Improving the QoS of Wireless Video Transmissions via Packet-Level FEC

Ghaida A. AL-Suhail, Liansheng Tan

***Abstract* — Wireless video transmission suffers errors from the dynamic wireless environment. Due to errors, the discarded link layer packets impose a serious limitation on the maximum achievable throughput over wireless channel. To face this challenge and to enhance the overall TCP-Friendly video throughput, this paper proposes an MPEG packet loss model which is based on Forward Error Correction (FEC) over wired-to-wireless channel. Within this model, a FEC packet level scheme is used to act as an inter-protection control, which is based on Reed-Solomon (RS) code with the aim of bringing a robust transmission against frequent packet loss. Further, a BPSK scheme is applied for an Additive White Gaussian Noise (AWGN) wireless channel. By using this model, the predicted frame rate for MPEG video streaming can be estimated. Quality of Service (QoS) in terms of frame rate and the quality factor are evaluated for the predicted quantizer scale. The numerical results demonstrate that the proposed scheme improves the QoS of video transmissions in the presence of high wireless channel bit errors.**

Keywords— Wireless video, Video quality, TCP-Friendly, Quality of Service (QoS), Forward Error Correction (FEC), Reed-Solomon (RS) code.

1. INTRODUCTION

In practice, the major challenges of video transmission over wireless links are to deal with low bandwidth and high error rates due to the noise, interference, fading and shadowing. The bit stream video over noisy channel introduces symbol or bit errors causing packets corruption, which leads to degradation in the quality of reconstructed video sequence. A robust error protection, hence, for video traffic is required in order to achieve an acceptable video quality [**1-7**].

To provide a high quality of service (QoS) for video applications, i.e. high video play-out quality, at high loss rates over wireless links, it is important to use error-control techniques [9]. The physical layer mainly introduces a quick estimate of the performance over wireless link e.g. symbol or bit error rate (BER) versus Signal-to-Noise ratio (SNR) due to an Additive White Gaussian Noise (AWGN) over wireless channel. To facilitate efficient support of QoS for video applications, measurements of physical layer; such as a radio-link BER, channel SNR, Doppler spectrum and channel capacity; are reported to the upper-layer for channel state estimation. TCP flow or TCP-Friendly flow at transport layer varies in a consequence to channel state estimation by controlling the sending rate in highly reliable transmission. Both are connection-oriented protocols and avoiding network congestion collapses comparing with UDP protocol [**4**].

While multimedia applications can tolerate some data loss, excessive packet loss during congestion over wired link and/or high bit errors over wireless channel yields unacceptable media quality. Since video coding involves interframe dependencies to achieve high compression rates, the random dropping of packets by routers and/or random bit errors due to a highly additive Whit Gaussian noise over wireless can both seriously degrade video quality. Hence in wired MPEG transmission [**8**], for example, dropping packets from an independently encoded I frame causes the following dependent P and B frames to be fully undecodable. In practice, interframe dependencies have been shown to cause a 3% packet loss rate to result in a 30 % frame loss rate.

To address the above interaction, we should provide a high quality of service (QoS) for video applications, by meaning high video play-out quality, at high loss rates over wireless link; whilst several studies [**3-12**] have pursued both error-control techniques of media adaptation, as well as network-adaptation. The network-adaptation can be efficiently employed by adapting the end-system to the changing network conditions, whereas adaptation in general meaning represents the ability of network protocols and applications to observe and respond to the channel variations. Thus there are three error control techniques widely used in various settings: *Retransmission*, *Redundancy* and *Interleaving* [**1-5, 14-20**]. These approaches are used either separately and/or jointly by cross-layer scheme in order to combat the overall packet loss over Internet network.

In this paper, BPSK (Bi-Phase-Shift-Keying) scheme is used over AWGN wireless channel to express the exponential packet loss over the channel [**4**] when the state condition is poor. Furthermore, to avoid the latency (delay) and variance in latency caused by re-transmission of lost packets over a hybrid network, MPEG packet loss model based packet-level FEC is considered as in [**21**] including Reed Solomon (R-S) code [**15**] in the application layer in order to reconstruct the overall lost video packets. A FEC adds a redundant repair data to the original video stream. Many current approaches [2**, 14-17**] use either a priori, static FEC choices or FEC that adapts to

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perceived packet loss on the network; meanwhile Wu and Claypool [**8, 21**] have dealt with adaptive FEC schemes which accounts for the additional FEC overhead against a capacity constraint. In fact, by adding FEC the capacity constraint means a significant reduction in the effective transmission rate of the original video content.

We therefore estimate the predicted frame rate for MPEG video streaming by using a Variable Frame Rate based on TCP-Friendly Rate Control model in [**21**] over a combined network of wired link and AWGN wireless channel for under utilized bandwidth. Meanwhile, the improvement in the effective range of channel SNR can be achieved when FEC based packet level is setting at a fixed certain value. As a result, Quality of Service (QoS) in term of SNR scalability is exploited for the predicted quantizer scale (Q) if the network throughput is assumed to be equal the available bandwidth.

The remainder of the paper is arranged as follows: Section 2 describes a brief background for the work in this paper; Section 3 presents the analytical MPEG packet loss model over combined wired/wireless channel. Through the illustrative results in Section 4, the video quality is depicted for the predicted frame rate as well as the QoS in terms of SNR scalability. Finally, Section 5 summarizes the paper and introduces possible future work.

2. BACKGROUND

2.1 MPEG Video

MPEG is a popular standard for video compression. **Figure 1** illustrates a typical GoP (Group of Pictures) structure of an MPEG stream. Each GoP consists of three types of frames: I-, P- and B-frames. An I-frame (Intra coded) located at the head of a GoP is coded as a still image and serves as a reference for P and B frames. P-frames (Predictive coded) depend on the preceding I or P-frame in compression. Finally, B-frames (Bi-directionally predictive coded) depend on the surrounding reference frames, that are, the closest two I and P or P and P frames. The loss of one P frame can make some of other P and B frames undecodable, and the loss of one I frame can result in the loss of the whole GoP. This implies that I frames are more important than P frames, and P frames are more important than B frames [**8-9**]. A GoP structure is expressed as (N , M), where M corresponds to the number of P frames in a GoP and M corresponds to the number of B frames between I and P frames. In Fig. 1, N =3 and M =2.

2.2 Video Quality

Traditionally video quality is measured by distortion given by Peak Signal-to-Noise Ratio (PSNR) [**18**]. It has been noticed that PSNR is proportional to the video goodput defined by useful data bits per second received by the end clients after adding FEC, which gives the residual packet error rate is below a certain low value $(p \le 3\%)$ [**19**]. In MPEG coding, specific quantizer scale against each block of 16x16 pixels is performed. For a large quantizer scale, the quality of decoded block becomes poor. It means this scale leads to degrade image SNR values [**9**]. On the other hand, the timely resolution is related to the number of frames per second [fps]. This rate can be regulated by means of a frame dropping technique. The required bandwidth $BW(R,Q,F)$ in [bps]

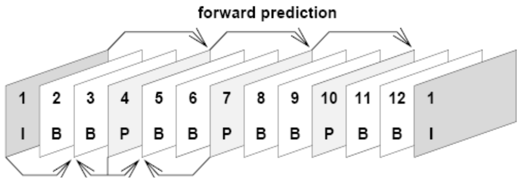


Figure 1. A typical MPEG Group of Pictures

can be estimated in terms of spatial resolution (R [pixels]), PSNR resolution (Q) and the timely resolution (F [fps]) as

$$BW_{R,Q,F} \cong \left(\frac{1}{3.1}\right)^{\log_4\left(\frac{R}{640x480}\right)} \left(0.151 + \frac{9.707}{Q} - \frac{4.314}{Q^2}\right) \frac{F}{30} BW_{Base}, \qquad (1)$$

where BW_{base} indicates the peak bit rate of the reference stream [9].

2.3 Forward Error Correction (FEC)

To improve the video quality under transmission errors, error-control schemes can be performed at the source or channel coding stage. Studies [2-5, 7, 9] introduce source coding schemes, like reversible variable-length coding (RVLC) and multiple description coding (MDC). Another approach by using channel coding schemes protects the integrity of bit stream, such as Forward Error Correction (FEC) codes or Automatic Repeat Request (ARQ). The choice of a particular scheme depends on channel characteristics, statistics of channel errors, delay constraint, and type of services at the end users. Since the network conditions generally cause errors on network packets, hence correction of these errors is in the subject of “Forward Error Correction” (FEC). A FEC is mainly divided into two categories: bit-level FEC and Packet-level FEC. These two categories are unfamiliar [**14**]. Recently, for example Demir and *et al.* [**17**] have studied two techniques: a *Reed-Solomon* FEC which is found widely on the wired Internet; and *Raptor code* which is a commercial and not used broadly yet unless in few new technologies such as MBMS, DVB.

More precisely, Reed-Solomon (R-S) code is a media-independent FEC technique that can be applied at the packet level for *m*-bit symbols (maximum *m* is 8 bits for byte-oriented computer applications) [**15**]. As shown in **Figure 2**, an application level video frame is modeled as being transmitted in K packets where K varies with frame type, encoding method, and media content. R-S code adds ($N-K$) redundant packets to the K original packets and sends the N packets over the network. Although some packets may be lost, e.g., packet 2 in **Figure 2**, the frame still can be completely reconstructed if any K or more packets are successfully received. For example, in [**14**] Lee and *et al.* investigated video delivery of optimal allocation FEC based on packet-level (i.e., the number of packet level FEC parity bits per second) as well as byte-level (i.e., the number of byte-level FEC parity bits per second) from the server over hybrid wired/wireless network in order to serve maximum video quality for multicasting transmission.

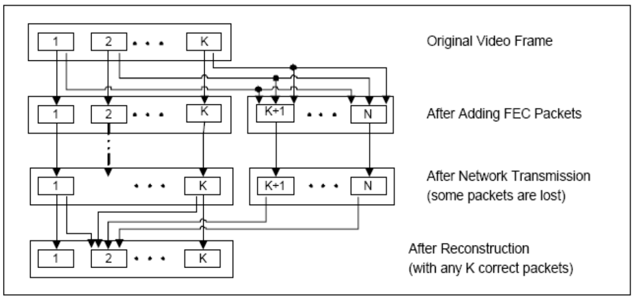


Figure 2. A block diagram of Reed-Solomon code.

In this paper, we present R-S code only on packet level for video streaming over a combined unicast wired and wireless network. To analyse the effects of FEC on the application layer frames, the sending of packets is modelled as a series of independent Bernoulli trials. Thus, the probability $q(N,K,p)$ that a K packets video frame is successfully transmitted with $N-K$ redundant FEC packets along a network path with overall packet loss probability p is

$$q(N,K,p)=\sum_{i=K}^{N}\left[\binom{N}{i}(1-p)^{i}\,p^{N-i}\right]. \qquad (2)$$

Hence, Section 3 examines the effective range of channel SNR when bit error rate is high and FEC-packet level is considered over the combined wired and wireless network.

3. Problem Formulation

3.1 Wireless Channel Model

In this paper, we consider using a TCP-Friendly Rate Control (TFRC) scheme [**4, 21**] as an underlying rate control and adjusting video traffic to the channel condition, i.e., the available bandwidth. The target sending rate T of a TFRC session is derived as,

$$T=\frac{S}{t_{RTT}\sqrt{\frac{2p}{3}}+t_{RTO}\sqrt{\frac{27p}{8}}\,p\left(1+32p^{2}\right)}, \qquad (3)$$

where p stands for the packet loss probability, i.e., loss event rate, S is the packet size [byte], t_{RTT} is the round-trip time [sec], and t_{RTO} is the TCP retransmit time out value [sec]. By regarding T as the available bandwidth for video streaming and adjusting the video traffic, we can expect the high-quality video play-out at a receiver. However, a source node cannot distinguish packet losses caused by bit errors on wireless links from those caused by buffer overflow.

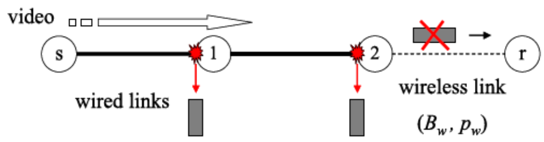


Figure 3. A typical wired/wireless video streaming model.

A typical model of video streaming over wired and wireless links can be considered as shown in **Figure 3**. The wireless link is characterized by available bandwidth B_w and packet loss rate p_w. Then, a brief scenario in [**4**] can be applied when there is no cross-traffic at either node 1 or node 2. We make the following assumptions:

1. The wireless link is assumed to be bottleneck of the network by meaning no congestion at node 1.
2. Packet loss is assumed only due to wireless channel bit errors and the buffer at node 2 does not overflow, as $p_c = 0$.
3. In consequence $t_{RTT} = t_{RTT\,\min}$, i.e., the minimum RTT, if $T \le B_w$ [**4**].
4. B_w and p_w are constants. p_w is assumed to be random and stationary [**4, 11**].
5. The backward route from receiver r to server s is assumed to be congestion-free but not error-free due to bit errors.

Here, the video sending rate is smaller than the bottleneck bandwidth and should not cause any network instability, i.e., congestion collapse. Additionally, the optimal control should result in the highest possible throughput and the lowest packet loss rate. To derive the target sending rate which satisfies them by using (3), packet loss rate p is now defined by two independent loss rates p_w and p_c as, $p = p_w + (1-p_w)p_c$. Since p_w gives the lower-bound for p for $p_c = 0$, the upper-bound of the network throughput becomes

$$T \le \frac{S}{t_{RTT\min}\sqrt{\frac{2p_w}{3}}+t_{RTO}\sqrt{\frac{27p_w}{8}}\,p_w\left(1+32p_w^{\,2}\right)} = T_b. \qquad (4)$$

Hence, for an under-utilized channel, $T_b < B_w$ holds when only one TFRC connection exists.

To obtain p_w, we have to consider frequent bit errors of a wireless channel with AWGN ignoring fading effect where BPSK scheme is applied. With an ideal assumption that any bit error in a packet leads to a loss of the whole packet, we can estimate the packet loss probability p_w as the channel bit error rate p_e. BER performance of uncoded BPSK scheme is given by [**1, 5**] as

$$p_e = Q\left(\sqrt{\gamma}\right) = Q\left(\sqrt{\frac{2E_b}{N_o}}\right), \tag{5}$$

where E_b stands for the bit energy, N_o is the noise power, and $\gamma = 2E_b/N_o$ represents the total channel SNR of a BPSK scheme. The Gaussian cumulative distribution function is being $Q(.)$.

For robust transmission over the hybrid link when a bit stream are highly corrupted due to AWGN wireless channel environment and without altering the sending rate it needs to repair the losses locally using either Forward-Error-Correction (FEC) or Automatic Repeat-Request (ARQ). In this case, wireless model considers protected packets using an inter-packet FEC in application-layer at the video source s. The protection deals with Reed Solomon (R-S) code to encode the video packets before transmitting them over the network.

3.2 MPEG Packet-loss Model

This section considers the details of Wu and Claypool's VFR-TCP model in [**21**] as an algorithm to estimate the number of playable frames at a receiver behind wired links and a wireless link, where random and stationary packet losses occur. In this model, we employed TFRC to control the sending rate in accordance with loss of packets caused by packet corruptions for bit errors over a wireless channel. Here, we adopt the assumption of a frame-dropping mechanism to compensate the varying TCP-Friendly sending rate. Frames are also dropped, or lost, by corruption of packets. If the quality of a frame in terms of PSNR falls below a pre-determined threshold $PSNR_{threshold}$, the frame is considered lost. The resultant frame rate F can be estimated as follows. When we consider the Bernoulli packet loss model, the observed frame rate F can be expressed as,

$$F = f_o\left(1-\phi_R\right), \tag{6}$$

where ϕ_R stands an effective "*frame drop rate*", i.e., the fraction of frames dropped, and f_o [fps] is the frame rate of the original video stream [**6**]. If quality scaling is applied, a constant f_o is replaced with a variable f_r. The frame rate f_r is further replaced by $GS_{GOPsize}$, where G corresponds to the number of GoPs per second and $S_{GOPsize}$ is the number of frames in a GoP. Therefore,

$$F = GS_{GOPsize}\left(1-\phi_R\right). \tag{7}$$

The frame drop rate ϕ_R can be formulated by

$$\phi_R = 1 - \frac{X_R}{S_{GOPsize}}, \tag{8}$$

where X_R is assumed to run over the playable frame rates f_i's of the *i*-frame type in GOP, i.e., I-, P- and B- frames. By using Bernoulli trails model for the sending of packets the probabilities of successful frame transmission $q(N,K,p)$ for *i*-frame types are defined as,

$$q_i = q(S_i + S_{iF}, S_i, p), \tag{9}$$

where q_i is defined as in Equation (2) and consequently the term X_R can be expressed in terms of $R_i[X_R]$ using the total playable frame rate based TCP-Friendly [2**1**] to obtain

$$R = \sum_i R_i, \tag{10}$$

where R_i stands the playable frame rate of ***i***-frame type in GOP (N_P, N_{BP}) and in accordance the number of frames in each GOP is expressed as,

$$S_{GOPsize} = 1 + N_P + N_B, \tag{11}$$

In a similar manner as [21], by rewriting Equation (11), the predicable play-out frame rate can be derived as

$$R = G.q_I\left[1 + \frac{q_p - q_p^{\,N_P+1}}{1-q_p} + N_{BP}.q_B\left(\frac{q_p - q_p^{\,N_P+1}}{1-q_P} + q_I q_P^{\,N_P}\right)\right]. \tag{12}$$

The GOP parameters are treated as variables for MPEG video stream as follows:

N_P : Number of P-frames in a GOP

N_B : Total number of B-frames in GOP, $N_B = (1+N_P)\times N_{BP}$.

N_{BP} : Number of B-frames in a GOP in an interval of I- and P-frames.

S_i : Size of *i*-frame [in packets]

S_{iF} : Size FEC-packet level for *i*-frame [in packets].

The strategy in this model is to assume that the network is able to provide an estimate of the current network loss probability (due to congestion and/or high bit errors) and round-trip-time while the MPEG application can provide details on the video characteristics. In consequence, the model can be used to chose GOP pattern to obtain the reasonable expected playable frame rate that are compatible with the full video motion [**8, 21**].

4. NUMERICAL RESULTS

4.1 Methodology

Based on the above assumptions, we develop the following illustrative steps to find the optimal playable frame rate for QoS requirements based on the given scenario of Section 3.1.

1. Obtain a channel SNR per bit $\gamma/2$ on wireless link.
2. Assess the bit error rate (BER) p_e from the channel SNR by Equation (5) for uncoded BPSK modulation scheme. Then the packet loss rate over a wireless link is defined as $p_w = p_e$.
3. TFRC rate is evaluated by Equation (3), which must satisfy the condition of (4) substituted the obtained p_w.
4. Determine video quality in terms of the temporal scalability, i.e., frame dropping, to regulate the sending rate to the TFRC rate.
5. For all possible GoP structures, one with the maximum frame rate is chosen.
6. The overall frame drop rate (ϕ_R), hence, is estimated by using Equation (8).

Now if the base rate BW_{base} is known, quality scaling can be applied to all of the spatial, temporal, and SNR scalabilities by using (1). During a video streaming session, a server regulates R, F and Q to adjust the sending rate to the TCP-friendly rate as in VFR-TFRC model in [**11**].

As a consequence, a strategy to achieve the optimal performance for an application is to increase the number of connections until the total throughput reaches the *hard limit* of $B_w(1-P_w)$. With the fixed p_w, the total throughput increases with the number of connections up to a certain point, after which there is a saturation effect.

4.2 Results Analysis

Simulation results have been obtained for a typical 1xRTT CDMA wireless network model [**4, 11**]. On some reasonable constraints, the results are based on Table 1 for the fact that a typical maximum frame rate allowed over Internet is 30 [fps] for full motion video and a recommended typical GoP is 12 frames, such as 'IBBBPBBBPBBB' GoP(2,3), for optimal performance. Furthermore, a channel capacity is assumed not exceeding limited bandwidth B_w, which represents a maximum throughput for wireless link.

By using the given scenario in Section 3.1, node 2 is assumed within no congestion, i.e. p_c =0, hence we changed SNR of a wireless channel to evaluate the TCP-Friendly throughput for each video connection. **Figure 4** (a) and (b) show the maximum number of video connections n_{opt} over the effective channel SNR range and channel error rate. It should be noticed that with the packet loss rate $p_w = 4.3\%$ and without error control, which implies the channel SNR is 1.68 [dB], the optimal number of connections is around 4 or 5 as shown in [**4**].

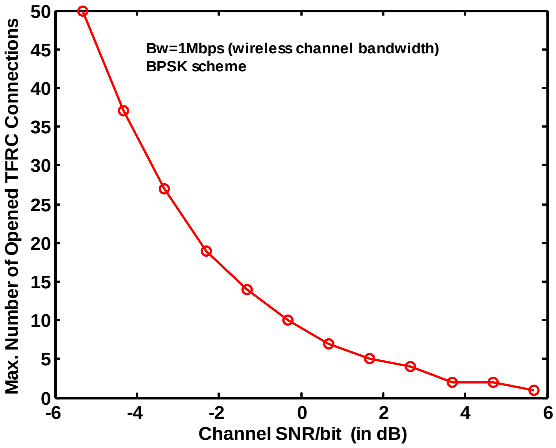
TABLE 1
PARAMETER SETTING IN SIMULATION OVER WIRELESS

$t_{RTT} = 168\,ms$ $t_{RTO} \cong 4t_{RTT}$	**S** =1 Kbytes **I**-Frame=25 packets
$B_w = 1Mbps$	**P**-Frame=8 packets **B**-Frame=3 packets
Peak rate =144kps for one user	
Channel SNR per bit	6dB to -10dB
Bit error rate (packet level) p_w	0.33% *to* 22%

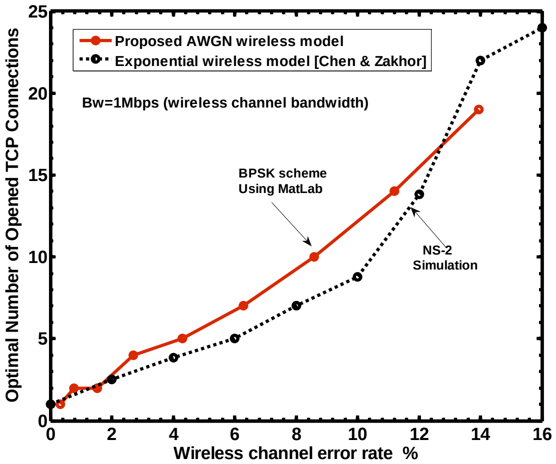
In order to evaluate the improvement in playable frame rate for each video TFRC throughput connection, we applied error control scheme based fixed FEC for small, medium and large code. **Figure 5** evaluates the total effective channel SNR per bit for certain FEC. For example, the playable frame rate is clearly increased at 5.68 [dB] to achieve 20.68 [fps] for small FEC (1,1,0) and degraded to 16 [fps] for large FEC (8,4,1); whilst the medium and large FEC values improves significantly the performance at low values of channel SNR as compared with no FEC employing. Also, it is noticed that the frame drop rate is degraded as FEC value increases. However, the resultant F changes depending on the interaction of GOP frames. In other words, chosen value of S_{PF} or S_{BF} has a slightly effect on the resultant F as compared with chosen values of S_{IF}.

Moreover, a comparison with Wu and Claypool VFR-TFRC model over Interne [**8**], depict more improvement in playable frame rate up to 30 [fps] for total packet loss rate $p \le 2\%$ (See **Figure 6**). This is the highest among all others frame rates, but the rate is not TCP-friendly over wireless Internet channel. Specifically, in Fig. 5 (b), the frame drop rate decreases as the wireless channel state improves using error control. This leads an increasing in playable frame rate at the receiver and achieving a reasonable video quality due to transmission over wireless channel.

Figure 7 depicts the video quality, in term Q, as a function of the resultant play-out frame rate for a single TFRC connection. An original video stream has the spatial resolution of 640x480 [pixels], the temporal resolution of 30 [fps], and the SNR resolution of 10 as a quantizer scale value. The coding rate of the original video stream is 144 [kbps]. Using (1), we derive the SNR scalability Q by substituting the TFRC sending rate as the resultant required bandwidth $BW(640x480, Q, 30)$. Therefore, X-axis and Y-axis are indirectly related to each other through the channel error rate or TFRC rate. In other words, it is noticed that the video quality Q is independent on the GoP pattern structure. Also, when error control of FEC-based packet level is used to evaluate the corresponding improvement, it is found that the quality scale decreases rapidly to be less than 5 on low SNR values of channel state.

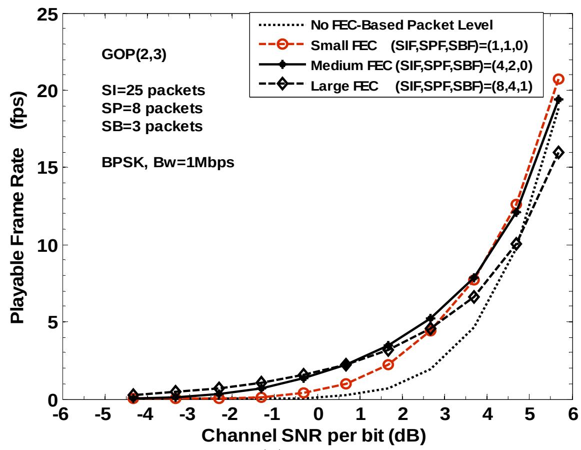


(a)

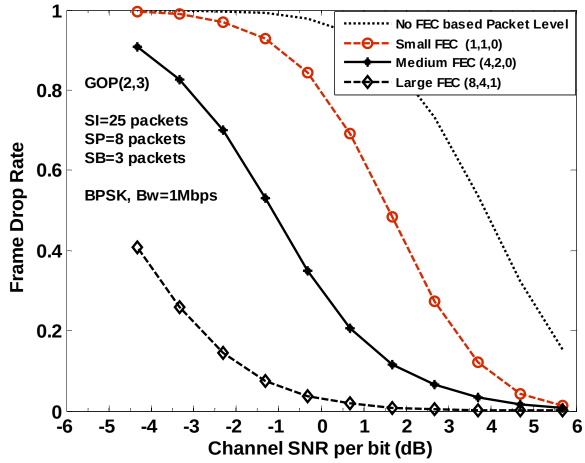


(b)

Figure 4. TFRC performance over wireless link. (a) Maximum number of TFRC connections, (b) Comparison between schemes



(a)



(b)

Figure 5. One video connection as a function of a channel SNR (a) Playable frame rate and, (b) The Predicted frame drop rate

Table 2 illustrates optimal video quality performance for GoP(2,3) over wireless link using the indirect dependence via the channel error and TFRC rate (throughput) of Fig. 4 (b). Depending on preferences on the perceived video quality, one can choose the temporal scalability or the SNR scalability as quality scaling. When the temporal scalability is applied, video play-out becomes choppy, intermittent, or like a series of still images [9]. On the other hand, the low SNR scalability results in coarse and mosaic appearances in the case of small FEC or ignoring of FEC.

5. Conclusions

This paper has applied a frame dropping mechanism for variable frame rate (VFR) model which is based on TCP-Friendly rate control assuming under utilized bandwidth over a combined wired/wireless channel. The proposed scheme has provided QoS estimation for the video streaming in terms of frame rate and as well as the quality factor (Quantizer factor Q). Numerical results showed that the proposed model introduces a good robustness QoS for video wireless link using Reed Solomon code as FEC-based packet level. It is also found that the model based TCP-Friendly rate control increases tolerance to packet loss due to high bit errors and achieves a good quality compared with TFRC rate transmission over wired Internet. Further work can be extended to involve multicasting video. Also, byte- or bit level FEC can be applied for physical layer over wireless link when multi-path fading channel is considered. Furthermore, a study can involve a number of TFRC connections for full-utilized bandwidth.

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TABLE 2
OPIMAL VIDEO QUALITY PERFORMANCE FOR GoP(2,3) OVER WIRELESS LINK

	Play-out Frame Rate (fps)			
Quality factor (Q)	**No FEC (0,0,0)**	**Small FEC (1,1,0)**	**Medium FEC (4,2,0)**	**Large FEC (8,4,1)**
2	18.8	20.68	19.4	15.98
4	4.6	7.7	7.86	6.65
6	1.92	4.42	5.24	4.55

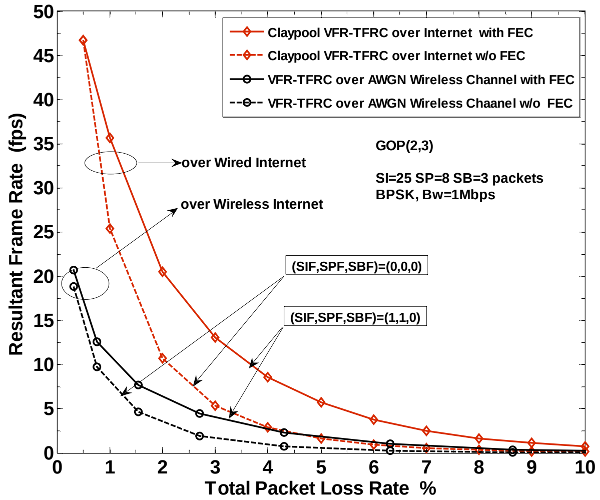

Figure 6. Comparison of playable frame rate for only one video connection as a function of total packet loss.

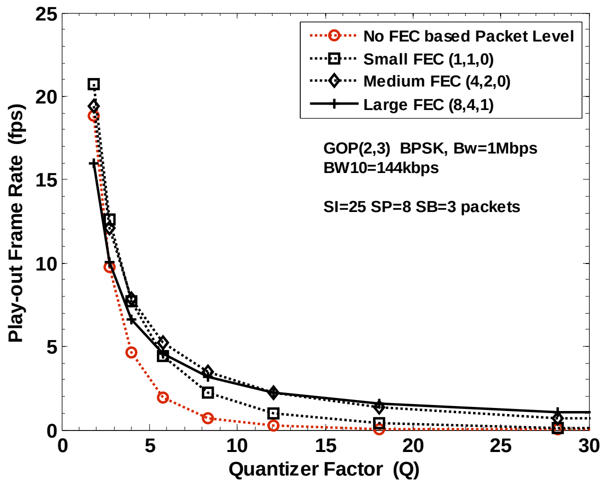

Figure 7. Play-out frame rate for only one video connection as a function of video quality factor

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