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IMPLEMENTATION AND VALIDATION OF AN ADAPTIVE FEC MECHANISM FOR VIDEO TRANSMISSION

Osman Ghazali [1] and Nur Ziadah Harun[2]

School of Computing
UUM College of Arts and Sciences
Universiti Utara Malaysia

osman@uum.edu.my[1]
s71497@student.uum.edu.my[2]

ABSTRACT

This research focuses on investigating the FEC mechanism as an error recovery over a wireless network. The existing adaptive FEC mechanism faces a major drawback, which is the reduction of recovery performance by injecting too many excessive FEC packets into the network. Thus, this paper proposes the implementation of an enhanced adaptive FEC (EnAFEC) mechanism for video transmission together with its validation process. There are two propositions in the EnAFEC enhancement, which include block length adaptation and implementation, and suitable smoothing factor value determination. The EnAFEC adjusts the FEC packets based on the wireless network condition so that excessive FEC packets can be reduced. The proposed enhancement is implemented in a simulation environment using the NS-2 network simulation. The simulation results show that EnAFEC generates less FEC packets than the other types of adaptive FEC (EAFEC and Mend FEC). In addition, a validation phase is also conducted to verify that the proposed enhancement is functioning correctly, and represents a real network situation. In the validation phase, the results obtained from the simulation are compared to the outputs of the other adaptive FEC mechanisms. The validation results show that the mechanism is successfully implemented in NS2 since the number of packet loss falls under the overlapping confidence intervals.

Keywords: Forward error correction, automatic repeat request, video transmission, smoothing factor, blocks length adaptation, validation.

INTRODUCTION

Video transmission over the wireless network is usually interrupted by video packet loss caused by interference, terrestrial obstructions and reflection of transmission signal (Ding, Chen & Wang, 2006). Thus, Forward Error Correction (FEC) can be used to recover the lost video packet to ensure that the video contents can be successfully played at the receiver. FEC is a technique to add a redundant packet into the original packet so it can be reconstructed in the occurrence of packet loss. There are two types of FEC packets generation, which are static and dynamic FEC. The static FEC generates a fix number of FEC packets while a dynamic FEC generates dynamic FEC packets based on the wireless network condition. Currently, a dynamic FEC is required because a static FEC produce extra load due to the fixed number of FEC packets generated on the network (Moid & Fapojuwo, 2008). Thus, the recovery performance will be reduced accordingly. In order to generate the dynamic number of FEC, an Automatic Repeat reQuest (ARQ) mechanism has been adopted with FEC mechanism to overcome the limitation in static FEC. The reason for using FEC with ARQ mechanism is because each mobile node needs to face different wireless network conditions. However, it is difficult to decide the number of FEC packets to be generated. The proper amount of redundant packets must be identified in order not to harm the network.

Recently, there are many researchers investigating this area, and they have proposed various types of adaptive FEC mechanisms. Latré, Staelens, De Turck, Dhoedt, and Demeester (2007) proposed a dynamic FEC mechanism, known as the Hybrid ARQ and FEC (AHAFEC), which is able to alter the amount of FEC packets and the number of maximum retransmission at the base station. The performance of AHAFEC is better than EAFEC as it is able to generate a low number of loss frame. Moreover, Tsai, Ke, Wu, Shieh and Hwang (2008) proposed the Burst Aware FEC (BAFEC) that generates FEC packets based on the information feedback to the sender regarding the average packet loss. Thus, the sender can decide the number of FEC packets. The simulation results showed that BAFEC achieved high PSNR quality. Unfortunately, AHAFEC and BAFEC did not provide any information regarding the amount of FEC packets produced to recover the loss packets. Thus, researchers have been unable to determine the recovery efficiency. Meanwhile, Lin, Ke, Shieh and Chilamkurti (2006a) proposed the Enhanced Adaptive FEC (EAFEC) to adapt to the varying nature of wireless networks. In this technique, queue length is used as the indicator to estimate network traffic load while packet retransmission time is used to indicate wireless channel status. Although EAFEC technique performs better than the static FEC, EAFEC has a limitation whereby the access point might not generate the FEC when the buffer at the queue length is full. If any video packets are lost during that time, no recovery for the lost packets is possible.

In order to solve the above problem of the EAFEC, the authors (Du, Liu & Guo, 2009) proposed Mend FEC. Mend FEC is an enhancement of the EAFEC mechanism that can improve the quality of video in a sudden video changing scene. However, the limitation of the EAFEC mechanism is that when queue length is too large, video packets will be transmitted without adding any FEC packets. This is due to the fact that if queue length is more than the threshold, the number of FEC is set to zero. If the wireless channel state is the worst at that time, the original packets might be dropped, and the receiver will not be able to recover the packets. Besides, the retransmission time in EAFEC is not a good indicator to estimate the number of FEC as it does not fully adapt to the various wireless network conditions. Unfortunately, the Mend FEC generates too many FEC packets recovering the packets from losses. The excessive FEC packets will consume the network bandwidth, and waste the resource. The other issue in wireless network is related to packet loss caused by burst error loss, for example radio interference, fading, and shadowing. However, the previous work ignored the packet loss impact in burst error wireless network (Lin, Ke, Shieh & Chilamkurti, 2006a; Du, Liu & Guo, 2009).

To reduce the excessive FEC packets, block length adaptation and a new smoothing factor value are needed. In addition, this will enhance the existing adaptive FEC mechanism. The objective of this paper is to investigate the distribution of the FEC packets on different adaptive FEC mechanisms over the burst error wireless network. Thus, validation for the experiment using wireless error model under burst error network is introduced to verify that the simulation is functioning correctly and is representing the real network situation.

WIRELESS VIDEO SYSTEM

Figure 1 depicts the structure of a video transmission system over the wireless network environment. Three components in the wireless video system include the video server, wireless access point, and video receiver. At the video sender site, the video encoder segments the original video packets stream into blocks of fixed size (k). Then, the encoded video packets are transmitted to the wireless access point. Access Point is responsible to determine how much FEC packets must be generated, because when the wireless node wants to sent data to other nodes, the data must be sent first to the wireless AP. The amount of FEC generated must be dynamic, based on the variation network conditions. Every k video packet is protected by $h=n-k$ FEC packets for each block, as n is the block size while h refers to as redundant packets (Nafaa, Taleb & Murphy, 2008). At the video receiver, the lost packets are recovered by the FEC packets. If k out of the n is successfully received, the block of packets can be completely reconstructed. After the packet lost recovery, the video decoder will decode the recovered video packets to be played at the receiver.

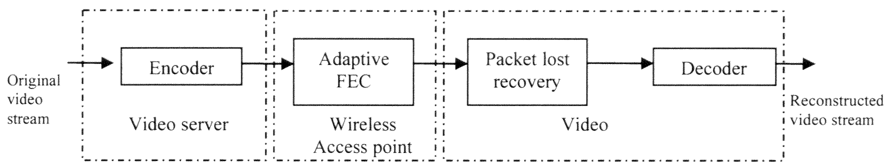

Figure 1. Video transmission system.

REVIEW OF ADAPTIVE FEC MECHANISM

The aspects of the design that influence adaptive FEC performance are described as cross-layer design, queue management policy, congestion level, and wireless error model. The first aspect, cross-layer design, refers to the information exchange within different layers about specific needs and capabilities, and aims to enhance system performance (Moid, 2009). There are two ways to implement cross-layer design: bottom up and top down. The bottom up approach enables a higher level, such as the application level. For example, the FEC has to be configured according to the requirement of lower level such as the physical layer. This allows the FEC packet at the application level to be configured according to the current wireless channel status. However, the top down approach uses the information from upper layer to configure at the lower approach. In this case, the particular channel status is determined according to the characteristic of the application status.

Since the bottom up approach is well suited for the video application due to the adaptation of application level with current network status, the available information from the current channel condition needs to be used as much as possible to enhance the video quality. Therefore, the main aims of implementing a cross-layer design are to increase wireless channel utilization and to adapt the FEC packets with the varying network and traffic conditions.

The conventional FEC approaches are implemented on the application layer in order to recover the packet loss. Unlike the conventional approach, this work proposed to enhance the adaptive FEC mechanism which operates over the wireless network. The adaptive FEC is the error recovery approach that focuses on utilizing functionalities of two different layers, which are the data link layer and transport layer. As shown in Figure 2, the adaptive FEC controller cooperates with the User Datagram Protocol on the transport layer and the Medium Access Control (MAC) protocol on the data link layer. This controller retrieves the failure information from the MAC layer, and uses it to adaptively control the FEC packets generation.

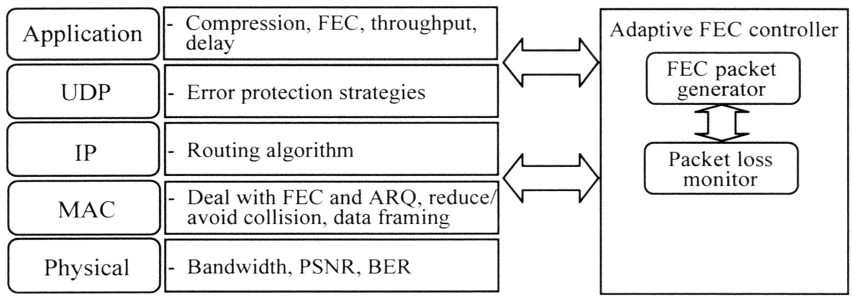

Figure 2. Hybrid cross-layer FEC architecture.

The second aspect is queue management policy, which refers to the scheduling algorithm process in selecting the next packet to be transmitted. It also manages the dropping packet when the buffer at the queue space is filled up (Han, Park, Kang & In, 2010). Apart from that, it is one of the important factors that influence the network throughput, end-to-end delay, and QoS. Even though there are many scheduling policies over the wired network such as First In First Out (FIFO), Stochastic Fair Queuing (SFQ), Fair Queuing (FQ), and Class-Based Queuing (CBQ), only FIFO and Priority Queue (PQ) are used in wireless network (Boukhalfa, Minet, Midonnet & George, 2005). PQ is the oldest scheduling policy implemented as PriQueue in the NS-2 simulator. Besides, PQ serves the packets based on their priority order. This means that a class with higher priority is always processed first compared with a lower priority.

PriQueue has only one queue, which is to receive all types of packet. Using PriQueue, the routing protocol packets gets higher priority, whereby they are inserted at the head of the queue. While the other lower priority packets are inserted at the tail of the queue which employed the FIFO policy. This means that the head of the queue is served first. Since the PriQueue drops the packets at the tail when the buffer queue is full, the generated FEC packets have to be controlled properly. When the network is overloaded with the video packets, generating too much FEC packets to the network may not improve the error recovery rates. Eventually, the quality of video is also decreased.

Another important aspect is the congestion level, which is responsible for measuring the level of congestion in the network. According to Manimekalai, Meenakshi and Abitha (2009), network congestion may happen due to the traffic load that exceeds the available network resource during a video transmission. On the other hand, some packets transmitted are either buffered or discarded by network traffic that monitors the network status. Therefore, queue length is a sufficient and reasonable indicator to measure the level of congestion for the network traffic which has multiple sources sending to the single sink across the wireless network (Rangwala, Gummadi, Govindan & Psounis, 2006; Hull, Jamieson & Balakrishnan, 2004). Besides, the average queue length illustrates the condition of congestion more accurately compared to the instantaneous queue length (Medhi & Ramasamy, 2007). The current Adaptive FEC mechanism uses Exponential Weighted Moving Average (EWMA) to smoothen the estimated value of average queue length. Whenever the packets queue in the buffer, the average queue length is updated according to the following equation:

$$avg_q = (1 - w_q) \times inst_q + w_q \times avg_q \quad (1)$$

avg_q the average queue length
w_q the smoothing factor
$inst_q$the instantaneous queue

Finally, the GE error model is the channel model used to measure the burst pattern error when the packets are lost consecutively over the wireless channel. The GE channel model is based on the two states of the Markov Chain, as illustrated in Figure 3. The "good" state (G) losses occur with the lower probability P_G while in the "bad" state, losses happen with higher probability P_B, where $P_B > P_G$.

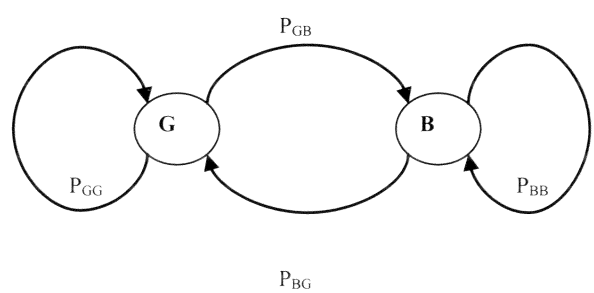

Figure 3. Gilbert-elliot channel model.

The other two parameters are introduced as P_{GB} = 1-P_{GG} and P_{BG} = 1-P_{BB}. P_{GB} stands for the probability of the state transition from a good state to a bad state, and P_{BG} is the transition from a bad state to a good state. The steady state probabilities that the channel is in good status is:

$$\pi_G = \frac{p_{BG}}{p_{BG} - p_{GB}} \tag{2}$$

And the steady state probabilities that the channel is in bad status is:

$$\pi_B = \frac{p_{GB}}{p_{BG} - p_{GB}} \tag{3}$$

Therefore, the average packet loss rate produced by the GE error model is:

$$P_{avg} = P_G \pi_G + P_B \pi_B \tag{4}$$

Finally, the possible status of the GE error model is $\pi_G + \pi_B = 1$.

Currently, the GE model is sufficiently complex to model the burst error behaviours over the wireless network. Compared to the random uniform model, the figure from the GE error model is closer to the real wireless error condition. It also generates a lower frame error rate because the GE model provides a characteristic burst error pattern. Work from Ke, Lin, Shieh and Hwang (2006) shows that when compared to the random uniform model, the GE model produces better video quality at the receiver end at the same packet error rate. Therefore, the GE model is the best model to measure burst error pattern that usually occurs in the wireless channel. The phenomenon of both error model distributions is explained in Figure 3, the grey slices represent the loss slices during the transmission.

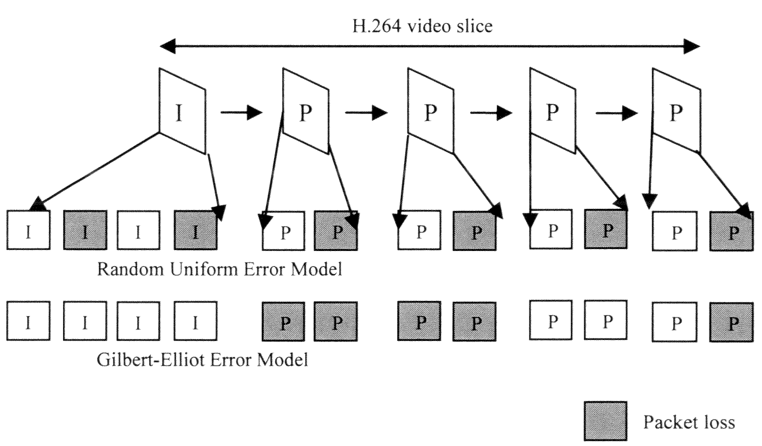

Figure 4. H.264 Video transmitted with different wireless error distributions.

The impact of random packet loss on video transmission is larger than burst packet loss. As shown in Figure 4, I-slice packets are more lost in a random distribution. The losses of I-slices packets cause more impact to the video quality since P and B slices depend on the succeeding of I-slice. Otherwise, they cannot be decoded appropriately at the receiver (Oh, Hua & Chen, 2008).

PROPOSED ENHANCEMENT FOR ADAPTIVE FEC MECHANISM

The proposed enhancement, Enhanced Adaptive FEC (EnAFEC) is an extension of the current Adaptive FEC mechanism. The enhancement is done by implementing block length adaptation and determining a suitable smoothing factor value. This is important in order to reduce the unnecessary FEC packets injected into the wireless network. To implement the FEC mechanism, several conditions have been considered. Firstly, a block of packet is considered for FEC packets generation, whereby the FEC packets are generated on the top of one video block. Secondly, the fixed numbers of video packets are grouped together into a video block. Thirdly, the maximum number of FEC packets must not exceed the number of original video packets in a block in order to minimize congestion in the network.

The block length adaptation is expected to solve the problem faced by the current adaptive FEC mechanism by increasing the video packet length to reduce the packet error rate. Therefore, the length of FEC packets is also increased as the video packet length increases because the FEC packets are generated on top of the video blocks. Thus, the recovery performance produced by the FEC mechanism can be enhanced. Based on the work from Whetten, Vicisano, Kermode, Handley Floyd and Luby (2001), the advantages of block length adaptation are specification reuse and complexity reduction. Due to these advantages, the enhancement of the Adaptive FEC mechanism has been designed based on block length adaptation. After determining the suitable block length, Access Point will then decide the appropriate number of FEC packets by using average retransmission times. The first set of FEC packets that will be obtained is referred to as numFEC1. Then, the dynamic values of the smoothing factor must be generated based on the number of retransmission times at the MAC layer. The dynamic value is needed to generate the FEC packets based on the variation network conditions. The new smoothing factor value is required to replace the existing static smoothing factor value. After that, the second set of FEC packets, i.e. numFEC2, is obtained based on the average queue length. Lastly, the final FEC packets can be obtained by calculating the set of redundant packets (numFEC1 and numFEC2).

This research presents a block length adaptation technique that can reduce the excessive FEC packets. In addition, a suitable replacement value for the static smoothing factor is also proposed in order to fit to the various network conditions. Based on the above description, Figure 5 proposes the steps in enhancing the adaptive FEC mechanism that can reduce the unnecessary FEC packets.

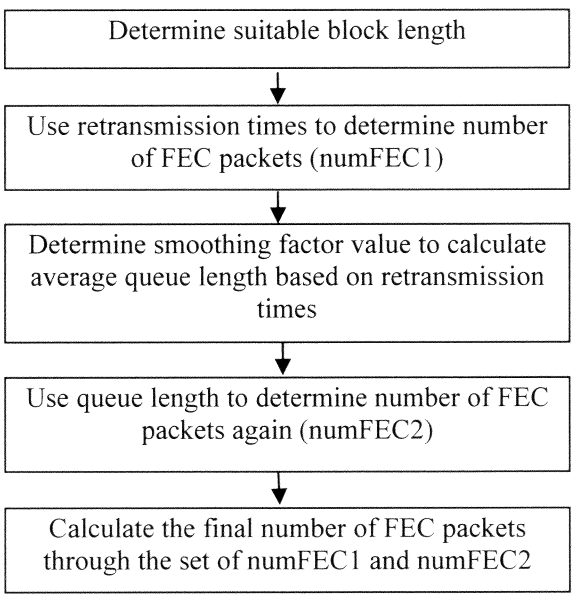

Figure 5. Steps in proposed enhancement.

SIMULATION SETTING

The video packets are delivered via multicast transmission with the GE error model. The P_{GG}, P_{BB} and P_G are set to 0.96, 0.94, and 0.001. In this case, P_{GG} and P_{BB} are set to high values while P_G is set to low value to represent the bursty nature of a wireless network. The packet error probability (P_B) represent the bad channel with values which vary from 0.2 to 0.5. For the simulation setting, the video traffic used for this experiment is “Highway.qcif” with 176 x 144 pixels. Video traffic is not the only traffic present in the network during the simulation. There are two background traffics generated to interfere with the video traffic. The first is FTP traffic that represents a bulk of file applications that are transmitted using TCP packets. The second is exponential traffic that represents the burst traffic transmitted using UDP packets. At first, the FTP traffic and exponential traffic are transmitted randomly between 0 to 1 second. The video traffic is then transmitted after 1 second.

At the network layer, No Adhoc routing protocol (NOAH) has been selected which support direct communication between wireless nodes, or base station nodes, or mobile nodes. At the MAC level, the video frames are segmented into small packets which are based on the maximum packet size of the network. Too small packets size will consume much time in transmitting queue, and will be dropped by the access point due to the overflow at the queue (Gopal, Ramaswamy & Wang, 2004). Kuo, Tsai, Shih and Shieh (2007) conducted an experiment to evaluate the FEC efficiency based on different packet sizes. They proved that a small packet size will increase the delay time because it produces more headers overhead. Therefore, in this study, the maximum packet size of video is set to 1500 bytes as suggested by Lin, Ke, Shieh and Chilamkurti (2006b) which resulted in the best video transmission quality.

The typical maximum of retry limit in IEEE 802.11 is set to 4 in the simulation in order to achieve the most wanted delay objective for the video packets (Sgardoni, Ferre, Doufexi, Nix & Bull, 2007). As mentioned in Lee and Kang (2006), the bad network condition will occur if the retransmission attempt at the MAC layer reaches retry limit.

The data analysis is based on the trace file generated during the simulation process, and data is taken between the range of 0 to 70 seconds, which refers to the time taken to transmit data from sender to the receiver. All simulations are run at least 20 times and at most 100 times. Finally, the results are calculated at 95% confidence intervals. Table 1 shows the detailed setting for the simulation parameter.

Table 1

Simulation Parameters

Network Parameters	
Wireless error model	Gilbert-Elliot (GE) Error Model
Packet error probability (P_B)	0.1, 0.2, 0.3, 0.4, 0.5
Source Coding	
Sequence name	Highway

(continued)

Network Parameters	
Resolution	QCIF (176 x 144 pixels)
Sequence length	2000 Frames
Video packet size	1500 bytes
Channel Coding	
Reed-Solomon	RS(n, k)
RS(n, k)	RS(24,12)
Simulation Environment	
Antenna type	OmniAntenna
Propagation model	TwoRayGround
Network size	500m x 500m
MAC protocol	MAC 802.11b
RTS/CTS function	Disable
Retry limit	4
MAC bandwidth	11 Mbit
IFQ length	50
Background traffic	FTP & CBR

SIMULATION RESULTS

The access point generates a certain number of FEC blocks on top of each block video packets. It is important for the adaptive FEC mechanism to dynamically adjust the number of redundant packets according to the changing network conditions in order to avoid network congestion and reduce bandwidth utilization. This part explains the trend of FEC packets generated at different packet error probabilities, which represents different network conditions.

FEC Packets for EAFEC

Based on Figure 6, EAFEC will only start to generate FEC packets after 40 seconds. If packet loss occurs before that time, no error recovery for the packet is lost. This is irrelevant because the unrecoverable loss packets lead to bad video quality at the receiver end. It can be proven that the FEC blocks for each video block will follow the the same trend. Figure 6 highlights that after 40 seconds, the FEC is generated for all the packet error probabilities (P_B).

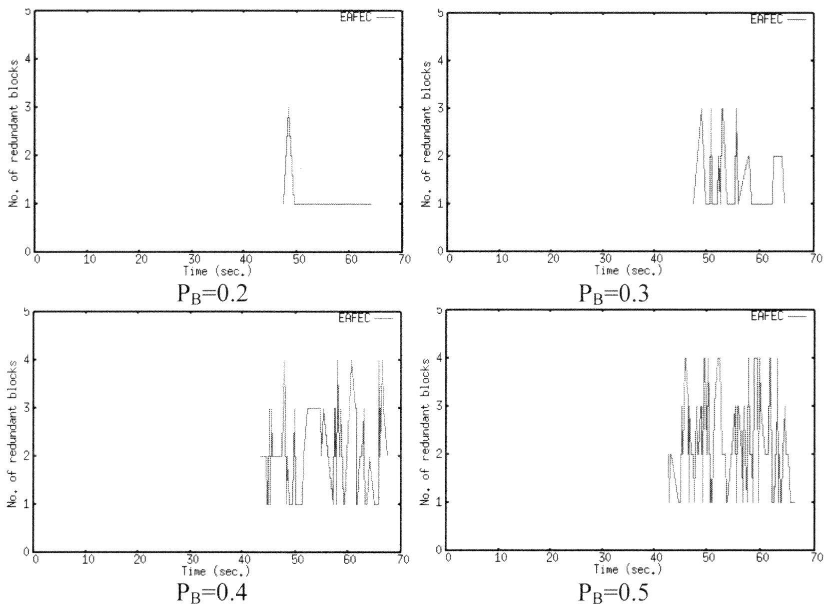

Figure 6. Number of redundant blocks vs. time for EAFEC.

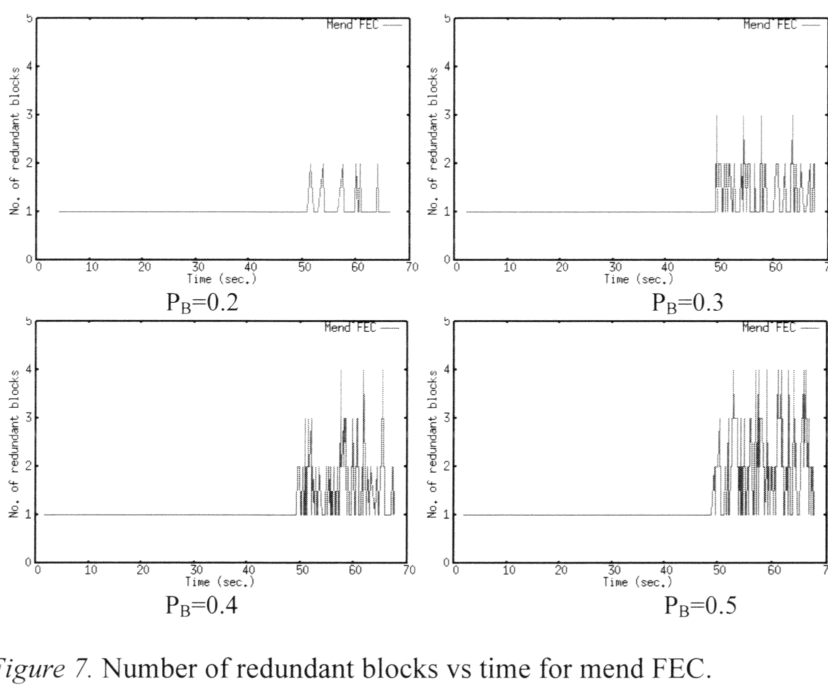

Figure 7. Number of redundant blocks vs time for mend FEC.

FEC Packets for Mend FEC

The Mend FEC generates FEC packets along the video transmission process. At the beginning of the simulation, the number of FEC blocks generated by the Mend FEC algorithm is small. After a few seconds, the number of FEC blocks increases as the time increases, and the FEC blocks continue to generate almost along the time. Based on Figure 7, it is proven that the number of FEC blocks for each video block achieves the same trend by generating the FEC blocks along the video transmission process for all packet error probabilities (P_B).

FEC Packets for EnAFEC

The EnAFEC generates only one block of FEC packets for each video block when packet lost happens, otherwise none of the FEC packets is generated. This mechanism is able to reduce the number of excessive FEC blocks in order to avoid network congestion, and at the same time, network bandwidth utilization can be saved. Based on Figure 8, the number of FEC blocks for each video block achieves the same trend by generating one block of FEC packets for all packet error probabilities (P_B).

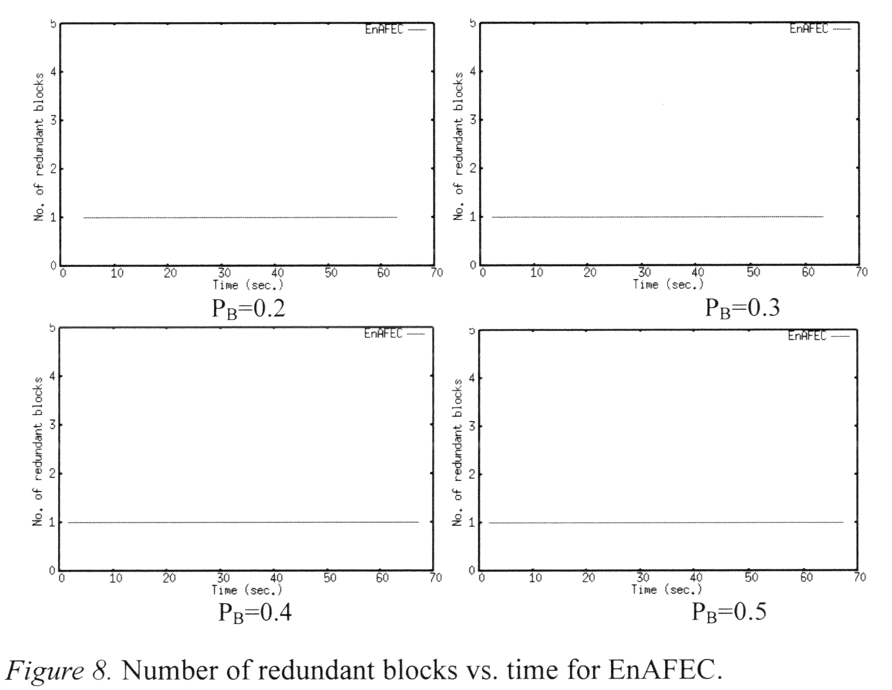

Figure 8. Number of redundant blocks vs. time for EnAFEC.

Figure 6, 7 and 8 show the number of FEC packets generated by EAFEC, Mend FEC and EnAFEC mechanisms under different packet error probability (P_B). Theoretically, only one packet of FEC is needed to recover one packet of source video. When more than one FEC block is injected into the network, congestion might arise and more video packets will be dropped, resulting in the wastage of FEC packets. Also generating multiple FEC blocks wastes network bandwidth which may also contribute to network congestion. The EnAFEC generates less FEC packets compared to the EAFEC and Mend FEC, conserving the precious bandwidth as well as protecting the network from congestion.

PACKET LOSS VALIDATION

The validation phase is an important task to make sure that the proposed mechanism is correctly implemented inside the network simulator. The results of simulation are valid if they match the known output. If the results do not match, the mechanism has to be corrected, and the validation process for the failed component will be repeated until the correct result is achieved. In this work, the number of packet lost, before recovering with the FEC packets, are used to validate the implemented adaptive FEC mechanism. Table 2 shows the partial code for the packet loss implementation using the Gilbert Elliot (GE) model that has been set up in Tcl file.

Table 2

Partial Code for Packet Loss Implementation

```
#$wl_phy set-error-level $P_GG $P_BB $P_G $P_B $loss_model
#loss model->(0:random uniform; 1:GE)
set wl_phy [$wl_node_(0) set netif_(0)]
$wl_phy set-error-level $opt(1) $opt(2) $opt(3) $opt(4) $opt(5)
```

The number of packet loss is then calculated for each adaptive FEC mechanisms. The P_{GG}, P_{BB}, and P_G are set at 0.96, 0.94, and 0.001 respectively, while P_B is set from 0.3 to 0.5. The simulations are run at least 30 times in order to get the confidence interval. The packet error probability (P_B) that represents the channel is in a bad state. It varies from 0.3 to 0.5, a range which is based on previous studies (Lin, Ke & Shieh, 2005; Lin, Ke, Shieh & Chilamkurti, 2006b; Oh, Hua & Chen, 2008; Moid 2008b; Han, Park, Kang, & In, 2010). Packet error probability is important to simulate packet loss pattern on WLANs. According to the formula , the average packet error rate is set to

0.12, 0.16 and 0.2. The example of the calculation to obtain the average packet error rate () is discussed here, using the parameter $P_{GG} = 0.96$, $P_{BB} = 0.94$, $P_G = 0.001$ and $P_B = 0.5$.

$$\begin{aligned}
P_{avg} &= P_G\pi_G + P_B\pi_B \\
&= p_\pi\left(\frac{p_{BG}}{p_{BG}-p_{GB}}\right) + p_B\left(\frac{p_{GB}}{p_{BG}-p_{GB}}\right) \\
&= 0.001\left(\frac{1-0.94}{(1-0.94)+(1-0.96)}\right) + 0.5\left(\frac{1-0.96}{(1-0.94)+(1-0.96)}\right) \\
&= 0.001\left(\frac{0.06}{0.06+0.04}\right) + 0.5\left(\frac{0.04}{0.06+0.04}\right) \\
&= 0.001\left(\frac{0.06}{0.1}\right) + 0.5\left(\frac{0.04}{0.1}\right) \\
&= 0.0006 + 0.2 \\
&= 0.2006
\end{aligned}$$

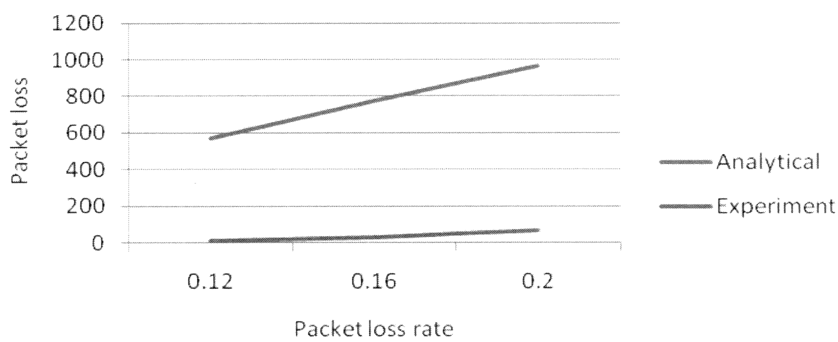

Figure 9. The packet loss of a GE model.

As shown in Figure 9, the number of packet loss in simulation is lower than in an analytical model. According to Lin, Ke and Shieh (2005) and Lin, Ke, Shieh and Chilamkurti (2006), the number of packet loss in the real wireless network must be better than the result from an analytical model. The analytical model only provides the predicted bounds of the number of packet loss over a wireless network. Besides, an analytical model requires more assumptions which may result in inaccuracies about the actual network (Abdullah, Ramly, Muhammed, & Derahman, 2009).

a) Packet error probability = 0.3

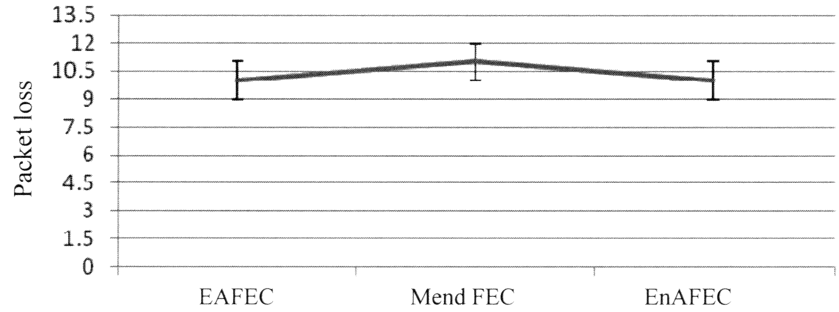

Figure 10. Packet loss vs. adaptive FEC mechanism (P_B =0.3).

b) Packet error probability = 0.4

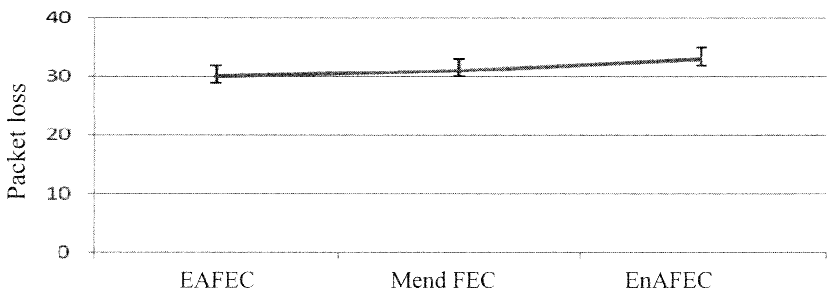

Figure 11. Packet loss vs. adaptive FEC mechanism (P_B =0.4).

c) Packet error probability = 0.5

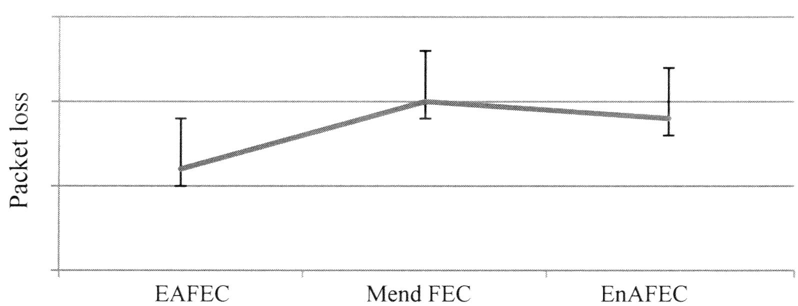

Figure 12. Packet loss vs. adaptive FEC mechanism (P_B =0.5).

The results show that the number of packet loss for different mechanisms is nearly the same. This is proven by the overlap plotted graph as shown below. Apart from that, the number of packet loss increased as the packet error probability increased. This is due to the fact that more packets were dropped during bad network conditions. Therefore, the number of packet loss trend indicates that the implementation of the adaptive FEC mechanism in the ns-2 is working correctly.

CONCLUSION

This paper presents the implementation and validation process of the adaptive FEC mechanism for video transmission. The background of wireless video transmission with adaptive FEC mechanism has been discussed briefly. There are four aspects of design that influence the adaptive FEC performance, and these include cross-layer design, queuing policy, congestion level, and wireless error. To improve the performance of the adaptive FEC mechanism, the enhancement on block length adaptation and smoothing factor determination is proposed. The enhanced mechanism is known as EnAFEC. Block length adaptation is important to reduce the packet error in the video packet sequences. In addition, a smoothing factor is required to eliminate the effect of short term fluctuation in network traffic in order to produce a weighted average of queue length. The work shows that the EnAFEC mechanism injected the lowest number of FEC packet into the network for all packet error rate compared to the other adaptive FEC mechanisms. When the network is fully loaded, less FEC packets are required. This is because it can avoid the congestion on network, and save the bandwidth. Experiments with different packet error probabilities show that the distribution of FEC blocks for each adaptive FEC mechanism achieved the same trend. This indicates that the generation of FEC packets are consistent in different network conditions. A part from that, the validation process based on the number of packet lost has been carried out to make sure that the simulation for the adaptive FEC mechanism is working correctly. The validation results show that the number of lost packets for different mechanisms decreased in the overlapping confidence intervals. The validation results prove that the implementation of the adaptive FEC mechanism is working correctly in the simulation environment.

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