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ผลงานทางวิชาการที่ได้รับการตีพิมพ์

1. P.MOUNGNOUL,A.KANWIBOON,K.DEJHAN, “Packet Loss Recovery in Media Specific FEC Audio Transmission by Least Square Method,” in *CITA’07 Fifth International Conference on Information Technology in Asia 2007, Social Computing : Engaging Communities,* vol.5, Kuching, Sarawak, Malaysia, July. 2007, pp. 300-302.


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Packet Loss Recovery in Media Specific FEC Audio Transmission by Least Square Method

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Abstract - *This paper proposes a packet loss recovery technique in audio stream at a sender side. The adaptive error control in the Internet telephony system uses this packet recovery scheme based on the media specific Forward Error Correction (FEC). By adapting with the least square method used recovery algorithm with the best fitting curve of a reward value. This technique can estimate the trend of outcomes, and improve the adjustment of redundant packets of the recovery algorithm. The results of experiments show that the adaptive error control algorithm can decrease the packet loss rate.*

Keywords: Internet telephony, packet loss recovery, least square method

1 Introduction

The current internet normally uses the best-effort service model. This cannot guarantee the quality of service (QoS). As a result the transmission of audio traffic over the internet is always affected by the packet loss problem. The FEC (Forward Error Correction) is one of the error recovery mechanisms that can reduce the effect of packet loss with the low latency [2]. FEC technique adds some redundancy that increases the usage bandwidth. The adaptive FEC-based algorithm which can set the quantity of audio redundancy based on the number of packet lost, therefore, is required. [4]

There are several proposed algorithms that can do this task. Some algorithms have to use a loss report forecast or estimate the reward parameters for the next interval time [2]. The least square method is very popular technique employed in computing estimation parameters and data curve fitting. This method can be used in part of increase or decrease combination number (pattern of redundant data in packet) for the adaptive FEC-based algorithm. In addition, the better forecasting can be performed because the result of this forecasting will be used in the optimization of the recovery mechanism.

The SES (single exponential smoothing) technique is a simple and pragmatic approach to forecasting, whereby the forecast is constructed from the exponentially weighted average of pass observation [6]. If the packet loss prediction use bounded adaptive smoothing constant from the forecast loss rate in the next interval.

This paper studies the performance of the media specific FEC-based algorithm by increasing the number of combination data and comparing the performances. The adaptive FEC control algorithm developed in [5] CNR (Centre for Network Research) algorithm is studied, and the reward value of this algorithm is forecasted using the least square and exponential method. Process of quantitatively is estimated and the trend of the outcomes known as *regression* or curve fitting becomes necessary [1]. The develop mechanism attempts to eliminate or to minimize the impact of packet loss and end-to-end delay on the quality of the delivered audio to the destinations. The effect of the variability parameters that use in process of each algorithm for improve audio transmission in the future is employed.

2 Packet Recovery Algorithm by Least Square Method

2.1 CNR Algorithm

CNR algorithm [6] is a media specific FEC-based algorithm. This algorithm improved from Bolot [4] algorithm by increase the combination number of audio redundancy in each period. G.723.1 codec is the most widely-use in IP telephony system because it has low bit rate enough (6.3 kbps), this can reduce waste time from Bolot algorithm.

CNR algorithm requires RTP (Real-time Transport Protocol) for send audio data from the sender side and RTCP (Real-time Transport Control Protocol) use for report the loss rate (packet loss rate before and after the reconstruction) of the received data packet to the sender [3]. The redundant packet used for recovery the lost packets and the better combination of codec for higher the recovery rate. By used Bolot formulation to calculate a reward value of each combination data as equation.

$$reward = \frac{packet\ recovery\ rate\ before\ reconstruction}{packet\ recovery\ rate\ after\ reconstruction} \qquad (1)$$

RTCP packet is received in sender side, the reward value (L_b / L_a) of current combination R_c to calculate the loss information as follow.

$$R_c = c_0 + c_1 (L_b / L_a) \quad (2)$$

Where R_c is the reward value of current combination, c_0 and c_1 are coefficients to be determined the least squares error solution.

2.2 The Least Square Method

The least square concept presented here will continue to develop for estimation, prediction or numerical analysis the reward value in CNR algorithm. Least square method with Hilbert space [1] can predict the optimized of reward value for system in Figure 1. From Figure 1, w represented the input data and z represented an output, the relationship between w and z as the transformation $z = f(w)$. Discrete input data is represented by w_i, the output result is $z_i = f(w_i)$, $i = 1,\ldots,m$. Linear regression method is sloved this problem by a linear function $f_a(w)$, as follow.

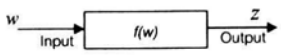

Figure1. System model of linear regression method.

$$f_a(w) = c_o + c_1 w \quad (3)$$

Let the individual deviations δ_i, defined as

$$\delta_i = f(w_i) - f_a(w_i) \qquad i = 1,\ldots,m \quad (4)$$

The solution problem by least square method is obtained to choose the coefficients c_0 and c_1 with minimize S.

$$S = \sum_{i=1}^{m} \delta_i^2 \quad (4)$$

$$= \sum_{i=1}^{m} [f(w_i) - f_a(w_i)]^2 \quad (5)$$

$$= \sum_{i=1}^{m} [z_i - (c_0 + c_1 w_i)]^2 \quad (6)$$

S is the sum of the individual deviation square, $f(w_i)$ is collected data and $f_a(w_i)$ is trend of reward data. The 2nd derivatives of S, as follow.

$$\frac{\partial S}{\partial c_0} = \sum_{i=1}^{m} 2[z_i - (c_0 + c_1 w_i)](-1) = 0 \quad (7)$$

$$\frac{\partial S}{\partial c_1} = \sum_{i=1}^{m} 2[z_i - (c_0 + c_1 w_i)](-w_i) = 0 \quad (8)$$

Rewrite (6) as follow.

$$\sum_{i=1}^{m} z_i = m c_0 + \left(\sum_{i=1}^{m} w_i \right) c_1 \quad (9)$$

$$\sum_{i=1}^{m} w_i z_i = \left(\sum_{i=1}^{m} w_i \right) c_0 + \left(\sum_{i=1}^{m} {w_i}^2 \right) c_1 \quad (10)$$

and

$$c_0 = \frac{\left(\sum_{i=1}^{m} z_i\right)\left(\sum_{i=1}^{m} {w_i}^2\right) - \left(\sum_{i=1}^{m} w_i\right)\left(\sum_{i=1}^{m} w_i z_i\right)}{\Delta} \quad (11)$$

$$c_1 = \frac{m \sum_{i=1}^{m} w_i z_i - \left(\sum_{i=1}^{m} w_i\right)\left(\sum_{i=1}^{m} z_i\right)}{\Delta} \quad (12)$$

$$\Delta = m \sum_{i=1}^{m} w_i^2 - \left(\sum_{i=1}^{m} w_i\right)^2 \quad (13)$$

2.3 The Proposed Method

Base on the CNR algorithm used the least square method for calculates the reward value in the next time interval. The reward value in the previous time interval used to decision the optimize combination number.

The reward value and combination number can predict by linear regression formula. Table 1 show the reward value in difference combination of Bolot.

Table 1. Reward value of difference combinations type

No.	Combination	Bolot
0	-	1
1	1-	2.5
2	-2	6
3	-1-2	6
4	-1-3	10
5	-1-2-3	18

3 Experiment Setup and result

This simulation setup with CNR algorithm uses NS-2 program for network simulator.

The network topology used for simulation model as shown in figure 2, all of links are drop tail. Most of them have 10 Mbps capacity and 1ms delay except the bottleneck link between N1 and N2 has 2 Mbps capacity and 50 ms delay.

The audio packets was sent from the sender (node RS) to the receiver (node RR) via the RTP in every 30 milliseconds. Node RR sends the receiver report packet in every 5 seconds to RS and another node sent the TCP (Transmission Control Protocol) and UDP (User Datagram Protocol) traffics.

From Figure 2, all average value calculated from source node N1 and destination node N2. The packet size was 24 bytes and audio traffic started for send a packet data at 0.1 second and stopped at 2400 seconds. Table 2 shows result data from NS-2 trace files in network. This data determined the performance of the recovery algorithm. The number period that L_a>high threshold of least square method has lower than Bolot. It can increase number of data recovery and received packet data.

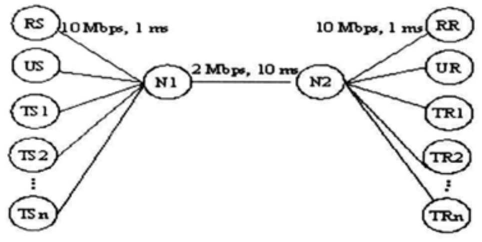

Figure 2. Topology model of the simulation

Then the performance is increased by the number of average throughput and average of delay is decreased.

Parameters	Least Square	Bolot
No. period that L_a >high threshold	13	17
Received Packets	1,056,000	1,052,190
Average Throughput (kbps)	1,950.67	1,950.46
Average Delay (ms)	62.56	70.14

Table 2 Simulation results from NS-2 trace file.

From (2) can plot graph of difference combination scheme and compare with the other the results [4][5][6] shows in Figure 3. The least square method is better than Bolot and exponential smoothing method.

Figure 4 and 5 shows loss rate and combination adjustment by Bolot and least square method, respectively. The adaptive algorithm of CNR can control the loss rate in the next time interval and reduce the number of loss rate after reconstruction as L_a more than the HIGH threshold.

4 Conclusions

In this paper, proposes a reward value forecasting by the least square method to perform the better forecasting and solve the drawback of the media specific FEC algorithm in part of increasing combination number. The experimental result shows the better performance of loss rate and important parameters in network. The recovery algorithm at sender side can send the appropriate redundant packets to audio receiver.

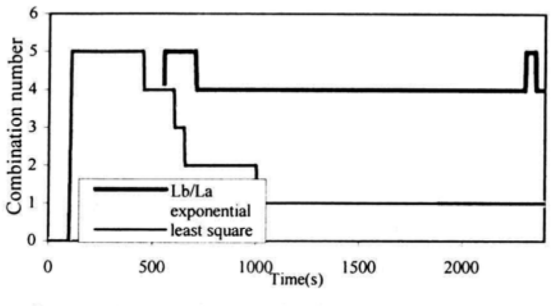

Figure 3. Comparative combination adjustment with different formula of R_c

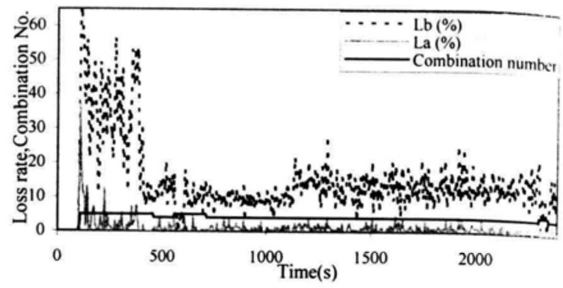

Figure 4. Loss rate of Bolot method

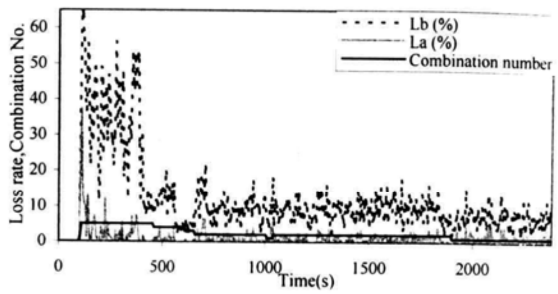

Figure 5. Loss rate of least square method

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ประวัติผู้เขียน

นางสาวอรุณี กาลวิบูลย์ เกิดเมื่อวันที่ 25 มกราคม 2519 ที่จังหวัดอุดรธานี ปีการศึกษา 2542 สำเร็จการศึกษาปริญญาตรี วิศวกรรมศาสตรบัณฑิต สาขาวิศวกรรมไฟฟ้า จากมหาวิทยาลัยขอนแก่น จังหวัดขอนแก่น