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Effective TCP-Friendly Transmission for Video Streaming over AWGN Wireless Channel

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Abstract: - In this paper, video streaming over AWGN wireless channel combined with wired links is investigated based on VFR-TCP model (Variable Frame Rate based on TCP-Friendly Rate Control). The model is proposed to evaluate the predicted frame rate for MPEG-4 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 required bandwidth. Simulation results show that the VFR-TCP model increases tolerance to packet loss due to network congestion as well as channel bit errors and achieves a reasonable quality.

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

1 Introduction

Multimedia communications over wireless networks have grown over the last decade involving real-time video applications, such as video conferencing, video phone, and on-demand video streaming. For example in [1], transmission of a video stream over a wireless channel is considered. In practice, wireless video communications face several challenges such as high bit error rates, bandwidth variations, limitations on the power for multimedia services, and processing capability constraints on handhold devices. Among these, the most influential is variations in the quality of wireless channels caused by AWGN, time-varying fading and shadowing, and interfering conditions, which lead to corruption of packets.

To provide an acceptable end-to-end QoS for video applications, i.e., high-quality video play-out, at high loss rates of wireless links, several approaches have been pursued. They include adaptive rate control, passive error recovery, Forward Error Correction (FEC), and adaptive modulation [1-6]. Studies [1,2] combined adaptive modulation and joint source channel coding over fading wireless channels and verified significant performance advantages in the worst channel conditions.

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 [5].

In this paper, we focus on influences of bit errors caused by the channel SNR of an AWGN wireless link on the video quality, whose sending rate is controlled by TFRC. A wireless channel is assumed in a bad condition where the BPSK (Bi-Phase Shift Keying) scheme is applied.

The rest of the paper is organized as follows. First Section 2 states the problem formulation of video streaming over a typical wireless channel model and then VFR-TCP (Variable Frame Rate based on TCP-Friendly Rate Control) algorithm is proposed to predict the effective playable frame rate. Section 3 describes the problem solution for the optimal frame rate and the strategy to reach the desired throughput. Simulation results are presented in Section 4 for multiple video sessions on a fully-utilized wireless channel. Also, QoS in term of the SNR scalability, i.e., the quantizer scale, is evaluated when the network capacity is assumed to be equal to the required bandwidth. Finally, Section 5 concludes the paper.

2 Problem Formulation

Transmission of a video stream over wireless networks is challenging because of the limited bandwidth and channel errors. To achieve a good video quality at a receiver, a robust transmission scheme over a wireless link is required. An ARQ error control scheme is helpful for wireless channels, but the effective channel bandwidth will become variable due to retransmissions in a poor channel condition.

In this paper, we consider using a TCP-Friendly Rate Control (TFRC) scheme [1,5,6] 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,

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

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 this section, 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.

A reasonable video quality can be achieved by allocating enough bandwidth to VBR traffic of an MPEG coded video stream. The allocated bandwidth can be equal to the actual peak rate of a video stream. However, it apparently deteriorates the efficiency of the bandwidth usage. For example, as the temporal resolution is degraded by dropping one or more frames from a GoP and the traffic smoothing is applied to the rest of frames, the required bandwidth decreases in inversely proportional to the number of dropped frames. Figure 2 illustrates an example of such reduction of the bandwidth by discarding B frames from GoP (1,2)=IBBPBB. The frame rate of video play-out, in consequence, is reduced by one thirds; i.e., from 30fps to 10fps. By displaying the preceding frame repeatedly, the empty time slots can be filled.

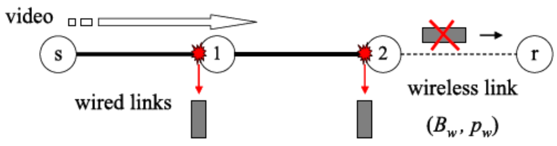


Fig. 1 Typical wired/wireless video streaming model.

There is a common tendency in the relationship between scaling parameters and the required bandwidth independently of the video content. The required bandwidth $BW(R,Q,F)$ [bps] of an MPEG video stream with the spatial resolution (R [pixels]), the temporal resolution (F [fps]), and the SNR resolution (Q) can be estimated as,

$$BW(R,Q,F) \cong (\frac{1}{3.1})^{\log_4(\frac{R}{640\times480})}(0.151 + \frac{9.707}{Q} - \frac{4.314}{Q^2})\frac{F}{30}BW_{Base}, \quad (2)$$

where BW_{base} indicates the peak bit rate of the reference stream [4].

2.1 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.

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 [5].
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 Eq. (1), packet loss rate p is now defined by two independent loss rates p_w and p_c as,

$$p = p_w + (1 - p_w) p_c \ , \qquad (3)$$

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

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

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 $n_{opt} \equiv B_w / T_b$ and thus,

$$n_{opt} = \frac{B_w}{S\left[t_{RTT\min}\sqrt{\frac{2p_w}{3}} + t_{RTO}(3\sqrt{\frac{3p_w}{8}})p_w(1+32p_w{}^2)\right]^{-1}} \qquad (5)$$

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 [3] as,

$$p_e = Q\left(\sqrt{\gamma}\right) = Q\left(\sqrt{\frac{2E_b}{N_o}}\right) \qquad (6)$$

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 $Q(\cdot)$.

2.2 Analytical Packet-Loss Model

This section provides the details of our VFR-TCP [7], 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 corrupsions 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,

$$F = f_o(1 - \phi), \qquad (8)$$

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 [8]. 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 $G \cdot S_{GOPsize}$, where G corresponds to the number of GoPs per second and $S_{GOPsize}$ is the number of frames in a GoP. Therefore,

$$F = G \cdot S_{GOPsize}(1 - \phi) \qquad (9)$$

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

$$\phi = \sum_i P(f_i) \cdot P(\overline{F} \mid f_i) \ , \qquad (10)$$

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 [7-8]. 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.

3 Problem Solution

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

1. Obtain a channel SNR per bit $\gamma/2$.
2. Determine the bit error rate (BER) p_e from the channel SNR by Eq. (6). Then the packet loss rate over a wireless link is defined as $p_w = p_e$.
3. TFRC rate is determined by Eq. (1), which must satisfy the condition of Eq. (4) substituted the obtained p_w.
4. Consider quality scaling in terms of the temporal resolution, i.e., frame dropping, to regulate the sending rate to the TFRC rate.
5. For all possible GoP structures, one with the maximum frame rate is chosen.
6. The 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 Eq. (2). During a video streaming session, a server regulates R, F, and Q to adjust the sending rate to the TCP-friendly rate [4,7].

A strategy to achieve the optimal performance is for an application to increase the number of connections n_c 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 upto a certain point, after which there is a saturation effect. When any extra connections or too many connections are opened, the total throughput goes beyond the capacity of a wireless channel, i.e., $n_cT_b > B_w$. As a result, the packet loss rate p_c increases and the round trip time t_{RTT} becomes higher than $t_{RTT\min}$. In that case, the round trip time is expressed as,

$$t_{RTT} = \frac{S/T_b}{\sqrt{\frac{2p}{3}} + 4\left(3\sqrt{\frac{3p}{8}}\right)p\left(1+32p^2\right)}, \qquad (11)$$

where p is derived by Eq. (3).

Table 1: Parameter setting in simulation

$t_{RTT} = 168ms$ $t_{RTO} \cong 4t_{RTT}$ $B_w = 1Mbps$	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 -6dB $0.33\% to 22\%$

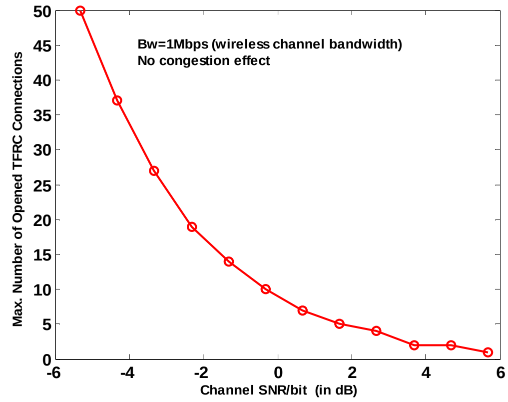
4 Simulation Results

We conducted simulation experiments using a typical 1xRTT CDMA wireless network model summarized in Table 1 [5].

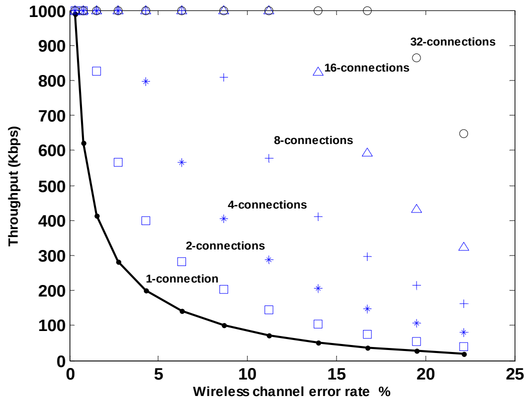
We changed SNR of a wireless channel to evaluate the maximum number of video connections n_{opt}. We also evaluate the total throughput n_cT_b with different sets of the number of TFRC connections and BER p_e, thus, packet loss rate p_w. The expected round trip time obtained by Eq. (11) is evaluated with different sets of the number of extra TFRC connections and BER p_e. Results are shown in Fig. 2. It should be noticed that with the packet loss rate p_w=0.0043, which implies the channel SNR is 1.68 [dB], the optimal number of connections is around 4 or 5 as shown in [5].

As shown in Fig. 2(b), the total throughput increases as the number of TFRC connections increases up to the channel capacity. In the case of one TFRC connection, the throughput does not exceed the limitation B_w =1 [Mbps]. Therefore, the expected RTT does not change and stays at 168 [ms] for small BER as shown in Fig. 2(c). However, as the number of extra TFRC connections n_{extra}, increases, the total sending rate exceeds the limitation and, as a result, t_{RTT} becomes higher. Too many connections also lead to the increase in the end-to-end packet loss rate p with frequent packet losses due to congestions as shown in Fig. 2(d). It is found that the packet loss rate due to congestion decreases as wireless channel error rate increases. It is because that the throughput per connection decreases for higher packet loss rate as shown in Fig. 2(b) and the total throughput is kept below the limitation.

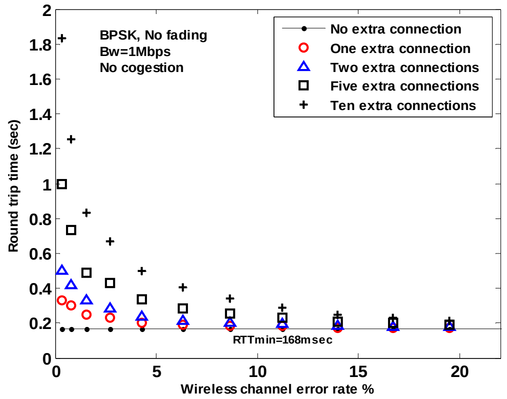
The congestion effect due to extra TFRC connections is also depicted in Fig. 3. It is found that, as the number of extra TFRC connections increases from



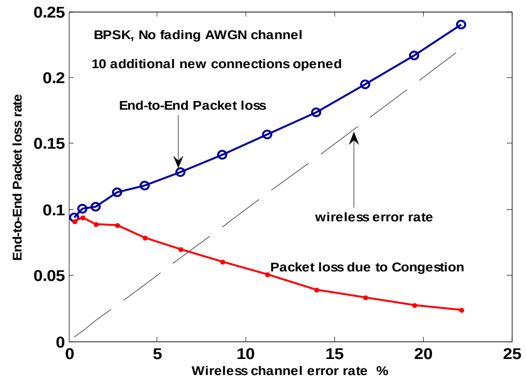
(a)



(b)

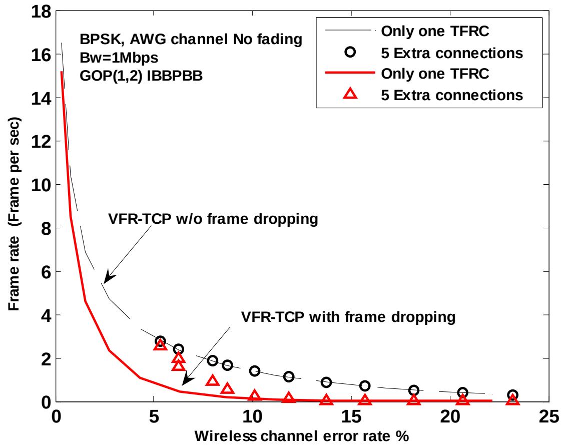


(c)

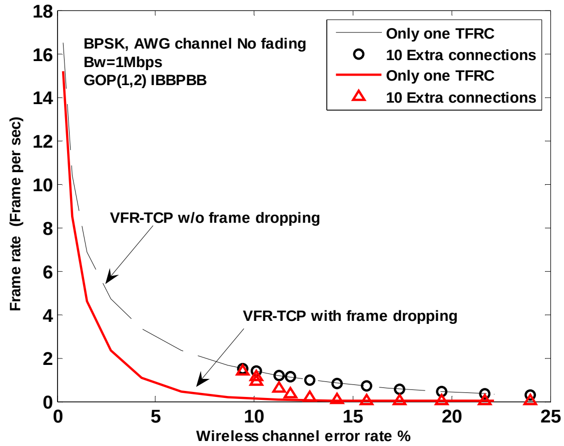


(d)

Fig. 2 TFRC performance: (a) maximum number of TFRC connections, (b) total throughput, (c) expected round trip time, (d) end-to-end packet loss rate with 10 extra connections



(a) with 5 extra connections



(b) with 10 extra connections

Fig. 3 Predicted frame rate versus channel error rate

Fig. 3(a) to Fig. 3(b), the minimum packet loss rate increases and it affects the maximum throughput achievable on each video connection.

As a result, the predicted performance decreases as the number of extra connections is increased due to the end-to-end packet loss rate. In general, it is evident that the performance of VFR-TCP model can be controlled not only by adjusting the number of extra connections opened but also GOP pattern can be chosen for maximum frame rate.

Figure 4 depicts the video quality, in terms of the quantizer scale Q, as a function of the playable frame rate for a single TFRC connection. A dashed line shows the relationship between a quantizer scale and a wireless channel SNR. We assume that the state of a wireless channel is affected only by channel bit errors. 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]. The GoP structure is GoP(1,2)=IBBPBB. During a session, a server regulates the video rate in accordance with the TFRC rate by quality scaling. Y-axis on the left of Fig. 4 corresponds to the playable frame rate of the case of our VFR-TCP, where a frame-dropping is employed as a means of quality scaling. On x-axis, a quantizer scale is derived by substituting the TFRC sending rate as the resultant required bandwidth $BW(640\times 480, Q, 30)$ in Eq. (2). Therefore, X-axis and Y-axis are indirectly related to each other through the channel error rate or the TFRC rate. 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. On the other hand, the SNR scalability results in coarse and mosaic appearances.

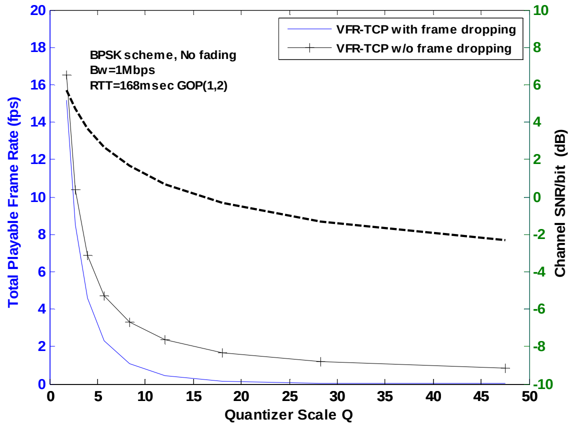


Fig. 4 Video quality of one TFRC connection versus the predicted frame rate and the channel SNR

5 Conclusion

In this paper, we present first a wireless channel model for under and full utilized bandwidth. A variable frame rate model based on TCP-Friendly rate control is also considered over a wireless channel. The proposed work estimates QoS for the video streaming in terms of frame rate as well as the quality factor (Quantizer factor Q). Simulation results showed that the proposed model introduced a good performance. FEC schemes can be proposed for further work to achieve more robust transmission of TFRC video flow over a wireless channel.

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