Improving Wireless Video Performance via Packet-Level FEC

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ABSTRACT:

Wireless video transmission is often vulnerable to errors from the noisy 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. A FEC packet level scheme is used to act as an inter-protection control based on Reed-Solomon (RS) code providing a robust transmission against random packet loss. Hence, the predicted frame rate for MPEG video streaming can be estimated. Quality of service (QoS) in terms of frame rate and quality factor (Quantizer Scale) is also evaluated under various FEC code conditions. The new results demonstrate that the proposed scheme improves a video quality performance in high wireless channel bit errors in spite of an increment in the overhead packets due to FEC code.

Key words: *Video Streaming, Wireless Video, TCP-Friendly, FEC, Error Control*.

تحسين أداء الفيديو اللاسلكي بأستخدام تقنية تصحيح الخطأ المتقدم في مستوى الرزمة

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الخلاصة

إن أرسال الفيديواللاسلكي يكون بشكلٍ كبيرٍعرضـه للأخطـاء بسـبب البيئـة المشوشة. فالرزم المهملة في طبقة التوصيل بسبب هذه الأخطاء تفرض حدود جـادة اللاسكية على الكفاءة القصوى الممكن تحقيقها عبرالقناة اللاسلكية. ومن اجـل تحسـين الكفـاءة الكليـة ، فإن هذا البحث يقترح نموذج (موديل) فقـد TCP-Friendly لفيديويعتمد بروتوكول نقل نوع عبر قناة سلكية-الـى-(FEC) المعتمد تقنية تصحيح الخطأ المتقدم MPEG رزمة الفيديو نوع عند مستوى الرزمة inter-protection قناة لاسكلية. التقنية المستخدمة تعمل كسيطرة حماية نوع العشوائي للرزمـة. من اجل توفيرأرسال قوي مضاد للفقد Reed-Solomon باستخدام تشفير وبناء عليه، يتم تقييم معدل مقاطع الصورة المتوقع لتدفق الفيديو. وكذلك يتم تقيـيم جـودة الأداء عند حالات مختلفة مـن تشـفير (Quantizer Scale) بدلالة معدل المقاطع مع مقياس الجودة الجديدة تبين ان التقنية المقترحة تُحسن أداء جودة الفيديو لمعدلات خطأ البـت . النتائج FEC العالية في القناة اللاسلكية على الرغم من التزايد في حجم الرزم بسبب التشفير.

1. INTRODUCTION

Recently, the rapid improvement in wireless communications [Goldsmith, 2005] as well as the increasing demand for real-time multimedia applications such as video conference, video telephony, video-on-demand (VoD) are both attracted issues to transmit video over hybrid wired/wireless Internet [Tripathi 2002;Yuan 2006]. However, today's Internet does not provide the necessary Quality of Service (QoS) guarantees that are needed to support high-quality video transmission. In addition, multimedia data over Internet often suffers from bandwidth, delay, jitter and packet loss. Therefore, growing requirement towards fairly transport protocols for video streaming such as TCP-Friendly has been devoted in many researches in order to achieve the fairness among TCP and UDP [Wakamiya 2000; Aramvith 2001; Handley 2003; Yang 2004;; Chen 2006]. In those works, "TCP-Friendly" is defined as a non-TCP connection should receive the same share of bandwidth as a TCP connection if they traverse the same path".

On the other hand, in practice the major challenges of video traffic through the wired and wireless Internet links are to deal with high packet loss rate due to the congestion of buffer overflow over wired networks; whilst wireless links are suffering from low bandwidth and high error rates due to the noise, interference, fading and shadowing. More precisely, the bit stream video over a noisy channel introduces bit errors causing packets corruption, which leads to a significant degradation in the quality of reconstructed video sequence. Thus a robust transmission of real-time video over wireless channels is still open issue to achieve good perceptual quality at the client terminal end [Chen 2006; Yang 2004].

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 MPEG video coding involves interframe dependencies to achieve high compression rates, the random dropping of packets by routers and/or random bit errors over a noisy wireless networks can both seriously degrade video quality. In wired Internet [Wu 2005], the 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, a high quality of service (QoS) for video applications, by meaning high video play-out quality, is required at high loss rates over wireless link. Several studies [Aramvith 2001; Li 2001; Fukuda 1997; Lee 2001; Chen 2006; Tan 1999; Wu et. al. 2005] 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 [Feamster 2001; Rejaie 1999; Yuan 2006; Tripathi 2002; Wang 1998]. These approaches are used either separately or jointly in order to combat the overall packet loss over Internet network.

In this paper, to avoid the latency (delay) and variance in latency caused by re-transmission of lost packets over a hybrid network, we propose MPEG packet loss model [Wu and Claypool, 2003] including Reed Solomon (R-S) code [Reed and Solomon, 1960] as a Forward-Error-Correction (FEC) 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 approaches [Tripathi 2002; Lee 2001; Demir 2006, Al-Suhail 2007] use either a priori, static FEC choices or FEC that adapts to perceived packet loss on the network; meanwhile Wu and Claypool, in 2005, have improved an adaptive FEC scheme by adjusting also the quality scaling (in terms of quantization level). This approach 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.

In addition, the physical layer of wireless link can estimate the performance such as bit error rate (BER) versus Signal-to-Noise Ratio (SNR) in accordance to the varying in the channel state [Ericsson 1999]. 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. Contrarily, TCP or TCP-Friendly flow at transport layer varies in a consequence to channel state estimation by controlling the sending rate in a highly reliable transmission. Both are connection-oriented protocols and avoiding network congestion collapses comparing with UDP protocol.

We therefore estimate the predicted video quality for MPEG video streaming by using a variable frame rate based on TCP-Friendly Rate Control model in [Wu 2003] over a combined network of wired link and wireless channel.

The random bit errors and a wireless channel in bad condition are both assumed for under utilized bandwidth. A Bi-Phase-Shift-Keying (BPSK) scheme is considered to define the exponential packet loss over a noisy wireless channel, and to match MULTFRC model [Chen and Zakhor, 2006; Al-Suhail and Wakamiya 2006] over 1xRTT CDMA wireless network. Within this model, we investigate the improvement in the effective video quality when FEC codes are applied at fixed certain values. Quality of Service (QoS) in terms of frame rate and SNR scalability (Quantizer Scale) is also evaluated 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 related to the work; Section 3 presents the proposed approach for MPEG video streaming based FEC scheme over a wired-to-wireless network. The simulation results are presented in Section 4 for the predicted video quality. Finally, Section 5 summarizes the paper and introduces possible future work.

BACKGROUND

MPEG VIDEO QUALITY

Traditionally video quality is measured by distortion given by Peak Signal-to-Noise Ratio (PSNR) [Ortego et al., 1998]. It has been noticed that PSNR is proportional to the video good put defined by useful data bits per second received by the end clients after adding FEC, which gives the residual packet error rate below a certain low value $(\leq 3\%)$ [Wang 1998]. In MPEG coding shown in Fig. 1, a 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 [Li 2001]. On the other hand, the timely scalability is related to the number of frames per second [fps]. This rate can be regulated by means of a frame dropping technique. Each video sequence consists of a cyclic sequence of GOPs, such as IBBPBB for GOP(1,2).

Additionally, several quality scaling schemes have been devoted on such as adaptive quantization values [Tan 1999] to adapt the encoding quantization value to network capacity; signal-to-noise ratio (SNR) scalability [Rejaie 1999] which encodes a video clip into multiple layers and streams it as many layers as possible; MPEG-4 fine granularity scalability (FGS) [Li 2001] as a special case of SNR scalability that provides continuous scalability using partial enhancement; and scalable MPEG (SMPEG) [Yuang 2004], which transcodes MPEG's DCT coefficients to a base level plus three enhanced levels and transmits different numbers of levels. In 2005, Wu and et al. investigated adjusting FEC with quality scaling using analytical model (QAFEC) to capture the quality distortion of MPEG stream in the presence of quality scaling and frame loss. In 2006 [Yuan et al.], a GOP based FEC for the source video has also been developed providing only a better playable frame rate (temporal scalability) than the classical packet-level FEC techniques for MPEG-2 video stream over wired Internet but with high complexity of packet generation.

In this paper, we use a common relationship of rate-distortion (R-D) formula to describe the QoS parameters of MPEG video sequences where the required bandwidth is independent on the video content. Thus the required bandwidth $BW(R,Q,F)$ in [bps] can be estimated in terms of spatial scalability (R [pixels]), PSNR scalability (Q) and the timely scalability (F [fps]) as

$$BW_{R,Q,F} \cong (3.1)^{-\log_4(\frac{R}{R_{\max}})} \times \left(\alpha_1 + \alpha_2 \times Q^{-1} - \alpha_3 \times Q^{-2}\right) \times F_{30} \times BW_{Base}$$

(1)

The coefficients $\alpha_1, \alpha_2,$ and α_3 are related to the video encoding at the server, and F_{30} is being an effective temporal scaling normalized by a reference frame rate 30 [fps]. BW_{base} indicates the peak bit rate of the reference video stream [Fukuda 1997].

FEC: FORWARD ERROR CORRECTION

To improve the video quality under transmission errors, error control schemes can be performed at the source or channel coding stage. Studies [Tripathi 2002; Aramvith 2001; Chen 2004; Li 2001; Wang 1998] 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) 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. Recently, two alternative packet-level FEC codes [Demir 2006] are used with different network conditions: 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 and DVB-H. Figure 2 shows the idea of FEC in a packet-oriented video transmission scheme by generating redundant packets at the sender. These codes can be used at the receiver to recover lost video data packets.

In this subsection, we briefly introduce Reed-Solomon (R-S) code which is a media-independent FEC technique that can be applied at the packet level [Reed and Solomon, 1960]. It enables high- and stable-quality of video transmission by protecting the video data against packet loss, whereas this code has maximum distance separable code. In fact, there are no other codes that can reconstruct erased symbols from a smaller number of received code symbol.

As shown in Fig. 3, an application level video frame is modeled as being transmitted in K packets where K varies with frame type, encoding method, and media content. RS 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 Fig. 3, the frame still can be completely reconstructed if any K or more packets are successfully received. For example, in 2001 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.

In this paper, we present this RS (N,K) code at a packet-level under a lossy wireless environment for video streaming over a combined unicast wired and wireless link in Fig. 4. Not that the scheme depicted in Fig. 3 has no additional delay at the sender. The sender has only to store copies of the information packets until K packets have been sent. Then $N-K$ redundancy packets are generated and transmitted after the last information packet. RS (N,K) can generally correct $\varepsilon = \left\lfloor \frac{N-K}{2} \right\rfloor$ packet errors. With the knowledge of the packet position, it can correct up to $\varepsilon = N-K$ packet errors, that is, the information packets can be reconstructed from any subset of K correctly received packet using erasure decoding [McAuley, 1990]. Thus our FEC scheme requires a receiver buffer which can at least hold K packets. However, the receiver does not need defer play-back as far as there is no packet loss. Even when one or more packets are lost, they are recovered as soon as the receiver obtains K packets.

To evaluate the effectiveness of FEC on the application layer frames, the sending of packets is modelled as a series of independent Bernoulli trials. It means we need to know the probability that more than $N-K$ packets are lost in a certain network condition. Then, we can compute this probability $P_e(n,N)$ that n out of N packets are lost. $P_e(n,N)$ is called the block error density function. It is a simple binomial distribution in the case of memoryless channel with packet loss probability as follows [Miyabayashi 2002],

$$P_e(n,N) = \binom{N}{n} P_{link}^n (1-P_{link})^{N-n} \tag{2}$$

where P_{link} is the actual packet loss probability observed in the network. We can derive the probability P_{video} that a K packets video frame is successfully transmitted with $N-K$ redundant FEC packets along a network path with overall link packet loss probability P_{link} by using the following equation.

$$P_{video} = 1 - P_{lost} = 1 - \sum_{m=N-K+1}^{N} P_e(m,N) \tag{3}$$

That is, P_{lost} denotes the probability that the video packet is lost, and the probability that the video packet is successful transmitted can be expressed as,

(4) .

$$P_{video} = P(N,K,P_{link}) = 1 - \sum_{n=N-K+1}^{N} P_e(n,N) = \sum_{n=K}^{N} \left[\binom{N}{n} (1-P_{link})^n P_{link}{}^{N-n} \right]$$

Note that the probability of (3) ignores the burst packet effect [Wu et. al. 2003].

TFRC: TCP-FRIENDLY RATE CONTROL OVER HYBRID LINKS

TFRC [Handley et al., 2001] is a mechanism to have a non TCP connection behave similarly to, but more stable than a TCP connection which traverse the same path. For this purpose, a TFRC sender estimates the network conditions by exchanging control packets between the sender and receiver to collect the feedback information. In fact, the sender transmits one or more control packets in one RTT. On receiving the control packet the receiver returns feedback information required for calculating RTT and estimating the packet loss event rate. In this paper, we use a TCP-friendly protocol over a wired to wireless link in Fig. 4 for several reasonable advantages such as highly reliable transmission due to being a connection-oriented protocol and avoiding network congestion collapses. By adjusting the sending rate to the desirable rate determined by an underlying TCP-Friendly Rate Control (TFRC), one can achieve the required QoS of video applications over a wireless link [Chen et al., 2004]. Thus, we consider a TFRC scheme 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,

$$T_{TFRC} = \frac{S}{t_{RTT}\sqrt{\frac{2P_{link}}{3}} + t_{RTO}\sqrt{\frac{27p}{8}}\,P_{link}\left(1+32P_{link}^{2}\right)} \qquad (5)$$

where P_{link} stands for the overall 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 retransmission time out value [sec]. By regarding T_{TFRC} 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 link from those caused by buffer overflow of wired links. Therefore, in Section 3, we employ an MPEG packet loss model described in [Wu and Claypool, 2003], which considers a variable frame rate based on TFRC for single TFRC connection. This model estimates the number of playable frames at a receiver when a video stream is transmitted over a hybrid wired/wireless network.

PROPOSED APPROACH

WIRELESS CHANNEL MODEL

A typical model of video streaming over hybrid links can be considered as shown in Fig. 4. Video server s in a wired network sends a video stream to receiver r behind a wireless link. The wireless link is characterized by available constant bandwidth B_w and a random stationary packet loss rate p_w due to embedded additive noise. Then, a following brief scenario can be applied when there is no cross-traffic at either node 1 or node 2.

Proposed Scenario [Chen et. 2006]: (i) The wireless link is assumed to be bottleneck of the network by meaning no congestion at node 2 as shown in Fig. 4. (ii) Packet losses are assumed to occur at a wireless channel only by channel bit errors and the buffer at node2 does not overflow. Therefore, the packet loss probability at node 2, denoted as p_c, is assumed to be zero. (iii) $t_{RTT} = t_{RTT\min}$, i.e., the minimum RTT, if $T \le B_w$ and both B_w and p_w are constants, and (iv) The backward route from receiver r to server s is assumed to be congestion-free but not error- free due to bit errors.

Following the above scenario, 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 (5), the overall link packet loss rate P_{link} is now defined by two independent loss rates p_w and p_c as,

$$P_{link} = p_w + (1 - p_w)\,p_c \,, \qquad (6)$$

Since p_w gives the lower-bound for P_{link} for no-congestion (due to multiple TFRC connections at the same time at node 2), i.e. $p_c = 0$, then the upper-bound of the network throughput T_b becomes, $T_{TFRC} \le T_b$ at $t_{RTT\min}$ and P_{link} equals only p_w using (4). Hence, for an under-utilized channel, $T_b < B_w$ holds when only one TFRC connection exists. To achieve the full utilization of a wireless channel, an application opens a number of connections as far as the total throughput is less than $B_w(1-p_w)$. If the channel capacity B_w, the packet loss rate p_w, and packet size S are identical among connections, the optimal number of connections must satisfy $n_{opt} \equiv B_w / T_b$ [Chen et al. 2006].

BER PERFORMANCE

To simplify the analysis of wireless link characteristic for obtaining p_w, we have to consider random bit errors due to only additive White Gaussian noise (AWGN) ignoring fading effect. For robust wireless transmission a 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_b. Hence, a BER performance of BPSK scheme is given by [Goldsmith, 2005] as,

$$\mathrm{P_{link}} = \mathrm{p_b} = \mathrm{Q}\left(\sqrt{\gamma}\right) = \mathrm{Q}\left(\sqrt{\frac{2\mathrm{E_b}}{\mathrm{N_o}}}\right) \qquad (7)$$

E_b denotes 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 $Q(.)$.

PROPOSED PACKET-LOSS MODEL

This section considers the details of VFR-TCP model [Wu and Claypool, 2003] to estimate the number of playable frames at a receiver behind wired links and a wireless link, where random and stationary packet losses occur. We employ TFRC to control the sending rate over network in accordance with loss of packets caused by packet corruptions for bit errors over a wireless channel.

Also, we adopt the assumption of a frame-dropping mechanism to compensate the varying TCP-Friendly sending rate where frames are also dropped, or lost, by corruption of packets. If the quality of a frame (in terms of Peak SNR) falls below a pre-determined threshold $PSNR_{threshold}$, the frame is considered lost. The effective frame rate can be estimated as follows. When we consider the Bernoulli packet loss model, the observed frame rate F_{eff} can be expressed as,

$$F_{eff} = f_o(1-\phi_R),$$

(8)

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 [Feamster 2001]. If quality scaling is applied, we replace a constant f_o with a variable f_{link}. The frame rate f_{link} is further replaced by $G_{link}.S_{GOP}$, where G_{link} corresponds to the number of GOPs per second over hybrid link and S_{GOP} is the number of frames in each GOP pattern. Therefore,

$$F_{eff} = G_{link}.S_{GOP}(1-\phi_R) = \frac{S_{GOP}(T_{TFRC}/S)(1-\phi_R)}{(S_I + S_{IF}) + N_P(S_P + S_{PF}) + N_B(S_B + S_{BF})}$$

(9)

where $S_{GOP} = 1 + N_P + N_B$.

Hence, the frame drop rate ϕ_R can be formulated from (8) by these equations,

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

(10)

,

$$X_R = P_I\left[1 + \frac{P_p - P_p^{N_P+1}}{1-P_p} + N_{BP}.P_B\left(\frac{P_p - P_p^{N_P+1}}{1-P_P} + P_I P_P^{N_P}\right)\right]$$

(11)

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 probability of successful frame transmission $P(N, K, P_{link})$ for each i-frame type is defined as,

$$P_i = P(S_i + S_{iF}, S_i, P_{link}),$$

(12)

where P_i is defined as in (4) and consequently the term X_R of (11) can be rewritten in terms of $R_i[X_R]$ in order to obtain the total effective playable frame rate as in [Wu and Claypool, 2003],

$$F_{eff} = G_{link} .X_R = \sum_i R_i = R_I + R_B + R_P \qquad (13)$$

R_i stands the playable frame rate of i-frame type in GOP (N_P, N_{BP}). As a result, 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]

ILLUSTRATIVE RESULTS

METHODOLOGY

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 high bit errors) and the round-trip-time; while the MPEG application can provide details on the video characteristics at the server according to the network feedback report. The model can choose a suitable GOP pattern at server in order to obtain the reasonable expected playable frame rate at the receiver.

Based on these assumptions, we develop the following steps to find the optimal playable frame rate for QoS requirements using a proposed scenario in Section 3.1.

Obtain a channel SNR per bit $\gamma/2$ on wireless link.

Assess the bit error rate from the channel SNR by (7) using BPSK modulation scheme.

The effective link packet loss rate (P_{link}) due to only bit errors is defined in (6).

TFRC rate is evaluated by (5), which must satisfy the condition of $t_{RTT\,\min}$ without exceeding B_w.

Determine video quality in terms of the temporal scalability, i.e., frame dropping, to regulate the sending rate to the TFRC rate at the server.

For all possible GOP patterns, one with the maximum frame rate is chosen such as GOP(2,3).

After reading the network feedback report, if the bit errors are high enough, then a suitable FEC code must be added to the packets at video server to enhance the perceptual video quality at client.

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.

Furthermore, to achieve the optimal performance over wireless link a strategy of increasing the number of video connections can be applied until the total throughput reaches the hard limit of $B_w(1-P_w)$ where there is a saturation effect.

RESULTS ANALYSIS

In this section, simulation results have been obtained for a typical 1xRTT CDMA wireless network model used by [Chen 2006] to describe a wireless link. Moreover, the network and MPEG parameters used by MPEG packet loss Model [Wu 2003] are also considered as shown in Table 1. On some reasonable constraints, we use a 30 [fps] as a typical maximum frame rate allowed over Internet for full motion video and a recommended typical GOP is 12 frames, such as 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. All video quality performances assume only one video TFRC connection over hybrid network.

TABLE 1
PARAMETER **S**ETTING USED **I**N **S**IMULATION FOR WIRELESS LINK

network parameters	
) t_{RTT} RTT (	168 [ms]
) t_{RTO} retransmission timeout (	t_{RTT} 4
wireless channel parameters	
) B_w channel capacity (	1 [Mbps]
) $\frac{\gamma}{2}$ channel SNR per bit (	-6 ... 6 [dB]
) p_w BER and packet loss rate (	0.33 … 22 %
MPEG-4 parameters	
GOP(2,3) IBBBPBBBPBBB	12 frames/GOP
maximum peak rate	144 [kbps]
) $R_{\max}$ Spatial scalability ( ) Q Quantizer scalability ( ) F Temporal scalability ($\alpha_1, \alpha_2, \alpha_3$ Video encoder coefficients	640x480 [pixels] 10 30 [fps] 0.151, 9.707,4.314
) S packet size (	1 [Kbytes]
S_I size of an I-frame	25 [packets]
S_P size of a P-frame	8 [packets]
S_B size of a B-frame	3 [packets]
) S_{BF} , S_{PF} , S_{IF} FEC Code (	Small (1,1,0) Medium (4,2,0) Large (8,4,1)

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 6 (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 [Chen et al., 2006, Al-Suhail et al., 2006].

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 codes. Figure 67evaluates 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 play-out frame rate 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 frame rate as compared with chosen values of S_{IF} .

Moreover, a comparison with VFR-TFRC model over Internet [Wu and Claypool, 2003], depicts a more improvement in playable frame rate up to 30 [fps] for total packet loss rate $p \leq 2\%$ (See Fig. 8). This is the highest among all others frame rates, but the rate is not TCP-friendly over wireless Internet channel. Additionally, the frame rate increases over wireless link as far as a suitable FEC code is chosen well. Specifically, in Fig. 7 (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 in case of wireless channel [Al-Suhail, 2007].

Figure 9 depicts the video quality, in terms of 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 $BW(640 \times 480, Q, 30)$. 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 [Al-Suhail et. al. 2006]. 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. Hence, Table 2 and Fig. 10 illustrate an example of optimal video quality performance for GOP(2,3) over wireless link using the indirect dependence via the channel error and TFRC rate of Fig. 6 (a). Depending on preferences on the perceived video quality, one can choose the temporal scalability or the SNR scalability (Quantizer Scale) as quality scaling. As a result, when the temporal scalability is applied, video play-out becomes choppy, intermittent, or like a series of still images at high bit errors on wireless channel. On the other hand, the low SNR scalability results in coarse and mosaic appearances in the case of non well-chosen FEC or ignoring of FEC. Thus, within a suitable FEC code achieving low values of Q scale means certainly a good perceptual temporal video quality at the client end.

CONCLUSION

This paper has applied a robust packet loss model for MPEG-4 video streaming using FEC scheme at the packet-level of Group of Pictures, whereas a TCP-Friendly rate control (TFRC) runs over a combined wired-to-wireless network to regulate video traffic. Our proposed model has evaluated the video quality-of service (QoS) in terms of play-out frame rate as well as the Quantizer factor (Q) under various FEC code conditions at application layer. Illustrative results showed that the proposed model introduces a good and robust wireless video transmission in the case of only one TFRC connection using small FEC Reed Solomon code (Table 2). It is also found that the model 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 a number of TFRC connections for full-utilized bandwidth. Moreover, a FEC bit-level can also be applied at the physical layer over wireless link when multi-path fading channel is considered.

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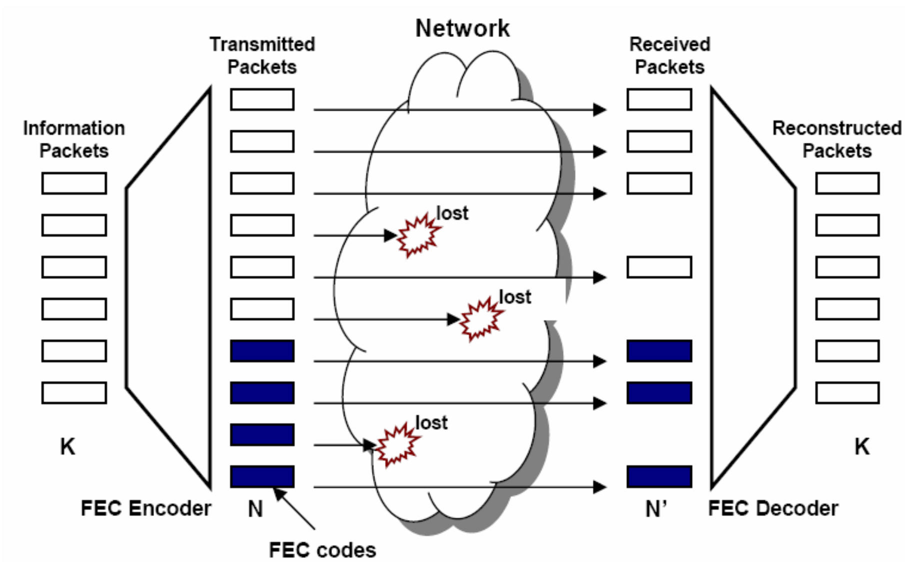


Fig. 1 A typical MPEG Group of Picture (GOP) and its inter-frame dependency relationship.

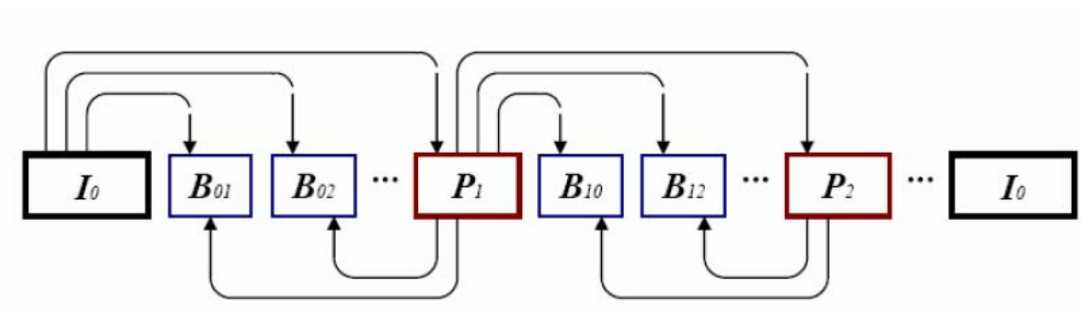


Fig. 2 An example of FEC technique over a lossy network

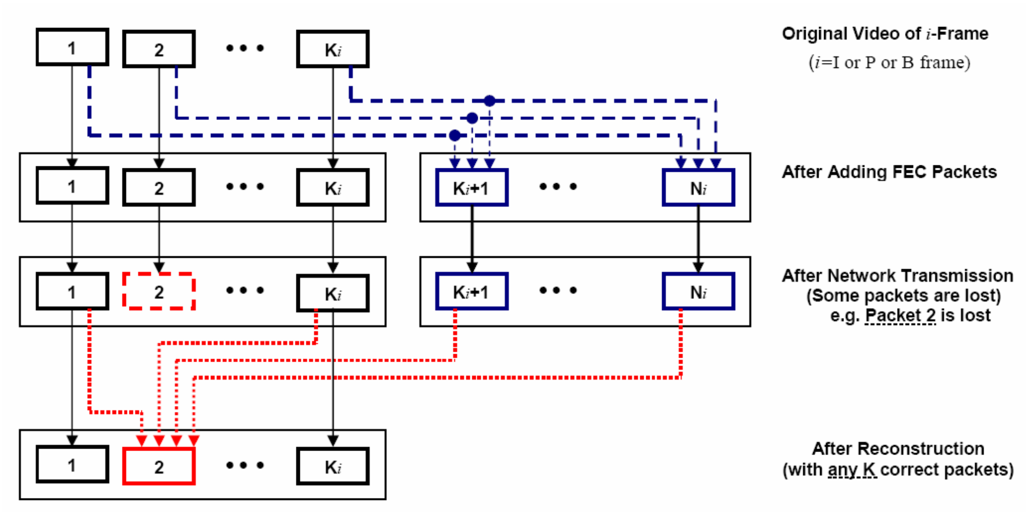


Fig. 3 A block diagram of Reed-Solomon code in each i-frame type of GOP of MPEG video.

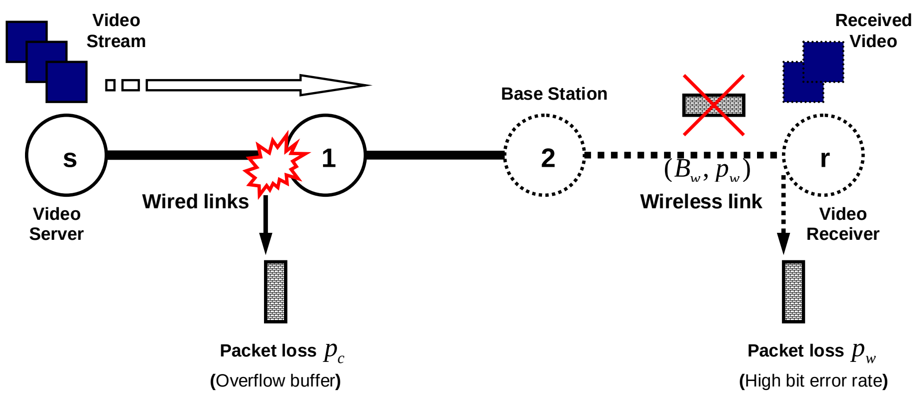


Fig. 4 A typical proposed wired-to-wireless video streaming model

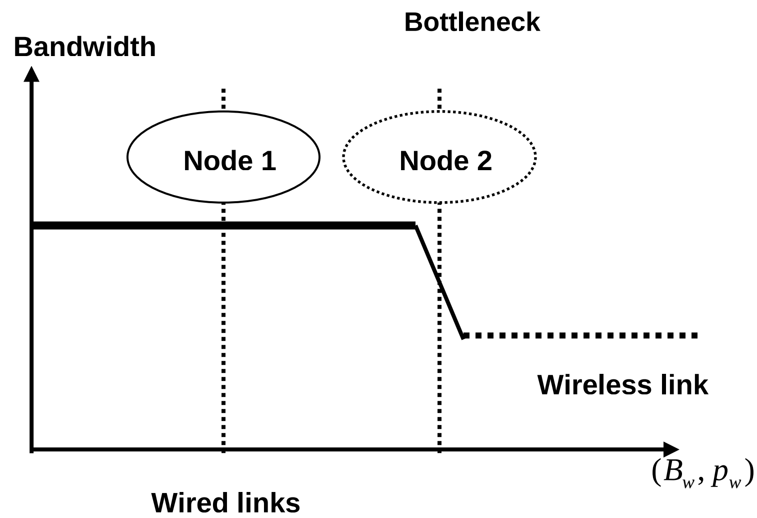
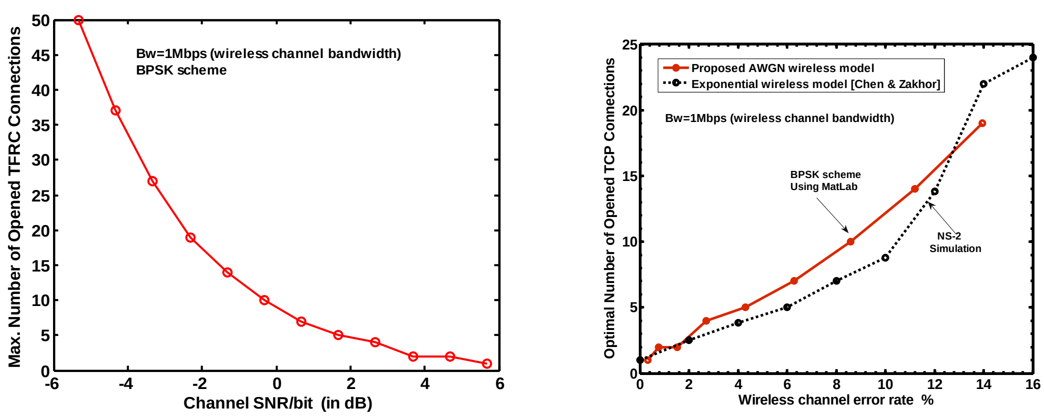
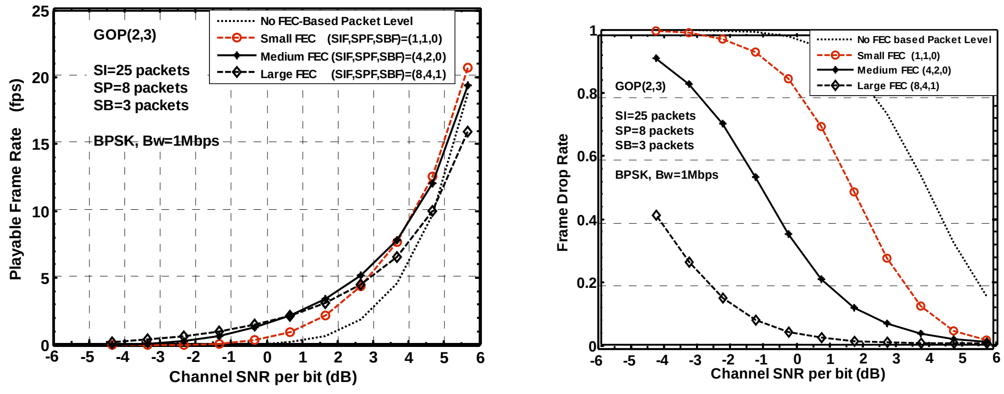


Fig. 5 Bandwidth condition for wired-to-wireless video streaming model



(a) (b)

Fig. 6 TFRC performance over wireless link. (a) Maximum number of TFRC connections and (b) Comparison between schemes [Chen *et. al.*, **2006**, Al-Suhail *et. al.*, **2006**].



(a) (b)

Fig. 7 Video quality performance of only one video connection (a) Playable frame rate and (b) Predicted frame drop rate

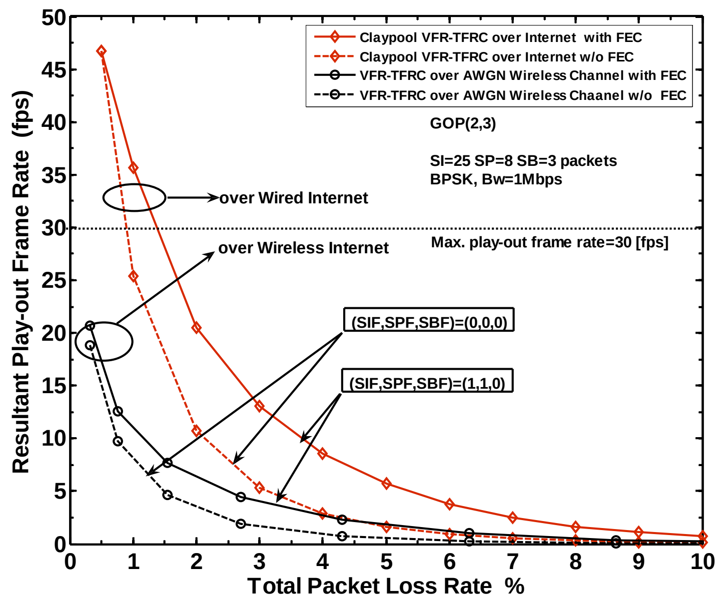


Fig. 8 Comparison of playable frame rate of only one video connection for wired and wireless Internet.

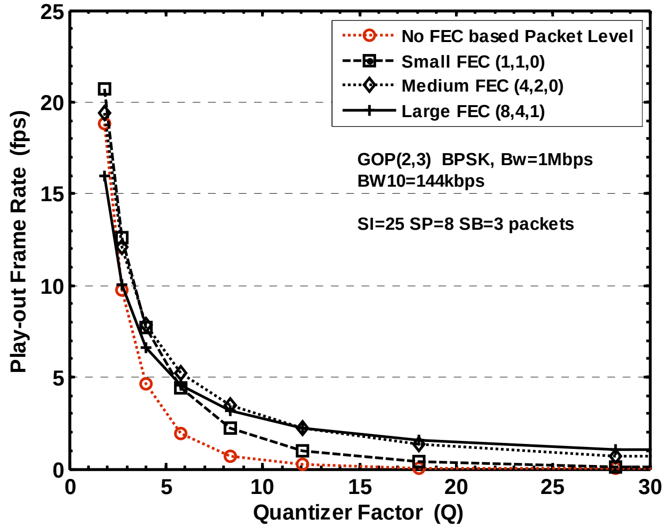


Fig. 9 Video quality scaling for only one video connection under various FEC code conditions.

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

$(640 \times 480{,}10{,}30)$ Original video rate 144 [kbps] with

		Play-out frame rate [fps]			
Quality factor (Q)	**Channel SNR (dB),** %) p_w (	**No FEC (0,0,0)**	**Small FEC (1,1,0)**	**Medium FEC (4,2,0)**	**Large FEC (8,4,1)**
2	**5.68,** (0.33 %)	18.8	20.68	19.4	15.98
4	**3.68,** (1.54 %)	4.6	7.7	7.86	6.65
6	**2.68,** (2.7 %)	1.92	4.42	5.24	4.55

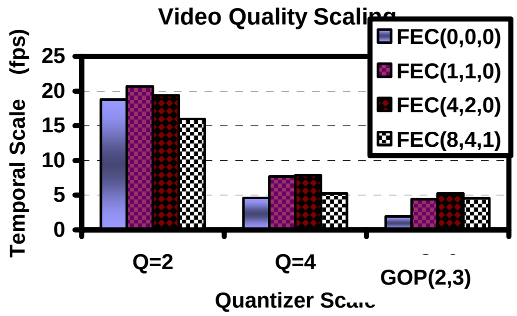


Fig. 10 Video quality scaling chart under various FEC code conditions