Enhance Quality of Video Transmission Based on Improving the RED Algorithm

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Summary
Nowadays, application on Video transmission on Internet tends to be challenging in meeting the end users' demand. There have been numerous researches on improving video transmission quality when network congestion occurs. In this paper, we propose a control mechanism of the packets mark (drop) in the Random Early Detection (RED) active queue management. To enhance the quality of video transmission, we built a probability adjusting function of the packets mark (drop) based on buffer size in routers, network status. Our proposal is validated by NS-2 simulation, using objective video quality measurement peak signal to noise ratio (PSNR) (dB). The results of simulation experiment in IP network environment show that the video transmission quality has improved approximately the 6.9 percent (in average) compared with the original RED queue.

Key words:
RED, NS-2, Mpeg, Active Queue Management.

1. Introduction

With the fast growing of information technology transmission, multimedia applications (video, audio) on the Internet are widely applied in many in many areas of social life such as education, health and so on. The transmitting video applications online always require more resources (bandwidths), meet real-time requirements and the end-user requirements. Originally designed Internet network is not available to meet QoS [1, 2] requirements so that there are many encountered obstacles in the multiple or video streaming process, especially in the network which has financial dispute of multiple data streams. There are many methods to increase the quality of video transmission such as bandwidth increasing. However this solution caused expensiveness and resource wastage. An alternative method is to use the algorithm of the active queue management (AQM) [3] to solve the congestion which causes network quality degradation in general and network quality of video transmission services in particular.

One of the active management mechanism used to avoid congestion at the router is the RED [4, 5, 6] queue. RED is designed to early random detect congestion on the network, and to conduct to mark or remove packets to avoid congestion. The original RED is designed based on queue size in the buffer to mark or (drop) packets in the buffer of the router when congestion occurs without distinction of video packets with other packets. As a result, the video packets are randomly discarded, leading to image quality reduction in the receiving machine. We have analyzed the basic parameters of the RED algorithm to solve this problem. On that basis, we proposed a preference solution for video data integrated in the RED queue management mechanism, called ViRED, by integrating additional packet classification mechanism before the queue marking (drop) in an acceptable time. This improvement is both theoretically and practically significant when using ViRED mechanism this will be clearly evaluated in section 4. The left of the paper is as following: section 2 represents the effect of an information packet on the video transmission quality. Section 3 is about RED algorithm the video transmission quality. Section 4 proposes the improved ViRED algorithm based on RED by defining linear function adjusting the probability of marked (drop) packet with priority given to video packet. Section 5 shows the experimental results and simulation compared between RED and ViRED. Finally, the section 6 is the conclusion and further research proposal.

2. Effect of the packet loss on video transmission quality

2.1 Encoding Mpeg video

Mpeg [7, 8] referred to Moving Picture Experts Group. Characteristics of Mpeg compression are three different frame types (I, P, and B frames) and inter-frame coding structure GoP (Groups of pictures).
i) I-frame (Intra Pictures) is independently encoded and decoded of the other frames. I-frame contained all the needed information to reconstruct the image after decoding, I-frame compressed rate is relatively low. Therefore, I-frame is an important button to access the video segment.

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ii) P-frame (Predicted Pictures) is encoded from the precious I-frame, P-frame by using the motion compensation algorithm. Therefore, as a result of motion compensation limitation, the number of P-frame between two I-frames cannot be too large.
iii) B-frame (Bi-directionally Predicted Pictures) is encoded by interpolation between the previous and posterior I-frames and P-frames.

The entire videos, decoded in smaller unit, are called GoP. A GoP, specifying the time of P-frame and B-frame between two I-frames, are characterized by two perimeters: the distance among I-frames (N) and the distance from I-frame to P-frame (M). This hierarchical structure depends on encoding of a number of frames in each GoP illustrated in Figure 1 [7].

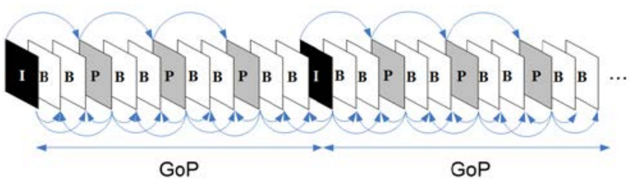

Fig 1: A GoP structure, N = 12, M = 3

2.2 Video quality measurements

At present, among several measures used to evaluate video quality transmission over the network, peak signal to noise ratio (PSNR)(dB) [8, 9] is considered the most famous one. Therefore, PSNR(dB) is chosen for analysis in experimental simulations in this paper.

2.3 The decrease of video transmission quality over network

Figure 2 is a simple example of online video transmission. The IP packets from the video server are transmitted across the network. Handling packets may encounter problems due to the buffer delay, the processing delay or transmission delay. On the other hand, packet loss can be caused due to buffer overflow or link error.

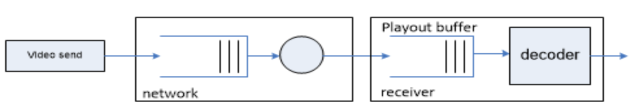

Fig 2: Network video transmission applications

Mpeg frame size is greater than Maximum Transmission Unit (MTU) size of IP packet; therefore, they will be fragmented into smaller units [10]. Due to structure for inter-coding frame of Mpeg video, the packet loss may cause high frame-loss ratio [11]. For example, the loss of I-Frame is considered the loss of GoP. If the GoP size is small, it will cause bandwidth recourse wastage. If the GoP size is large, the large transmission error will cause loss of many frames and bad quality for video streaming receiver. The RED active queue management algorithm plays an important role in dealing with congestion to reduce the packet loss in the router buffer. However, RED marks or drops packets in pa standard for probability without video packet distinction. In this paper, we focus on improve RED research to improve the video quality.

3. Red queue management mechanism

RED algorithm is proposed by Sally Floyd and Van Jacobson for active queue management (AQM) function in 1993 [4, 5, 6], and then it is standardized by the IETF request [12]. RED will reduce likelihood of TCP global synchronization; maintain low latency as well as high throughput and low latency fair treatment by using multiple TCP connections. RED calculates the average queue size based on the low-pass filter and a weighted moving average of exponential growth (EWMA-Exponential Weighted Moving Average). Average queue size is compared to two threshold values: the minimum threshold (min_{th}) and a maximum threshold (max_{th}). Figure 3 shows the active queue management mechanism of RED.

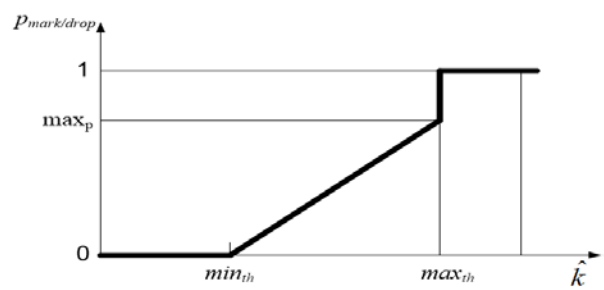

Fig 3. RED operation mechanism

When the average queue size is less than the minimum threshold (min_{th}), no packet is marked (or marked with probability 0). When the average queue size is greater than the maximum allowed threshold (max_{th}), all incoming packets are marked, and in fact the package can be dropped (or marked with probability as 1).

When the average queue size is between minimum threshold and maximum threshold values, each coming packet is marked with a probability p_a, which is a function of the average queue size $\hat{k}$. At each time a packet is marked or (drop), the probability of marked packet must match the bandwidth of the shared connection in the router. Three cases above are considered as 3-phase process of congestion: the first phase is operating normally, the second phase is to avoid congestion and the third phase is congestion control. RED marks or (drop) packets according to probability p_a, which is linear function of $\hat{k}$ and is defined as follows:

$$p_b = \max_p \frac{\hat{k} - \min_{th}}{\max_{th} - \min_{th}} \quad (1)$$

$$p_a = \frac{p_b}{1 - count * p_b} \quad (2)$$

Here, RED does not use the actual size of the queue to determine p_a but the average queue size $\hat{k}$ instead. The purpose is to avoid rapid fluctuations of the queue when the batch is sent for a short time. RED calculated with ω factor for each coming packet to the following assignment.

$$\hat{k} = (1 - \omega) * \hat{k} + \omega * k \quad (3)$$

With ω ($0 < \omega < 1$) is the queue weight and k is the current queue size. If the value of ω is small enough, the average value will be less inclined to change, and will be less susceptible to the short period queue (for example: ω = 0.002). The descriptions of RED algorithm are shown below [4, 5, 6].

For each packet arriving:
Calculate the average queue size
if (queue is not empty) {

$$\hat{k} = (1-\omega) * \hat{k} + \omega * k \; \}$$

else {

$$\hat{k} = (1-\omega)^{f(time - q_{time})} * k;$$

$$q_{time} = time;$$

}
Decide to dropped packets

$$if\,(\min_{th} \le \hat{k} \le \max_{th}) \; \{$$

Calculate marking probability p_a:

$$p_b = \max_p \frac{(k - \min_{th})}{(\max_{th} - \min_{th})};$$

$$p_a = \frac{p_b}{(1 - count * p_b)};$$

Mark the packet with probability p_a;
}
else $if\,(\max_{th} < \hat{k})$
Mark the packet (p = 1)
else
put the packet into the queue (p = 0);
End for

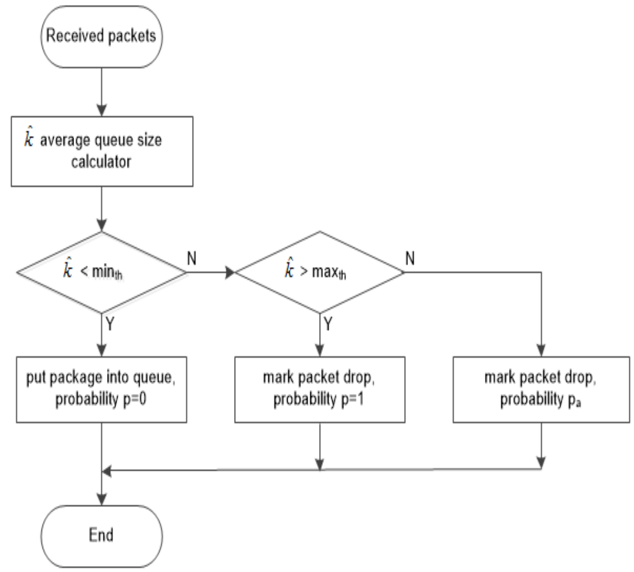

Fig 4: The flow chart presented RED algorithm

However, like many other AQM algorithms, RED shows its weakness in network packets management of multithreaded involvement of the video stream, which marks mechanism or remove the RED packet without distinction of data packets and video packets. Therefore, to improve the of video transmission quality it is necessary to build priority mechanism to classify priority the marked or dropped packets to buffer packets, we propose a linear function affected the pa probability during packet marking or drop as in Section 4.

4. Proposal to improve red algorithm

4.1 The idea ViRED algorithm

On the basis of original RED algorithm, we have built a linear function to adjust the probability of mark or (drop) the packets based on the average buffer size element at the router, and the characteristics of the flow data to the buffer. We propose integrating the adjust probability linear function in RED algorithm in process of packet marking (drop) packets as follows:

If the received packet is a video
Updates the value
$p_a = u.\, p_a$;
else
$p_a = p_a$;

The proposed improvement is illustrated in Figure 5 so-called ViRED.

4.2 Definition of the function u

To classify video packet priority, a function is built for $u \in [0;1]$

$$u(\hat{k}) = 1 - \alpha \frac{\hat{k}}{L}$$

Where

- L : the buffer size,
- $\alpha \in [0, 1]$;
- $\hat{k}$ the current average size of the buffer.

U function gets a less than 1 positive value, is inversely proportional to the value. In the experimental simulation, we choose $\alpha = 0.02$ because The ViRED algorithm always converges to the original RED and improves video quality when packet loss occurs or network status is complicated.

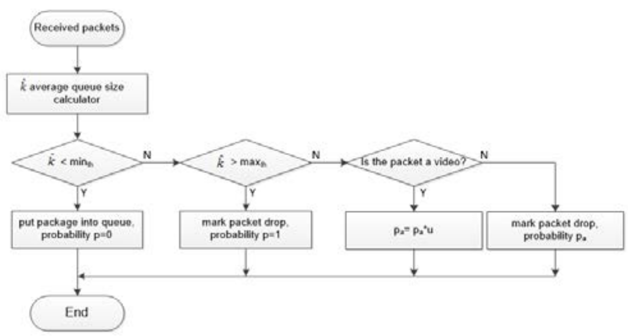

Fig 5. The flow chart presented ViRED algorithm

5. Simulation results and evaluation

We use EvalVid [13, 14], a video streaming simulation framework for evaluating video transmission quality over IP networks. Network configured simulation scenario (topology) has 32 nodes (Fig 6) and uses UDP and TCP protocols. The video transference is made from n_0 node to n_1 node, the button su_1.. su_5 sends data which has a constant bit rate over UDP protocol to the ru_1 .. ru_5 destination node, the node st_1, st_2, ... st_{10} transfer data over FTP on TCP protocol to destination node rt_1, rt_2,. .. rt_{10}, simulation execution time is 10 seconds.

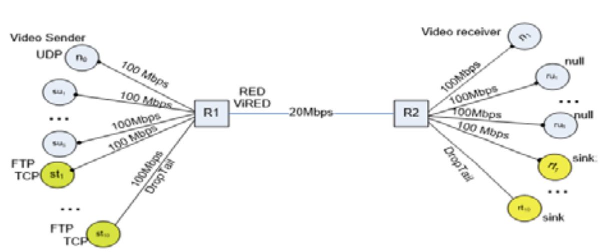

Fig 6: Network configuration using video streaming stimulation with the RED and ViRED queue management mechanism at router R1

Video trace files (video trace) [15] used in the simulation is Akio.yuv [16], 352x288 resolution with 300 frames were found at a rate of 30 frames per second (30 fps). Queue mechanism used at the router R1 is RED, ViRED, queues at the other line is DropTail. The default Drop Tail and RED mechanism was built in NS-2[17], ViRED source is integrated to the NS-2 simulator. Using the framework with ViRED and RED mechanism to evaluate the video transmission quality shows that improve ViRED algorithm has increased the average PSNR (dB) quality (Table 3), compared with RED approximately 6.9%.

The results in Table 1 and Figure 7 show the transmission bandwidth on the R1-R2 fluctuations, increased from 0.5 Mbps to 5 Mbps transmission latency when using RED and almost ViRED approximation, matched exactly with amplitude deviation average of approximately 0.25% and approaching 0 when bandwidth increases. Therefore, the integration of packet classification mechanism, although the processing time increases buffers and leads to increased latency without affecting the quality of video transmission.

Table 1. The relationship between delay and bandwidth on the link R1-R2

Bandwidth (Mbps)	***RED (ms)***	***ViRED (ms)***
0.5	0.160738	0.169738
1.0	0.092226	0.096226
1.5	0.063430	0.066430
2.0	0.054839	0.055839
2.5	0.051118	0.050118
3.0	0.048264	0.049264
3.5	0.046763	0.048763
4.0	0.045548	0.046548
4.5	0.044808	0.046808
5.0	0.044203	0.047203

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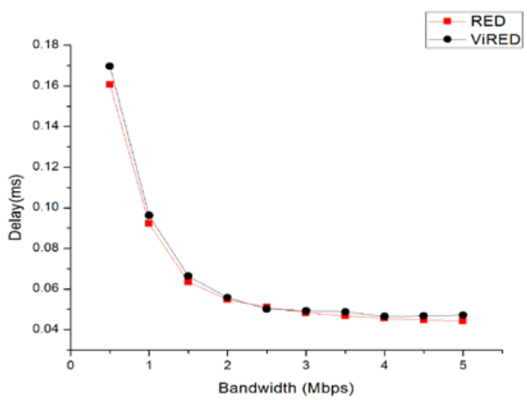

Fig 7. Delay and bandwidth relationship

Table 2 shows the results of the video packet loss rate in simulation experiments corresponding to the bandwidth value changes in R1-R2 bottleneck transmission when using and ViRED and RED. As it can be seen in Figure 8, the video packet loss rate when using ViRED algorithm decreased significantly at approximately 9.33% over the original RED algorithm.

Table 2. Packet loss rate when using RED and ViRED

Bandwidth (Mbps)	***Packet loss rate***	
	RED	***ViRED***
20	0.366243	0.266243
25	0.344008	0.244008
30	0.312154	0.212154
35	0.301234	0.201234
40	0.293421	0.193421
45	0.273454	0.173454
50	0.232323	0.132323
55	0.203232	0.123232
60	0.173323	0.113323

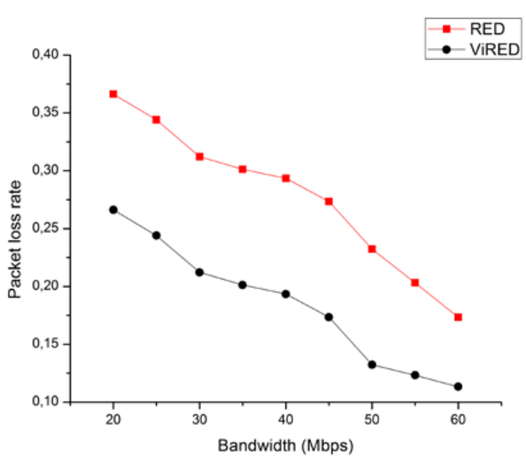

Fig 8. Video packet loss rate when using ViRED and RED

In simulation, we calculate, matching PSNR values (dB) when using RED mechanisms and ViRED. In Table 3 the PSNR value of received frames when video transmission over the network can be configured as shown in Figure 9. We show that when using the new queue mechanism ViRED average PSNR values increased approximately 6.9% compared with RED. Use common sense in the figure 10.a, 10.b corresponding to 222 frames with PSNR value of 35.78 dB and 39.78 dB respectively, also showed better image quality when using RED.

Table 3: PSNR values (dB) of the received video frames when using RED and ViRED

Bandwidth (Mbps)	***RED (ms)***	***ViRED (ms)***
…	…	…
220	35.47	39.47
221	35.27	39.27
222	35.78	39.78
223	36.19	39.19
224	36.63	39.63
225	37.32	39.32
226	39.16	39.16
227	40.66	40.66
228	42.83	42.83
229	46.11	46.11
230	46.99	46.99
231	43.73	43.73
232	42.23	42.23
233	38.99	39.99
234	36.33	39.33
235	35.71	39.71
236	35.37	39.37
237	35.36	39.36
238	37.23	39.23
239	41.67	41.67
…	…	…

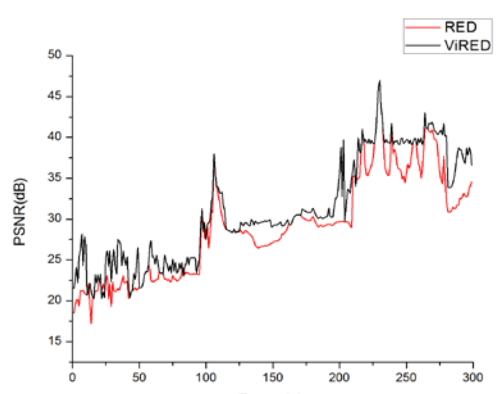

Fig 9. Comparison of PSNR values (dB) using RED and ViRED

(a) (b)

Fig 10. The corresponding frames received with: (a) RED and (b) ViRED at router R1

6. Conclusion

Video quality reduction in transmission over a network can occur; especially there are multiple competitive resources on a network which cause congestion at the center node. In this paper we have proposed solution which uses a linear function to adjust the dropping packet probability on the basis RED queue management mechanism. It is based on the average queue size which has integrated packet priority classification mechanism of the video stream. The ViRED algorithm has improved the video transmission quality over IP networks in the context of multi-threaded network participants and data loss.

In further research, we continue to improve active queues in process of marking or removing packet, determining the parameter of the u function on the characteristics of the queue management algorithm to achieve the better performance when transmitting video data over an IP network in complex network environments such as unstable wireless networks, multi-threaded network.

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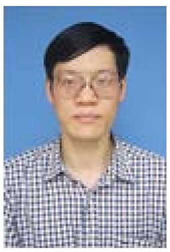

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