PAPER

Proactive Retransmission and Buffer Management for Layered Video Transmission over Wireless Channel

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SUMMARY Hybrid FEC/ARQ, which is a mixture of forward error correction (FEC) and automatic repeat request (ARQ), is a well-known technique aiming for packet-loss recovery to guarantee quality of service (QoS) for real-time communications. In this paper, focusing on layered video transmission over wireless network environment, we propose a proactive retransmission scheme for hybrid FEC/ARQ. In the proposed scheme, a receiver host periodically sends probe packets to a sender host in order to check wireless channel state. If the sender host does not receive any probe packet during a pre-specified interval, it regards the wireless channel as being in burst loss state, and it proactively retransmits packets expected to be lost during the burst loss period. The buffer management associated with layered video coding is also taken into consideration. The performance of the proposed scheme is investigated by simulation. Numerical examples show that the proposed scheme transmits packets of the base layer more successfully than the conventional FEC/ARQ.

key words: *FEC/ARQ hybrid system, layered video transmission, proactive retransmission, buffer management*

1. Introduction

Wireless camera system has attracted considerable attention for professional use in electronic news gathering (ENG) and covering live events such as sports and concerts [4]. Typical wireless camera system consists of wireless camera transmitter and receiver, and the wireless camera receiver is connected to an editing system. Note that in the professional use of wireless camera system, there exists only a pair of transmitter and receiver, and hence the wireless network is dedicated to the use of the pair. Video data taken by a wireless camera is transmitted from the transmitter to the receiver. Then the video data is stored and edited in the editing system. In case of broadcasting live events such as marathon and Olympic games, it is important for the wireless camera system to guarantee stringent quality of service (QoS) for video image over the wireless network.

In data communication for video applications, video data packets should be transmitted to a receiver host within a bounded amount of time for continuous playout. Therefore, much effort has been devoted to guaranteeing video image quality over wireless networks.

There are two basic techniques for packet-loss recovery: forward error correction (FEC) and automatic repeat request (ARQ) [1]. FEC generates redundant data from the original data, and both original and redundant data are transmitted to a destination. The redundant data is called parity, and lost data can be reconstructed with the redundant data. There are some methods to generate parity. The simplest one is the use of exclusive OR (XOR), which generates one parity for some amount of original data. Reed-Solomon code is a famous technique to generate multiple parities, providing efficient recovery of lost data.

As FEC is one-way recovery technique based on open-loop error control, FEC is suitable for real-time applications. In general, FEC works well against random packet loss event, however, FEC is not robust enough to handle packet burst loss, which is likely to occur in wireless channels. This is because the amount of data FEC can recover is pre-determined with the estimate of the packet loss probability. The unequal error protection scheme [3], [5], [7] dynamically determines FEC redundancy according to the importance of bits or frames, however, this scheme cannot adapt FEC redundancy to ongoing loss process.

On the other hand, ARQ is an acknowledgment-based error recovery technique in which lost data packets are retransmitted by the sender host [1]. There are two well-known retransmission schemes of ARQ: Go-back-N and Selective Repeat. The former is superior in implementation simplicity, while the latter transmits packets efficiently in terms of the link utilization. In this paper, we consider Selective Repeat ARQ. ARQ can cope with burst losses, however, ARQ is not suitable for a network environment with a large round-trip time, resulting in a large delay.

In order to overcome drawbacks of FEC and ARQ, hybrid FEC/ARQ has been proposed and studied [9], [10]. In hybrid FEC/ARQ, a data block containing original and redundant data packets is transmitted to the receiver host. If a packet loss occurs, the lost packet can be recovered by ARQ retransmission or by FEC recovery. This results in a small block-loss probability, and hence QoS at application level is highly guaranteed in the sense of data loss. In wireless networks, however, available bandwidth greatly varies due to mobility and interference, resulting in a burst packet loss where consecutive packets are likely to be lost. It is difficult to recover the consecutive lost packets only with hybrid FEC/ARQ.

Recently, multi-layered scalable video coding schemes such as JPEG-2000 [8], MPEG-4 FGS [6] and SVC (H.264)

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[13] have received considerable attention, which provide encoding adaptability to time-varying available bandwidth. The scalable video coding can provide error resilience with minimal impact on the underlying coding algorithms and bandwidth efficiency, and much effort has been devoted to developing efficient and robust transmission scheme for scalable video coding over the wireless network [2], [3], [11], [12]. Most of previous work, however, focused on rate control at the transport layer, and hybrid FEC/ARQ with scalable video coding has not been fully studied yet.

In this paper, we focus on the layered video transmission for wireless camera systems, and propose a proactive retransmission scheme for hybrid FEC/ARQ. We consider a professional-use case for wireless camera system where only a pair of sender and receiver hosts for the wireless camera system exists in the wireless network. In the proposed scheme, a receiver host periodically sends probe packets to a sender host in order to check wireless channel state. If the sender host does not receive any probe packet during a pre-specified interval, it regards the wireless channel as being in burst loss state, and it proactively retransmits packets expected to be lost during the burst loss period. The buffer management related to the proactive retransmission is also taken into consideration. The performance of the proposed scheme is investigated by simulation.

The rest of this paper is structured as follows. In Sect. 2, we describe the conventional hybrid FEC/ARQ scheme. The proactive retransmission scheme for wireless channel is proposed in Sect. 3. In Sect. 4, the performance of the proposed scheme is investigated in comparison with the conventional hybrid FEC/ARQ. Finally, we conclude this paper in Sect. 5.

2. Hybrid FEC/ARQ

In this section, we describe the hybrid FEC/ARQ that our proposed scheme is based on. Note that the hybrid FEC/ARQ works in the application layer, and the functions and resources described in the following are implemented in the application layer.

Consider a video transmission for a pair of sender and receiver. The sender transmits video frames to the receiver, each of which is layer-encoded and packetized into a set of packets. Let N_L denote the number of layers provided by the encoder and S the number of packets generated from a video frame. Layer-0 is called the base layer, and it provides a basic picture of the frame. Layer-n ($1 \leq n \leq N_L - 1$) is called the nth enhancement layer, and for m and n ($m > n$), layer-m provides better quality picture than layer-n. Note that the video quality associated with layer-n can be achieved only when all layers-0 to n are available. Layer-n ($0 \leq n \leq N_L - 1$) is composed of D_n data packets and F_n FEC packets, and the total number of packets of a frame is given by

$$S = \sum_{k=0}^{N_L-1} (D_k + F_k).$$

Packets generated by the encoder are forwarded to a FIFO buffer, and then transmitted to the receiver at a constant rate. If the sender receives a negative acknowledgment (NACK) packet from the receiver, it retransmits the corresponding packet.

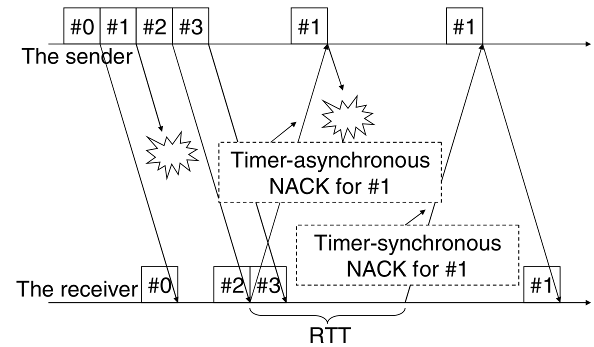

Fig. 1 Automatic repeat request.

Each packet has its own sequence number, and the receiver detects packet loss via sequence numbers. When the receiver finds that a packet from the sender was lost, it sends a NACK packet of the lost packet. This type of NACK is called a timer-asynchronous NACK. If the packet corresponding to the timer-asynchronous NACK doesn't arrive within an expected round-trip time RTT, the receiver retransmits the NACK packet. The NACK packet triggered by timeout is called a timer-synchronous NACK. An example of the ARQ is shown in Fig. 1. Note that for each group of packets consisting in a frame, this ARQ process is performed until the scheduled time for the frame to be played out.

Arriving packets are classified into layer-groups according to their contained data, and the receiver checks the number of lost packets in each layer-group. If the number of lost packets in some layer is smaller than or equal to the number of the corresponding redundant packets, those lost packets are reconstructed and the corresponding layer data is retrieved. If layers-0 to n are eventually retrieved, the frame having video quality associated with layer-n is played at its playout time. If the receiver fails to retrieve layer-0, the frame is skipped.

3. Proactive Retransmission Scheme

In this section, we describe the proactive retransmission scheme in detail. The proactive retransmission is composed of burst loss detection, packet retransmission, and buffer management.

In this paper, we consider a single-hop wireless network in which the frequency for data transmission from the sender to the receiver is different from that for reverse direction. We assume that when the wireless link is in bad condition, data transmissions in both directions are likely to fail concurrently. We consider long-term fading for the wireless links, ignoring short-term fading such as frequency-selective fading.

3.1 Burst Loss Detection

The sender has two modes for burst loss detection: normal mode and detection mode. We assume that the receiver sends a probe packet to the sender with regular interval T_{Probe}. These probe packets are used for investigating wireless link condition. If the sender in normal mode has not received any packet during a fixed period θ_B, the sender regards the wireless link as being in burst loss state where packets in transmission are lost consecutively. In the following, we call θ_B the retransmission control interval. Then the sender mode changes to detection mode. When the sender in detection mode receives a probe packet, the sender considers that the wireless link is back in good channel condition, and the sender mode transits to normal mode.

3.2 Proactive Retransmission

According to the burst loss detection, the sender retransmits the packets expected to be lost. The details are as follows.

When the wireless link is in good channel condition, the sender consecutively receives probe packets which are transmitted from the receiver with regular interval T_{Probe}. The event of probe packet loss occurs just after the wireless link state changes to burst loss state.

Suppose that a burst loss event occurs with a burst loss period u. Note that all the probe packets transmitted during u are lost. Let t_0 denote the time epoch at which the probe packet transmitted just before the burst loss period begins. Note also that this probe packet eventually reaches the sender.

Let RTT denote the round-trip time between the sender and the receiver. We define the retransmission control point t_n $(n \geq 1)$ as

$$t_n = t_0 + n\theta_B.$$

Let P_n $(n \geq 1)$ denote the period from $t_{n-1} - RTT/2$ to $t_n - RTT/2$.

Consider the first retransmission control point t_1. If $u > \theta_B$ and if no probe packet arrives at the sender for the period from t_0 to t_1, the sender mode changes to detection mode, and the sender retransmits the packets that were transmitted in period P_1. (See Fig. 2.)

If any probe packet does not arrive at the sender during the period from t_0 to t_1, the sender remains in detection mode. Note that if $t_{n-1} < t_0 + u \leq t_n$, the burst loss detection mode ends at $t'_n = t_0 + u$. In this case, we have $n-1$ retransmission control points. Note also that at t_m $(1 \leq m < n)$, the sender retransmits the packets that were transmitted in P_m.

The sender receives a probe packet at t'_n, and the sender mode changes to normal mode. At t'_n, the sender retransmits the packets sent in P'_n, the period from $t_{n-1} - RTT/2$ to $t'_n - RTT/2$.

3.3 Buffer Management

If the burst loss detection period is long, a number of packets to be transmitted are accumulated in the FIFO buffer of the sender. These packets suffer from a large delay, resulting in playout-buffer starvation in the receiver. In this subsection, we consider a buffer management scheme to prevent the degradation of video quality due to starvation. A key idea of the buffer management is that packets are prioritized based on layer type and allowable delay, and that packets with a lower priority are discarded when the FIFO buffer is congested.

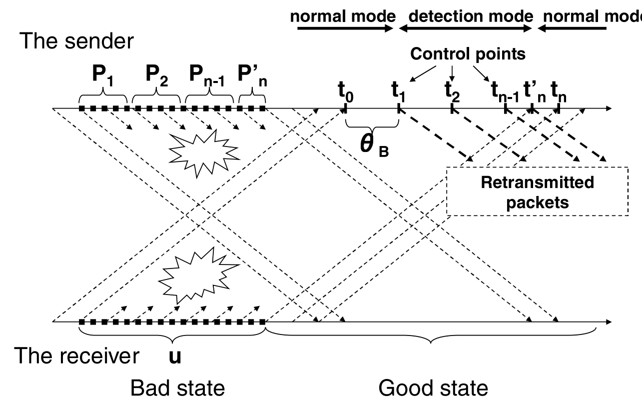

Fig. 2 Proactive retransmission.

We classify packets being in the FIFO buffer into three transmission types: *normal*, *proactive*, and *arq*. Packets with attribute *normal* are those just generated by the encoder. Packets with attribute *proactive* are those sent by proactive retransmission, while packets with attribute *arq* are those retransmitted by ARQ.

Let $F_{attribute}(u)$ define the value function of packet attribute and $F_{layer}(n)$ the value function of layer-n. Finally, we define the packet importance factor $V(u, n)$ as

$$V(u, n) = \alpha F_{attribute}(u) + (1 - \alpha) F_{layer}(n), \quad (1)$$

where α is a constant in the interval $[0, 1]$.

The buffer management policy is to guarantee high video quality, as well as to reduce the number of packets in the FIFO buffer. We consider the following buffer management policy

$$\begin{aligned} F_{attribute}(normal) &< F_{attribute}(proactive) \\ &< F_{attribute}(arq). \end{aligned}$$

The reason is as follows. First of all, packets with attribute *normal* (*normal*-packets) belonging to enhance layers can be discarded owing to the scalability of layer-encoding. Therefore, the lowest priority is allocated to *normal*-packets. Next, note that the proactive retransmission aims to recover from packet burst loss, and it works inefficiently for randomly lost packets, which can be recovered by FEC/ARQ. Note also that the number of packets with attribute *proactive* (*proactive*-packets) is likely to be larger than that of packets with attribute *arq* (*arq*-packets). This is because the original packets of the *proactive*-packets are consecutively lost during a burst-loss period, while those of the *arq*-packets are randomly lost. This bursty nature of *proactive*-packets causes a rapid saturation of the buffer. Therefore, the highest priority is allocated to *arq*-packets.

$F_{layer}(n)$ should be a decreasing function of n because for $n < m$, layer-m packets are less important than layer-n ones in terms of video quality.

The sender checks the FIFO buffer at a constant interval T_{BM}. At each checkpoint, if the number of packets in the FIFO buffer is larger than B_{Th}, the sender locates the layer-n packets with such attribute u that $V(u, n)$ takes the minimum, and then discards those packets.

4. Numerical Results

In order to evaluate the performance of the proposed scheme, we conducted simulation experiments. In the simulation experiment, we modeled the wireless link as a Gilbert-Elliot model with good and bad states. The good (bad) state period is exponentially distributed with a mean T_G (T_B). When the wireless link is in good (bad) state, a packet loss occurs with probability $P_{loss}^{(g)}$ ($P_{loss}^{(b)}$). An example of this link model is shown in Fig. 3. In the following, we set $P_{loss}^{(b)} = 1$ and $P_{loss}^{(g)} = 0.03$.

Basic parameter settings of our numerical experiments are shown in Table 1. In terms of the network environment, we assumed IEEE 802.11g and hence the maximum data transmission rate (54 Mbps) was set to the link band width. We considered a situation in which the round-trip time is not small. Note that this is a severe condition for the proposed scheme because the burst-loss detection of the proactive retransmission greatly depends on the round-trip time. In terms of video traffic, we assumed high-definition television (HDTV) in which the frame rate is 60 frame/s, and hence the frame interval is 16.6 ms. We consider JPEG2000 for video codec. JPEG 2000 is a wavelet-based image compression standard, providing efficient codestream organizations. In JPEG 2000, the amount of video data for each layer can be flexibly determined by choosing appropriate parameters. Because JPEG 2000 provides signal-to-noise ratio (SNR) scalability, video quality can be improved progressively by increasing the amount of video data. In order to evaluate the proposed scheme under a severe network condition, the parameters for video codec and the streaming bit rate were determined such that the offered load is greater than 50%.

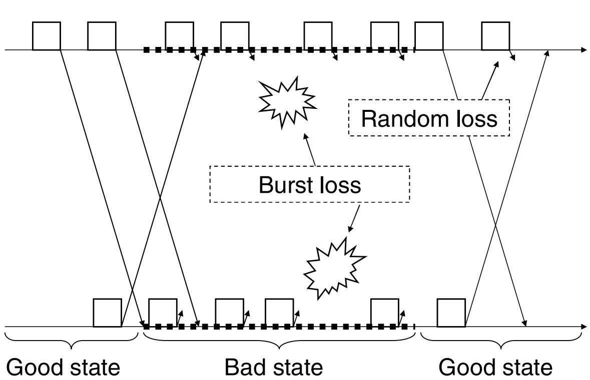

Fig. 3 Wireless link model.

As a performance measure, we consider the layer-n loss probability that one or more packets among a set of packets belonging to layers-0 to n are eventually lost at the receiver. The bandwidth usage is also investigated. We calculate the performance measures, keeping $T_B : T_G = 1 : 19$. Note that in this case, the mean packet loss probability is 0.0785.

We compare the performance of the proposed scheme with that of the conventional FEC/ARQ. Note that the conventional FEC/ARQ is equivalent to the proposed scheme without proactive retransmission.

4.1 Packet Importance Factor

First of all, we investigate how packet impact factor $V(u, n)$ affects video quality. Tables 2 and 3 show the values of $F_{attribute}(u)$ and $F_{layer}(n)$, respectively. For both $F_{attribute}(u)$ and $F_{layer}(n)$, two types are considered. In Table 2 for $F_{attribute}(u)$, packets with attribute *proactive* for type A are less important than those for type B. Similarly, in Table 3 for $F_{layer}(n)$, layer-1 packets for type C are less important than those for type D. We consider four combinations of $F_{attribute}(u)$ and $F_{layer}(n)$, as shown in Table 4.

Figures 4 to 6 show the loss probabilities of layers-0, 1, and 2, respectively, against the weighting parameter α of $V(u, n)$ in (1). We set the mean bad period $T_B = 10$ (ms), and calculated each layer loss probability in cases of $\alpha = 0.25$, 0.50 and 0.75.

Table 1 Simulation parameters.

Parameter	Value
Number of layers	$N_L = 3$
Number of data packets	$D_0 = D_1 = D_2 = 20$
Number of FEC packets	$F_0 = 2, F_1 = 1, F_2 = 0$
Packet loss probability	$P_{loss}^{(g)} = 0.03$, $P_{loss}^{(b)} = 1.0$
Probe interval	$T_{Probe} = 1.0$ [ms]
Retransmission control interval	$\theta_B = 10.0$ [ms]
Buffer management threshold	$B_{Th} = 120$
Buffer management interval	$T_{BM} = 1.0$ [ms]
Frame interval	16.6 [ms]
Data packet size	1000 [byte]
Streaming bit rate	30.2 [Mbps]
Startup delay	100.0 [ms]
Link bandwidth	54.0 [Mbps]
Round-trip time	30.0 [ms]

Table 2 Values of $F_{attribute}(u)$.

	u		
Type	*arq*	*proactive*	*normal*
A	100	25	0
B	100	75	0

Table 3 Values of $F_{layer}(n)$.

	n		
Type	0	1	2
C	100	25	0
D	100	75	0

Table 4 Combination of $F_{attribute}(u)$ and $F_{layer}(n)$.

Case	$F_{attribute}(u)$	$F_{layer}(n)$
I	A	C
II	A	D
III	B	C
IV	B	D

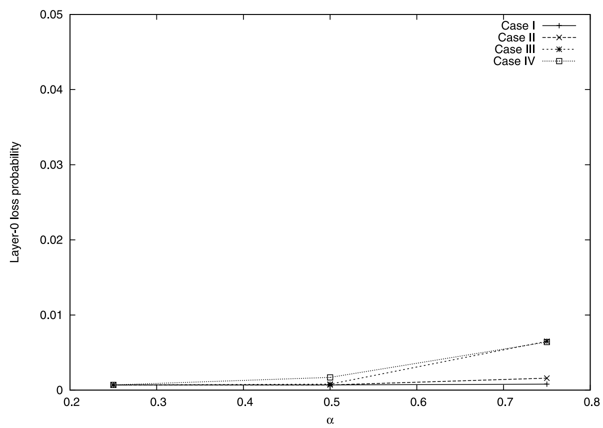

Fig. 4 Layer-0 loss probability vs. α.

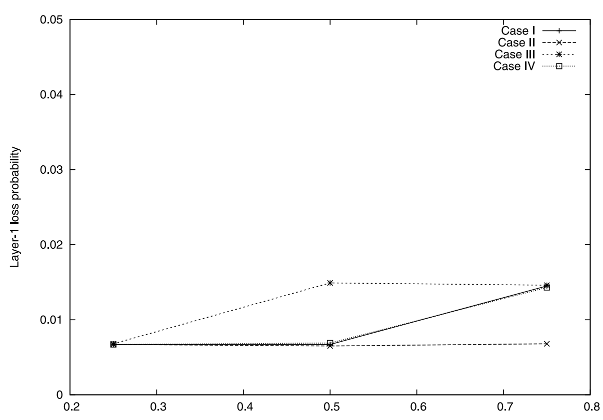

Fig. 5 Layer-1 loss probability vs. α.

In Fig. 4, the layer-0 loss probabilities for cases I and II remain constant against α, while those for cases III and IV increase gradually. Note that the layer-0 loss probabilities for all the cases are almost the same when α is smaller than or equal to 0.5. In Fig. 5, the layer-1 loss probability for case II achieves the smallest and remains constant, while other cases provide large layer-1 probabilities with an increase in α. In Fig. 6, the layer-2 loss probability for case II also remains constant, however, it achieves the largest when $\alpha = 0.75$. The layer-2 loss probabilities for cases I, III and IV decrease when α increases.

From Figs. 4 to 6, we observe that when $\alpha = 0.25$, discrepancy among four cases is quite small. When α is small, the packet importance factor gives a large value to layer-0 packets, and hence the lost packets of layer-0 are more likely to be recovered than those of layer-1 and layer-2. In this case, the buffer management mechanism does not work

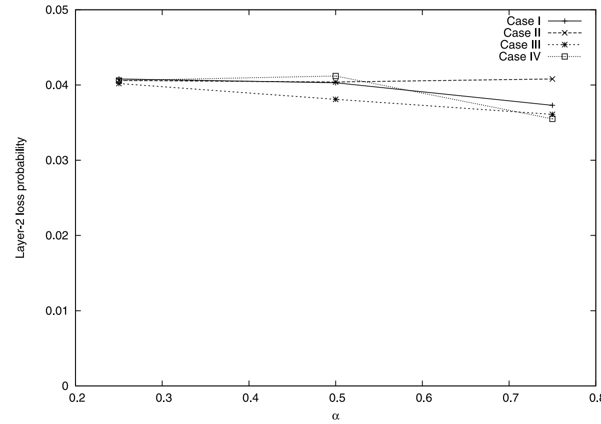

Fig. 6 Layer-2 loss probability vs. α.

efficiently for the loss recovery of layer-1 and 2 packets. With the increase in α, on the other hand, the packet importance factor for layer-0 packets is decreasing, while those for layer-1 and layer-2 packets are increasing. As a result, an enhanced-layer packet loss can be alleviated, while the layer-0 loss probability is increasing.

It is also observed from Figs. 4 to 6 that in case I, layer-0 packets are most likely to be alleviated from packet loss events. In cases III and IV, layer-2 packets are likely to be guaranteed in compensation for the increase in the layer-0 packet-loss probability. A remarkable point is that case II is significantly insensitive to α. In addition, packets of layer-0 and layer-1 are highly guaranteed for case II. In the following, we adopt case II and $\alpha = 0.25$ for $V(u,n)$ in order to investigate fundamental performance of the proposed scheme.

4.2 Effect of Retransmission Control Interval

In this subsection, we investigate the impact of the retransmission control interval θ_B on the layer-n loss probability and bandwidth usage. In this experiment, the following four cases are compared: the conventional FEC/ARQ and the proposed schemes with $\theta_B = 4.0, 10.0$, and 20.0.

Figure 7 represents the layer-0 loss probability against the mean bad period T_B. In Fig. 7, the layer-0 loss probability for the proposed scheme is smaller than that for the conventional FEC/ARQ, regardless of the values of θ_B. It is also observed that the proposed scheme with $\theta_B = 20.0$ gives the maximum loss probability among the three θ_B's. Remind that θ_B is the threshold to detect a burst loss event. A smaller θ_B enables to detect a shorter burst loss period. On the other hand, a large θ_B identifies only the burst loss event with a long duration, resulting in a large layer-0 loss probability.

In Fig. 7, the layer-0 loss probability of the proposed scheme decreases first and then increases. When the mean bad period T_B is small, the packet-loss process is likely to be random and hence lost packets are likely to be alleviated by ARQ. When T_B increases, the packet-loss process exhibits

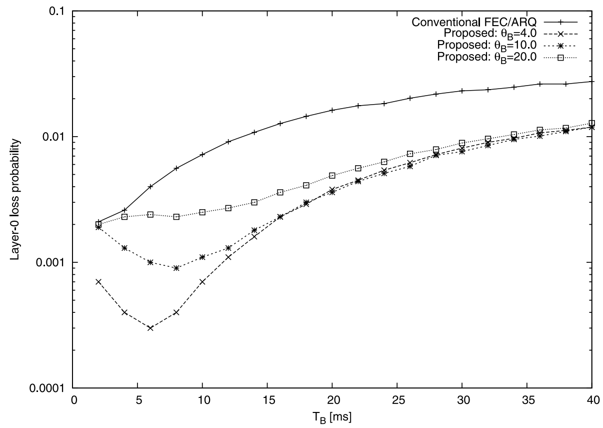

Fig. 7 Layer-0 loss probability vs. mean bad period length.

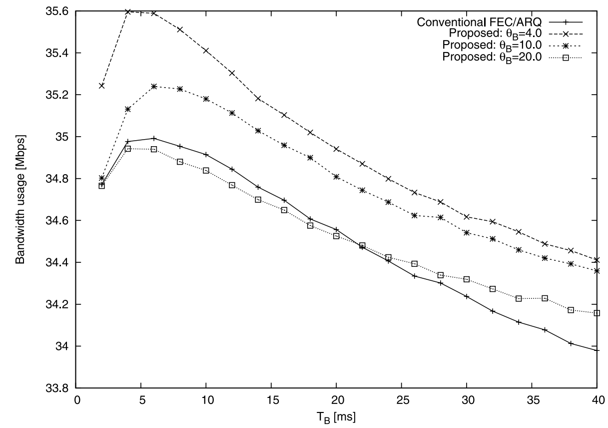

Fig. 9 Bandwidth usage vs. mean bad period length.

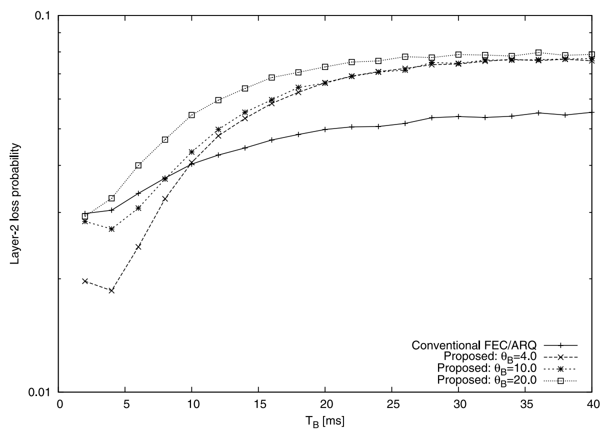

Fig. 8 Layer-2 loss probability vs. mean bad period length.

a bursty nature. In this case, the proactive retransmission mechanism works well, decreasing the layer-0 loss probability. We also observe that the layer-0 loss probabilities for the three cases of the proposed scheme increase and converge to the same value when T_B increases. When the bad period is long, packets retransmitted by the proactive retransmission are likely to be largely delayed and discarded, resulting in a large layer-0 loss probability. Note that even when the mean bad period is long, the proposed scheme achieves better performance than the conventional FEC/ARQ. This implies that the burst-loss detection is indispensable for preventing a long-term burst loss.

Figure 8 shows the layer-2 loss probability against T_B. The layer-2 loss probability for the proposed scheme exhibits the same tendency as in Fig. 7. A remarkable point in Fig. 8 is that the layer-2 loss probability for the conventional FEC/ARQ is smaller than those for the proposed scheme when T_B is greater than 10 ms. When T_B is large, packets are likely to be lost in the wireless link. In this case, the number of packets in the sender FIFO buffer increases due to the proactive retransmission. As a result, the buffer management is activated, and the packets of layer-2, which have lower priority than those of layer-0 and layer-1, are likely to be discarded.

Figure 9 represents the bandwidth usage against T_B. In this figure, the bandwidth usages for all the cases increase first and then decrease when T_B is large. Because the packet-loss process exhibits a random nature for a small T_B, the proactive retransmission is rarely activated. When T_B increases, a burst packet-loss event is likely to occur, resulting in a large number of packet retransmissions. It is also observed that the bandwidth usages of all the cases decrease from $T_B = 8.0$ to 40.0. This is due to the nature of ARQ, that is, when packets are lost in a burst loss period, the receiver can send timer-asynchronous NACK packets only after the burst loss period ends. Note that the number of *proactive*- and *arq*-packets are bounded due to delay constraint. This results in less frequent retransmission and the decrease in the bandwidth usage.

Figure 9 also shows that the proposed scheme with a smaller θ_B requires more bandwidth. This is because the proposed scheme with a small θ_B can detect burst losses more successfully and thus can retransmit packets timely.

The above results imply that the proposed scheme with a small θ_B is effective for providing a small layer-0 loss probability against a short burst loss period. However, this results in large high-layer's loss probabilities and high bandwidth usage. Note that there exists a trade-off between quality and the bandwidth usage.

4.3 Effect of Probe Interval

In this subsection, we investigate how the probe interval T_{Probe} affects the layer loss probability and bandwidth usage. In this experiment, the following four cases are compared: the conventional FEC/ARQ and the proposed schemes with $T_{Probe} = 1.0, 5.0$, and 10.0. The other parameters are the same as Table 1.

Figure 10 shows the layer-0 loss probability against T_B. We observe that the layer-0 loss probability for the proposed scheme is smaller than that for the conventional FEC/ARQ, as expected. It is also observed that when T_B is small, the proposed scheme with $T_{Probe} = 10.0$ achieves the smallest layer-0 loss probability, while that with $T_{Probe} = 1.0$ pro-

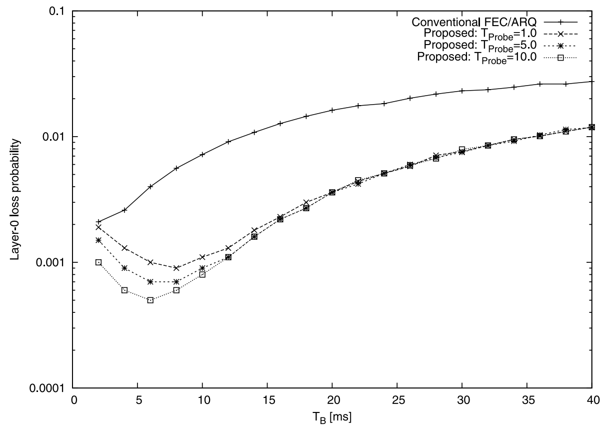

Fig. 10 Layer-0 loss probability vs. mean bad period length.

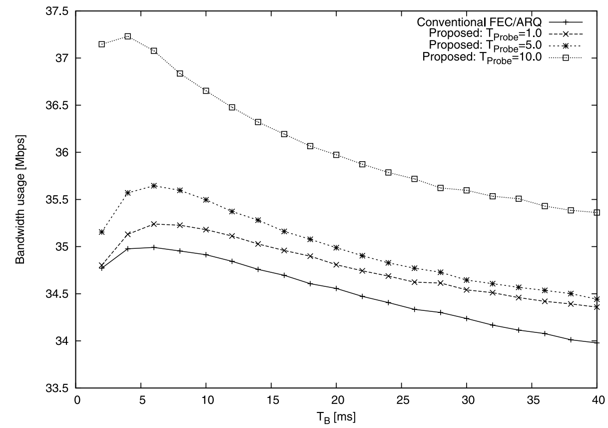

Fig. 12 Bandwidth usage vs. mean bad period length.

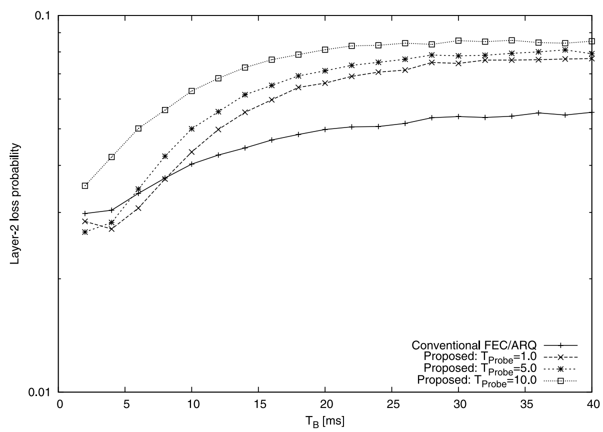

Fig. 11 Layer-2 loss probability vs. mean bad period length.

vides the largest one. Note that the estimation of a burst loss event is inaccurate when T_{Probe} is large. That is, the proactive retransmission can be activated even when a random packet loss occurs. In this case, the sender retransmits not only the lost packet but also the packets transmitted during the burst loss period estimated by the sender. The latter packets are unnecessary for the recovery of the lost packet, however, these unnecessary packets are likely to be effective to recover another packet loss.

In Fig. 10, the layer-0 loss probabilities for three cases increase and converge to the same value when T_B is greater than 10.0 ms. Note that when T_{Probe} is large, the burst-loss estimation by the sender is inaccurate and hence the proactive retransmission does not work effectively. Note also that the buffer management mechanism works independently of the proactive retransmission. Therefore, a layer-0 packet loss can be alleviated by the buffer management even for a large T_{Probe}.

Figure 11 represents the layer-2 loss probability against T_B. In contrast to Fig. 10, the proposed scheme with T_{Probe} = 10.0 exhibits the worst performance. This is because the above-mentioned unnecessary retransmission activates the buffer management control frequently. Therefore, layer-2 packets are more likely to be discarded than layer-0 packets.

Figure 12 illustrates the bandwidth usage against T_B. It is observed that the bandwidth usage of proposed scheme with T_{Probe} = 10.0 is the highest. This is also caused by unnecessary retransmission due to a long probe interval.

From Figs. 10 to 12, we can observe that the length of the probe interval does not affect the layer-0 loss probability significantly, while an excessively large T_{probe} results in the increase in the enhanced-layer loss probability and the bandwidth usage. Note that basement-layer video quality can be guaranteed even when the probe interval is not small. In terms of the probe interval setting, T_{Probe} should be determined so that $T_{Probe} < \theta_B$. Our numerical results suggest that the proposed scheme works effectively even when T_{Probe} equals a half of θ_B.

4.4 Effect of Buffer Management

In this subsection, we investigate how much the buffer management contributes to guaranteeing video quality. We conducted simulation experiments for the following five cases: the conventional FEC/ARQ, the proposed scheme without buffer management, and the proposed scheme with B_{Th} = 90, 120, and 180. We set T_B = 10.0 and change the bandwidth of the wireless link channel from 33.0 to 54.0 [Mbps].

Figure 13 shows the layer-0 loss probability against the link bandwidth. The proposed scheme with a small B_{Th} achieves a small layer-0 loss probability. This is because the buffer management is activated more frequently, and layer-1 and layer-2 packets are discarded. The proposed scheme without buffer management provides the largest layer-0 loss probability when the link bandwidth is small. This is simply because layer-0 packets are likely to wait in the FIFO buffer. When the link bandwidth is small, the layer-0 loss probability for the proposed scheme without buffer management is greater than that for the conventional FEC/ARQ. This is because the proactive retransmission of the proposed scheme forwards more packets to the FIFO buffer than the conventional FEC/ARQ.

Figure 14 represents the layer-2 loss probability against

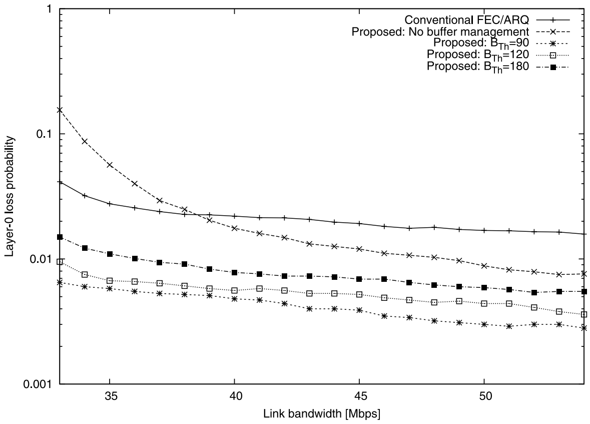

Fig. 13 Layer-0 loss probability vs. link bandwidth.

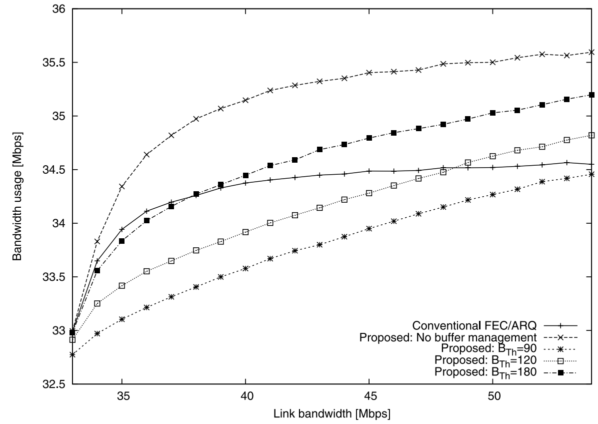

Fig. 15 Bandwidth usage vs. link bandwidth.

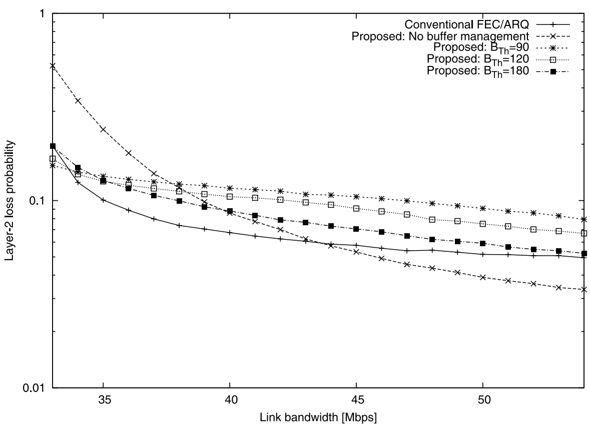

Fig. 14 Layer-2 loss probability vs. link bandwidth.

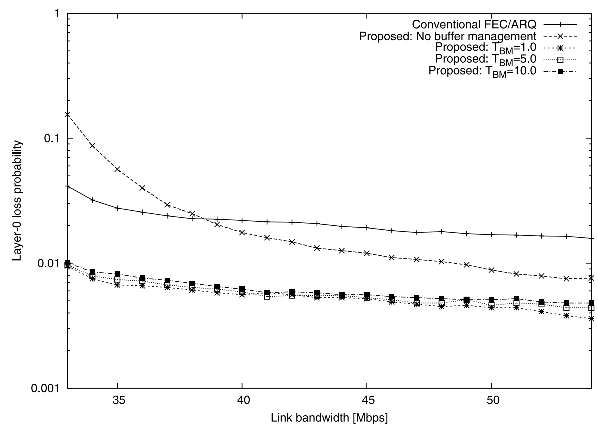

Fig. 16 Layer-0 loss probability vs. link bandwidth.

the link bandwidth. The tendencies of the conventional FEC/ARQ and the proposed scheme without buffer management are almost the same as Fig. 13. In Fig. 14, the layer-2 loss probabilities for the proposed schemes with buffer management are larger than that of the conventional FEC/ARQ. This is because layer-2 packets are likely to be discarded for the proposed scheme. It is also observed from Fig. 14 that the proposed scheme with a large B_{Th} provides a small layer-2 loss probability. This is because layer-2 packets are not discarded frequently for a large B_{Th}.

Figure 15 shows the bandwidth usage against the link bandwidth. The bandwidth of the proposed scheme without buffer management is the highest. It is also observed that the proposed scheme with a large B_{Th} uses a large bandwidth. Note that the proposed scheme without buffer management is equivalent to that with infinite B_{Th}.

From the above results, the proposed scheme with buffer management can efficiently provide a small layer-0 loss probability even when the link bandwidth is small. A large B_{Th} increases the mean number of packets in the FIFO buffer and the queueing delay, resulting in the increase in the loss probabilities of all layers when the link bandwidth is small.

4.5 Effect of Buffer Management Interval

In this subsection, we investigate the impact of the buffer management interval on video quality. The following five cases are considered: the conventional FEC/ARQ, the proposed scheme without buffer management, and the proposed scheme with T_{BM} = 1.0, 5.0, and 10.0. We set T_B = 10.0 and change the bandwidth of the wireless link channel from 33.0 to 54.0 [Mbps].

Figure 16 (Fig. 17) illustrates the layer-0 (layer-2) loss probability against the link bandwidth. Since a large T_{BM} makes the mean number of packets in the FIFO buffer large, the tendency observed in Figure 16 (Fig. 17) is similar to the case of a large B_{Th} in Fig. 13 (Fig. 14). However, discrepancies of the proposed schemes observed in Figs. 16 and 17 are small. This implies that the buffer interval T_{BM} does not affect the loss probabilities significantly.

Figure 18 shows the bandwidth usage against the link bandwidth. The proposed scheme with T_{BM} = 10.0 uses the largest bandwidth as expected, but no remarkable differences are observed.

Figures 16 to 18 indicate that the performance of the proposed scheme does not degrade drastically when T_{BM} is

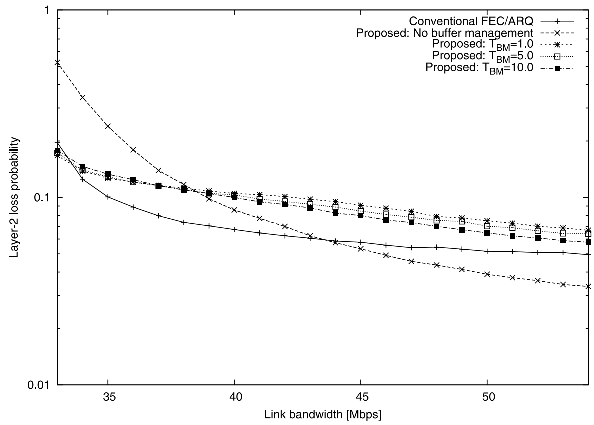

Fig. 17 Layer-2 loss probability vs. link bandwidth.

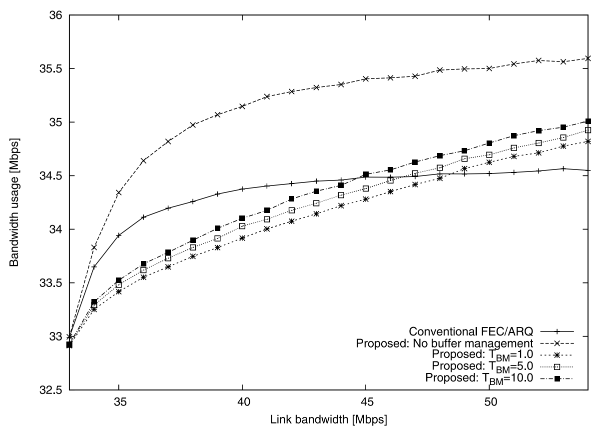

Fig. 18 Bandwidth usage vs. link bandwidth.

large. When a number of packets are accumulated in the FIFO buffer, it takes much time to transmit those packets in a low speed link. For example, suppose that the wireless link bandwidth is 40.0 Mbps and that 170 packets are in the FIFO buffer. There must be at least 120 packets in the FIFO buffer after 10 ms because about 5 packets per ms are sent from the FIFO buffer to the wireless link. In this case, the buffer management with $T_{BM} = 10.0$ [ms] works well for decreasing the layer-0 loss probability. Noting that the frame interval is 16.6 ms, these results suggest that the buffer management in which the FIFO buffer is checked once a frame is efficient enough to reduce the layer-0 loss probability.

5. Conclusion

In this paper, we have proposed a proactive retransmission scheme for hybrid FEC/ARQ, which consists of burst loss detection, packet retransmission, and buffer management. Burst loss detection, in which the receiver sending probe packets to the sender at regular interval, enables the sender to estimate the wireless link state and to detect burst losses. When the sender detects a burst loss, the sender retransmits packets expected to be lost without receiving NACK packets from the receiver. Buffer management is also taken into consideration for reduction of queueing delay.

Numerical results showed that the base layer loss probability of the proposed scheme is smaller than that of the conventional FEC/ARQ. The enhancement layer loss probability of the proposed scheme, on the other hand, is larger than that of the conventional FEC/ARQ, because packets of enhancement layer are likely to be discarded by buffer management when the mean burst loss period is large.

The retransmission control interval affects the layer loss probabilities remarkably when the mean burst loss period is small, while it has little influence when a burst loss period is large. This is because the retransmission control interval directly corresponds to the minimum burst loss period that the proposed scheme can detect. Therefore it should be decided according to the characteristics of the burst loss process in wireless channel.

The probe interval must be smaller than the transmission control interval, and it is desirable to set the probe interval to a half of the transmission control interval in order to prevent unnecessary retransmission.

Buffer management is effective to prevent the increase in the base-layer loss probability when the wireless link bandwidth is restricted. A small buffer management threshold gives a strong priority on the base layer, resulting in a small loss probability of the base layer and large loss probabilities of the enhancement layers. Buffer management interval has little impact on the loss probabilities and the bandwidth usage.

In this paper, we considered only long-term fading for wireless links. However, the wireless links are significantly affected by short-term fading due to node mobility. For future work, the effect of short-term fading on the performance of the proposed scheme should be investigated.

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