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Study on the Throughput Limits over the HDLC Protocol

MITSUHIRO AZUMA*, YOSHIHIKO EBIHARA** and KATSUO IKEDA**

This paper describes the throughput evaluation of the high level data link control protocol. The analytical models are outlined and the throughput analysis is carried out under fully loaded condition for the following three error recovery schemes: (1) Checkpoint retransmission scheme, (2) SREJ recovery scheme and (3) REJ recovery scheme. We derive the optimal packet text length which achieves the maximum throughput for each error recovery scheme. Also we denote the optimal window size for the checkpoint retransmission scheme.

1. Introduction

There has been remarkable developments recently in both the architecture and the technology of data communication systems. Line control procedures, in particular, have had a basic roll in developing communication network systems with high efficiency and reliability, and have helped the progress in the evolution of heterogeneous computer networking. Many observations and research projects have been done to estimate the characteristics of the performance [1]–[4]. The HDLC (High Level Data Link Control) protocol has been specified and recommended by ISO [5].

In a recent paper [6] the behaviour of the NRM (Normal Response Mode) of the HDLC was analyzed in respect to packet loss to determine the optimum time-out duration to minimize transmission delays. In article [7] the performance limits, the maximum achievable throughput, for the NRM of HDLC was derived and some of its analytical results have been confirmed by our measurements [2].

The analysis of the HDLC presented in this paper allows us to compute its performance limits as a function of window size, the bit error rate of a line, packet processing time and packet length for point-to-point communication.

2. Analytical Model

This paper shows analytical model of error recovery in the HDLC protocol for the following three cases:

(1) Checkpoint retransmission scheme
(2) SREJ (Selective reject) recovery scheme
(3) REJ (Rejective) recovery scheme

The checkpoint retransmission scheme is well suited for the half duplex channel of the NRM. The SREJ and REJ recovery schemes are each suited for the full duplex channel. This analysis is carried out under fully loaded packet transmission.

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3. Throughput Analyses

3.1 Checkpoint Retransmission Scheme

We suppose that each site for point-to-point communication can execute error recovery operations by the checkpoint function of the *P/F* bit. One of the nodes will send up to $(m-1)$ packets to its partner's node before stopping and waiting for an acknowledgment, where m is called a number of modulus. If a transmission error exists in a packet from the opposite site, the erroneous packet and the ones that followed are retransmitted. After completion of this retransmission, new packets are transmitted successively. If a transmission error occurs at the last frame, error recovery operations are performed after the time-out duration of the primary reply timer.

We analyze the checkpoint retransmission scheme under the following assumptions:

(1) Transmission from each site is carried out under fully loaded condition. The number of $(m-1)$ packets of fixed size are transmitted from each site.
(2) There is no propagation delay time.
(3) Zero insertion delay time is not considered.
(4) Time-fill between frames is not considered.

The checkpoint retransmission scheme is shown in Fig. 1. The upper plot shows I-command frame error recovery operation when *P* bit equals 0. The lower is for

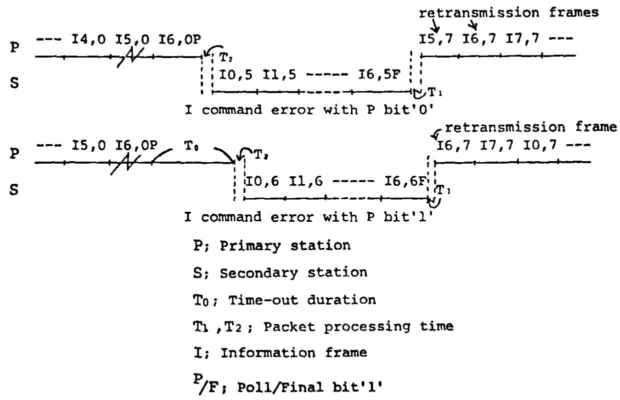

Fig. 1 Checkpoint retransmission scheme.

the case when P bit equals 1.

One cycle time is defined to be the time between the start of the transmission of the first frame and the start of next one from the primary station. Let C_i be the cycle time at the i-th cycle. The total number of bits transferred without error is referred to as the ETI (Effectively Transmitted Information) in this paper, and let B_i be the ETI at the i-th cycle. We define the average throughput G as follows.

$$G \triangleq \lim_{n\to\infty} \frac{\sum_{i=1}^{n} B_i}{\sum_{i=1}^{n} C_i} \tag{1}$$

Applying the strong law of large number [8] to Eq. (1), we readily obtain the next equation.

$$G=\frac{EB}{EC} \tag{2}$$

where EB is the mean value of the ETI and EC is the mean cycle time.

Since data transmission from the primary station and one from the secondary are independent events, we have,

$$\left.\begin{aligned} EB&=EB_1+EB_2\\ EC&=EC_1+EC_2 \end{aligned}\right\} \tag{3}$$

where EB_i and EC_i denote the ETI and the mean cycle time from the i-th station.

Let p_x be the probability of a frame being transmitted successfully from the primary station. Then we have,

$$P_x=(1-b)^{h+x} \tag{4}$$

where h, x and b represent packet header length, packet text length and the bit error rate of a physical line, respectively.

Therefore, the error probability of i-th frame transmission from the primary station to the secondary one, P_1, can be represented as follows.

$$P_1=\begin{cases} P_x^{i-1}(1-P_x) & (i=1, 2, \cdots, m-1)\\ P_x^{m-1} & (i=m) \end{cases} \tag{5}$$

Where, $i=m$ shows a case of no error on all transmission frames. It is evident that the above equation is probability density function (p.d.f.) because,

$$\Sigma P_1=\sum_{i=1}^{m-1} P_x^{i-1}(1-P_x)+P_x^{m-1}=1 \tag{6}$$

Thus, the mean value of the ETI from the primary station, EB_1, is expressed as follows:

$$\begin{aligned} EB_1&=\sum_{i=1}^{m-1} (i-1)x\cdot P_x^{i-1}\cdot(1-P_x)+(m-1)x\cdot P_x^{m-1}\\ &=\frac{P_x-P_x^m}{1-P_x}x \end{aligned} \tag{7}$$

And the one from the secondary station to the primary one, EB_2, is

$$EB_2=\frac{P_y-P_y^m}{1-P_y}y \tag{8}$$

where,

$$P_y=(1-b)^{h+y} \tag{9}$$

Here packet text length from the secondary station to the primary one and the error probability of a frame are denoted by Y and P_y, respectively.

Now, we can derive the mean cycle time, EC_1 as the sum of the transmission time with and without error.

$$\begin{aligned} EC_1&=\left\{\frac{(m-1)(h+x)}{R}+T_2\right\}P_x\\ &\quad+\left\{\frac{(m-1)(h+x)}{R}+T_0+T_2\right\}(1-P_x)\\ &=\frac{(m-1)(h+x)}{R}+T_2+(1-P_x)T_0 \end{aligned} \tag{10}$$

where R is the data transmission rate over the physical line, T_0 is the time-out period of the primary reply timer, and T_2 is the packet processing time of the secondary station including modem turn around time.

The mean cycle time of the secondary station, EC_2, is expressed by,

$$\begin{aligned} EC_2&=\left\{\frac{(m-1)(h+y)}{R}+T_1\right\}P_y\\ &\quad+\left\{\frac{(m-2)(h+y)}{R}+T_0+T_1\right\}(1-P_y)\\ &=\frac{(m-2+P_y)(h+y)}{R}+T_1+(1-P_y)T_0 \end{aligned} \tag{11}$$

where T_1 is the packet processing time including the modem turn around time.

According to Eqs. (2), (3), (7), (8), (10) and (11), we can get the average throughput. The average throughput $\bar{G}$ in balanced traffic, i.e. packet text length from the primary station and that from the secondary one are equal, is expressed as follows:

$$\bar{G}=\frac{\dfrac{2(P-P^m)}{1-P}x}{\dfrac{(h+x)(2m-3+P)}{R}+T_1+T_2+2(1-P)T_0} \tag{12}$$

where $P=P_x=P_y=(1-b)^{h+x}$.

The relationship between throughput and packet text length is depicted in Fig. 2 on the supposition that $R=$ 48 Kbps, $T_1=T_2=0.1$ sec, $T_0=10$ sec and $h=48$ bits.

Fig. 3 shows the relationship between throughput and the number of transmission frames where $x=2048$ bits, $R=48$ Kbps, $T_1=T_2=0.1$ sec, $T_0=10$ sec and $h=48$ bits.

The relationship between time-out duration and throughput is plotted in Fig. 4, in terms of packet text length, where $R=48$ Kbps, $T_1=T_2=0.1$ sec $h=48$ bits and $b=10^{-6}$.

3.2 SREJ Recovery Scheme

We consider the analysis of SREJ recovery scheme in

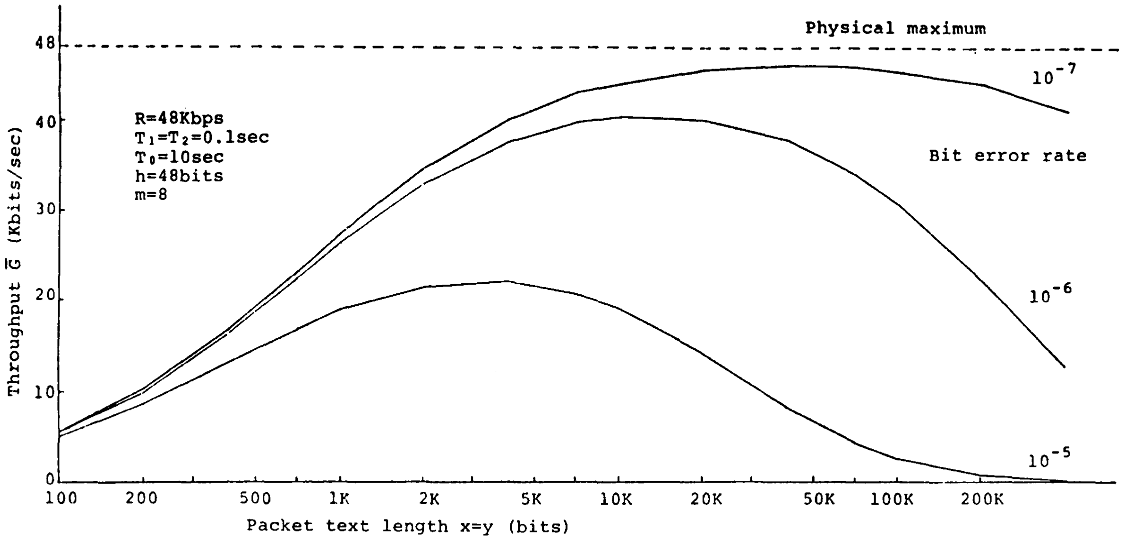

Fig. 2 Relationship between the throughput and the packet text length (Checkpoint retransmission scheme for balanced traffic).

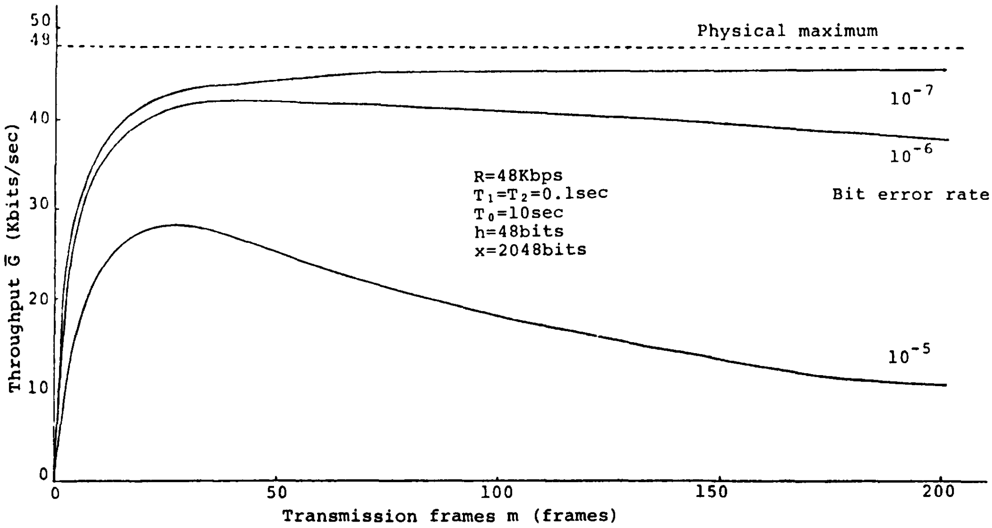

Fig. 3 Relationship between the throughput and the transmission frames (Checkpoint retransmission scheme for balanced traffic).

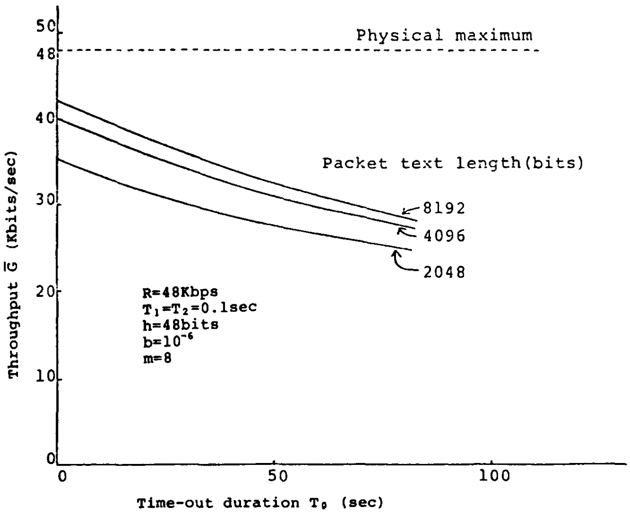

Fig. 4 Relationship between the throughput and the time-out duration (Checkpoint retransmission scheme for balanced traffic).

the environment of full duplex operation. In this scheme the receiving site transmits a SREJ control frame when an error is detected and when the transmitting site receives the SREJ control frame, it retransmits the erroneous frame. Packets with fixed length are transmitted continuously from each site. We form a model under the following assumptions:

(1) No transmission error occurs in any SREJ command response frame.

(2) The number of outstanding frames is below $(m-1)$ at any time.

(3) Fixed size packet is transmitted from each site under fully loaded condition.

(4) There is no propagation delay time.

(5) Zero insertion delay time is not considered.

(6) The time-fill is not considered.

The error recovery retransmission scheme at the primary station for a SREJ response sent from the secondary station in case of an I-command frame error from the primary station is depicted in Fig. 5.

We assume that transmissions from both sides start

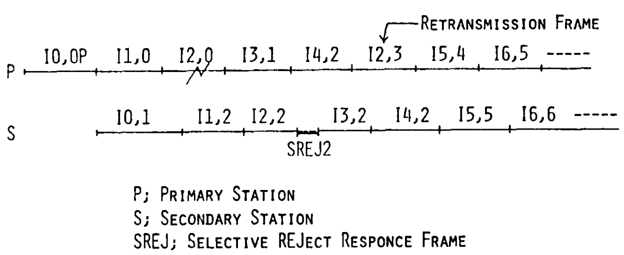

Fig. 5 SREJ response recovery of I command error (full duplex).

at the same time. The average throughput is defined as follows.

$$G \triangleq \lim_{t \leftarrow \infty} \frac{B(t)}{t} \tag{13}$$

where t denotes the time interval from the start of the communication. And $B(t)$ denotes the ETI within the time interval t. For the transmission of information, the next equations hold.

$$\left.\begin{aligned} \frac{m_1(h+x)}{R} + n_1 T_s = t \\ \frac{m_2(h+y)}{R} + n_2 T_s = t \end{aligned}\right\} \tag{14}$$

where $m_i (i=1, 2)$ is the number of transmission frames from the site i within the time interval t including retransmission frames. And $n_i (i=1, 2)$ is the number of SREJ frames transmitted from the site i within the time interval t. T_s is the time required to transmit a SREJ frame and is 48/R (no extended HDLC).

Then the success probability P_x of a I-command frame transmission from the primary station to the secondary one and unsuccessful one Q_x are given as follows:

$$\left.\begin{aligned} P_x &= (1-b)^{h+x} \\ Q_x &= 1-(1-b)^{h+x} \end{aligned}\right\} \tag{15}$$

In the same manner, P_y and Q_y are given as follows:

$$\left.\begin{aligned} P_y &= (1-b)^{h+y} \\ Q_y &= 1-(1-b)^{h+y} \end{aligned}\right\} \tag{16}$$

The number n_2 is expressed as follows provided $m_1 Q_x$ SREJ frames are transmitted from the secondary station:

$$n_2 = m_1 Q_x \tag{17}$$

In the same way, the number n_1 is

$$n_1 = m_2 Q_y \tag{18}$$

According to Eqs. (14), (17) and (18), the number of frames m_2 is

$$m_2 = \frac{\dfrac{h+x}{R} - T_s Q_x}{\dfrac{h+y}{R} - T_s Q_y} m_1 \tag{19}$$

Effectively transmitted informations of the site 1 and the site 2 in the time interval t, B_1 and B_2 are expressed as follows:

$$\left.\begin{aligned} B_1 &= m_1 x P_x \\ B_2 &= m_2 y P_y \end{aligned}\right\} \tag{20}$$

We can get the average throughput G_1 (site 1) and G_2 (site 2) from Eqs. (13), (14), (17), (18), (19) and (20):

$$\left.\begin{aligned} G_1 &= \frac{x P_x}{\dfrac{h+x}{R} + \dfrac{h+x-Q_x T_s R}{h+y-Q_y T_s R} Q_y T_s} \\ G_2 &= \frac{y P_y}{\dfrac{h+y}{R} + \dfrac{h+y-Q_y T_s R}{h+x-Q_x T_s R} Q_x T_s} \end{aligned}\right\} \tag{21}$$

The average throughput $\bar{G}$ under balanced traffic, is obtained by substituting $x=y$ in Eq. (21).

$$\bar{G} = \frac{x P_x}{\dfrac{h+x}{R} + T_s Q_x} \tag{22}$$

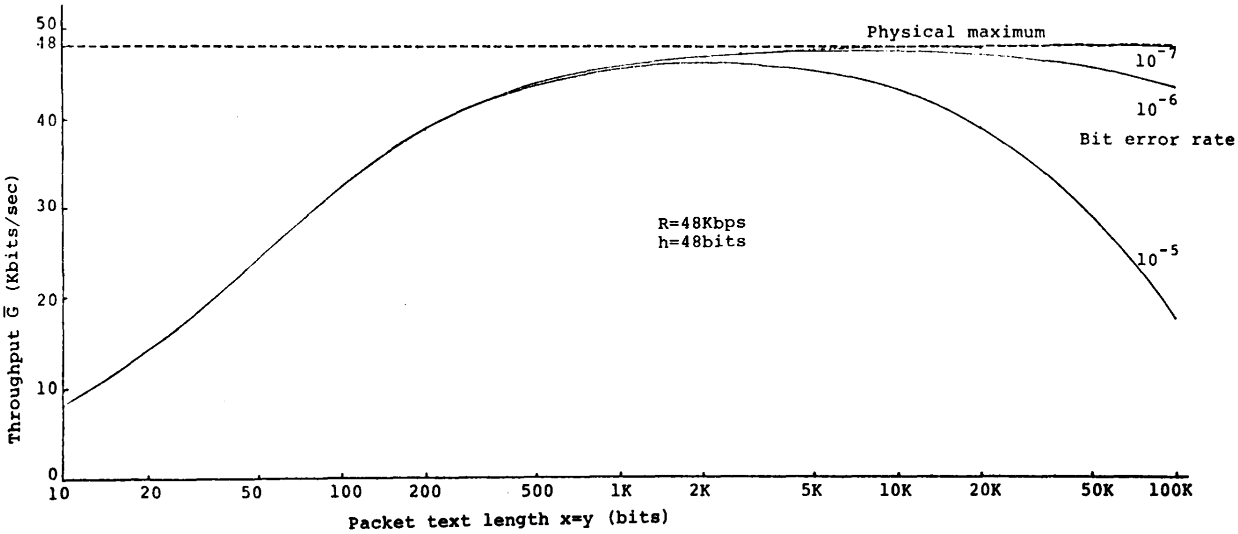

Fig. 6 Relationship between the throughthput and the packet text length (Selective Reject frame recovery).

The relationship between throughput and packet text length in connection with the bit error rate of the physical communication line is shown in Fig. 6, where $R=48$ Kbps and $h=48$ bits.

3.3 REJ Recovery Scheme

We suppose the ability to transmit REJ frame is implemented in each site. Each site can recover from error conditions by this frame. Fixed size packets are transmitted continuously in each site. When a transmission error is detected in one site, a REJ frame will be transmitted, which requests the retransmission of the erroneous frame and transmission of frames that followed the erroneous one. We consider the throughput analysis of REJ recovery scheme under the following assumptions:

(1) No transmission error occurs in any REJ command response frame.

(2) The number of outstanding frames is below $(m-1)$ at any time.

(3) Fixed size packets are transmitted from each site under fully loaded condition.

(4) There is no propagation delay time.

(5) Zero insertion delay time is not considered.

(6) The time-fill is not considered.

When an error occurs in an I-command frame sent from the primary station, the error recovery retransmission scheme at the primary station for a REJ response sent from the secondary station, is depicted in Fig. 7.

The average throughput is defined as in the previous section.

$$G \triangleq \lim_{t\to\infty} \frac{B(t)}{t} \tag{23}$$

The following equations hold for the transmission of information within the time interval t.

$$\left.\begin{aligned} &\frac{n_1P_x(h+x)}{R}+n_1Q_1Te_1+n_2Q_2Tr=t \\ &\frac{n_2P_y(h+y)}{R}+n_2Q_2Te_2+n_1Q_1Tr=t \end{aligned}\right\} \tag{24}$$

where $n_i(i=1,2)$ is the number of transmitted frames from the site i excluding those frames transmitted or retransmitted between an erroneous frame and the last of the retransmitted frames. Tr is the time required to send a REJ frame, and equals 48/R. Te_i $(i=1,2)$ is the time between the transmission of the erroneous frame

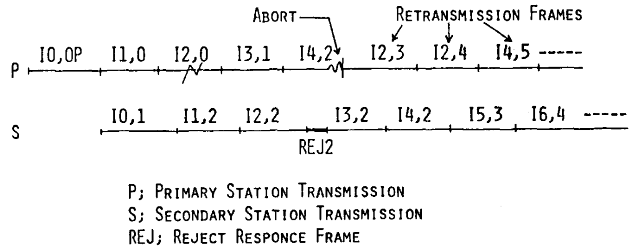

Fig. 7 REJ responce recovery of I command error (full duplex).

and the retransmission of this frame at the site i, and is expressed as follows:

$$\left.\begin{aligned} Te_1&=\frac{h+x}{R}+T_2+W_2+Tr+Ta \\ Te_2&=\frac{h+y}{R}+T_1+W_1+Tr+Ta \end{aligned}\right\} \tag{25}$$

where $T_i(i=1,2)$ denotes the detecting time of the REJ error in site i. Ta represents the abort time until the transmission stopped after a REJ frame is received. $W_i(i=1,2)$ is the mean value of waiting time for the transmission at site i, and is defined as follows:

$$\left.\begin{aligned} W_1&=\frac{h+x}{2R} \\ W_2&=\frac{h+y}{2R} \end{aligned}\right\} \tag{26}$$

We obtain the next equation from Eq. (24).

$$\frac{n_2}{n_1}=\frac{P_x(h+x)+Q_xR(Te_1-Tr)}{P_y(h+y)+Q_yR(Te_2-Tr)} \tag{27}$$

In order to make it simple, let A be the right term of Eq. (27) and we get,

$$n_2=An_1 \tag{28}$$

Next, ETI, B_1 and B_2, transmitted from each site within the time interval t, are expressed as follows:

$$\left.\begin{aligned} B_1&=n_1P_xx \\ B_2&=n_2P_yy \end{aligned}\right\} \tag{29}$$

We can obtain the average throughput G_1 and G_2 according to Eqs. (23), (24), (28) and (29), respectively.

$$\left.\begin{aligned} G_1&=\frac{xP_x}{\dfrac{P_x(h+x)}{R}+Te_1Q_x+ATrQ_y} \\ G_2&=\frac{yP_y}{\dfrac{P_y(h+y)}{R}+Te_2Q_y+\dfrac{TrQ_x}{A}} \end{aligned}\right\} \tag{30}$$

The average throughput $\bar{G}$ under the balanced traffic is given as follows:

$$\bar{G}=\frac{P}{\dfrac{P(h+x)}{R}+(Te+Tr)Q} \tag{31}$$

where $P=P_x=P_y=(1-b)^{h+x}$, $Q=Q_x=Q_y=1-(1-b)^{h+x}$, and $Te=Te_1=Te_2=(h+x)/R+T_1+W_1+Tr+Ta$.

Strictly, T_1 and T_2 are different, but even if we make them equal, generality of the analysis will not be lost.

The relationship between throughput and packet text length is shown in Fig. 8.

4. Considerations

The relationship between packet text length and

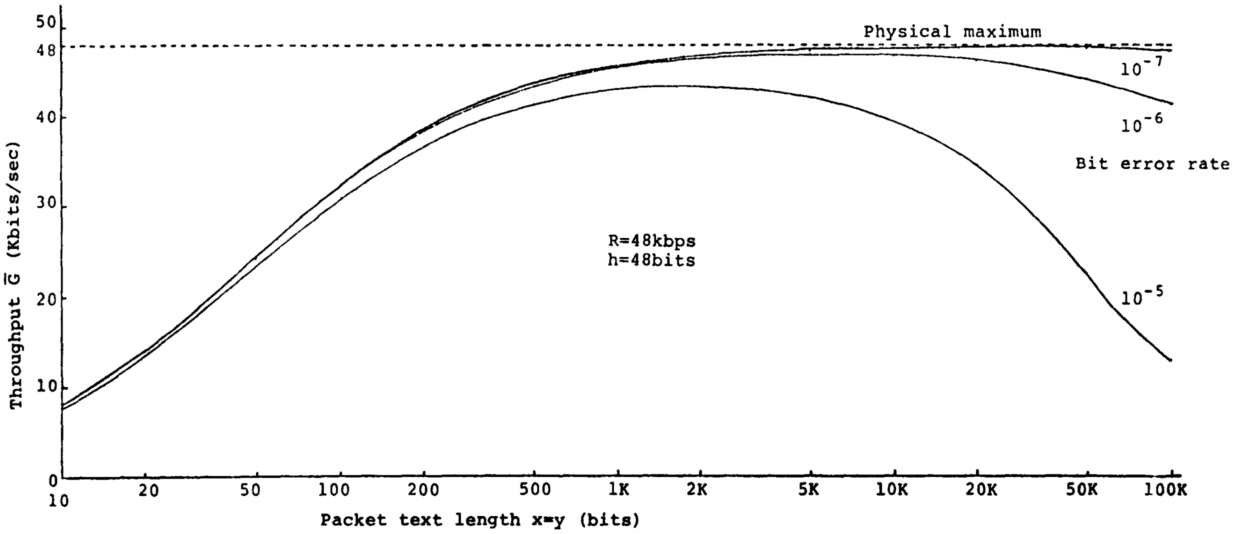

Fig. 8 Relationship between the throughput and the packet text length (Reject frame recovery).

throughput for checkpoint retransmission scheme is shown in Fig. 2. The transmission of short packets causes lower throughput, because packet header overhead per packet becomes large and can not be reduced. The transmission of long packets also makes the throughput lower, as packet error rate becomes larger in proportion to packet length and retransmission rate increases. Therefore, there exists an optimal packet text length that makes the throughput maximum for a physical line of specified error rate. The packet text length that maximizes the throughput is estimated to be about 40 Kbits, for a physical line whose data rate is 48 Kbps and the bit error rate is 10^{-7}, as is expected in the case of DDX packet exchanging network. However, it is common to use shorter packets in practical environments of communication network systems because memory capacity of CCPs (Communication Control Processor) is restricted.

The relationship between the number of transmission frames and the throughput is shown in Fig. 3. The number of transmission frames is called the window size. The additional control field, a packet header, is needed to increase the modulus number more than 8, and the supplementary overhead should be taken into consideration for the calculation of the throughput. However, the control field length for modulus 8 is adopted to outline the essential characteristics of the relationship, even though other cases of modulus more than 8 should be considered. It is also made clear that larger modulus causes lower throughput as is shown in Fig. 3, because of the line control procedure that retransmits the frames following the erroneous frame. There exists an optimal modulus number which makes the throughput maximum. The modulus number which maximizes the throughput is about 35, for a physical line whose data rate is 48 Kbps and the bit error rate is 10^{-6}. Modulus 128 is often used on satellite communications, mainly because of large transmission time delays on satellite link. But the results of analyses suggest that the throughput of transmission depends significantly on not only transmission time delays but also the number of modulus. Therefore, even for a local or in-house communication network systems with shorter transmission time delay, high throughput will be expected by using the extended packet header with modulus 128.

The relationship between time-out period of the primary reply timer and throughput for checkpoint transmission is shown in Fig. 4. It shows that shorter time-out period makes the throughput higher.

The relationship between packet text length and throughput for SREJ recovery scheme is shown in Fig. 6. When packet text length is short, the bit error rate of the physical line affects the throughput little. This fact is caused by the characteristics of SREJ recovery scheme that only erroneous frames will be transmitted when a transmission error is detected. When packet text length becomes shorter, the overhead of retransmission decreases. Therefore, the throughput does not change much with changes in the bit error rate of the physical link. The packet text size which maximizes the throughput is about 15 Kbits, for a physical line whose data rate is 48 Kbps and the bit error rate is 10^{-7}.

The relationship between packet text length and throughput for REJ recovery scheme is described in Fig. 8. The throughput of REJ recovery scheme is lower than that of SREJ for shorter packet length, because of the characteristics of REJ recovery scheme that demands the transmission of the erroneous frame and that of all the frame sent after it.

5. Conclusion

We showed analytical results of the three recovery schemes of the HDLC: (1) Checkpoint retransmission scheme, (2) SREJ recovery scheme and (3) REJ recovery scheme. We showed the relationship with the throughput and packet text size. We also discussed the optimal window size in checkpoint retransmission scheme.

Our discussion is restricted to fully loaded conditions. However, line speeds supplied by common carriers are not fast enough to allow processing-bound speed for long file transfers. Therefore, it can be said that our results are still practical and useful for local or in-house communication systems. The basic contribution of this paper is that it provides a quantitative basis for the choice of the window size of the HDLC protocol, and that it gives a comprehensive comparison between three different line control schemes by simulating the system operations. Discussion for other message arrival distributions than fully loaded is left for further research.

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