Impact of Packet Size on the Temporal Quality of Video Transmission over Wired-to-Wireless Network

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ABSTRACT
In this paper, an analytical packet loss model is applied for a Group of Pictures (GoPs) pattern of MPEG-4 video stream over wired to wireless Internet. The model assumes Additive White Gaussian Noise (AWGN) wireless channel causing frequent bit errors associated with packets. To achieve high efficient video transmission over network, the effect of packet size is first investigated on the video temporal quality in case of TCP-Friendly Rate Control (TFRC) transmission, and then the video stream quality is compared in case of non-TFRC transmission. The new simulation results show that a good predicted video quality performance (temporal scalability) can be evaluated in terms of a number of successful playable frames per second. The findings reveal that under different levels of channel Signal-to-Noise Ratios (in dBs), an appropriate packet size must be well-chosen before transmission commences. Further, the wireless video provides a reasonable visual perception compared to that one of wired-video Internet.

General Terms
Performance, Model, Analysis.

Keywords
MPEG, TCP-Friendly, Video streaming, Wireless video, Video quality.

1. INTRODUCTION
Recently, the demand on multimedia applications is widely increased over wired and/or wireless Internet, including such as real-time video streaming, video conference, and video on demand. However, wired/wireless Internet does not provide the necessary Quality of Service (QoS) guarantees that are needed to support high-quality video transmission. Many major challenges of video traffic on the wired and wireless Internet links [1-2] are encountered. Some of these challenges deal with high packet loss rate due to the congestion of buffer overflow over wired networks; and others are faced by the characteristic of wireless links, which are mostly suffering from 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 a significant degradation in the quality of reconstructed video sequence. Thus robust transmission of real-time video over wireless channels is becoming critical problem to achieve good perceptual quality at the client terminal end.

Unlike typical Internet traffic, streaming video is sensitive to delay and jitter, but can tolerate some data loss. In fact, video transmission can yield better video play-out when the underlying protocol provides smooth data rate than a bursty data rate. For this purpose, video streaming applications often use UDP (non-TFRC) or TCP-Friendly Rate Control (TFRC) as a transport protocol rather than TCP. Unfortunately, UDP does not reduce its data rate when an Internet router drops packets to indicate congestion. It means that there is no congestion control within UDP and no response to relieve the saturation of a bottleneck due to congestion. It means that there is no congestion control within UDP and no response to relieve the saturation of a bottleneck due to congestion. Thus recent researches have proposed rate-based TCP-Friendly protocols for steaming media [3-5] as alternatives to UDP over wired/wireless networks. There are three advantages to rate control using TFRC: first, avoid congestion collapse (providing network stability); second, it is fair to TCP flows and third, TFRC's rate fluctuation is lower than TCP, making it more appropriate for streaming applications. However, the *key* assumption behind TCP and TFRC is that the packet loss is a sign of congestion, whereby neither TFRC nor TCP can distinguish between packet loss due to buffer overflow and that due to physical channel errors, resulting in underutilization of wireless bandwidth. Hence video streaming rate control and congestion control over wireless are still open issues.

In traditional communications systems, channel variations are dealt with in a worst-case manner. For wireless systems this implies the use of a simple modulation scheme, and a complex error-correcting code. When the coding fails to compensate for temporary bad conditions, higher layers in the protocol will ensure that the information is correctly and completely transmitted, by requiring a retransmission of the erroneous data.

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Therefore, to avoid this problem adaptive modulation schemes are used to adapt the demands on the channel as it varies. By changing the modulation format as the channel SNR (or SINR) varies, this will accomplish less retransmission.

Therefore, to provide an acceptable video quality; i.e. high-Playable Frame Rate (PFR), at high loss rates of wireless links, several approaches have been either separately or jointly pursued. They include adaptive rate control, passive error recovery (re-transmission), Forward Error Correction (FEC), and adaptive modulation [6-13]. For example, Wu *et al.* [7] extended the packet loss model for MPEG-2 video streaming and derived analytically the playable frame rate (PFR) for a given packet loss probability over wired Internet. They employed VQM method to regulate the level of quantization via the estimation of the amount of distortion. Yaun *et al.* [8] proposed a FEC scheme in the GoP-level with cost of high complexity of the packet generation. Furthermore, Feamster *et al.* [6] proposed error-model for MPEG-4 based on constant frame rate using UDP and I-frame re-transmission. Finally, AL-Suhail *et al.* [9] have applied a Wu's model [7] over wired/wireless channel and improved the QoS using FEC in the packet-level. Additionally, other studies [1,12,13] combined adaptive modulation and joint source channel coding over fading wireless channels and verified significant performance advantages in the worst channel conditions. Zahi [1] proposed optimal cross-layer resource allocation over wired/wireless link for real-time video transmission without taking into account TFRC protocol. In contrary, Khan *et al.* [12] proposed CLD for wireless video streaming based on Peak Signal-to-Noise ratio (PSNR) versus the communication cost associated with the transmission of rate-distortion profile from the video server. Meanwhile, Tong *et. al.* [13] presented TFRC-LERD scheme using loss event rate discounting to improve TFRC over wireless networks. Recently, Pack *et. al.*[14] have studied analytically TFRC performance in mobile hotspots, and they specifically have developed the throughput model in steady state that is necessary to support multimedia application with QoS guarantee in these mobile hotspots.

In this paper, we assume that the physical layer of wireless channel can estimate the performance of symbol or bit error rate (BER) versus Signal-to-Noise Ratio (SNR) when the channel is corrupted by an Additive White Gaussian Noise (AWGN). Thereby a Binary-Phase-Shift-Keying (BPSK) modulation is chosen to provide a robust transmission against high bit errors. By using packet-loss model, a video playable frame rate can eventually be predicted in the application layer when an appropriate small packet size is chosen well for TFRC transmission mode as well as in case of non-TFRC.

The rest of the paper is organized as follows. Section 2 describes a brief background. Section 3 presents the proposed approach, and followed by Section 4 for performance evaluation. Simulation results are explained in Section 5, and finally conclusions are summarized in Section 6.

2. SYSTEM DESCRIPTION

2.1 MPEG Video

Video Streaming is promising multimedia application which is recently gaining popularity as a key factor of the success of 3G systems. Using mobile devices, users can access online video clips such as news, sports, etc. by clicking on a hyperlink using web browser. With streaming, the video content need not be completed downloaded, but the client can begin playback the video few seconds after its receiving part of content from the streaming server. However, a raw video stream requires a high bit rate for transmission so video compression is usually employed to carry out transmission efficiency. MPEG-4 is therefore a video compression standard adopted by most mobile networks (for example UMTS, WLANs, etc). In practice, MPEG-4 technology can produce good video quality at bit rates suitable for mobile and wireless transmission.

In this paper, we evaluate the video quality of MPEG-4 stream over wired-to-wireless network when a last hop is mobile device as shown in **Figure 1**. A typical Group of Pictures (GoPs) structure of an MPEG stream is considered. 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. In fact, GoP pattern can be identified for MPEG-4 in similar manner in MPEG-2, for simplicity, as $G(N_P, N_{BP})$ and $N_B = (1+N_P)\times N_{BP}$, where N_B corresponds to the number of B-frames, N_P corresponds to a number of P-frames in a GoP, and N_{BP} corresponds to the number of B-frames between I and P frames (for example, GoP(2,2) "IBBPBBPBB" where N_P =2 and N_{BP} =2. However, another new error-resilient GoP pattern can be found in [15].

Furthermore, there are three user's preferences related to video QoS parameters in terms of spatial scalability, peak SNR scalability, and timely scalability (frames per second) [16]. In this paper, the QoS of MPEG-4 video is defined only in terms of successful play-out frame rate at client end.

2.2 TFRC Equation Model

In the original TFRC [**5**], the sender uses TCP throughput model to estimate the current available network bandwidth, i.e.,

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

where T is the smooth target sending rate of a TFRC session, t_{RTT} is the round-trip time [sec], t_{RTO} is the TCP retransmit time out value [sec], S is the packet size [byte], and p is the loss event rate reported by the receiver. Meantime, t_{RTO} is estimated and calculated by TFRC sender. Initially, TFRC sender sets its sending rate to one packet per second and doubles the rate every RTT until packet loss occurs. Thereafter, the sending rate is determined by (1) via estimating p , t_{RTT} , and t_{RTO} to qualify the throughput of the TFRC flow. By regarding T as the available bandwidth for video streaming and adjusting the video traffic, the high-quality video play-out at a receiver can be expected.

3. PROPOSED APPROACH

In this paper, two packet loss models for MPEG-4 video streaming over wireless Internet are investigated on AWGN wireless channel; one model is under TFRC transmission mode (i.e. variable frame rate depends on TFRC throughput over network), and the other model is assuming a fixed or constant playable frame rate during non-TFRC transmission mode, i.e. using UDP transport protocol.

3.1 Wireless Channel Model

A realistic wireless video transmission system can be represented by the model shown in **Figure 1 (a)**. The source coder provides compression (usually lossy) of the video while the channel coder introduces redundancy in order to combat error caused by a noisy channel. The concealment stage is a post-processing stage (usually found only in lossy compression systems such as video) which is useful for reducing the effects of residual channel errors. In this stage, operations such as spatial or temporal filtering are carried out to improve the quality of corrupted video [1].

In this paper, the concealment stage and the channel coding are *not* considered in our proposed approach. Thus we assume **Figure 1 (a)** is a part of a typical model of video streaming over wired and wireless links in Figure 1 (b); whereby a video server s in a wired network sends a video stream to a receiver r behind a wireless link. The wireless link is characterized by available bandwidth B_w and packet loss rate p_w due to bit errors. Meanwhile, p_c at node1 and/or node 2 denotes the packet loss rate due to the congestion (buffer overflow). In this model, a source node cannot distinguish packet losses caused by bit errors on wireless links from those caused by buffer overflow [1,5]. Thus we consider only the bit error rate over wireless link is the considerable reason to generate this packet loss.

3.2 BER Performance

At physical layer, a frequent bit error of a wireless channel with AWGN is considered and BPSK scheme is therefore applied when the fading effect is ignored. Within proposed model, we will refer to the term "*mod m*" to indicate to a specific choice of a modulation and coding scheme. The probability $P_{e,m}(l)$ of error in a packet of length l bytes (*also referred* to as the physical layer packet loss rate, PLR), for a given *mod m*, as a function of the bit error probability p_b can be expressed by [11],

$$P_{e,m}(l) \leq 1-(1-p_{b,m})^{8l} \tag{2}$$

where $S \equiv 8l$ denotes a packet size (in bits), and the inequality in (2) represents the fact that one can recover from bit errors in a packet, due to the coding scheme used at the packet level (intra-protection). Also, the packet error probability in (2) can be denoted as packet loss rate without any error-correction procedure when the inequality is replaced by equality [2].

In this paper, we simulate the results ignoring the procedure of error-correction code in order to specify only the effective operating ranges of channel SNR ratios at various packet lengths.

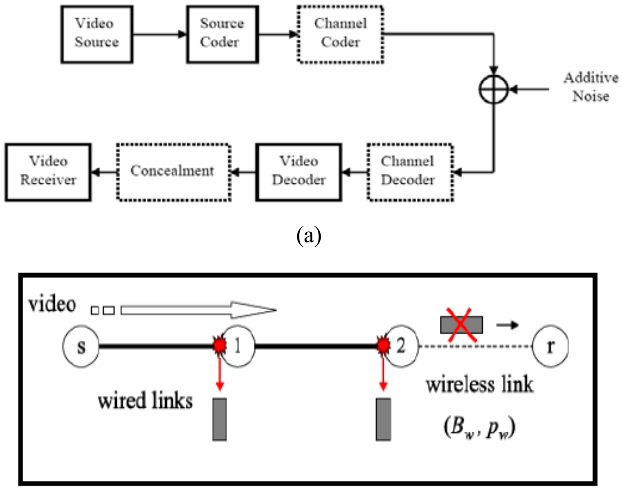


Figure 1. Video transmission system (a) A wireless channel model (b) A typical wired/wireless video streaming model

Therefore, here the BER performance of *non-coded* BPSK scheme over AWGN channel ignores the channel coding, and the probability of bit error is given by [1],

$$p_b = Q\left(\sqrt{\gamma}\right) = Q\left(\sqrt{\frac{2E_b}{N_o}}\right), \tag{3}$$

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 *non-coded* BPSK scheme assuming there is no channel encoding for this modulation scheme. The Gaussian cumulative distribution function is being $Q(.)$ and can be approximated in a close formula for a limited range of γ.

3.3 Packet loss Model Based-TFRC

Because TFRC was initially designed for wired networks, when applied to wireless environment, the original TFRC receiver cannot distinguish the congestion packet loss from the wireless packet loss. Thus the loss event rate p calculated by the receiver consists of wireless bit errors part and congestion part. However, TFRC sender actually only needs the congestive loss event rate, so it may result in bandwidth underestimation if the original loss event rate is directly used. Hence, our solution is to follow this scenario.

When there is no cross-traffic at either node 1 or node 2, this scenario can be applied as follows. The wireless link is assumed to be bottleneck of the network by meaning no congestion at node 1 as shown in **Figure 2**. Packet loss is assumed only due to wireless channel bit errors and the buffer at node 2 does not overflow, as $p_c = 0$. In consequence $t_{RTT} = t_{RTT\,\min}$, i.e., the minimum RTT, if $T \leq B_w$. Here, B_w is assumed limited constant bandwidth and p_w is to be random and stationary packet loss due to bit errors [5,9]. 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 (i.e. B_w) 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 (1), 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,

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

Hence, for an under-utilized channel, $T_b < B_w$ holds when only one TFRC connection exists. Here, the reported p_w at the receiver depends on the source packet length (size) before TFRC transmission starts.

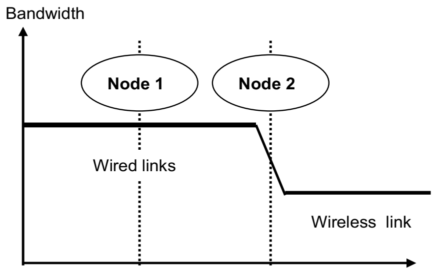


Figure 2. Bandwidth condition for wired-to-wireless video streaming model

To estimate the number of playable frames at a receiver, random and stationary packet losses are considered over wireless network. Thus the analytical model designed over wired Internet for MPEG-2 video stream, which is proposed by Wu *et al.* [**7**], is applied in this paper for GoP pattern of MPEG-4 video stream over wireless link. This model basically employs TFRC protocol to control the sending rate in accordance with loss of packets caused by packet corruptions for bit errors over a wireless channel. Subsequently, a GoP rate can be analytically expressed using TFRC protocol and the frame dependency relationship of I, P, and B frames. Hence, the resultant playable frame rate R can be computed as follows.

Under the TCP-Friendly constraint of (1), the GoP rate G (in GoP per second) is computed as,

$$G = \frac{T_b / S}{S_I + + N_P S_P + N_B S_B}, \tag{5}$$

where G corresponds to the number of GoPs per second. S_I, S_P, and S_B are the frames' sizes of the I, P, and B frames in GoP pattern (in packets). Then the total playable frame rate can be evaluated as [7],

$$R_{TFRC} = \sum_i R_i = R_I + R_P + R_B \tag{6}$$

3.4 Packet-Loss Model Based non-TFRC

When UDP protocol is considered to stream MPEG-4 video over wireless channel, an extra control algorithm is required to control the congestion over wired-to-wireless links. In the model of **Figure 1 (b)**, we assume the following scenario. The packet size starts with 1Kbytes at the video source, and this packet size should be changed at the node 2 to avoid the congestion (by meaning $p_c = 0$) at the base station (BS) of the wireless network. Furthermore, a proper packet size will introduce a significant effect on the network performance. Hence, the effective physical layer throughput can be expressed as [11],

$$T_{Phy,m} = A_m\left(1 - P_{e,m}(l)\right), \tag{7}$$

The factor A_m represents the maximum achievable data rate in (Kbps) for *mode m*. The probability of packet error $P_{e,m}(l)$ can be defined as in (2). Thus for *non-coded* BPSK mode, the coding rate is not taken into account, and the factor A_m should be known in terms of channel SNR in order to evaluate the effective network throughput. On the other hand, Equation (7) can be reformatted in terms of frame-dropping mechanism in order to explain the variability in the original video sending rate due to packet loss over wireless link ignoring the effect of any congestion at node 2. Hence, the frames are assumed to be dropped, or lost, by corruption of packets due to bit errors on wireless link. If the frame quality in terms of Peak Signal-to-Noise Ratio (PSNR) falls below a pre-determined threshold $PSNR_{threshold}$, then the frame is considered lost at the receiver (i.e., mobile side).

$$R_{non-TFRC} = F_o\left(1 - \phi_R\right), \tag{8}$$

where ϕ_R stands as 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. If quality scaling is applied, a constant F_o can be employed to support non-TFRC transmission mode. At node 2, the frame drop rate can be predicted and formulated as a sum of conditional probabilities as [6],

$$\phi_R = \sum_i P(f_i) \cdot P(\overline{F} \mid f_i), \tag{9}$$

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**]. 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, i.e. dropped.

The strategy in this model is to assume that the network will depend on the pre-defined probabilities at video source of node 2 in Figure 1 (b). In consequence, the model can choose a GoP pattern before UDP transmission starts in order to obtain the reasonable expected playable frame rate that are compatible with the full video motion at the client terminal end.

4. PERFORMANCE EVALUATION

To evaluate the received window size at the client in (Kbytes), a TFRC bandwidth-delay product can be considered. Here, a propagation delay is based on only fixed round-trip-time. As this product increases due to the network throughput then the expected video quality may eventually enhance at the client in terms of playable frame rate. In TFRC mode, this product is affected by the packet length over wireless channel.

Furthermore, one metric used to evaluate the video quality performance is the successful playable frame rate in (6), which can be also expressed as a ratio (in %) as follows.

$$PFR \text{ (in \%)} = \frac{R}{F_o} \times 100, \qquad (10)$$

Where F_o is the source frame rate in frames/sec. Equation (10) provides the percentage of the frame rate in a GoP that can be perceived correctly at the receiver [8]. In practice, the source frame rate typically varies between 15 and 30 [fps] depending on the applications. In the simulation results in Section 5, we assumed 25 [fps] is as a reference frame rate (**Table 1**).

5. SIMULATION RESULTS

5.1 Methodology

To find the resultant TFRC-playable frame rate (PFR):

1. The video source initially starts with the packet size S at 1 Kbytes in wired links.
2. As soon as the video flow has arrived node 2, a bottleneck bandwidth problem will allow the video system to reduce its packet size before the wireless transmission starts.
3. At node 2, an appropriate low value of S should be chosen (such as 512 bits, 640, 960, etc.) depending on the wireless channel state in order to ensure a desired quality of video streaming at client.
4. Thus the video system obtains a channel state in terms of SNR per bit ($\gamma/2$) and assesses the bit-error rate p_e by (4) on the wireless link for *non-coded* BPSK modulation scheme.
5. Then, it estimates the corresponding packet loss rate (PLR) using (2). Hence, the packet loss rate over a wireless link can be defined as $p_w = P_e$.
6. Estimate TFRC rate for MPEG video, which must satisfy the condition of the obtained p_w equals P_e, by (4). Then, we can determine the video quality in terms of the temporal scalability, i.e., playable frame rate (R) by (6).

Non-TFRC transmission mode over wireless link can also be evaluated using (9) if the prior probabilities of I-, P-, and B-frames are per-defined at node 2 before UDP transmission starts. Hence, the predicted frame rate over network is estimated by using (10) if the original video frame rate is 25 [fps].

5.2 Results Analysis

Simulation results have been conducted for a typical wireless network model. **Table 1** introduces a typical parameters setting for wireless network and GoP pattern parameters [**5,7**]. A channel capacity is assumed at the limited bandwidth (B_w), which represents a maximum throughput for wireless CDMA link.

Table 1. Parameters setting used in simulation at node 2

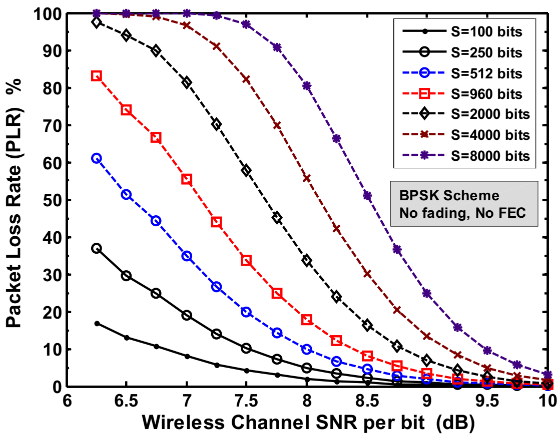
Wireless network parameters	
t_{RTT} =168 [ms] $t_{RTO} = 4\,t_{RTT}$ S = 64, 120, 250, 500, 1000 bytes	B_w =1 Mbps (1x RTT CDMA) BPSK (uplink/downlink) Data Speed 144 Kbps
Channel SNR / bit γ 6 ... 11 [dB]	
Source video F_o = 25 [fps]	
GoP Pattern parameters	
I-frame S_I = 25 packets	
P-frame S_P = 8 packets	
B-frame S_B = 3 packets	

In **Figure 3**, the performance of the wireless channel in terms of packet loss rate (PLR) is clearly evaluated through the change in the channel SNR per bit and its corresponding bit error rate for various values of packet size. It is noticed that as packet size increases, then the overall resultant PLR highly raises. For example, $p_e = 1 \times 10^{-4}$ introduces PLR of 5% for 512 bits packet size as the results in [**13**]. Therefore, **Figure 4** draws the effect of packet size on TFRC performance in terms of TCP-Friendly throughput and its related window size (in Kbytes), which is received at the client terminal. The window size is defined by the network throughput-Delay product; and the t_{RTT} can be considered as the minimum effective delay on network. Hence, it is found that although the packet size S becomes large, the expected PLR increases due to the bit errors occurrence and the expected window size will remain also large. As a result, such large values of S cause a poor perception for video quality at the receiver in case of low values of channel SNR ratios.

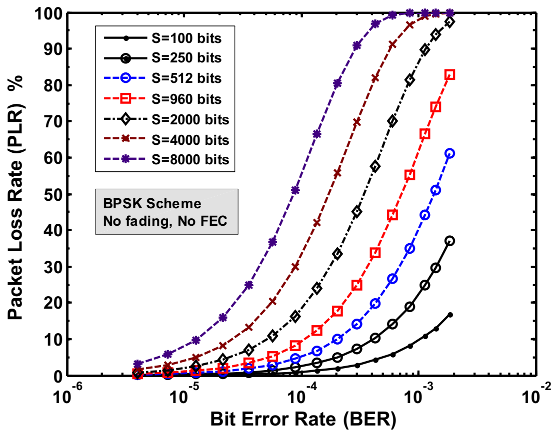
Figure 5 (a) shows a significant improvement in the resultant PFR when S is chosen within small values (i.e., when S is being 512 bits) for TFRC mode. It is found that small packet size allows low resultant PLR and it also provides an opportunity for wireless channel to operate at a lower range of channel SNR as compared to the SNR range of large values of S.

Specifically, at node 2 when packet size S changes from large values (e.g. 8000 bits) to a small value (e.g. 512 bits) in a typical chosen GoP(2,3) pattern, this will provide a rapid full motion (high video quality) for video at client. In effect, to guarantee the video quality (5 to 25 fps), the corresponding channel SNR can be between 9.2 to 10 dB in case of 512 bits, and between 10.3 to 11 dB in case of 8000 bits.

Figure 5 (b) displays also a resultant PFR which is based on non-TFRC transmission mode. The PFR can introduce a good tolerance for packet loss rate in terms of channel SNR. When packet size is chosen to be 100 bits, then the corresponding channel SNR can be between 7.25 dB to 10 dB for 5 to 25 fps, respectively. However, this range can be greater than 10 dB in case of packet size of 8000 bits.

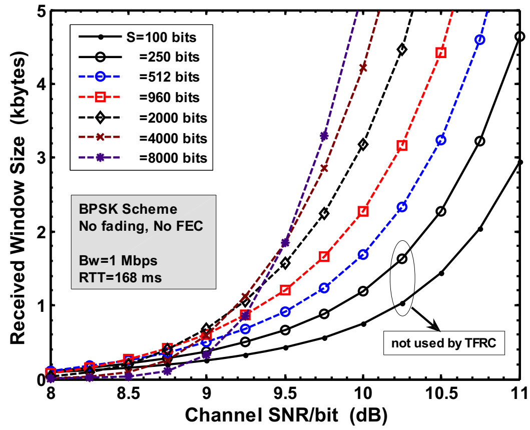


(a)

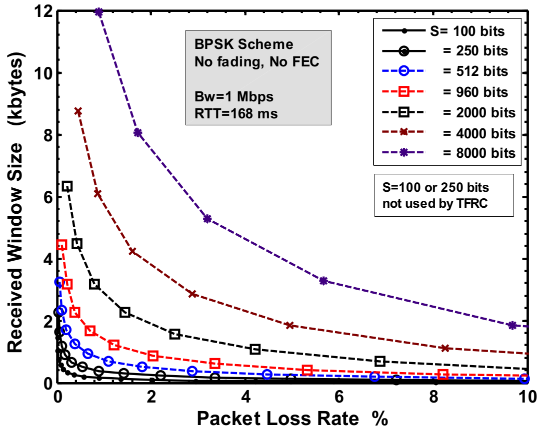


(b)

Figure 3. The performance of wireless channel versus packet loss rate for various values of packet size



(a)



(b)

Figure 4. TFRC performance of only single video stream at the client for various values of packet size

Furthermore, **Figure 6** explains the required window size at client as a function of PFR% in case of TFRC-mode for various values of packet size. It is noticed that a small packet size, for instance 512 bits; will need small window size, however, the temporal scalability (in PFR %) can rapidly achieve high values of full motion when the channel SNR becomes around 10 dB. In contrary, as S increases highly to be 8000 bits then there is a significant degradation in PFR % ratio although the received window size is being high at client.

As a result, in both modes, the study introduced a new guideline to investigate the effect of packet size of MPEG-4 video on temporal scaling (video quality) over wireless channel as compared with other recent studies for wired Internet [7,8] (See **Table 2**). The findings can be summarized as follows:

(1) A proper choice for packet size (i.e., by setting small values at node 2 in radio link transport layer) will provide a lowest or a minimum packet drop rate during transmission.

Consequently, the play-out frame rate can achieve high values quickly when channel SNR becomes high enough using BPSK scheme over wireless link.

(2) Since TFRC mode is originally UDP controlled by TCP equation model at the sender, hence, packet size can be typically chosen not less than 512 bits (i.e. 64 bytes); to provide the *fairness* with TCP flows [5] over channel (by meaning the header of TCP packet is at least 20 bytes).

(3) Since the header of UDP packet is typically 8 bytes in case of non-TFRC, then small values of S can be less than 512 bits. For example, setting S to be 100 bits will provide a highest video quality (full motion) at about 11 dB as shown in **Figure 5 (b)**.

(4) A full motion can be achieved at 0.2% packet loss rate (channel SNR 10 dB) when S is chosen to be 512 bits in case of TFRC-mode; and roughly at 11 dB in case of non-TFRC (See **Figures 4 & 5 & Table 2**).

(5) TFRC-mode takes into account the network constraints such as network throughput; then the maximum channel SNR required for full motion will not be less than 10 dB and 11 dB when packet size changes from 512 bits to 8000 bits, respectively.

(6) The non-TFRC is based on the prior probabilities of frame dropping in terms of PSNR of perceived video stream before video transmission starts. Hence, the maximum channel SNR required for full motion will roughly be much greater than 10 dB when S changes to large values.

(7) Although the large values of S trends to increase the received window size at client, the successful PFR % remains lower than that of small values of S in case of TFRC-mode (See **Figure 6**).

6. CONCLUSIONS

This paper has dealt with the effects of packet size on the video quality in terms of temporal scalability (frame per sec) over wireless Internet. We have applied a packet loss model based TCP-Friendly Rate Control (TFRC) for low bit rate MPEG-4 video streaming over an AWGN wireless channel. The model has estimated the effective playable frame rate (PFR) and its corresponding PFR% ratio. As a result, the findings highlight that the proposed model based-TFRC introduces a good tolerance in video quality over wireless link as compared to the non-TFRC mode. Furthermore, the simulation results provide a direct guideline to track the effects of small packet sizes over radio link layer based TFRC. However, future work can involve channel coding and adaptive modulation in order to improve the low value range of SNR over the wireless channel; and eventually to provide more perceptive video quality at client in case of large values of packet size when a bad channel state is considered.

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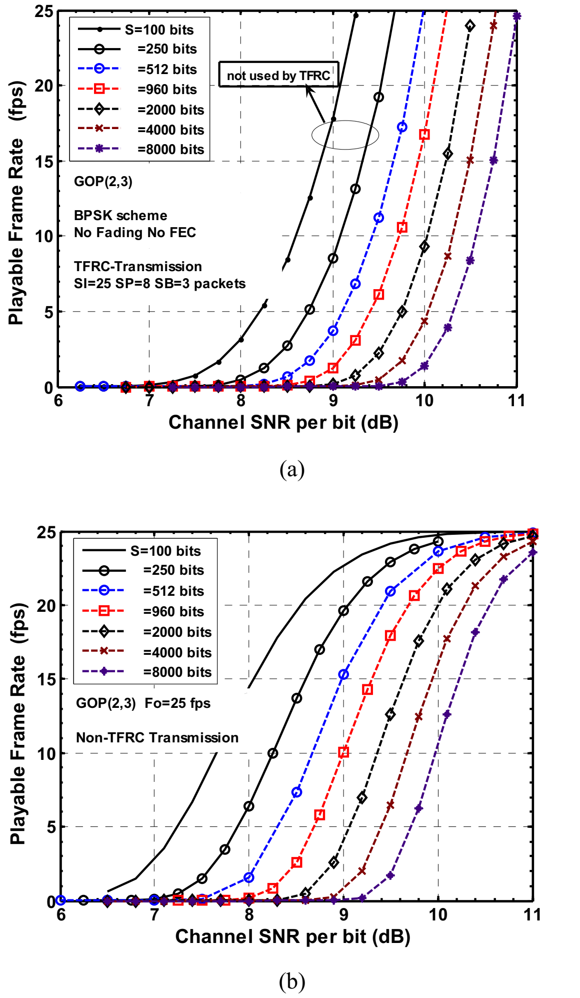

Figure 5. Temporal video quality versus channel SNR/bit (a) based-TFRC Mode (b) based-Non TFRC Mode

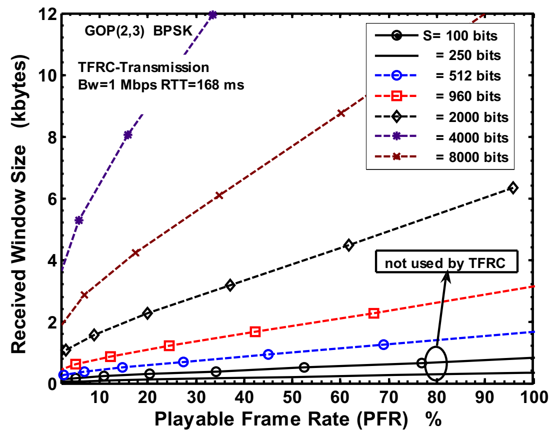

Figure 6. Temporal video quality in terms of PFR ratio for TFRC transmission Mode

Table 2. Performance comparison amongst packet loss models for MPEG-4 video stream with GoP(2,3) and original frame rate of 25 [fps] under various values of packet size

Transmission-Mode	Packet –Loss Model	S (bits) Packet Size	PLR %	PFR (fps), PFR%
Non-TFRC Mode Wireless UDP	**Frame-Based** w/o FEC	**8000 bits**	5% (due to bit errors)	6.35 [fps], **25.3%** Channel SNR (9.8 dB)
		512 bits	5% (due to bit errors)	6.32 [fps], **25.2%** Channel SNR (8.43 dB)
TFRC-Mode Wired Internet [7]	**Frame-Based** w/o FEC	**8000 bits** (or 1 Kbytes)	5% (due to congestion, RTT=50 ms)	1.58 [fps], **6.32%**
TFRC-Mode Wired Internet [8]	**GoP-Based** w/o FEC	**8000 bits** (or 1 Kbytes)	5% (due to congestion, RTT=50 ms)	1.25 [fps], **5%**
Proposed TFRC-Mode Wireless Internet	**Frame-Based** w/o FEC	**512 bits** (or 64 bytes)	**0.2%** (due to bit errors, RTT=168 ms)	Full-motion **100%** Channel SNR (10 dB)
		960 bits (or 120 bytes)	**3.3%** (due to bit errors, RTT=168 ms)	1.26 [fps], **5%** Channel SNR (9.0 dB)
		8000 bits (or 1 Kbytes)	**3%** (due to bit errors, RTT=168 ms)	1.4 [fps], **5.6%** Channel SNR (10 dB)

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