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ROBUST NETWORK FEC-EMBEDDED VIDEO TRANSMISSION OVER ERROR-PRONE WIRELESS CHANNELS

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ABSTRACT

Most wireless networks incline to include FEC code (Forward error correction) to recover the corrupted packets without retransmission. The sequential packets transmitted over wireless channel are likely to be infected with various errors like noise or deep fading. In this paper, therefore, a frame-level FEC is added across packets at the application layer to define two analytical packet loss models. At the network layer level, Gilbert Elliot Model (GEM) and Extended Gilbert Model (EGM) are considered to characterize the MPEG-4 packets loss over the bursty error-prone wireless channels. The resultant video quality in terms of playable frame rate (PFR) is estimated at the client end when using the underlying TCP-Friendly protocol. Numerical results point out that the EGM-FEC model introduces a robust measure in estimating the perceived video quality in particular at higher packet loss rate and lower effect of packet correlation. Moreover, it is also found that the GEM-FEC model outperforms EGM-FEC but when ignoring the effect of packet correlation.

KEYWORDS

FEC, Gilbert Model, Extended Gilbert Model, Video Streaming & Wireless Network

1. INTRODUCTION

Recently, the IP wireless networks, like Universal Mobile Telecommunications Systems (UMTS), High-Speed Uplink/ Downlink Packet Access (HSUPA/HSDPA) and WiMAX technologies are developed for high data rate transmission [1]-[4]. Among these, wireless environment is unreliable, acknowledges the fact that all wireless real-time video transmission introduce distortions into transmitted streaming packets due to channel fading, multi-path, noise, congestion and interference which may cause to transmit useless packets.

Specifically, a packet-loss in such networks is still a crucial issue in estimating the probability of correctly received video packets at the clients. 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, mobility, and multi-path fading effects. It is found that a small amount of network error / packet loss can have a dramatic impact on the playable frames at client side. Thus robust video transmission in wireless environment is very challenging and it requires more robust schemes to adapt the variations of network and channel conditions [5]-[7].

In fact, the packet-loss process in IP networks needs an accurate mathematical model to describe it for the design and performance analysis of network applications (e.g., real-time applications). One way is to concentrate on the observed packet loss by itself without considering explicitly the causes that lead to such process [8] [9]. Thus, Markov process (chain)

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based models are classified to describe packet loss. 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 [10].

Another simple Markov chain model is the case *k*-1, known as the Gilbert model [11], which has been widely used to model end-to-end packet loss processes for real-time applications [7]-[9]. The former model has also been extended to Gilbert-Elliott model [11]. Although these low-complexity Markov chain models have been frequently used to model such processes [9][11], the accuracy of these Markovian models in capturing the correlation characteristics of real-world packet-loss processes needs to be investigated. For example, Sanneck *et al.* [12] 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 [13] to be applied for MPEG-4 video packet loss process over wireless channel.

Moreover, error-control mechanisms like Automatic Repeat reQuest (ARQ), and Forward Error Correction (FEC) [14]-[15] are also widely used to improve the throughput of a wireless network and eventually compensate the network error (i.e., packet loss). Moreover, since the final performance (e.g. optimal achievable throughput) at the transport level which consequently introduces the optimal play-out rate at client end, depends on the FEC code at video server based on the feedback of radio link layer for wireless client, the consistent TCP model is used to capture interactions and simplify the design complexity. Basically there are advantages and disadvantages for each error-control mechanism. ARQ is simple and needs less bandwidth, where sender retransmits the erroneous packet. However, the retransmission incurs large round trip time (RTT) delay to recover lost packets. FEC is more complexity and incurs less RTT where the erroneous packets are corrected with the help of redundant packets. However, this scheme will incur a bandwidth overhead [16][17].

The studies [18]-[21] which use adaptive packet –level FEC code to improve video quality mostly assume a simple Bernoulli model to describe the packet loss effects. Therefore, in this paper, we investigate two packets loss models: Gilbert-Elliot Model (GEM) and Extended Gilbert Model (EGM) using frame-level FEC at the application layer to correct packet loss due to wireless environment conditions. The models employ the underlying TCP-Friendly Rate Control for MPEG-4 video transmission. As a result, the video quality can be estimated in terms of playable frame rate at the client end under various packet-level FEC codes at the application layer. The results obtained introduce a good comparative study for robust network FEC – embedded video transmission over a highly correlated packet loss over wireless channel.

The rest of this paper is organized as follows. Section 2 describes several related works on video quality over wireless networks. Section 3 provides background to the work. Our proposed analytical models of MPEG video transmission are presented in Section 4. Results and performance comparison are addressed in Section 5. Finally, Section 6 summarizes the conclusion and outlines some future works

2. RELATED WORK

To carry out a high video quality, most wireless networks incline to include FEC code to recover the corrupt packets and avoid retransmissions. Video streaming have strict constrains on delay latency and delay jitter. This makes FEC as an optimal choice for many real-time applications. Many studies therefore have devoted to improve the video quality performance using different strategies. One solution may tackle the problems of how modifying the TCP model [22][23]; and the other is how improving the link reliability observed by TCP [5][6][24].

The later solution mainly includes a cross-layer design of hybrid schemes to provide a required quality link reliability over wireless links. There exist many recent cross-layer approaches, for example: adaptive rate control [6], adaptive selective Repeat (ASR) protocols (re-transmission) [24], finite-length queuing at the data link layer coupled with adaptive modulation and coding (AMC) at the physical layer [5], optimal time slot for optimal resource allocation [25], and finally adaptive FEC at packet/byte/bit level (*e.g*., BCH, RCPC and RS codes) [2][5][21].

On the other hand, many researches [18]-[28] 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 [28], for example, they propose a client-driven video transmission scheme by utilizing multiple HTTP/TCP streams. In [6], they evaluate video streaming based on TCP-Friendly Rate Control (TFRC) using Bernoulli packet loss model for a single and Multiple TFRC flows in UMTS network.

3. Network Preliminaries

3.1. MPEG

In the last years, several video standards have been developed for 3G mobile multimedia communications like H.263, MPEG-4, and H.264/AVC [9]. 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 Figure 1 [11][18].

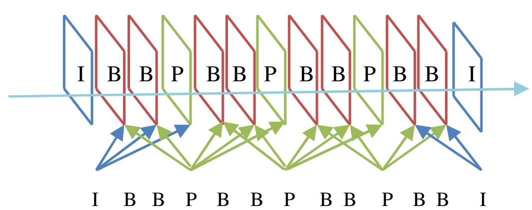


Figure 1.The structure of a GOP and the inter-frame dependency relationship

3.2. Forward Error Correction (FEC)

In a lossy channel packets are dropped due to routing disruption, interference, and congestion in the intermediate nodes [4], therefore to protect video packets from losses in a wireless environment, FEC blocks are applied at the application–layer. In [18], Wu *et al.'s* proposed an analytical model which is known as the frame-level FEC technique. The FEC packets are generated based on individual frames (I, P, or B) as shown in Figure 2.

Thus, when K packets are transmitted with (N-K) redundant FEC packets with packet loss probability P_{avg} , the successfully probability after encoding is defined by a binomial trial;

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

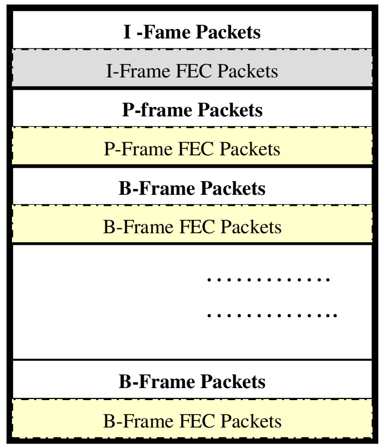


Figure 2.Arrangement of source and FEC Packets in frame-level FEC technique.

3.3 TCP-Friendly Throughput

We focus on the video traffic controlled by TCP-Friendly protocol over wireless network because of some protocol advantages. It does not cause network instability and avoids congestion collapse. It is fair to TCP flows, which is the dominant source of traffic on the internet, and finally this protocol has lower fluctuation compared to TCP. That makes TCP-Friendly more appropriate for real time applications, which requires a constant video quality [3].

In this paper, 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][6]. 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,

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

where S is the TCP packet size [byte], P_{avg} 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. t_{RTT} is the round-trip time [sec], and t_{RTO} is the TCP retransmit time out value [sec].

3.4 GOP Rate

As T is TFRC throughput, the GOP rate can be written as;

$$G = \frac{T/S}{(S_I + S_{IF}) + N_P(S_P + S_{PF}) + N_B(S_B + S_{BF})} \quad (3)$$

N_P represents the number of *P* frame in a GOP and N_B is the number of *B* frame in a GOP.

3.5 Playable Frame Rate (PFR)

The total PFR by Wu *et al.'s*, technique is given by [18];

$$PFR = G.P_{SI}.(1 + \frac{P_{SP} - P_{SP}^{N_P+1}}{1 - P_{SP}} + N_{BP}.P_{SB}.(\frac{P_{SP} - P_{SP}^{N_P+1}}{1 - P_{SP}} + P_{SI}.P_{SP}^{N_P})) \quad (4)$$

Where P_{SI}, P_{SP} and P_{SB} are the probabilities of success transmission of I, P, or B frame.

4. THE PROPOSED MODEL

4.1 GEM-FEC Video Model

Recently many studies used the measurements of burst error over a wireless channel by the well known Gilbert Elliot model (GEM) [8][13]. 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 3, the model represents two states; the good state (good packet P_0) and the bad state (bad packet P_1).

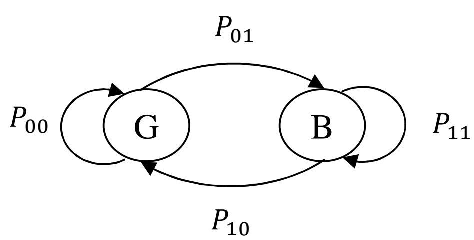


Figure 3.Gilbert Elliot state diagram for packet level

The P_{01} is the state probability for transition form good to bad, and P_{10} is the state probability from transition from bad to good .The π_1 and π_0 study state probability being in state P_0 and P_1 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 defined as

$$P_{avg} = P_0\pi_0 + P_1\pi_1 \quad (5)$$

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.

With a specific I, P, and B frame sizes and number of FEC blocks, (1) can be used to compute the probabilities of successful transmission for each frame type as following;

$$GEMP_{S_I} = \sum_{i=S_I}^{S_I+S_{IF}} \binom{S_I + S_{IF}}{i} (1-P_{avg})^i P_{avg}^{\ S_I+S_{IF}} \tag{6}$$

$$GEMP_{S_I} = \sum_{i=S_P}^{S_P+S_{PF}} \binom{S_P + S_{PF}}{i} (1-P_{avg})^i P_{avg}^{\ S_P+S_{PF}} \tag{7}$$

$$GEMP_{S_I} = \sum_{i=S_B}^{S_B+S_{BF}} \binom{S_B + S_{BF}}{i} (1-P_{avg})^i P_{avg}^{\ S_B+S_{BF}} \tag{8}$$

The total PFR using GEM-FEC is expressed as

$$PFR_{GEM-FEC} = G.GEMP_{SI}.(1 + \frac{GEMP_{SP} - GEMP_{SP}^{\ N_P+1}}{1 - GEMP_{SP}} + N_{BP}.GEMP_{SB}.(\frac{GEMP_{SP} - GEMP_{SP}^{\ N_P+1}}{1 - GEMP_{SP}} + GEMP_{SI}.GEMP_{SP}^{\ N_P})) \tag{9}$$

4.2. EGM-FEC Video Model

The Second proposed model for packet loss probability is a model which was derived by Wu and Radha [13] 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

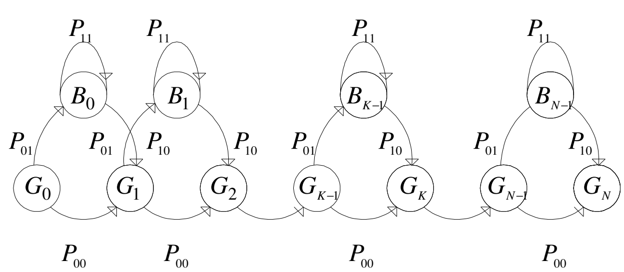


Figure4. The state transition diagram of Extended Gilbert Model (EGM).

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

$$\phi(N,K) = \pi_0(\phi_{G_0G_K}(N) + \phi_{G_0B_K}(N)) + \pi_1(\phi_{B_0G_K}(N) + \phi_{B_0B_K}(N)) \tag{10}$$

Where

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

$$\phi_{G_0B_K}(N) = \begin{cases} \sum_{m=0}^{K} \binom{K}{m}\binom{N-K-1}{m-1} P_{01}{}^{m+1} P_{10}{}^{m} P_{00}{}^{K-m} P_{11}{}^{N-K-m-1} & 0 < K < N \\ 0 & K=N \end{cases} \quad (12)$$

$$\phi_{B_0G_K}(N) = \begin{cases} \sum_{m=1}^{K-1} \binom{K-1}{m}\binom{N-K}{m} P_{01}{}^{m} P_{10}{}^{m+1} P_{00}{}^{K-m-1} P_{11}{}^{N-K-m-1} & 0 < K < N \\ 0 & K=0 \end{cases} \quad (13)$$

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

The state probability P_{01} and P_{10} (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 [13], the authors used the average loss rate P_{avg}, and the packet correlation $\boldsymbol{\rho}$, to give another perception to the state transition probabilities.

$$P_{01} = P_{avg}(1-\rho) \quad (15)$$

$$P_{10} = (1-P_{avg})(1-\rho) \quad (16)$$

For FEC codes, the probability that the node can receive enough packets to decode an FEC block when K packet is transmitted is;

$$P_{suc} = \sum_{i=K}^{N} \phi(N, K) \quad (17)$$

According to (17), the computation of the successful transmission probabilities for a specific *I*, *P*, and *B* frame sizes and FEC block can be written as;

$$EGMP_{S_I} = \phi(S_I + S_{IF}, S_I) \quad (18)$$

$$EGMP_{S_P} = \phi(S_P + S_{PF}, S_P) \quad (19)$$

$$EGMP_{S_B} = \phi(S_B + S_{BF}, S_B) \quad (20)$$

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

$$PFR_{EGM-FEC} = G.EGMP_{SI}.(1 + \frac{EGMP_{SP} - EGMP_{SP}^{N_P+1}}{1 - EGMP_{SP}} + N_{BP}.EGMP_{SB}. \quad (21)$$

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

5. Simulation Result

Based on the above assumptions, we can develop the following illustrative steps to find the optimal playable frame rate using both GEM-FEC and EGM-FEC video models defined in Section 4.

1. In order to compare GEM-FEC and EGM-FEC, one can determine the P_{avg} from (1) by defining a specified state probability. For each set of other system variables, we can compute the PFR.

2. For GEM-FEC model, assuming specific average loss rate, *and* packet correlation ρ, PFR defined in (9) can be estimated using (6), (7) and (8), respectively. Using the three different cases of FEC weights with a certain probability of packet loss; we can evaluate the robust optimal PFR.

3. For EGM-FEC model, the PFR can be evaluated using (18), (19), (20) and (21) for given packet correlation, FEC weights, and certain loss probability. This allows us to characterize the variation of the PFR under different conditions of packet correlation and FEC weights.

5.1. System Settings

Table 1 describes network characteristic of many typical network connection [13][18][26]. A GOP (N_P, N_{BP}) pattern is considered with N_P=2 and N_{BP}=3. In Table 2 a list of three FEC weights is defined to be used in our experiments.

Table 1. Simulation Parameters

Parameter	value
t_{RTT} , t_{RTO}	168ms, $4 \times t_{RTT}$
N_P , N_{BP}	3, 2
S_I	24.64KB [25 packet]
S_P	7.25KB [8 packet]
S_B	2.45KB [3 packet]
P_{00} , P_{11}	0.96, 0.94
P_0 , P_1	0.001, 0.001 to 0.1
ρ	0.9

Table 2. FEC Weights

FEC weight	S_{IF}	S_{PF}	S_{BF}
Light	1	1	0
Medium	4	2	0
High	8	4	1

5.2 Performance Evaluation

This section describes the PFR of an MPEG4 Video streaming over wireless channel using two packets loss models: the GEM-FEC video model and EGM-FEC video model addressed in Section 4. The performance evaluation is carried out by considering the parameters in Table 1 for the fact that maximum frame rate allowed over Internet is 30fps. The results are conducted using Matlab package.

Experiment 1: Figure 5 shows the video quality vs. the average packet error rate using GEM-TFRC and EGM-TFRC packet loss models with no FEC weights. We consider there is no packet correlation for GEM-TFRC because it was found through our experimental results that there is no significant effect on video quality when ρ factor varies between 0 and 1. On the other hand, we evaluated the EGM-TFRC model when ρ factor is considered to be close to 1. As a result, it is clearly noticed that a highest PFR can be achieved for packet loss probability less than 2% in both models. However, ***ρ*** factor has a slight impact on the resultant video quality at packet loss rates greater than 2% especially in EGM-TFRC model. In other word, one can conclude that the GEM-FEC becomes closer to the EGM-FEC at packet loss probabilities below 2%.

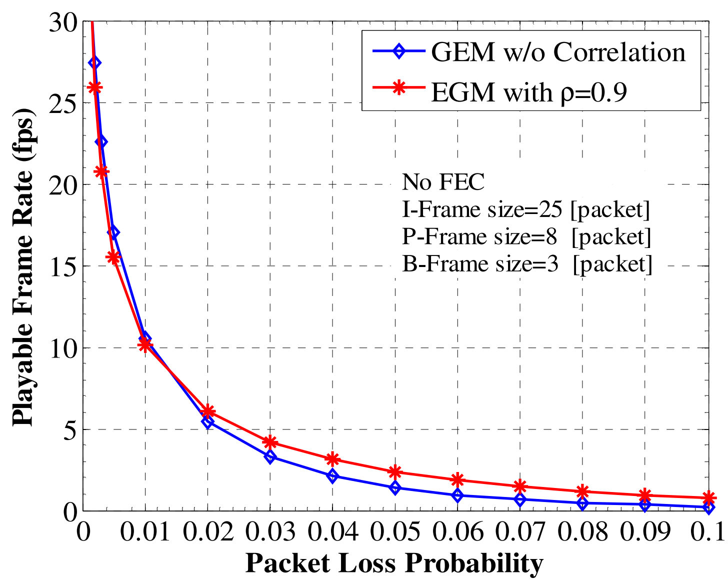
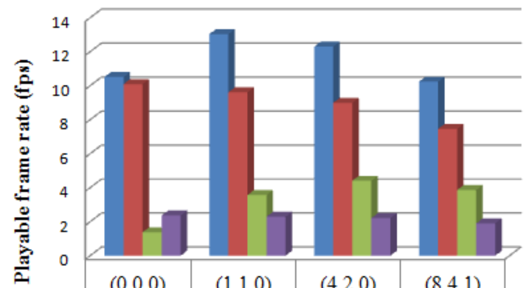


Figure 5.Playable Frame Rate of GEM and EGM models with no FEC



	(0,0,0)	(1,1,0)	(4,2,0)	(8,4,1)
GEM-FEC@P=1%	10.512	13.0337	12.2973	10.2349
EGM-FEC@P=1%	10.0929	9.6112	8.9918	7.4572
GEM-FEC@P=5%	1.3787	3.5662	4.4046	3.862
EGM-FEC@P=5%	2.3648	2.2891	2.2093	1.8966

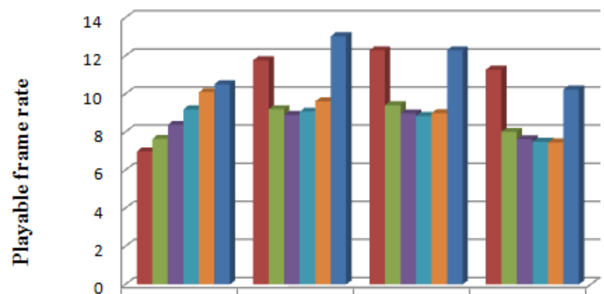
FEC code

Figure 6.Playable Frame Rate of GEM-FEC and EGM-FEC with correlation factor of 0.9

Experiment 2: The results obtained in Figure 6 depict the PFR variations when we add three different FEC weights to both GEM-FEC and EGM-FEC model, respectively. As mentioned in Fig. 5, only EGM-FEC model is evaluated with correlation factor of 0.9; and this factor effect is discarded on the performance of GEM-FEC model. When average loss probability is considered to be 1% (less than 2%), it is clearly observed that GEM-FEC provides highest PFR at light weight FEC (1,1,0) compared to EGM-FEC. Moreover, the former model responds less quality performance at high weight of FEC (8,4,1). Now, when average loss probability loss increases to be 5% it is also found that GEM-FEC outperforms EGM-FEC particularly at medium FEC weight (4,2,0).

Experiment 3: In Figure 7, the effect of packet correlation factor is examined for both proposed packet loss models under different FEC weight codes presented in Table 2. Figure 7 (a) shows the resultant PFR when the average loss rate is set to be 1%, and $\boldsymbol{\rho}$ correlation factor is changed from 0.1 to 0.9 for only EGM-FEC model; meanwhile the $\boldsymbol{\rho}$ correlation factor has no significant role in GEM-FEC model (as shown in Fig. 5). Therefore, it is shown that when $\boldsymbol{\rho}$ varies in the range of 0.5 to 0.9 and FEC weight increases, the perceived PFR may clearly decline. In contrary, when $\boldsymbol{\rho}$ decreases to be less than 0.5 (say, 0.1), the resultant PFR indicates much better performance on FEC weight (4,2,0).

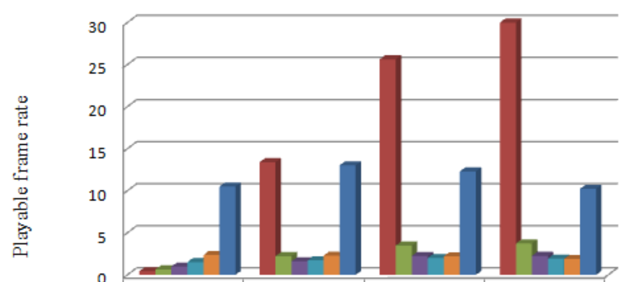
In Figure 7 (b), the effect of higher average loss rate (say 5%) has another serious impact on video quality at the client end. From the chart, one can see that when packet correlation varies from 0.5 to 0.9, the PFR refers to the better performance on FEC weight (4,2,0). However, once the packet correlation decreases to be 0.1, there is a clear improvement in video quality. The perceived PFR achieves the highest quality especially when FEC weight becomes (8,4,1). Thus, we can conclude that EGM model with high FEC provides a robust measure in video transmission over wireless channel compared to GEM model.



	(0,0,0)	(1,1,0)	(4,2,0)	(8,4,1)
EGM @ ρ=0.1	6.9823	11.7739	12.2949	11.2764
EGM @ ρ=0.3	7.6481	9.2055	9.4108	8.0095
EGM @ ρ=0.5	8.383	8.8989	8.9857	7.6278
EGM @ ρ=0.7	9.195	9.073	8.8433	7.4901
EGM @ ρ=0.9	10.0929	9.6112	8.9918	7.4572
GEM	10.512	13.0337	12.2973	10.2349

FEC code

(a)



	(0,0,0)	(1,1,0)	(4,2,0)	(8,4,1)
EGM @ ρ=0.1	0.4323	13.3934	25.6401	41.4504
EGM @ ρ=0.3	0.6498	2.2262	3.4852	3.735
EGM @ ρ=0.5	0.9867	1.6206	2.2529	2.2703
EGM @ ρ=0.7	1.5166	1.7332	2.0185	1.9366
EGM @ ρ=0.9	2.3648	2.2891	2.2093	1.8966
GEM	10.512	13.0337	12.2973	10.2349

FEC code

(b)

Figure7.The impact of packet correlation on video quality for two various packet loss rates (a) P_{avg} =1% and (b) P_{avg} =5% ρ

6. CONCLUSIONS AND FUTURE WORK

In this paper, a frame-level FEC at the application layer is considered to define two analytical packet loss models. The models are applied to estimate the video quality over Gilbert wireless channels with a highly correlated error. The results clearly introduce a good comparative performance. The GEM-TFRC video model shows a good performance in high packet loss probability; meanwhile GEM and EGM both show reasonable performance but for average packet loss rate below 2%. With FEC, it is found that GEM-TFRC at light FEC of (1,1,0) provides the best performance. In contrary, EGM-TFRC with a small packet correlation below 0.5 and a FEC weight of (4,2,0) outperforms GEM-TFRC; otherwise a clear degradation in video quality can be achieved with the increase of packet correlation. As a result, we can conclude that these models are applied to be helpful in predicting the robust video quality over wireless channels. The Future work can be extended to involve the GOP-level FEC and retransmission techniques to improve the performance of video quality under different conditions of wireless environment.

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