NETWORK ADAPTIVE RATE CONTROL FOR TRANSCODER

Takayasu Takaoka†, Isao Nagayoshi‡, Tsuyoshi Hanamura‡, and Hideyoshi Tominaga†

† GITS, Waseda University, 29-7 building 1-3-10 Nishi-Waseda, Shinjuku-ku, Tokyo 169-0051 Japan
‡ Media Glue Corp. Ramdax Bldg. 8th floow, 2-4-12 Okubo, Shinjuku-ku, Tokyo 169-0072 Japan
phone: 03-5286-3385, fax: 03-5273-7368
E-mail: {takaoka, isao, hana, tominaga}@tom.comm.waseda.ac.jp

ABSTRACT

In this paper, we propose a rate control scheme for MPEG video transcoder over the network which has a variable bandwidth. MPEG video transcoder can reduce the required bitrate by using re-quantization in the DCT domain. For video streaming service over the network, bitrate of a video stream needs to adjust dynamically to available bandwidth. We focus on the effect of congestion control on data-processing speed in transcoder. Here, the data-processing speed means the amount of processed data per unit time in transcoding. This effect can be used to estimate the available bandwidth without a special implementation on the network. First, we clarify the relation among the data-processing speed of the transcoder, the target bitrate for the transcoder, and the channel bandwidth. Next, we propose a dynamic rate control scheme based on the relation mentioned above. Finally, by simulating experiments, we show the effectiveness of the proposed scheme.

1. INTRODUCTION

Since a broadband network is extending, the demand to access multimedia data such as video streaming. However, the Internet is a best-effort network whose traffic fluctuates over time. The quality of service that users wish for video streaming can not be assured. Therefore, it is necessary to control a bitrate of video for keeping playback stable at the client terminal and maintaining the decoded video quality, while the channel throughput is fluctuated. For video streaming service over the Internet, in existing technology, the same streams of several qualities are ready and delivered at constant bitrate. So the streaming server requires the enormous quantity of storage capacity. In comparison with the above technology, transcoder[1, 2, 3, 4] has been studied as one of the effective solutions to realize the one source and multiple-use on the video streaming. And it can recompress the compressed video signals in order to adjust to available bandwidth. However, transcoder which has an adaptive rate control mechanism for the video streaming has not been discussed enough.

In this paper, we propose a rate control scheme for the transcoder for the video streaming. Our scheme uses an effect of congestion control on data-processing speed in the transcoder to estimate available bandwidth, and controls a bitrate of video streaming for keeping playback stable at the client terminal. And, we evaluated the performance through simulations.

2. EFFECT OF CONGESTION CONTROL ON DATA-PROCESSING SPEED IN TRANSCODER

In this section, we define the assumed transmission model. And, we discuss about the relation among the data-processing speed of the transcoder, the target bitrate for the transcoder (the conversion bitrate of video), and the channel bandwidth.

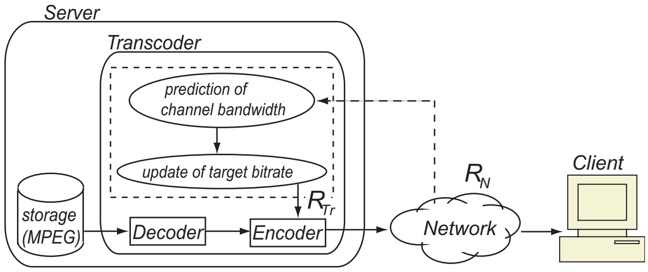


Figure 1: Assumed transmission model

2.1 Assumed Transmission Model

First, the assumed transmission model is shown in Fig.1.It is assumed that video streams which have already encoded are stored in the server, the transcoder as a bitrate reducer is implemented on the server. Here, we use the core algorithm of the transcoder in [2], which can convert the compressed video signal to the required quality in the DCT domain. Video streams whose bitrate are maintained by the transcoder are transmitted to the client.

We focus on the effect of congestion control on data-processing speed of the transcoder. The data-processing speed of the transcoder changes depending on the channel throughput[5]. The higher the throughput is, the faster the data-processing speed is. So, it is possible to understand indirectly the channel bandwidth by comparing the transcoding processing speed. And then, the estimated channel bandwidth is used to update the target bitrate for the transcoder.

2.2 Data-processing Speed of Transcoder Based on SCR

We clarify the relation among the data-processing speed of the transcoder, the target bitrate of the transcoder, and the channel bandwidth. In this case, we use System Clock Reference (SCR) timestamps which are set in system Pack headers and control synchronization between encoder and decoder[6]. SCR timestamps use 90kHz clock (MPEG-1) or 27MHz clock (MPEG-2). These frequency is defined as f. Transcoding elapsed time is defined as t, and playback time of processed data from start of transcoding to t is defined as τ. τ is calculated by (1).

$$\tau = \frac{SCR}{f} \tag{1}$$

We define a channel bandwidth as $R_N[bps]$, and a target bitrate for the transcoder as $R_{Tr}[bps]$. If it is assumed that

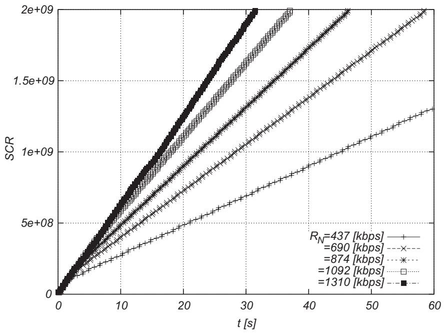


Figure 2: Relationship between transcoding time and SCR (fixed R_N)

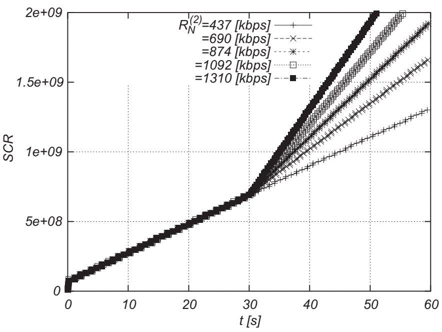


Figure 3: Relationship between transcoding time and SCR (variable R_N)

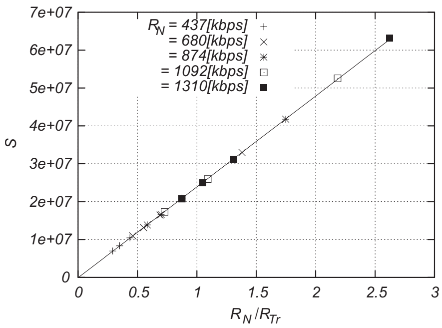


Figure 4: Relationship between R_N/R_{Tr} and S

an actual amount of the transmission data is equal to R_N, R_N is calculated by (2).

$$R_N = \frac{R_{Tr} \times \Delta\tau}{\Delta t} \tag{2}$$

In transcoding nth frame, $i_1[bit]$ denotes the first data position in the nth frame, and $i_2[bit]$ denotes the end one in the nth frame. $\Delta\tau$ is calculated by (3) using (1).

$$\Delta\tau = \frac{SCR(i_2) - SCR(i_1)}{f} \tag{3}$$

From (2) and (3), (4) can be derived.

$$\frac{R_N}{R_{Tr}} = \frac{SCR(i_2) - SCR(i_1)}{\Delta t \times f} \tag{4}$$

From (4), it is clear that R_N/R_{Tr} is related to SCR and t. So, R_N can be estimated by using R_{Tr}, SCR, and t.

2.3 Preparative Experiment

We investigate the data-processing speed of the transcoder in order to prove the validity of the relation derived in (4) from experiments. The experimental condition is as follows. Original video is encoded in an MPEG-1 video at 4:2:0 format at $1.8[Mbps]$ with group of pictures (GOP) structure of N=15 and M=3. The video stream was transmitted from the server to the client for 60 seconds.

We performed two kinds of experiments.

(1)Fixed channel bandwidth

The first one is that R_{Tr} is fixed ($= 500[kbps]$), R_N is fixed. The channel bandwidth R_N is shown as follows.

$$R_N = 437, 690, 874, 1092, 1310[kbps] \qquad (fixed) \tag{5}$$

(2)Variable channel bandwidth

The second one is that R_{Tr} is fixed ($= 500[kbps]$), R_N is changed at 30 seconds. The channel bandwidth R_N is shown as follows.

$$R_N = \begin{cases} 437[kbps] & (0 \rightarrow 30[s]) \\ 437, 690, 874, 1092, 1310[kbps] & (30 \rightarrow 60[s]) \end{cases} \tag{6}$$

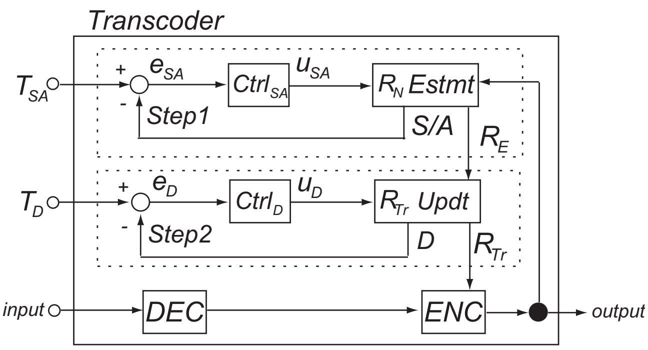


Figure 5: Block diagram of the proposed rate control scheme

We show the experimental results about the value of SCR in Fig.2, and Fig.3. From Fig.2, these lines can be approximated to the linear function. From Fig.3, after the channel bandwidth changes, there is the same characteristics to Fig.2.

And so, each slope of the straight line from Fig.2 is defined as S. It means the increasing rate of processing time, which is amount of processed data per unit time in transcoding. The relation among S and R_N/R_{Tr} is shown in Fig.4. From Fig.4, (7) can be derived.

$$S = A \times \frac{R_N}{R_{Tr}} \tag{7}$$

Here, A is a constant of proportion. From (7), the validity of the relation derived in (4) was clarified.

3. PROPOSED RATE CONTROL SCHEME

In this section, we propose a dynamic rate control scheme based on (7).

The purpose of this proposed scheme is to control the target bitrate for the transcoder by predicting the channel bandwidth in order to keep playback of video stream stable at the client terminal.

Fig.5 is a block diagram of the proposed rate control scheme. Proposed scheme consists of 2 steps. In Step 1, we predict a channel bandwidth based on the fact discussed in Section 2 by using a proportional control. In Step 2, we update the target bitrate for the transcoder based on the predicted channel bandwidth by using a proportional control.

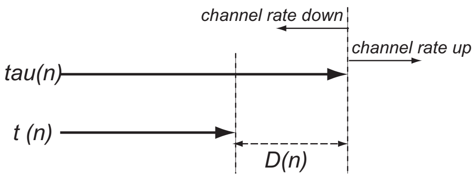


Figure 6: Concept of $D = \tau - t$

3.1 Step 1 : Prediction of Channel Bandwidth

In this step, we predict a channel bandwidth based on (7). In Fig.5 $R_N Estmt$ is the block which calculates an estimation value of $R_N(n)$. In the proportional control block $Ctrl_{SA}$, $S(n)/A$ is stabilized at the target value T_{SA}.

First, we get the $S(n)/A$ in transcoding nth frame using least-square method. $S(n)$ is calculated by (8) using $N(> 1)$ samples.

$$S(n) = \frac{N \sum_{i=0}^{N-1} t(n-i) SCR(n-i) - \sum_{i=0}^{N-1} t(n-i) \sum_{i=0}^{N-1} SCR(n-i)}{\sum_{i=0}^{N-1} \{t(n-i)\}^2 - \left(\sum_{i=0}^{N-1} t(n-i) \right)^2} \tag{8}$$

And then, proportional coefficient $K_{SA}(< 1)$ is multiplied by the difference of between $S(n)/A$ and T_{SA}. This value is defined as $u_{SA}(n)$, which is shown as (9). The lower K_{SA} is, the more stable R_N is.

$$u_{SA}(n) = \left\{ T_{SA} - \frac{S(n)}{A} \right\} \times K_{SA} \tag{9}$$

$u_{SA}(n)$ is passed to $R_N Estmt$, then $R_N(n)$ is predicted based on (7). The estimation value of $R_N(n)$ is defined as $R_E(n)$, $R_E(n)$ is shown as (10).

$$R_E(n) = \left\{ T_{SA} - u_{SA}(n) \right\} \times R_{Tr}(n) \tag{10}$$

3.2 Step 2 : Update of Target Bitrate

In this step, we update the target bitrate for the transcoder $R_{Tr}(n+1)$ by using $R_E(n)$ from (10). Before we discuss about the update of the target bitrate, we define the parameter that prevents interruption of stream as follows.

3.2.1 Parameter that Prevents Interruption of Stream

In order to prevent interruption of stream, it is necessary to consider how much data is transcoded. If stream playback time of the transcoded data is less than the transcoding time, the video stream becomes interrupted at the client terminal.

For problem-solving mentioned above, it is necessary to be processed with $\tau - t$ maintaining the fixed time. And so, this $\tau - t$ value is defined as D in order to avoid an interruption of stream. If R_{Tr} is below R_N, transcoder continues with the future frames. And then, the increasing rate of the stream playback time τ is greater than that of the transcoding time t. That increases D. On the other hand, if R_{Tr} is above R_N, there is an opposite characteristics. This concept is shown in Fig.6.

3.2.2 Target Bitrate for the Transcoder

In Fig.5 $R_{Tr}Updt$ is the block which updates a value of $R_{Tr}(n)$. In the proportional control block $Ctrl_D$, $D(n)$ is stabilized at the target value T_D.

First, we get the $\tau(n)$ and $t(n)$ in transcoding nth frame. And then, proportional coefficient K_D ($K_D < 1$) is multiplied by the difference of $D(n) = \tau(n) - t(n)$ and T_D. This value is defined as u_{SA}, which is shown as (11). The lower K_D is, the more stable $R_{Tr}(n)$ is.

$$u_D(n) = \Big[T_D - \Big\{ \tau(n) - t(n) \Big\} \Big] \times K_D \tag{11}$$

$u_D(n)$ is passed to $R_{Tr}Updt$, then $R_{Tr}(n)$ is updated by using (10) and (11). $R_{Tr}(n+1)$ is shown as (12).

$$R_{Tr}(n+1) = \Big\{ 1 - u_D(n) \Big\} \times R_E(n) \tag{12}$$

4. SIMULATING EXPERIMENTS

In this section, we evaluate (i) stability of the target bitrate for the transcoder to channel bandwidth, and (ii) stability of the parameter D that prevents interruption of stream, (iii) received data at the client terminal, when the channel bandwidth changes dynamically.

The experimental condition is almost same as the one in section 2.3. Difference of 2.3 is as follows. The video stream was transmitted from the server to the client for 180 seconds. Then, the channel bandwidth is shown as follows.

$$R_N = \begin{cases} 1160[kbps] & (0 \rightarrow 60[s]) \\ 732[kbps] & (60 \rightarrow 120[s]) \\ 1160[kbps] & (120 \rightarrow 180[s]) \end{cases} \tag{13}$$

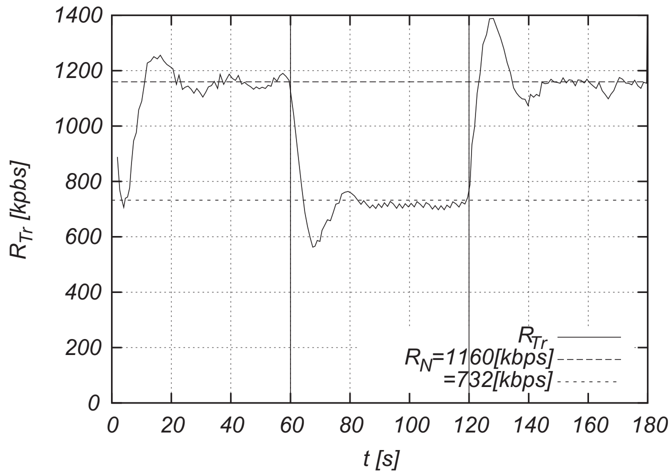
The parameters of proposed rate control are set as follows. $T_{SA} = 1.0$, $T_D = 2.5$, $K_{SA} = 0.2$, $K_D = 0.08$. The parameter values used here are chosen from the ones that gave good performance in many simulations we performed. We recognize that the parameter setting is one important issue to be investigated in the near future.

4.1 Stability of the Target Bitrate for the Transcoder

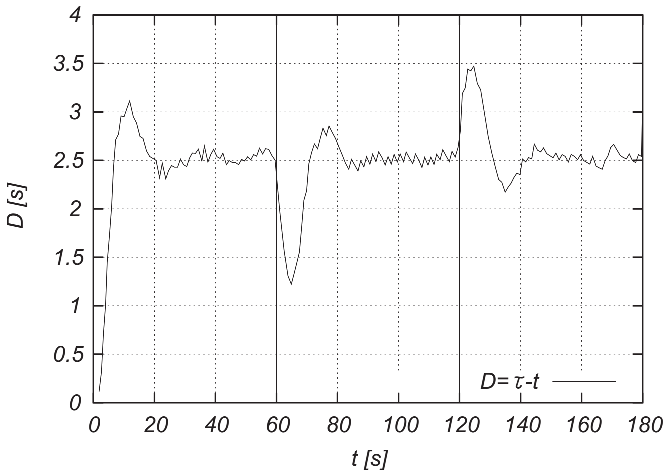
We show the experimental results about the behavior of R_{Tr} by the change of R_N in Fig.7(a). From Fig.7(a), when the channel bandwidth changes at 60 seconds and 120 seconds, proposed scheme can adjust the target bitrate for the transcoder to available bandwidth quickly. R_{Tr} is less than R_N from start to 10 seconds, it is caused by convergence of D to the target value T_D. And, because the value of R_{Tr} in relation to R_N can be stabilized within 100[kbps], the stream with stabilized quality can be provided.

4.2 Stability of the Parameter that Prevents Interruption of Stream

We show the experimental results about the behavior of D by the change of R_N in Fig.7(b). From Fig.7(b), proposed scheme can adjust D to the target value T_D quickly as well as the above-mentioned results. And, a big gap is not caused for the rapid change of R_N. Because of feedback control of $D(= \tau - t)$ (> 0) to keep it fixed, stream playback time τ of the processed data is always greater than the transcoding time t. So, the data that should reach at the playback time can be transmitted to the client without the stream's becoming interrupted. When the R_N decrease at 60 seconds,



(a) Relationship between transcoding time and R_{Tr}



(b) Relationship between transcoding time and D

Figure 7: Response of the proposed scheme at the server (R_N:1160→ 732→ 1160$kbps$)

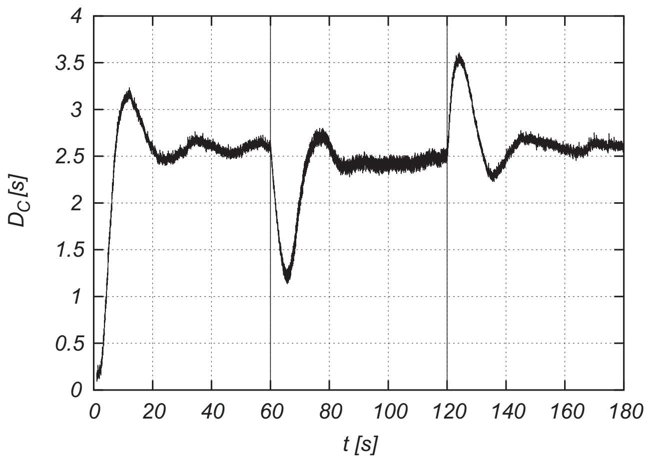


Figure 8: Relationship between transcoding time and D_C (at the client terminal)

D also decrease. But D doesn't reach 0. It means that the interruption of stream can be avoided.

4.3 Received Data at the Client Terminal

In 3.2.1, we defined D to prevent the interruption of stream. Here, we evaluate the influence of D at the client terminal. At the client terminal, the time when the nth frame are reproduced is defined as $t_C(n)$, and playback time of received data from start of the reception to $t_C(n)$ is defined as τ_C. If $\tau_C(n) - t_C(n)$ is maintained the fixed time (> 0), it means that the interruption of stream can be avoided. This $\tau_C(n) - t_{C(n)}$ value is defined as $D_C(n)$. We show the experimental results about the behavior of $D_C(n)$ at the client terminal in Fig.8. Because the behavior of $D_C(n)$ in Fig.8 is as same as that of $D(n)$ in Fig.7(b), the interruption of stream can be avoided.

5. CONCLUSION

This paper described using a video transcoder as a bitrate reducer and proposed a rate control scheme for video streaming. Proposed scheme uses the data-processing speed of the transcoder to estimate channel bandwidth and adjusts target bitrate accordingly. Simulating results show that our scheme prevents interruption of stream and is effective for video streaming. We did perform experiments over operating network and the experimental results support the above findings. If some kind of protection mechanism such as retransmission control and error correction function is applied to the proposed scheme, a more effective control becomes possible. We recognize that the above-mentioned items is one important issue to be investigated in the near future.

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