Performance evaluation of proposed packet recovery systems using forward error correction technique over wireless link

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ABSTRACT

Real time communication system's success depends on the ability of service provider to give better quality of service to their end users. The reliability of real time service like video, audio and image streaming depends on the packet loss rate of wireless channel. The packet loss rate of wireless channel can be resolved by using appropriate technique such as forward-error correction (FEC) technique. In designing media streaming applications, media (video, audio or image) quality, packet loss rate are of prime concern. So the packet recovery has the greater impact on the quality of received media frame. These issues can be handled by using adaptive FEC.In this technique redundancy of FEC code is controlled by extracting some parameters of received frame like average packet loss rate,MBL and MILD metric and power to signal noise ratio. In this paper we propose Method1 and Method2 and does comparative analysis of both methods with Conventional Method.

Keywords - Interleaving, FEC, MBL, MILD, PSNR.

1. INTRODUCTION

The wireless channel are much nosier and have both multipath fading and shadowing effect, thus making bit error rate very high. This will directly influence the quality of service of received media frame. The media is being sent on the packet based wireless network. The packet lost determines the quality of received signal. If there are more packets lost the quality of received signal will be degraded, so we use the error control techniques to recover the lost packets in order to enhance the received signal quality.

Packet recovery at the receiver end is done in two ways, the first is the automatic repeat request (ARQ) technique and another is the forward Error correction (FEC) technique [1]. In the ARQ technique, if the packet is lost, the receiver sends the request to the transmitter to transmit the packet which had been lost. And then transmitter transmits the same packet to receiver. If we use such technique in real time application like streaming of video, audio or image signals, it will affect the media quality of the signal. So we should have the technique which recovers the packet at the receiver end. Means we need the error correction technique to recover the data. Such error correction technique is called as forward error correction technique. This technique is positioned at the level 4 which is the transport layer of OSI model provides server/client or transmitter/receiver communication via wireless network. For the sake of adjusting optimal amount of redundancy of FEC codes, we derive two methods, Method1 controlles redundancy of FEC codes by calculating MBL and MILD metric, another Method2 controlles redundancy by calculating PSNR of received frame [2].

The remainder of this paper is organized as follows. Section II gives background related to forward error correction technique. In Section III, we present conventional and proposed packet recovery methods and their path loss models. Section IV is devoted to result analysis. Finally, Section V elaborates conclusion.

2. FORWARD ERROR CORRECTION TECHNIQUE

Forward error correction technique is used to handle the losses in real-time communication [1]. This technique correct the losses at the receiver end without interacting with sender. (n, k) block code converts k source symbols into n coded symbols. Generally the first k data in each group is identical to the original k source data, the remaining n-k data is referred as parity data.

These codes are able to correct both errors and erasures in a block of n symbols. Error is defined as corrupted symbol in an unknown position and erasure is a corrupted symbol in a known position.FEC codes are LDPC, XOR and Reed-Solomon codes. We prefer the Reed-Solomon codes because it is maximum distance separable code (MDS).Means it has the capability to recover lost source data symbols from fewer received code symbols

3. COMPONENTS OF PACKET RECOVERY PROCESS

3.1. Reed-Solomon Encoding and Decoding:

Reed-Solomon codes are non-binary cyclic codes with symbols having m bit sequences, where m>2[3].The Reed-Solomon code is specified as (n, k) = (2^m-1,2^m-1-2t_c) where k is number of data symbols, n is the number of code symbols, m is number of bits in each symbol ,$h=n-k=2t_c$ is the number of parity symbols and t_c is error correcting capability of the code. If $m(x)$ represents message polynomial, then the systematic form of RS code is given by $c(x) = x^{n-k}m(x) + p(x)$ where parity polynomial $p(x)=Rem[x^{n-k}m(x)/g(x)]$.The corrected codeword $c(x)=r(x)+e(x)$, where r(x) is received polynomial and e(x) is error polynomial.

3.2. Interleaving and Deinterleaving:

Time interleaving is commonly used in video streaming systems to reduce the effect of noise burst [1], [3]. The purpose of time interleaver is to disperse the set of symbols over time using block interleaver.The block interleaver accepts a set of symbols and rearranges them, without repeating or omitting any of the symbols in the set. It disperses the symbols by using permutation table. The key advantage of interleaving is that it provides the better error recovery while not increasing the bandwidth requirement of a stream. It is particularly effective for multimedia streams with short term dependencies between the data, here adjacent losses result in error propagation, which affects the quality of decoded content as even correctly received data might not be decoded properly.

However, the pitfall of interleaving consists in the fact that it increases delay due to additional buffering at the sender. This limit the application of this technique to delay sensitive interactive applications.

Interleaved FEC protection is based on the combination of two well-known techniques, FEC and interleaving. This combination may increase FEC efficiency and consequently reduce the amount of FEC transmission at servers.FEC techniques are not sufficient to safeguard data transmission from burst errors, as they are only effective in counteracting random losses.FEC interleaving is capable of minimizing the effect of burst errors at the decoding level, although its efficiency still depends on the amount of FEC redundancy being transmitted and interleaving stride used.

At the receiver we use deinterleaving.Now after deinterleaving the effect of burst losses is minimized and now they become the scattered losses and are available for decoder for decoding.

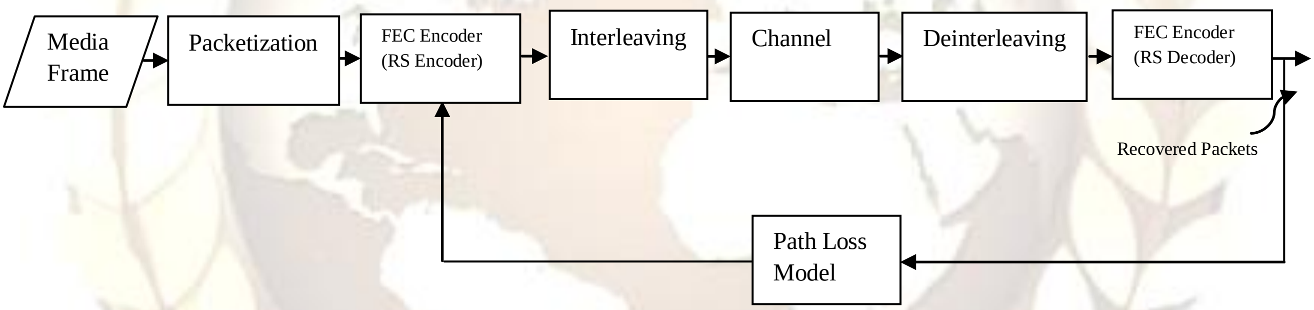

Fig.1 Block Diagram of Packet Recovery Process

3.3. Channel

In this work, additive white gaussian noise (AWGN) channel is used in which the only impairment to communication is a linear addition of white noise with a constant spectral density expressed as watts per hertz of bandwidth and a gaussian distribution of amplitude. The model does not account for fading, frequency selectivity, interference, nonlinearity or dispersion. However, it produces simple and tractable mathematical models which are useful for gaining insight into the underlying behavior of a system before these other phenomena are considered. If the average received power is p in watts and noise power spectral density is No in watts/Hz, then AWGN channel capacity is

$$\mathrm{c} = Blog_2\left(1 + \frac{p}{NoB}\right) \text{ Bits/sec} \qquad (1)$$

Where p/NoB is the received signal to noise ratio and B is bandwidth in Hz.

3.4. Path Loss Model

At the output of RS decoder, we get recovered frame, sometimes this recovered frame is not fullely recovered but it is partially recovered. Means some packets in the frame are recovered and some are not recovered i.e. they are corrupted .The effect of channel noisy pattern is reflected on received frame. The loss model calculates the redundancy h for the RS encoder by calculating the parameters like average packet loss rate, loss metric and video quality of received frame.

In conventional method, the loss model calculates the redundancy h of RS code by calculating average packet loss rate p of received frame and in proposed method1 the loss model calculates redundancy by calculating loss metric like mean burst loss length (MBL) and mean interloss distance (MILD) metric. In proposed method2, it calculates redundancy from video quality of received frame which is expressed in terms of power to signal noise ratio (PSNR)[4]. According to redundancy calculated by path loss model, the RS encoder will change its specification expressed as (n, k). So the dimension of the input frame applied to the RS encoder will change adaptively. The loss model is different for conventional, proposed method1 and proposed method2.

3.4.1. Path Loss Model for Conventional Method:
In this model we count number of corrupted and received packets in the received frame. The average packet loss rate (p) is defined as follows:

$$p = \frac{Number\ of\ corrupted\ packets}{Number\ of\ received\ packets} \qquad (2)$$

So the path loss model first calculates the average packet loss rate p by the above mentioned relation [2]. The adaptive FEC scheme in this model is based on systematic RS (n, k) codes, where n and k are reassigned for each frame to be transmitted based on current channel loss rate. Therefore it is easy to calculate the amount of redundant packet h using the average packet loss rate (p) and the number n of coded packets of current frame to be transmitted as

$h=p.n$ (3)

So we get the new value of redundancy h for each received frame. For this new value of h, the RS encoder's specification changed, since $h=n-k$.Means as h changes n and k of RS encoder also changes. So the dimension of input frame applied to RS encoder changes as k of RS encoder has been changed [5]. This process is repetitive and the changed specification of RS encoder has the effect on media quality of received frame.

This model poses a problem when dealing with channels that exhibit varying packet loss rates over time. In reality, the packet loss process is often variable over time, so the use of the average packet loss rate to adjust the FEC redundancy may not produce optimal FEC redundancy transmission. Obviously, the average packet loss rate is more useful for a very large FEC block, but fails to capture short-term loss process fluctuations. To address this issue, proposed method1 is used to capture the correlation between packet losses, where it is shown that the path loss model in proposed method1 improves the performance of FEC system by using loss metric like MBL and MILD instead of the average packet loss rate p to control redundancy of RS codes.

3.4.2. Path Loss Model for Proposed Method1:
The path loss model calculates, MBL and MILD metric from the channel noisy pattern [2][6]. The systems keep a counter l, *which* counts the number of consecutively corrupted packets and is reset whenever the next packet is delivered successfully.
Let m_i,i=1,2,…,n-1 is loss bursts having length i,where n-1 is the longest loss burst,m_0 denotes the number of delivered packets Note that a model can be entirely described by its burst loss length occurrence vector M, $M=(m_0,m_1,\ldots,m_{n-1})$ .The mean burst loss length is calculated as:

$$MBL = \frac{Total\ number\ of\ corrupted\ packets}{Total\ number\ of\ lost\ burst} \sum_{k=1}^{n-1} P(X=k) \qquad (4)$$

$$MBL = \frac{\sum_{k=0}^{n-1} i m_i}{\sum_{k=1}^{n-1} m_i} \qquad (5)$$

MBL gives the expected loss run length based on the previously observed loss pattern. The another metric that we are calculating is MILD loss metric. It is the distance between packet losses in terms of sequence number. Larger the value of ILD metric better the performance of FEC codes. Let d_i,$i=1,2,\ldots,n-1$ denote the number of ILDs having length i. The ILD model is completely described by its ILDs occurrence vector given by

$D=(d_1,d_2,\ldots,d_{n-1})$ (6)

The Mean Inter-Loss Distance (MILD) is given by

$$MILD = \frac{\sum_{k=0}^{n-1} i d_i}{\sum_{k=0}^{n-1} d_i} \qquad (7)$$

The Inter-Loss Distance gives the idea about separation between the packet losses. It is useful to the loss model for an enhanced loss pattern prediction and multimedia application adaption. Fig.2 illustrates MBL and MILD metric where consecutive corrupted packets are shown by cross mark representing noise burst whose average length is calculated by MBL metric and average distance between two noise burst is calculated by MILD metric. By taking appropriate amount of redundancy h=MBL improves the FEC efficiency and by taking number of source data k=MILD improves the bandwidth efficiency of FEC codes [5].

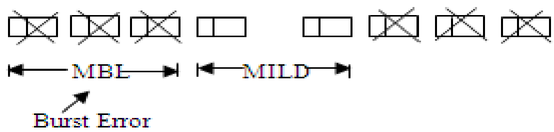

Fig.2 Illustration of MBL and MILD metric

3.4.3. Loss Model for Proposed Method2:

Here Loss Model is used to capture the channel noisy dynamics by calculating the power to signal noise ratio PSNR between the transmitted and received frame[4]. Higher the PSNR better the quality of received frame. To compute PSNR, first calculates the mean squared error using the following equation.

$$MSE = \sum_{i=1}^{x}\sum_{j=1}^{y}\left(\frac{\left(|A_{ij} - B_{ij}|\right)^2}{x.y}\right) \qquad (8)$$

Where *x* and *y* are the number of rows and columns in the frame respectively. *A* is transmitted frame and *B* is received frame. The PSNR is calculated by the following equation

$$PSNR = 10log_{10}\left(\frac{255^2}{MSE}\right) \qquad (9)$$

We go on increasing the redundancy *h* of RS encoder by calculating media quality of received frame till we get fullely recovered frame. So the loss model gives new *n* and *k* specification of RS encoder.

4. RESULT ANALYSIS

The comparative analysis of conventional, proposed method1 and proposed method2 is shown in the following Table 1

Conventional Method				Proposed Method1			Proposed Method2		
Frame index	Video quality in PSNR(dB)	Packet error rate	FEC Efficiency (%)	Video quality in PSNR(dB)	Packet error rate	FEC Efficiency (%)	Video quality in PSNR(dB)	Packet error rate	FEC Efficiency (%)
1	64.72	0.035	83.83	59.54	0.046	**82.88**	**64.20**	**0.032**	83.73
2	59.10	0.244	84.16	64.34	0.132	**85.60**	**75.40**	**0.036**	81.84
3	100	0	71.46	45.53	0.043	**84.50**	**56.15**	**0.014**	80.97
4	39.42	0.325	70.48	59.20	0.111	**85.06**	**100**	**0**	80.08
5	100	0	70.42	59.21	0.088	**85.75**	**100**	**0**	80.18

Table1: Comparative analysis of Conventional Method, Proposed Method1 and Proposed Method2

Simulation is carried out by using five image frames for conventional, proposed method1 and proposed method2 for 33.8 dB signal to noise ratio. Video quality expressed in decibels (dB), packet error rate and FEC efficiency for both methods is calculated. The results of all three methods are shown in Table1 and their plots are shown in Fig.3, Fig.4, Fig.5, Fig.6, Fig.7, and Fig.8.The simulation session involves transmission of 1193,853 and 1177 packets respectively for conventional, proposed method1 and proposed method2.Thus we have observed the 140, 71 and 20 packets corrupted during whole streaming session for conventional, proposed method1 and proposed method2 respectively. The simulation is carried out in two parts, first compares conventional and proposed method1 and another compares conventional and proposed method2.

In first simulation, for proposed method1,the video quality, packet error rate and FEC efficiency is 20% less(which is 15.08dB less),30-31% less(which is less by 0.037),and 10-11 % more respectively compared to conventional method..In second simulation, for proposed method2,the video quality, packet error rate and FEC efficiency is 8-9% more(which is 6.52dB more),86-87% less(which is less by 0.105),and 6-7 % more respectively compared to conventional method.

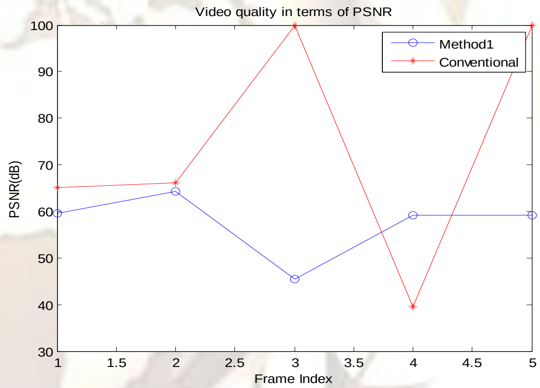

Fig.3 Comparing video quality of conventional and proposed method1

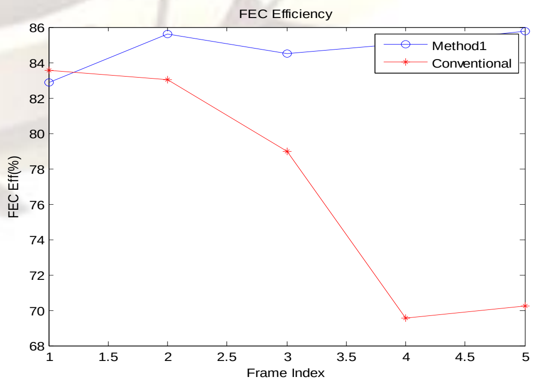

Fig.4 Comparing FEC efficiency of conventional and proposed method1

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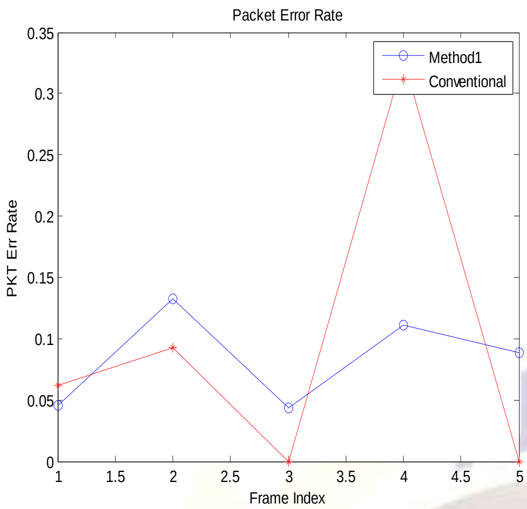

Fig.5 comparing packet error rate of conventional and proposed method1

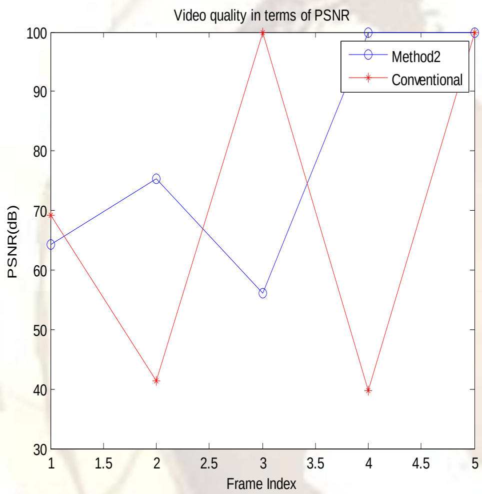

Fig.6 comparing video quality of conventional and proposed method2

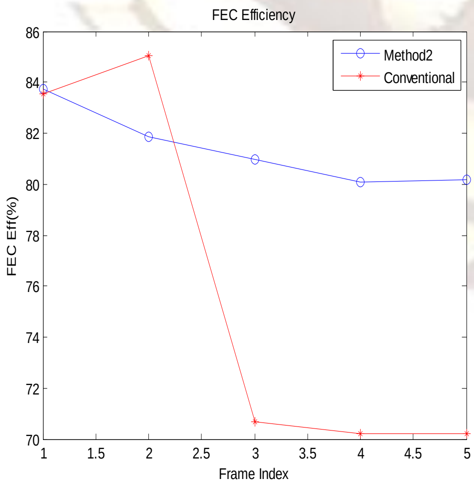

Fig.7 comparing FEC efficiency of conventional and proposed method2

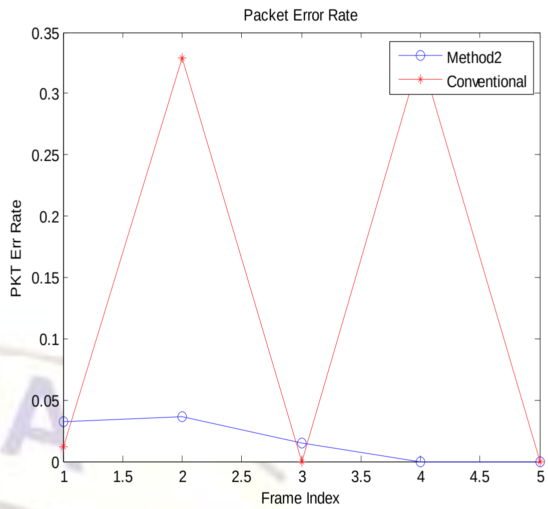

Fig.8 comparing packet error rate of conventional and proposed method2

5. CONCLUSION

By combining the best features of forward error correction technique and interleaving the proposed method1 and proposed method2 is integrated to the transport layer. The simulation results demonstrated that proposed method1 where the path loss model controlles redundancy by calculating MBL and MILD metric from the received frame is advantageous in improved FEC efficiency compared to conventional and proposed method2.In proposed method2 where loss model controlles redundancy between transmitted and received frame has better video quality, reduced packet error rate and moderate FEC efficiency compared to conventional and proposed method1.So among all three methods, proposed method2 is a better choice for packet recovery.

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BIOGRAPHIES

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