Estimating MPEG-4 Video Transmission over Gilbert Wireless Channels

Ghaida. A. AL-Suhail ***Sarah A. Subber***
Computer Engineering Department, University of Basrah, Basrah, Iraq
0780 60 12 354 ***0770 277 26 40***
ghaida_alsuhail@yahoo.com **sarah_subberidsco@hotmail.com**

Abstract

In this paper, the aim is to propose an analytical packet loss models to estimate the video quality for MPEG-4 video transmission over error-prone correlated wireless channels. Two packet loss models are considered at the packet level in the network layer using TCP-Friendly protocol: (i) Gilbert-Elliott Model (GEM) and (ii) Extended Gilbert Model (EGM). The resultant video quality in terms of playable frame rate (PFR) is estimated at the end client over wireless cellular network. The obtained results point out the effect of packet correlation factor on the perceived video quality under different conditions of packet loss probability. It is found that the proposed EGM model outperforms in estimating the perceived video quality.

Keywords— Gilbert-Elliott, Extended Gilbert, MPEG-4 Video, TCP-Friendly, Wireless Network

Introduction

The IP packet-based wireless cellular systems, like Universal Mobile Telecommunications Systems (UMTS) and High-Speed Uplink/ Downlink Packet Access (HSUPA/HSDPA) technologies are recently developed for high data rate transmission [1][2][3]. However, a packet-loss process in such networks is still a crucial issue in estimating the probability of correctly received video packets at the clients. The real-time video transmission over the internet and wireless channel in heterogeneous packet networks is subject to various errors types. In wired networks (such as the Internet), packets are lost mainly due to buffer overflows (congestion) at the routers, meanwhile in the wireless hop, packets are often lost due to random bit errors caused by channel variations like noise, co-channel interference, mobility, and multi-path fading effects. Thus maintaining video quality in wireless environment is very challenging and it requires more strategic planning and robust video transmission which is adaptable to the network and channel conditions [4][5].

More specifically, understanding the packet-loss process in IP-based packet networks and providing an accurate mathematical model to describe it are of great importance for the design and performance analysis of network applications (real-time applications). One way of describing the packet-loss process is to concentrate on the observed packet loss by itself without considering explicitly the causes that lead to such process [5] [6]. Thus, Markov process (chain) based models are classified in this category. In k-th order Markov chain model characterizing a loss process, the loss of every packet is assumed to be dependent only on the loss status of the previous k packet transmissions. The simplest Markov chain model is the case k=0, known as the Bernoulli model. In this model, the packet loss probabilities are independent of each other [7][8]. Another simple Markov chain model is the case k-1, known as the Gilbert model [9], which has been widely used to model end-to-end packet loss processes for real-time applications [6]- [8]. The former model has also been extended to Gilbert-Elliott model [10]. Although these low-complexity Markov chain models have been frequently used to model such processes [7][8], the accuracy of these Markovian models in capturing the correlation characteristics of real-world packet-loss processes needs to be investigated. Sanneck et al. [11] develop a different Markov chain model, called the extended Gilbert model. There are two categories of extended Gilbert models; those which describe reception run-lengths (RRL) and those which describe loss run-lengths (LRL). In our approach, we concentrate on RRL extended Gilbert models which is derived by Wu and Radha [12] to be applied for MPEG-4 video packet loss process over wireless channel.

On the other hand, many researches [13]-[18] have devoted on video streaming based on TCP/UDP protocols in different wired/wireless IP-packet networks. Specifically, TCP has become popular protocol because of its easy handling and deployment. TCP features in order delivery and reliable end-to-end transport, which makes additional tools like error concealment, unnecessary at the clients. In low-latency networks, TCP introduces good throughput performance and low end-to-end delays, which makes TCP-based interactive services possible. In [18], for example, they propose a client-driven video transmission scheme by utilizing multiple HTTP/TCP streams. In [3], they evaluate video streaming based on TCP-Friendly Rate Control (TFRC) using Bernoulli packet loss model for a single and Multiple TFRC flows over a 1xRTT CDMA in UMTS network.

In this paper, we focus on video traffic controlled by TCP-Friendly protocol over cellular network for the advantages of this protocol: (i) it does not cause network instability and avoid congestion collapse. (ii) it is fair to TCP flows, which is the dominant source of traffic on the internet, and finally (iii) this protocol has lower fluctuation compared to TCP. That makes TCP more appropriate for Real time applications, which requires a constant video quality [3]. To our best knowledge, the studies [13]-[18] did not investigate the extended Gilbert model for TCP video transmission over wireless channel. Therefore, we propose an approach to introduce the two packets loss models: Gilbert-Elliot Model (GEM) [7]-[10] and Extended Gilbert Model (EGM) [12]. The models employ TCP-Friendly Rate Control for MPEG-4 video transmission, and then estimate the resultant video quality in terms of playable frame rate at the client end.

Network Preliminaries

MPEG Standard

In the last years, several video standards have been developed for 3G mobile multimedia communications like H.263, MPEG-4, and H.264/AVC [8]. MPEG video is one of the most commonly used video compression standard which is encoded into three different types of frames- I (Intra-coded), P (Predictive-coded), and B (Bi-directional Predictive coded) frames.

In this standard, I-frame is independently coded, while P-frame is coded based on the prediction of object movement of the previous I or P frame. The B-frame is coded based on the differences between the previous I or P frame and the next I or P frame. Thus, there is a certain dependency relationship between I, P, B frames as shown in Fig. 1 [13][17].

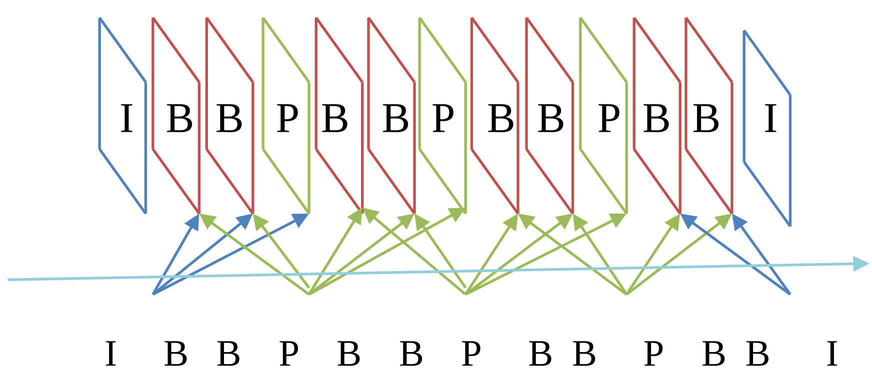

Figure1. Frame dependency

TCP-Friendly Throughput

We consider a video flow in a point-to-point network which is simply composed of one base station (access point) in UMTS and a single user end. This last wireless link is connected to a wired Internet via this base station [3][14]. By adjusting the sending rate to the desirable rate determined by an underlying TCP-Friendly Rate Control (TFRC), one can achieve the required quality of service (QoS) of video applications over a wireless link. Thus the normalised available bandwidth of a TFRC video session with respect to TCP packet size can be expressed as,

$$T = \frac{S}{t_{RTT}\sqrt{\frac{2P}{3}} + t_{RTO}(3\sqrt{\frac{3P}{8}})P(1+32)} \qquad (1)$$

S is the TCP packet size [byte], P stands for the average packet loss probability, i.e., loss event rate due to only the channel bit errors and there is no buffer overflow effect at the base station. The t_{RTT} is the round-trip time [sec], and is the t_{RTO} TCP retransmit time out value [sec].

GOP Rate

Since (1) provides an expression for T as TFRC throughput, then GOP rate can be written as;

$$G = \frac{T/S}{(S_I + S_{IF}) + N_P(S_P + S_{PF}) + N_B(S_B + S_{BF})} \qquad (2)$$

where is the size of I-frame, is the size of P-frame, is the size of B-frame, represents the number of P-frame in a GOP, represent the number of B-frame in a GOP [13].

Playable Frame Rate

The total PFR by Wu et al.'s, technique is given by [13,14];

$$PFR = G.P_{SI}.(1 + \frac{P_{SP} - P_{SP}^{N_P+1}}{1-P_{SP}} + N_{BP}.P_{SB}.(\frac{P_{SP} - P_{SP}^{N_P+1}}{1-P_{SP}} + P_{SI}.P_{SP}^{N_P})) \qquad (3)$$

Where P_{SI}, P_{SP} and P_{SB} are the probabilities of success transmission of I, P, or B frame. The detail of the derivation of this equation is outlined in.

The Proposed Approach

GEM-TFRC Video Model

Recently many studies used the measurements of burst error over a wireless channel by the well known Gilbert-Elliot model (GEM) [7][10]. In the present work, we further assume GEM as virtual channel with two nodes, to improve the video quality by estimating the playable frame rate. The state diagram of the model is shown in Figure 2, the model represents two states; the good state (good packet P_0) and the bad state (bad packet P_1).

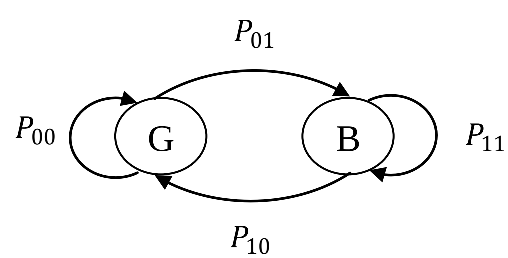

Figure 2. Gilbert Elliot state diagram for packet level

The P_{01} is the state probability for the transition from good to bad, and P_{10} is the state probability from the transition from bad to good .The π_1 and π_0 study state probability being in state P_1 and P_0 and $\pi_1 = \frac{P_{01}}{P_{01}+P_{10}}$, $\pi_0 = \frac{P_{01}}{P_{01}+P_{01}}$ respectively . The average packet loss rate product by the GE model is [7]-[10];

$$P_{avg} = P_0\pi_0 + P_1\pi_1 \quad (4)$$

The GEM is memory less model, where Packet error is produced by a sequence of independent trial. Each packet has P_{avg} being flipped and $1-P_{avg}$ being successfully transmitted, P_{avg} is then the packet loss probability for the wireless channel. The network throughput becomes;

$$T = \frac{S}{t_{RTT}\sqrt{\frac{2P_{avg}}{3}} + t_{RTO}(3\sqrt{\frac{3P_{avg}}{3}})P_{avg}(1+32)} \quad (5)$$

Giving N packet being transmitted over a Gilbert Elliott channel, with K packet received, the probability of successful transmission is;

$$P_{suc} = \sum_{i=K}^{N} \binom{N}{i} (1-P_{avg})^i P_{avg}{}^{N-i} \quad (6)$$

With a specific I, P, and B frame sizes, Equation (5) can be used to compute the probabilities of successful transmission for each frame type as following;

$$GEMP_{S_I} = \sum_{i=K}^{S_I} \binom{S_I}{i} (1-P_{avg})^i P_{avg}{}^{S_I - i} \quad (7)$$

$$GEMP_{S_P} = \sum_{i=K}^{S_P} \binom{S_P}{i} (1-P_{avg})^i P_{avg}{}^{S_P - i} \quad (8)$$

$$GEMP_{S_B} = \sum_{i=K}^{S_B} \binom{S_B}{i} (1-P_{avg})^i P_{avg}{}^{S_B - i} \quad (9)$$

The total PFR using GEM-TFRC is expressed as

$$PFR_{GEM-TFRC} = G * GEMP_{SI} .(1+\frac{GEMP_{SP} - GEMP_{SP}{}^{N_P+1}}{1-GEMP_{SP}} + N_{BP} * GEMP_{SB} .(\frac{GEMP_{SP} - GEMP_{SP}{}^{N_P+1}}{1-GEMP_{SP}} + GEMP_{SI} * GEMP_{SP}{}^{N_P})) \quad (10)$$

EGM-TFRC Video Model

The Second proposed model for packet loss probability is a model which was derived by Wu and Radha [12] have been used, where the authors extends the two-state Gilbert model using the number of correctly received packets as a indexes for Gilbert states.

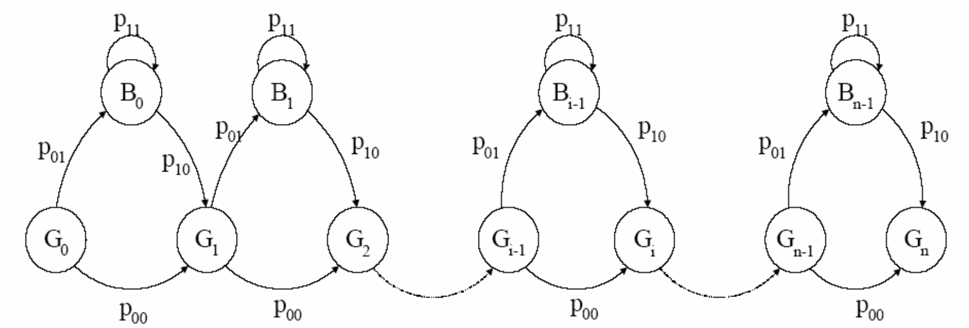

Figure 3. The state transition diagram of Extended Gilbert Model (EGM) [12].

The Extended Gilbert model is build on the probability of receiving correctly K packets among N packets transmitted over Gilbert is giving by [12];

$$\phi(N,K) = \pi_0(\phi_{G_0G_K}(N) + \phi_{G_0B_K}(N)) + \pi_1(\phi_{B_0G_K}(N) + \phi_{B_0B_K}(N)) \tag{11}$$

Where

$$\phi_{G_0G_K}(N) = \begin{cases} \sum_{m=1}^{K} \binom{K}{m}\binom{N-K-1}{m-1} P_{01}{}^{m} P_{10}{}^{m} P_{00}{}^{K-m} P_{11}{}^{N-K-m} & 0 < K < N \\ 0 & K=0 \\ P_{00}{}^{N} & K=N \end{cases} \tag{12}$$

$$\phi_{G_0B_K}(N) = \begin{cases} \sum_{m=0}^{K} \binom{K}{m}\binom{N-K-1}{m-1} P_{01}{}^{m+1} P_{10}{}^{m} P_{00}{}^{K-m} P_{11}{}^{N-K-m-1} & 0 < K < N \\ 0 & K=N \end{cases} \tag{13}$$

$$\phi_{B_0G_K}(N) = \begin{cases} \sum_{m=1}^{K-1} \binom{K-1}{m}\binom{N-K}{m} P_{01}{}^{m} P_{10}{}^{m+1} P_{00}{}^{K-m-1} P_{11}{}^{N-K-m-1} & 0 < K < N \\ 0 & K=0 \end{cases} \tag{14}$$

$$\phi_{B_0 G_K}(N) =$$

$$\begin{cases} \sum_{m=0}^{K-1} \binom{K-1}{m} \binom{N-K}{m+1} P_{01}{}^{m+1} P_{10}{}^{m+1} P_{00}{}^{K-m-1} P_{11}{}^{N-K-m-1} & 0 < K < N \\ P_{11}{}^{N} & K= \\ 0 & K= \end{cases} \quad (15)$$

The state probability P_{01} and P_{01}(or $P_{00} = 1 - P_{01}$ and $P_{11} = 1 - P_{10}$) are the effective parameters on the evaluation of $\phi(N, K)$. More useful parameters can be involved to improve the performance. In [12], the author used the average loss rate P, and the packet correlation ρ, to give another perception to the state transition probabilities, where packet correlation is the correlation between two errors symbols P_{00} and P_{11}, which can be expressed as;

$$P_{01} = P(1-\rho) \quad (16)$$

$$P_{10} = (1-P)(1-\rho) \quad (17)$$

The computation of the successful transmission probabilities for a specific I, P, and B frame sizes can be written as;

$$EGMP_{S_I} = \phi(S_I, K) \quad (18)$$

$$EGMP_{S_P} = \phi(S_P, K) \quad (19)$$

$$EGMP_{S_B} = \phi(S_B, K) \quad (20)$$

Then the total PFR using EGM-TFRC can be written as;

$$PFR_{EGM-TFRC} = G.EGMP_{SI}.(1 + \frac{EGMP_{SP} - EGMP_{SP}{}^{N_P+1}}{1 - EGMP_{SP}} + N_{BP}.EGMP_{SB}.$$

$$(\frac{EGMP_{SP} - EGMP_{SP}{}^{N_P+1}}{1 - EGMP_{SP}} + EGMP_{SI}.EGMP_{SP}{}^{N_P})) \quad (21)$$

Simulation Results

Methodology

Based on the above assumption, we develop the following illustrative steps to find the optimal playable frame rate based on the given models GEM-TFRC video model, and EGM-TFRC video model) mentioned in Section II.

1. Giving a specified state probability one can determine the from equation 3.For each set of other system variables, compute the PFR using equations 4 and 12.For EGM-TFRC, assign specific average loss rate *p, and* packet correlation, determine PFR using (4) and (21).We characterize the variation of the PFR in both Models in term of two dimensions; Packet loss Probability and PFR dimension, to fined the optimal PFR.
2. Used the four different cases of transmitted and received packet with a certain probability of packet loss, to found the optimal PFR.

3. Giving a combination of packet correlation, and a set of an average loss probability, compute the PFR using (21). Characterize the variation of the PFR in term of two dimensions: packet correlation and PFR. To investigate the effect of packet correlation on the PFR, in four different cases transmitted and received packet.

System Settings

Table 1 describes network characteristic of many typical network connection.[2] in which we used in our simulation. Encoded video is transmitted as a repeating sequence of GOP (N_P, N_{BP}) pattern, where N_P =3 and N_{BP} =2. The P_{00}, P_{11} and P_0 are set at 0.96, 0.94 and 0.001 respectively. The packet error rate is set between 0.001 to 0.1 with 0.001 intervals. Also, the packet correlation is set between 0.1 to 0.9 with 0.1 intervals.[13][12][17]

Table 1: Network Settings [3][13][12]

Parameter	Value
t_{RTT}	168ms
t_{RTO}	$t_{RTT} \times 4$
S_I	24.64KB ~ 25 packet
S_P	7.25KB ~ 8 packet
S_B	2.45KB ~ 3 packet

Table 2 MPEG Sending, Receiving Frames Size

	Sending	Receiving
	(N_I , N_P, N_B)	**(K_I , K_P, K_B)**
1	(25,8,3)	(25,8,3)
2	(25,8,3)	(20,6,1)
3	(25,8,3)	(25,6,1)
4	(25,8,3)	(24,6,1)

Performance Evaluation

This section describe the PFR of an MPEG4 Video streaming over wireless channel using two packets lost models; the GEM-TFRC video model and EGM-TFRC video model. The performance evaluation is carried out by considering the parameters in Tables 1, 2 and 3 for the fact that maximum frame rate allowed over internet is 30fps. Result was conducted using Matlab programming.

Figure 4 shows the PFR over the effect of packet error rate, using proposed models. The PFR clearly indicates better performance in the range from 0.02 to 0.1 packet loss probability. Meanwhile, the GEM-TFRC becomes closer to the EGM-TFRC at packet loss probabilities below 0.02. However, the special case of the GEM-TFRC for which the correlation parameter had been added, gives us a poor performance in comparison with the other.

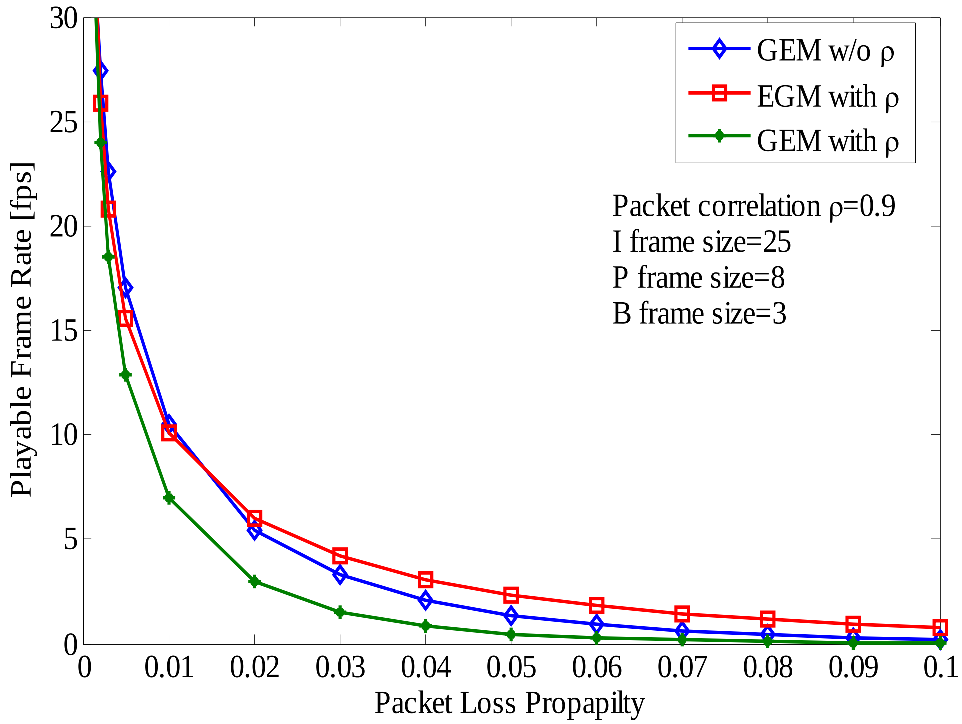

Figure 4 The Playable Frame Rate comparison using GEM-TFRC without memory, GEM-TFRC with memory, and EGM-TFRC models

Table 3 summaries different sending and receiving packets that are applied in the two models for video streaming. It is clearly noticed that there are two effects as follows: dependency relationship between frames in GOP, and packet correlation factor. The PFR achieves the highest value at P=1% and in contrary there is a clear degradation in video quality when packet loss increases to 5%. However, EGM-TFRC model introduces nonzero values of PFR when sending (25,8,3) and receiving (20,6,1); and this result obtained is caused due to the packet correlation factor.

Figure 5 shows the PFR over the effect of correlation factor in EGM-TFRC, when sending (NI=25, NP=8, NB=3) and receiving (KI=25, KP=8, KB=3) packets, when the packet correlation is low, it will produce low PFR; but this is not always the case when the `correlation increase. In Fig. 5 we consider a case where a packet loss occur, where sending packet sizes are (NI=25, NP=8, NB=3), receiving packets (KI=20, KP=6, KB=1), as shown in Fig. 6 it is clearly shown that when *P* set to1% we have got higher PFR at $\boldsymbol{\rho}$ =0. The fluctuation in PFR values is also clearly noticed when $\boldsymbol{\rho}$ increases to 0.9. The PFR has approximately constant value over a combination of *P*; as the correlation $\boldsymbol{\rho}$ is strong. Once the process initially starts in bad or good state, it has the inertia to stay at that state.

Table 3: The MPEG4 sending and receiving Frame sizes and PFR for *P*=1% and *P*=5%.

GEM-TFRC			
Sending (N_I, N_p, N_B)	Receiving (K_I, K_p, K_B)	PFR P=1%	PFR P=5%
(25,8,3)	(25,8,3)	10.5119	1.37873
(25,8,3)	(20,6,1)	0.00	0.0005
(25,8,3)	(25,6,1)	1.4404	0.3022
(25,8,3)	(24,6,1)	0.0044	0.0062
EGM-TFRC			
Sending (N_I, N_p, N_B)	Receiving (K_I, K_p, K_B)	PFR P=1%	PFR P=5%
(25,8,3)	(25,8,3)	10.0929	1.3787
(25,8,3)	(20,6,1)	0.0028	0.0042
(25,8,3)	(25,6,1)	1.2677	0.3022
(25,8,3)	(24,6,1)	0.2366	0.2416

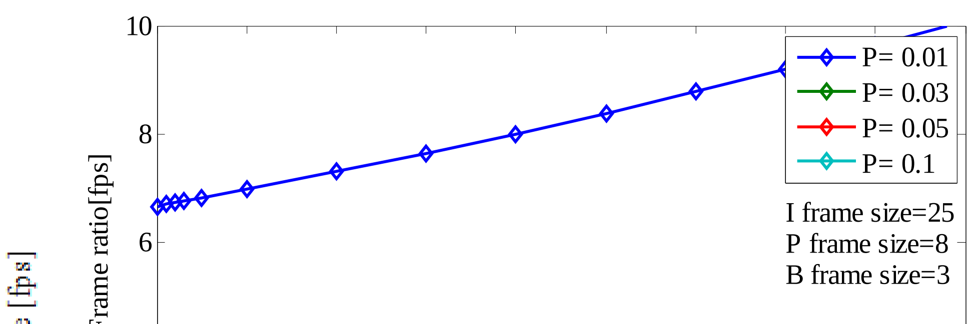

Figure 5. The playable frame rate over the effect of correlation factor in EGM-TFRC video model when sending (NI=25, NP=8, NB=3) and receiving (KI=25, KP=8, KB=3)

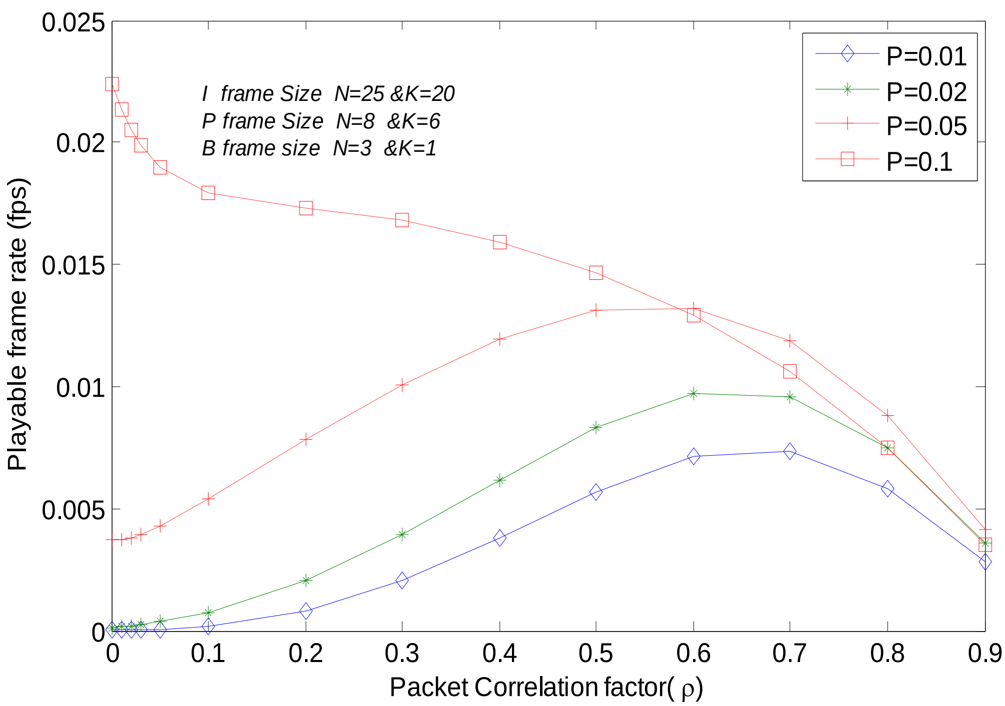

Figure 4.b The playable frame rate over the effect of correlation factor in EGM-TFRC video model when sending (NI=25, NP=8, NB=3) and receiving (KI=20 ,KP=6 ,KB=1)

Conclusion

In this paper, we have proposed analytic packet loss models to estimate video quality over wireless channels with highly correlated errors using TFRC protocol for MPEG-4 streaming. The models are based on Gilbert Elliot and Extended Gilbert model of packet loss probability. The results of GEM-TFRC video model show good performance in high packet loss probability; meanwhile GEM and EGM both show reasonable performance as well but in the range of average packet loss rate below 2%. As a result, we conclude that these proposed models can be helpful to predict the video quality using packet loss models over wireless channels. The Future work can be extended to involve the FEC techniques the two to improve the performance of video quality under different conditions of wireless channels.

References

[1] H. Holma and A. Toskala(2006), HSDPA /HSUPA for UMTS High Speed Radio Access for Mobile Communications, John Wiley, England.

[2] Q. Liu, S. Zhou, and G. B. Giannahis(2005), "Queuing with Adaptive Modulation and Coding over Wireless Links: Cross-Layer Analysis and Design," IEEE Trans. Wireless Comm., vol. 4, no. 3.

[3] M. Chen and A. Zakhor(2005), "Rate Control for Streaming Video over Wireless", IEEE Wireless Communication, pp. 32-41.

[4] A. Khan, L. Sun, E. Ifeachor, J.O. Fajardo, F. Liberal (2010), "Video Quality Prediction Model For H.264 Video over UMTS Networks and their Application in Mobile Video Streaming", IEEE Communication (ICC).

[5] H. Han(2010), "Performance Improvement of TCP Reno Based on Monitoring the Wireless Packet Loss Rate", Physics & Electronics Engineering Department, Xiangfan University, China.

[6] X. Yu, W. Modestino and X. Tian(2005), "The Accuracy of Gilberts Models in Predicting Packet-Loss Statistics for Single-Multiplexer Network Model", IEEE International Conference.

[7] G. Haßlinger and O. Hohlfeld (2008), "The Gilbert-Elliott Model for Packet Loss in Real Time Services on the Internet", Darmstadt University of Technology, Germany.

[8] Y. Baguda, N. Fisal, S. Syed, S. Yusof, M. Abdullah, and A. Zulkarmawan(2008)" Mobile Streaming of H.264 Video over Gilbert-Elliotts Channel ", World Academy of Science, Engineering an Technology 48.

[9] E. N. Gilbert(1960), "Capacity of a Burst Noise Channel", Bell Sys. Tech. Journal, vol. 39, p. 1253.

[10] E. O. Elliott (1963) "Estimates of Error Rates for Codes on Burst-Noise Channels," Bell System Technical Journal, vol. 42, pp. 1977-1997.

[11] H. Sanneck, G. Carle, and R. Koodli(2000) " A Framework Model for Packet Loss Metrices Based on Loss Run Length," Proceedings of SPIE/ACM SIGMM Multimedia Computing and Networking Conference.

[12] M. Wu and H. Radha(2005) "Network Embedded FEC for Overlay and P2P Multicast

over Channels with Memory," Conf. on Information Sciences and Systems, The John Hopkins University,.

[13] H. Wu, M. Claypool, R. Kinicki (2003) "A Model for MPEG with Forward Error Correction and TCP-Friendly Bandwidth ", In Proceedings of Workshop on Network and Operating Systems Support for Digital Audio and Video (NOSSDAV),Monterey, California, USA.

[14] Y. Yuan, B. Cokburn, T. Sikora, and M. Mandal(2006) "A GoP Based FEC Technique for Packet Based Video Streaming", Proceedings of the 10th WSEAS International Conf. on Communications, Greece.

[15] Y. Shan, S. Yi, S. kalyanaraman, and J. Woods(2009) "Two-stage FEC Scheme for Scalable Video Transmission over wireless Network "Image comm.24 (9):718-729.

[16] J. R. Yee, E. J. Weldon(1995) "Evaluation of the Performance of Error-Correcting Codes on a Gilbert Channel", IEEE Trans. on Communications, vol. 43, no. 8, pp. 2316-2323.

[17] C. H. Lin, C. H. Ke, C. K. Shieh(2004)"A Study of MPEG Video Transmission in Lossy Wireless Networks", Electrical Engineering Department, National Cheng Kung University, Taiwan.

[18] R. Kusching, I. Kofler, and H. Hellwanger(2010) "Improving Internet Video Streaming Performance by Parallel TCP-based Request Response Streams", Proc. of IEEE CCNC.

تقييم إرسال الصورة المرئية نوع MPEG-4 في قنوات جلبرت اللاسلكية

غيداء عبد الرزاق السهيل ***سارة عبد الرضا صبر***

قسم هندسة الحاسبات كلية الهندسة، جامعة البصرة

الخلاصة

في هذا البحث، الهدف هو اقتراح نماذج تحليلية لفقد الرزمة لغرض مقايسة جودة الصورة المرئية من نوعMPEG4 المٌرسلة عبر القنوات اللاسلكية ذات الاخطاء المترابطة .تم اقتراح نموذجين من فقدالرزمة عند مستوى الرزمة في طبقة الشبكة باستخدام بروتوكول النقلTCP-Friendly; نموذج جلبرت - اليوت (2) نموذج جلبرت الموسّع. يتم تقييم جودة الصورة الناتجة بدلالة معدل الفريم العامل عند نهاية المستخدم في الشبكة الخلوية اللاسلكية .النتائج المستحصلة تبين تاثير معامل ترابط الرزمة على جودة الصورة المرئية المحسوسة تحت ظروف مختلفة من احتمالية فقد الرزمة .وقد وجد ايضاً ان نموذج جلبرت الموسّع قد تفوق في مقايسة جودة الصورة المستلمة.