DYNAMIC QUALITY ADAPTATION MECHANISMS FOR TCP-FRIENDLY MPEG-4 VIDEO TRANSFER

Naoki Wakamiya, Masaki Miyabayashi, Masayuki Murata, Hideo Miyahara

Graduate School of Engineering Science, Osaka University
1–3 Machikaneyama, Toyonaka, Osaka 560-8531, Japan
E-mail: {wakamiya, miyabays, murata, miyahara}@ics.es.osaka–u.ac.jp

ABSTRACT

When a considerable amount of UDP traffic is injected into the Internet by distributed multimedia applications, the Internet is easily driven congested. Consequently, bandwidth available to TCP connections is oppressed and their performance significantly deteriorates. In order that both multimedia applications and TCP-based ones fairly co-exist in the Internet, it becomes increasingly important to consider the inter-protocol fairness.

In this paper, we propose a video quality adjustment mechanism which accomplishes the high-quality, stable, and TCP-friendly video transfer under a lossy environment in cooperation with the FEC (Forward Error Correction) technique. Our mechanism adjusts the video quality in accordance with the TFRC (TCP-Friendly Rate Control) rate, the packet loss probability, and the resultant video quality. Through simulation experiments, we show that our proposed method can provide high-quality, stable and TCP-friendly video transfer even in the unstable and lossy Internet.

1. INTRODUCTION

It has been pointed out that selfish UDP traffic injected by multimedia applications easily dominates network bandwidth and drives the network into congestion. As a result, the bandwidth available to TCP connections is oppressed and their performance extremely deteriorates. In recent years, several researches have been focused on the investigation of "TCP-friendly" rate control [1-3]. "TCP-friendly" is defined as "a non-TCP connection should receive the same share of bandwidth as a TCP connection if they traverse the same path" [2]. A TCP-friendly system regulates its data sending rate according to the network condition, typically expressed in terms of the round-trip-time (RTT) and the packet loss probability, to achieve the same throughput that a TCP connection would acquire on the same path. In particular, TCP-Friendly Rate Control (TFRC) proposed in [3] has a feature of adjusting a transmission rate so smoothly while coping with network congestion.

If an application successfully adjusts its sending rate to the TFRC rate, TCP-friendly data transfer can be accomplished. However, TFRC itself does not consider the influence of the TCP-friendly rate control on the application-level performance. We have proposed some rate control methods to accomplish TCP-friendly MPEG-4 video transfer with consideration of the application-level performance, i.e., the video quality. Through simulation experiments, it has been shown that high-quality, stable and TCP-friendly video transfer can be accomplished by regulating video rate at an interval of group of pictures. However, we have not taken into account packet loss within the network so that we could investigate the ideal performance of the TCP-friendly MPEG-4 video transfer.

In this paper, taking into account the quality degradation caused by packet loss, we propose the video quality adjustment mechanism which accomplishes the high-quality and stable video transfer under a lossy environment in cooperation with the FEC (Forward Error Correction) technique. Our mechanism adjusts the video quality in accordance with the TFRC rate, the packet loss probability, and the resultant video quality. Through simulation experiments, we show that our proposed method can provide high-quality, stable and TCP-friendly video transfer even in the unstable and lossy Internet.

The paper is organized as follows. In Section 2, we briefly introduce the FGS video coding algorithm, the TFRC mechanism, and our "G-G *smooth*" method proposed in [4]. In Section 3, we investigate the influence on the perceived video quality by packet loss. Then, we propose a dynamic quality adaptation mechanism for a lossy Internet environment. We evaluate our mechanism in Section 4. Finally, we summarize our paper and outline our future work in Section 5.

2. TCP-FRIENDLY MPEG-4 VIDEO TRANSFER

In this section, we briefly introduce (1) the FGS (Fine Granular Scalability) algorithm [5, 6] which is excellent in adaptation to the bandwidth variation among MPEG-4 video-coding standards, (2) TFRC [3], which accomplishes fair-share of bandwidth among TCP and non-TCP connections, and (3) our FGS video rate control method proposed in [4].

2.1. FGS Video Coding Algorithm

In this paper, expecting highly flexible and scalable rate adjustment capability, we employ a Fine Granular Scalability (FGS) video coding algorithm [5, 6] that is a compression method suitable for video streaming applications and is being introduced into MPEG-4 standards. Figure 1 illustrates a basic structure of the FGS video stream. A FGS video stream is composed of a sequence of VOPs (Video Object Plane). The VOP is the basic unit of image data and is equivalent to the frame or picture of MPEG-1 and MPEG-2. A sequence beginning from an I-VOP is called GOV (Group Of VOP). An FGS video stream consists of two layers, Base Layer

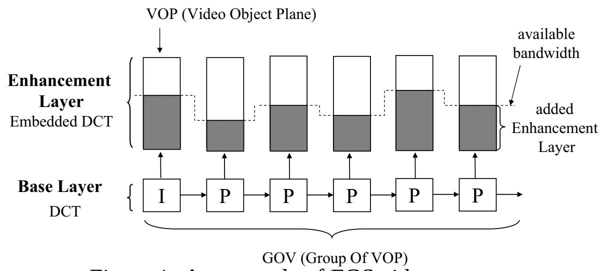

Figure 1: An example of FGS video structure

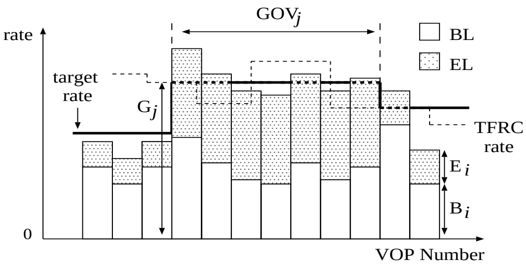

Figure 2: "G-G *smooth*" video rate adjustment method

(BL) and Enhancement Layer (EL). The BL is generated using motion compensation and DCT (Discrete Cosine Transform)-based conventional MPEG-4 coding algorithm, and it provides minimum picture quality. The EL is generated from the BL data and the original frame. The embedded DCT method is employed for coding EL to obtain fine-granular scalable compression.

The video quality depends on both the encoding parameters (quantizer scale, etc.) and the amount of supplemental EL data added. Even if only little EL data is used in decoding a VOP, the perceived video quality is improved. Losses of the BL data have a significant influence on perceived video quality because BL is indispensable for decoding VOP.

2.2. TFRC: TCP-Friendly Rate Control

TFRC [3] is a mechanism to have a non-TCP connection behave similarly to, but more stable than a TCP connection which traverses the same path. During a session, a sender transmits one or more control packets every RTT. On receiving the control packet a receiver returns a feedback information required for calculating RTT and estimating the loss event rate p_{tfrc}. The sender then derives the estimated throughput of a TCP connection which competes for bandwidth on the path that the TFRC connection traverses. The estimated TCP throughput r_{TCP} is given as:

$$r_{TCP} \approx \frac{MTU}{RTT\sqrt{\frac{2p_{tfrc}}{3}} + T_0(3\sqrt{\frac{3p_{tfrc}}{8}})p_{tfrc}(1+32p_{tfrc}^2)}, \quad (1)$$

where T_0 stands for retransmission timeout [1]. Finally, an application on TFRC adjusts its data rate to the estimated TCP throughput r_{TCP} (called "TFRC rate" in this paper) by means of, for example, video quality regulation.

2.3. FGS Video Transfer on TFRC Connection

To accomplish TCP-friendly video transfer with consideration of the application-level performance, i.e., video quality, we have considered the several issues such as control interval, video rate adjustment, and BL rate violation, in [4]. Then, we have proposed "G-G *smooth*" video rate adjustment method.

The interval that TFRC notifies the upper application of a new sending rate does not match the point that the application can change its data structure or amount. Considering the FGS video structure shown in Fig. 1, the target rate G_j of GOV$_j$ is defined as the TFRC rate observed at the beginning of GOV$_j$ as illustrated in Fig. 2. Then, the FGS video rate is adjusted to the target rate by adding a portion of the EL data to the BL data. In the G-G *smooth* method, the video rate averaged over GOV$_j$ satisfies the target rate G_j. The rate E_i of the EL data added to each VOP$_i$ in the GOV is given as $(NG_i - \sum_{VOP_k \in GOV_j} B_k)/N$ where N and B_k stands for the number of VOPs in a GOV and the BL data rate, respectively. It is intended to achieve the smooth variation of video quality by equalizing the amount of supplemental EL data among VOPs, while the

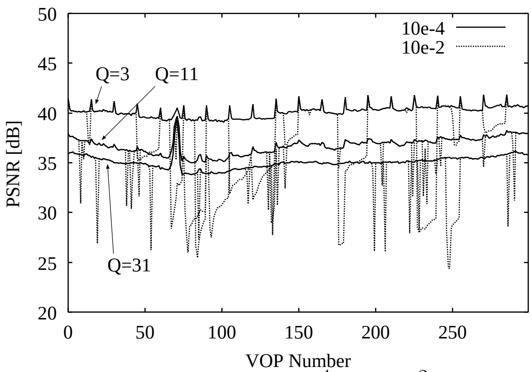

Figure 3: FGS video quality ($p = 10^{-4}$ and 10^{-2}, rate = 1 Mbps)

video rate may instantaneously exceed the target rate. In the case that the available bandwidth becomes smaller by the occurrence of network congestion, the BL rate occasionally exceeds the target rate. Since the BL data are crucial for video decoding, they are always sent out and an excess is managed by reducing the EL rate of the following VOPs or GOVs. The excess is divided and equally assigned to the rest of VOPs in the GOV, thus averaged rate over several VOPs matches the target rate.

3. TCP-FRIENDLY MPEG-4 VIDEO TRANSFER UNDER A LOSSY ENVIRONMENT

In [4], we have demonstrated that G-G *smooth* method proposed for the video rate control provides a high- and stable-quality video transfer on TFRC sessions. In those simulations, we have not taken into account packet loss within the network so that we could investigate the ideal performance of the TCP-friendly MPEG-4 video transfer. However, any kind of TCP-friendly rate control such as TFRC cannot avoid packet loss in nature.

In this section, we propose a video quality adjustment mechanism which accomplishes the high-quality video transfer under a lossy environment in cooperation with the FEC. We first start by investigation of basic characteristics of our G-G *smooth* under a lossy environment. Then we propose a new mechanism and evaluate it.

3.1. Investigation of Influence of Packet Loss on Video Quality

In Fig. 3, we show results of experiments for the case that a video sequence "coastguard" is transferred over a lossy connection with packet loss probability p of 10^{-4} or 10^{-2}. The video is coded at an average rate of 1 Mbps. "coastguard" is a QCIF-large video sequence and played back at 30 fps. The figure shows the video quality variation when only successfully received video data are decoded. In this figure, results for three quantizer scales, i.e., 3, 11, and 31, are depicted. In the experiments, each 1-Kbyte packet is examined and lost at the given packet loss probability.

For the packet loss probability of 10^{-4}, none of about 1,860 packets was lost and no quality degradation was observed. As the packet loss probability increases, a video stream with a smaller quantizer scale begins to be affected. It is shown that the smaller the quantizer scale is, the higher the level of degradation is. The degradation of video quality is notable in the case of $Q = 3$ owing to the loss of BL data. In the cases of $Q = 11$ and 31, annoying spike-shaped degradations were observed. The subjective video qualities in terms of MOS (Mean Opinion Score) were 3.95, 3.88, 2.63 for video streams with of $Q = 3$, 11, and 31, respectively, at 10^{-2} of packet loss probability. From the result of MOS evaluation, we can see that users tend to feel uncomfortable by spike-

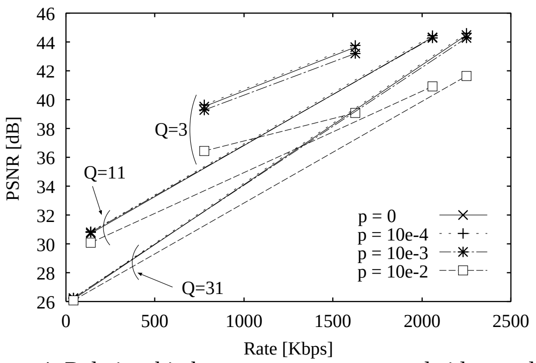

Figure 4: Relationship between average rate and video quality

shaped degradations, that are perceived as flickers. On the other hand, degradations caused by the loss of BL data, which last during a while, is rather acceptable. Therefore, we conclude that we need to protect both EL and BL data and it is better to employ as small a quantizer scale as possible.

We also conducted experiments for the other settings of the video rate. All results are summarized in Fig. 4 in terms of the average SNR, the average video rate, and the quantizer scale, for different level of packet loss. Despite trajectories of quality variation depicted in the figures, average SNR is higher when we choose smaller quantizer scale for given rate and packet loss probability.

3.2. Dynamic Quality Adaptation Mechanism with Packet-Loss Protection for TCP-friendly MPEG-4 FGS Video Transfer

In accordance with the considerations given in the previous subsection, we propose a dynamic quality adjustment mechanism which obtains the video transfer of stable quality.

Our mechanism is based on the G-G *smooth* but reacts against packet loss and employs the FEC technique to protect video data from packet loss. The idea of the Forward Error Correction (FEC) [7, 8] in a packet oriented video transmission scheme is to generate redundant packets at the sender, which can be used at the receiver to recover lost video data packets. When we send f redundant packets in addition to k information packets, all information packets can be reconstructed from successfully received packets as far as the number of lost packets is below f. We use Reed-Solomon (RS) codes for forward error correction in this paper.

To accomplish the efficient control, we have to consider how much redundant packets should be added to the video packets under a lossy environment. Specifically, introducing redundancy suppresses the bandwidth that the server can use to transfer video data within given TFRC rate. When the system faces to a congestion, the target rate given by the TFRC algorithm is decreased. At the same time, the sender should increase the number of the redundant packets to achieve a higher level of protection against the higher loss probability. Consequently, the bandwidth left for the video data become suppressed very much and as a result, the video quality is expected to deteriorate.

Although results are omitted from the manuscript due to a space limitation, we have conducted several experiments where were applied varying packet loss probability and target loss probability to sending FGS video data coded with various quantizer scales over a 1 Mbps session. We verified that independently of actual packet loss probability and the quantizer scale value, setting the target loss probability to 10^{-3} or 10^{-4} leads to the highest video quality.

A new dynamic quality adaptation mechanism in cooperation

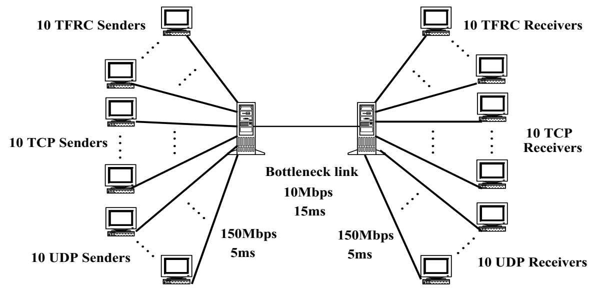

Figure 5: Simulation network model

with FEC becomes as follows.

1. Determine the TCP-friendly sending rate G_j by Eq. (1) at the beginning of GOV$_j$, based on the feedback information (RTT, p_{tfrc}) obtained by the TFRC algorithm.

2. Derive the packet loss probability observed in the preceding interval, P_{j-1}, by dividing the number of lost packets by that of sent packets in the preceding GOV time.

3. Calculate the smoothed packet loss probability P_{ave} by applying the exponential moving average employed in TFRC.

$$P_{ave} = \frac{\sum_{i=1}^{8} w_i P_{j-i}}{\sum_{i=1}^{8} w_i}.$$

Weighting parameter w_i is, 1 for $1 \leq i \leq 4$ and $(9-i)/5$ for $4 < i \leq 8$.

4. Determine the redundant rate r_{FEC} to achieve the target loss probability P_{target}. r_{FEC} is determined as $G_j \times f/n$ where f and n are the number of redundant packets and the number of packets admitted in a GOV time. f is derived by using the following equation,

$$\min_f \left\{ \sum_{i=f+1}^{n} \binom{n}{i} P_{ave}^{i} (1-P_{ave})^{n-i} \leq 1-(1-P_{target})^{n-f} \right\}.$$

5. Determine the sending rate R_{video} allocated to the video data as $G_j - r_{FEC}$ and apply the G-G *smooth* mechanism to R_{video}.

6. Generate redundant packets at the redundant rate r_{FEC} and send them in addition to video packets.

4. SIMULATION RESULTS

In this section, we evaluate the effectiveness of our dynamic quality adaptation mechanism proposed in the previous section through simulation experiments. A simulated network is depicted in Fig. 5, where ten TFRC connections, ten TCP connections and ten UDP connections compete for the bottleneck bandwidth. In the following experiments, the frame rate of coded video is 30 fps and the number of pictures in GOV is 30.

In Figs. 6 and 7, we show simulation results of the method with the FEC technique and without it, respectively. In the case of the original G-G *smooth* method, the quantizer scale of 5 is used during a session. On the other hand, the quantizer scale of 6 is applied in the case of the method with the FEC since the actual video rate is suppressed by FEC packets. Figure 6 shows the rate variation of the bandwidth allocated to the video data, R_{video}. Trajectories of perceived video quality under the lossy environment are shown in Fig. 7. Through these figures, the target packet loss probability P_{target} is set to 10^{-3} or 10^{-4}.

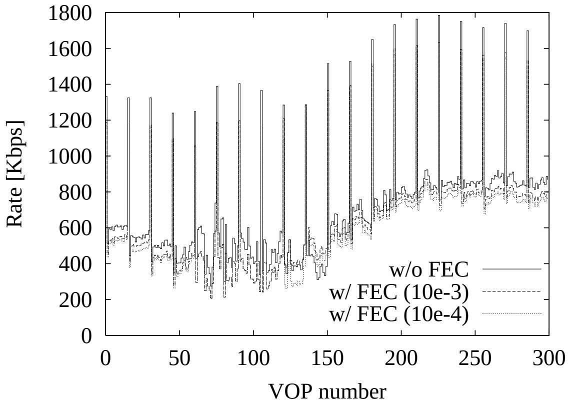

Figure 6: Video rate variation (w/ FEC vs. w/o FEC)

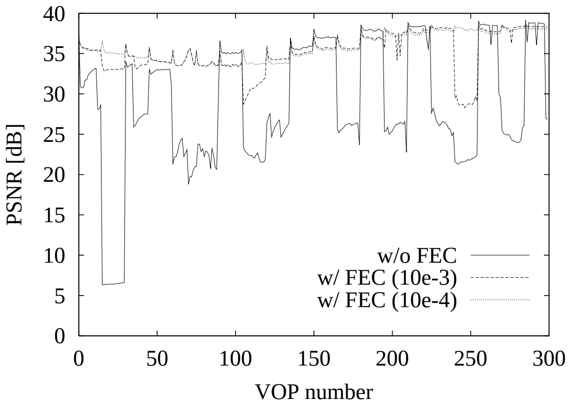

Figure 7: Video quality variation (w/ FEC vs. w/o FEC)

We can see that the FEC technique is effective for protecting the video quality from degradation caused by packet loss. Although packets are lost at probability 10^{-2} to 10^{-1} in the network, it is suppressed toward 10^{-6} to 10^{-2} with a help of FEC. However, quality degradation due to loss of BL data is noticeable when FEC is targeting at the probability of 10^{-3}. This implies that the protection targeted at 10^{-3} is not effective enough against instantaneous but serious packet loss.

When we apply the target loss probability of 10^{-4}, the bandwidth available to the video data becomes obviously lower than that in the case of $P_{target} = 10^{-3}$ because of the increased FEC redundancy. As a result of increased redundancy, the video quality becomes lower for loss-free GOVs. However, trajectories of quality are kept stable owing to a higher level of protection. Thus, on the basis of the above observations, we conclude that choosing the target packet loss probability of 10^{-4} is effective to protect the video packet from loss at the sacrifice of only a slight quality degradation at most 1.5 dB in our experiments. Examples of images displayed on the monitor are shown in Fig. 8 for the method without FEC (left) and with FEC (right), respectively.

5. CONCLUDING REMARKS

In this paper, we have proposed the video quality adjustment mechanism which accomplishes the high-quality video transfer under the lossy environment. Our mechanism employs the FEC to protect video packets from loss and dynamically regulate video rate in accordance with the network condition.

Although the effectiveness of our mechanism is verified through

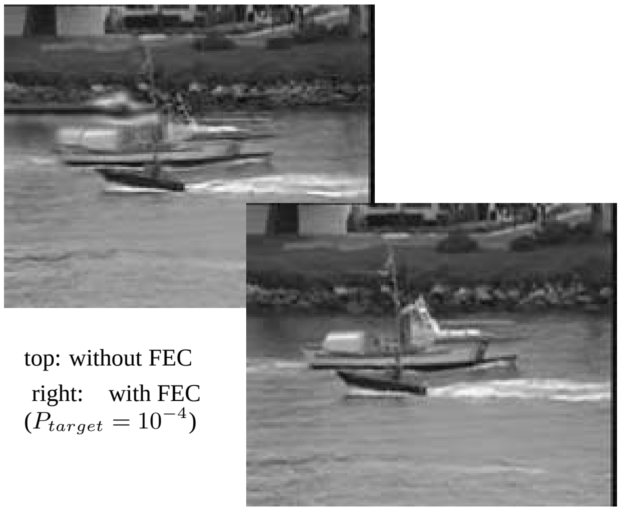

Figure 8: Displayed VOP 70

simulation experiments, there still remains some research issues. When the video application employs TFRC, the video data injected into the transport layer should be smoothed to fit to the TFRC rate, but such a smoothing delay is not considered in this paper. Furthermore, we should consider the improved mechanisms to change the quantizer scale dynamically in accordance with the packet loss probability, in order to attain the real-time video transfer with high and stable video quality under both loss-free and lossy conditions.

6. REFERENCES

[1] Jitendra Padhye, Victor Firoiu, Don Towsley, and Jim Kurose, "Modeling TCP throughput: A simple model and its empirical validation," in *Proceedings of ACM SIGCOMM'98*, September 1998, vol. 28, pp. 303–314.

[2] Jitendra Padhye, Jim Kurose, Don Towsley, and Rajeev Koodli, "A model based TCP-friendly rate control protocol," in *Proceedings of NOSSDAV'99*, June 1999.

[3] Mark Handley, Jitendra Padhye, Sally Floyd, and Jorg Widmer, "TCP Friendly Rate Control (TFRC): Protocol Specification," Internet-Draft draft-ietf-tsvwg-tfrc-03.txt, work in progress, July 2001.

[4] Naoki Wakamiya, Masaki Miyabayashi, Masayuki Murata, and Hideo Miyahara, "MPEG-4 video transfer with TCP-friendly rate control," in *Proceedings of IFIP/IEEE MMNS2001*, October–November 2001, vol. 1, pp. 29–42.

[5] Hayder Radha and Yingwei Chen, "Fine-granular-scalable video for packet networks," in *Proceedings of Packet Video'99*, April 1999.

[6] Hayder Radha, Mihaela van der Schaar, and Yingwei Chen, "The MPEG-4 fine-grained scalable video coding method for multimedia streaming over IP," *IEEE Transactions on Multimedia*, vol. 3, no. 1, pp. 53–68, March 2001.

[7] Uwe Horn, Klaus Stuhlmüller, M. Link, and B. Girod, "Robust Internet video transmission based on scalable coding and unequal error protection," *Image Communication*, vol. 15, no. 1–2, pp. 77–94, September 1999.

[8] Qian Zhang, Wenwu Zhu, and Ya-Qin Zhang, "Resource allocation for multimedia streaming over the Internet," *IEEE Transactions on Multimedia*, vol. 3, no. 3, pp. 339–355, September 2001.