RELIABLE MULTICAST PROTOCOL
A modified retransmission-based loss-recovery scheme based on the selective repeated ARQ protocols

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Abstract: This paper proposes a modified retransmission-based loss-recovery mechanism based on the selective repeated ARQ protocols for reliable multicast delivery. With the operation of XOR, it minimizes the number of packets for retransmissions, reduces the network burden, and increases the throughput. Furthermore, the analysis infers a basis for choosing the suitable loss-packed size of retransmission to achieve higher throughput with lower network cost. The delays of processing time are also evaluated as well. Some interested effects of key system parameters on the delay performance are observed.

1 INTRODUCTION

IP multicast is an Internet Engineering Task Force (IETF) standard that allows a single packet to be sent to a potentially large number of receivers. Despite its unreliability, it is both selective (each packet is delivered only to receivers who have subscribed to the multicast address) and efficient (a packet is never transmitted on a link more than once). However, most applications such as video-conferencing, distributed interactive simulation, and news distribution still require some form of reliability to recover themselves from link-level packet losses even though the packet losses are an inherent by-product of Internet congestion control (Whetten, 1998).

In data communication, one of the most widely used techniques for handling transmission errors at the data link layer is error detection with Automatic-Repeat-reQuest (ARQ). This scheme is simple and highly reliable. The protocol is usually classified into three basic categories: Selective repeat, Go-back-N, and Stop-and-wait. Many researches have analyzed these basic protocols ((Benice, 1964), (Bruneel, 1986), (Burton, 1972)) and thoroughly studied the variants to the three basic protocols ((Towsley, 1987), (Weldon, 1982)). One principle feature those variants share is to send multiple copies of each loss packet instead of just one copy of each loss packet, which makes heavier the network burden. This paper showcases an improved recovery retransmission mechanism to decrease significantly this kind of network burden. It is based on (a) automatic-repeat-request protocols to enhance the capacity of a communication channel and (b) the idea of XORing several combined retransmitted packets to minimize the number of retransmissions and to hence increase the throughput.

Section 2 describes the Loss-Recovery Retransmission Scheme equipped in the protocol. Several analyses for the Pure SR and Loss-Recovery Retransmission protocols are discussed in Section 3. Finally, concluding remarks are given in Section IV.

2 LOSS-RECOVERY RETRANSMISSION SCHEME

2.1 System Scenario Description

The scenario considered in this paper consists of one sender and *m* receivers, where the communication between the sender and receivers takes place over multicast channel. The data packets transmitted by the sender can be received by all the *m* receivers. Once a packet is lost, a receiver will send back a negative acknowledgement (NACK) to the sender. Data messages are sent as fixed length data block.

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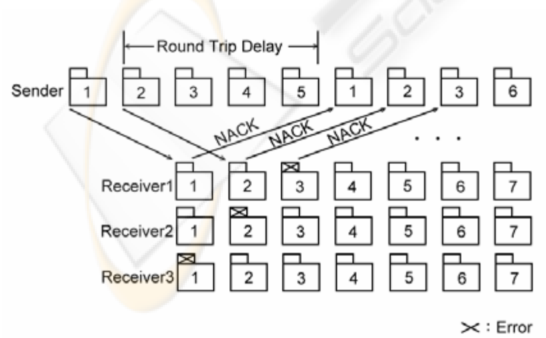
Assume that the lost packets at all receivers for all transmissions are independent and that the probability of packet loss, p, is the same among all channels. Furthermore, NACKs and retransmitted packets are never lost.

The assumption that losses among receivers occur independently still holds when the multicast backbone loss is small (observed experimentally in the MBone in (Yajnik, 1996)) and when there is low loss from the sender into the backbone. It is possible that losses could correlate, according to the concept of association of random variables (Esary, 1967). The pessimistic bounds on throughput resulted from our analyses are hence regarded as a consequence of this independence assumption. Another assumption that NACKs is never lost is reasonable as control packets are small. If necessary, the assumption can be relaxed by following the analysis given in (Towsley, 1985).

In this context, we now describe the two approaches, pure selective repeat (Pure SR) and loss-recovery retransmission strategies, for reliable transmission of the data packet from sender to multiple receivers.

Pure selective repeat strategy is a receiver-initiated protocol that assigns the responsibility for ensuring reliable packet delivery to the receivers, whose role is to check for the lost packets. That is, sender keeps transmitting new data packets until it receives a NACK from a receiver. When this occurs, the sender then retransmits (i.e. again multicasts) the lost packet to all its receivers. In order to guard against the loss of the NACK and the subsequent packet retransmission, the receiver use timers in a manner analogous to the sender's use of timers in sender-initiated protocols (Towsley, 1997). Figure 1 illustrates an example of pure selective repeat strategy.



Fiqure 1: Pure selective repeat strategy.

With loss-recovery retransmission, the sender does not retransmit requested packet upon receiving a NACK. Rather, it gathers NACKs to a certain number L ($L \geq 1$) after a period of the round trip propagation delay, and then decides and groups among these unacknowledged packets into so-called a *parity-packet*. Of course, each receiver must be able to reconstruct all its lost packets from parity-packet; namely it must recognize all the packets included in the parity-packet. With the XOR operation conducting on all the packets, correctly received and parity-packet, the receiver obtains the expected lost packet.

Figure 2 illustrates an example of loss-recovery retransmission strategy. Each NACK represents a negative acknowledgement reported by at least one of the receivers. The size of L is 3 packets. The sender sums the NACKed packets 1, 2, and 3 by elementary modulo 2 addition (XOR) into a parity-packet and retransmits these 3 NACKed packets in this parity-packet instead of retransmitting them separately. After error-free reception of the parity-packet, each receiver checks the header in order to identify the sequence numbers of data packets in the parity-packet. With the help of the sequence numbers it can reconstruct the expected packet from the parity-packet as described below.

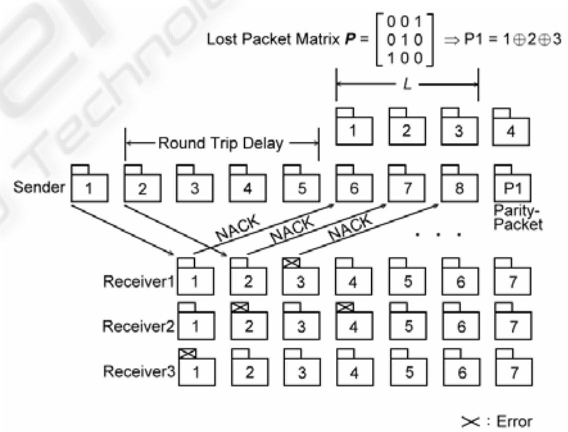


Figure 2: Loss-recovery retransmission strategy.

The difficulty of this Loss-Recovery Retransmission strategy lies in combining as many NACKed packets in a parity-packet as possible and also ensuring that each related receiver can recognize its lost packets from the parity-packet in no time.

2.2 Loss-Recovery Retransmission Mechanism

To analyze this strategy, we have to describe mathematically which receiver needs the retransmission of what packets in advance. We use ***P*** to denote a lost packet matrix, which is generated after the arrival of a certain number of NACKs, *L*, at sender's site, with the receiver numbers as rows and packet numbers as columns. If receiver *i* has lost packet *j*, the matrix element at position (*i*, *j*) is set to one; otherwise, the element is set to zero. The lost packet matrix ***P*** is shown below:

	Packet1	Packet2	Packet3
Receiver1	0	0	1
Receiver2	0	1	0
Receiver3	1	0	0

Algebraically, the above matrix consists of column vectors: $\begin{bmatrix}0\\0\\1\end{bmatrix}$, $\begin{bmatrix}0\\1\\0\end{bmatrix}$, and $\begin{bmatrix}1\\0\\0\end{bmatrix}$. Assume that there is a *m* by *n* matrix ***P***. Then we denote *n* column vectors as P_1, P_2, …, P_n. If they are all orthogonal, namely, no receiver has lost more than one packet, the sender can exclusive-or all these *n* packets to obtain a single recovery packet, *parity-packet*. To prove the two vector's orthogonality, we use inner product.

$\boldsymbol{A} = (a_1, a_2, \ldots, a_n)$ and $\boldsymbol{B} = (b_1, b_2, \ldots, b_n)$ are orthogonal if $a_1b_1 + a_2b_2 + \ldots + a_nb_n = 0$.

We can see that if each retransmission request comes from different receiver, the Loss-Recovery Retransmission mechanism will work very well. Yet once such situation disappears and no modification is given, the reliability will not be guaranteed. For example, in case Receiver2 also loses Packet3, each column vector of the matrix ***P*** goes like $\begin{bmatrix}0\\0\\1\end{bmatrix}$, $\begin{bmatrix}0\\1\\0\end{bmatrix}$, and $\begin{bmatrix}1\\1\\0\end{bmatrix}$. If the parity-packet is made up of $\text{Packet1} \oplus \text{Packet2} \oplus \text{Packet3}$, Receiver2 may not be able to reconstruct Packet2 and Packet3, as efficiently as both Receiver1 and Receiver3.

Notice that, columns P_1 and P_2 are orthogonal, so are columns P_1 and P_3. If sender sets one parity-packet as $\text{Packet1} \oplus \text{Packet2}$ or $\text{Packet1} \oplus \text{Packet3}$, and treats the remaining packet (Packet3 or Packet2) as another parity-packet, Loss-Recovery Retransmission strategy still works well because every two column vectors in the packet are orthogonal.

To make good use of Loss-Recovery Retransmission mechanism, we should find as many orthogonal column vectors as possible from matrix ***P***, which forms a submatrix of ***P***, denoted as ***P′***. This set of packets that are to be XORed for an arbitrary error pattern[1] can be pre-computed and stored in tables to avoid high computational cost during transmission. The algorithm of finding out ***P′*** is shown in next subsection.

2.3 Algorithm of Building a Matrix *P′*

In (Aghadavoodi Jolfaei, 1993), Aghadavoodi Jolfaei et al interpret lost packet matrix ***P*** as the incidence matrix of a hypergraph (Berge, 1989). By that the problem of generating parity-packets from a given error matrix was linked with the edge coloring problem of graph theory (Aghadavoodi Jolfaei, 1993). In this subsection, we present a simple straightforward algorithm of finding out submatrix ***P′***.

Provided that sender has already achieved the lost packet matrix ***P***, which is *m* by *n*. Recall that *m* stands for the number of receiver which submits retransmission request, and correspondingly *n* stands for the number of packet which is lost at the receiver site. In this system scenario, any receiver might lose more than two packets, and any packet might be lost by two or more receivers. We can rewrite matrix ***P*** as $[P_1, P_2, \ldots, P_n]$ (P_i is a $m \times 1$ column vector). The algorithm is shown as follows:

Algorithm 1.

Step 1: Get an arbitrary element from ***P*** and add it to submatrix ***P′***.

Step 2: Check a vector *P*, not in ***P′*,** which is orthogonal with every element of ***P′***, add *P* to the ***P′***, and at same time delete *P* in the matrix ***P***.

Step 3: Repeat Step 2 until no remaining vector *P* is orthogonal with every element of ***P′*** or ***P*** is empty.

Step 4: If ***P*** is not empty, then back to Step 1 to find another submatrix ***P′*** and repeat Step

[1] The task to determine a minimum subset is presumably NP-complete [7]. Nevertheless an exact calculation is possible if the number of packets per parity-packet is limited.

2 ~ 3.

Step 5: Repeat Step 1 ~ 3 until ***P*** is emptied.

Step 6: Output all the submatrice ***P′***.

3 ANALYTICAL RESULTS

3.1 Expected Number of NACKs Received by the Sender

For an analysis of the transmission cost, the following three parameters are needed:

m = number of receivers,
h = number of hops between sender and receivers,
p = packet loss probability per hop.

Assume that h is constant and p is equal among all links. Let p_b denote the probability that a packet is lost on a link branch in the multicast tree from the sender to a receiver. Then p_b can be expressed as

$$p_b = 1 - (1 - p)^h. \quad (1)$$

Let the random variable X denote the number of retransmissions by the sender necessary for all m receivers to successfully receive a packet and X_r is the number of transmissions required for receiver r to receive the lost packet correctly, i.e. $X = \max\{X_r ; 1 \le r \le m\}$. The number of NACKs reported to the sender by receiver r is $X_r - 1$. Hence, the mean number of NACKs received by the sender from the receiver r is

$$E(X_r - 1) = \sum_{k=2}^{\infty}(k-1)p_b^{k-1}(1-p_b) = \frac{p_b}{1-p_b} \quad (2)$$

Consequently, the expected number of NACKs received by the sender from all the receivers is $mp_b/(1 - p_b)$.

Assuming all retransmissions are to be performed by the sender and loss events independent for each receiver the following probabilities for X can be derived (cf. D. Towsley et al (Towsley, 1997)):

$$P(X \le n) = \prod_{r=1}^{m} P(X_r \le n) = \sum_{i=0}^{m}\binom{m}{i}(-1)^i p_b^{in}, \quad (3)$$

where n is positive integer and thus

$$P(X = n) = P(X \le n) - P(X \le n-1) = \sum_{i=0}^{m}\binom{m}{i}(-1)^i p_b^{in}(1 - p_b^{-i}). \quad (4)$$

The expected number of retransmissions is

$$E(X) = \sum_{k=1}^{\infty}[kP(X = k)] = \sum_{i=1}^{m}\binom{m}{i}(-1)^{i+1}\frac{1}{(1-p_b^{i})}. \quad (5)$$

If the sender has to send D packets, the sender should expect to receive $D \cdot E(X)$ NACKed packets statistically.

3.2 Expected Number of Parity-Packets Sent by the Sender

For further analysis, let the random variable Y denote the number of parity-packet generated for a L loss-recovery retransmissions. If $\boldsymbol{P} = [P_1, P_2, \ldots, P_n]$, Y is exactly the minimum number of the submatrices, within which each two vectors are orthogonal. Therefore, the retransmitted parity-packet RP is

$$RP = \frac{D \cdot E(X)}{L} \cdot E(Y). \quad (6)$$

Apparently, Equation (6) fails to offer a general way to determine $E(Y)$ as this value depends greatly on the packet loss rate as well as the number or receivers, total packets, and the size of loss-recovery retransmissions. Viewing from different perspective, it might be necessary to find another way out. We assume now for this analysis that $E(Y) = \frac{L}{\mathrm{M}}$ where M $\ge$ 1 (M is the mean number of packets in a parity-packet). This assumption leads to

$$RP = \frac{D \cdot E(X)}{\mathrm{M}}. \quad (7)$$

Ideally, every NACK within on collected window asks for distinct packet, thus

$$\mathrm{M} = L \text{ and } RP = \frac{D \cdot E(X)}{L}. \quad (8)$$

However, it can not be guaranteed throughout the whole multicast communication. The worst case is when every two vectors are not orthogonal, M = 1, and

$$RP = D \cdot E(X). \quad (9)$$

It is hence degenerated into a Pure SR (Selective Repeat) strategy, which is the worst case of all.

Figure 3 gives the differences between various numbers of receivers with each curve indicating how number of retransmitted packets change along with mean number of packets in a parity-packet.

The retransmitted packet number has inverse ratio relationship with M, mean number of packets in a parity-packet. The trends indicate while M is 5 ~ 10 there appears a good expected value of the retransmitted packets; whereas while M > 10, the throughput receives marginal effect. We infer from this finding a basis for choosing a suitable size of *L*, size of collected window, for network communication with various situations.

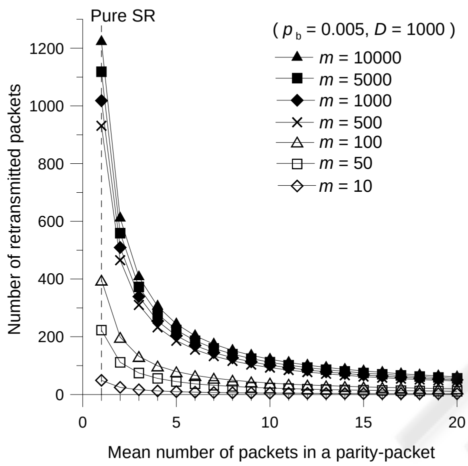


Figure 3: Comparison of the retransmitted packet as a function of M between Pure SR and Loss-Collected Retransmission strategy for various receivers *m*

3.3 Delay Evaluation

This paper presents an improved method to recover lost packets in multicast applications. The source is the only network node able to retransmit lost packets (global recovery mechanism) and, to decrease the amount of retransmissions (and hence, network congestion), several lost packets are assembled in a single packet (parity-packet) before retransmission. However, the delay by waiting for several lost packets to be assembled into a single retransmission packet might be significant.

Our goal in analyzing both the sender and receivers will be to compute the necessary amount of *processing time* (at both the sender and a receiver) for a packet to be successfully delivered from the sender to all of the receivers. The processing time includes the amount of time needed to send/receive the original packet as well as any retransmissions of that packet, and the amount of time needed to send/receive NACK packet. These delay measures will be of our primary interest in this subsection.

We begin by considering the transmission of a packet from one participant, henceforth referred to as the sender, to *m* identical participants, henceforth referred to as receivers. As the behavior of the sender differs from that of a receiver, we consider their behaviors separately.

We now analyze the Pure SR protocol first by considering the sender. Let $T^{\mathrm{SR}}_{\mathrm{sender}}$ denote the packet processing time required at the sender under Pure SR protocol. This processing time can be expressed as

$$T^{\mathrm{SR}}_{\mathrm{sender}} = \sum_{k=1}^{X} T_t(k) + \sum_{i=1}^{R} T_{NACK_s}(i) , \qquad (10)$$

where the first term corresponds to the processing time associated with the *X* different transmissions of the packet and the second term corresponds to the processing time for the NACKs that are transmitted from the receivers to the sender. $\{ T_t(k) \}$ and $\{ T_{NACK_s}(i) \}$ are sequences of identically distributed random variables. As before, *X* is the number of transmissions required and *R* is the number of NACKs received.

The mean processing time is given as

$$E(T^{\mathrm{SR}}_{\mathrm{sender}}) = E(X)E(T_t) + E(R)E(T_{NACK_s}) . \qquad (11)$$

The number of NACKs reported to the sender by receiver *r* is $X_r - 1$ with mean $p_b/(1 - p_b)$. Hence the mean number of NACKs reported by all receivers is $E(R) = mp_b/(1 - p_b)$, the mean per packet processing time at the sender is

$$E(T^{\mathrm{SR}}_{\mathrm{sender}}) = E(X)E(T_t) + \frac{mp_b E(T_{NACK_s})}{1 - p_b} . \qquad (12)$$

We focus next on the mean per packet processing time at a receiver. In a similar fashion, the mean processing time required at the receiver for a randomly chosen packet is

$$\begin{aligned} E(T^{\mathrm{SR}}_{\mathrm{receiver}}) = {} & E(X)(1 - p_b)E(T_p) + \\ & E\big((X_r - 1)^+\big)E(T_{NACK_r}) + \\ & E\big((X_r - 2)^+\big)E(T_{tout}) , \end{aligned} \qquad (13)$$

where $(X)^+ = \max\{0, x\}$. T_p , T_{NACK_r} , and T_{tout} are times to receive a packet, NACK transmission times, and times to process timeout at receiver respectively.

Here the first term corresponds to the processing required to receive a packet. The second term corresponds to the processing required to prepare and return NACKs. Note that this only occurs each time the receiver determines the packet to be lost prior to the first correct receipt of this packet. The last term corresponds to the processing of the timer when it expires. Again, this is only required after the first transmission (if lost) up to, but not including, the first correct reception of a given packet.

From the distribution of X_r, it follows that

$$E\left((X_r-1)^+\right)=\frac{p_b}{1-p_b},$$

$$E\left((X_r-2)^+\right)=\sum_{k=3}^{\infty}(k-2){p_b}^{k-1}(1-p_b)$$

$$=\frac{{p_b}^2}{1-p_b}. \quad (14)$$

Substituting into Equation (13) gives

$$E(T^{SR}_{receiver})=E(X)(1-p_b)E(T_p)+$$

$$\frac{p_b E(T_{NACK_r})}{1-p_b}+\frac{{p_b}^2 E(T_{tout})}{1-p_b}. \quad (15)$$

We end up this subsection with the analysis of the Loss-Recovery Retransmission strategy. Loss-Recovery Retransmission Strategy differs from Pure SR in that the sender does not retransmit requested packets immediately upon receiving a NACK. Instead gathers L ($L \geq 1$) numbers of NACKs after the period of the round trip propagation delay, and then groups among these unacknowledged packets into several so-called parity-packets. This delay is $\sum_{k=0}^{L}T_t(k)+\sum_{i=0}^{L}T_c(i)$, where $\{T_c(i)\}$ is the sequence of identically distributed random variable which corresponds to the time to process parity-packet. Let T^{LR}_{sender} denote the packet processing time required at the sender under Loss-Collected Retransmission protocol. The mean processing time can be expressed as

$$E(T^{LR}_{sender})=E(T_t+T_c)+E(R)E(T_{NACK_s})$$

$$=\sum_{i=0}^{L}{p_b}^i(1-p_b)^{L-i}E(T_t)+E(T_c)+$$

$$E(R)E(T_{NACK_s}). \quad (16)$$

The mean number of NACKs returned by all receivers is $E(R)=mp_b/(1-p_b)$ and the mean time of parity-packet process, $E(T_c)$, by the sender is

$$E(T_c)=\sum_{k=1}^{L}\frac{1}{L}kT_{pp}$$

$$=\frac{L+1}{2}T_{pp}, \quad (17)$$

where T_{pp} is the time to generate a parity-packet. Substituting into Equation (16) gives

$$E(T^{LR}_{sender})=\sum_{i=0}^{L}{p_b}^i(1-p_b)^{L-i}E(T_t)+$$

$$\frac{L+1}{2}T_{pp}+\frac{mp_b}{1-p_b}E(T_{NACK_s}). \quad (18)$$

Similar to the analysis of Equation (18), we have the mean processing time at the receiver under Loss-Recovery Retransmission protocol, $E(T^{LR}_{receiver})$, which can be expressed as

$$E(T^{LR}_{receiver})=E(X)(1-p_b)E(T_p)+$$

$$E\left((X_r-1)^+\right)E(T_{NACK_r})+$$

$$E(T_p+T_d), \quad (19)$$

T_d, similar as T_c, is the time for a receiver to recover the lost packet.

Hence, the mean processing time required at the receiver for a randomly chosen packet is

$$E(T^{LR}_{receiver})=E(X)(1-p_b)E(T_p)+$$

$$\frac{p_b}{1-p_b}E(T_{NACK_r})+$$

$$\sum_{i=0}^{L}{p_b}^i(1-p_b)^{L-i}E(T_p)+$$

$$\frac{L+1}{2}T_{pp}. \quad (20)$$

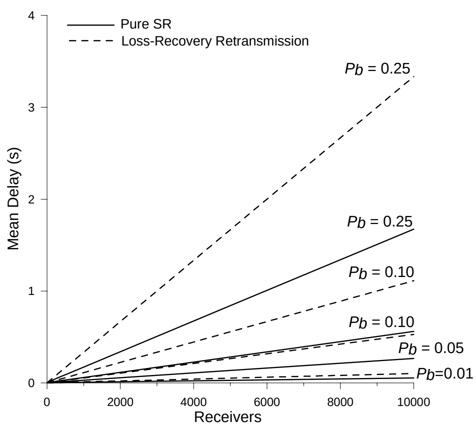


Figure 4: Mean processing time delayed at the sender for Pure SR and Loss-Recovery Transmission Protocols.

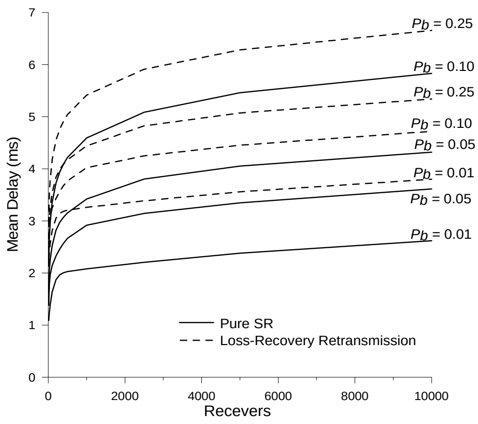


Figure 5: Mean processing time delayed at the receiver for Pure SR and Loss-Recovery Transmission Protocols.

Figure 4 and Figure 5 show the mean values of processing time delayed at the sender and receiver for Pure SR Protocol and Loss-Recovery Retransmission Protocol. In our numerical examples we have $E(T_t) = E(T_p) = 1000\,\mu\text{s}$ for the processing time needed to send or receive a 2K data packet and $E(T_{NACK_s}) = E(T_{NACK_r}) = 500\,\mu\text{s}$ as the processing time to send or receive a small NACK packet (Kay, 1993). We use $E(T_{tout}) = 24\,\mu\text{s}$ (Kay(1), 1993) to indicate the timer overhead and $T_{pp} = 6.5\,n\text{s} \times 2\text{KB} \cong 100\mu\text{s}$ the time to generate a parity-packet. We examine such delay for loss probabilities in the range 0.01 – 0.25 as they typify the loss characteristics of the MBone (Yajnik, 1996).

4 CONCLUSIONS

By *fully reliable* we mean that the protocol should provide recovery from losses even at the expense of reduction of throughput. Rather, at the end of transmission, the sender has to guarantee that every receiver in its membership set has received all the data packets it transmitted. In this paper we have discussed an improved retransmission-based approach to packet loss recovery schemes for multicast communication protocol, the Loss-Recovery Retransmission strategy. In this strategy the sender does not retransmit requested packets immediately upon receiving of a NACK; instead, it gathers a number of selected NACKed packets by XORing them to minimize the number for retransmissions and thus actually reduces the network burden and increases the throughput. The analytical results for a Pure SR strategy show the decrease in retransmission of the Loss-Recovery Retransmission strategy with the growing mean number of packets in a parity-packet, packet loss rate, and the quantity of the participating receivers. Furthermore, one significance in our analysis of this strategy is that we can estimate a suitable collected window size of the loss-recovery retransmission for the present various network characteristics.

We also analyze both the sender and receiver and evaluate the expected amount of *processing time* required by Pure SR and Loss-Recovery Retransmission protocols for a packet to be successfully delivered from the sender to all of the receivers.

REFERENCES

Aghadavoodi Jolfaei, M., Martin, S.C., and Mattfeldt, J., 1993. A new efficient selective repeat protocol for point-to-multipoint communication. *In IEEE Proc. ICC'93*. IEEE Press.

Berge, C., 1989. *Hypergraphs*. Amsterdam: North-Holland.

Benice, R.J. and Frey, Jr. A.H., 1964. An analysis of retransmission system. *In IEEE Trans. Comm. Tech.*. IEEE Press.

Bruneel, H. and Moeneclaey, M., 1986. On the throughput performance of some continuous ARQ strategies with repeated transmission. *In IEEE Trans., Comm.*. IEEE Press.

Burton, H.O. and Sullivan, D.D., 1972. Errors and error control. *In Proceeding IEEE*. IEEE Press.

Esary, J.D., Proschan, F., and Walkup, D.W., 1967. Association of random variables with applications. *In Ann. Math. Stat*.

Hoyer, I., 1981. The NP-completeness of the edge-coloring. *In SIAM J. Comput*.

Kay, J. and Pasquale, J., 1993. Measurement, analysis, and improvement of UDP/IP throughtput for the DECstation 5000. *In Proceeding 1993 Winter Usenix Conference*.

Kay(1), J. and Pasquale, J., 1993. The importance of nondata touching processing overhead in TCP/IP. *In Proceeding ACM SIGCOMM '93.* ACM Press.

Towsley, D., Kurrose, J., and Pingali, S., 1997. A comparison of sender-initiated and receiver-initiated reliable multicast protocols. *In IEEE Journal on Selected Area in Comm.*

Towsley, D. and Mithal., S., 1987. A selective repeat ARQ protocol for a point to multipoint channel. *In Proceeding INFOCOM*.

Towsley, D., 1985. An analysis of a point-to-multipoint channel using a go-back-N error control protocol. *In IEEE Trans. Comm.*.

Weldon, E.J., 1982. An improved selective-repeat ARQ strategy. *In IEEE Trans. Comm.*.

Whetten, B. and Conlan, J., 1998. A rate based congestion control scheme for reliable multicast. *GlobalCast Commun. Tech.* White Paper.

Yajnik, M., Kurose, J., and Towsley, D., 1996. Packet loss correlation in the mbone multicast network. *In IEEE Global Internet Conf.* IEEE Press.