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An Efficient Video Transmission by Reducing the Packet Loss Using HEVC

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Abstract: to protect real-time video transmission over unreliable networks, packet-level forward error correction (fec) codes are widely studied. When the number of redundancy packets are determined, with fec coding block size the performance of fec coding improves. The decoding delay for a real-time video transmission cannot be tolerable and the packet loss in one frame affects not only the current frame, but all the other frames which is associated with that frame. Lesser bit error rate is achieved in the fec coding compared to the conventional coding. To reduce the video size, and get a better transmission without loss of packets which would give a better quality of video causing less delay during the video transmission, fec coding method is applied to a real-time video streaming scheme which is known as high efficiency video coding(hevc). In this scheme, there is not only improvement in the compression ratio but also in the quality of frame. No delay will be caused for waiting for the video or parity packets of the following frames both at encoding and decoding sides, since video packets of the frames are not encompassed in the hevc block. The performance is excellent, but it can still be improved when burst packet loss occurs. In this paper, a novel scheme is proposed by replacing randomized expanding reed-solomon (re-rs) scheme with hevc scheme to get improved compression ratio and good quality frames. Therefore packet loss is evaded in a video by reducing the number of frames and increasing its quality of video by applying fec coding to hevc instead of re-rs scheme which improves the compression ratio and causes less delay during video transmission. As a result, we simply get a good quality video with lesser frames and less delay during any video transmission.

Keywords: packet-level forward error correction (fec), redundancy packets, high efficiency video coding (hevc), randomized expanding reed-solomon (re-rs).

I. INTRODUCTION

The wireless networks for example wireless local area networks prominently 4G networks have been emerged as a growing technology. The packet loss rate, delay and the bandwidth are worst in wireless network than the wired ones. Packet loss pattern are burst than random in wireless network, which makes correcting errors a difficult task. Excessive packet loss rate can lower the video quality perceived by the users and the packet loss of current frame affects not only the decoding of the current frame but also the following frames which is associated with it. This phenomenon is error propagation. The distortion caused by this varies and it relays on the packet of lost and the content of video.The performance of FEC depends on the FEC coding block size. The larger the FEC coding block size, the better performance. But users are sensitive to delay in real-time video transmission applications, such as videoconference, video-telephone, tele-consultation and so on. As a result, the FEC coding block size usually equals the number of packets per frame, which makes the performance of FEC low. These are the typical characteristics of real-time FEC for video transmission.

To overcome this impairment of packet loss on video quality, an error control scheme has been introduced which is called Forward Error Correction (FEC). This scheme brings less delay when the FEC block is constrained in a few frames. Therefore, FEC is used widely in real-time applications.

FEC can be classified to bit-level schemes and packet-level schemes. Bit-level schemes are devoted to eliminating bit-errors in wireless cases. This scheme may be soft or hard decoded at the lower layers of the OSI stack. Packet-level schemes mitigate the packet-loss events, as exemplified by the packets lost in Internet-routers. It achieves the ability of correcting the lost packets by adding redundant packets into FEC coding block. It can correct the lost packets when the number of lost packets is less than or equal

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to the number of redundant packets. A simple packet-level scheme is adding the same number of redundant packets into each frame which is treated as a FEC coding block. The advantage of the scheme is simple and induces no additional delay. But the performance of the scheme is low during to the small FEC coding block size. To improve the performance, large numbers of modified schemes are proposed. Enlarging the FEC block size can improve the performance of FEC scheme. In [7], the FEC coding block includes the whole group of pictures (GOP) and one GOP of delay is caused. The disadvantage of the scheme is that it leads to additional delay which may not be tolerable for real-time video transmission.

To improve the performance of FEC and not lead to extra delay, the Dynamic Sub-GOP FEC Coding (DSGF) approach was proposed in [8,9]. This adds parity packets into each sub-GOP which makes the FEC coding block. To avoid delay, the DSGF corrects the lost packets at the end of each sub-GOP. When a packet loses in one frame which is not the last of the sub-GOP, the frame will be concealed and displayed first and then the current and following frames will be recovered when decoding the last frame of the sub-GOP. In this way there will be no extra delay introduced and the performance of this approach is high with large FEC coding block size. The FEC error correction capability can only be used by the last frame of each sub-GOP, the performance is lower and consequently the video quality can fluctuate.

II. SCHEMES TO OVERCOME THE CHALLENGES

A. Tetrys Scheme

To overcome the hardships occurred various scheme was introduced. Here the parity packets are generated using the video packets of the current and all the previous frames of the current GOP. The tetrys sender uses an elastic encoding window buffer which includes all the source packets sent and not yet acknowledged. For every k source packets, the sender sends a parity packet which is generated using all the previous packets of the current window buffer. When the number of lost packets is larger than one, tetrys could not recover the packets when receiving the first parity packet. Because the following parity packets are generated using all the previous packets of the current window buffer, the packets will be recovered after enough parity packets received. After re-decoding the previous frame, error propagation will be erased.

B. Re-Rs Scheme

RE-RS scheme adds equal parity packets into each frame which are generated using the video packets of the current and all the previous frames of the current GOP. At the encoder, there will be no delay introduced due to no packets of following frames used. In the decoder RE-RS scheme uses the parity packets of current frame to recover the lost packets. The frame will be concealed if the number of lost packets are larger than the number of parity packets of the current frame. But the lost packets will be recovered after enough parity packets are received. After re-decoding the previous frames concealed, error propagation will be eased. As a result, RE-RS gives an excellent performance with no delay introduced. The performance of this scheme is ideal when random packet loss occurs.

The performance of RE-RS scheme is improved by changing the parity packets position of it. Besides, the parity packets are generated using packets of current window buffer rather than of current and all previous frames of the current GOP.

C. hevc scheme (our scheme)

HEVC scheme is similar to the RE-RS scheme. The main different between this scheme and the RE-RS scheme is that the Redundant packets are in middle of the video packets rather than those are in end of the video packets. This scheme is described in Figure 1.

Here sender uses an elastic encoding window buffer which includes all the video packets sent and not yet acknowledged. The Redundant packets are generated using the video packets of the current window buffer. At the encoder, there will be no delay introduced due to no packets of following frames used. In decoder, RE-RS uses the parity packets of current frame to recover the lost packets.

The frame will be concealed if the number of lost packets larger than the number of parity packets of current frame. But the lost packets will be recovered after enough following parity packets received which lost packets are used to generate. After re-decoding the previous frames concealed, error propagation will be eased and we get an excellent performance similar to the RE-

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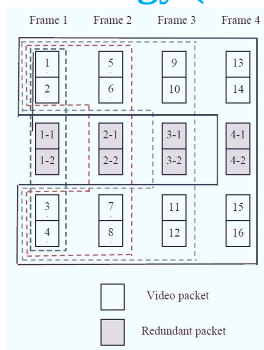

. Fig 1 : HEVC scheme where each frame has 4 video packets and redundant packet rate is 0.5

The detailed differences between HEVC and the RE-RS scheme are as follows.
Firstly, the Redundant packets of the RE-RS scheme is at the end of video packets, and the Redundant packets of the HEVC scheme is in the middle of video packets, which is the main reason of better performance of our scheme. Secondly, the Redundant packets are generated using the video packets of the current frame and all the pervious frames of the current GOP in the RE-RS scheme, but the Redundant packets are generated using the video packets of the current window buffer which is an elastic encoding window buffer and includes all the video packets sent and not yet acknowledged when there is a feed-back channel. When there is no feed-back channel, the redundant packets will be generated and simplified in the same way as the RE-RS scheme.
There are two methods to send the video packets and redundant packets of each frame at the video sender. One is sending the packets at the maximum rate. Another is sending the packets at uniform rate which is determined by the interval time between frames and the packets of each frame. We adopt the second method here. This is because we pay more attention to the performance of FEC schemes when the available bandwidth is a bottleneck, because it's easy to fulfill the user experience when the resource is unlimited and optimizing the performance in this moment will be not that meaningful. Secondly, is the performance of FEC schemes in burst packet loss condition. Thirdly, the decoder keeps a buffer of some frames to make the frames displayed fluently. So the delay of the frames displaying between the two methods is smaller than the delay caused by different sending rate.

III. HEVC SCHEME AT THE TRANSMITTER AND THE RECEIVER

Since HEVC scheme is used for real-time video transmission which is sensitive to delay and to simplify the problem, the Bframe will not be used, so the GOP structure is the IPPP. To make the code efficient, fixed length slice scheme will be used where each slice makes up a packet. In this way, other slices will not be influenced when one slice is lost. In practice, the length of the last slice of a frame may be shorter than that of previous slice. We can pad a few zero bytes at this moment. Then the slices are packetized using protocol UDP/RTP whose sequence number could help to determine whether there are packets lost at the video decoder. The redundant packets are generated using the previous method mentioned. In the end, the packets of each frame are sending at uniform rate.
The HEVC coding procedure at the video sender side works as following and is shown in the below figure 2:
Step 1: Redundancy packet rate, encoding window buffer, packet loss rate are determined.
Step 2: using equation (1) in the group of pictures (GOP) redundant packets are allocated for the current frame.

$$R(i) = \begin{cases} \lceil \mu S(1) \rceil & if\ i == 1 \\ \left\lceil \mu \sum_{k=1}^{i} S(i) \right\rceil - \sum_{k=1}^{i-1} R(i) & if\ k > 1 \end{cases} \quad (1)$$

Step 3: video packets are gathered. When the packet length is shorter than the target length the zero bytes are padded and as directed by the encoder the video packets are ordered as they are generated.
Step 4: There is a generation of redundant packets from all the packets of the current and previous frames at the current window buffer if there is a feedback channel. Otherwise, the redundant packets are generated using all the packets of current and previous frames of the current GOP.
Step 5: At the uniform rate all the packets of the current frame are sent.
Step 6: steps 2-5 are repeated.

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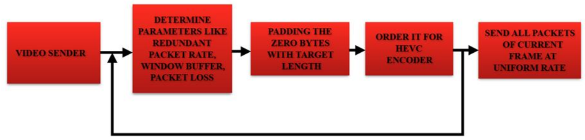

Fig 2 : Transmission process

At the receiver side, HEVC coding will works as following and is shown in the below figure 3:
Step 1: lost video packets or late video packets are denoted as Esum and the available redundant packets are denoted as Rsum which is used to recover the lost packets. ESum and RSum are initialed to zero.
Step 2: All the video packets and redundant packets of the current frame are received by the receiver and the lost packets or late packets are determined whose arrival times exceed the deadline according to the RTP protocol sequenced number.
Step 3: According to the number of packets lost or last and the number of redundant packets received of current frame the Esum and Rsum values are updated. The packets are recovered , the distorted frame is re-decoded, decode and display the current frame, and reset the values of ESum and RSum to zero, If the value of ESum is less than or equal to RSum . Otherwise, the current frame is concealed and display it.
Step 4: Window buffer value is updated when packets received is acknowledged, if there is a feedback channel.
Step 5: Steps 2-4 are repeated .

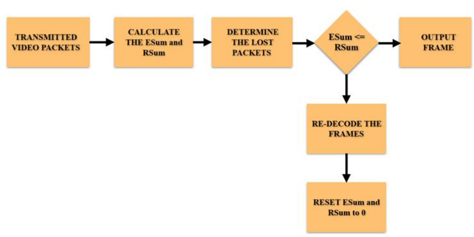

Fig 3 : Receiver process

IV. EXPERIMENTAL RESULTS

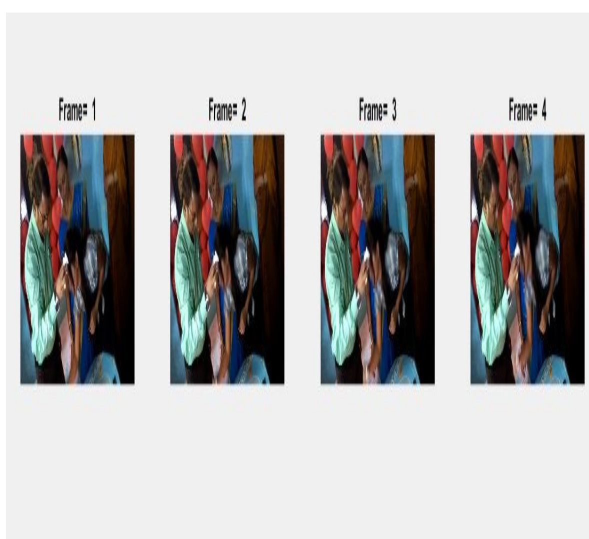

Fig 4 : Video Frames

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These frames have been generated from a video where the frames are compressed to give a better compression ratio. The redundant packets are acknowledged and the unnecessary packets are eliminated from the video to give a compressed video. This would result in lower bit error.

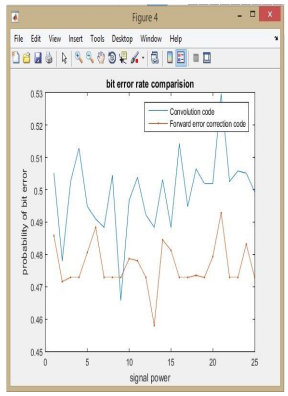

Fig 5 : Bit error rate comparison

The Frames have been compressed under two method that would result in a lesser bit error. The first technique is an conventional method and the second method is the forward error correction code. From the above figure it is proved that the use of forward error correction gives us a lesser bit error rate compared to the conventional method. Lesser bit error rate would result in reduced packet loss that would give a better video quality.

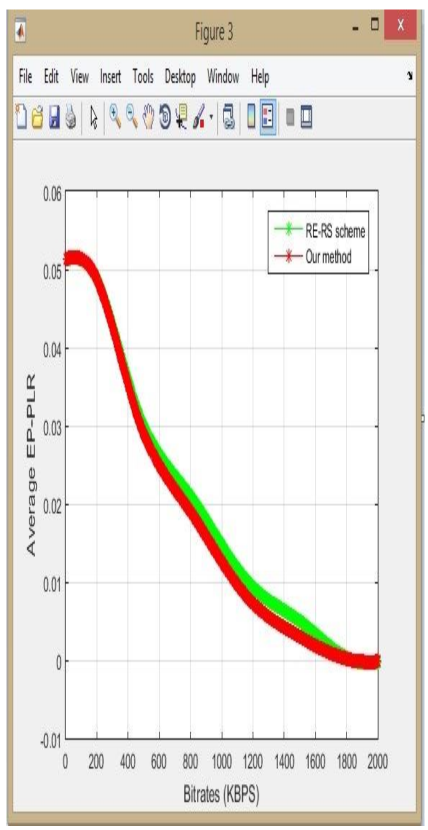

Fig 6 : comparison graph of RE-RS and HEVC scheme

As discussed before the lesser the bit error rate, the packet loss is reduced. Reduction in the packet loss would result in improving the quality of the video which ultimately improves the efficiency of the video during the video transmission. The above figure gives us the comparison between the performance of the two scheme discussed before that is the RE-RS scheme and the HEVC scheme. In the graph the bit error rate and packet loss rate have been compared and The curves of the two schemes are almost superposition together. But it is proved that the HEVC scheme gives us better results compared to the previous scheme used and using HEVC scheme gives us excellent compression ratio which ultimately improves the efficiency of he video during video transmission.

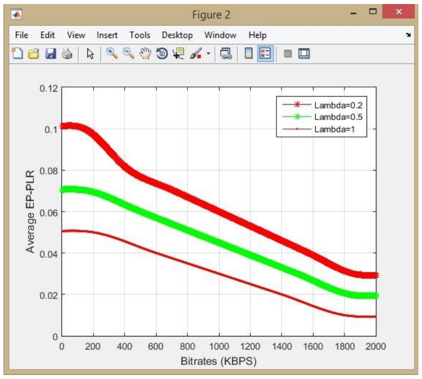

Fig 7 : error rate for various lama values

After the scheme has been applied and the frames are received at receiver in order to eliminate the error which is still present the values of lamda are inserted at the receiver to give error free frames that would make the video more efficient and accurate. The above figure give us the comparison of various values of lamda that would be accurate to be inserted along the receiver for the video to be more efficient. As in the graph it is seen that the lamda = 1 gives us a less error compared to the other values and this is how this value is inserted at the receiver.

V. CONCLUSION

Finally , The scheme is simple and easy when it is to be implemented using HEVC. Experimental results show that the scheme improves quality of frames and also improve compression ratio without packet loss by using HEVC method which reduces packet loss with burst length. And in the condition with random packets loss, the performance of this method is almost improved.

VI. ACKNOWLEDGMENTS

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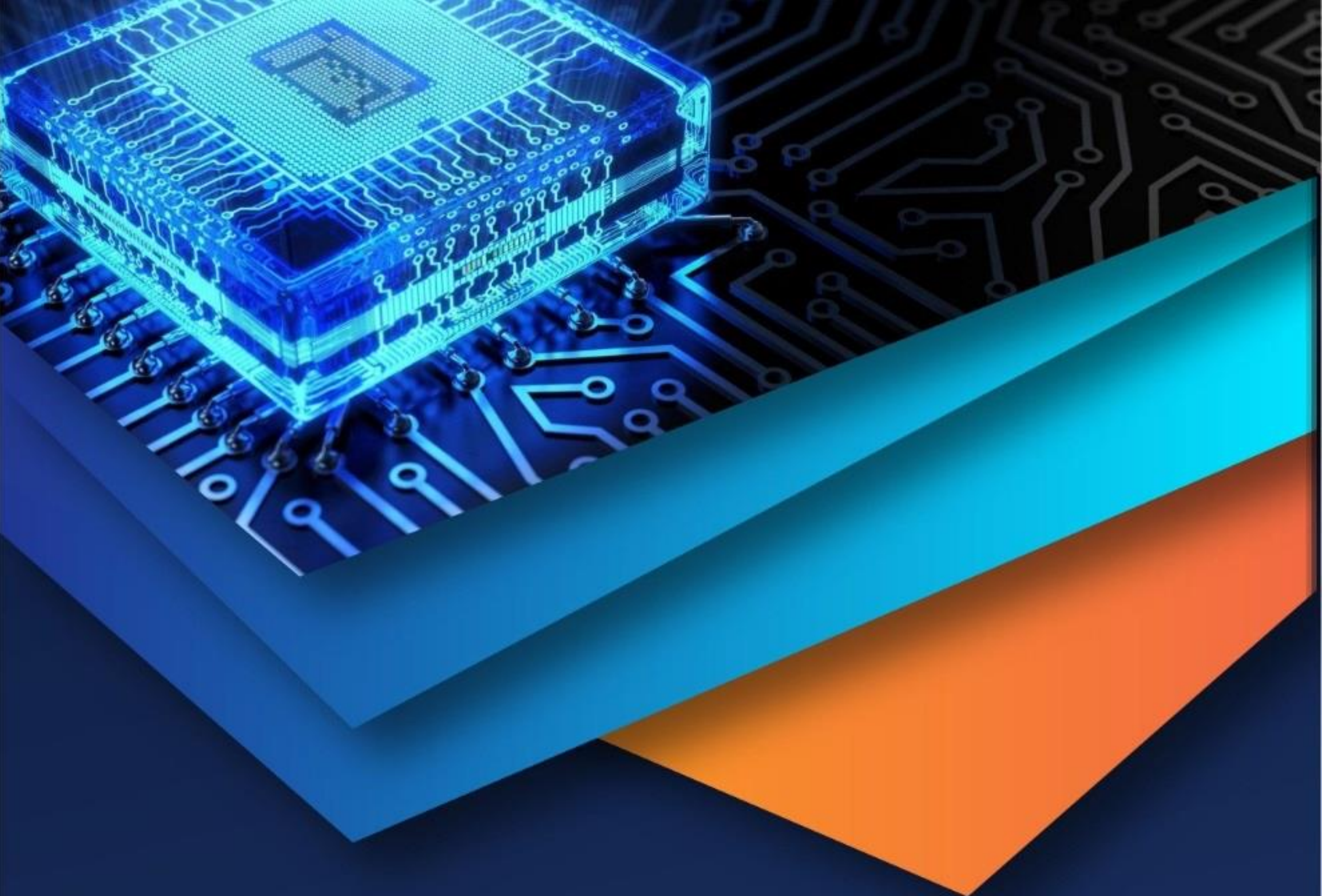

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