MPEG VIDEO TRANSMISSION OVER ERROR-RESILIENT AWGN WIRELESS CHANNEL

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Abstract -- The goal is how to estimate video quality of MPEG TCP-Friendly video streaming over robust wireless channel against frequent packet loss. In this paper, a Forward-Error-Correction (FEC) scheme is used as an intra-protection control over an Additive White Gaussian Noise (AWGN) wireless channel behind wired links. For this purpose, we propose Variable Frame Rate based on TCP-Friendly Rate Control (VFR-TCP) algorithm to evaluate the predicted frame rate of MPEG-4 video streaming. Quality of Service (QoS) is also evaluated by the predicted quantizer scale Q for the case that the network throughput is assumed to be equal to the required bandwidth. As a result, we obtained a good and reasonable perceived video quality over a noisy wireless channel, by varying the channel error rate or the channel SNR where AWGN and a coded BPSK scheme are dominated.

إرسال الإشارة المرئية عبر قناة لاسلكية مُضّادة لأخطاء الضوضاء البيضاء

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الخُـــلاصة: الهدف من هذا البحث هو تقييم جودة ارسال الاشارة المرئية (الفيديو) نوع MPEG المّعتمد برتوكول النقل TCP-Friendly عبر قناة لاسكلية ضد الفقد المتكررلرزمة الفيديو. تـّم استخدام تقنية تصحيح الخطأ الامامي(FEC) كطريقة سيطرة حماية-داخلية عبر قناة لاسلكية ذات ضوضاء مضافة من نوع كاوس (AWGN) واقعة في نهاية التوصيلات السلكية من الشبكة. ومن اجل ذلك، اقترحنا خوارزميـة جديـدة تدعى بـ(VFR-TCP) -اي خوارزمية معدل النقل للبروتـوكول TCP-Friendly المّعتمدة على معدل تغيـر الصـورة - لغرض تقييم معدل الصورة المتوقع لسريان الفيديو نوع MPEG-4.

تـّم ايضاً تقييم جودة الخدمة عن طريق معامل الجودة المّتوقع لحالة أفتراض عطاء الشبكة يكون مساوياً الى سعة النطاق المطلوبة. وبالنتيجة، تـم الحصول على نتائج معقولة لجودة الفيديو المُتوقع عبر قناة لاسكلية ذات ضوضاء، عندما يكون هناك تغير في معدل حدوث الخطأ في القناة او نسبة قوة الاشارة-الى-الضوضاء للقناة العاملة في حال استخدام كل من الضوضاء البيضاء ومعها تضمين مفتاح الازاحة ثنائي الطورBPSK عبر القناة.

Key-Words: - MPEG, TCP-Friendly, Video streaming, Wireless video, QoS, FEC.

1 Introduction

Practically, the major challenges of video transmission over wireless links are to deal with low bandwidth and high error rates due to 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 [1-4]. Several researches introduce different approaches such as adaptive modulation, error-resilience and adaptive rate control to reduce or mitigate the effects of wireless channel variations on the video quality at the receiver [2-5].

To provide a high quality of service (QoS) for video applications, i.e., high video play-out quality, at high loss rates of wireless links, it is important to use error-resilience [3-7]. 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. In this paper, we analyzed influences of a noisy wireless channel on the perceived video quality in terms of the frame rate and the quantizer scale, by varying the channel error rate or the channel SNR where AWGN and a coded modulation scheme are dominated.

The goal is how to estimate video quality of MPEG TCP-Friendly flow over wireless channel based FEC. A redundancy of Forward-Error Correction (FEC) depends on adding repair data along the original one such that packets can be repaired at the receiver without any additional transmission from the sender. FEC requires no feedback so it is efficient for random bit errors or burst errors of limited length over real-time multimedia traffic. For this purpose, we consider an encoded BPSK (Bi-Phase-Shift-Keying) scheme over AWGN wireless channel when the channel condition is in bad state. Hamming code is employed as a FEC in order to improve the effective range of channel SNR. In addition, we propose MPEG VFR-TCP model (Variable Frame Rate based on TCP-Friendly Rate Control) in [8] to estimate the predicted frame rate for MPEG-4 video streaming. Quality of service (QoS) is also evaluated by the predicted quantizer scale Q.

The rest of paper is organized as: Section 2 investigates the problem formulation. Simulation results are explained in Section 3. Finally, Section 4 concludes the paper.

2 Proposed Model Formulation

2.1 Video Quality

In MPEG coding, specific quantizer scale against each block of 16x16 pixels is performed. For large quantizer scale, the quality of decoded block becomes poor. It means this scale leads to degrade image SNR values [6]. 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] can be estimated in terms of spatial resolution (R [pixels]), PSNR resolution (Q) and the timely resolution (F [fps]) as

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

BW_{base} indicates the peak bit rate of the reference stream [7,8].

2.2 Wireless Channel Model

A typical model of video streaming over wired and wireless links can be considered as shown in Fig. 1. The wireless link is characterized by available bandwidth B_w and packet loss rate p_w. We consider using a TCP-Friendly Rate Control (TFRC) scheme [5,6] 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 [6],

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

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. In effect, a server cannot distinguish packet losses caused by bit errors on wireless links from those caused by buffer overflow.

A brief scenario can be applied when there is no cross-traffic at either node 1 or node 2. The wireless link is assumed to be bottleneck of the network by meaning no congestion at node 1. Packet loss is assumed only due to wireless channel bit errors and the buffer at node 2 does not overflow, as $p_c = 0$. In consequence, $t_{RTT} = t_{RTT\,min}$, i.e., the minimum RTT, if $T \le B_w$ [5,9]. Here, B_w is assumed limited constant bandwidth and p_w is to be random and stationary [5]. The backward route from receiver r to server s is assumed to be congestion-free but not error-free due to bit errors.

In this 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 Eq. (2), 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 if $p_c = 0$, the upper-bound of the network throughput becomes,

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

Hence, for an under-utilized channel, $T_b < B_w$ holds when only one TFRC connection exists. To obtain p_w, for robust transmission over wireless link when a bit corruption probability is high and without altering the sending rate it needs to repair the losses locally using FEC. A coded BPSK scheme based on Hamming code is examined to improve effective channel SNR per bit over wireless link. Therefore, $p_b(n,m)$ block error probability for coded BPSK is defined by [1] as,

$$p_b(n,m) \le (2^m - 1)\left(\sqrt{2d_{\min}\frac{R_c E_b}{N_o}}\right) \tag{4}$$

where $d_{\min} = n - m$ represents a minimum distance in Hamming code, m is being symbol length, for example $m = 4$ or 10 bits, and n is coded block. R_c is code rate. $\gamma = 2E_b/N_o$ represents the total channel SNR of a BPSK channel. The Gaussian cumulative distribution function is being $Q(.)$. 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 symbol error rate, $p_b(n,m)$, when channel bit error rate is high.

2.3 MPEG Packet-loss Model

We propose our MPEG VFR-TCP model in [8,9], as an algorithm to estimate the number of playable frames at a receiver. Random and stationary packet losses occur over robust wireless channel. We adopt a frame-dropping mechanism to compensate the varying TCP-Friendly sending rate. Frames are dropped, or lost, by corruption of packets. If the quality of a frame in terms of PSNR falls below a $PSNR_{threshold}$, 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,

$$F = G \cdot S_{GOPsize}(1 - \phi) \tag{5}$$

where ϕ is the "*frame drop rate*", G corresponds to the number of GoPs per second [6] and $S_{GOPSIZE}$ is the number of frames in a GoP (Group of Picture). The frame drop rate ϕ can be formulated as a sum of conditional probabilities as [8],

$$\phi = \sum_i P(f_i) \cdot P(\overline{F} \mid f_i) \qquad (6)$$

where i runs over the three frame types (I, P, and B), $\overline{F}$ represents the event that a frame is "useless" because the quality falls below quality threshold $PSNR_{threshold}$, and f_i is the event that the type of the frame is i. The *a priori* probability $P(f_i)$ can be determined directly from the structure of a stream [10].

3 Simulation Results

We obtained simulation results using a typical 1xRTT CDMA wireless network model summarized in Table 1 [5,6,9].

We have changed SNR of a wireless channel to evaluate the TCP-Friendly throughput of only one video connection. Figure 2 shows the code gain of Eq. (4) which improves the effective range of channel SNR (or channel error rate) as compared with that in [5,8]. As shown in Fig. 3 (a), the expected frame rate increases over operating channel SNR range up to 20 [fps] at 5.68 [dB] with $p_w = 0.33\%$. While Fig. 3 (b) illustrates predicted frame rate of 20 [fps] obtained over improved channel SNR range usingerror control scheme of Hamming code (7,4), by meaning at low range of channel SNR when the wireless channel state is poor. Moreover, Nicholas model, in [10], depicts more improvement in playable frame rate up to 30 [fps]. This is the highest among all, but the rate is not TCP-Friendly. But, also it is found that VFR-TCP model introduces a reasonable and good video quality in term of frame rates compared with the rate over Internet [6].

Figure 4 depicts the video quality, in term of Q, as a function of the channel SNR in [dB] 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 $BW(640 \times 480, Q, 30)$. As a result, it is noticed that the video quality Q is independent on the GoP pattern structure. Also, when error control is used to evaluate the corresponding improvement, it is evident that the quality scale decreases rapidly to be less than 10 on low SNR values of channel state. Therefore an improvement in the perceived user's video quality can be predicted at low channel SNR when FEC is employed.

4 Conclusion

In this paper, we presented an MPEG VFR-TCP model for under utilized bandwidth of wireless channel. The proposed work has estimated QoS for the video streaming in terms of frame rate and as well as the quality factor (Quantizer factor Q). Simulation results show that the proposed model introduces a good robust algorithm for only one TFRC connection over wireless link using FEC Hamming codes. It is also found that the VFR-TCP model increases tolerance to packet loss due to high bit errors and achieves a good quality compared with non-TFRC rate transmission in [10]. Further work can be proposed for multi-path fading channel as well as a number of TFRC connections opened during transmission.

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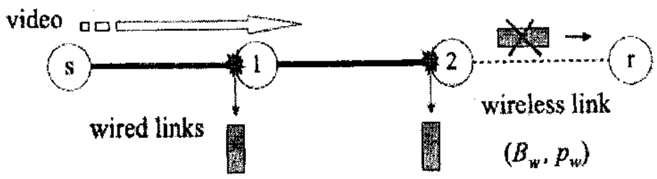


Figure 1: Typical wired/wireless video streaming model.

Table 1 Parameters setting in simulation

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

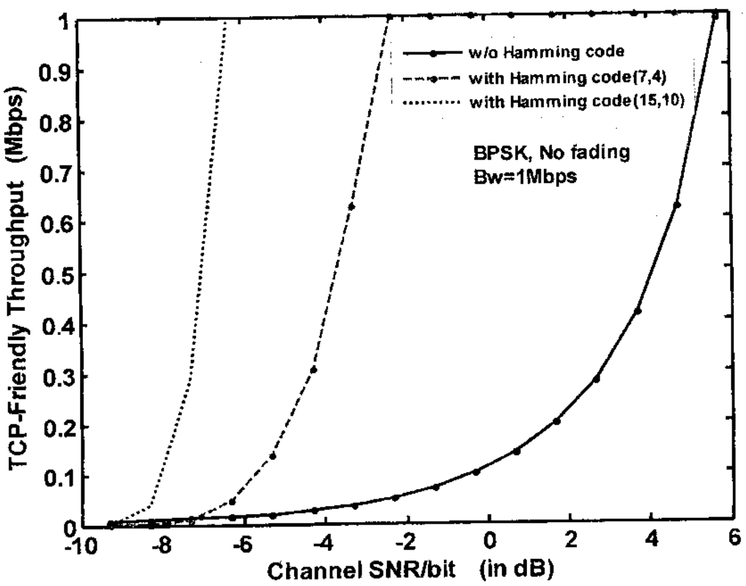
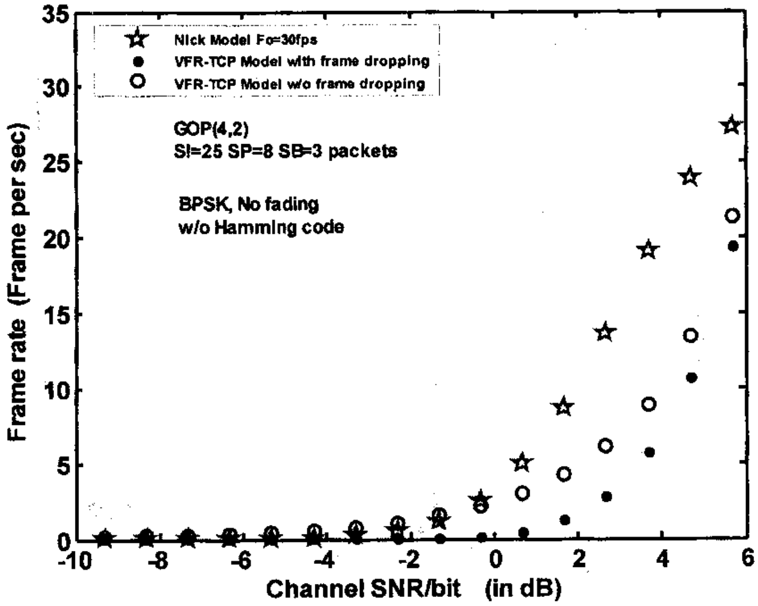
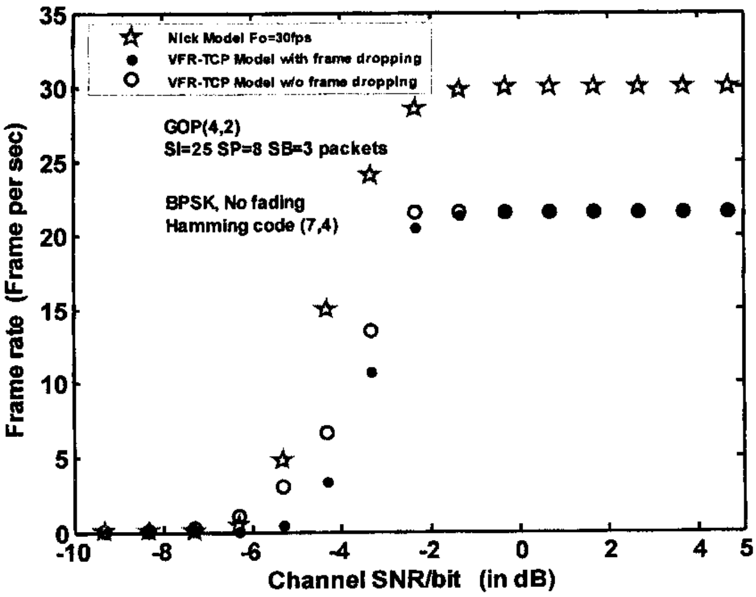


Figure 2: TCP-Friendly throughputs versus channel SNR.



(a)



(b)

Figure 3: Play-out frame rate of only one video connection as a function of channel SNR.

(a) w/o Hamming code (b) with Hamming code (7,4)

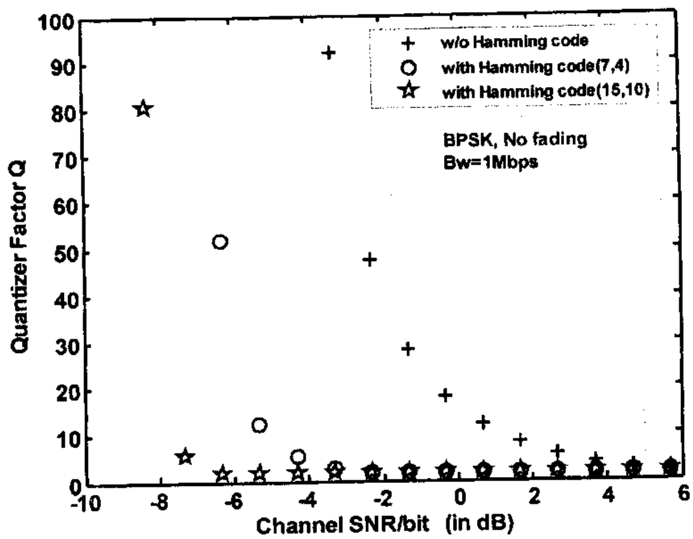


Figure 4: Video quality of one TFRC connection versus the channel SNR.