An Efficient Packet Discarding Scheme for On-demand Video Streaming

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Abstract

Streaming services over the Internet are expected to grow rapidly in the near future. However, bursty packet losses caused by buffer overflow at routers are a serious problem in these services, and are responsible for degradation of the Quality of Service (QoS). Active Queue Management (AQM) that discards the packets at routers is effective for avoiding the buffer overflow but results in the degradation of QoS as well unless it takes into account the characteristic of the streaming media. Generally, the streaming media implement forward error correction (FEC) that is a promising technique to recover the lost packets during transmission at receiver ends. In this paper, a packet discarding scheme considering the FEC recovery ability and the video encoding method at routers is proposed. This packet discarding scheme is provided by combining the assigning priority to the media packets at the sender end and the simple packet discarding scheme at the routers. Thus, these proposed methods reduce the packet loss ratios and improve the QoS of streaming media. The results of computer simulations show that the packet loss ratio of the proposed method is highly improved compared to the conventional method even for highly congested networks.

Keywords*: video streaming, video on demand, packet loss ratio, packet discarding, forward error correction*

1 Introduction

The demand for the video streaming such as Video on Demand (VoD), remote lecturing system and Internet TV is expected to increase rapidly with development of the broadband networks. However, the quality of Service (QoS) provided by current best effort Internet is not good enough for these video applications. According to [1], the multimedia QoS over the Internet has a layered structure and can be divided into six levels: physical-level, link-level, network-level, end-to-end-level, application-level and user-level. Since these applications are provided for the end users, they require at least a guaranteed end-to-end QoS. Therefore, this QoS is the focus of this study. In the end-to-end QoS, main factors responsible for degradation of QoS are delay, jitter and packet losses. The effects of delay and jitter could be overcome by pre-buffering of media stream at the receiver end. Therefore, packet losses are considered as the main factor that degrades the QoS of the video streaming applications. In order to recover packet losses during transmission, the end-to-end and network-side approaches have been already proposed [2-14]. Although the network-side approaches are effective compared to the end-to-end approaches, the complicated mechanism in the network-side approaches and their overhead limit their efficiency and scalability (e.g., some of them require data caching at router sites). Therefore, an end-to-end approach with a simple network-side scheme is considered as an attractive solution.

Packet losses in the Internet are primarily caused by buffer overflows at routers when the network is congested. These losses are bursty in nature and the video streaming applications are very much affected by them since the generations of the video streaming traffic is also bursty. Forward Error Correction (FEC) is a promising end-to-end technique to recover the lost packets and improve the QoS of streaming media. However, it is well-known that conventional FEC scheme may degrade the network condition due to the increased congestion caused by FEC overhead when the number of streaming sessions increases. This leads to more packet losses and the degradation of QoS. Also, the packet discarding scheme at routers in the network side is effective to avoid the bursty packet losses due to the buffer overflow. However, if the media packets are discarded without considering the characteristics of the video applications, this also results in the degradation of QoS. Here, MPEG-2 is assumed as the video encoding. In this paper, an efficient video streaming that combines a novel FEC method with the simple packet discarding scheme considering the FEC recovery ability and the video encoding method is proposed. The proposed method avoids the increased congestion caused by FEC overhead during media streaming. It also gives priority to selected media packets depending on the FEC recovery ability and the video encoding method at the sender end. A simple algorithm of priority-based selective packet discarding (SPD) is performed at routers as an active queue management (AQM) using the above priority assigned by the sender end. Thus, the packet discarding scheme considering the FEC recovery ability and the video encoding can be achieved and it avoids the buffer overflow. Consequently, the proposed method will reduce the packet losses in the video streaming of the future Internet and is suitable for the on-demand streaming services.

The balance of this paper is organized as follows. Next Section briefly introduces an overview of MPEG-2. Section 3 describes the related works of this study. In Section 4, we discuss the proposed method in detail. The performance of the proposed method is evaluated by way of computer simulations and results are presented in Section 5. Finally this paper is concluded in Section 6.

2. An Overview of MPEG-2

In this paper, we focus on the on-demand streaming media delivery of MPEG-2 [15] video. MPEG-2 has been widely used as an encoding scheme for the video streaming applications. In MPEG-2, sequence of the video stream consists of a series of frames. Compression algorithms are used to reduce both spatial and temporal redundancy of the video data stream. The spatial redundancies (intraframe) are reduced by discrete cosine transform and entropy coding. The temporal redundancies (interframe) are reduced by prediction of future frames based on motion vectors. This is achieved using three types of frames, as shown in Fig. 1. I-frames use only intraframe coding. P-frames use a similar coding algorithm to I-frames, but with addition of motion compensation with respect to the previous I- or P-frames. B-frames are similar to P-frames, except that the motion compensation can be with respect to the previous I- or P-frames and the next I- or P-frames. Typically, I-frames require more bits than P-frames and B-frames. After encoding, the frames are arranged in a deterministic periodic sequence like IBBPBBPBBPBB, which is called Group of Pictures (GoP). The loss of I-frames affects the video quality more than P- and B-frames. Therefore, I-frames are the most important frames in MPEG-2. Similarly, P-frames are considered more important than the B-frames.

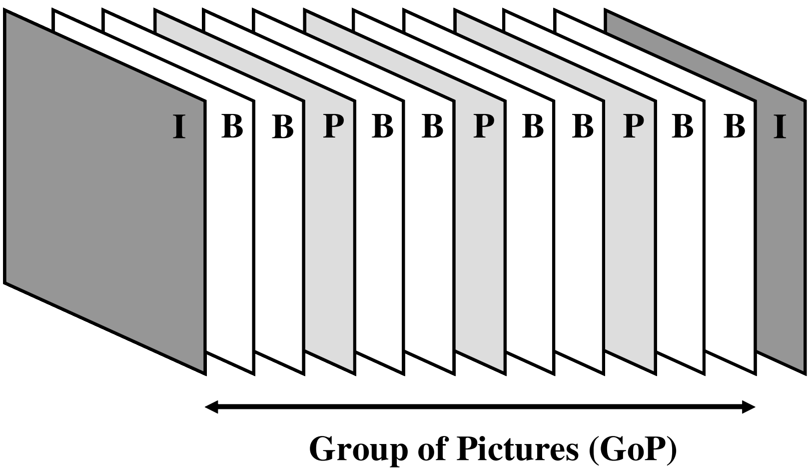

Figure 1. Sequence of frames in MPEG-2

3. Related Works

3.1 Types of Packet Loss Recovery Methods

There are 2 types of packet loss recovery methods that guarantee the end-to-end QoS in the Internet [2]. First is a closed loop type, which is called Automatic Repeat reQuest (ARQ) and second is an open loop type, which is called Forward Error Correction (FEC).

In ARQ, when packet losses occur, the sender will retransmit lost packets according to the notification of lost packets from the receiver end. This method needs to allow large delay time for the retransmission request to return to the sender and be serviced. Therefore, the retransmission delay is unacceptable for video streaming applications because they are very sensitive for delays during media streaming. Moreover, packet losses in the Internet occur due to the network congestion. Even if the network is congested, these applications cannot stop their streaming services because of the requirement of delays. When retransmission is done, it may cause more congestion leading to more packet losses, which results in a network collapse. Therefore, it is necessary for the retransmission method in streaming media to control the source transmission rate according to the network condition [16-18]. This control scheme is based on the Additive-Increase Multiplicative-Decrease (AIMD) algorithm [24]. When network is congested, source coding rate is reduced multiplicatively at the sender end. Thus, this scheme avoids network congestion. In other words, this scheme gives a high priority to the retransmitted packets, making sure that they are transmitted to the receiver end at the expense of the source rate. The multiplicative decreasing of source rate in streaming media results in the degradation of the user-level QoS. When network congestion is reduced, the source rate is increased additively. Therefore, it takes a longer time for the reduced source rate to revert back to the original rate before congestion. This also results in the degradation of the user-level QoS. Even if TCP is used for the streaming, the above described situation still occurs since TCP is performed according to AIMD algorithm as well. From these descriptions, although it is clear that ARQ is an effective technique to solve network congestion, it is not considered as a suitable scheme for the media streaming. Therefore, FEC technique is preferred.

In FEC, redundant packets, which are generated from original media packets by the use of an error correction code, are transmitted along with the original media packet so that the lost original media packet can be recovered using the redundant packet [4,24]. This is illustrated in Fig. 2. As this method of packet loss recovery provides resilience to packet losses without increasing latency, it is suitable for streaming media. Generally, implementation of the FEC method requires a redundant bandwidth that is called an FEC overhead. When FEC with (n,k) block code is applied, where n is the total number of packets and k is the number of data packets, it adds $(n-k)$ redundant FEC packets for every k data packets. Notation n and k are called the block length and the data length, respectively. When there are packet losses and if any k packets of n block length are received at the receiver end, all original media packets within the n block length can be recovered using FEC. In contrast to ARQ, the quality of the lost packets retrieved using FEC packets is the same as that of the original media packets.

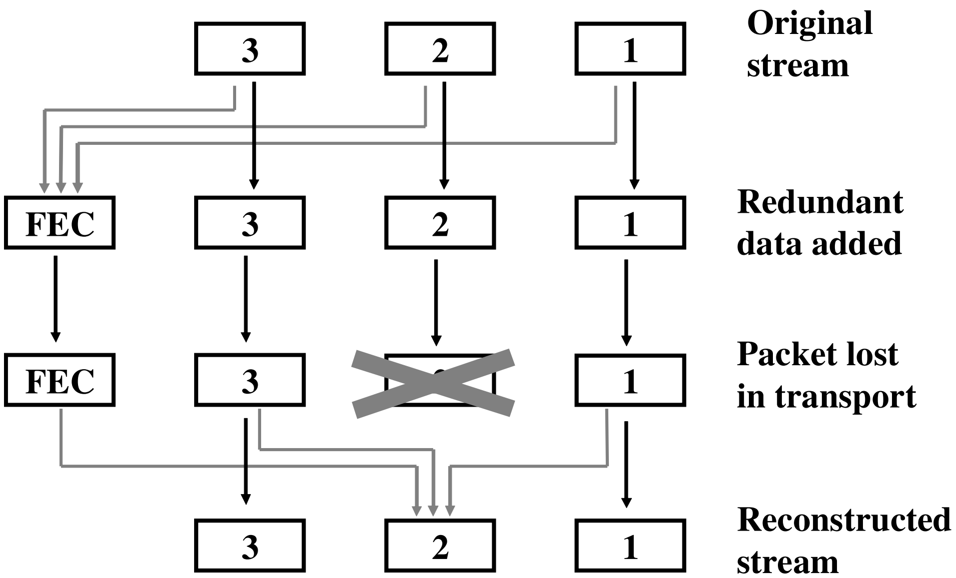

Figure 2. Repair using parity FEC

3.2 Validity of Current Assumptions in FEC Control for Future Streaming Applications

Generally FEC overheads will increase network congestion with the increase of streaming sessions using FEC technique, leading to more packet losses. This increased network congestion caused by the FEC overheads may degrade the QoS of all communications sent to the networks. This is a well-known problem in an FEC technique [13, 14, 19, 24]. Adding FEC packets to real-time applications in order to guarantee their QoS causes more network congestion and packet losses. These packet losses are beyond the FEC recovery ability and cannot be recovered using FEC. One popular technique to avoid this adverse effect is to limit the bandwidth on which FEC is applied. When FEC is applied to just a part of the whole bandwidth, the QoS of that particular part is considerably improved, but the QoS of the remaining part is slightly degraded [13, 14]. If this limitation is applied, it is valid to assume that the FEC overhead is negligible. This assumption is widely used in all conventional FEC methods and is valid to both past and current networks, where the bandwidth of access networks is small and the amount of real-time traffic is not much.

However, broadband networks are already deployed and getting popular day by day. As a result, the on-demand video applications that consume lots of bandwidth will increase drastically in the near future. Future networks will have more real-time traffic. Furthermore, session durations of these real-time applications will tend to increase and therefore redundant FEC packets will continuously be added to the media stream for longer duration. Therefore, the assumption to neglect FEC overheads might be invalid for future networks. Therefore, a new FEC method for on-demand video streaming that does not increase the congestion caused by the FEC overhead during the media streaming is required.

3.3 Conventional Packet Discarding Schemes

Packet losses due to buffer overflow at router often occur with the increase of the network load. These losses are characterized by bursty losses and highly degrade the QoS of the video applications. In order to avoid these bursty losses, active queue managements (AQM) such as RED (Random Early Detection) [20] and RIO (RED with In and Out) [21] have been proposed. RED monitors the average queue size and drops packets based on statistical probabilities. If the buffer is almost empty, all incoming packets are accepted. As the queue grows, the probability for dropping an incoming packet grows too. When the buffer is full, the probability has reached 1 and all incoming packets are dropped. The basis of the RIO mechanism is RED-based differentiated dropping of packets according to the traffic marked as In and Out. Therefore, these algorithms discard the packets subject to the probability determined only by the queue length of the buffer. However, these probabilistic discarding is not always suitable for streaming communications. Any packet discarding scheme suitable for streaming communications should consider not only the queue length of buffer but it also should take into account the FEC recovery ability depending on the FEC block code used and the video encoding method. Implementation of such a scheme is impossible without additional FEC and video encoding information at the routers and is complicated. However, this is very important because the process of recovering lost data packets using FEC packets should not be hampered by the packet-discarding scheme at routers. If the packet discarding scheme considering the FEC recovery ability and the video encoding method can be simply provided, this is an attractive solution.

To summarize these descriptions, it is desirable to have a packet loss recovery and a packet discarding schemes with the following features.

- Avoid the increased congestion caused by FEC overheads during media streaming as the packet loss recovery scheme of the end-to-end approach.
- Discard the packets considering the FEC recovery ability and the video encoding method at routers, and simply implement this AQM as the packet discarding scheme of the simple network-side approach.

4. Proposed Method

4.1 Referential Loss Recovery

Streaming media is very sensitive to transmission delay. However, regarding the on-demand media streaming, there is a permissible delay time before its communication starts. It is called the start-up delay [3]. We propose a referential loss recovery (RLR) method, which separates the FEC packets from the original media packets and send them to the receiver end before media streaming starts. An FEC content consisting of the FEC packets is created when the media content is created. This FEC content is transmitted using TCP in order to make sure they are received correctly at the receiver end. Therefore, the adverse effects of FEC overheads can be ignored because of the congestion control based on the AIMD algorithm in TCP. TCP provides a reliable transmission at the expense of transmission time when it is applied for data communications. Also it can adjust its transmission rate according to the network congestion. As usual, media packets are transmitted using RTP and UDP. The transmission sequence of our proposed method is shown in Fig 3. It is clear that the RLR method can avoid the increased congestion caused by FEC overhead during media streaming, although session durations of recent streaming media tend to increase. If media packets are lost, they are recovered by referring to the FEC content already received at the receiver end. Here, FEC is applied only for I-frame packets, which are the most important packets of MPEG-2 in order to control the FEC overhead. In this method, the length of the start-up delay time to download all FEC content is one of the key factors to be considered. However, both broadband access network environment and restricting FEC only for I-frames packets will make sure the delay time is well within the tolerable limits. [22]

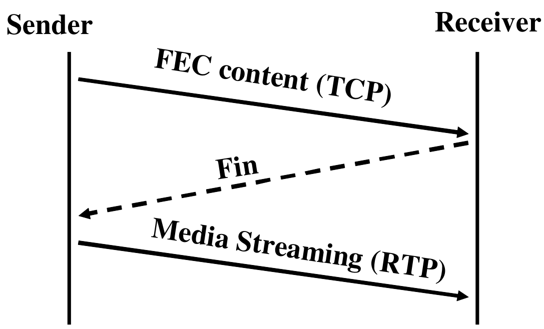

Figure 3. Transmission sequence of proposed method

4.2 Assigning Priority for Selective Discarding

It is suggested to implement simple packet discarding scheme considering the FEC recovery ability and the video encoding method by combining the assigning priority to the media packets at the sender end and the simple packet discarding scheme at the routers.

In the conventional scheme the FEC packets that are sent along with data packets are assigned low priority 1, as shown in Fig. 4-(a). Thus, it intends to protect media packets from bursty losses caused by buffer overflow. In the proposed assigning scheme, since all FEC packets are transmitted to the receiver end before the media streaming starts, 2 levels of low priorities are given to some of the media packets at the sender end, considering the FEC recovery ability and video coding method. Fig. 4-(b) shows a low-priority assignment of proposed method. The low priority 1 is assigned to some I-frame packets starting from the first packet of each FEC block. If the FEC recovery ability is one, only the first data packet in FEC block is assigned low priority 1 and if the FEC recovery ability is two, first two data packets in FEC block are assigned low priority 1. In general, if the FEC recovery ability is r, first r data packets in FEC block are assigned low priority 1. These priority-assigning schemes were decided based on the results of lots of simulations we performed. Even when these low-priority 1 packets are discarded at routers, they could be recovered using FEC content already sent to the receiver end. The low priority 2 is assigned to all B-Frames packets that are considered as the least important frames of MPEG-2.

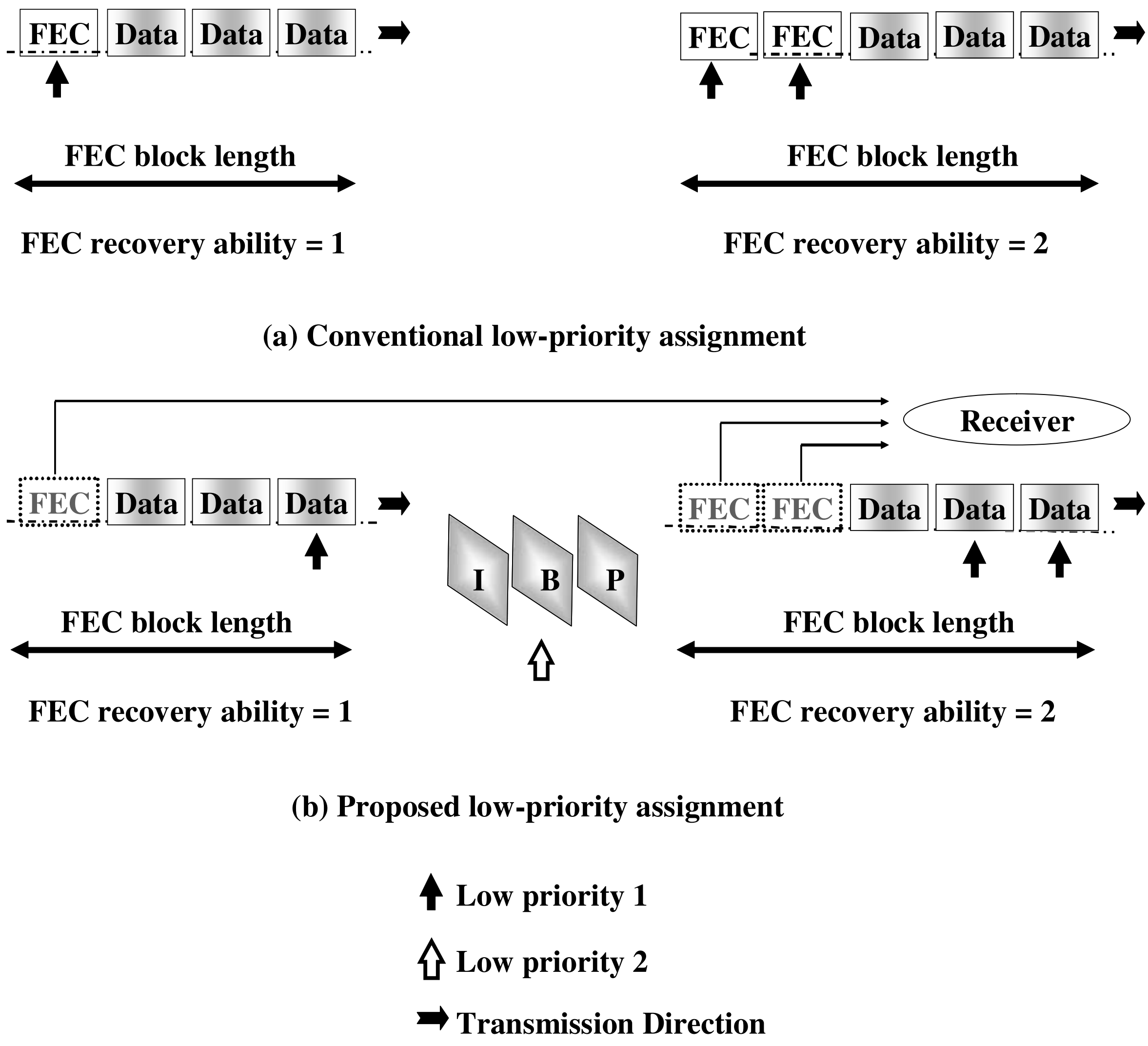

Figure 4. Low-Priority Assignment

4.3 Selective Packet Discarding Method

A very simple algorithm of the priority-based selective packet discarding (SPD) is implemented to avoid buffer overflows at routers. As shown in Fig. 5, the low threshold Q_L and the high threshold Q_H are set in the buffer at router. If the queue length exceeds the threshold Q_L, the router discards the packets with the low priority 1. In other words, if it exceeds Q_L, some I-frame packets are discarded in the proposed method whereas FEC packets are discarded in the conventional method. Furthermore, if the queue length exceeds the threshold Q_H, the packet discarding scheme related to video encoding method is also activated and the low priority 2 packets are discarded in the proposed method. Selective discarding of low priority B-frames packets will save the loss of I- and P-frames packets due to bursty losses.

The router just discards the packets based on their priorities and queue length in buffer. Therefore, this selective packet discarding method is very simple.

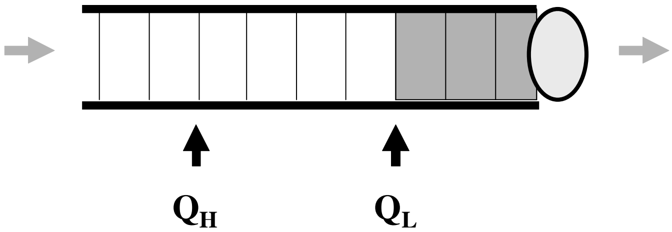

Figure 5. Selective Packet Discarding Scheme at Routers

5. Performance Evaluations

The simulation network topology is depicted in Fig. 6. It consists of video sources, edge routers and core routers. Each edge router has ten video sources and they are connected to the core router with a capacity of 10 Mbps.

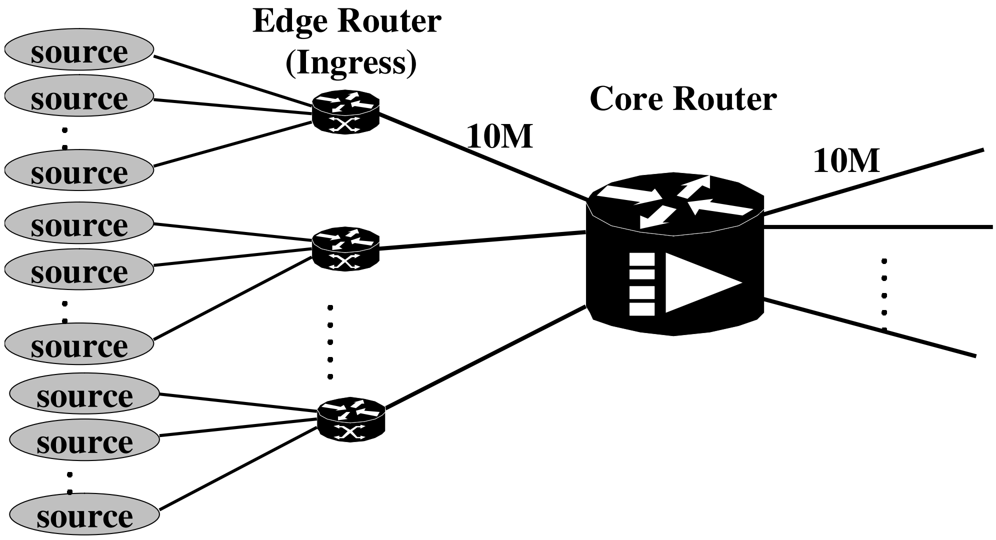

Figure 6. Simulation Model

5.1 Video Source

The video sources generate MPEG-2 VBR traffic modeling of movies from Alien, Blade Runner, City of Joy, Dick Tracy and Road to Utopia described in [23]. Their average bit rates are resized to 1 Mbps. 10 MPEG-2 sources are realized by duplicating each of the 5 movies. The simulation parameters of source traffic are shown in Table 1. Every data frame is cut into small packets with fixed length of 500 bytes.

When the video content is created, at the same time FEC content is created only for I frame packets. Simulations are done to compare FEC contents generated using Single Parity Check (SPC) code described in the example of [4] and Reed Solomon (RS) code. The FEC contents are sent by TCP flow before the media streaming starts. The media content is sent by RTP and UDP.

Table 1. Simulation Parameters of Source Traffic

Type of traffic	Variable Bit Rate (VBR)
Number of source	10
Average bit rate	1 Mbps
Frame rate	30 frames/second
GoP sequence	IBBPBBPBBPBBPBB
Packet size (fixed)	500 bytes

5.2 Edge Router

It is assumed that packet losses do not occur at ingress edge router when video traffic is multiplexed. The destination address of each video source is manipulated to make sure the buffer overflow occurs only at the core router. This enables to observe the performance of core router subject to the network load more accurately.

5.3 Core Router

Core router has 8 input and 8 output ports, and each link capacity is 10 Mbps. The buffer size of each input port is set to 50Kbytes, which equals to 100 packets since the packet size of generated traffic is 500 bytes. The waiting time based round robin algorithm is implemented as the scheduling algorithm. The simulation parameters of core routers are shown in Table 2. When the queue length in the buffer exceeds Q_L, low priority 1 packets are discarded selectively. When it exceeds Q_H, low priority 2 packets are discarded.

Table 2. Simulation Parameters of Core Router

Buffer size	100 packets
High threshold	80 packets
Low threshold	50 packets

Performance of the proposed framework is compared with the conventional method for different network loads. Moreover, FEC with the selective packet discarding (SPD) and FEC without SPD are compared. The (10,9) SPC code is applied for both conventional and proposed methods as an example. In the proposed method, the (11,9) RS code and the (5,4) SPC code are also applied because they have similar amount of FEC content as shown in Table 3, and therefore they are comparable.

Table 3. Amount of FEC Data for Each MPEG2 Source in the case Each Content Length is 90 Min.

Media Contents	FEC contents size (MB)	
	(5,4) SPC code	(11,9) RS code
Alien	23.7	23.4
Blade Runner	20.7	21.2
City of Joy	25.0	24.8
Dick Tracy	24.0	23.9
Road to Utopia	28.1	28.3

5.4 Results Analysis

The packet loss ratios of I-, B-, and P-frames according to the network load are shown in figures 7, 8 and 9 respectively. All figures show that the packet loss ratio increases with the network load. The conventional method has higher packet loss ratio compared to the others because of the adverse effect of the FEC overhead. Fig. 7 shows that the proposed method reduces the packet loss ratio of I-frame compared to the conventional method and the No FEC condition. When the selective packet discarding at router is not applied, the RS code marginally improves the packet loss ratio compared to the SPC code. It is clear that the RS code can cope with the bursty packet losses compared with the SPC code when no active queue management is performed at the router. However, there is a vast improvement of packet loss ratio in the case of FEC with SPD scheme. When network load is less than 0.9, no packet loss is observed for all proposed method that uses FEC with SPD. It means that all lost packets are recovered at the receiver end. Even when the network load is 0.9, the (11,9) RS code with SPD and the (5,4) SPC code with SPD could recover all losses while the packet loss ratio is less than 10^{-4} in the case of the (10,9) SPC code with SPD.

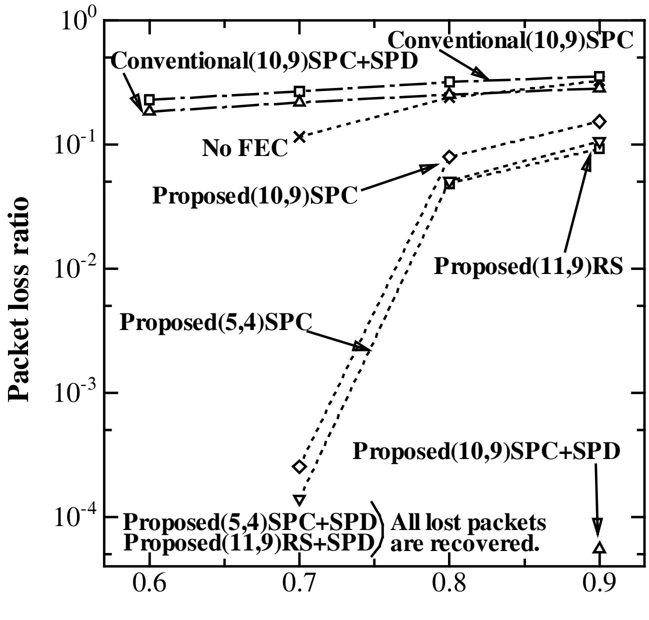

Figure 7. Packet Loss Ratio of I-frames

Figures 8 and 9 show that the proposed FEC without SPD schemes have the same performances as the No FEC condition because in our proposed method, FEC is not applied for B- and P-frame packets. Regarding the B-frames, the proposed FEC with SPD schemes slightly degrade their performances compared to the No FEC condition, as shown in Fig. 8. This is because in our proposed method the B-frame packets are discarded actively when the queue length exceeds the high threshold Q_H. However, the proposed FEC with SPD schemes improve their performances when network load is equal to 0.7. This is because the selective packet discarding of I-frame packets considering the FEC recovery ability reduces the queue length after reaching the low threshold Q_L. In other words, when the SPD scheme related to the FEC recovery ability is activated, very often the queue length will not exceed the high threshold Q_H. Therefore the SPD scheme related to video encoding method is not often activated. Regarding the P-frames, Fig. 9 shows the packet loss ratio of the proposed FEC with SPD method is less than all the other methods even though FEC is not applied for P-frames. When the network load is less than 0.8, no packet loss is observed for the proposed FEC with SPD. It can be considered that this improvement in P-frames is achieved by the SPD scheme. In other words, FEC will recover all I-frames if they are lost and SPD will improve the packet loss ratio of P-frames. Since the priority of importance of frames in MPEG-

2 is I, P, and B, the proposed combination of FEC and SPD provides a huge improvement in the overall QoS of the on-demand video streaming.

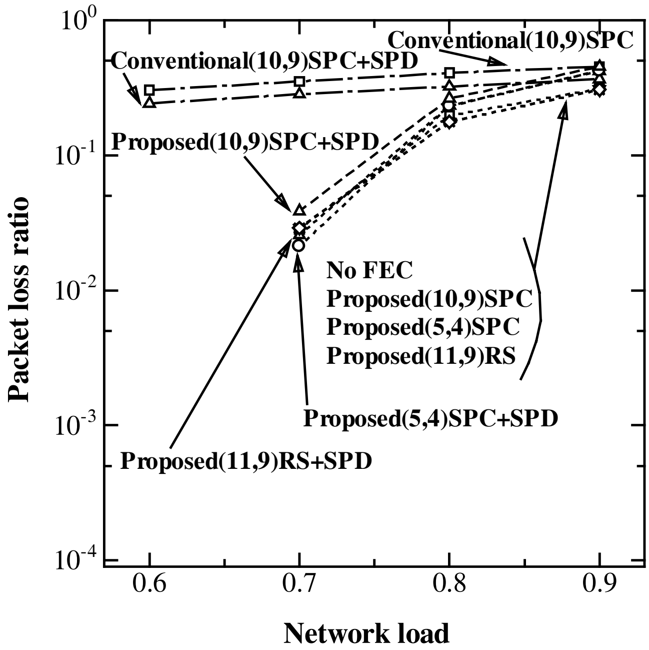

Figure 8. Packet Loss Ratio of B-frames

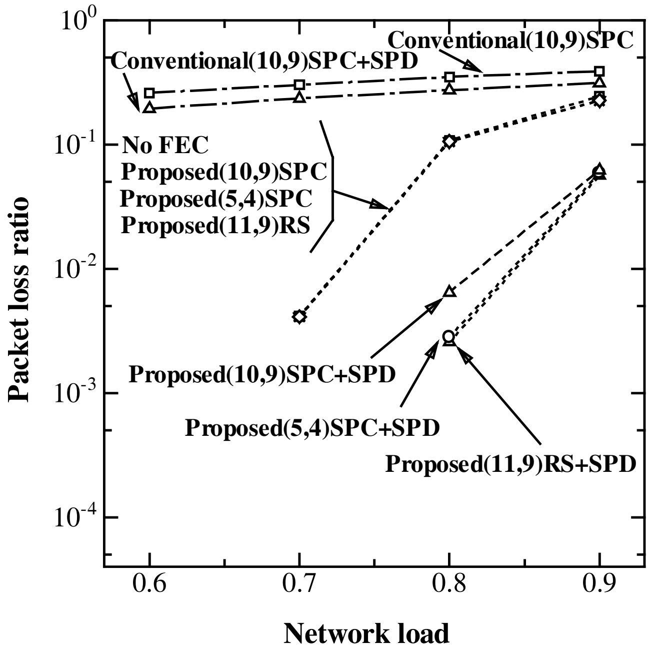

Figure 9. Packet Loss Ratio of P-frames

When (5,4) SPC code and (11,9) RS code with SPD are applied for I-frame packets, no packet loss is observed in I fames, as shown in Fig. 7. In order to evaluate the difference of the performance between these FEC codes whose FEC contents size are almost the same, simulations are performed when buffer size and Q_L are set to 50 and 5 packets respectively. The high threshold value Q_H is ignored in these simulations to observe the performance difference between these FEC codes accurately. Thus, the (5,4) SPC and the (11,9) RS codes are applied only for I-frame packets and packet losses will occur without the SPD scheme related to the video coding method. The simulation results where the network load is 0.9 are summarized in Table 4. It shows the proposed (5,4) SPC with SPD reduces the packet loss ratio compared to the proposed (11,9) RS code with SPD. This is because the block length of the (5,4) SPC code is about half of the (11,9) RS code, and therefore the SPD scheme considering the FEC recovery ability can quickly discard the low-priority packets for the lower FEC block length and can avoid the buffer overflow. Therefore, it can be concluded that the SPC code reduces the packet loss ratio compared to the RS code when both FEC contents sizes are similar and selective packet discarding scheme considering FEC recovery ability is applied.

Table 4. Packet Loss Ratio of I–Frames according to FEC Codes where Network Load is 0.9.

Method	Packet Loss Ratio
No FEC	3.22×10^{-1}
Proposed (11,9) RS code with SPD	7.76×10^{-4}
Proposed (5,4) SPC code with SPD	6.59×10^{-5}

The parameters for core router are reset to the values in Table 2. Table 5 shows the simulation results where different FEC block lengths of the SPC codes are applied for I-frames. From Table 5, as the block length increases the packet loss ratio increases too. When the FEC block length is less than 10, all lost packets can be recovered even if network load is 0.9. It is clear that the FEC recovery ability is improved with shorter block length although shorter block length in SPC code increases the FEC contents size. Therefore, there is a trade-off between FEC block length and FEC content size that affects the start-up delay time.

Table 5. Packet Loss Ratio of I–Frames for Different FEC Block Length in SPC code where Network Load is 0.9.

Method	Packet Loss Ratio
No FEC	3.25×10^{-1}
Proposed (14,13) SPC code with SPD	2.58×10^{-3}
Proposed (13,12) SPC code with SPD	1.93×10^{-3}
Proposed (12,11) SPC code with SPD	1.25×10^{-3}
Proposed (11,10) SPC code with SPD	7.64×10^{-4}
Proposed (10,9) SPC code with SPD	5.53×10^{-5}
Proposed (9,8) SPC code with SPD	All lost packets are recovered.
Proposed (8,7) SPC code with SPD	All lost packets are recovered.
Proposed (7,6) SPC code with SPD	All lost packets are recovered.
Proposed (6,5) SPC code with SPD	All lost packets are recovered.

The start-up delay time is one of the important factors to be considered in the proposed method. It depends on the FEC contents size, and FEC contents size depends on the media contents length, the average bit rate and the applied FEC code. Here, (9,8) SPC code is applied because its block length is the maximum among SPC codes that give no packet loss from Table 5. Table 6 shows the FEC contents size for each MPEG-2 video sources when the average bit rate of 1 Mbps is adopted. In the future, the FTTH technology will be deployed and the capacity of the access networks will be 100 Mbps. In this scenario the download time of FEC contents is less than 2 second even for 90 minutes of media contents length and we believe that this is an acceptable start-up delay time.

Table 6. Amount of FEC data for each MPEG2 source in the case of (9,8) SPC code.

Media Content Length	FEC contents size (MB)				
	Alien	Blade Runner	City of Joy	Dick Tracy	Road to Utopia
30 min.	4.4	3.7	4.6	4.5	5.2
60 min.	8.8	7.4	9.3	8.9	10.3
90 min.	13.2	11.0	13.9	13.4	15.5

6. Conclusion

In the near future most of the applications over the Internet will be using FEC as a solution for packet losses because the on-demand video streaming traffic will be more compared to the conventional non real-time traffic. The conventional FEC methods cannot be applied in this scenario as they may degrade the QoS of all communications due to the increased congestion caused by FEC overhead.

In this paper, we proposed and proved the effectiveness of a new FEC method with selected packet discarding (SPD). It is proposed to generate the FEC contents and transmit them during the start-up delay time using TCP. FEC contents are generated based on the video encoding method. The SPD is simply achieved by combining the assigning priority to certain media packets considering the FEC recovery ability and video encoding method at sender ends with the simple algorithm of the priority based packet discarding at routers. The proposed method restrains the degradation caused by the conventional FEC overhead and highly improves the packet loss ratio with the use of the simple active queue management at the routers even for the highly congested networks.

Future work will focus on implementation and evaluation of its performance in the real Internet. Also, we wish to simulate the possibility of improving the proposed SPD by assigning hierarchical priorities and have more than two threshold values in the buffer of core routers.

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Article received: 2006-11-28