Error-Resilience of TCP-Friendly Video Transmission over Wireless Channel

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***Abstract*—The paper investigates TCP-Friendly video streaming over wireless channel using Forward-error-Correction (FEC). A FEC scheme is used as an intra-protection control based on Hamming code. Coded BPSK scheme is applied over AWGN wireless channel to be robust against frequent packet loss. For this purpose, we propose Variable Frame Rate based on TCP-Friendly Rate Control (VFR-TCP) model. The model estimates the predicted frame rate for MPEG video streaming. Quality of service (QoS) is also accounted for the predicted quantizer scale Q if the network throughput is assumed to be equal the available bandwidth. Simulation results show that the VFR-TCP model increases tolerance to packet loss due to channel bit errors and achieves a good quality.**

***Keywords*— Wireless video, TCP-Friendly, FEC, Error Control.**

I. INTRODUCTION

The continuing growth in wireless communications as well as the increasing demand for real-time multimedia applications such as video conference, video telephony, video-on-demand (VoD) are both attracted considerable issues to transmit video over wireless channels. 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 a 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 quality [1-7].

To provide a high quality of service (QoS) for video applications, by meaning high video play-out quality, at high loss rates over wireless links, it is important to use both error-resilience, i.e., media adaptation, and network-adaptation [2-9]. The error resilience is how to control the bit-rate of a video stream and adjust error control behaviours according to the varying wireless network conditions. It can be achieved by three error control techniques widely used in various settings to combat the channel errors: *Retransmission*, *Redundancy* and *Interleaving* [1-7]. The network-adaptation can be efficiently utilized by adapting the end-system to the changing network conditions. Adaptation represents the ability of network protocols and applications to observe and respond to the channel variations. Several studies [4-9] attempt to improve the video quality by employing adaptability.

In addition, the physical layer 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].

In this paper, the goal is how to estimate video quality of 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.

We estimate the predicted frame rate for MPEG video streaming by proposing VFR-TCP model based TCP-Friendly rate control. The wireless channel is assumed under utilized bandwidth and in a bad condition. Encoded Bi-Phase-Shift-Keying (BPSK) scheme is considered. Hamming code is employed as a FEC in order to improve the effective range of channel SNR. As a consequence, Quality of Service (QoS) in term of SNR scalability is accounted for the predicted quantizer scale (Q) if the network throughput is assumed to be equal the available bandwidth.

The rest of paper is organized as follows: Section 2 describes background on video quality, FEC and TCP-Friendly over wireless link. Section 3 investigates the proposed Variable-Frame-Rate TCP-Friendly (VFR-TCP) model over wireless channel to evaluate the predicted frame rate for video streaming in the presence of AWGN channel. Simulation results in Section 4 are for the predicted frame rate as well as the QoS in term of SNR scalability. Section 5 includes conclusion.

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II. BACKGROND

A. Video Quality

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 [7]. 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. Each video sequence consists of a cyclic sequence of GoPs, such as IBBPBB for GoP(1,2). Therefore, the frame rate can be reduced by dropping some frames and since the least influential frames are B-frames then B frames can be dropped first. The resulting GoP becomes I BP B or I P only. In consequence, the original frame rate is reduced by two or one thirds; i.e. from 30 [fps] to 20 [fps] or 10 [fps], respectively. By displaying frame repeatedly, the empty frame time should be filled and then the expected video quality is not degraded by dropping B frames.

There is a common tendency in the relationship between QoS parameters and the required bandwidth independently of the video content. 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

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

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

B. Forward Error Correction (FEC)

To improve the video quality under transmission errors, error-resilience 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.

The concept of FEC depends on adding redundancy bits to the block of video data for the purpose to detect and correct bit errors. To be able to correct a long burst of errors, significant amount of redundancy bits need to be added. It reduces the effective channel bandwidth for video data even when the channel is in good condition. In this paper, we present an encoded BSPK scheme based Hamming code over wireless channel. It is called intra-protection in physical layer. Section 3 examines the effective range of channel SNR when bit error rate is high.

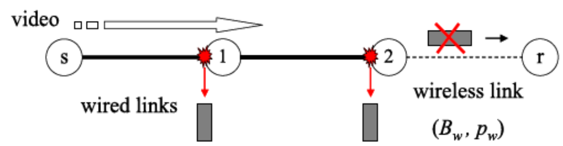

Fig. 1 Typical wired/wireless video streaming model.

C. TCP-Friendly over wireless links

TCP flows and TCP-friendly flows, in which the sending rate is controlled in accordance with network conditions as TCP does, are dominant in the wired and wireless Internet. In this paper, we use a TCP-friendly protocol over a wireless link 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 [4].

In this paper, we consider using a TCP-Friendly Rate Control (TFRC) scheme [10,11] 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,

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

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. However, a source node cannot distinguish packet losses caused by bit errors on wireless links from those caused by buffer overflow. Therefore, in Section 3, we propose an algorithm, called VFR-TCP (Variable Frame Rate based on TCP-Friendly Rate Control) to estimate the number of playable frames at a receiver when a video stream is transmitted over a network with wired and wireless links.

III. PROBLEM FORMULATION

A. Wireless Channel Model

A typical model of video streaming over wired and wireless links can be considered as shown in Fig. 1. 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 bandwidth B_w and packet loss rate p_w.

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Then, a following brief scenario can be applied when there is no cross-traffic at either node 1 or node 2.

1. The wireless link is assumed to be bottleneck of the network by meaning no congestion at node 1.
2. Packet losses are assumed to occur at a wireless channel only by channel bit errors and the buffer at node 2 does not overflow. Therefore, the packet loss probability at node 2, denoted as p_c , is assumed to be zero.
3. In consequence $t_{RTT} = t_{RTT\min}$, i.e., the minimum RTT, if $T \le B_w$.
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.

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 (2), packet loss rate p is now defined by two independent loss rates p_w and p_c as,

$$p = p_w + (1 - p_w) p_c \tag{3}$$

Since p_w gives the lower-bound for p for $p_c = 0$, the upper-bound of the network throughput becomes,

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

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 $n_{opt} \equiv B_w / T_b$.

B. Optimal Throughput over Wireless Channel

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 [6] as,

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

Here 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. channel. The Gaussian cumulative distribution function is being $Q(.)$.

In contrary, 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 Forward-Error-Correction (FEC) or Automatic Repeat-Request (ARQ). In this paper, wireless model considers protected code as an intra-packet FEC in link-layer. The protection deals with bits of the physical layer in forms of blocks. A coded BPSK scheme based on Hamming code is examined to improve effective channel SNR per bit over wireless link. This reflects in consequence on video streaming quality by regulating network TFRC throughout according to the effective channel SNR. The code gain of block error probability for BPSK scheme improves TFRC flow. Therefore, $p_b(n,m)$ block error probability for BPSK is defined by [1] as,

$$p_b(n,m) \le (2^m - 1) Q\left(\sqrt{2d_{\min}\frac{R_c E_b}{N_o}}\right) \tag{6}$$

$d_{\min} = n - m$ represents a minimum distance in Hamming code, where m being symbol length, for example $m = 4$ or 10 bits, and n is coded block. R_c is code rate. Since bit errors are randomly and independently occurred, then the symbol error probability can be written by [7] as,

$$p_s = 1 - \left(1 - p_e\right)^m \tag{7}$$

p_s denotes error probability of symbol length, m , before adding repair bits. Hence, $p_b(n,m)$ becomes the effective error probability for a Hamming code (n,m) . To achieve optimal TFRC throughput over wireless, a scenario of TFRC in Section 3.1 should be verified by setting packet loss $p_w = p_b(n,m)$ based on the same assumption for uncoded BPSK scheme ignoring the effect of packet size at high bit error rate values.

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C. Packet-loss Model

This section provides the details of our VFR-TCP [10], 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. TFRC is considered to control the sending rate in accordance with loss of packets caused by packet corruptions for bit errors over a wireless channel. We adopt 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,

$$F = f_o(1-\phi), \tag{8}$$

where ϕ is the "*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 $G.S_{GOPsize}$, where G corresponds to the number of GoPs per second and $S_{GOPsize}$ is the number of frames in a GoP. Therefore,

$$F = G.S_{GOPsize}(1-\phi) \tag{9}$$

The frame drop rate ϕ can be formulated as a sum of conditional probabilities as,

$$\phi = \sum_i P(f_i).P(\overline{F} \mid f_i) \tag{10}$$

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 [6,10]. The conditional probabilities for each frame type of size S_I, S_P and S_B can be derived under the assumption that if one or more packets within a frame are lost or one or more packets are lost in a reference frame, the frame is considered useless.

IV. Simulation Results

A. Methodology

Based on the above discussions, we develop the following steps to find the optimal playable frame rate for QoS requirements.

TABLE 1
PARAMETER SETTING IN SIMULATION

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

1. Obtain a channel SNR per bit $\gamma/2$.
2. Estimate the bit error rate (BER) p_e from the channel SNR by (5) for uncoded BPSK scheme. Then the packet loss rate over a wireless link is defined as $p_w = p_e$.
3. Using (6), we estimate the symbol, i.e. block, error rate $p_b(n,m)$ for Hamming code (n,m). The packet loss is defined as $p_w = p_b(n,m)$ for coded BPSK scheme ignoring effect of packet size S on packet corruption.
4. TFRC rate is determined by (1), which must satisfy the condition of (4) substituted the obtained p_w.
5. Consider quality scaling in terms of the temporal resolution, i.e., frame dropping, to regulate the sending rate to the TFRC rate.
6. For all possible GoP structures, one with the maximum frame rate is chosen.
7. The frame drop rate ϕ is estimated by using VFR-TCP.

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 [11].

A strategy to achieve the optimal performance is for an application 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.

B. Results Analysis

We obtained simulation results using a typical 1xRTT CDMA wireless network model summarized in Table 1 [4,11]. The results are based on some reasonable constraints. The maximum frame rate allowed is 30fps for typical full motion video and a recommended typical GOP is 15 frames, such as 'IBBPBBPBBPBBPBB' GOP(4,2), for optimal performance and the channel capacity B_w, which represents a maximum throughput for wireless link.

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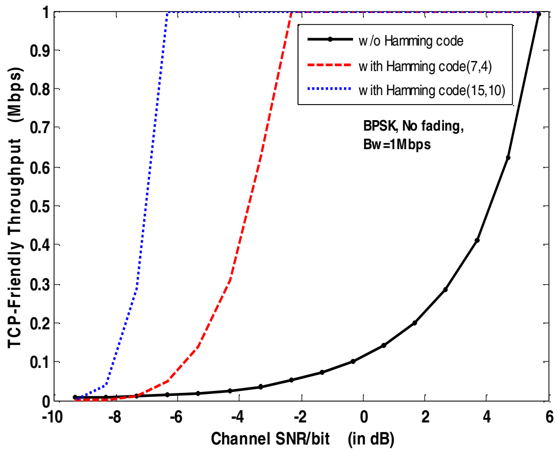

Fig. 2 TCP-Friendly throughput versus channel SNR.

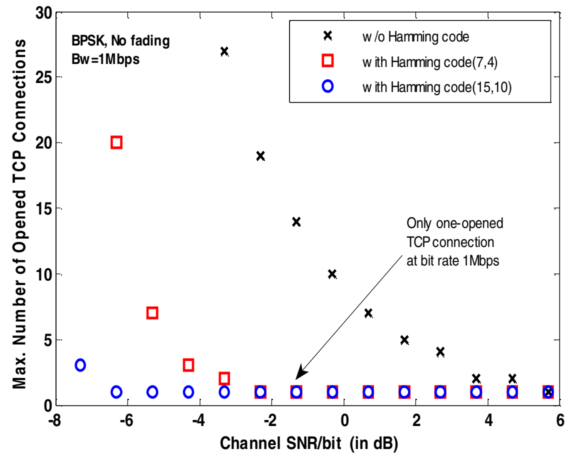

Fig. 3 The maximum opened TFRC connections versus channel SNR.

We changed SNR of a wireless channel to evaluate the TCP-Friendly throughput of only one video connection. We considered an error control Hamming code for encoded BPSK scheme ignoring fading effect on the channel. Fig. 2 shows the relationship between channel SNR and the network throughput. The code gain of (6) improves the effective range of channel SNR at a channel capacity, i.e.1Mbps, by approximately 9dB and 13dB for Hamming code (7,4) and (15,10) respectively.

In Fig. 3, we evaluate the maximum number of video connections n_{opt} over the effective channel SNR range. It should be noticed that with the packet loss rate $p_w = 0.0043$ 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 [6]. Whilst, the error control scheme introduces degradation in the number of opened video connections. The reason is due to FEC scheme which reduces the total effective channel bandwidth for video data connections even when the channel SNR becomes in a good condition. In other words, the improvement in channel SNR allows approximately one or few number of video connection(s) when error control scheme is employed. However, Fig. 4 shows results on the predicted number of playable frames and frame drop rate of only one video connection. As shown in Fig. 4(a), the playable frame rate consider uncoded BPSK scheme. The expected frame rate increases over the effective channel SNR range up to 20 [fps] at 5.68 [dB]. In Fig. 4(b) and (c), the predicted frame rate increases up to approximately 20 [fps] but over improved channel SNR range using error control scheme, by meaning 20 [fps] is achieved at low range of channel SNR when the wireless channel state is poor. Moreover, Nick model, in [8], depicts more improvement in playable frame rate up to 30 [fps]. This is the highest among all, but the rate is not TCP-friendly. In Fig. 4(d), the frame drop rate decreases as the wireless channel state improves using error control. This leads increasing in playable frame rate at the receiver and achieving a high video quality.

Fig. 5 depicts the video quality, in term Q, as a function of the wireless channel SNR 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 is used to evaluate the corresponding improvement, it is found that the quality scale decreases rapidly to be less than 10 on low SNR values of channel state. 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 [7]. On the other hand, the SNR scalability results in coarse and mosaic appearances.

V. CONCLUSION

In this paper, we present a variable frame rate model based TCP-Friendly rate control for under utilized bandwidth over wireless channel. The proposed work estimates 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. 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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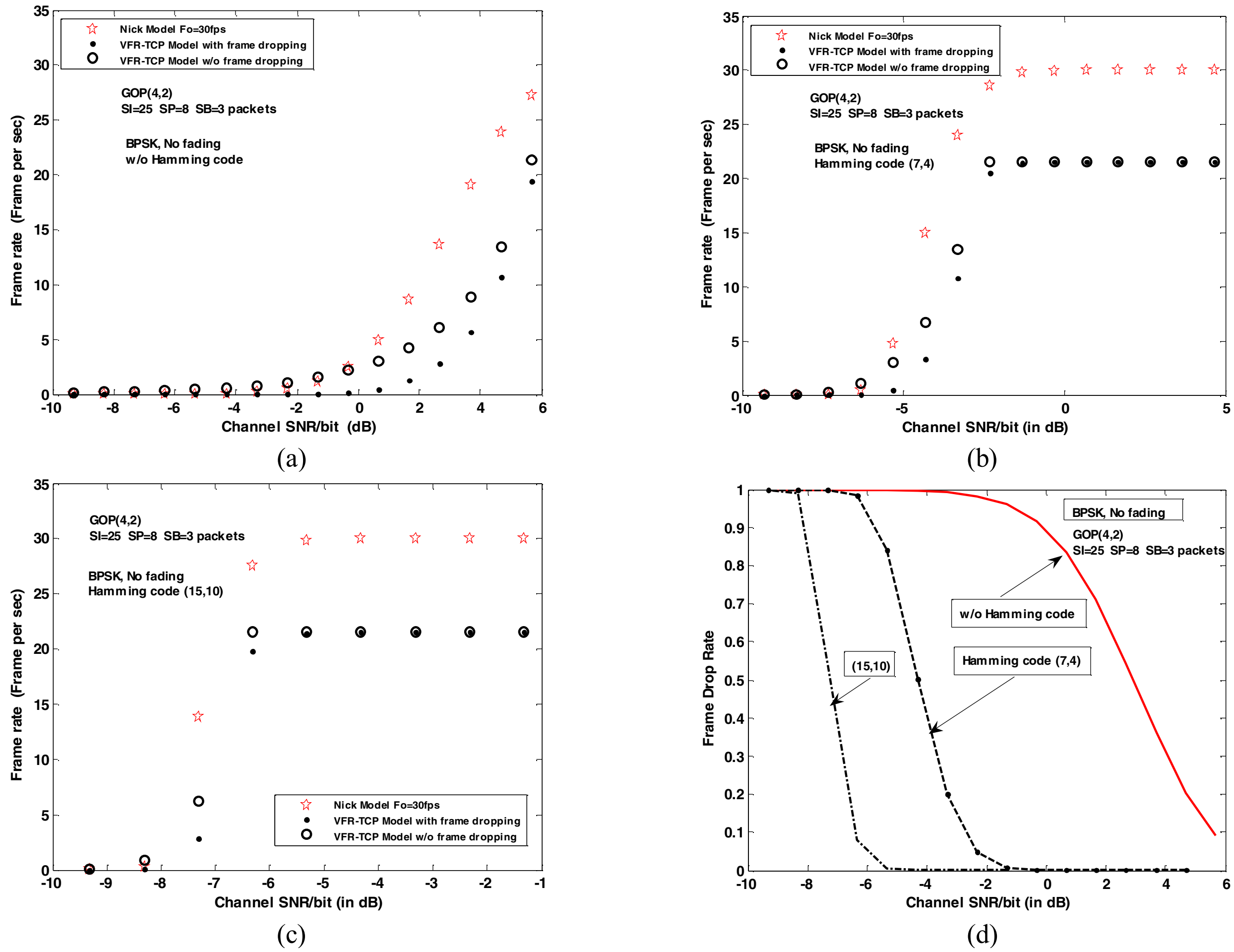

Fig. 4 Play-out frame rate and predicted frame drop rate of only one video connection as a function of channel SNR.

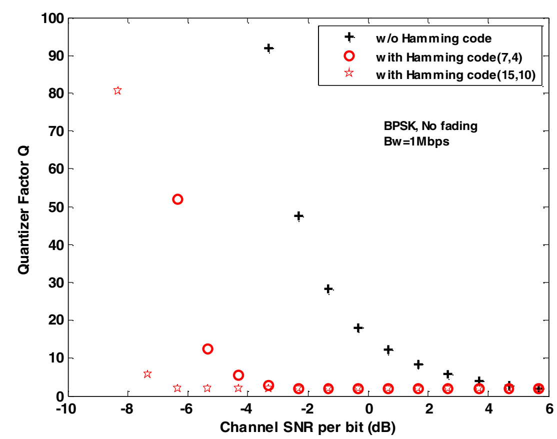

Fig. 5 Video quality of one TFRC connection versus the channel SNR.

REFERENCES

[1] A. Goldsmith, *Wireless Communications*, New York, NY: Cambridge University, 2005.

[2] R. Tripathi, " Pre-interleaving Error-Correction Coding for Reliable Transmission over Fading Channels", Dept. of Elec. and computer Engg, University of Texas, pp 1-11, 3 May 2002.

[3] S. Aramvith, I. Pao and M. Sun, "A Rate-Control scheme for Video Transport over wireless Channels," IEEE Trans. on Circuit and System for Video Tech. , vol. 11, no. 5, pp. 569-580, May 2001.

[4] M. Chen and A. Zakhor, "Rate Control for streaming Video over Wireless", Proceed. of INFOCOM, Hong Kong, chain, March 2004.

[5] N.C. Ericsson, "Adaptive Modulation and Scheduling For Fading Channels", Uppsala University, Uppsala, Sweden, 1999.

[6] N. Feamster, "*Adaptive Delivery of Real-Time Streaming Video,*" Master thesis, Dept. of Elec. Eng. and Comp. Sc., Massachusetts Institute of Technology, May 2001.

[7] H. Wu, M. Claypool, and R. Kinicki, "Adjusting Forward Error Correction with Quality Scaling for Streaming MPEG," *NOSSDAV'05*, pp. 111-116, June 2005.

[8] K. Fukuda, N. Wakamiya, M. Murata and H. Miyahara, " QoS Mapping between User's Preference and Bandwidth control for video transport", IFIP, 1997.

[9] F. Yang, Q. Zhang, W. Zhu, and Y.Q. Zhang, "End-to-End TCP-Friendly Streaming Protocol and Bit Allocation for Scalable Video over Wireless Internet", IEEE Journal on Selec. Area In Comm., vol. 22, no.4, May 2004.

[10] G. A. AL-Suhail and R.S. Fyath, "Analytical Model for Packet Loss Recovery with TCP-Friendly Bandwidth Based MPEG," Proc. of 6th JIEEEC, March 14-16, 2006.

[11] G. A. AL-Suhail and N. Wakamiya, "Effective TCP-Friendly Transmission for Video Streaming over AWGN Wireless Channel," Accepted in 6th WSEAS MUSP'06, China, April 16-18, 2006.

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