Adaptive Redundancy Control for Systematic Erasure Code Based Real time Data Transmission in Internet.

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Abstract— In this paper, we propose an adaptive redundancy control scheme for systematic erasure coding scheme to transmit real time data in Internet. Since recovery rate of packet loss depends on the amount of redundancy data, we use the redundancy estimation algorithm which is considering burstness of loss. Gilbert model is used for modeling the loss process and adjusting the number of redundant packet. Analysis and experimental results show that proposed scheme is efficient for loss recovery in consecutive loss environments like Internet.

I. INTRODUCTION

In the past few years, owing to the growth of the network bandwidth in Internet and the improvement of transmission techniques, the teleconferencing system has been researched and popularized very much. However, in spite of remarkable advance of network technique, real time audio applications over best-effort networks such as Internet suffer from frequent packet loss by the traffic congestion\cite{1}\cite{2}. To reduce the effect of packet loss, several researches have been done\cite{3}. The Forward Error Correction (FEC) technique is useful as the packet loss resilient mechanism. In FEC mechanism, the lost information could be recovered without retransmission by transmitting redundant information in the initial transmission. This technique is appropriate for real time applications, because it reduces the time needed to recover the lost packet, and the additional channel is not needed for retransmission.

The systematic erasure code is one of the FEC methods used in various types of media (audio, video, etc.). In this technique, redundant packets are generated from original data packets based on linear block code algorithm. Linear block codes have been previously used for both error and loss recovery, but in transport or application layers of communication systems, they can be employed for loss recovery, since loss location is easily gripped using sequence number of Real time Transport Protocol (RTP). For usage of erasure code based FEC, a RTP payload format is needed\cite{6}. In this paper, we proposed an adaptive redundancy control method using feedback information.

This paper is organized as follows. In section II, as a related work, we describe the erasure code performance for consecutive packet loss. The analysis of systematic erasure code and redundancy control algorithm is presented in Section III. In section IV, we discuss numerical results. Finally, we make conclusions in Section V.

II. ERASURE CODE PERFORMANCE FOR CONSECUTIVE LOSS

In this section we describe erasure codes and performance analysis method of erasure code for consecutive loss\cite{7}. The basic principle of the erasure coding is explained as following: If we assume that \( k \) data packets are encoded to \( n \) packets, Transmission Group (TG) size is \( n \) and we name this coding mechanism as \((n, k)\) code. Receiver can recover \( k \) original data packets based on any \( k \) packets out of \( n \) encoded packets. In the systematic erasure code, \( k \) data packets and \( n - k \) redundant packets for loss recovery are inside TG\cite{8}. We call \( n - k \) redundant packets as parity packets. Erasure code method is relatively simple in view of processing than redundant transmission technique using multi-rate audio vocoder\cite{2}. However, it needs relatively high bandwidth overhead and brings out a playout delay of block coding \cite{8}.

In the case that packet loss is identical and independent, the probability that the length of consecutive packet loss becomes \( n \) is \( p^n(1 - p) \) and the mean length of packet loss is \((1 - p)^{-1}\). If mean packet loss rate is \( p \), and mean loss length is not \((1 - p)^{-1}\), Gilbert model can be used\cite{4}. This model is shown in Fig. 1. Gilbert model presents the consecutive packet loss process based on on-off Markov process\cite{7}. In Fig. 1, \( L \) indicates the state of packet loss and \( R \) indicates the non-loss state. \( 1 - \beta \) is the conditional probability losing the next packet when a packet...
III. PERFORMANCE ANALYSIS OF SYSTEMATIC ERASURE CODE AND REDUNDANCY CONTROL

A. Performance analysis of systematic erasure code

The most important part in packet transmission using erasure code is the decision of code rate according to the loss characteristics, and we know the recovery rate of the erasure code due to the loss characteristics. The redundancy control method that we want to suggest in this section could be used for systematic erasure code. In Fig. 2, the size of TG is n and the number of date packet is k. If a data packet in a TG is lost, and the receiver could not receive more than k – 1 packets out of n packets in the TG, the lost packet is not recovered during the process of decoding.

Let individual packet loss probability $P_{Li}$ be a loss probability of i-th packet of a TG in (n,k) code. $P_{Li}$ is the probability that i-th packet of the TG is lost and the number of received packets in the TG is less than k. Then, the mean data packet loss rate $P_{DL}$ is given by

$$P_{DL} = \frac{1}{k} \sum_{i=1}^{k} P_{Li}$$

(11)

In equation (11), let $PK_i$ be the i-th packet of the TG, $N_r$ be the number of received packets in the TG, and $N_{r+}$ be the number of received packet after i-th packet transmission, and $N_{r-}$ be the number of received packets until i-th packet transmission. Then, $P_{Li}$ is given by

$$P_{Li} = \sum_{j=0}^{k} D(i,j,L)P(N_{r+} < k - j | N_{r-} = j, PK_i is lost)$$

$$= \sum_{j=0}^{k} D(i,j,L)D^*(i,j,k,n)$$

(12)

In equation (12), $D^*(i,j,k,n)$ is the probability that the number of received packet after i-th packet of the TG is less than k – j. This probability is divided into two cases. The first case is that the number of lost packets is so big before the transmission of i-th packet that it could not be recovered, although all packets are transmitted without loss after i-th packet. In this case, $j + n - i$ is less than k, and $D^*(i,j,k,n)$ becomes 1, because i-th packet can not be recovered regardless of the number of received packets after i-th packet in the TG. The second case is that $j + n - i$ is the same as k or larger than k. In this case, $D^*(i,j,k,n)$ is the probability that received packets after i-th packet is less than k – j, if i-th packet is lost. Using the time reversible property of Gilbert model, $D^*(i,j,k,n)$ in the second case is given by

$$D^*(i,j,k,n) = \sum_{l=0}^{k-j-1} P(N_{r+} = l | PK_i is lost)$$

(13)
\[ D(i, j, k, n) = \begin{cases} \frac{1}{k} \sum_{l=0}^{j-1} \binom{n-i-l}{j-1} \frac{D(n-i-l)}{P_L} & \text{if } j+n-i < k \\ \sum_{l=0}^{j-1} \binom{n-i-l}{j-1} \frac{D(n-i-l)}{P_L}, \text{otherwise} \end{cases} \]

(14)

Using the above equations, the mean data packet loss rate after recovery \( P_{DL} \) is given by

\[ P_{DL} = \frac{1}{k} \sum_{i=1}^{k} P_{L_i} \]

(15)

\[ = \frac{1}{k} \sum_{i=1}^{k} \sum_{j=0}^{i} D(i, j, L) D^*(i, j, k, n) \]

B. Redundancy Control algorithm

It is important to adjust the number of data packet \( k \) in the TG as well as the entire TG size that combines data packet with parity packet when the packet is transmitted by erasure code. The number of data packet \( k \) in the TG is normally given by conditions like transmission delay and packetization time. In this paper, we assume that original data packet is transmitted without any delay, and redundancy packet is generated after data packet is transmitted. So, the relation of \( k \) and delays is given as following.

\[ (k + 1) \cdot D_p + D_{network} < D_{QoS} \]

(16)

In equation (16), \( D_p \) is the packetization interval of original data packets, \( D_{network} \) is the network delay, and \( D_{QoS} \) is the transmission delay that is given as QoS. \( k \) must satisfy equation (17).

\[ k < \frac{D_{QoS} - D_{network}}{D_p} - 1 \]

(17)

Also, TG size \( n \) should be estimated based on the characteristics of packet loss. In the proposed algorithm, feedback information that is transmitted by feedback packet like RTCP is the state transition probability of Gilbert model. If \( N_{total} \) is the number of total packets that is transmitted, \( N_{loss} \) is the number of lost packets, and \( N_{on} \) is the length of consecutive packet loss, the loss rate \( P_{loss} \) and the mean length of consecutive loss \( L_{ave} \) are obtained by

\[ P_{loss} = \frac{N_{loss}}{N_{total}} \]

(18)

\[ L_{ave} = \frac{N_{loss}}{N_{on}} \]

(19)

State transition probability of Gilbert model could be derived from \( P_{loss} \) and \( L_{ave} \) as following.

\[ \alpha = \frac{1}{L_{ave}} \]

(20)

\[ \beta = \frac{\alpha \cdot P_{loss}}{1 - P_{loss}} \]

(21)

We can get \( k \) from \( D_{QoS} \) and \( D_{network} \) using the equation (17), and we can estimate the TG size \( n \) from feedback information \( \alpha \) and \( \beta \) using equation (15). If we assume that \( k \) is given, the sequence of redundancy estimation is as follows:

1) Get the transition probability \( \alpha \) and \( \beta \) from feedback information.
2) Let \( n = k \).
3) While \( P_{DL} < \text{Target loss rate} \) increase \( n \) and do the following.
   a) Calculate individual \( P_{L_i} \) using (12).
   b) Calculate \( P_{DL} \) using (15).
4) Stop.

Let \( NMEA_{current} \) be the estimated TG size from the feedback data. Then, smoothed TG size \( NSMO_{current} \) can be obtained from \( NMEA_{current} \) and previous smoothed TG size \( NSMO_{before} \) as following.

\[ NSMO_{current} = SMO \cdot NSMO_{before} + (1 - SMO) \cdot NMEA_{current} \]

(22)

In equation (22), \( SMO \) is the smoothing factor, and \( SMO \) has the value between 0 and 1. For considering the rapid change of loss characteristics, \( NDEV_{current} \) which is the deviation of \( NMEA_{current} \) can be obtained as following.

\[ NDEV_{current} = SMO \cdot NDEV_{before} + (1 - SMO) \cdot |NMEA_{before} - NMEA_{current}| \]

(23)

Then, the TG size for real data transmission \( NSND \) can be calculated as following.

\[ NSND = \lfloor NSMO_{current} + CDEV \cdot NDEV_{current} \rfloor \]

(24)

In equation (24), \( CDEV \) is the deviation factor.

IV. NUMERICAL AND SIMULATION RESULTS

A. Systematic erasure code performance

Fig. 3 is the packet loss rate according to the mean length of consecutive loss and loss rate. In this figure, the relative mean length of consecutive loss is the relative value of mean length of consecutive loss compared with the mean length of consecutive loss in the case that each packet loss is independent and identical. If (5,3) erasure code is applied to the case that the transmission loss rate is 12%, and each packet loss is independent and identical (relative mean length of consecutive loss is 1), the loss rate of data packet after loss recovery is 0.9%. But, if the transmission loss rate is 12%, and the relative mean length of consecutive loss is 2, the loss rate of data packet after loss recovery is 6%. From this figure, we can see that the packet loss rate after recovery is considerably increased when the mean length of consecutive loss becomes larger.

Fig. 4 shows packet loss rate after loss recovery according to the mean length of consecutive loss during the transmission as well as various erasure codes. This figure shows that packet loss probability is decreased as TG size becomes larger. For example, when (4,2) erasure code is used, the transmission loss rate is 10%, and mean length of consecutive loss is 1.5, the packet loss rate after recovery is 1.8%. But, when (8,4) erasure code is used in the same case, the packet loss rate after recovery is 0.5%.
Fig. 3. Packet Loss Rate according to relative mean length of consecutive loss (5.3) systematic erase code.

Fig. 4. Packet Loss Rate according to TG size ($P_{loss} = 0.1$).

Table I shows the estimated TG size $NMEA$ according to the target loss rate, loss rate $P_{loss}$ and mean length of consecutive loss $L_{ave}$ when $k$ is 3. In this table, if $k$ is 3, target loss rate is 1%, packet loss is 0.12, and mean length of consecutive loss is 1.2, the estimated TG size $n$ is 6. Also, as target packet loss is increased, TG size must be increased.

<table>
<thead>
<tr>
<th>Target loss rate</th>
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<th>Mean length of consecutive loss</th>
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<tr>
<td></td>
<td>1%</td>
<td>0.6</td>
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<td>1%</td>
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<td>1%</td>
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is 12.94% and mean length of consecutive loss is 1.2563. From these results, we know that mean length of consecutive loss is decreased as packetization interval becomes larger.

Fig. 5. Number of consecutively lost packets (packetization interval of PCM audio data is 20 msec).

C. Simulations of Redundancy Control Algorithm

Using the measured packet loss processes that is shown in previous subsection, we perform the simulations of redundancy control algorithm. The mean loss rate after loss recovery according to SMO and CDEV is shown in Fig. 7 and 8. In this simulation, we assume that the transmission interval of feedback packet is 500 msec, the feedback delay of feedback packet is 250 msec, and $k$ is 3. In Fig. 7, when $SMO$ is 0.7 and $CDEV$ is 0.8, loss rate after recovery is 0.8%. From these figures, we can know that the loss rate is decreased as SMO and CDEV becomes larger, because the system stability could be improved as SMO becomes larger, and the number of transmitted redundancy packets is increased as CDEV becomes larger. So, CDEV might be small for reducing redundancy.
TABLE II
PACKET LOSS CHARACTERISTICS.

<table>
<thead>
<tr>
<th>Packetization interval</th>
<th>Loss rate</th>
<th>Number of burst</th>
<th>Mean length of consecutive loss</th>
</tr>
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<tbody>
<tr>
<td>20 msec</td>
<td>9.62%</td>
<td>486</td>
<td>1.9794</td>
</tr>
<tr>
<td>60 msec</td>
<td>12.94%</td>
<td>1030</td>
<td>1.2563</td>
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V. CONCLUSIONS

We have described the redundancy control algorithm for systematic erasure code based real time data transmission in Internet. This mechanism causes the bandwidth overhead due to the additional transmission of redundancy data block. So, it is important to decide how much redundant data is transmitted. Therefore, in this paper, we adjust the size of TG adaptively so that the target loss rate as well as audio application requests could be satisfied. For this purpose, we used Gilbert Model that is suitable for modeling the characteristics of consecutive packet loss. Measurement and simulation results show that adaptive redundancy control method can be used as an efficient loss recovery algorithm for real time data transmission in Internet. The proposed techniques could be used for reliable transmission of audio data for interactive audio applications such as Teleconferencing system, Internet phone, remote collaboration and distance learning.

REFERENCES


Fig. 6. Number of consecutively lost packets (packetization interval of PCM audio data is 60 msec).

Fig. 7. Loss rate after loss recovery according to CDEV and SMO (packetization interval of PCM audio data is 20 msec, and target loss rate = 1% ).

Fig. 8. Loss rate after loss recovery according to CDEV and SMO (packetization interval of PCM audio data is 60 msec, and target loss rate = 1% ).