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Modeling and Analysis of Frame-Level Forward Error Correction for MPEG Video over Burst-Loss Channels

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Abstract: Forward error correction (FEC) is a common error control technique to improve the quality of video streaming over lossy channels. To optimize the data recovery performance, frame-level FEC schemes have been proposed for streaming video to maximize playable frame rate (PFR) within transmission rate constraint on a random binary symmetric channel (BSC). However, burst loss is a commonplace in current Internet architecture, and the FEC efficacy can be degraded since the burst data losses easily exceed the error correction capacity of FEC. Accordingly, an estimated video quality model over burst loss channels is proposed in this paper to evaluate the impact of burst loss for FEC-based video applications. In addition to the model analysis, the simulation experiments on the NS-2 network simulator are conducted at a given estimate of the packet loss probability and average burst length. The results suggest a useful reference in designing the FEC scheme for video applications, and as the video coding and channel parameters are given, the proposed model can provide the optimal FEC solution to achieve a better reconstructed video quality than the FEC model based on a random BSC channel.

Keywords: Forward Error Correction, Bust Loss Channels, Video Streaming

1 Introduction

Packet losses can severely affect the quality of delay sensitive streaming video over the wired/wireless Internet. In the current Internet architecture, the packet loss tends to be bursty due to the network congestion and/or wireless errors. In order to improve the quality of media streaming, forward error correction (FEC) is a well-established technique to maintain video quality for combating losses and errors in end-to-end networks [9]. In a generic FEC scheme, the redundant packets are generated by FEC encoder and transmitted along with source packets. If the receiver side can collect enough packets, the source data can be reconstructed by FEC decoder without error. Compared with the automatic repeat request (ARQ), which retransmits lost data through an end-to-end acknowledgement mechanism, FEC is usually preferred for real-time video applications due to lower end-to-end delay.

The optimal FEC control schemes have been proposed for streaming video based on a playable frame rate (PFR) model to evaluate the quality of video streaming. In [7], Wu et al. derive an analytical FEC model within TCP-friendly rate constraint for MPEG video streaming to obtain the optimal reconstruction quality of group of picture (GOP). Yuan et al. in [8] apply FEC at the GOP level to increase the error correction capacity of FEC with more computational complexity in average since a larger amount of video data need to be process. Their optimal frame-level FEC regards the transmission channel as a random binary symmetric channel (BSC) or uniform loss model, and assume that the adjacent packet losses are independent.

The burst losses induced by network congestion and/or wireless errors have a great impact on the efficiency of FEC. Related works in [12,13,14,15] have shown that the FEC schemes become very ineffective over burst-loss transmission channels since the receiver

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may not receive a sufficient number of packets with each FEC-encoded block to reconstruct the original data. Attempting to resolve this problem, different FEC control approaches can be adopted to increase the transmission reliability for video streaming applications, with an additional end-to-end delay. Typical approaches under consideration are to use longer block size for FEC coding, and to adopt interleaving cross multiple blocks. Using longer block size can provide higher error correction capacity, but causes the additional packet buffering latency and FEC processing delay, both of whom contribute to the end-to-end delay. In the FEC interleaving, the sender reorders the transmission sequence of the FEC blocks such that the continuous losses are distributed amongst different blocks at the receiving side. By converting burst losses into random errors, the FEC interleaving mitigates the impact of long loss burst on FEC efficiency [10,11]. Generally, the interleaving degree (i.e., the amount of interleaved FEC blocks) can be determined by the average loss burst length. An alternative approach to achieving graceful video quality degradation in processing burst losses is to use unequal error protection (UEP). The UEP technique differentiates the FEC protection level for prioritized video data.

For FEC-based video applications over burst-loss transmission channels, the FEC coding and UEP should be different from those given by the FEC model based on the assumption of random BSC channels. It is therefore necessary to consider the burst loss pattern in building the FEC model. Accordingly, this paper presents an analytical FEC model to compute an expected quality of decoded video for the video streaming over burst-loss channels. Assuming the average packet loss rate and average burst loss length can be available in the network, a two-state Markov chain is adopted as the burst packet loss process. By integrating the burst loss process into the frame-level FEC coding model, our analytical model presents the performance of FEC over burst-loss channels and further provides the optimal FEC solution as the video coding and channel parameters are given. The analytical and simulation experiments are conducted to observe the differences in PFR performance between the random BSC channel and the two-state Markov channel. The experimental results can suggest a useful reference in designing the FEC scheme over burst-loss channels, and also shows that the proposed model outperforms the FEC model based on the uniform loss process.

The remainder of this paper is organized as follows. Section II introduces the background about the analytical model in previous researches, while Section III formulates the PFR model to evaluate the video quality over burst-loss channels. Section IV presents and discusses the performance results. Finally, Section V provides some brief concluding remarks.

2 Background

2.1 Forward Error Correction

Systematic Reed-Solomon (RS) erasure codes are the linear erasure code used to protect video data from channel losses. RS codes, RS (n, k), groups the source data packets into FEC blocks of a predetermined size k, and then encodes $n = k + h$ packets for network transmission, where $h \geq 0$ is the number of redundant packets. Provided that k or more packets are successively received, the FEC block can be successfully reconstructed. For simplicity, a Bernoulli/uniform loss model can be used for modeling the packet loss process with the parameter of average loss rate. Given the packet loss probability, P_B, the probability that at least k packets is successfully reconstructed can be computed using

$$B(n,k,P_B) = \sum_{i=k}^{n} \left[\binom{n}{i} (1-P_B)^i (P_B)^{n-i} \right] \tag{1}$$

2.2 Playable Frame Rate for MPEG Video

In MPEG, a video can be divided into several groups of pictures (GOP), and each GOP contains three types of frames in a periodic sequence. Thus, the raw video data of MPEG video are encoded as Intra-coded (I), Predictive (P), and Bidirectional (B) video frames. An I frame is just a frame coded as a still image and encoded without dependence on any past frames. A P frame is encoded based on motion differences from the previous I frame or P frame. B frames are encoded based on the motion differences from the immediate past and future I or P frame. Due to the coding dependency, three types of frames have a descending order of importance (i.e., I, P, B). The organization of frames in a typical GOP is arranged as follows:

$$IB_{0,0}\cdots B_{0,N_{BP}-1}P_1\cdots P_m B_{m,0}\cdots B_{m,N_{BP}-1}P_{m+1}\cdots P_{N_P}B_{N_P,0}\cdots B_{N_P,N_{BP}-1} \tag{2}$$

where N_P is the number of P frames, N_B is the number of B frames in the GOP, and N_{BP} is the number of B frames in between an I and a P frame or two P frames. Denote the encoding frame rate per second as R_F, the effective GOP transmission rate is given by

$$G = \frac{R_F}{1+N_P+N_B} \tag{3}$$

According to Eq. 1 the probabilities of successful transmission for three frame types are obtained from

$$\begin{aligned} Q_I &= B(S_I+S_{IF}, S_I, P_B) \\ Q_P &= B(S_P+S_{PF}, S_P, P_B) \\ Q_B &= B(S_B+S_{BF}, S_B, P_B) \end{aligned} \tag{4}$$

where Q_I, Q_P, and Q_B are the probability of successful transmission of an I, P, or B frame, respectively; S_I, S_P, and S_B are the I, P, B frame size (in packets); S_{IF}, S_{PF}, and S_{BF} are the number of FEC packets for I, P, and B frames.

To evaluate the video streaming performance, the playable frame rate (PFR) is a good measure in a lossy network. The PFR is defined as the expected number of decodable frames at the receiver, and can be calculated according to [7]:

$$R = G \cdot Q_I \left[1 + \frac{Q_P - Q_P^{N_P+1}}{1-Q_P} + N_{BP} \cdot Q_B \left(\frac{Q_P - Q_P^{N_P+1}}{1-Q_P} + Q_I Q_P^{N_P} \right) \right] \tag{5}$$

3 Analytical Model

The proposed analytical model aims at providing the reference performance of FEC on MPEG video over burst-loss channels. In Section 2.2, the relationship between the burst loss and PFR is not addressed. In this section, the burst packet loss model is introduced first, and we then present a playable frame rate model over burst-loss channels.

3.1 Burst Packet Loss Model

In this study, the burst packet losses over the network are modeled by the Gilbert-Elliott model (i.e., two-state Markov model). Many studies show that the Markov model is a good approximation of Internet packet loss pattern for both wired and wireless transmissions [1,2,3,4,5]. In the Good state, a packet is dropped with a probability of 0, while in the Bad state, a packet is dropped with a probability of 1. The transition probability from the Bad state to the Good state is denoted as p, while the transition probability from the Good state to the Bad state is denoted as q. In the two-state Markov model, the average packet loss probability shows as:

$$P_B = \frac{q}{p+q} \tag{6}$$

Then the average burst length (ABL), L_B, is the average number of consecutive packet losses:

$$L_B = \frac{1}{p} \tag{7}$$

Packet loss based on two-state Markov chain can be modeled as a renewal error process [6]. The lengths of consecutive error-free gaps are distributed identically. Following [6], let the gaps length v be the event that after a lost packet, v-1 packets are received successfully and another packet is lost. Let $g(v)$ be the gap density function which gives the probability of a gap length v, i.e., $g(v) = Pr(1^{v-1}0|0)$, where the 1^{v-1} denotes consecutively received packets. Therefore, the gap distribution function $G(v)$ is the gap length which is greater than $v-1$, i.e., $G(v) = Pr(1^{v-1}|0)$. That is

$$\begin{aligned} g(v) &= \begin{cases} 1-p & for\ v=1 \\ p(1-q)^{v-2}q & for\ v>1 \end{cases} \\ G(v) &= \begin{cases} 1 & for\ v=1 \\ p(1-q)^{v-2} & for\ v>1 \end{cases} \end{aligned} \tag{8}$$

According to Eq.6 and Eq.7, Eq.8 can be modified to

$$\begin{aligned} g'(v,P_B,L_B) &= \begin{cases} 1-\frac{1}{L_B} & for\ v=1 \\ \frac{1}{L_B}\left(1-\frac{P_B}{(1-P_B)L_B}\right)^{v-2}\frac{P_B}{(1-P_B)L_B} & for\ v>1 \end{cases} \\ G'(v,P_B,L_B) &= \begin{cases} 1 & for\ v=1 \\ \frac{1}{L_B}\left(1-\frac{P_B}{(1-P_B)L_B}\right)^{v-2} & for\ v>1 \end{cases} \end{aligned} \tag{9}$$

It is noted that the receiver is responsible for periodically returning channel information, such as average packet loss probability (P_B) and average packet loss burst length (L_B), to the sender. Based on Eq. 9, when the sender receives the feedback information from the receiver, the sender can use P_B and L_B to build the bursty loss model based on the current channel conditions. The probability of m-1 packet losses within the next n-1 packets following a lost packet is R(m,n,P_B,L_B) and can be calculated by recurrence. Thus,

$$R(m,n,P_B,L_B) = \begin{cases} G'(n,P_B,L_B) & for\ m=1 \\ \sum_{v=1}^{n-m+1}[g'^{(v)} \cdot R(m-1,n-v,P_B,L_B)] & for\ 2 \le m \le n \end{cases} \tag{10}$$

Finally, $P(m,n,P_B,L_B)$ is the probability of m loss packets within a block of n packets

$$R(m,n,P_B,L_B) = \sum_{v=1}^{n-m+1}[P_B G'(v,P_B,L_B) \cdot R(m,n-v+1,P_B,L_B)]\ for\ 1 \le m \le n \tag{11}$$

For a block of n packets containing no loss packets, its probability can be computed as

$$P(0,n,P_B,L_B) = 1 - \sum_{m=1}^{n} P(m,n,P_B,L_B) \tag{12}$$

Finally, to analyze the effects of FEC for the bursty network, given the packet loss rate and average packet loss burst length, the probability $B'(n,k,P_B,L_B)$ that at least k packets is successfully reconstructed can be given by

$$B'(n,k,P_B,L_B) = 1 - \sum_{i=n-k+1}^{n} P(i,n,P_B,L_B) \tag{13}$$

3.2 Playable Frame Rate with Burst Loss Model

For MPEG video, the frame-level FEC produces redundancies for video frames and protects each video frame individually. Thus, according to Eq. 13, the probabilities of successful transmission for different frames can be expressed as follows:

$$\begin{aligned} Q'_I &= B'(S_I + S_{IF}, S_I, P_B, L_B) \\ Q'_{P_u} &= B'(S_{P_u} + S_{PF_u}, S_{P_u}, P_B, L_B) \\ Q'_{B_{u,w}} &= B'(S_{B_{u,w}} + S_{BF_{u,w}}, S_{B_{u,w}}, P_B, L_B) \end{aligned} \tag{14}$$

where Q'_I is the successful transmission ratio of the I frame; Q'_{P_u} is the successful transmission ratio of u-th P frame in a GOP; $Q'_{B_{u,w}}$ is the successful transmission ratio of w-th B frame following u-th P frame; $S_{PF_u}$is the number of FEC packets for the u-th P frame in a GOP; and $S_{BF_{u,w}}$ is the number of FEC packets for the w-th B frame following u-th P frame. To cater for the TCP-friendly bandwidth constraints in streaming video with FEC, the temporal scaling approach can be used to adjust the amount of video data. In the temporal scaling approach, the video frames with lower priority are discarded before transmission to match the available TCP-friendly transmission rate.

Therefore, the playable frame rate of I frame in a GOP is simply the I frame transmitted successfully

$$R'_I = G \cdot Q'_I \cdot D_I \tag{15}$$

where G defines the GOP rate using Eq. 3 and D_I is a binary parameter to indicate the temporal scaling decision of I frame. Specifically, the value of D_I is set to 0 as the I-frame is dropped; on the other hand, the value of D_I is set to 1 as the I-frame is transmitted. This binary setting similarly applies for other video frames. Since each subsequent P_u in a GOP depends upon the success of P_{u-1} and its own successful transmission. Let the temporal scaling parameter of frame P_j be D_{P_j}, the playable frame rate of P_u is shown:

$$R'_{P_u} = R'_I \cdot \prod_{j=1}^{u} (Q'_{P_j} \cdot D_{P_j}) \tag{16}$$

The playable frame rate for all P frames in a GOP as following:

$$R'_P = \sum_{u=1}^{N_P} R'_{P_u} = G \cdot Q'_I \cdot \sum_{u=1}^{N_P} \prod_{j=1}^{u} (Q'_{P_j} \cdot D_{P_j}) \tag{17}$$

In MPEG, all B frames following the same P frame have the similar dependency relationship and these B frames have the same playable frame rate by denoting the temporal scaling parameter of frame $B_{u,w}$ as $D_{B_{u,w}}$:

$$R'_{B_{u,w}} = R'_{P_{u+1}} \cdot Q'_{B_{u,w}} \cdot D_{B_{u,w}}, \ where\ 0 \le u \le N_P - 1 \tag{18}$$

When the B frames locate in the end of the reference GOP, these B frames not only depend on the last P frame (i.e., P_{N_P}), but also depend on the leading I frame of the next GOP:

$$R'_{B_{u,w}} = R'_{P_u} \cdot Q'_{B_{u,w}} \cdot D_{B_{u,w}} \cdot Q'_I, where\ u = N_P \tag{19}$$

Finally, the playable frame rate for all B frames in a GOP can be computed:

$$R'_B = \sum_{u=0}^{N_P} \sum_{w=0}^{N_{BP}-1} R'_{B_{u,w}} \tag{20}$$

Thus, the total playable frame rate of a GOP is expressed by

$$R' = R'_I + R'_P + R'_B \tag{21}$$

The main difference between our proposed and the previously introduced model in [7] is that our proposed model takes the burst packet loss into consideration and adopts a two-state Markov model as the underlying packet loss process. Furthermore, Eq. 9 provides a parameter conversion method to realize the model calculation by means of using the available channel parameters of average packet loss rate and average burst length. As a result, given the transmission channel parameters (i.e., P_B and L_B) and the video details (e.g., the amount of I, P, and B packets), the model can calculate an optimal solution of the required FEC redundant packets for video frames to achieve the highest PFR value under the transmission rate constraint. A detailed discussion of the optimal FEC control is presented in the Section 3.3.

3.3 Optimal Frame-level FEC Control

Uniform datagram protocol (UDP) is generally used in video streaming in order to provide a steady data transmission rate. Moreover, it is widely agreed that the UDP transmission should be TCP-friendly to avoid network congestion loss and delay. A TCP-friendly video flow needs to regulate its output rate to match the TCP-friendly transmission rate. Padhye et al. [16] have shown that the upper bound on the TCP-friendly bandwidth T (in bytes/sec) of a network flow is given by

$$T = \frac{S}{t_{RTT}\sqrt{\frac{2P_{CB}}{3}} + t_{RTO}(3\sqrt{\frac{3P_{CB}}{8}})P_B(1+32{P_{CB}}^2)} \tag{22}$$

where S is the packet size in bytes, t_{RTT} is round-trip time in seconds, P_{CB} is the congestion-induced packet loss probability, and t_{RTO} is the TCP retransmit timeout value in seconds.

In the GOP, the video frames can be classified in terms of their type and their distance from the leading I frame. The loss of an I frame has a significant effect on the video quality of the entire GOP, and hence the I frame is assigned the highest priority. The P frames have a temporal dependency, and thus P frames which are closer to the I frame are assigned a higher priority than those located further from the I frame. Finally, the B frames are not used as references by any of the other frames in the GOP, but cannot be dropped continuously since this will result in a temporal quality degradation. Thus, the B frames are selected evenly throughout the GOP in accordance with their distance from the reference I frame, and are assigned a progressively reducing priority accordingly. That is, the first B frame after the I frame is assigned the highest priority, the first B frame after the first P frame is assigned the next highest priority, the first B frame after the second P frame is assigned the next highest priority, and so on. As a result, the original GOP pattern shown in Eq. 2 is re-organized to form a frame priority sequence as follows:

$$IP_1P_2\cdots P_{N_P}B_{0,0}B_{1,0}\cdots B_{N_P,0}B_{0,1}\cdots B_{N_P,1}\cdots B_{N_P,N_{BP}-1} \quad (23)$$

According to the video frame priority, the temporal scaling approach drops frames starting from the tail of the sequence and moving towards the head of the sequence. Combined with the temporal scaling approach, the FEC allocation problem to obtain an optimal PFR value under the constraint of TCP-friendly transmission rate becomes

$$\begin{aligned}
&Maximize\\
&R' = R'((N_{PD},\{N_{BD_u}\}),(S_{IF},\{S_{PF_u}\},\{S_{BF_{u,w}}\}),P_B,L_B)\\
&Subject\ to:\\
&G\cdot\left(\begin{array}{c}(S_I+S_{IF})+\sum_{u=1}^{N_P-N_{PD}}(S_{P_u}+S_{PF_u})+\\ \sum_{u=0}^{N_P-N_{PD}}\sum_{w=0}^{N_{BP}-N_{BD_u}-1}(S_{B_{u,w}}+S_{BF_{u,w}})\end{array}\right)\cdot S\le T \quad (24)\\
&0\le N_{PD}\le N_P, 0\le N_{BD_u}\le N_{BP}\\
&0\le S_{IF}\le S_I, 0\le S_{PF_u}\le S_{P_u}, 0\le S_{BF_{u,w}}\le S_{B_{u,w}}
\end{aligned}$$

where N_{PD} is the number of dropped P frames, and N_{BD_u} is the number of dropped B frames which following u-th P frame. Note that the leading I frame is never dropped, even under low-bit-rate transmission conditions, in order to avoid interrupting the video presentation.

Table 1: SYSTEM PARAMETER SETTINGS.

The number of I Frame (packets)	25
The number of P Frame (packets)	8
The number of B Frame (packets)	3
GOP Length	12
N_{BP}	2
N_P	3
P_B	1%-10%
L_B	1-10
Packet size (bytes)	1024
$FEC(S_{IF},S_{PF},S_{BF})$	(0,0,0),(10,0,0), (10,4,0),(10,4,1)

4 Performance Analysis

This section presents the analytical results for PFR models based on the uniform packet loss process (uniform-PFR model) and the renewal packet loss process (renewal-PFR model), respectively. We further conduct the NS-2 simulation experiments to observe the video streaming performance under the realistic network environment. The simulation topology is shown in Figure 1. The wireless link was assumed to have length of 10 meters and the transmission was performed in accordance with the IEEE 802.11b 11 Mbps protocol. Table I gives the system settings. For all experiments, four different FEC UEP settings were adopted to determine the differentiated protection degrees at the video frame level. In the stronger protection degree, more low-priority frame types received FEC redundancies. For instance, the strongest protection degree had the redundant packets of 10, 4, and 1 for I, P, and B frame, respectively. All results were presented in PFR ratio (%).

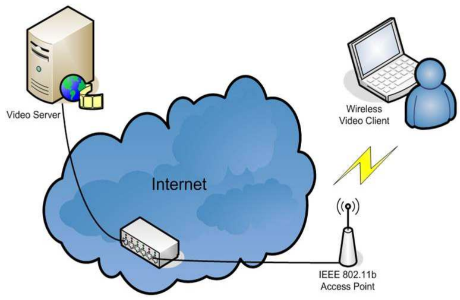
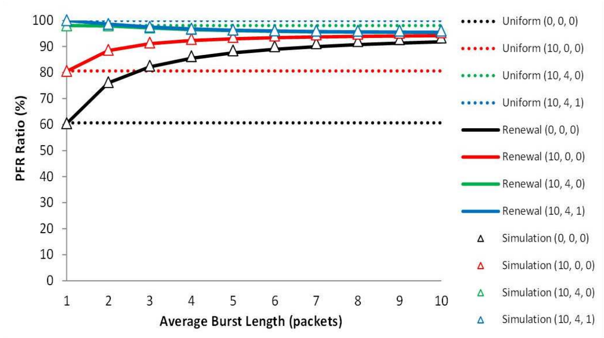


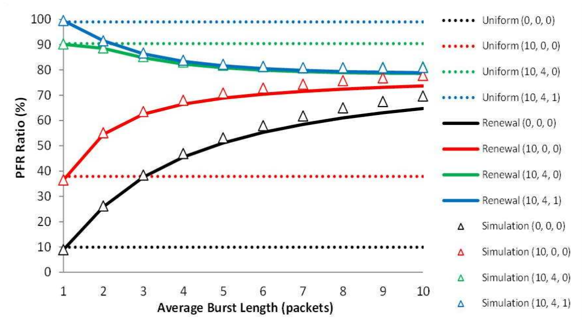
Fig. 1: The simulation topology

4.1 Analysis Results

The analytical results are obtained by using Eq. 5 for the uniform loss model and Eq. 21 for the renewal loss model. Figure 2 shows the PFR results as the average burst length varies from 1 to 10. The simulation results are also plotted to observe the PFR behavior in the realistic network environment. In Figure 2 (a), the average packet loss rate is 1% for all analytic and simulation cases, while the average packet loss rate is set to 5% in Figure 2 (b). As shown in Figure 2, it can be seen that the uniform-PFR model is irresponsive to the changes in the burst packet loss length and presents the differentiated PFR values for different FEC UEP settings. For the renewal-PFR model, we can observe that (1) both two cases with lower UEP protection degree- FEC (0, 0, 0) and FEC (10, 0, 0), has the increased PFR values as the average burst loss length increases, and (2) other two cases with higher UEP protection degree- FEC (10, 4, 0) and FEC (10, 4, 1), receives the decreased PFR values as the average burst loss length is increased and their PFR values are still higher than that obtained by the lower UEP protection degrees. It is noted that the larger burst packet loss length produces the lower packet loss rate since more packet losses easily aggregate for a long burst packet loss length with the fixed packet loss rate. For the

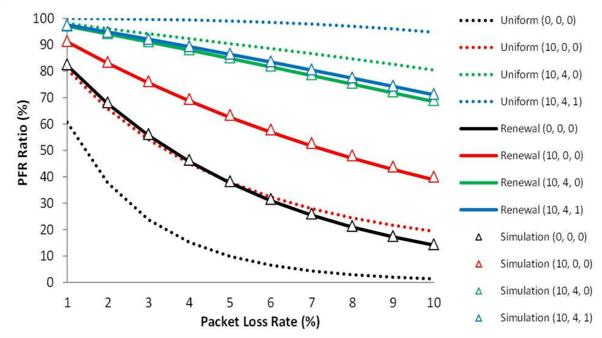


(a)

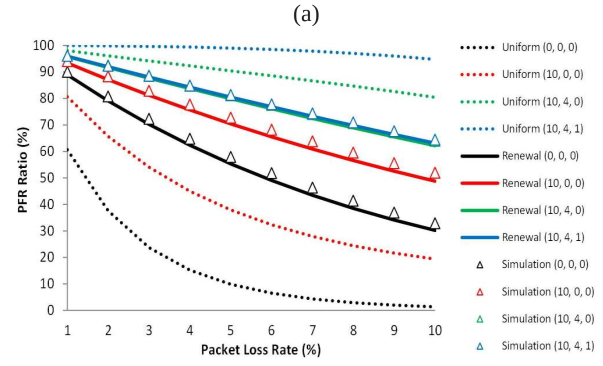


(b)

Fig. 2: The PFR ratio comparison with varied average burst length: (a) PLR=1%, (b) PLR=5%



(a)



(b)

Fig. 3: The PFR ratio comparison with varied packet loss rate: (a) ABL=3; (b) ABL=6

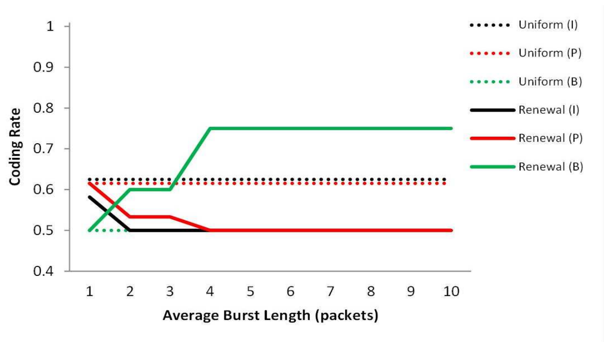
cases with lower UEP protection degree, the long burst loss length thus causes the increased PFR values. As to the cases with higher UEP protection degree, it is observed that the longer burst loss length easily decreases the FEC efficacy since the receiver may not receive a sufficient number of packets with the FEC-encoded P and B frames to reconstruct the original video frames.

Figure 3 shows the PFR results as the average packet loss rate varies from 1% to 10%. In Figure 3 (a), the average burst length is 3 for all analytic and simulation cases, while the average burst length is set to 6 in Figure 3 (b). Comparing Figure 3 (a) with Figure 3 (b), for the renewal-PFR model, the cases with higher UEP protection receives lower PFR levels as the average burst length is increased to 6, and the cases with lower UEP protection receives higher PFR levels. For the uniform-PFR model, all PFR curves in Figure 3 (a) are the same as those in Figure 3 (b) since the uniform-PFR model is unaware of burst packet loss. From Figure 2 and Figure 3, the uniform-PFR model under-estimates the PFR for the cases with lower UEP protection degree, while it over-estimates the PFR for the cases with higher UEP protection degree. In addition, the PFR curves of the simulation and the proposed renewal-PFR model are close to each other.

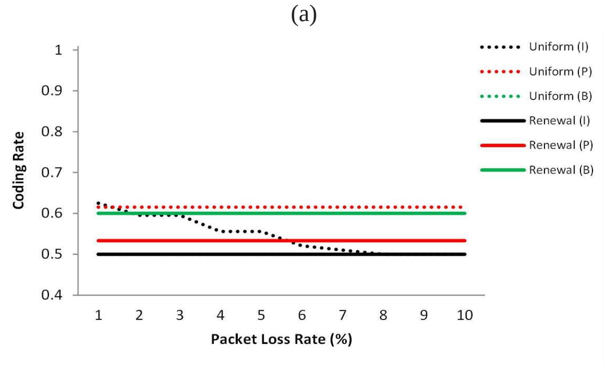
4.2 Optimal Result Comparison

In order to evaluate the optimal PFR performance for two different packet loss processes (i.e., Eq. 5 and Eq. 21), a series of NS-2 simulations were conducted to observe the video streaming performance as the optimal solutions for two different packet loss processes are calculated to obtain the frame-level FEC coding parameters (S_{IF}, S_{PF}, S_{BF}) and frame discarding parameters (N_{PD}, N_{BD}). In the simulation experiments, the congestion-induced packet loss rate in the wired link was 1%, and the TCP-friendly transmission rate was calculated as in Eq. 22. In the wireless link, the wireless packet loss rate was set in the range of 1% to 10% and the average burst packet loss length was assigned values of 1 to 10. The round trip time (RTT) was 50 ms. We assume ACK packets are always successful in transmission. The remaining settings keep unchanged. In the video sender, the test video clip (“Foreman”) was encoded in an MPEG-4 QCIF format (176 x 144) and transmitted to the wireless client at a rate of 30 frames per second.

The FEC coder applies the Reed-Solomon code *RS* (n, k) to the video packets, and the FEC coding rate is given by $R_c = k/n$. The lower coding rate indicates a larger amount of redundant packets ($n-k$) for k source video packets to provide the higher protection degree. According to the constraint on FEC redundancies defined

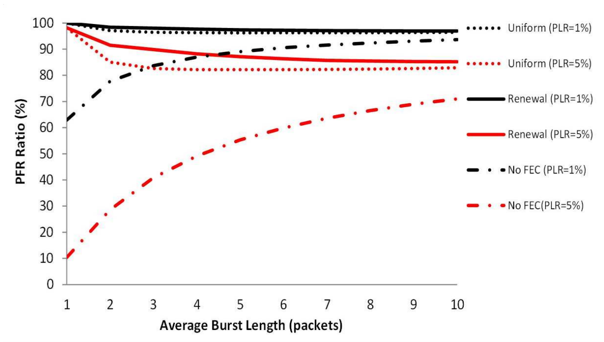


(a)

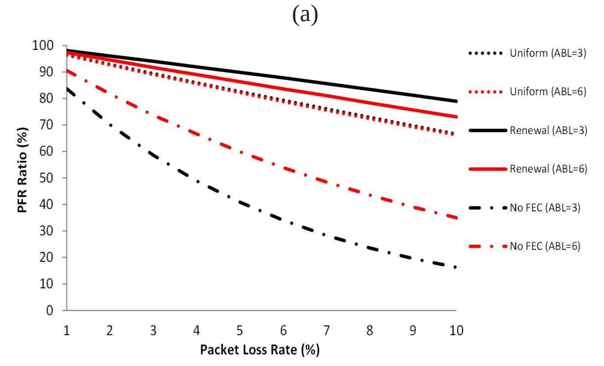


(b)

Fig. 4: The coding ratio comparison between uniform-PFR model and renewal-PFR model: (a) varied average burst length, (b) varied packet loss rate. For each compared model, coding rate curves for I, P, and B frame are plotted respectively



(a)



(b)

Fig. 5: Performance comparison for three different FEC schemes: (a) varied ABL; (b) varied PLR

in Eq. 24, the FEC coding ratio is ranged from 0.5 to 1. Figure 4 (a) shows the coding ratios of three frame types: I, P, and B, as the average burst loss length is varied from 1 to 10 and the packet loss rate is fixed to 5%. Noted that for P and B frame type, their coding ratio results are obtained from the statistic data of all available P and B frames in the GOP. From Figure 4 (a), we can observe that for the renewal-PFR model, the required coding ratios of I and P frame decrease to obtain the optimal FEC results as the burst loss length is increased. On the other hand, the coding ratio of B frame is increased to receive a lower error protection level with the longer average burst length. For the uniform-PFR model, which is unaware of burst loss, the coding ratios of three frame types keeps unchanged, and B frames have lowest coding rate among all frame types since the amount of B-frame packets is typically small to obtain the coding ratio of 0.5.

Figure 4 (b) shows the coding ratios of three frame types as the packet loss rate is varied from 1% to 10% and the average burst loss length is fixed to 3. For the renewal-PFR model, the UEP level provided by the three different frame types is apparent. Furthermore, all coding ratio curves maintain constant as the packet loss rate varies. This is because the renewal-PFR model attempts to combat with the long loss burst length of 3 in the presence of increased packet losses. For the uniform-PFR model, the coding ratio of I frame is decreased to obtain more FEC redundancies as the packet loss rate increases. As the packet loss rate is lower than 8%, the renewal-PFR model has lower coding rates of I frame than the uniform-PFR model. Generally, the renewal-PFR model applies stronger protection to high-priority video frames over the burst-loss transmission channel, comparing to the uniform-PFR model.

Figure 5 shows the optimal PFR ratios achieved by three cases: “No FEC”, “Uniform”, and “Renewal”. In the Figure 5 (a), the average burst length is varied from 1 to 10 as the packet loss rate is fixed to 1% and 5%. The PFR ratio is defined as the ratio of the measured PFR at the receiver over the encoding frame rate (i.e., 30fps). We can observe that both two cases of No “FEC”, has the increased PFR values as the average burst loss length increases, and all the other cases receives the decreased PFR values as the average burst loss length is increased. However, either the uniform-PFR model or the renewal-PFR model outperforms the “No FEC” case. Another result is shown in Figure 5 (b) as the average packet loss rate varies from 1% to 10% and as the average burst length is fixed to 3 and 6. As the packet loss rate increases, the PFR gap between the renewal-PFR model and the uniform-PFR model is increased accordingly for both cases of ABL=3 and ABL=6. From Figure 5, it can be seen that, for the renewal-PFR model: (1) the measured PFR value is higher than that of the uniform-PFR model and (2) the largest PFR gain of 12.34% occurs as the average burst length is 3 and packet loss rate is 10%.

5 Conclusions

In this paper, an analytical model is derived to evaluate the MPEG video delivery performance of frame-level FEC scheme over the burst-loss channels. The analytical results show that the burst loss affects the FEC efficacy, and the PFR model based on the uniform loss process easily lead a performance bias under the burst-loss condition. Through a series of NS-2 simulation experiments, our proposed model can calculate the optimal FEC coding parameters for the given system settings and obtains a better PFR results than the FEC model based on the uniform loss process. The future works are to include more FEC techniques, such as the interleaving, into the proposed model, and to use an n-state Markov model as the packet loss process to adapt the proposed model to the dynamically varying network situation in wireless mobile communications.

References

[1] J. Bolot and A. V. Garcia, "The Case for FEC-based Error Control for Packet Audio in the Internet," in Proc. ACM Int. Conf. Multimedia, (1996).

[2] H. Sanneck, G. Carle, and R. Koodli, "A Framework Model for Packet Loss Metrics Based on Loss Run length," in Proc. SPIE/ACM SIGMM Multimedia Comput. Network Conf., San Jose, CA, 177-187 (2000).

[3] B. Vuectic and J. Du, "Channel Modeling and Simulation in Satellite Mobile Communication Systems," IEEE J. Selected Areas Commun., **10**, 1209-1218 (1992).

[4] C. C. Tan and N. C. Beaulieu, "On First-order Markov Modeling for the Rayleigh Fading Channel," IEEE Trans. Commun., **48**, 2032-2040 (2000).

[5] Z. Li, J. Chakareski, X. Niu, Y. Zhang, and W. Gu, "Modeling and Analysis of Distortion Caused by Markov-Model Burst Packet Losses in Video Transmission," IEEE Trans. On Circuits and Systems for Video Technology, **19**, (2009).

[6] P. Frossard, "FEC Performance in Multimedia Streaming," IEEE Communication Letters, **5**, (2001).

[7] Huahui Wu, Mark Claypool, and Robert Kinicki, "Adjusting Forward Error Correction with Temporal Scaling for TCP-Friendly Streaming MPEG," ACM Transactions on Multimedia Computing, Communications, and Applications (TOMCCAP), **1**, (2005).

[8] Y. Yuan, B. F. Cockburn, T. Sikora, and M. Mandal, "Efficient Allocation of Packet-level Forward Error Correction in Video Streaming over the Internet," Journal of Electronic Imaging, (2007).

[9] S. R. Kang, and D. Loguinov, "Modeling Best-effort and FEC Streaming of Scalable Video in Lossy Network Channels," IEEE/ACM Trans. On Networking, **15**, 187-200 (2007).

[10] K. Kang and H. Shin, "Reduced Data Rates for Energy Efficient Reed-Solomon FEC on Fading Channels," IEEE Trans. On Vehicular Technology, **58**, 176-187 (2009).

[11] X. Yu, J. W. Modestino, R. Kurceren, and Y. S. Chan, "A Model-based Approach to Evaluation of the Efficacy of FEC Coding in Combating Network Packet Losses," IEEE/ACM Trans. on Networking, **16**, 628-641 (2008).

[12] A. Nafaa, "Forward Error Correction Strategies for Media Streaming over Wireless Networks," IEEE Communications Magazine, **46**, 72-79 (2007).

[13] C. C. Lai, P. W. Tiang, M. K. Abdullah, B. M. Ali, and M. A. Mahdi, "FEC Performance Analysis Based on Poisson and Bursty Error Patterns for SDH and OTN Systems," Photonic Network Communications, **11**, 265-270 (2006).

[14] W. T. Tan, and G. Cheung, "SP-frame Selection for Video Streaming over Burst-loss networks," In Proc. 7th IEEE. Symp. Multimedia, (2005).

[15] Z. Li, J. Chakareski, L. Shen, L. Wang, "Video Quality in Transmission over Burst-Loss Channels: A Forward Error Correction Perspective," IEEE Communications Letters, **15**, 238-240 (2011).

[16] J. Padhye, V. Firoiu, D. Towsley, and J. Kurose, "Modeling TCP Throughput: A Simple Model and Its Empirical Validation," In Proceedings of ACM SIGCOMM, (1998).

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