Hybridized Interleaved Automatic Repeat Request and Enhanced Adaptive Sub-Packet Forward Error Correction mechanism for Cloud

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Abstract—With the increase in the demand for the high quality video streaming services over the mobile networks, the wireless link capacity cannot practically cope up with the rising traffic load of users across the globe. Packet loss in the network affects the perceptual transmission quality of the video streams. To avoid this issue, this paper presented a hybridized Interleaved Automatic Repeat Request (IARQ) and Enhanced Adaptive Sub-Packet Forward Error Correction mechanism (EASP-FEC) (IARQ-EASP-FEC) to improve the video quality for effective video streaming in the cloud environment. EASP-FEC mechanism splits a video packet into a group of sub-packets and generates redundant sub-packets for each packet, to enhance the error recovery rate and video streaming quality. Based on the traffic condition, traffic load and significance of the video frame types, the required number of sub-packets is generated. One-Dimensional (1D) interleaved parity codes named as Exclusive OR (XOR) FEC codes are applied for handling the burst loss problem in the packet transmission over the wireless network. Our proposed scheme provides better error recovery rate by applying the Automatic Repeat Request to retrieve the lost packets. This avoids the network congestion problem. The experimental analysis shows that the proposed scheme yields better decodable frame rate and minimum packet error rate than the existing FEC schemes.

Keywords—*Adaptive Forward Error Correction, Automatic Repeat Request, Interleaving Parity Codes, Packet Error Rate, Wireless Sensor Network*

I. INTRODUCTION

Due to the rapid development of mobile communication technology, more people are attracted to relish the high quality video streaming services on the smartphones and tablets [1]. While accessing the video streams through the Third Generation (3G)/Fourth Generation (4G) mobile networks, mobile users have to wait for long buffering delays due to the bandwidth variations and fluctuations in the network link caused by the multi-path fading and mobility of the user [2]. Hence, it is vital to improve the video streaming quality while efficiently utilizing the networking and computing resources [3]. Packet loss in the network creates a substantial impact on the perceptual quality of the video streams. Error control techniques are applied to reduce the packet loss and transmission delays due to the network congestion, without any support from the network infrastructure.

Error control techniques employ Forward Error Correction (FEC) mechanism in which the redundant information is added to the data stream to facilitate the reconstruction of the stream during the packet loss [4]. FEC deals with the streaming video errors based on the usage of redundant data transmitted along with the original data. The original data stream can be reconstructed at the receiver end even though some data streams are lost during the transmission. Due to the introduction of redundant sub-packets, the length of the packet is increased and this increases the energy consumption and latency. The main problem of the existing FEC mechanisms is the network overloading due to the high redundancy rate that increases the collisions and interferences and burst errors.

This paper describes about the IARQ-EASP-FEC, by allowing the sender and relay user terminals to calculate the redundant sub-packets based on the network conditions such as effective packet error rate, queue length and types of video frames. The objective of the proposed work is to increase the efficiency of data recovery and guarantee a high quality video streaming by eliminating the network congestion. ARQ is a simple and convenient tool to reduce the packet errors. Poor channel quality in the wireless network often leads to retransmissions that are controlled by the ARQ schemes. Data retransmission leads to the increase in the energy consumption and latency. The use of FEC with the ARQ scheme offers numerous advantages. The performance of the EASP-FEC is improved by using the interleaving technique. 1D interleaved parity codes are highly suitable for protecting the large size packets. When compared with the PL-FEC codes, 1D interleaved parity codes can reduce the complexity of encoding and decoding operations and can save a lot of computing power. Processing power of the parity codes increases linearly with the increase in the size of the packet. 1D interleaved parity codes recovered more packets than the Packet Level FEC (PL-FEC) scheme [5].

The manuscript is systematized in the following order: Section II describes about the existing FEC mechanisms for improving the video streaming quality. Section III outlines the proposed work including network model, 1D-interleaving, ARQ and EASP-FEC for video streaming in the cloud. The simulated study and performance evaluation of the proposed IARQ-EASP-FEC are explained in Section IV. Finally, the concluding remarks and advantages of the proposed work are discussed in Section V.

II. RELATED WORKS

Many variations of FEC mechanisms such as PL-FEC) scheme [5], Forward Looking FEC (FL-FEC) [6], Enhanced Random Early Detection FEC (ERED-FEC) [7], Adaptive and Interleaving FEC (AIFEC) [8], FEC with Path Interleaving (FEC-PI) [9] are proposed to deal with the video streaming quality issues in wireless network. These FEC mechanisms dealt with the burst errors occurring at the receiver end. But, these schemes suffer from the limitations such as long delay and high redundancy rate. A Sub-Packet Forward Error Correction (SPFEC) [10] is proposed to improve the video streaming quality in terms of data recovery performance and jitter when compared with the traditional FEC mechanisms. SPFEC divides the video packet into 'n' sub-packets. FEC mechanism is applied on these sub-packets. SPFEC could be improved by adding traffic load control in the FEC to avoid network congestion. SPFEC mechanism outperforms PFEC as SPFEC reduces the Effective Packet Error Rate (EPER), redundancy overhead and end-to-end delay as the sender generates only one packet to perform the FEC mechanism.

Wu et al. [11] investigated the cross-layer design of FEC schemes using the Unequal Error Protection (UEP) Raptor codes and UEP Rate Compatible Punctured Convolutional (RCPC) codes at the Application Layer (AL) and physical layer (PHY) for prioritized video packets. A Genetic Algorithm (GA)-based optimization algorithm is proposed to find the optimal parameters for both codes. This reduces the video distortion and improves the Peak Signal-to-Noise-Ratio (PSNR). As the proposed design uses two different time-scales at the application and physical layers, the physical layer can adapt quicker to the varying channel quality. However, the performance of the proposed design degrades more for faster mobile velocity because reliable channel estimation becomes difficult when faster variations are introduced in the radio channel.

Alotaibi [12] proposed a new multi-path FEC control scheme with path interleaving to defeat burst loss problems in a multi-path communication environment for video streaming. The transmission order of the FEC blocks is changed and the path interleaving technique is applied to send these blocks. The video streaming quality is improved due to the higher PNR and a lower Packet Loss Rate (PLR) for high Burst Length (BL). But, it is not applicable in the real-time environment.

Kuo et al. [13] derived an analytical model to evaluate the video delivery performance of the frame-level FEC scheme over the burst-loss channels. This model provides the optimal FEC solution and channel parameters based on the uniform loss process. The proposed model can yield better Playable Frame Rate (PFR) than the FEC process. However, it does not adapt with the dynamically varying network scenarios.

Kuo et al. [14] developed unequal error protection system based on the FEC technique for effective video streaming over the bursty channels. The proposed model can provide a more accurate tool for video streaming and help to evaluate the impact of FEC performance on different burst-loss parameters. However, the complexity of the proposed model is too high to apply to the high-quality video due to the inclusion of a larger data source.

III. PROPOSED WORK

A. *Network model*

The network model comprises three user terminals (N_1, N_2, N_3) that are located at equidistant from each other and connected to a Gateway (GW). Fig.1 shows the network model. Each user terminal can relay the packets in a multi-hopping way. The packets are sent to the GW from the user terminal N_3 that are located farthest away either through the single-hop or multi-hop way. There exist three different routes H_1, H_2, H_3 to the GW through three hops. The GW is responsible for error control including routing. Let us assume that the user terminals in the network to be updated with current error control information from the GW through the periodic synchronization packets. Based on the Channel State Information (CSI), the GW decides on the route and codes for the transmission. The GW also communicates this information to the respective user terminals. The channel quality may vary due to the positioning of user terminals and distance between user terminals and GW. To estimate variation in the Signal to Noise Ratio (SNR) between the hops, a path loss model that approximates the effects of signal propagation through the wireless channel is used. The standard path loss model is described as

$$P_r = P_t K \left(\frac{d_0}{d}\right)^{\gamma} \quad (1)$$

Where P_r and P_t denote the received and transmitting powers respectively. γ denotes the path loss exponent, K indicates a constant and d_0 shows the reference distance [15]. K, d_0 *and* γ are same for all hops. The received power P_{r3} after transmitting over a short hop h_3. The received power over the longer hops h_1 *or* h_2 is expressed as

$$P_{rj} = P_{r3}(\delta_j)^{\gamma} \quad (2)$$

Where δ_j denotes the relative hop distance over h_j with respect to the short hop h_3. $\delta_j = \frac{d_3}{d_j}, where\ j = 1,2.$ d_j denotes the distance over hop h_j [16].

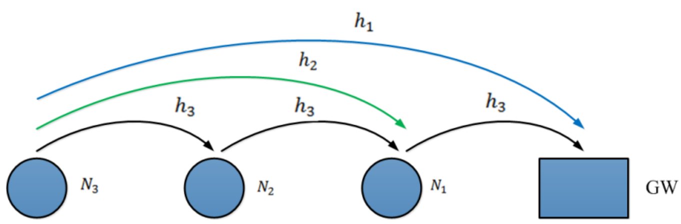

Fig.1 Network model

B. EASP-FEC

SPFEC [10] divides the video packets into 'n' number of sub-packets and applies the FEC mechanism on these packets. The redundant sub-packets are calculated according to the effective packet error rate in the network. An Enhanced Adaptive SPFEC (EASP-FEC) [17] which is an improved version of SPFEC is applied to avoid the network congestion problem and increase the video streaming quality.

EASP-FEC comprises three components such as traffic condition estimator, FEC redundant sub-packet generator and traffic load monitor. EASP-FEC is applied at the sender level and relays the user terminals in the network. When the relay user terminals received the packet, the current Bit Error Rate (BER) and Sub-Packet Error Rate (SPER) are estimated. The redundant sub-packets for the received packet are generated based on the SPER that provides maximum video quality in terms of Decodable Frame Rate (DFR) at the next relay user device. The traffic load monitor adjusts the number of generated redundant sub-packets according to the network load to avoid the congestion problem. High quality of video streaming for the relay user device in the next hop is allowed according to the significance of video sub-packets (I, P, B). The source terminal applies EASP-FEC mechanism and transmits the packet to the relay user terminal in the next hop. When a relay user terminal received the packet, the redundant sub-packets are recalculated and the packet is transmitted to the next hop, until the packet arrives to the destination terminal. The redundant sub-packets are regenerated at each hop. Fig.2 shows the architecture of the EASP-FEC mechanism.

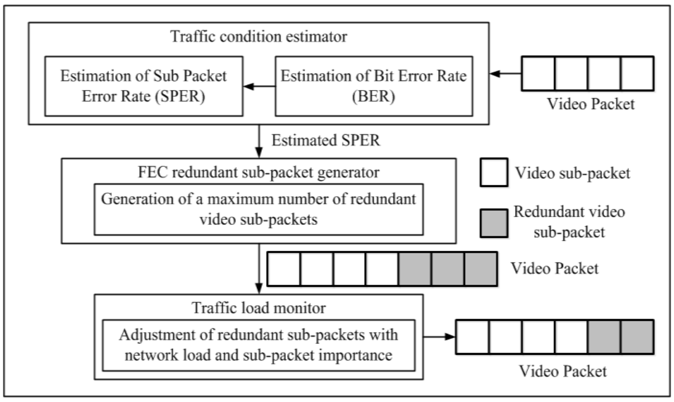

Fig.2 Architecture of EASP-FEC mechanism

C. Analytical model of EASP-FEC

The Effective PER (EPER) is estimated based on the SPER of video streaming in the network. SPER is estimated based on the BER. Let us assume that the video stream comprising 'N' number of video packets whose maximum size is 'n' bit. 'm' denotes the total number of sub-packets contained in a video packet.

BER

The BER is estimated at each time interval 'dt' using the following equation

$$BER(dt) = 1 - \left(1 - \frac{success(dt)}{Total(dt)}\right)^{\frac{1}{Total(dt)}} \quad (3)$$

Where $success(dt)$ denotes the number of successful received packets without FEC mechanism to current relay terminal during the time interval 'dt'. $Total(dt)$ indicates the total number of transmitted packets to the relay terminal during the time interval 'dt'.

SPER

SPER indicates the non-recoverable probability of a sub-packet. It is defined as

$$SPER = 1 - (1 - BER)^{\frac{n}{m}} \quad (4)$$

EPER

EPER denotes the non-recoverable probability of a packet. It is described as

$$EPER = 1 - \left(\sum_{i=k}^{k+h} C_i^{k+h} * (1 - SPER)^i * SPER^{k+h-i}\right) \quad (5)$$

Where 'k' denotes the number of original sub-packets and 'h' represents the number of redundant sub-packets.

D. Estimation of video streaming quality

DFR model is used to estimate the quality of video streaming at the next hop according to the EPER. DFR evaluates the quality of GoP at the application layer and provides highly accurate evaluation than the EPER. DFR model computes the number of decodable frames Intra-coded (I), Predicted (P) and Bi-directional predicted (B) frames in a given EPER. The DFR value varies from zero to one. Higher value of DFR indicates the best quality of video streaming in the network. The number of decodable frames 'I' is given as

$$NdecodeI = (1 - EPER)^{aI} * NGoP \quad (6)$$

Where $NGoP$ shows the total number of GoPs in the video stream and aI represents the average packet number in 'I' frame. The number of decodable frames 'P' is defined as

$$NdecodeP = (1 - EPER)^{aI} * \sum_{i=1}^{nP}(1 - EPER)^{i*aP} * NGoP \quad (7)$$

Where nP shows the total number of 'P' frames in a GoP and aP represents the average packet number in 'P' frame. The number of decodable frames 'B' is defined as

$$NdecodeB = \left[(1 - EPER)^{aI*nP*aP} * \sum_{j=1}^{nP}(1 - EPER)^{j*aP} * (1 - EPER)^{aB}\right] * (M - 1) * (1 - EPER)^{aI+aB} * NGoP \quad (8)$$

Where aB denotes the average packet number in B frame and 'M' represents the distance between 'I' and 'P' frames in a GoP. The ratio of the total number of decodable video frames 'I', 'P' and 'B' at the next relay terminal is given as

$$DFR = \frac{(NdecodeI + NdecodeP + NdecodeB)}{Ntotal} \quad (9)$$

E. EASP-FEC

When a new packet arrived at the relay terminal, the current BER and SPER are estimated and maximum number of redundant sub-packets is recalculated to improve the video streaming quality in terms of DFR at the next hop. The number of redundant sub-packets is adjusted to avoid the network congestion according to the network load. This resulted in the increase in the PER and end-to-end (E2E) delay in the network, if the density of the user terminals is high. In the EASP-FEC mechanism, the relay terminal estimates the network load based on the length of queue in the user terminal. The queue of the current user terminals contains two threshold levels referred as Low Threshold (LT) and High Threshold (HT).

- If the queue length (Qlength) is greater than the HT, the current network load is high and adjusted number of redundant sub-packets is set to zero.
- When the (Qlength) is lesser than the LT, the current network load is low. Then, the number of redundant sub-packets is set to the maximum number of redundant sub-packets.
- When the Qlength is between LT and HT, the adjusted number of redundant sub-packets is computed based on the following equation

$$No\ of\ redundant\ subpackets = Maximum\ no\ of\ redundant\ subpackets * \frac{(HT-Qlength)}{(HT-LT)} \quad (10)$$

EASP-FEC is based on the concept of asymmetrical protection of the sub-packets, such that the most important sub-packets are highly protected. Asymmetrical protection is specified by the dynamic update of the LT and HT according to the type of redundant sub-packets. The HT for the highly significant sub-packets is higher than the least significant sub-packets. When the network is congested heavily, the redundant sub-packets 'B' and 'P' are set to zero and redundant sub-packet 'I' is adjusted. Unlike HT, LT for all types of sub-packets is same. In the lightly congested network, the maximum number of redundant sub-packets should be transmitted to allow a high quality video for the user terminal in the next hop.

EASP-FEC

Step 1: Estimate the current BER and SPER using eqns (1 and 3)

Step 2: Determine the number of redundant sub-packets (h) in accordance with the network conditions. This provides maximum video streaming quality (MaxDFR) at the next hop

$h \leftarrow 0$

while $DFR < MaxDFR$ **do**

Determine EPER, number of decodable I-frames, P-frames and B-frames and DFR using eqns (2, 4, 5, 6 and 7)

$h \leftarrow h + 1$

$Maximum\ number\ of\ redundant\ sub-packets \leftarrow h$

end while

Step 3: Adjust the number of redundant sub-packets in accordance with the network load

if $Qlength < LT$ **then**

$\text{Number of redundant sub} - \text{packets} \leftarrow \text{Maximum number of redundant sub} - \text{packets}$

else if $Qlength < HT$ **then**

$\text{Number of redundant sub} - \text{packets} \leftarrow \text{Maximum number of redundant sub} - \text{packets} * \left(\frac{HT - Qlength}{HT - LT}\right)$

else

$\text{Number of redundant sub} - \text{packets} \leftarrow 0$

end if

F. 1D-Interleaving

The performance of the EASP-FEC is improved by applying the 1D-interleaving technique [5] to avoid the burst errors of video. There are a group of $(n-1)$ data packets $P_j, where\ j = 0,1,2,\ldots,n-2$ and a single redundant parity packet P_{n-1}. The packet P_{n-1} is formed by the Exclusive OR (XOR) operations between $(n-1)$ data packets. This group of packets is called as XOR group. In the XOR group, if the data in a single packet is damaged, it can be recovered from other packets. Interleaving technology is utilized by the 1D interleaved parity codes to deal with the burst of packets lost over the networks.

The encoder and decoder of the 1D interleaved parity codes use the XOR groups to form columns in each interleaving window called as transmitting window and receiving window respectively. One column is one XOR group. The encoder transmits the packets row by row and the decoder receives the packets row by row in the transmitting window and receiving window respectively.

The data structure of a transmitting and receiving window with 4 rows and 4 columns is shown in Fig.2.

$$\begin{bmatrix} P_{00} & P_{01} & P_{02} & P_{03} \\ P_{10} & P_{11} & P_{12} & P_{13} \\ P_{20} & P_{21} & P_{22} & P_{23} \\ P_{30} & P_{31} & P_{32} & P_{33} \end{bmatrix} \quad (11)$$

The encoding and decoding algorithms are described as follows

FX function: Let $(P_j)_i$ represents the i[th] bit in the packet $P_j, where\ j = 0,1,2,\ldots,n-1$. Then, $FX(P_0, P_1, \ldots, P_{n-1}) = (P_0)_i \oplus (P_1)_i \oplus \ldots \oplus (P_{n-1})_i, where\ i = 1,2,3,\ldots,L$. 'L' represents the number of bits in a packet.

Encoding algorithm of a window of packets (Transmission window size: $M \times N$)

Input: $M-1$ rows of data packets in a single transmitting window

Output: Single window of encoded packets

$for\ j = 0\ to\ N-1\ do$

{

$P_{(M-1,j)} = \{0\};$

//All bits in the redundant packet except the packet header are set to 0.

/*Encoding single XOR group[j]*/

$for\ i = 0\ to\ M-2\ do$

{

$FX\left(P_{(i,j)}, P_{(M-1,j)}\right);$

}

//end for i;

}

//end for j

Each source data packet can be decoded by $FX(P_{(i,j)}, P_{(M-1,j)})$; A group of bytes in a data packet is encoded with $O(1)$ operation. Hence, the encoding rates can be high.

Decoding algorithm of a window of packets (Receiving window size: $M \times N$)

Input: Single receiving window in which some packets are lost
Output: Single receiving window in which none or all lost data packets are recovered
$for\ j = 0\ to\ N - 1\ do$
{
/*Decoding single XOR group[j]*/
if $(P_{(M-1,j)})\ is\ not\ lost, then$
{
//If the redundant packet is not lost.
$i = 0; badNum = 0;$
While $(i < (M-1))$ and $(badNum < 2)$ do
{
//While the number of lost source data packets in the XOR group is not beyond one
if $P_{(i,j)}\ is\ not\ lost, then$
{
$badNum = badNum + 1;$
//Record the number of lost source data packets
$I = i;$
//Record the location of the lost source data packet
};
//end if
$i = i + 1;$
};
//end while
if $(badNum == 1)$ then
{
//If there is one lost source data packet
$P_{(I,j)} = \{0\};$
$for\ i = 0\ to\ M - 1\ do$
{
$FX(P_{(I,j)}, P_{(i,j)}), (i \neq I);$
//recover it
}
//end for i;
}
//end if;
}
//end if;
}
//end for j

After the decoder receives single XOR group, a damaged packet can be recovered. One group of bytes in a data packet can be recovered with $O(k)$ operations. Hence, the decoding rates can be high. The decoding algorithm can produce high decoding rates. Hence, the decoder only performs the decoding operations in the XOR group that only includes a single lost source data packet. If a single XOR group includes none of the lost data packet or more than one lost data packet, the decoding operations will not be performed in the XOR group. This scheme significantly increases the decoding rates. Thus, the decoding algorithm is energy efficient.

IV. PERFORMANCE ANALYSIS

The effectiveness of the IARQ-EASP-FEC mechanism is validated through a set of simulations conducted using the Java IDE environment. The results are compared with the EASP-FEC and SPFEC mechanisms in terms of EPER, DFR and number of redundant sub-packets. The IARQ-EASP-FEC mechanism is applied at the sender and relay the user terminals in a network. Table 1 shows the parameter settings of the simulation experiment.

TABLE 1 PARAMETER SETTINGS OF THE SIMULATION EXPERIMENT

Parameter	Values
BER	{0,….,0.005}
Packet size	1000 bits
Number of sub-packets in a packet	10
Number of original sub-packets in a packet	8
Number of GoPs in the video stream	10
Average packet number in 'I, P and B' frames	10
Maximum desired DFR of video stream	1
Total number of P frames in a GOP	2
Queue length	{0 to 50} packets
LT	10
HT	25

The validation of the IARQ-EASP-FEC traffic condition estimation and effect of the EPER on the video quality is done by comparing the EPER and DFR for the PFEC, EASP-FEC and proposed IARQ-EASP-FEC. Fig.3 shows the variation of EPER with BER. Table 2 shows the variation of the EPER with BER. The EPER of the PFEC increases greatly than the EASP-FEC and proposed IARQ-EASP-FEC with a varying BER, as the sub-packet is more vulnerable to be recovered than the entire packet, in the same network condition. Fig.4 depicts the variation of the DFR for PFEC, EASP-FEC and IARQ-EASP-FEC. Table 3 presents the variation of the DFR with BER. When there is an increase in the BER with the same redundancy rate, the DFR of the PFEC decreases than the EASP-FEC and IARQ-EASP-FEC.

TABLE 2 VARIATION OF EPER WITH BER

BER	EASP-FEC	Packet-FEC	IARQ-EASP-FEC
0	0	0	0
1	0.02	0.2	0.01
2	0.12	0.4	0.09
3	0.2	0.6	0.12
4	0.4	0.8	0.35
5	0.56	0.9	0.4

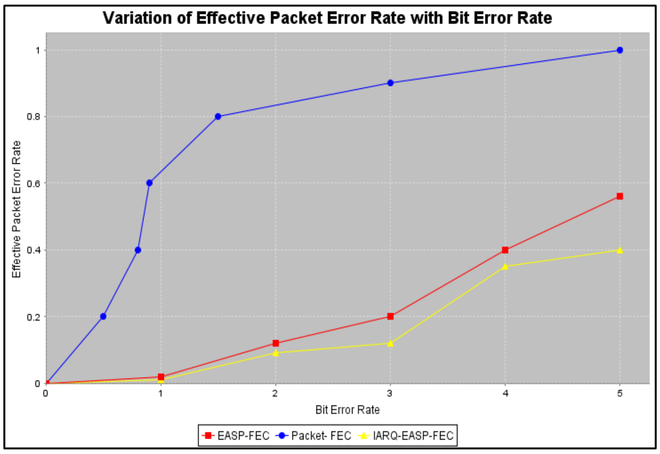

Fig.3 Variation of EPER with BER

TABLE 3 VARIATION OF DFR WITH BER

BER	EASP-FEC	Packet-FEC	IARQ-EASP-FEC
0	1	1	1.2
0.001	0.9	0.15	1.0
0.002	0.2	0.01	0.5
0.003	0.15	0.01	0.18
0.004	0.01	0.01	0.05
0.005	0.01	0.01	0.05

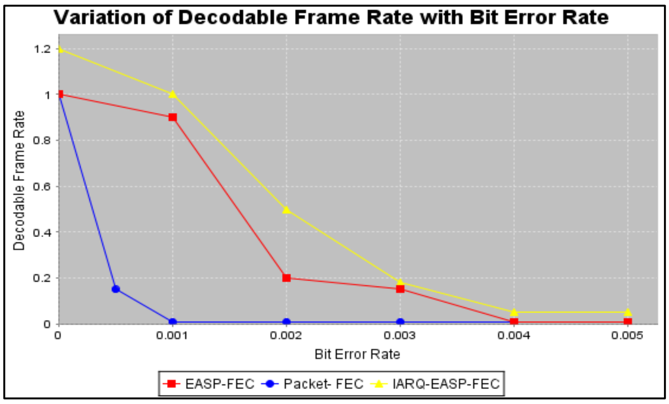

Fig.4 Variation of DFR with BER

To verify the effectiveness of redundant sub-packets used to recover the original sub-packets, the EPER is compared with the variation of the redundancy rate of the PFEC, EASP-FEC and proposed IARQ-EASP-FEC schemes. The redundancy rate represents the ratio of the redundant sub-packets to the original sub-packets. As shown in Fig.5, the EPER of the proposed IARQ-EASP-FEC decreases significantly than the EASP-FEC and PFEC schemes. Table 4 depicts the variation of EPER with redundancy rate. In the PFEC scheme, the packet is dropped, even if one part of the packet is found to be erroneous, whereas in the proposed scheme, the sub-packet containing the error part is dropped. Also, the lost packets can be retrieved using the ARQ scheme. When there is an increase in the BER value, the EPER decreases with the increase in the redundancy rate. The redundancy rate should be high to recover the erroneous bits. Fig.6 shows the variation in the DFR of video stream with the redundancy rate. Table 5 shows the variation of the DFR with respect to the redundancy rate. The graph shows that the DFR increases with the increase in the redundancy rate. The DFR of the proposed mechanism is higher than the PFEC and EASP-FEC.

TABLE 4 VARIATION OF EPER WITH REDUNDANCY RATE

Redundancy rate	EASP-FEC	Packet-FEC	IARQ-EASP-FEC
0	0.84	1	0.71
15	0.85	1	0.70
25	0.8	1	0.6
50	0.57	1	0.43
75	0.37	1	0.2
100	0.19	1	0.12

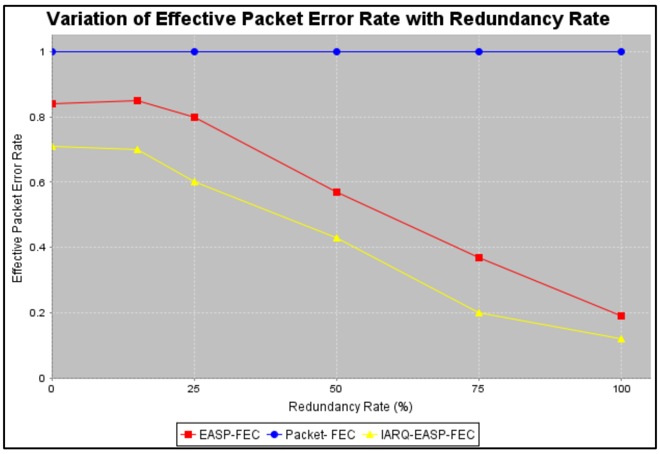

Fig.5 Variation of EPER with redundancy rate

TABLE 5 VARIATION OF DFR WITH REDUNDANCY RATE

Redundancy rate	EASP-FEC	Packet-FEC	IARQ-EASP-FEC
0	0	0	0
15	0	0	0
25	0	0	0
50	0	0	0
75	0	0	0
100	0.13	0	0.17

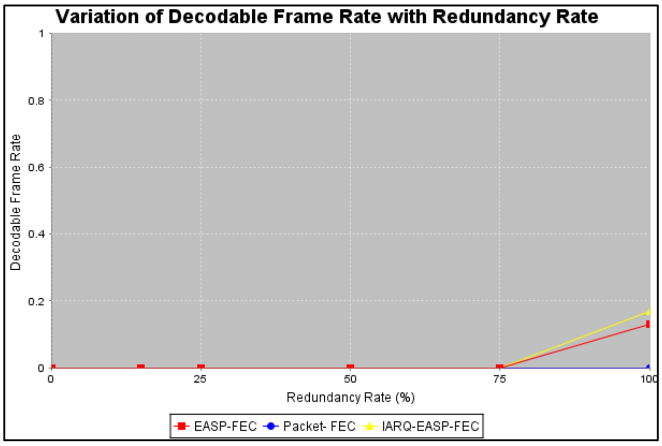

Fig.6 Variation of DFR with redundancy rate

Fig. 7 shows the graph depicting the variation in the number of redundant sub-packets corresponding to the length of queue. Table 6 presents the variation in the adjusted number of redundant sub-packets with the queue length. The number of redundant sub-packets is generated in a random way before adjusting the redundant sub-packets with respect to the queue length. When the queue length is lower than LT, the density of user terminals is low and adjusted number of redundant sub-packets of the proposed IARQ-EASP-FEC scheme, EASP-FEC and PFEC schemes is same. When the queue length is between the LT and HT, the density of the user terminals is medium. Then, the adjusted number of redundant sub-packets for the SPFEC is higher than the EASP-FEC and proposed IARQ-EASP-FEC scheme, as SPFEC does not have a mechanism that controls the network congestion issue. When the queue length is higher than HT, the density of user terminals is high. The adjusted number of redundant sub-packets of the SPFEC is high when compared to the IARQ-EASP-FEC and EASP-FEC scheme which is equal to zero. This is due to the detection of network load using the traffic load monitor. The generation of redundant sub-packets is stopped to avoid the congestion issues.

TABLE 6 VARIATION IN THE ADJUSTED NUMBER OF REDUNDANT SUB-PACKETS WITH QUEUE LENGTH

Queue length	Subpacket FEC	EASP-FEC	IARQ-EASP-FEC
0	0	0	0
5	0.5	0.5	0.4
10	0.3	0.3	0.2
15	4.8	3	2.8
20	7	2.2	0
25	0.8	0	0
30	4.8	0	0
35	8	0	0
40	3.8	0	0
45	6	0	0
50	8.8	0	0

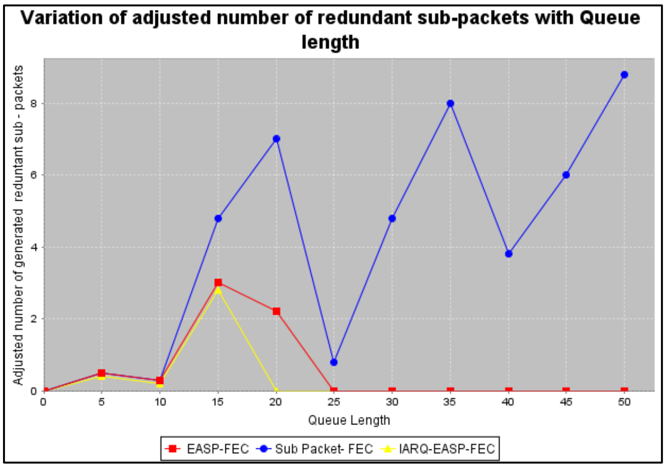

Fig.7 Variation in the adjusted number of redundant sub-packets with queue length

TABLE 7 VARIATION IN NUMBER OF DECODABLE FRAMES WITH BER

BER	Frames I	Frames P	Frames B
0	10	20	60
0.001	10	19	55
0.002	8	8	8
0.003	4	3.5	3
0.004	0	1	0
0.005	0	1	0

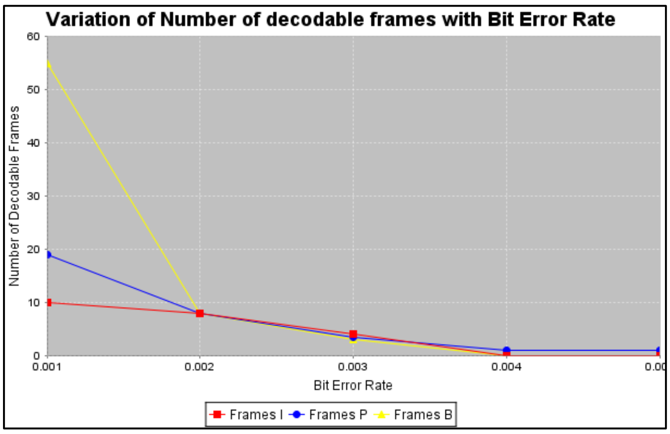

Fig.8 Variation in number of decodable frames with BER

Fig.8 shows the variation in number of decodable frames with BER. Table 7 depicts the variation in the number of decodable frames with BER. When the *BER* increases, the number of decodable frames *B* decreases greatly than the decodable frames *P* and this later decreases greatly than *I* frames. According to the dependencies between MPEG frames, error of *I* frame influences on the *P* and *B* frames and the error of *P* frame influences on the *B* frames. Hence, IARQ-EASP-FEC proposes to distinguish values of LT and HT for different types of video frame (*I*, *P*, *B*).

V. CONCLUSION

In this paper, a combined mechanism of IARQ-EASP-FEC is presented. Dissimilar to the PFEC that calculates the redundant packets for each block of packets, our proposed scheme calculates the redundant sub-packets for each packet allowing high error resiliency. Moreover, IARQ-EASP-FEC considers network load and the robustness of the video frame types to avoid the network congestion issue and increase the video streaming quality, when compared to the SPFEC that calculates the redundant sub-packets only in the basis of network condition. When compared with the PLFEC codes, 1D interleaved parity codes can reduce the complexity of the encoding and decoding operations. This can save a lot of computing power. 1D interleaved parity codes used minimum amount of redundant data than the PL-FEC codes. EASP-FEC is at the sender and relay user terminals to guarantee accurate estimation of the network condition and network load. Experimental results have shown that the IARQ-EASP-FEC yields higher DFR than the EASP-FEC and PFEC with the same redundancy rate. Our proposed scheme avoids the network congestion problem and improves the video streaming quality when compared to the SPFEC.

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