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AN EFFICIENT ERROR-ROBUST WIRELESS VIDEO TRANSMISSION USING LINK-LAYER FEC AND LOW-DELAY ARQ SCHEMES

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Video transmission often suffers from various errors over wireless networks. Due to errors, the discarded link layer packets impose a serious limitation on the maximum achievable throughput over wireless channel. To face this challenge and to improve the overall TCP-Friendly video throughput, this paper proposes a new robust error-model for MPEG-4 video stream over a point-to-point wireless network. A noisy wireless channel is modeled for random bit errors causing packet loss with some restrictions on the design parameters including packet length, modulation format, and channel SNR. By this model, efficient bandwidth access from wireless network is achieved via a hybrid scheme of channel coding which acts as a Forward Error Correction (FEC), and Automatic Repeat Request (ARQ) protocol at a radio link layer. The new results show that a good video quality of service (QoS) can be estimated in terms of play-out frame rate (in frames/sec) when a maximum channel coding throughput is achieved. Further, this proposed model can improve drastically the end-to-end video quality at high wireless channel errors and low-delay of ARQ scheme.

Key words: MPEG, TCP-Friendly, Wireless video, Video quality, QoS, Channel Coding, ARQ.
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1 Introduction

Recently, many popular wireless multimedia networks cannot provide a guaranteed quality of service (QoS) in spite of the increase in demand on multimedia applications such as real-time video streaming, video conference, and video on demand. To this end, it is essential to rely on QoS metrics pertinent to wireless links in terms of data loss, delay, and throughput. In practice, many major challenges of video traffic are faced on the wired and wireless Internet links [1-3]. Some of these challenges deal with high packet loss rate due to the congestion of buffer overflow over wired networks; and others are mainly faced by the characteristic of wireless links, which are mostly suffering from low bandwidth and high error rates due to the noise, interference, Doppler effect, multi-path fading and time-dispersive effects introduced by the wireless air interface [4-5].

On the other hand, the compressed video bitstream is very sensitive to bit errors over wireless channel. Since a compression algorithm often uses variable-length-coding (VLC) codes [5], errors affect not only the symbol located at the error point, but also the succeeding symbols. As a result, this error propagation will cause packets corruption and lead eventually to a significant degradation in the quality of reconstructed video sequence when the wireless channel conditions are bad. Therefore, a robust real-time video transmission over wireless links is still open issue to achieve good perceptual quality at the client end.

Unlike typical Internet traffic, streaming video is also 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 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. Thus recent researches [6-7] have proposed rate-based TCP-Friendly protocols for steaming media as alternatives to UDP over wired/wireless networks.

To construct wireless link models that can provide QoS metrics for video applications under diverse wireless conditions, many recent analytical models have been established separately or jointly across multiple layers or cross-layer. For example, at separate layers, energy-consumption models for hardware and path-loss SNR models and finite-state Markov chain (FSMC) channel model at the physical layer; Rician, Rayleigh, Nakagami fading models at the physical layer (i.e., the hardware and radio issues could be included in the physical layer); and queuing models at the data link layer [3]. These models are suitable for traditional computers or telecommunication networks; however these separately-layered models may not fit multimedia wireless networks. Thus QoS support involves radio-resource management (RRM), e.g. power control and bandwidth scheduling, across multiple-layers. Liu *et al.* [3] have focused analytically on adaptive wireless links using cross-layer model in order to support QoS in terms of queuing delay, packet loss, and throughput.

Additionally, several packet-loss models have also been devoted on video quality metrics. A QoS of video flows at the media server often denotes these metrics either temporal scaling (frames per second) or quantization level scale. One model [8] has analyzed the effects of packet loss on the observed frame of MPEG-4 video at the receiver using UDP, and by selective reliability they improved the quality of received video. Others models [9-11] were pursued in conjunction with TCP-Friendly protocols to deliver streaming media flows over the Internet. These flows often utilize lower latency repair approaches, such as static, prior, or adaptive forward error correction (FEC). To preserve real-time media play-out, the servers must scale back their data rate to match the TCP-Friendly data rate using media scaling.

In summary, to improve the video quality over wireless networks at high loss rates, there are many analytical approaches which can be pursued such as adaptive rate control [6], passive error recovery (re-transmission) [8], frame-interleaving, [12], error-concealment [13], adaptive modulation [14], forward-error-correction (FEC) at packet-level and/or channel bit-level [15]. Effectively, FEC adds redundancy codes to the original information via either convolutional codes, like RCPC [1,16]) or block codes [1-5], like CRC, RS, and BCH codes. These schemes help to combat the worst-case errors and sustain the quality of video. For example, H.261 and H.263 videos use a (511,492) Bose Chaudhuri Hochquenghem FEC checksum which can correct 2 bits of random errors per packet. However, one problem of FEC is that it cannot efficiently handle burst errors. Thus some systems use frame-interleaving to solve this problem, but such scheme introduces a large delay, which must be avoided for real-time video transmission.

Furthermore, ARQ scheme can efficiently recover packet loss and burst-errors. In fact, the transmitter needs an ARQ buffer to hold the sent-out packets until the receiver acknowledged receipt of correct data. Thus, if there is data corruption then the transmitter will resend the data packet until the delay constraint cannot be held. Many studies [5,17] have combined ARQ and FEC schemes plus error-tracking at video proxy server to provide error-resilience tools. These tools are used to enhance the error performance in terms of QoS under predefined error conditions. However, the main problems of ARQ involve a feedback channel and retransmission might fail when the round-trip-time is long.

In this work, we propose a robust error-model for TCP-Friendly MPEG-4 video traffic over point-to-point wireless network. A wireless channel is assumed under an additive White Gaussian noise (AWGN) or slowly fading and some restricted design parameters including packet length, modulation format, and a range of channel SNR. Thus the physical layer can capture a Signal-to-Noise Ratio (SNR) versus bit error rate (BER) through a simple Binary-Phase-Shift-Keying (BPSK) modulation. To maximize the network throughput and to enhance the perceptual video quality, a BCH FEC channel coding is applied at radio link layer according to the channel state estimation. Moreover, a hybrid scheme of FEC and ARQ protocol at data link layer are both considered to provide more error control model against frequent packet loss rate. As a result, the proposed model can drastically predict a good playable frame rate of MPEG-4 video under various error-corrections, and specifically a low-delay of ARQ scheme introduces a higher end-to-end video quality.

The rest of the paper is organized as follows. Section 2 presents the modeling preliminaries followed by Section 3 for wireless link model in terms of BER and throughput. In Section 4, we derive the analytical QoS model for MPEG-4 video. Numerical results are explained in Section 5, and finally Section 6 summarizes conclusions.

2 Modeling Preliminaries

2.1 Video Quality

MPEG video is considered to be a standard video compression for wireless network. Figure1 illustrates a typical Group of Pictures (GoPs) structure of an MPEG stream. 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. The loss of one P frame can make some of other P and B frames undecodable, and the loss of one I frame can result in the loss of the whole GoP [9]. A GoP pattern for MPEG-4 video can be identified in similar manner of MPEG-2 video. Let $G(N_P, N_{BP})$ and $N_B = (1+N_P)\times\ N_{BP}$, where N_B corresponds to the total 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. An example, GoP(2,2) “IBBPBBPBB”, where N_P =2 and N_{BP} =2.

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) [18]. In this paper, the QoS of MPEG-4 video is defined only in terms of play-out frame rate at client end.

2.2 Network Model

Most of studies on error control of video transmission today uses point-to-point model. This model is shown in **Figure 2**. Various errors are encountered when two terminals are linked. These errors can mainly be classified as packet loss due to overflow buffer (congestion) and/or error bits due to wireless features environment [5]. Video input goes to encoder part of codec to form bitstream and is then transmitted to the network. At the decoder side, the video is received first by the decoder and then displayed on the terminal. In this network model, the network is treated as a black box whereas the error probability and delay of the network are essential parameters for a perceptual video quality at the client end. This point-to-point network applies Internet video communications since end-users have no privilege altering the network configuration which may affect error performance.

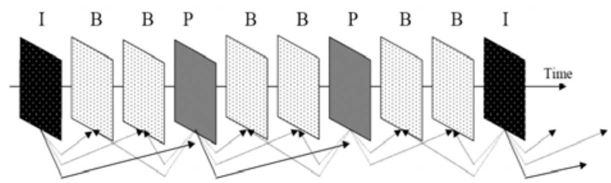


Figure 1. A structure of a GoP and inter-frame dependency relationship.

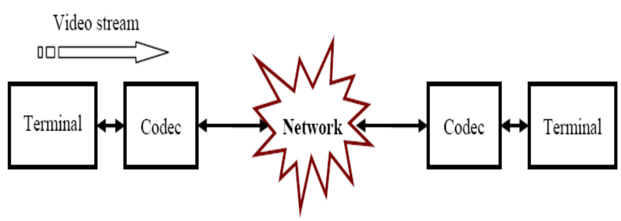


Figure 2. Block diagram of a traditional network model for point-to-point network.

2.3 Error Control Scheme

Shannon's channel coding theorem states that if the channel capacity is larger than the data rate, a coding scheme can be found to achieve small probabilities [19]. The basic idea behind Forward Error Correction (FEC) is to add redundancy to each information (payload) packet at the transmitter. At the receiver this redundancy is used to detect and/or correct errors within the information packet.

The binary (N,K) Bose-Chadhuri Hocquenghem (BCH) is a common FEC scheme based on block coding. This code adds redundancy bits to payload bits to form code words and can correct a certain number of bits, see [20] for details. An important subclass of non-binary BCH codes is the Reed Solomon (RS) code. An RS code groups the bits into symbol and thus achieves good burst error suppression capability.

We therefore consider a realistic video transmission system in **Figure 3,** which consists of a transmitter, a receiver, and a communication channel with a limited bandwidth B_w. The transmitter constructs packets of K bits and transmits the packets in a continuous stream. To ensure that bits received in error are detected, the transmitter attaches a C bit FEC (such as CRC or BCH) to each data packet, making the total packet length $K + C = L$ bits. This packet is then transmitted through the air and processed by the receiver. The FEC decoder at the receiver is assumed to be able to detect all the errors in the received packets. (In practical some errors are not decodable, but this probability is small for reasonable value of C and reasonable SNRs). Upon decoding the packet, the receiver sends an acknowledgment, either positive (ACK) or negative (NACK), back to the transmitter. For case of our analysis we assume this feedback goes through a separate control channel, and arrives at the transmitter instantaneously and without error. If the FEC decoder detects any error and issues a NACK, the transmitter uses a selective repeat protocol to resend the packet. It repeats the process until the packet is successfully delivered.

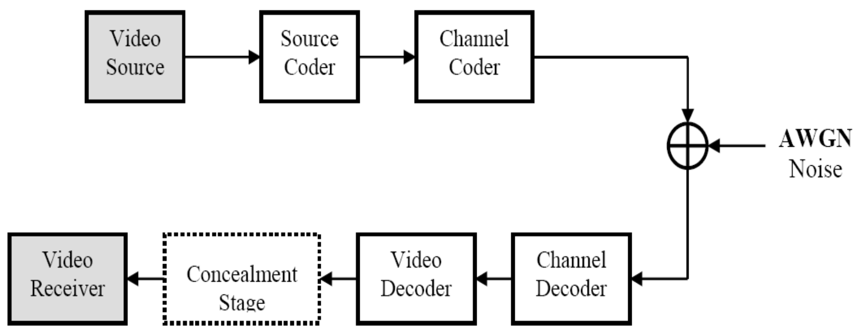
More precisely, in **Figure 3**, the source coder provides compression (usually lossy) of the video while the channel coder introduces redundancy in order to combat error caused by a noisy channel.

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.

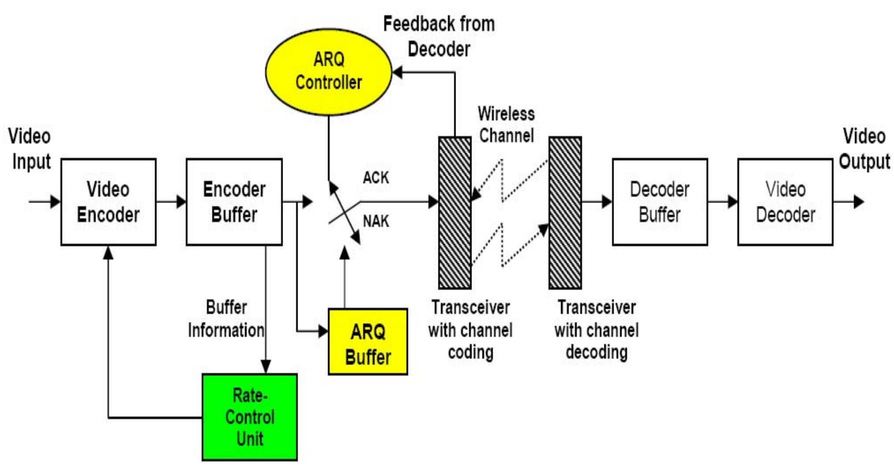
In this paper, the concealment stage is not considered in our proposed approach. Thus we assume a typical model of wireless video communication; whereby a video server sends a video stream to a receiver via a wireless channel corrupted highly by an AWGN, and no interference from other signals [21].

2.4 TCP-Friendly over Wireless

The wireless link is characterized by available bandwidth, i.e. B_w. Further, the effective packet loss rate p_w is mainly arising due to the corruption of bit errors ignoring the congestion due to opening many concurrent TFRC video connections on the same channel. Hence, we consider only the bit error rate (BER) over wireless link which is the substantial reason of generation this packet loss over channel. We use the following model for TFRC to analyze the problem as in [7],



(a)



(b)

Figure 3. A typical wireless video communication system (a) corrupted by AWGN noise (b) based-ARQ scheme

$$B_{TFRC} = \frac{k \cdot S}{T_{RTT}\sqrt{p_w}} \quad , \qquad (1)$$

where B_{TFRC} represents the upper-bound of the network throughput (i.e. effective sending rate), S is the packet size, T_{RTT} is the end-to-end round trip time, p_w is the end-to-end packet loss rate due to only bit errors over wireless link, and k is a constant factor between 0.7 [22] and 1.3 [23], depending on the particular derivation of (1). Although this model is simple, easy to analyze, it can captures all the fundamental factors that affect the sending rate. Note that the results we derive based on this simple model which can be extended to another more sophisticated models, such as the one used in [24].

3 Wireless Link Model

In this section, we develop a wireless link model across the physical layer (hardware-radio) and data link layer, which enables us to analytically derive the desired QoS metrics in terms of BER, packet loss and throughput.

At hardware-radio link layer, to obtain p_w, frequent and random bit errors of a simple noisy wireless channel are considered without taking any fast fading effect. In this model, we will refer to the term "*mod m*" to indicate to a specific choice of an uncoded modulation. Thus we define $P_{e,m}(L,\gamma_b)$ as the probability of error in terms of packet length in L bits and γ_b which is being SNR per bit for uncoded modulation scheme. Also it *refers* to the physical layer packet loss rate (PLR) for a given *mod m*. Then $P_{e,m}(L,\gamma_b)$ can be expressed as a function of the bit error probability p_b as in [3],

$$P_{e,m}(\gamma_b, L) \leq 1-(1-p_{b,m}(\gamma_b))^L \qquad (2)$$

where $L \equiv S = 8l$ denotes a packet length (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.

With the simplifying assumptions of Sub-section 2.3, we can define at the radio data link layer the maximum throughput of a channel coding as the number of payload bits per second received correctly for uncoded BPSK scheme [25],

$$B_{Phy,m} = \frac{L-C}{L}\Re_b\left[1-P_{ec,m}(\gamma_b, L)\right] \qquad (3)$$

Assume C may not only involve error-correction bits, but any *extra bits* which are related to a header of ARQ packet scheme (if ARQ scheme effect is taken into account). The term [1- $P_{ec,m}$] denotes the packet success rate (PSR) defined as the probability of receiving a packet correctly, $\Re_b$ is the bit rate (in bps), and γ_b is the SNR per bit given by,

$$\gamma_b = E_b/N_o = \frac{P}{N_o\Re_b} \qquad (4)$$

where E_b, N_o, and P represent the bit energy, the one-sided noise power spectral density, and the received power respectively.

To compensate for low SNR region in some technologies of spread spectrum modulation, where each bit is multiplied by a chip sequence and spread into K bit times, or time division multiplexing, a common idea of *non-extending* a period time of the packet can be applied. This does not increase the energy per information bit, and such variation is called namely an *adaptive* FEC.

We now consider a block code FEC scheme with redundancy of C error correction bits adding to the packet, but *without extending* the total packet length (in bits) to exceed a maximum length $L_{\max}$. In case of nine parity bits in BCH code, the packet error packet loss rate, $P_{ec,m}$, with maximum error capacity t can be expressed as [5],

$$P_{ec,m} = 1 - \sum_{i=0}^{t} \binom{L_{\max}}{i} p_{b,m}^{i} (1 - p_{b,m})^{L_{\max} - i} \qquad (5)$$

where,

$$t_{\max} = \frac{L_{\max} - L_{ARQ,H}}{9} \qquad (6)$$

Note that a maximum error-correcting bit is $C_{\max} \equiv 9 \times t_{\max}$, and $L_{ARQ,H}$ represents the extra bits of ARQ packet's header, i.e. 16 bits. Hence, a typical BER performance can be improved when the effectiveness of FEC coding is taken into account. On the other hand, the packet error in burst-error condition cannot easily be modeled by a single equation. The reason is that the distribution of error bits is not uniform. Thus Gilbert model is mainly used in this case. This model is out scope of this paper [3]. To simplify the estimation of BER performance, we apply a BPSK scheme over AWGN channel for upload/download streams. Since p_b in AWGN channel decays exponentially as γ_b increases, the probability of bit error can be given by [2],

$$p_b = Q\left(\sqrt{2\ \gamma_b}\right), \qquad (7)$$

$Q(.)$ is Gaussian cumulative distribution function.

The validity of the analysis above is not limited to BPSK bit error model. This model is used for the sake of simplicity. It can, however, be modified to take into account the multi-path effects of wireless channels. The log-normal shadowing path loss model can be used, for example [16].

4 Analytical MPEG Error Model

In this section, we adapt the QoS metrics of wireless link in **Section 3** for TCP-Friendly video traffic with the aim to achieve the desired QoS-oriented MPEG model at the application layer in terms of temporal scalability; i.e., providing high perceptual play-out frame rate of MPEG-4 video stream at the client end. The proposed analytical error model is based on the following scenario with three assumptions:

Assumption (1): A TCP-Friendly flow is considered with data rate (throughput) not exceeding the maximum data rate of TCP connection in the same network conditions. Here, the TCP-Friendly sending rate is controlled in accordance with network conditions as TCP does, on the wired Internet [24]. By adjusting the sending rate to the desirable rate determined by an underlying TCP-Friendly Rate Control (TFRC), one can achieve the required video quality of video applications over a wireless link.

Assumption (2): When there is no extra-traffic due to concurrent TFRC video connections on wireless channel, this scenario can be applied as follows. The wireless link is assumed having bandwidth limited and there is no congestion of video connections. Hence, a packet loss is only due to wireless channel bit errors. Furthermore, the minimum RTT in (1) (i.e., $T_{RTT} = T_{RTT\min}$) can be achieved if and only if $B_{TFRC} \leq B_w$. The backward route from video receiver to video server is assumed to be congestion-free but not error-free due to bit errors [7].

Assumption (3): Optimal control rate should result in the highest possible throughput and the lowest packet loss rate by using (2) or (5). To avoid any network instability, B_b is regarded as the available bandwidth for video streaming and adjusting the video traffic, the high-quality video play-out at a receiver can be expected. Hence, for an under-utilized channel, $B_{TFRC} \leq B_b < B_w$ holds when only one TFRC connection exists.

Within this scenario, the effective physical layer throughput in (3) can be again expressed under various error-correction conditions using BCH code as [13,25]

$$B_{Phy,\mathrm{mod}} = A_{\max,ec}\left[1 - P_{ec,m}(\gamma_b, L)\right], \qquad (8)$$

The factor $A_{\max,ec} = \Re_b (L - C)/L$ represents the maximum achievable data rate in (bps) for *mode m*. The probability of packet error $P_{ec,m}(\gamma_b, L)$ is defined as the effective p_w for maximum error capacity of t symbols. $A_{\max,ec}$ should be defined in terms of channel SNR in order to evaluate the effective TFRC network throughput, i.e., by setting $A_{\max,ec}$ as a maximum TFRC throughput defined in (1).

On the other hand, since TFRC sender needs the only congestive loss event rate, so it may result in bandwidth some underestimation if the original loss event rate ignores congestion effect and only uses directly the packet loss due to bit errors using (2) as the effective loss event rate. Thus our proposed solution is to discount the reported network throughput (i.e., a maximum throughput achievable at the receiver) by dynamically adjusted factor d [26]. Then,

$$B_{Phy,\mathrm{mod}} = B_b \times d\,, \qquad (9)$$

Where d is being the discounting factor and can take any value between 0 and 1 depending on error-correction condition. Under the TFRC constraint of (1), and by setting $A_{\max,ec}$ equals B_b in (9), then the achievable throughput can be rewritten as,

$$B_{Phy,\mathrm{mod}} = B_b\left(1 - P_{ec,m}(\gamma_b, L)\right), \qquad (10)$$

By equating (3), (9) and (10), the effect of the discounting factor d can be expressed in two formulas as,

$$d_1 = \frac{L - C}{L}\left[1 - P_{ec,m}(\gamma_b, L)\right] \qquad (11)$$

and,

$$d_2 \cong [1 - P_{ec,m}(\gamma_b, L)] \qquad (12)$$

Note that equation (12) ignores the effect of any extra bits C associated to ARQ packet.

4.1 Maximum Throughput of Channel-Coding

In order to achieve maximum performance in an erroneous noisy channel environment, a careful design of the channel coding is important. In this section, BCH is investigated under only random-error conditions.

When a typical ARQ packet is adopted as shown in **Figure 4,** the header needs 16 bits. This could be a big overhead in short packets (e.g. 511 or 640 bits). Since the delay is proportional with the packet length, hence a packet length is modelled with only 511 bits to fit with packet-length restriction of BCH code [5].

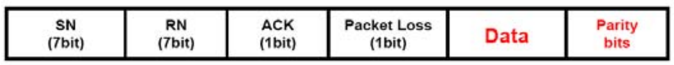


Figure 4. A typical ARQ packet format used in our proposed scheme

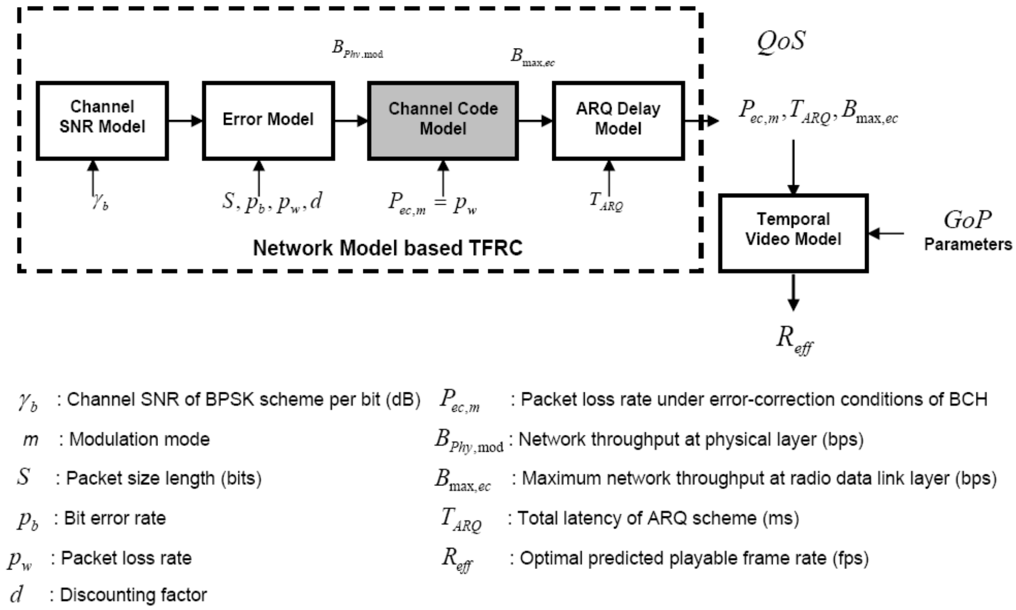


Figure 5. A QoS schematic diagram of a proposed scheme over wireless network

In BCH code, since the error capacity is nine parity bits per error bit for a 511-bits packet, then the maximum throughput (i.e., transmission efficiency) of this code can be calculated as,

$$\eta_{BCH} = \left(1 - PLR_{BCH}\right) \cdot \left(\frac{L_{ec}}{L_{\max}}\right) \qquad (13)$$

where $L_{ec} \equiv L_{\max} - L_{ARQ} - C_{\max}$ denotes the length of encoded packet, and and $C_{\max} \equiv 9 \times t$ is the length of *inclusive period* of total parity bits per packet. Note that $L_{\max}$ does not exceed 511 bits. For simplicity, we can rewrite (5) as,

$$PLR_{BCH} = 1 - \sum_{i=0}^{t} \binom{511}{i} p_b^i \left(1 - p_b\right)^{511-i} \qquad (14)$$

p_b is the bit error rate and PLR_{BCH} is the packet- error under error-correction condition with capacity of t symbols at radio link layer. The goal is to obtain t under determined p_b for maximum throughput. Using a simple computer program to search all possible t, we got the same result in [5] as shown in **Section 5**. As a result, by this maximum throughput of channel coding, the effective optimal network throughput can be evaluated as follows in this formula:

$$B_{\max,ec} = B_{phy,\mathrm{mod}} \times \eta_{BCH} \qquad (15)$$

4.2 Temporal Scaling Model

To estimate the number of playable frames at a receiver, packet loss rate is considered random and stationary over wireless point-to-point link. Thus the analytical model designed over wired Internet for MPEG-2 video stream in [9], is modified in this paper for a GoP pattern of MPEG-4 for point-to-point video communication channel. This model employs a TFRC protocol to control the sending rate on the frame-level in accordance with loss of packets caused by packet corruptions due to bit errors. Subsequently, a GoP rate (in GoP per second) can be analytically expressed using TFRC protocol and the frame dependency relationship of I, P, and B frames. Hence, the resultant playable frame rate (PFR) R can be computed as follows,

$$G = \frac{B_{\max,ec} / L_{\max}}{S_I + N_P S_P + N_B S_B}, \qquad (16)$$

For numerical example, we use $L_{\max}$ =511 bits. $B_{\max,ec}$ of (15) is the effective network throughput received at the client in (bps), 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 GoP size can be expressed as,

$$S_{GOP} = 1 + N_P + N_B, \qquad (17)$$

The total effective playable frame rate (PFR) can be derived as in [9],

$$R_{eff} = G.W_I \left[1 + \chi_P + N_{BP} \cdot W_B \cdot \left(\chi_P + W_I \cdot W_P^{N_P}\right)\right] \qquad (18)$$

where,

$$\chi_P = \frac{W_P - W_P^{N_P+1}}{1 - W_P}, \text{ and } W_i = (1 - p_w)^{S_i} \qquad (19)$$

where W_i stands for the successful transmission probability of the *i-th* frame type (I, P, and B) in a GoP pattern without taking into account any packet FEC correction at application layer, and S_i denotes packet size of the *i-th* frame type.

When BCH channel coding of (14) is employed at the radio link layer, the end-to-end packet loss rate is being p_w, and then the efficient bandwidth access (optimal network throughput) can be achieved over a highly corrupted wireless channel. Hence, the predicted video quality (temporal scaling) can be eventually regulated by the video server to fit with the QoS user's preferences.

4.3 ARQ Delay Analysis

Delay constraint is essential for real-time video communication system when automatic repeat request (ARQ) is combined with channel coding. Thus the total delay from encoder to decoder is given as [27],

$$T_{total} = T_{enc} + T_{net} + T_{buf} + T_{dec} + T_{ARQ} \;, \qquad (20)$$

where T_{enc} is the encoding time including channel code processing, T_{net} is the network delay from transmitter to receiver, and T_{buf} is the buffer delay, which is proportional to data in transmitter buffer and its affected by rate-control algorithm of TFRC. T_{dec} is the time interval from receiving all data to displaying them on screen. It can be considered as the decoder's latency. Finally, the last term is the ARQ delay.

The ARQ delay is mainly the largest part of the total delay items (i.e., dominated) if the network condition is bad. Thus it is important to reduce this ARQ delay as much as possible by rewriting (20) in terms of the effective p_w and $T_{RTT\min}$ of TFRC protocol. By using ARQ delay model of [28], we can approximate the total delay such as,

$$\begin{aligned} T_{total} = T_{ARQ} &= p_w \times T_{RTT\min} + p_w^2 \times T_{RTT\min} + \ldots\ldots \\ &= \frac{p_w}{1-p_w} T_{RTT\min} \qquad (21) \end{aligned}$$

The ARQ delay of (21) represents the delay penalty for resending the packets; meanwhile the packet-error probability p_w can be controlled by the FEC scheme. Note that Equation (21) is ignoring the effects of others propagation delays. Hence, the effective $T_{RTT\min}$ is reduced by the factor of $p_w/(1-p_w)$ due to the error-correction conditions. This significant reduction factor in the round-trip time will considerably enhance the video quality performance (in number of frames/sec) as compared to a case of assuming a fixed minimum RTT only.

5 Numerical Results

5.1 Methodology

In this section, we investigate *two* cases of delay constraints as follows. (*i*) When a minimum round-trip time (RTT) is fixed at a certain value, and (*ii*) when the effective end-to-end RTT is degraded by a factor of $p_w/(1-p_w)$ less than 1. Hence, to find the predicted QoS metrics for video stream, a following scenario is proposed as shown in **Figure 5**:

1. The video source must determine constantly a maximum fixed 511-bit packet according to BCH encoding restriction.
2. As soon as the video flow faces a network constraints in terms of QoS network (such as packet loss rate, delay and bandwidth,) over wireless channel, the feedback signal via channel state estimation will inform the video source to control its packet condition in order to adapt the rate of video streaming to the available network throughput using TFRC mode.
3. Effectively, the video system first obtains a channel state in terms of SNR per bit using BPSK scheme and then assesses the corresponding bit-error rate p_b on the wireless link.

4. For worst-channel state, video encoder must maintain a proper BCH code with a restriction of maximum packet size not exceeding 511 bits. Here, the packet loss rate p_w can be estimated using (14) for various error-correction conditions.
5. Then the video quality in terms of the temporal scalability, i.e., playable frame rate can be evaluated by (18).
6. As far as the total RTT of ARQ scheme is reduced by factor $p_w/(1-p_w)$, the perceptual video will considerably enhance under various FEC conditions.

5.2 Results and Performance Evaluation

Numerical results have been conducted for only one TFRC video connection over a typical point-to-point wireless network. **Table 1** describes a typical parameters setting used in the simulation for wireless network in GSM or CDMA systems including GoP pattern parameters for MPEG video stream [5,7]. A channel capacity is assumed at the limited bandwidth B_w, and upper-bound of the network throughput does not exceed B_w. The rate control of TFRC scheme which can handle packet loss on the encoder side will absorb the loss of throughput. The error-condition used here is only modeled for random errors. In order to get maximum performance, the BCH code is used. The optimal BCH code configuration is examined in **Figures 6-7**.

Table (1). Wireless Network settings and GoP parameters used in simulation

	Wireless network parameters
T_{RTT}	168 [ms]
k	1.2
B_w	1 Mbps
$L_{\max}$	511 bits for BCH code in radio-link layer
Modulation scheme	BPSK (upload/download)
γ_b	1 ... 10 [dB] Channel SNR/bit
	GoP Pattern parameters
F_o	30 [fps] reference frame rate at the video source.
S_I	25 [packets]
S_P	8 [packets]
S_B	3 [packets]

Figure 6 shows the QoS performance of the wireless channel in terms of packet error rate versus channel SNR/bit and bit error rate under various error-correction codes. It is noticed that there is a clear degradation in the resultant PLR when error capacity for correction increases as in [5]. Therefore, **Figure 7** draws the available channel state in terms of PLR, BER, and optimal channel coding throughput ratio (in %) before video traffic commences its transmission over a noisy wireless channel under these various error conditions and error-correction codes. It is clearly found that the optimal channel coding throughput decreases as the bit errors increases although error-correction capacity achieves 31 bits at roughly 12 % PLR.

To demonstrate the optimal video quality performance of our proposed scheme, **Figure 8** depicts optimal play-out frame rate (in fps) under various error-correction codes, when an optimal channel coding throughput is achieved. It is found that a significant improvement in the play-out frame rate when a RTT of ARQ scheme is reduced by the factor of $p_w/(1-p_w)$. For example, a full video motion (approximately becomes 26 fps; i.e. 87% of full-motion 30 fps) can be achieved at high

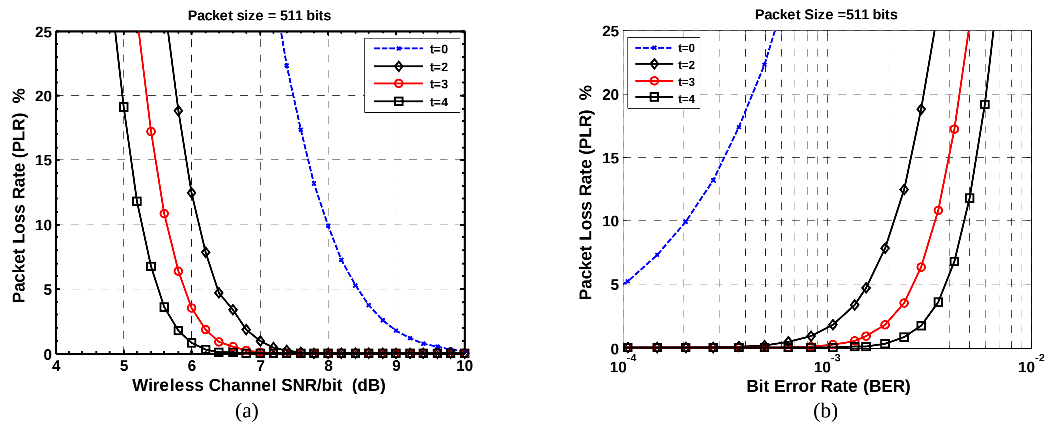


Figure 6. Packet error under various error conditions and error-correction codes of BCH (a) wireless channel SNR/bit using BPSK scheme, and (b) bit error rate

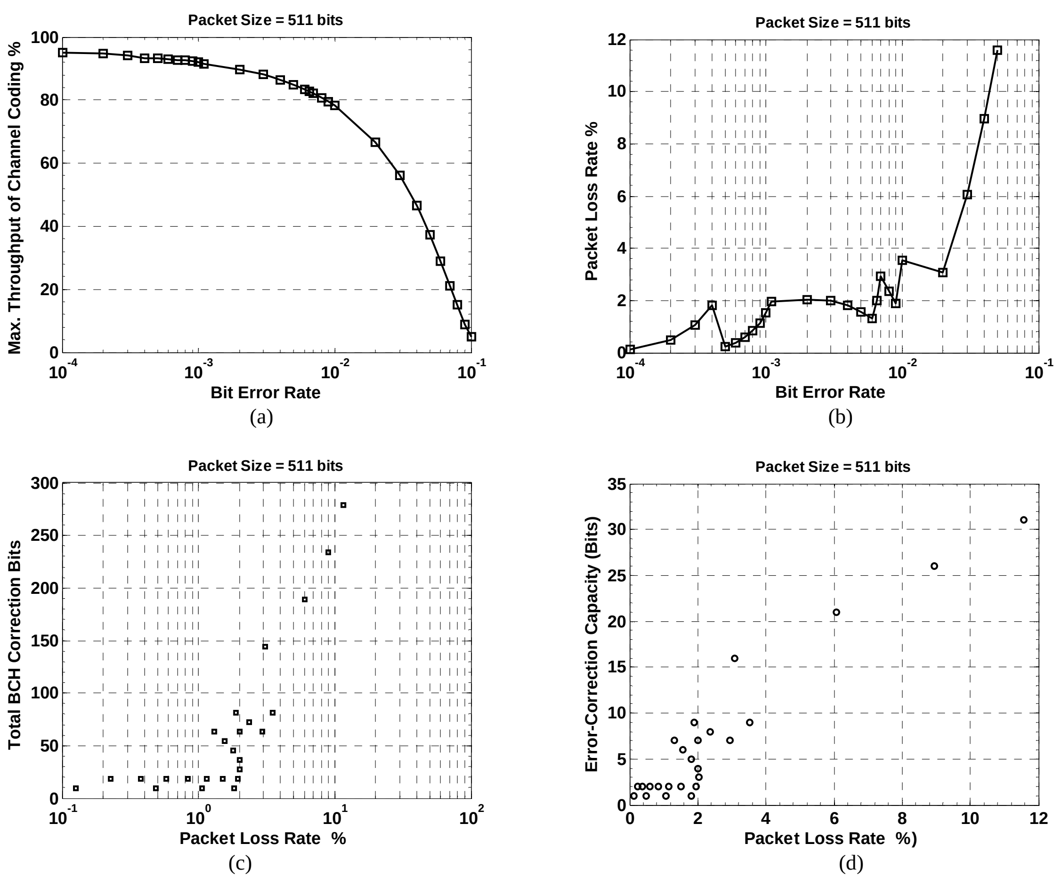


Figure 7. Channel Code Performance under various error conditions, **(a)** Maximum throughput [5], **(b)** Packet error at a maximum achievable throughput [5], **(c)** Total error-correction bits , and **(d)** Error-correction capacity

Channel SNR (above 8.4 dB) with only 1 or 2 bit-correction at bit error rate below $1x10^{-4}$ or $4x10^{-4}$ respectively. However, as far as bit errors estimation increases on wireless channel via the feedback channel state, the resultant play-out frame must be improved by adding extra error-correction bits to achieve the optimal end-to-end quality. For example, 6 bit-corrections can be applied for bit error rate below $3x10^{-3}$ (i.e., SNR is above 5.75 dB) providing 87% of full motion rate (**See Fig. 8**).

On the other hand, when the overall delay in terms of RTT is reduced; another significant improvement in video quality scaling can be clearly observed. An improvement achieves on average nearly 8-10 fps at optimal channel coding throughput 80-90 % as shown in **Fig. 8 (a)** and **Fig. 9 (b)**. Furthermore, it is observed that a good video quality can be achieved when error-correction

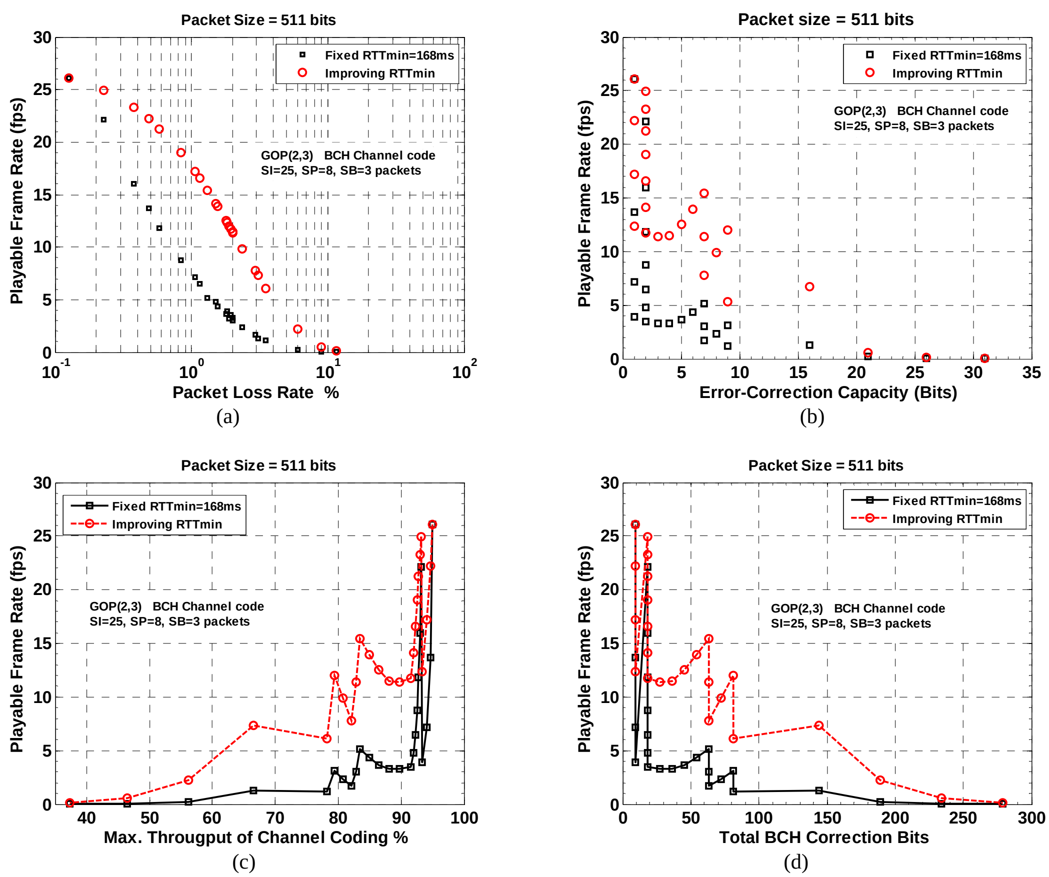


Figure 8. Video Quality Performance under various random packet loss conditions when a maximum Channel code throughput is achieved.

capacity *t* is below nine bits. As a result, a video transmission system can rely on this error-correction capacity when an improving in overall minimum RTT is taken into account. Hence, a higher video quality can be perceived with this robust video communication model under low-delay constraint as shown in **Fig. 8 (b). Table (2)** illustrates some examples in more detail.

As a result, we can summarize these findings as follows:

Table (2). Video Quality Performance when a maximum channel code throughput is achieved under various channel conditions.

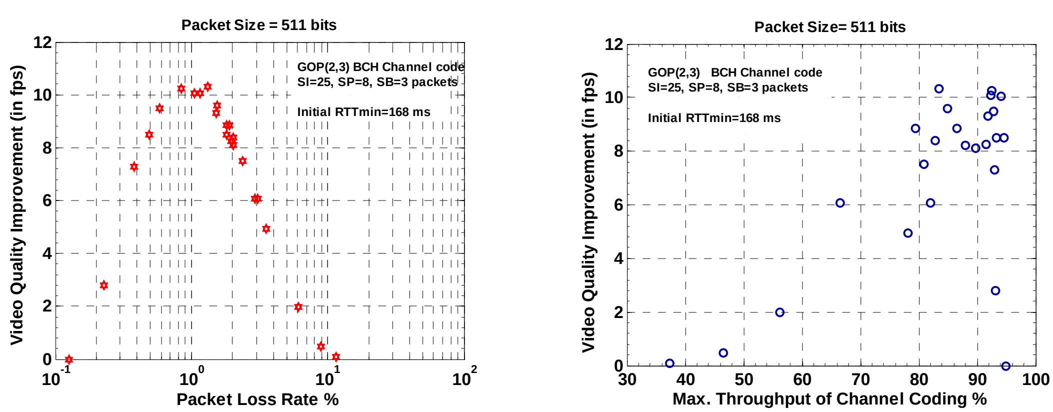
(a) Fixed RTT=168 ms, GOP(2,3)

Channel State	**Error Type**	**BCH Parity Bits**	PLR_{BCH} **%**	**Effective Bandwidth** $B_{\max,ec}$ (kbps)	η_{BCH} **%**	PFR (fps)
C1 (8.40 dB)	1×10^{-4} Random Error (t=1)	9	0.126	80.11*	**94.987**	26.13
C2 (7.35 dB)	5×10^{-4} Random Error (t=2)	18	0.228	71.08	**93.133**	22.13
C3 (6.85 dB)	1×10^{-3} Random Error (t=2)	18	1.5177	27.24	**91.929**	4.801
C4 (5.20 dB)	5×10^{-3} Random Error (t=6)	54	1.552	24.89	**84.962**	4.32
C5 (4.30 dB)	1×10^{-2} Random Error (t=9)	81	3.5225	15.20	**78.163**	1.15
C6 (1.30 dB)	5×10^{-2} Random Error (t=31)	**279**	**11.5745**	**4.01**	**37.377**	**0.0138**

(b) After Improving RTT, GOP(2,3)

Channel State	**Error Type**	**BCH Parity Bits**	PLR_{BCH} **%**	**Overall Delay** (ms)	**Effective Bandwidth** $B_{\max,ec}$ (kbps)	η_{BCH} **%**	PFR (fps)
C1 (8.40 dB)	1×10^{-4}	9	0.126	0.21	80.11*	**94.987**	26.13
C2 (7.35 dB)	5×10^{-4}	18	0.228	0.38	80.11	**93.133**	24.04
C3 (6.85 dB)	1×10^{-3}	18	1.5177	2.58	80.11	**91.929**	14.12
C4 (5.20 dB)	5×10^{-3}	54	1.552	3.1	80.11	**84.962**	13.91
C5 (4.30 dB)	1×10^{-2}	81	3.5225	6.1	80.11	**78.163**	6.08
C6 (1.30 dB)	5×10^{-2}	**279**	**11.5745**	**22**	**30.6**	**37.377**	**0.11**

* Upper-bound bandwidth (network throughput) achievable is 80.11 kbps in our proposed scheme



(a) (b)

Figure 9. An improvement in video Quality when a maximum channel code throughput is achieved.

(i) The obtained optimal channel coding throughput is one good QoS metric over point-to-point wireless link. By a proper choice of error correction capacity *t* under various bit error-conditions, a lowest packet loss rate (PLR) can be achieved.

(ii) Since optimal (maximum) channel coding is achieved under various error-correction codes, a good play-out frame rate (PFR) can be estimated at the client end. However, as far as the error-correction capacity *t* of FEC scheme increases higher than 9 bits (i.e., a code efficiency degrades);

then the predicted video quality will not introduce more additional enhancement in number of frames per second. (**See Fig. 9**).

(iii) A delay of ARQ scheme is reduced by the factor of $p_w/(1-p_w)$ after adding redundancy codes of BCH scheme. Hence, the resultant end-to-end RTT (i.e., the overall network delay) will affect the optimal channel code throughput. In consequence, a significant increase in PFR is achieved when PLR is not greater than 4%. In this case, a significant increment in PFR is 5-10 [fps] as compared to a case of fixed end-to-end RTT when a proper choice of error-correction codes at the server is taken before starting the transmission again on the channel (**See Fig. 8 a** and **Fig. 9 a**).

(iv) **Table 2** reveals examples of random error-conditions used in the simulation. **C1-C6** are channel states with errors ranging from 10^{-4} to 10^{-2}, which are most frequently used in practical conditions. A proper FEC coding can greatly reduce packet-error rate with a significant improvement in the resultant number of play-out frames. The video quality degradation for **C1-C2** is no more than 4 frames in case of fixed RTT, and no more than 2 frames when low-delay is achieved via ARQ protocol used in our proposed scheme. In contrary, [5] introduces PSNR degradation no more than 1.2 dB for the experimental H.236 “Foreman” video sequence and frame rate setting is 10 [fps].

(v) For high error-conditions such as **C3-C4**, the perceptual video at client is still image, where video quality degradation increases as far as FEC code increases if total delay is fixed. After improving RTT, **C3-C4** can attain nearly 14 [fps]. It means that there is an extra improvement by 10 [fps] when we take the effect of maximum channel coding throughput on the total delay over the network.

(vi) The Channels **C5**-**C6** are completely useless in spite of increasing FEC code but after improving RTT, only **C5** can play-out at 6 [fps] despite maximum network throughput is 80.11 kbps. As a result, **Table 3** provides video quality no more than 7.17 [fps] as compared with others models.

Table (3). Video quality comparison among models for wired and wireless networks.

Approach	**Packet-Loss Model**	**Error Control**	**Packet Length, PLR%**	**FEC Code**	*PFR* (fps)
TFRC Wired link [9] GOP(2,3), 12 frames	Frame-level **(due to congestion)** RTT=50 ms	Fixed RS-Code (Application layer) Packet-level	1 Kbytes **PLR=2%**	(1,0,0)	23.58
TFRC Wired link [10] GOP(3,2), 12 frames	GOP-level **(due to congestion)** RTT=50 ms	RS-Code (Application layer) Packet-level	1 Kbytes **PLR=2%**	(1,1,0)	25
TFRC wired-to-Wireless link [11] GOP(2,3), 12 frames	Frame-level **(due to bit errors)** RTT=168 ms	RS-Code (Application layer) Packet-level	1 Kbytes **PLR=1.5%**	(1,1,0)	7.7
Proposed TFRC wireless link GOP(2,3), 12 frames	**Frame-level (due to bit errors)** RTT=168 ms	**BCH code** (Radio data link layer) Bit-level	**64 bytes** (short packet) **PLR= 1%**	**(511,492)** 9 parity bits	7.17

6 Conclusion

This paper has presented a new robust error-model for MPEG-4 video stream over a point-to-point wireless network. The analytical model applies BCH FEC channel coding at the radio link layer to improve the bandwidth access from the wireless link. The video traffic is controlled by TCP-Friendly rate control and Automatic Repeat Request (ARQ). As a result, the network QoS in terms of throughput, packet loss, and delay were evaluated under various error-conditions where a BPSK scheme is applied for upload/download streams. Moreover, the QoS in terms of temporal scalability (frame per sec) at the client has also been evaluated when a maximum channel coding throughput is achieved. A further improvement in video quality can also be achievable when low-delay of ARQ scheme. The results demonstrate that a proposed hybrid scheme introduces a good predicted video quality at high channel bit-errors under various error-correction conditions as compared to other models [9-11] over wired and wireless Internet. However, the future work can be extended to involve another adaptive modulation formats to attain more robust video transmission with lowest propagation delay at the low value range of channel SNR.

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