

A new long-distance communication retransmission control scheme

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Abstract: This paper proposes a new retransmission scheme for achieving high-speed transmission by reducing the retransmission delay time caused by packet loss. High-speed data packet transmission can be realized by sending user datagram protocol (UDP) packets continuously over long-distance wireless systems. The UDP characterizes connectionless communication, requiring the use of a retransmission scheme for reliable quality. However, the transmission speed will reduce with repeated retransmissions over time. The key advantage of the proposed scheme is that the retransmission waiting time can be dramatically reduced by transmitting data packets *in descending order* from the last data packet. It is verified via a computer simulation that the average peer-to-peer network configuration receiving time can be approximately halved and a 20% improvement in throughput with a packet error rate of 0.1 can be achieved by the proposed scheme.

Keywords: retransmission scheme, selective repeat, large latency communication network system

Classification: Network System

References

- [1] J. Nakamura, A. Makishima, D. Umehara, and M. Kawai, “High-speed data transmission performance on Ka-band GEO-synchronous satellite communication link using TCP,” *2002 Korea-Japan Joint Conference on Satellite Communications*, Daejeon, Korea, pp. 33–38, Oct. 2002.
- [2] K. Ienaga and M. Hirahashi, “A Study of Fast File Transfer Protocol on PCs,” *IEICE Technical Report*, NS2002-102, pp. 35–40, Sept. 2002.
- [3] RFC 793: Transmission Control Protocol.
- [4] RFC 1323: TCP Extensions for High Performance.

- [5] K. Hasegawa and I. Sasase, “A Controlling Retransmission Size H-Arq Scheme by BLER based on results of CRC Decoding,” *The 18th IEEE Annual International Symposium on Personal Indoor and Mobile Radio Communications (PIMRC 2007)*, Athens, Greece, Sept. 2007.
- [6] S. Lin, D. Costello Jr., and M. Miller, “Automatic-repeat-request error-control schemes,” *IEEE Commun. Mag.*, vol. 22, no. 12, pp. 5–17, Dec. 1984.
- [7] [Online] <http://www.opnet.com/index.html>

1 Introduction

Long-distance wireless transmission systems have been widely used in various applications and services. Such conventional transmission systems cannot handle large amounts of data, resulting in long transmission delay times. For example, it is difficult to reasonably provide Web services owing to network delays. Efforts have been expended to overcome this difficulty. User datagram protocol (UDP) packet usage is effective for real-time transmission. Consecutive UDP packet transmission enables efficient data transmission [1, 2]. However, the UDP cannot compensate for packet loss; hence, a highly reliable scheme is essential for long-distance UDP wireless transmission.

TCP is a highly reliable communication protocol because of its flow control retransmission function [3] and [4]. This retransmission function uses a slow start procedure to decrease the transmission speed when packet loss occurs. TCP is highly efficient in multi-traffic flow environments, such as the Internet. However, this study assumes wireless relay by peer-to-peer networks, making TCP flow control unsuitable. Therefore, a retransmission function that is suitable for this condition is required. Moreover, same frequencies are assumed for both transmitter and receiver in order to enhance the frequency utilization. In such scenarios, automatic repeat-request (ARQ) [5, 6], which can reliably transmit data packets, is suitable for long-distance wireless transmission.

In this paper, the suitability of selective-repeat (SR) conventional long-distance ARQ wireless transmission is first described. Problems with SR are also discussed. Next, a novel retransmission scheme based on SR, which will be widely adopted in long-distance wireless transmissions, is proposed. The key advantage of the proposed scheme is that the retransmission waiting time can be dramatically reduced by transmitting data packets *in descending order* from the last data packet. Finally, it is shown that the proposed scheme outperforms *conventional* SR from the viewpoints of the average received time and throughput.

The rest of the paper is organized as follows. The conventional scheme and its issues are discussed in Section 2. The proposed scheme is described in Section 3. A computer simulation that verifies that the proposed scheme can better improve the average receiving time and throughput performance

than the conventional scheme is presented in Section 4. Finally, conclusions of this paper and future studies are presented in Section 5.

2 Conventional retransmission control schemes and their issues

ARQ is classified into three main categories: stop-and-wait (SAW), go-back-N (GBN), and selective-repeat (SR) [6]. All these types of ARQs have advantages and disadvantages. SAW is suitable for short-range transmission and environments with high bit errors, such as wireless LAN systems, because of its simplicity. GBN is employed in the TCP/IP employed for the world wide web (WWW). Unlike SAW and GBN, SR is generally used for long-distance wireless communications that have long delays, such as satellite systems, and so on. We adopted SR in this study because of our focus on systems having long delays owing to long-distance transmission.

Fig. 1 (a) shows a *conventional* SR transmission procedure. Acknowledgment (ACK) and Selective ACK (SACK) messages are used in ARQ to determine whether or not a data packet is successfully received at a receiver. As shown in Fig. 1 (a), the data packets are successfully received in the conventional SR.

The receiver sends an SACK message, including the successful packet number information, to the transmitter after a certain time lag, called a “Timeout”, when the success/failure of reception of the last data packet is determined, regardless of other packet losses. Therefore, the transmitter will retransmit only failure packets because the successful data packet numbers are transmitted at the transmitter by the SACK message. The receiver waits for a certain time before the SACK, which is retransmitted if appropriate packets are not received. Retransmission is not employed at the transmitter site unless the SACK is received after the transmission of the last data packet.

The receiver must wait for a period corresponding to the Timeout when the final data packet cannot be received. Hence, an SACK cannot be instantaneously sent by the receiver site. This problem results in system delays and degradation of frequency utilization. Increased transmission times owing to retransmission control schemes must be avoided in long-distance wireless transmission systems in which real-time TV broadcasts are employed.

3 Proposed scheme: instantaneous SACK (I-SACK)

The retransmission flow of the proposed scheme is shown in Fig. 1 (b). The proposed scheme employs consecutive data packet transmission, similar to the conventional scheme. The key advantage of the proposed scheme is that data packets from the *last* data packet are transmitted so that the transmitter and receiver share the transmission time at which all the data packets are sent between them in advance. In other words, the data packets from the last data packet are transmitted *in descending order* in the proposed scheme. SACK can be instantaneously returned by the receiver to the transmitter by employing this scheme. We call this scheme instantaneous SACK (I-SACK).

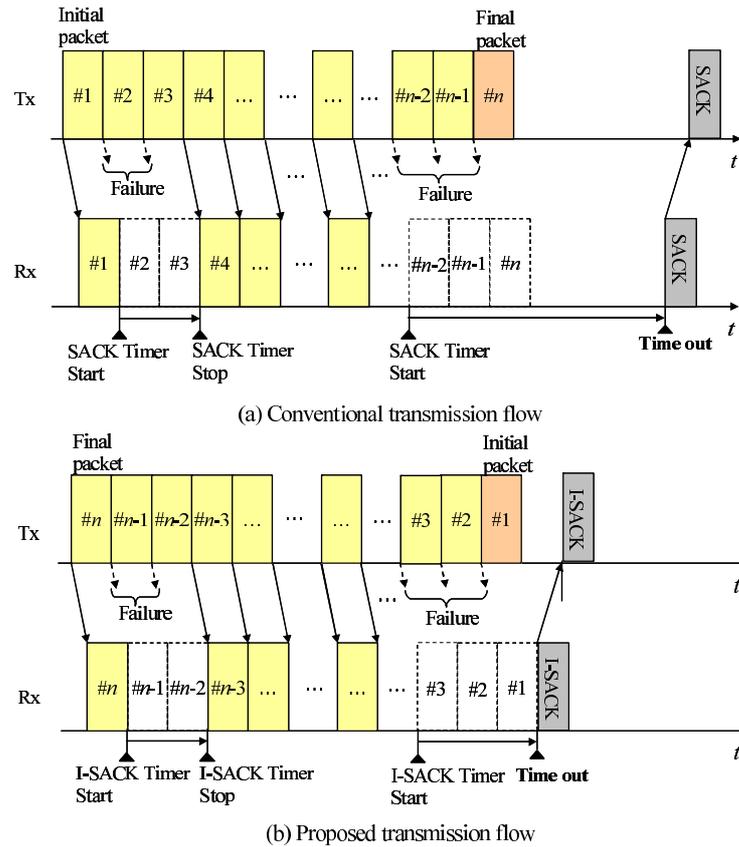


Fig. 1. Conventional/proposed selective-repeat transmission procedures.

As shown in Fig. 1 (b), five data packets are not received ($\#n-1$, $\#n-2$, $\#3$, $\#2$, and $\#1$). However, SACK can be instantaneously returned, i.e., I-SACK can be transmitted from the receiver to the transmitter because the receiver can understand the total transmission time per burst in time. The same procedure is repeated when retransmission packet loss occurs. Obviously, the proposed scheme can reduce the total reception time and enhance the throughput as compared to that enhanced by the conventional scheme.

As the scheme of achieving operation equal with the proposed scheme, there is the scheme of using a negotiation function. In this scheme, negotiation between the transmitter and the receiver is employed before consecutive data packets are transmitted and the receiver obtains the information regarding the number of total consecutive data packets from the transmitter. This scheme enables a similar reduction in reception time to the proposed scheme, as the receiver knows the reception time of the final packet because of the negotiation. However, a transmission overhead is required for this negotiation, and the transmitting efficiency will decrease. On the other hand, the proposed scheme is LIFO (last in first out), which simply reverses the consecutive data packets. The objective is achieved because the receiver obtains the sequence number.

A system that uses the negotiation function is adopted for multiuser satellite communication, etc., on TDMA (time division multiple access) circuit switching, in order to reserve the bandwidth beforehand. However, the tar-

get of this study is a peer-to-peer network, and a system in which the traffic data volume is dynamically transmitted, such as an FPU, is assumed. If the proposed method is achieved, it will be adopted with a new FPU system, etc., because the conventional system has not yet been achieved.

4 Effectiveness of the proposed scheme

The average receiving time and throughput of the proposed and conventional schemes were determined by a computer simulation and compared in order to verify the effectiveness of the proposed scheme.

The proposed scheme assumes that the system does not achieve negotiation functions in a peer-to-peer network; therefore, the proposed scheme was compared with the conventional simple SR scheme using an SACK timer.

Table I lists the simulation parameters. OPNET 11.5A [7] was used for the computer simulation. The data packet size considered in this simulation, i.e., 1,500 bytes, is the maximum Ethernet frame size. In order to evaluate the long-distance communication with high data rate, the transmission rate and delay are set to be 100 Mbps and 100 ms, respectively. The packet error was set in the range 0~0.3 because the proposed scheme is expected to be more effective than the conventional scheme when many bit errors occur.

Fig. 2 shows the normalized average receiving times and throughput versus the packet error rates (PER) of the proposed and conventional schemes. Normalized averaged received time and throughput values were obtained by considering the results for PER = 0. The average receiving time indicates the time taken for reception to be completed within one burst, including retransmission control, as shown in Fig. 2, in which the average receiving time of the conventional scheme is 1.8 times that of the proposed scheme for a PER of 0.1. However, the average receiving time for the proposed scheme did not increase much, regardless of the PER. As shown in Fig. 2, a 20% improvement in throughput was obtained for the proposed scheme relative to the conventional scheme for a PER of 0.1. Therefore, the simulation results confirm that the proposed scheme is effective from the viewpoints of both the average received time and the throughput.

5 Conclusion

In this paper, a novel retransmission scheme for long-distance communication based on selective-repeat is proposed. The main advantage of this scheme is

Table I. Simulation parameters.

Transmission rate	100 [Mbps]
Transmission delay	100 [ms]
Number of bursts	1000
Number of data packets per burst	43
Data packet size	1500 [byte]
ACK size	20 [byte]
Timeout (Conv. scheme)	1 [s]
Error generation method	random
Packet error rate	0~0.3

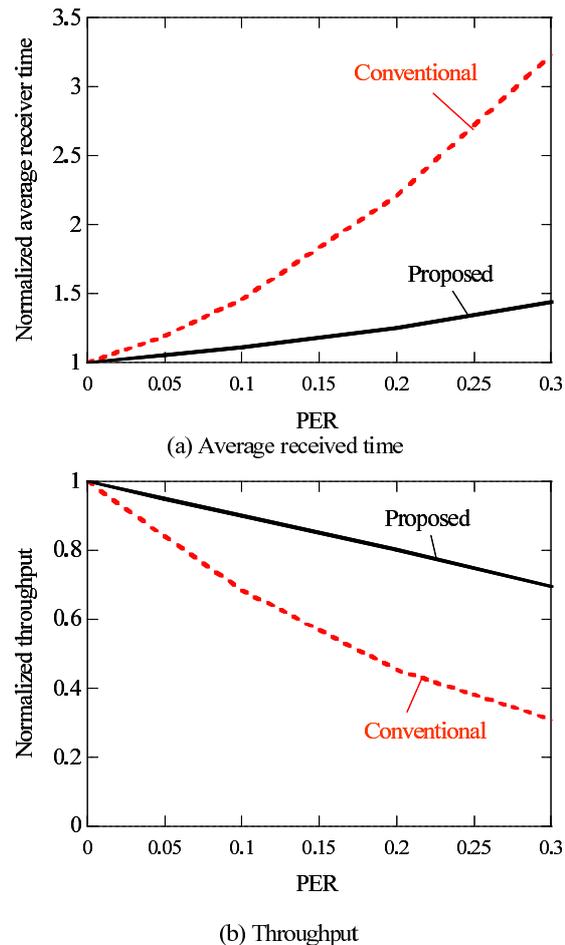


Fig. 2. Throughput and average received time versus PER.

that the retransmission waiting time can be dramatically reduced by transmitting data packets *in descending order* from the last data packet. Computer simulation results confirm that the average receiving time can be approximately halved and a 20% improvement in throughput can be achieved with a PER of 0.1 using the proposed scheme. As a future work, the evaluation when considering concrete parameters in certain systems such as satellite communications is essential.

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