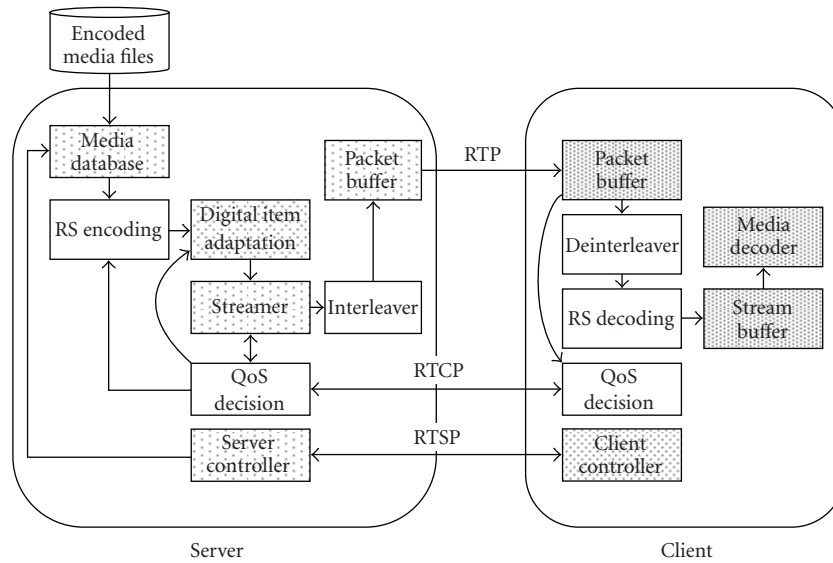


FIGURE 14: Architecture of the proposed system.

based on the MPEG-21 test bed for resource delivery [33]. The test bed includes an IP transmission channel emulator (based on the NIST net [34]) that allows real-time emulation of various network conditions. We have added Reed-Solomon coding modules, a data interleaving module, and a data deinterleaving module to the original test bed.

The CIF version of the standard MPEG test sequences STEFAN, MOBILE, TABLE TENNIS, FOREMAN, and COASTGUARD is used for the experiments. Those sequences are encoded using MSRA 3D wavelet video coding software [35] at 15 frames per second and a GOP is composed of 64 frames. Four levels of 5/3 MCTF temporal decomposition and three levels of 9/7 wavelet spatial decomposition are used for subband coding. The number of luminance ( $Y$ ) blocks is around 1024 block bit stream segments, and the number of chrominance ( $U$  and  $V$ ) blocks is around 608 block bit stream segments.

To evaluate the performance of the proposed system, reasonable range of packet loss rates should be used. Over wired links, studies showed that based on MPEG compressed video using the RTP and UDP transport protocols reported the average packet loss rates, ranging from 3.0 to 13.5 percent [36]. Over wireless links, Lai et al. [37] reported the characteristics of the MosquitoNet wireless network. The packet loss rates were 25.6% when packets were sent from a mobile host to a router, and 3.6% when packets are sent from a router to a mobile host. Risueño et al. [38] did a comprehensive study of the handover mechanisms during the disruption time in the wireless network. They reported that the packet loss caused by the handover mechanism was below 0.3%. Based on these published studies, we have set the packet loss rates of our experiments to 5%.

The proposed content-adaptive FEC protection framework is compared against a fixed-level FEC protection scheme



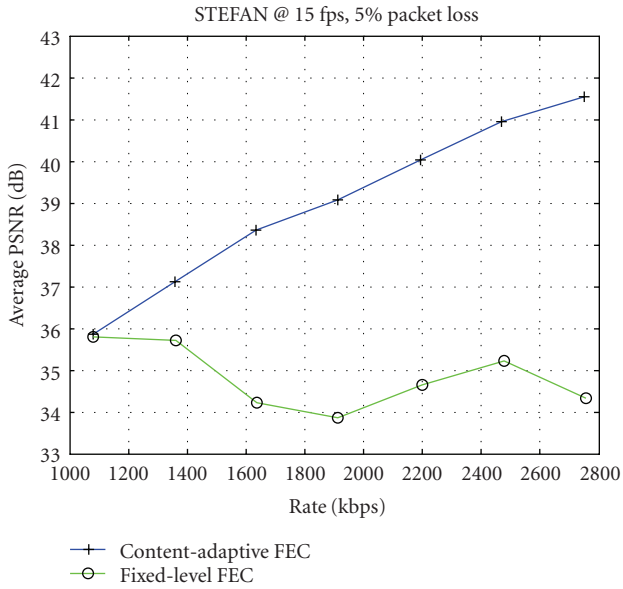


FIGURE 15: Comparison between fixed and content-adaptive FEC protection (both protection levels are for 4% packet loss) for the STEFAN sequence.

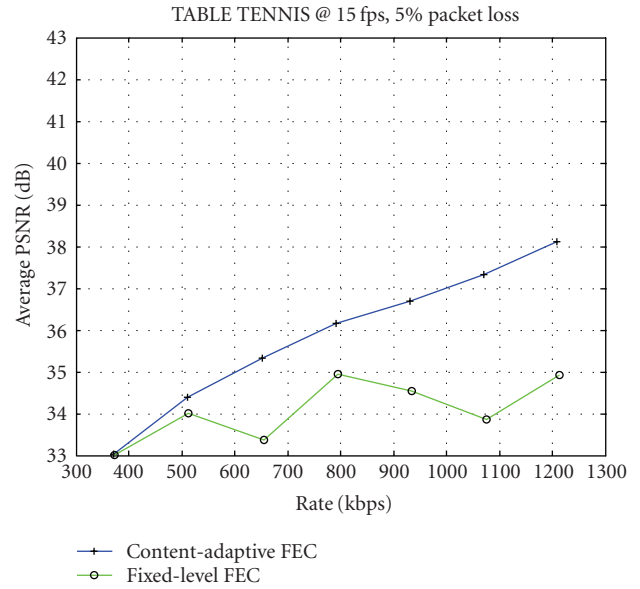


FIGURE 17: Comparison between fixed and content-adaptive FEC protection (both protection levels are for 4% packet loss) for the TABLE TENNIS sequence.

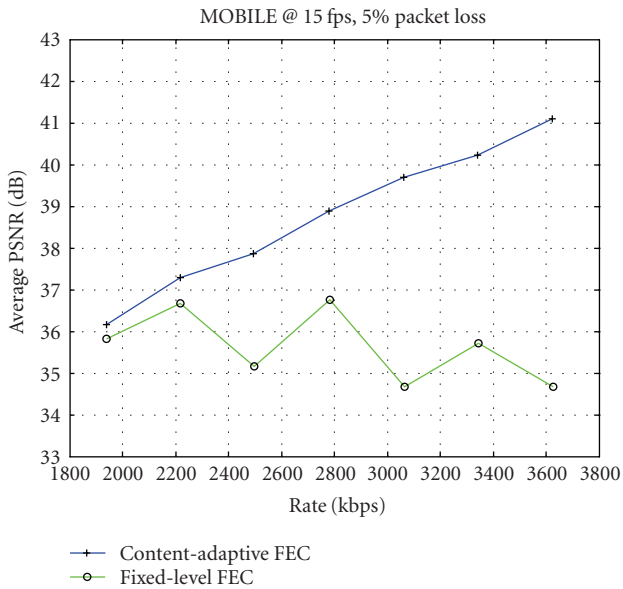


FIGURE 16: Comparison between fixed and content-adaptive FEC protection (both protection levels are for 4% packet loss) for the MOBILE sequence.

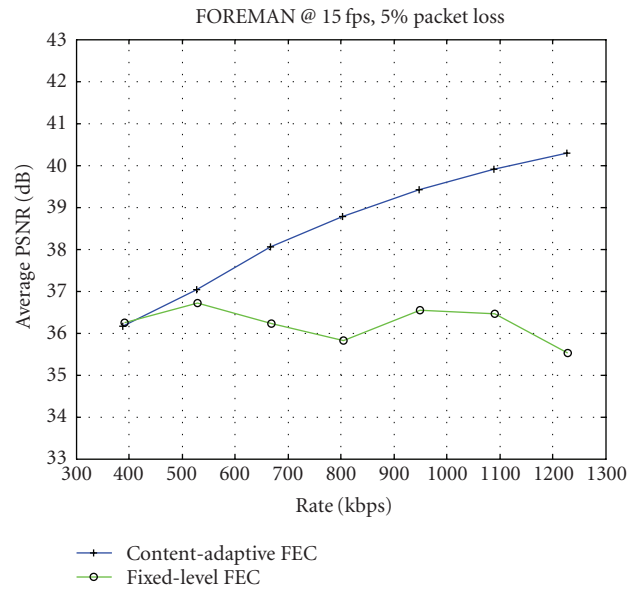


FIGURE 18: Comparison between fixed and content-adaptive FEC protection (both protection levels are for 4% packet loss) for the FOREMAN sequence.

for video streaming over a 4% packet loss channel. The R-D curves of the luma channel of the reconstructed video sequences are shown in Figures 15–19. The level of protection for different segment of video data with the content-adaptive FEC scheme is computed using (2), while the level of protection for video data protected using the fixed-level FEC is determined by the (predicted) average number of packet losses per second. In either case, the maximal packet loss

protection level can only recover up to 4% packet losses on average. It is important to point out that the overall number of bits used for FEC protection is the same for both the content-adaptive scheme and the fixed-level scheme. However, for content-adaptive protection, more protection bits are applied to more important data (based on (2)). Note that the PSNR of the reconstructed video does not increase with the bitrate for the fixed-level FEC protection

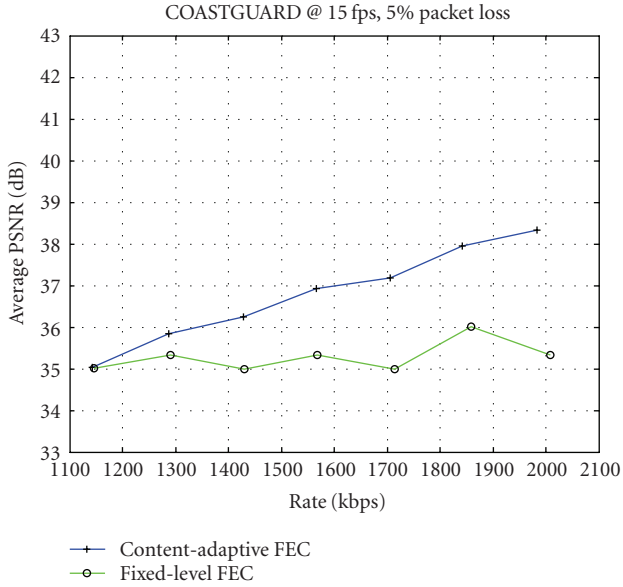


FIGURE 19: Comparison between fixed and content-adaptive FEC protection (both protection levels are for 4% packet loss) for the COASTGUARD sequence.

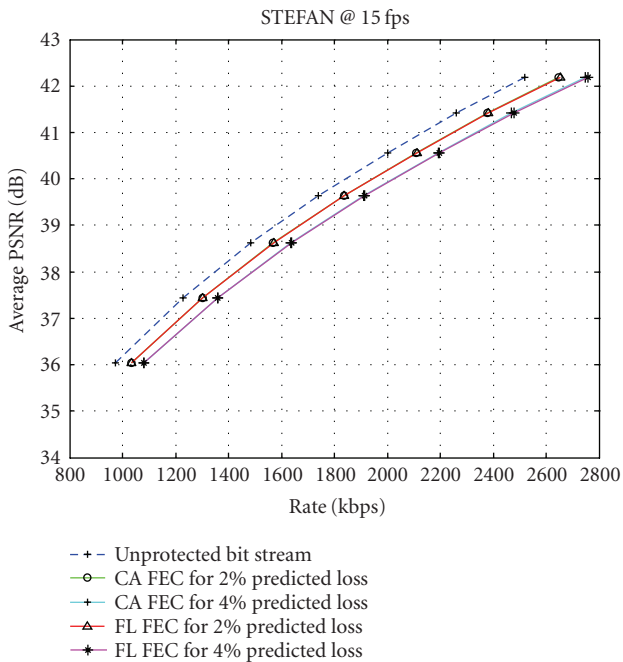


FIGURE 20: RD curves of STEFAN without and with different FEC protections in an error-free environment (CA: content-adaptive, FL: fixed-level).

mechanism. The reason is that if the small set of crucial subband data is corrupted, the PSNR will stay low even if more (less important) data is transmitted. As one can see from the figures, the content-adaptive FEC protection scheme works much better than the fixed-level protection

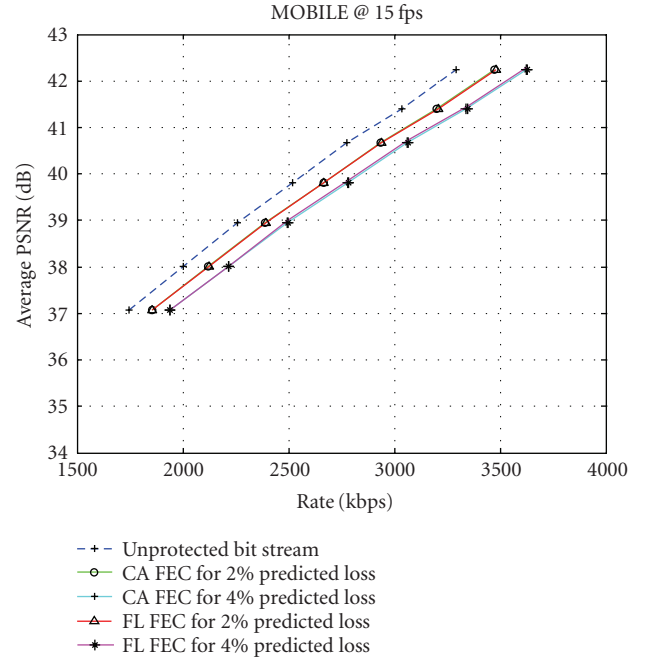


FIGURE 21: RD curves of MOBILE without and with different FEC protections in an error-free environment (CA: content-adaptive, FL: fixed-level).

scheme. The RD curves of unprotected bit streams are not shown in the figures because packet losses can severely corrupt an unprotected wavelet video bit stream. Take the STEFAN sequence for example, when the first few coding passes of coding block 0 of  $P(LLLL_t, Y)$  are lost, the PSNR is usually less than 10 dB, no matter how high the bitrate is.

To demonstrate the bitrate overhead of the content-adaptive FEC protection scheme, the error-free R-D curves of the video bit streams with and without FEC protection are shown in Figures 20–24. For the bit streams that are protected using FEC schemes, the level of protection is computed based on an assumption that the channel has estimated packet loss rates of 2% and 4%. As one can see from these figures, the overhead of the proposed content-adaptive FEC protection is quite reasonable (about 0.2 to 0.5 dB quality drop across a wide range of bitrates for 2% packet loss protection).

## 5. CONCLUSIONS AND FUTURE WORK

In this paper, a content-adaptive FEC protection and packetization framework for wavelet video streaming is proposed. The adaptive packet loss protection scheme using Reed-Solomon coding and data interleaving is based on detail analysis of rate-distortion tradeoff of wavelet subband data. The experimental results show that with an adaptive fine-granularity FEC protection level packetization scheme, one can achieve much better quality than with a fixed-level FEC protection scheme.

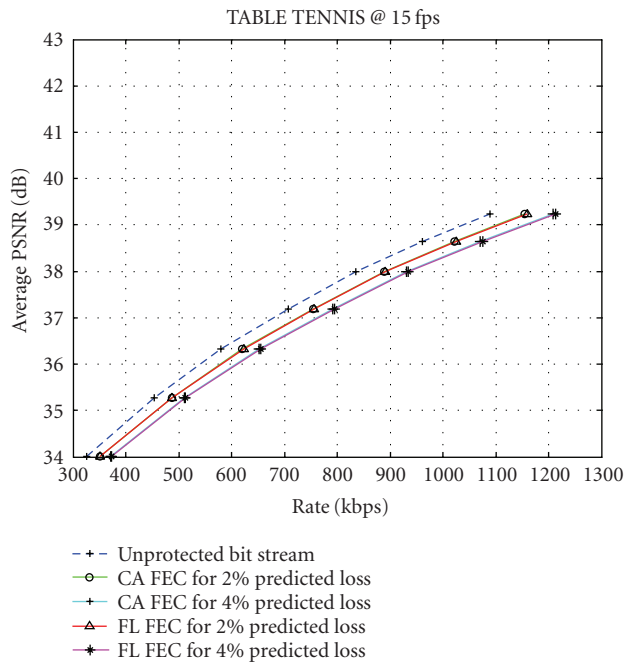


FIGURE 22: RD curves of TABLE TENNIS without and with different FEC protections in an error-free environment (CA: content-adaptive, FL: fixed-level).

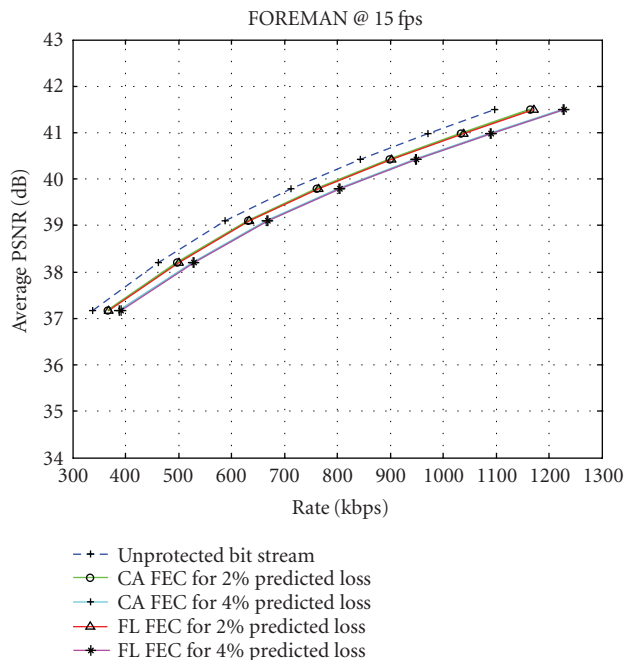


FIGURE 23: RD curves of FOREMAN without and with different FEC protections in an error-free environment (CA: content-adaptive, FL: fixed-level).

For future work, a run-time operational rate-distortion optimized streaming policy with joint optimization for minimal source coding distortion and packet loss distortion will

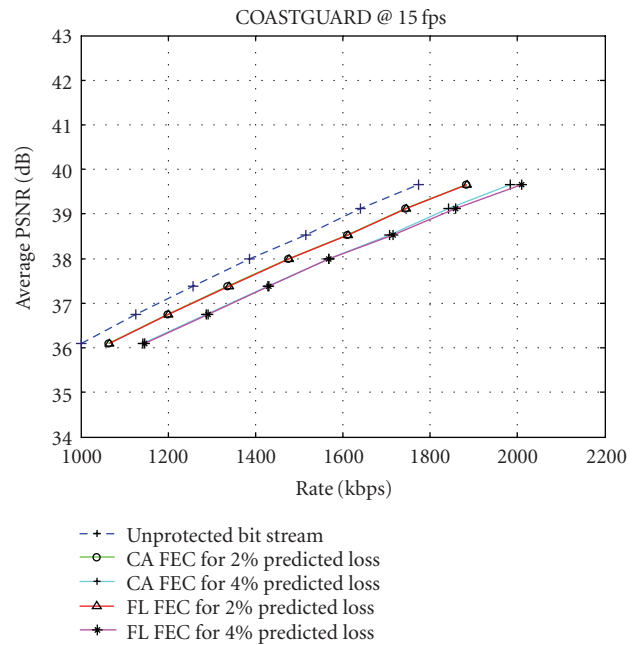


FIGURE 24: RD curves of COASTGUARD without and with different FEC protections in an error-free environment (CA: content-adaptive, FL: fixed-level).

be investigated. Furthermore, the equation used for the determination of FEC protection level given estimated packet loss rate is designed based on empirical analysis. More rigorous derivation of the FEC protection level function is under investigation.

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