P- ISSN 1991-8941 E-ISSN 2706-6703 Journal of University of Anbar for Pure Science (JUAPS) Open Access
2008,(2), (1) :83-93

EFFICIENT WIRELESS VIDEO TRANSMISSION VIA LINK-LAYER FEC FOR VIDEO COMMUNICATIONS

Ghaida A. AL-Suhail

Computer Engineering Department, College of Engineering, University of Basrah

ARTICLE INFO

Received: 1 / 4 /2008
Accepted: 24 / 4 /2008
Available online: 30/4/2008
DOI: 10.37652/juaps.2008.15451

Keywords:
MPEG,
TCP-Friendly,
Wireless video,
Video quality,
QoS,
Channel Coding.

ABSTRACT

In this paper, new analysis and performance of robust error-model are presented for MPEG-4 video stream over wireless point-to-point network. Analytical expressions assume a noisy wireless environment causing frequent and random bit errors associated with packets. By this model, the temporal video scalability can be evaluated under TCP-Friendly Rate Control (TFRC) transmission when the Bose-Chaudhuri-Hochquenghem (BCH) channel coding is employed as a forward-error-correction (FEC) at a radio link layer. A FEC provides an efficient throughput access on wireless network. The numerical results clearly indicate that a quality of service (QoS) can be improved at low channel SNR region when the maximum channel coding throughput is achieved.

Introduction

Wireless communication channels are prone to errors due to various physical impairments. Error correcting codes are used to overcome or reduce the impact of these errors. On the other hand, 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]. Therefore, a robust real-time video transmission over wireless links is still open issue to achieve good perceptual quality at the client end.

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 [4], passive error recovery (re-transmission) [3], frame-interleaving, [5], error-concealment [6], adaptive modulation [7], forward-error-correction (FEC) at packet-level [8-10] and/or channel bit-level [11]. Effectively, FEC adds redundancy codes to the original information via either convolutional codes, like RCPC [1]) 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 [12] 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

* Corresponding author at: Computer Engineering Department, College of Engineering, University of Basrah, Iraq.E-mail address: **gaida_alsuhail@yahoo.com**

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.

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). 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. As a result, the proposed model can drastically predict a good playable frame rate of MPEG-4 video under various error-corrections.

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

System Description

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. 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 [8].

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 [3]. 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.

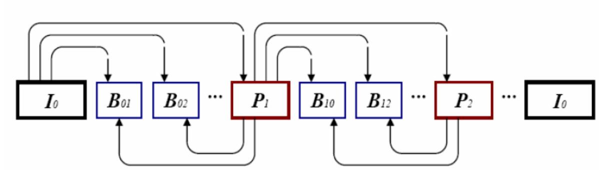


Figure 1: A structure of a GoP and inter-frame dependency

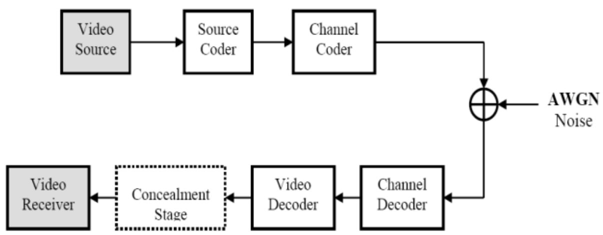


Figure ٢: A typical wireless video communication system corrupted by AWGN noise

Error Control Scheme

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 [13]. 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 ٢, 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).

More precisely, in Figure ٢, 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.

Wireless Link Model

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],

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

where $L \equiv S = 8l$ denotes a packet length (in bits), and the inequality in (1) 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 (1) 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 [14],

$$B_{Phy,m} = \frac{L-C}{L}\Re_b\left[1-P_{ec,m}(\gamma_b, L)\right] \quad (2)$$

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,

$$\gamma_b = E_b / N_o = \frac{P}{N_o \Re_b} \quad (3)$$

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

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 [3],

$$P_{ec,m} = 1 - \sum_{i=0}^{t}\binom{L_{\max}}{i} p_{b,m}^{i}\left(1-p_{b,m}\right)^{L_{\max}-i} \quad (4)$$

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. 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 [14],

$$p_b = Q\left(\sqrt{2\ \gamma_b}\right), \quad (5)$$

$Q(.)$ 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 [15].

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 [4],

$$B_{TFRC} = \frac{k \; . \; S}{T_{RTT}\sqrt{p_w}} \quad , \qquad (6)$$

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 [16] and 1.3 [17], depending on the particular derivation of (6).

MPEG ERROR MODEL

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 [4]. 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 $B_{TFRC} \le B_w$. The backward route from video receiver to video server is assumed to be congestion-free but not error-free due to bit errors [4].

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, $B_{TFRC} \le B_b < B_w$ 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 [7]

$$B_{Phy,\text{mod}} = A_{\max,ec}\left[1 - P_{ec,m}(\gamma_b, L)\right], \qquad (7)$$

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).

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 by dynamically adjusted factor [18]. Then,

$$B_{Phy,\mathrm{mod}} = B_b\left(1 - P_{ec,m}(\gamma_b, L)\right) \quad (8)$$

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 Automatic Repeat Request (ARQ) packet is adopted 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 modeled with only 511 bits to fit with packet-length restriction of BCH code [3].

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,

$$\eta_{BCH} = \left(1 - PLR_{BCH}\right)\cdot\left(\frac{L_{ec}}{L_{\max}}\right) \quad (9)$$

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 (4) as,

$$PLR_{BCH} = 1 - \sum_{i=0}^{t}\binom{511}{i} p_b^i \left(1 - p_b\right)^{511-i} \quad (10)$$

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. As a result, the effective optimal network throughput can be evaluated as follows:

$$B_{\max,ec} = B_{phy,\mathrm{mod}} \times \eta_{BCH} \quad (11)$$

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 [8], 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,

$$G = \frac{B_{\max,ec} / L_{\max}}{S_I + N_P S_P + N_B S_B}, \quad (12)$$

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,

$$S_{GOP} = 1 + N_P + N_B, \qquad (13)$$

The total effective playable frame rate (PFR) can be derived as in [8],

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

(14)

where,

$$\chi_P = \frac{W_P - W_P^{N_P+1}}{1 - W_P}, \text{ and } W_i = (1 - p_w)^{S_i} \qquad (15)$$

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 (10) 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.

NUMERICAL RESULTS

In this section, to find the predicted QoS metrics for video stream, a following scenario is proposed as:

The video source must determine constantly a maximum fixed 511-bit packet according to BCH encoding restriction.

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.

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.

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 (10) for various error-correction conditions.

Then the video quality in terms of the temporal scalability, i.e., playable frame rate can be evaluated by (14).

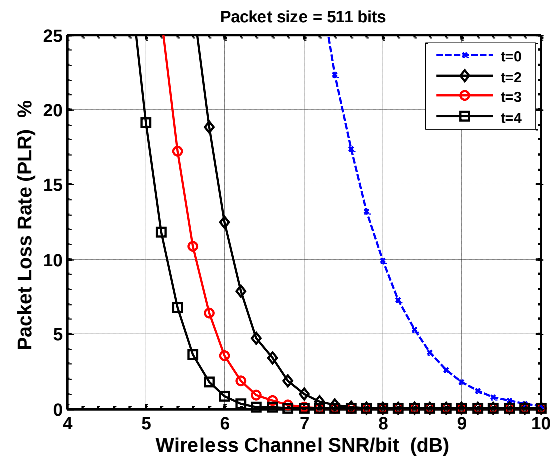
Table 1 describes a typical parameters setting used in the simulation for wireless network in GSM or CDMA systems including GoP pattern parameters for MPEG video stream [3-4]. A channel capacity is assumed at the limited bandwidth B_w, and upper-bound of the network throughput does not exceed B_w. The 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

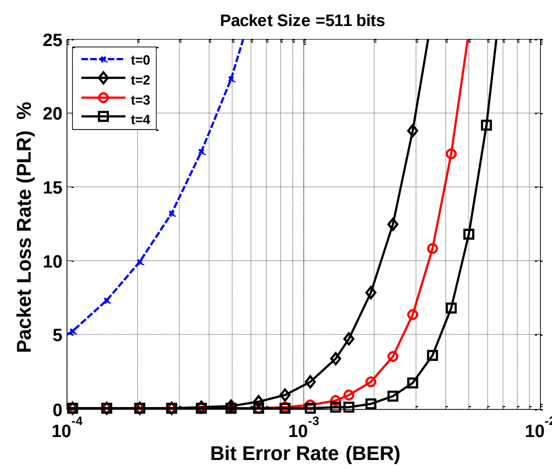
	Wireless network parameters
T_{RTT}	168 [ms]
k	1.2
B_w	1 Mbps
$L_{\max}$	511 bits for BCH code in radio-link layer
Modulation scheme	BPSK (upload/download)
γ_b	1 ... 10 [dB] Channel SNR/bit
	GoP Pattern parameters
F_o	30 [fps] reference frame rate at the video source.
S_I	25 [packets]
S_P	8 [packets]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

Table 2 reveals examples of random error-conditions used in the simulation. C1-C6 are channel states with errors ranging from 10-4 to10-2, 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, [3] introduces PSNR degradation no more



(a)



(b)

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

than 1.2 dB for the experimental H.236 “Foreman” video sequence and frame rate setting is 10 [fps].

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.

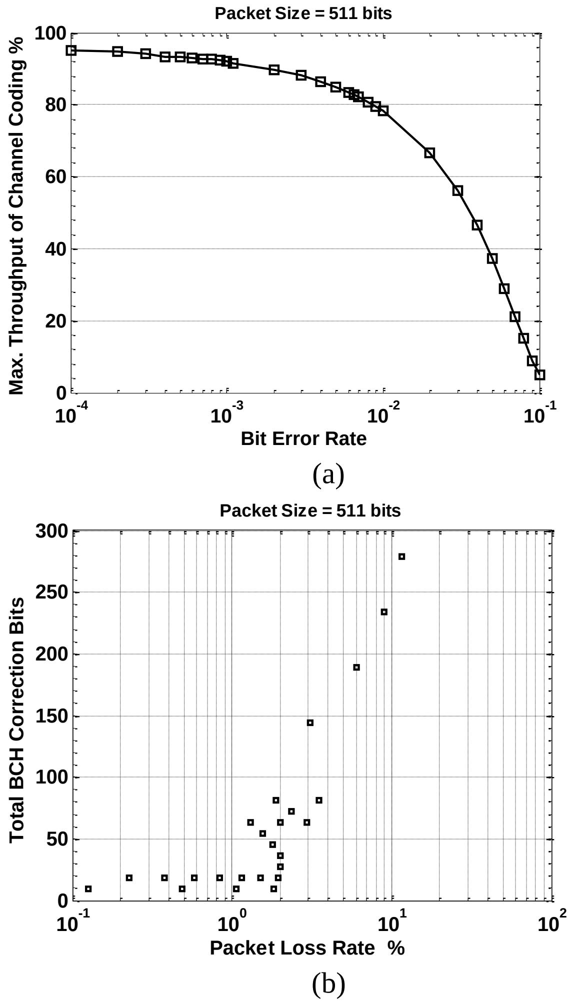


Figure 7: Channel Code Performance under various error conditions, (a) Maximum throughput [3], and (b) Total error-correction bits

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.

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.

As a result, 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.

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.

CONCLUSIONS

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, a QoS in terms of temporal scalability (frame per sec) at the client has been improved when a maximum channel coding throughput is achieved. The results demonstrate that a proposed scheme introduces a good predicted video quality at high channel bit-errors under various error-correction conditions as compared to other models [8-10] over wired and wireless Internet.

REFERENCES

[1].Y. Pei and J. W. Modestino, "Interactive Video Coding and Transmission over Heterogeneous Wired-to-Wireless IP Networks using an Edge Proxy," EURASIP J. on App. Sig. Proc., pp. 253-264, 2004.

[2].W. S. Lee, M. R. Pickering, M. R. Frater, and John F. Arnold, "A Robust Codec for Transmission of Very

Low Bit-Rate Video over Channels with Bursty Errors," IEEE Trans. on Circ. and Sys. for Video Tech. , vol. 10, pp. 1403-1412, Dec. 2000.

[3].T. Wang H. Fang, and L. Chen, "Low-Delay and Error-Robust Wireless Video Transmission for Video Communications", IEEE Trans. on Circ. and Sys. for Video Tech., vol. 12, no.12, pp. 1049-1058, Dec. 2002.

[4].M. Chen and A. Zakhor, "Multiple TFRC Connections Based Rate Control Wireless Networks", IEEE Trans. On Multimedia, vol. 8, no. 5, Oct. 2006.

[5].M. Claypool and Y. Zhu, "Using Interleaving to Ameliorate the Effects of Packet Loss in a Video Stream," Dept. of Computer Science, USA, May 2003.

[6]. Y. Wang and Q.-F. Zhu, "Error Control and Concealment for Video Communications: A Review," Proc. IEEE, vol. 86, pp. 974-997, May 1998.

[7].D. Krishnaswamy and M. Schar, "Adaptive modulated Scalable video Transmission over Wireless Networks with Game Theoretic Approach," IEEE Multimedia Signal Process. Workshop, MMSP'04, 2004.

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

[9].G. A. AL-Suhail and L. Tan, "Improving the QoS of Wireless Video Transmission via Packet-level FEC," Proc. of Inter. Conf. on Signal Processing and Communication Systems (ICSPCS'2007), Gold Coast, Australia, pp. 530-536, 17-19 Dec 2007.

[10]. Y. Yuan, "A GoP Based FEC Technique for Packet Based Video Streaming," Proc. of the 10th WSEAS Inter. Conference on Comm., Greece, pp. 187-192, July 10-12, 2006.

[11].T. Lee, S. Chan, Q. Zhang, W. Zhu, and Y. Zhang, "Optimal Allocation of Packet-Level and Byte-Level FEC in Video Multicasting over Wired and Wireless Networks," GLOBECOM'01, IEEE, vol. 3, pp.1994-1998, 2001.

[12].E. Steinbach, N. Faber, and B. Girod, "Standard Compatible Extension of H.263 for Robust Video Transmission in Mobile Environments," IEEE Trans. Circ. Sys. Video Tech.., vol. 7, pp. 872-881, Dec. 1997.

[13].S. Lin, Error Control Coding: Fundamentals and Applications. Prentice-Hall, 1983.

[14].J. B. Cain and D. N. McGregor, "A Recommended Error Control Architecture for ATM Networks with Wireless Links," IEEE J. on Selec. Area in Comm., vol. 15, no. 1, Jan 1997.

[15].W. Kumwilaisak, J. W. Kim, and C.-C., Kuo, "Reliable Video Transmission over Fading Channels via Channel State Estimation," SPIE Image and Visual Comm, Process., 2000.

[16].M. Mathis, J.Semke, J. Mahdavi, and T. Ott, "The Macroscopic Behavior of the TCP Congestion Avoidance Algorithm," ACM Computer Comm. Rev., no. 3, July 1997.

[17].R. Shinkkuma, M. Okada, and S. Komaki, "A New Resource Management Scheme for Multiple Video Transmission in Wireless Environment," IECE Trans. Comm., vol. E85-B, no. 10, pp. 2153-2160, Oct. 2002.

[18]. X. Tong, W. Gao, Q. Huang, "A Novel Rate Control Scheme for Video Streaming over Wireless Networks," Proc. of the 3rd Inter. Conf. on Image and Graphs (ICIG'04), IEEE Computer Society, 2004.

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

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**

- Upper-bound bandwidth (network throughput) achievable is 80.11 kbps
- Fixed RTT=168 ms, GOP(2,3)

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 [8] 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 [9] 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**

إرسال الفيديو اللاسلكي الكفوء بأستخدام تصحيح الخطأ المتقدم في طبقة التوصيل الراديوية في الاتصالات المرئية

غيداء عبدالرزاق السهيل

Email: gaida_alsuhail@yahoo.com

الخـلاصة

في هذا البحث، يتم تقديم تحليل ودراسة لأداء نموذج جديد قوي مُضاد للأخطاء من اجل إرسال اشارة فيديو (MPEG-4)– عبر شبكة لاسلكية (point-to-point). التعابير التحليلية تفترض بيئة لاسلكية مشوشة تسبب أخطاء بت عشوائية متكررة مصاحبة للرزم. وفي هذا النموذج يتم تقييم المقياس الزمني لجودة الاشارة المرئية تحت بروتوكول التحكم بمعدل النقل TCP-Friendly عندما يتم تشفير القناة بتقنية BCH لتصحيح الاخطاء المتقدمة في طبقة التوصيل الراديوية. ان تقنية FEC توفرولوج كفوء لسعة الكفاءة في الشبكة اللاسلكية. النتائج المستحصلة تبيّن بوضوح ان جودة الخدمة يمكن ان تتحسن عندما يتم الوصول الى القيم القصوى لكفاءة تشفير القناة في منطقة اشارات القناة الضعيفة.