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EVADING PACKET LOSS DURING VIDEO TRANSMISSION USING HEVC CODING

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Abstract

Packet-level forward error correction (FEC) codes are widely studied to protect real-time video transmission over unreliable networks. The performance of FEC improves with the FEC coding block size when the number of redundancy packets is determined. But for real-time video transmission, the decoding delay which it leads to cannot be tolerable. Besides, the packet loss in one frame affects not only the current frame, but could also include all the following frames which refer to it. To solve this dilemma, real-time video streaming scheme using HEVC code was proposed. In this scheme, the HEVC coding not only improves compression ratio but also improve quality of frame. Since video packets of the following frames are not encompassed in the HEVC coding block, no delay will be caused for waiting for the video or parity packets of the following frames both at encoding and decoding sides. The performance of the scheme is excellent, but it can still be improved when burst packet loss occurs. In this paper, we proposed a novel scheme by replacing AVC with HEVC encoder to get improved compression ration and good quality frames. As a result, we can simply replace HEVC which get improved quality of frames.

Keywords : *Forward Error Correction (FEC), Redundancy Packets, HEVC Coding.*

Objective

The objective is to evade packet loss in a video by increasing its quality of video by applying HEVC encoding instead of AVC and also by improving the compression ratio.

1. Literature Survey

[1] "Error control techniques for interactive low-bit rate video transmission over the Internet" by Rhee I. Acm Sigcomm, Computer Communication Review, 1998, 28(4):290-301.

This paper the recent progress of digital signal processing (DSP) in the high-speed optical transmission system using single-carrier based multi-level modulation formats is highlighted. A novel DSP-based interference mitigation algorithm for the single-ended coherent receiver has also been included.

[2] "Error control and concealment for video communication" by Wang Y, Zhu Q F., Proceedings of the IEEE, 1998, 86(5):974-997.

This paper reviews the techniques that have been developed for error control and concealment. These techniques are described in three categories according to the roles that the encoder and decoder play in the underlying approaches. Forward error concealment includes methods that add redundancy at the source end to enhance error resilience of the coded bit streams. Last, interactive error concealment covers techniques that are dependent on a dialogue between the source and destination.

[3] "ARQ Protocols and Unidirectional Codes" by Anantha M, Bose B, Tallini L G., Computers IEEE Transactions on, 2007, 56(4):433-443.

Forward error control (FEC) and automatic-repeat request (ARQ) is two main techniques used for reliable data transmission in computer and communication systems. In this paper, some simple, low cost error control techniques for ARQ protocols

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used with binary unidirectional channels are described. The proposed schemes can correct up to [t/2] unidirectional errors using t-unidirectional error detecting codes and code combining with a much smaller number of retransmissions.

[4] "A Real-time Forward Error Correction of Changing Redundant Packet's Position for Video Transmission" by Guobo Zhang, Xuan Zhang, Chongrong Li, Guangxing Han, Transactions 978-1-5090-5521-0/16/$31.00 ©2016 IEEE

When the video streams are transmitted over the unreliable networks, forward error correction (FEC) codes are usually used to protect them. Reed-Solomon codes are block-based FEC codes. On one hand, enlarging the block size can enhance the performance of the Reed- Solomon codes. On the other hand, large Reed-Solomon block size leads to long delay which is not tolerable for real-time video applications. In this paper an approach is proposed to improve the performance of Reed-Solomon codes. With the proposed approach, more than one video frame are encompassed in the Reed-Solomon coding block. Experimental results showed that the proposed approach had a grater compression ratio and encountered delay which world cause buffering of video streams and the efficiency of the video was not up to the mark.

1. Introduction

The wireless networks have achieved huge success in the past years for example Wireless local area networks (WLANs) and 4G networks and people use their Smartphone, notebooks and so on to access these wireless networks. Compared to wired networks, the packet loss rate, end-to-end delay, available bandwidth and other network metrics of wireless networks are much worse. Besides, the pattern packet loss of wireless networks is usually burst rather than random, which makes correcting the errors more difficult.

Excessive packet loss rates can dramatically decrease the video quality perceived by users and the packet loss of current frame affects the decoding of current frame, and all the following frames which refer to it. This is called error propagation (EP). The distortion caused by error propagation 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.

FEC can be classified to bit-level schemes and packet-level schemes. Bit-level schemes are devoted to eliminating bit-errors in wireless scenarios. This scheme may be soft or hard decoded at the lower layers of the OSI stack. 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, HEVC 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 current frame. But the lost packets will be recovered after enough parity packets received. After re-decoding the previous frames error propagation will be eased. As a result, HEVC coding reach an excellent performance with no delay introduced.

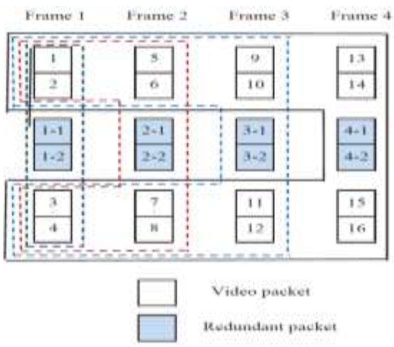

Figure 1 : HEVC coding where each frame has 4 video packets and redundant

2. HEVC coding at the transmitter and receiver

The HEVC coding procedure at the video sender side works as following:

Step 1: Determine the parameters such as the redundancy packet rate, window buffer, packet loss.
Step 2: The number of redundant packets is allocated for the current frame in the group of pictures (GOP) using the equation (1).

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

Step 3: All video packets used are collected. Pad the zero bytes when the length of the packet is shorter than the target length. And video packets are ordered as they are generated by the HEVC video encoder.

Step 4: If there is a feed-back channel, the redundant packets are generated using all the packets of the current and previous frames at the current window buffer. Otherwise, the redundant packets are generated using all the packets of current and previous frames of the current GOP.

Step 5: Send all the packets of the current frame at uniform rate.

Step 6: Repeat steps 2-5 process.

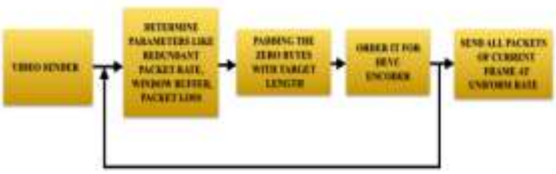

Figure 2 : Transmission Process

At the receiver side, HEVC coding will works as following:

Step 1: Let *ESum* denote the number of lost video packets or late video packets and *RSum* denote the available redundant packets which could used to recover the lost packets. Initialize *ESum* and *RSum* to zero.

Step 2: The receiver receives all the video packets and redundant packets of current frame and determines the lost packets or late packets whose arrival times exceed the deadline according the sequence number in protocol RTP.

Step 3: Update the values of *ESum* and *RSum* according the number of packets lost or last and the number of redundant packets received of current frame. If the value of *ESum* is less than or equal to *RSum*, then recover the packets, lost or late-arrival, re-decode the distorted frame, decode and display the current frame, and reset the values of *ESum* and *RSum* to zero. Otherwise, conceal the current frame and display it.

Step 4: Update the value of window buffer when packets received is acknowledged, if there is a feedback channel.

Step 5: Repeat the steps 2-4.

Figure 3: Receiver Process

3. Experimental Results and Discussion

In this section, we conduct a lot of experiments to compare the performance of the earlier used RS-RS coding technique and HEVC. We use the method of Monte Carlo to get the value of EP-PLR . We set the numb of frames as 500 and get the object of each experiment by averaging the object of 1000 trials. And we assume the MTU is 400Bytes [14].We generate the source packets of each frame and redundant packets using the current window buffer. Put the Redundant packets in right position according to the FEC scheme. Then, we drop the packets according to Gilbert model or Bernoulli model. After that we recover the lost packets using redundant packets and compute the value of EP-PLR .

Figure 4: Video frames reduced due to HEVC coding

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From the above figure it is evident that the unnecessary frames in a video are evaded using the HEVC coder and the unique frames are retained thus maintaining a good compression ratio. Through this method the size of the video also reduces to a greater extent and the bit error rate is also reduced.

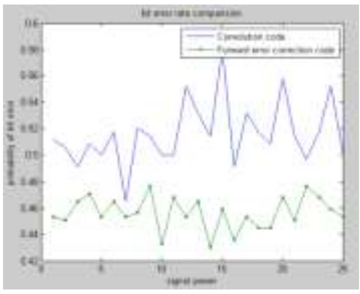

Figure 5: Bit error rate comparison

From the above figure it is quite evident that the bit error rate is lesser in forward error correction (FEC) coding than in convolution coding. Because of its lower bit rate, we have taken into consideration forward error correction in our method.

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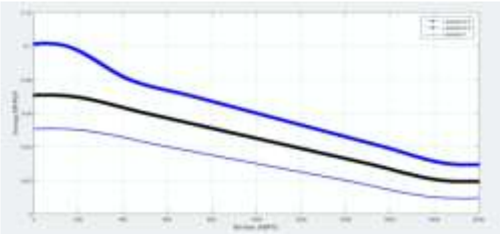
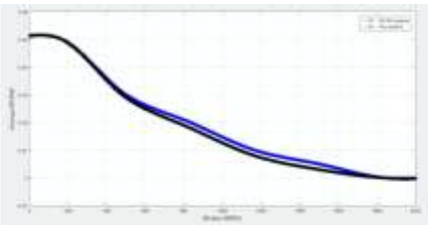


Figure 6: Average EP-PLR versus bit-rate for various lamda which includes {0.2,0.5,1.0}; average packet loss rate is 5%; average burst length is 2; the redundant packet rate is 0.3.

Figure 5 shows the average EP-PLR versus bit-rate for various lamda which includes {0.2,0.5,1.0}, the average packet loss rate is 5%, average burst length is 2, and the redundant packet rate is 0.3. The red curves represent the results of the earlier used RE-RS coding result and the blue curves represent the result of HEVC. We conduct this experiment in order to find how the values of lamda influence the experimental results. Divide the curves into three group where the values of lamda are the same. As we can see, they are consistent. So the metric of EP-PLR is insensitive to lamda .We then set lamda=1 and will not worry about other values of lamda bringing us a different experimental result. Xiao [14] points out that the appropriate redundant packet rate is $4P_L$ where P_L is the i.i.d average packet loss rate. And Zhang [22] also find that the redundant packet rat is 4.5 P_L . But the redundant packet rate should be increased in the burst packet loss condition. Here we set the redundant packet rate $M = 6P_L$ for the



HEVC.

Figure 7: Average EP-PLR versus bitrate curves;average burst length is 2; packet loss rate P_L ={5%,10%,15%, 20% } and redundant packet rate $M = 6P_L$

Figure 7 compare the performance of HEVC scheme and the previously used RE-RS coding scheme versus bit-rate when average burst length is 2. Figure 7 shows that HEVC outperforms the RE-RS scheme for various packet loss rate when average burst length is 2. The average EP-PLR of both RE-RS scheme and HEVC scheme decrease with the increasing of bit-rate. Note that the less the value of average EP-PLR , the better. The performance of the two scheme increases with the increasing of bit-rate. This phenomenon is consistent with our instinct. The number of packets for each frame will increase with the increasing of bitrate. And the FEC block size increases at the same time, which could increase the performance of the schemes. Besides, the difference of the two schemes becomes smaller when bit-rate increases. This is also easy to understand. Imagine the number of packets of each frame is so large that the packet loss rate could be compute by dividing the number of packets of each frame into the number of packet lost in the frame. Then, we only need to add little redundant packets whose number equals the number of packet lost. The performance arrival to the limitation and can never to be higher and the performance of all kinds of FEC schemes equals each other. So the difference of all kinds of FEC schemes decreases in the way to the limitation point.

4. 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.

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