Impact of Packet Spacing Time on Packet Loss under LW for QoS of FEC-based Applications

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Abstract In this paper, we describe certain relationships between successive packet loss patterns and packet spacing. To observe a successive packet loss pattern, one possible method is to investigate test packets that are generated with some intervals such as Poisson or some second par packet. However, successive packet loss strongly depends on the generated interval. If test packets are generated with long interval, then, successive loss pattern cannot be shown. Thus, in such a method, where the packet intervals may sometimes be long or short, a successiveness of the packet loss should be considered in terms of the packet spacing. To clarify the relationship between the successive packet loss and the packet spacing, we analyze data based on observation of an actual network with the loss window size as a parameter. We find that when the packet spacing is narrower, i.e., a shorter interval, the probability becomes higher that the packet immediately following a single packet loss would also be lost.

1. Introduction

Internet applications using continuous streams, such as audio and video, are increasingly popular. Interactive and real-time audio applications over the Internet, such as Internet phone calls, require low loss and short playout delay; Packet loss may deteriorate the decoded audio/video quality, and long playout delay will hinder interactiveness. In order to alleviate the effect of packet losses, several methods are developed. Forward Error Correction (FEC) coding is one solution to this problem[1), 2), 3)].

FEC methods, add redundancy to communication data, and try to (partially) recover original data carried in lost packets. Added redundancy, off course, lengthens transmission times. Therefore, appropriate redundancy factor should be used for ensuring a real-time response, minimizing the delay, and maintaining playouted media quality.

The optimal redundancy factor depends on the pattern of successive packet losses, since FEC methods will fail to recover data if several successive packets are lost.

Conventional network QoS(Quality of Service) parameters such as loss and delay, however, are based on each packet only, and ignore loss patterns in a packet stream. Therefore, the probability of long successive losses, for example, cannot be monitored, even though they strongly affect FEC-based application QoS.

In order to examine such successive packet losses, we previously proposed a method of expressing network and audio qualy for FEC-based audio communication[4)]. In this work, we have shown that the loss probability of one packet is strongly dependent on the loss/non-loss of its preceding packet. We have proposed a method for expressing such kind of conditional loss probabilities.

In our previous work[4)], all packets were generated with spacing of 40 ms, which is close to the standard value for an algorithmic delay under G.723.1[6)].

In general, loss patterns strongly depend on packet intervals. When the packet spacing between two successive packets is narrow (a short interval), the loss of the preceding packet is more likely to imply the loss of following one. However, we know no existent papers discussing on the dependency between loss pattern and packet intervals.

In this paper, we focus this aspect of packet loss patterns. We compute the successive loss patterns changing packet spacing, and discuss the relationship between these results.

In Section 2, we introduce the loss window size (LW), which extends the definitions for packet loss and delay to groups of packets. In Section 3, we describe our experimental set-up. In Section 4, we examine the packet loss rate as a function of LW and compare the packet loss rate for different packet interval times. We dis-

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cuss these results in Section 5, In Section 6, we summarize our conclusions.

2. Expression of network quality based on LW

In this section, we propose new measures for expressing network and application QoS using an FEC method.

2.1 The need to express network quality based on LW(N)

We assume that newly arrived data (or highly compressed data) d_i are copied on N successive packets as shown in Fig. 1. We treat these packets as one group, so we extend the definition of the conventional expression of network quality, such as loss and delay d, based on one packet.

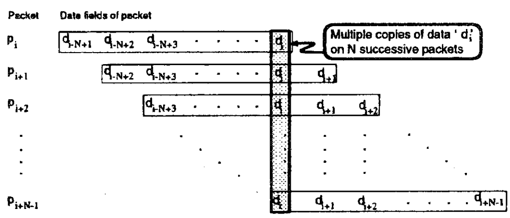

Fig. 1 Data fields of each packet.

To handle duplicated data on N successive packets, the loss window size is introduced. Intuitively speaking, packets are analyzed through a loss window of size N (packets). The window moves one packet at a time like a computed moving average.

We write LW for loss window size and LW(N) means LW=N. In the following, we extend the loss and delay using LW.

Network quality is conventionally based on one packet. Using LW, this situation corresponds to LW(1). Under our assumption above that successive packets carry the same data, this definition is not suitable.

For example, when k successive packets are lost, the packet loss is equal to k in the conventional definition. However, in the case of LW(N), for example, if $k < N$, there are no packet losses under LW(N). Losses happen only when $k \geq N$.

The data redundancy also affects the definition of delay. Since a group of packets carry the same data, we need to define the start time and the received time of the group. Furthermore, in interactive / real-time applications, packets that arrived too late are not suitable for use, and should be treated as losses. We call this loss and deadline-missed condition the playout error. Loss, delay, and playout error should thus be generalized using LW.

2.2 Loss based on LW(N) : LW(N)-L

Loss based on LW(N) (hereafter called LW(N)-L) means "all N successive packets have been lost". Therefore, LW(1)-L matches the conventional packet loss.

For example, compare LW(1)-L to LW(3)-L. For the loss pattern of Fig. 2(a), LW(1)-L occurs once; however, by changing to LW(3), no losses occur under the definition of LW(3)-L. For the loss pattern of Fig. 2(b), on the other hand, LW(1)-L occurs three times; however, LW(3)-L occurs only once.

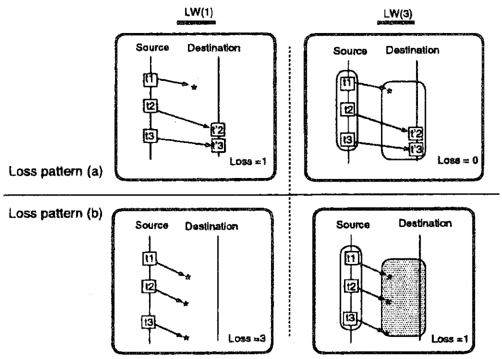

Fig. 2 Comparison of LW(1)-L and LW(3)-L.

2.3 Calculation of the loss rate based on LW(N)

To calculate all loss rates LW(N)($1 \leq N \leq n$) as a directly, we calculate the number of N successive packet losses for each LW(N). For all received packets (total: T), this method needs to scan all received packets n times. Thus, the cost is $O(T * n)$ to compute the LossRate [N] for every N ($1 \leq N \leq n$).

We propose a faster method in which we can reduce the calculation cost.

First, we define losses with run length : k: when a packet p_i reaches a destination, packets $p_{i+1}, p_{i+2}, ..., p_{i+k}$ are lost but packet p_{i+k+1} reaches the same destination.

c_k : = the count of losses with run-length k.

We assume K to be the longest loss run-length, and T to be the total number of received packets.

As shown in Fig. 3, in a pattern of losses with run-length = k ($1 \leq k \leq K$), LW(N)-L occurs. The number of LW(N)-L occurrences in this pattern follows (where we denote the number of occurrences for LW(N)-L as $|LW(N) - L|$)

is:

$$|LW(N) - L| = \begin{cases} k - N + 1 & (N \le k) \\ 0 & (N > k) \end{cases}$$

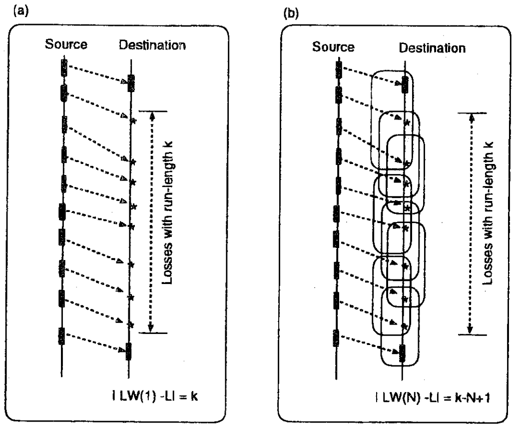

Fig. 3 Losses with run-length k under LW(1)-L and LW(N)-L.

Packet losses with run-length k occur c_k times, so the number of occurrences of LW(N)-L (we denote this as $L_N[k]$) for each k is given by:

$$L_N[k] = \begin{cases} (k - N + 1) * c_k & (N \le k) \\ 0 & (N > k) \end{cases}$$

The total number of LW(N)-L($1 \le N \le n$) referred to as, *L[N]*, is

$$L[N] = \sum_{k=1}^{K} L_N[k].$$

The total number of received packets under LW(N), referred to as T_N, becomes T-N+1. Thus, the LossRate[N] based on LW(N) is:

LossRate[N] $= L[N]/T_N$.

Based on the above, we can scan once for all T packets to calculate c_k; then, we can calculate the LossRate[N] for any N ($1 \le N \le n$). Consequently, using this method, the cost is $O(T+nK)$ for computing LossRate[N] for every N ($1 \le N \le n$). In general, the longest loss run-length K << T, and n << T, so $T+nK << Tn$. Thus, we can reduce the number of scans.

For example, consider the packet-loss pattern in Fig. 4. In this pattern, $c_1 = 2$, $c_2 = 1$, and $c_3 = 1$. So LossRate[1], LossRate [2], and LossRate [3] under LW(1), LW(2), and LW(3) are:
N=1; LossRate[1] $= (1 * c[1] + 2 * c[2] + 3 * c[3])/12 = 7/12$,

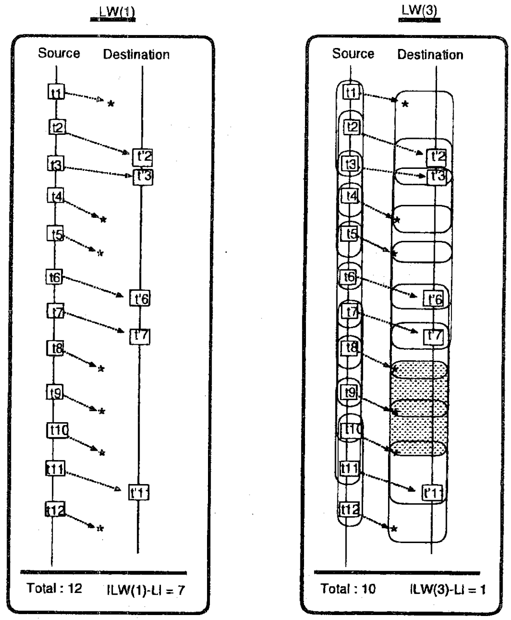

Fig. 4 Calculation for LossRate under LW(3)

N=2; LossRate[2] $= (1{*}c[2]+2{*}c[3])/11 = 3/11$, and
N=3; LossRate[3] $= (1 * c[3])/10 = 1/10$.

3. Experimental monitoring network

In order to show the effectiveness of the proposed measures, we have made the monitoring system on a working network as shown in Fig. 5. We have monitored packets flowing from a stream generator HOST 3 at a network 2 to HOST 1 at the network 1. Packets are probed at network 2 using HOST 2, and also at network 2 using HOST 1. Both hosts were equipped with the QoS Visualizer[10),11)]. Thus, each probed packet was given precise timestamp synchronized to UTC (Universal Coordinated Time) within 0.5 *ms*. HOST 2 also works as an NTP server, and HOST 3 is its client.

Generated packets are transported using UDP, with the fields shown in Fig. 6. The sequence of numbers begins from 0, and is incremented by one when a packet is generated. The packet length is 320 octets, and the packets are generated at 40-*ms* intervals.

By comparing these timestamps between HOST 1 and HOST 2, we can collect information on network quality such as one-way delay and jitter. Checking packets only observed at HOST 2, losses can also be detected. Thus we

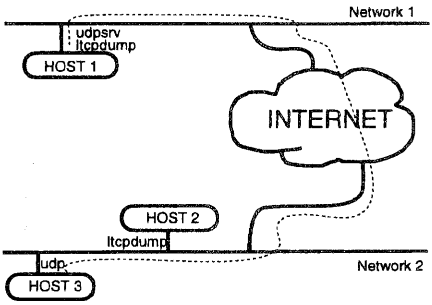

Fig. 5 Experimental monitoring network.

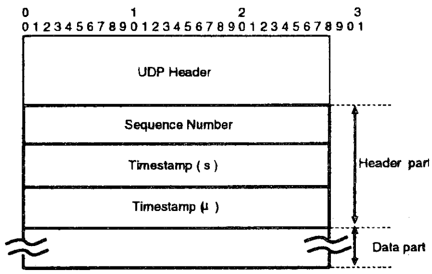

Fig. 6 Construction of UDP data segment

can also obtain packet loss patterns.

4. Experiments results

In this section, we examine some network quality, applying LW(N) based on Internet measurements. These data are the same one in our previous experiment data[4)] which introduced some results about packet loss rate, delay, and playout delay under LW(N). In this work, we are also interested in packet losses under changing packet spacing time that will be explained in this section.

4.1 LW(N)-L

We used two hosts: HOST 1 is located in Tokyo, HOST 2 is outside Tokyo. We measured from 12:30 to 13:30. The network's (average) loss rate was 1.27%, and the longest loss run-length was 27. The run-length distribution of the packet losses is shown in Fig. 7.

Using this distribution and applying the calculation method described in Sec. 2, we obtained each LW(N)-L rate for $1 \leq N \leq 27$ (Fig. 8).

We also measured the loss rate for different days and times (Fig. 9). The measurement periods were all one hour and the loss rates for each network are shown below, where K means the longest loss run-length. The network with

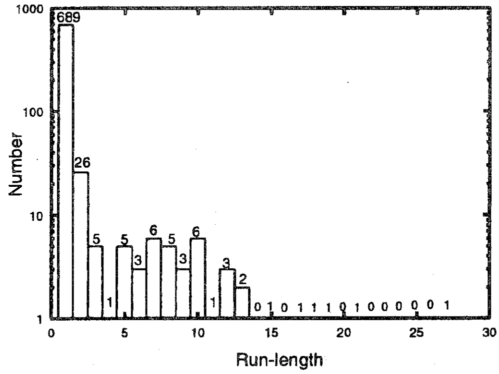

Fig. 7 Run-length distribution of successive packet losses for July 04.

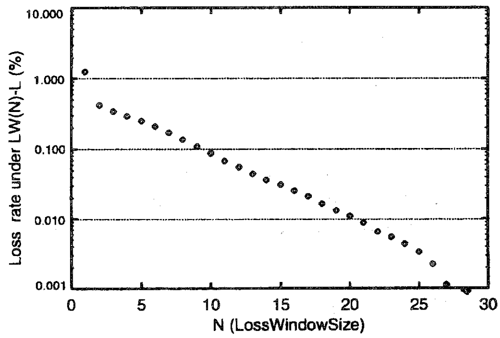

Fig. 8 LW(N)-L rate for July 04.

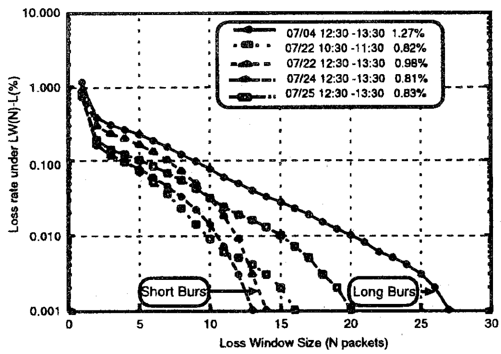

Fig. 9 LW(N)-L rate for all five days.

the largest K (Table. 1).

Date	Loss Rate	K
07/04 12H30 - 13H30	1.27	27
07/22 10H30 - 11H30	0.82	16
07/22 12H30 - 13H30	0.98	14
07/24 12H30 - 13H30	0.81	13
07/25 12H30 - 13H30	0.83	20

Table 1 Measurement results for each day.

4.2 Impackt of Packet spacing time on LW(N)-L

In this section, we observed some relations between a successive packet loss pattern and a packet spacing time.

Suppose a packet loss pattern to get such as an upper one Fig. 10. For simple example, we can make a packet loss pattern reducing each two packets. These two patterns, an upper packet loss pattern and a lower one, are different. So as easily understand with this example, we can estimate some impact of packet spacing time between successive two packets on packet loss under LW.

First, we observe a loss rate under LW(N)-L reducing each N packets from an original packet loss patter. Fig. 11 shows a relationship between LW and LW(N)-L changing packet spacing time from 40 *ms* to 320 *ms*. This data is based on one of results in Fig. 9 which has a longest successive packet loss, which means that this data has a long burst time. This ordinary loss rate, i.e., LW(1)-L, was 1.27 %.

Secondly, we examined a same observation applying an another data that has a shorter burst time. This ordinary loss rate, i.e., LW(1)-L, was 1.27 %. This LW(1)-L was 0.98 %. We define each points P_i in Fig. 11 and P'_i in Fig. 12 correspond to loss window size $i = 1, 2, 3$. Comment tendencies between these two difference burst cases is following: from Fig. 11, refshort.burst, when a packet spacing time become longer, absolute values of gradients for line (P_1, P_2), (P_2, P_3), (P'_1, P'_2), and (P'_2, P'_3) increase, i.e., a conditional probability that happen a following packet loss after happening the first or second packet loss become low. Specially, the values of gradient for line (P_1, P_2) and line (P'_1, P'_2) become close a value of an independent event model, i.e., each packet loss event will happen with no relationship each other. On the other hand, when a packet spacing time become shorter, all absolute values of gradients for all four lines decrease, i.e., a conditional probability of successive loss after happening a first packet loss or a following second packet loss become higher.

To confirm an above relation between a conditional probability of successive loss after happening some packet loss, we observed following two probability:

- Event P: a conditional probability of the second packet loss after happening the first packet lost.
- Event Q: a conditional probability of the third packet loss after happening the second packet lost.

The conditional probability P in Fig. 13 corresponds with an absolute gradient value for Line(P_1, P_2) in Fig. 11, and similarly in Fig. 14, P corresponds with an absolute gradient value for Line(P'_1, P'_2) in Fig. 12. As similarly, the conditional probability Q in Fig. 13 is an absolute gradient value for Line(P_2, P_3) and in Fig. 11, Q is an absolute gradient value for Line (P'_2, P'_3) in Fig. 12. Applying these correspondence, we observed a probability of P and Q for changing packet spacing time to 2000 *ms* with 40 *ms* interval. This result are Fig. 13 for a long burst case and Fig. 14 for a short burst one. We set independent model values of probability for P and Q in both figures. The independent model means that has no relationship between any packet loss probability, so the conditional probabilities of P and Q equal an ordinal packet loss rate, i.e., a loss rate under LW(1)-L. The independent model value for the long burst model and the short burst mode are following (we calculate any gradients in logarithm):

Long burst model:

$$\log_{10}(LW(N))(\forall N) = \log_{10}(LW(1)) = \log_{10}(0.0127) \approx -1.8962$$

Short burst model:

$$\log_{10}(LW(N))(\forall N) = \log_{10}(LW(1)) = \log_{10}(0.0098) \approx -2.00877$$

5. Discussion

Successive packet losses and LW(N)-L

The run-length distribution of successive packet losses (Fig. 7) shows that single or two successive packet losses happen frequently, while three or more successive losses are rare.

As shown in Fig. 8, LW(N)-L decreases when N increases. Even considering this, LW(1)-L seems to be a singular point, since its probability is especially high compared with other

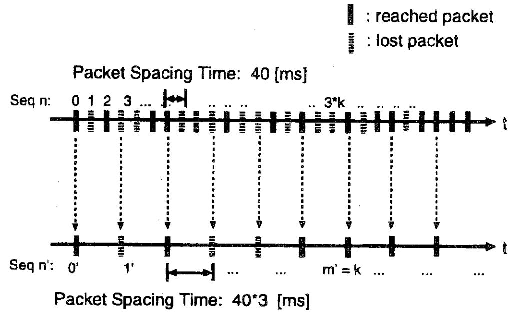

Fig. 10 Example for relationship between packet loss pattern and packet spacing time

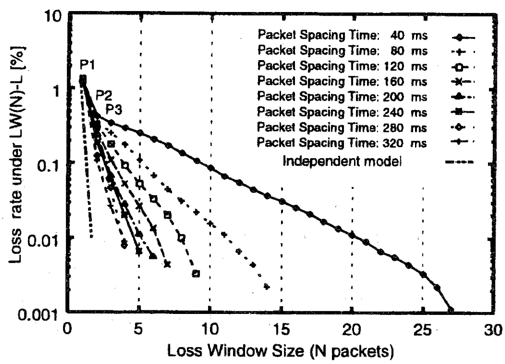

Fig. 11 Long burst model

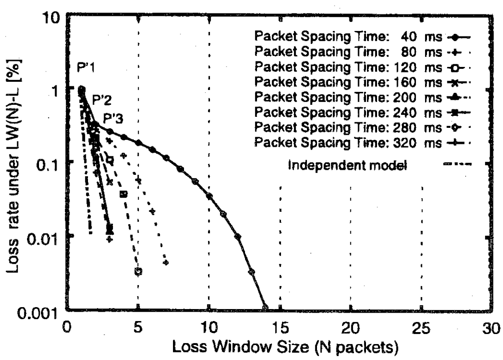

Fig. 12 Short burst model

LW(N)-Ls. Furthermore, as shown in Fig. 9, we have observed a partial linear logarithmic relation among LW(N)-L($2 \leq N \leq 8$). Note that coefficients of this linear relation have changed considerably when observation periods were different.

The conventional packet loss rate is an average rate for LW(1). Thus, it cannot express the difference of quality between long burst and short burst case. Applying only the ordinary

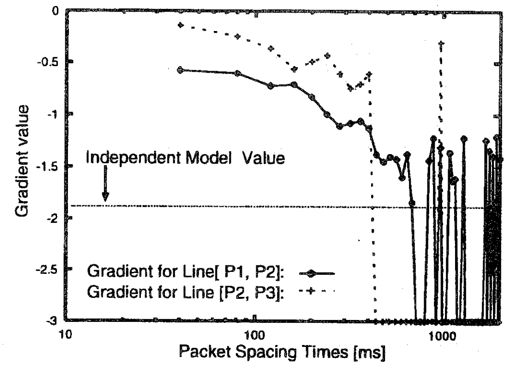

Fig. 13 Conditional probablity for successive loss in Long burst model

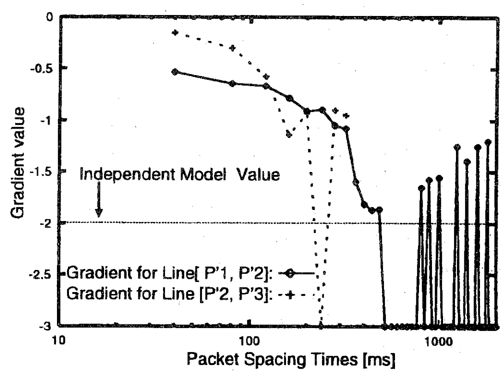

Fig. 14 Conditional probablity for successive loss in Short burst model

loss rate, QoS difference such as burst are not so apparent, however our new parameter of loss window size can show some information concerning successive packet loss pattern as we described.

Impact of packet spacing time

From Fig. 11, 12, we can confirm that the packet spacing time impact on successive packet loss. It depends on a network burst, however there is a comment tendency between a long burst network and a short burst network. The comment tendency is that: if a packet spacing time become longer, a successive loss dependency to occur a following packet loss become small and to be close to an independent relation. It make us mention that to observe a successive packet loss pattern, observer should consider about packet spacing time.

About concerning the conditional probability for the event P and Q in case for Fig. 13, 14, we can confirm that: if a packet spacing time is within a given threshold, the condition probability of P become close to the independent model. About Q, it seems to have a same tendency, however the sampling number was not enough to observe an event Q.

6. Conclusion

In this paper, we described some relations between successive packet loss patterns and packet spacing. Our main finding were that:

- Given longer packet spacing of two packets, single or successive packets loss has tend to close an independent model, which model means that each probability of packet loss does not relate with preceding or following one.
- Given shorter packet spacing of two packets, a probability that a following packet of single lost packet is lost successively becomes higher.

In our future work, we are interesting in measuring and modeling a successive loss pattern. With our previous work for a proposal of measurement for loss window size and this work about some impacts of packet spacing time on packet loss under LW, we would make clear a relationship of successive loss pattern applying Markov chain model.

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