

## Peer-to-Peer Multimedia Streaming with Guaranteed QoS for Future Real-time Applications

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Peer-to-Peer multimedia streaming is expected to grow rapidly in the near future. Packet losses during transmission are a serious problem for streaming media as they result in degradation of the quality of service (QoS). Forward Error Correction (FEC) is a promising technique to recover the lost packets and improve the QoS of streaming media. However, FEC may degrade the QoS of all streaming due to the increased congestion caused by the FEC overhead when streaming sessions increase. Although streaming media can be categorized into live and on-demand streaming contents, conventional FEC methods apply the same FEC scheme for both contents without distinguishing them. In this paper, we clarify the effective ranges where each conventional FEC and Retransmission scheme works well. Then, we propose a novel FEC method that distinguishes two types of streaming media and is applied for on-demand streaming contents. It can overcome the adverse effect of the FEC overhead in on-demand streaming contents during media streaming and therefore reduce the packet loss due to the FEC overhead. As a result, the packet loss ratios of both live and on-demand streaming contents are improved. Moreover, it provides the QoS according to the requirements and environments of users by using layered coding of FEC. Thus, packet losses are recovered at each end host and do not affect the next-hop streaming. The numerical analyses show that our proposed method highly improves the packet loss ratio compared to the conventional method.

### 1. Introduction

In recent years, peer-to-peer (P2P) multimedia streaming is becoming increasingly popular with the development of broadband networks. In P2P streaming, if the quality of service (QoS) of streaming media at a current hop is degraded during transmission, the degraded media stream is forwarded to the next-hop end host and this results in all remaining end hosts receiving the lower quality of streaming media. Therefore, it is important to recover QoS degradation at each end host before forwarding streaming media. However, the QoS currently provided by the Internet is not good enough for these media and packet losses are considered as the main factor that degrades their QoS. This can be solved by the implementation of forward error correction (FEC) scheme. However, it is well-known that conventional FEC methods may degrade the QoS of all streaming media due to increased congestion caused by the FEC overhead when the number of streaming sessions increases. Therefore, it is necessary to find a proactive technique to avoid this degradation to improve their QoS. This paper is focused on a P2P streaming and proposes a novel FEC scheme that minimizes the adverse effects

of FEC overheads. Furthermore, it provides a QoS according to the requirements and equipments of users by using the layered FEC coding. Thus, packet losses are recovered at each end host and do not affect the next-hop streaming. Consequently, it can provide a higher quality media even in a wide-area streaming by repeating this loss recovery process.

The rest of this paper is organized as follows. Section 2 describes the related works of this study. In Section 3, we discuss the proposed method in detail. The performance of the proposed method is numerically evaluated and the results are presented in Section 4. Consideration for implementation is described in Section 5. Finally this paper is concluded in Section 6.

### 2. Related Works

Packet loss recovery techniques that guarantee the end-to-end QoS can be categorized into Retransmission and FEC.

In Retransmission, the sender retransmits lost packets according to the notification from the receiver. This simple retransmission mechanism is called Automatic Repeat reQuest (ARQ). The retransmission delay is unacceptable for streaming media as they are sensitive to delays. Moreover, ARQ may cause more congestion leading to a network collapse because

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packet losses in the Internet generally occur due to network congestion. It is clear that ARQ is not a suitable recovery scheme for a packet loss due to network congestion<sup>14)</sup>. Generally, the retransmission methods for streaming media need to control the source transmission rate according to the network condition<sup>1),2)</sup>. It is based on the Additive-Increase Multiplicative-Decrease (AIMD) algorithm<sup>14)</sup>. In this paper, retransmission represents the retransmission scheme with the rate control for simplicity. However, the multiplicative decrease and additive increase of the source rate result in degradation of the user-level QoS. When the P2P streaming with this technique is used, the degraded streaming media is forwarded to the next-hop end host. The QoS of these media is highly degraded as the number of end hosts forwarding the streaming increases. From these descriptions, although it is clear that Retransmission is effective to solve network congestion, it is not considered as a suitable scheme for the P2P streaming. Therefore, FEC is preferred.

In FEC, the redundant packets, which are generated from original media packets by the use of an error correction code, are transmitted along with the media packets so that the lost original packets can be recovered using these redundant packets<sup>3),14)</sup>. As this method provides resilience to loss with an acceptable latency, it is suitable for streaming media. This technique requires a redundant bandwidth called an FEC overhead. When an FEC of  $(n,k)$  block code is applied, where  $n$  is the total number of packets and  $k$  is the number of media packets, it adds  $(n-k)$  redundant FEC packets for every  $k$  media packets. Notation  $n$  and  $k$  are called block and data lengths respectively. When there are packet losses and if any  $k$  packets of  $n$  block length are received at the receiver end, all original media packets within the  $n$  block length can be recovered using FEC. In contrast to Retransmission, the quality of the lost packets retrieved using FEC packets is the same as the original media stream. When P2P streaming with the FEC is used, packet losses can be recovered using FEC if they are within the FEC recovery ability. Then, lost packets recovered by FEC are forwarded to the next hop end host.

### 2.1 Issues in FEC Techniques

Generally FEC overheads increase network congestion with the increase of streaming sessions using FEC, leading to more packet losses. These losses are beyond the FEC recovery abil-

ity and cannot be recovered using FEC. It is well-known that this increased congestion due to FEC overheads may degrade the QoS of all streaming media sent to the network<sup>4),5),14)</sup>. One popular technique to avoid this adverse effect is to limit the bandwidth where FEC is applied. When FEC is applied to a part of the whole bandwidth, the QoS of that particular part is considerably improved, but the QoS of the remaining part is slightly degraded<sup>5)</sup>. If this limitation is applied, it is valid to assume that the FEC overhead is negligible. This assumption is widely used in all conventional FEC methods and is valid for both past and current networks, where the bandwidth capacity of access networks is small and the amount of real-time traffic is not great. However, in the near future, streaming media will increase drastically due to the implementation of broadband access networks. Future networks will have more real-time traffic. Moreover, session durations of these media will tend to increase and therefore redundant FEC packets will continuously be added to the media stream for a longer duration. The assumption to neglect FEC overheads might be invalid for future networks.

In conventional streaming schemes, the same FEC techniques are applied to both live and on-demand streaming contents without distinguishing them. Since FEC is created instantaneously for live streaming contents and is sent along with media packets, it is difficult to avoid the FEC overhead being added to media packets throughout the duration of the session. However, if the FEC overheads of the on-demand streaming contents can be avoided, this will be an attractive solution because the FEC overheads are caused by only the live streaming contents. Since the demand for on-demand streaming contents such as Video on demand, e-learning system and Internet TV increases, it is likely to accommodate the adverse effect of the FEC overhead within the limitation of bandwidth where FEC can be applied without degradation. Therefore, a new FEC method for on-demand streaming contents that avoids the adverse effects of the FEC overhead during media streaming is required. As a result, the QoS of both live and on-demand streaming contents can be improved because the FEC overheads are negligible as in past and current networks.

From these, it is desirable to have the following features in such an FEC scheme.

- Recover packet losses at each end host us-

ing P2P streaming.

- Avoid QoS degradation due to FEC overheads during media streaming.

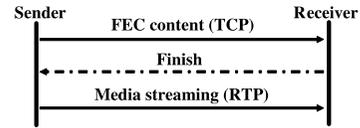
### 3. Proposed Method

The repaired stream using FEC will be the same as the original stream as described in Section 2. We focus on this advantage and propose a novel FEC method for the on-demand streaming contents.

#### 3.1 Referential Loss Recovery

Streaming media is sensitive to transmission delay. However, regarding the on-demand content, there is a permissible delay time before its communication starts. It is called the start-up delay<sup>6)</sup>. We propose the referential loss recovery (RLR) method, which separates the FEC packets from the original media packets and creates an FEC content consisting of the FEC packets when the media content is created. This FEC content is sent to the receiver end using TCP before media streaming to make sure the receiver end receives it correctly. TCP adjusts its transmission rate based on the AIMD algorithm according to the network congestion. When the retransmission due to the packet loss is done, TCP multiplicatively decreases the transmission rate since it assumes the packet loss occurs due to the network congestion. Therefore, the adverse effects of FEC overheads can be ignored. As usual, media packets are transmitted using RTP and UDP. Generally, UDP flows are unfair to competing TCP flows<sup>15)</sup>. The TCP flows reduce their sending rates in response to congestion, leaving the UDP flows to use the available bandwidth. This behavior of competing TCP and UDP flows is evaluated by way of computer simulations<sup>15)</sup>. Also many studies have addressed this well-known problem<sup>1),2),15)</sup>. In other words, TCP provides a reliable transmission at the expense of transmission time. The transmission sequence of our proposed method is shown in **Fig. 1**. It is clear that the RLR method can avoid the adverse effects of FEC overhead during media streaming at the expense of the start-up delay time. If media packets are lost, they are recovered by referring to the FEC content already received at the receiver end. Thus, the proposed method can limit the adverse effects of FEC overheads to live streaming contents and these overheads can be minimized.

When P2P streaming with RLR is used, the FEC content is delivered to each end host be-



**Fig. 1** Transmission sequence in proposed concept.

fore media streaming. Packet losses can be recovered using it at each end host before the stream is forwarded to the next-hop end host if they are within the FEC recovery ability. In other words, the degradation of QoS at one hop is minimized for the next-hop streaming and each end host can receive a higher quality of streaming media. Therefore, this technique is suitable for P2P streaming.

#### 3.2 Layered FEC Coding for Delivery of FEC Content

In RLR, the delivery method of FEC content for each end host is an important factor that must be considered. When the FEC content is generated, a layered coding is applied to provide a different FEC recovery ability according to the user requirement of QoS and equipment. The concept of layered FEC is already proposed<sup>7)</sup>. In this method, multicast groups of media streaming and FEC provisions are created separately. Some FEC multicast groups are created to provide a different loss recovery ability. The packet loss of media stream is recovered by combining appropriate FEC multicast groups. However, the adverse effects of the FEC overhead cannot be solved in this scheme. Here, we propose a layered coding method of FEC suitable for P2P streaming using RLR. Many types of FEC block codes such as Reed-Solomon (RS)<sup>18)</sup>, Tornado<sup>19)</sup> and Single Parity Check (SPC)<sup>3)</sup> codes have been deployed. It is clear that the RLR concept can be applied using these codes because the FEC content created by them are sent before media streaming starts. According to Ref. 8), SPC codes have sufficient ability to recover lost packets for audio applications. In Ref. 9), it is shown that the SPC code with a packet discarding scheme at routers provides sufficient loss recovery ability for video applications. The SPC code is the simplest approach. It is easy to compute and adapt<sup>14)</sup>. Moreover, the block length of the SPC code is generally shorter than that of RS and Tornado Codes when the coding rate of FEC is the same<sup>14)</sup>. This makes the recovery of lost packets fast. Therefore, we focus on the SPC codes. There are two layers, namely the base and the extended (ext) layers in layered

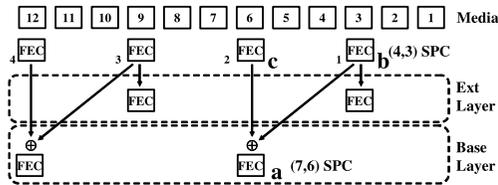


Fig. 2 Encoding process of layered FEC coding.

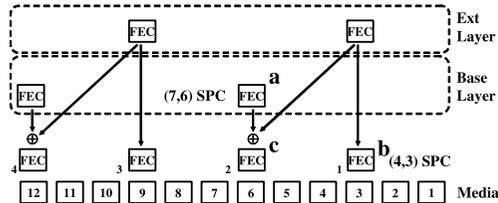


Fig. 3 Decoding process of layered FEC coding.

coding. It is assumed that FEC with (4,3) SPC code is applied. As shown in **Fig. 2**, the encoding process of layered coding at the sender end is as follows. FEC packets are created by (4,3) SPC code from media packets. The base layer will have only one FEC packet ('a') for every two FEC packets ('b' and 'c') generated by (4,3) SPC code. This results in one FEC packet in base layer for every 6 media packets. In other words, base layer FEC ('a') is generated by (7,6) SPC code and this can be easily obtained by sending the two FEC packets ('b' and 'c') through an exclusive-or gate ('b' XOR 'c' = 'a'). The ext layer consists of all odd FEC packets generated by (4,3) SPC code. The decoding process of layered coding at the receiver end is as follows. When the receiver end receives only the base layer of FEC content, lost packets are recovered using (7,6) SPC code. If both base and ext layers are received, it is possible to obtain the original (4,3) SPC generated FEC packets, as shown in **Fig. 3**. In other words, FEC packet ('b') can be directly obtained from the ext layer and FEC packet ('c') can be obtained by sending FEC packets ('a' and 'b') through an exclusive-or gate. The generations of original (4,3) SPC FEC packets are done before receiving the media packets since FEC content is sent in advance. Thus, lost packets are recovered using (4,3) SPC code.

#### 4. Performance Evaluations

The performance of the proposed RLR is numerically evaluated compared to conventional FEC methods. First, the problem due to the FEC overhead in conventional FEC methods is

presented. After the basic characteristic of the proposed method is analyzed for unicast communications, the performance of RLR in P2P streaming is evaluated. Finally, the FEC content size is evaluated and then the contributions of the proposed RLR are summarized.

Packet loss probability is increased with the network load at router/switch of the bottleneck link. Here, the network load is defined as the ratio of the bandwidth used for the traffic to the bandwidth capacity of a network link. In packet switching networks, Ref. 16) introduces that the performance of a packet switch with multiple input and output ports can be approximately analyzed using M/M/1 or M/G/1 model. Packet loss events of the switch at the bottleneck link can be treated as a queuing system with a definite queue length. The M/M/1/K queuing system can be easily used to model the loss process and the performance of the FEC technique in packet switching networks is evaluated using this queue<sup>17)</sup>. Therefore, in order to show the improvement of the proposed method compared to conventional methods, their performances are evaluated using M/M/1/K queue here.

##### 4.1 Analysis of Issues in Conventional FEC Methods

Let  $\rho_0$  denote the initial network load without considering the FEC overhead. When FEC is applied to all real-time traffic, it requires an extra bandwidth, resulting in an increase of the network load. The realistic network load,  $\rho_{fec}$  considering the effect of the FEC overhead is given by

$$\rho_{fec} = (1 - \alpha)\rho_0 + \frac{\alpha\rho_0}{R_{cr}} \quad (1)$$

where  $\alpha$  and  $R_{cr}$  are the percentage of real-time traffic with FEC and the coding rate respectively. The  $R_{cr}$  is given by  $k/n$ , where  $k$  and  $n$  are data and block lengths, respectively. Packet loss rate considering the FEC overhead during transmission is calculated as follows.

$$P_{loss\_fec} = \frac{1 - \rho_{fec}}{1 - \rho_{fec}^{K+1}} \rho_{fec}^K \quad (2)$$

where  $K$  is buffer size. From Eqs. (1) and (2), it is clear that the FEC overhead increases the network load, leading to the increase of packet loss rate during transmission. Effective packet loss ratio (EPLR) of the conventional FEC,  $P_{eplr\_fec}$  is given by Eq. (3) as follows.

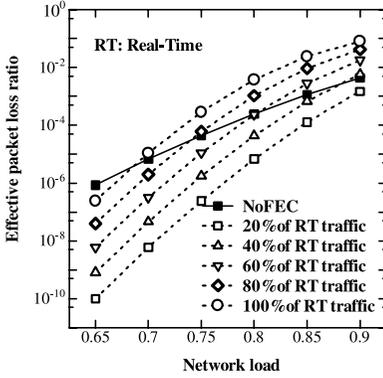


Fig. 4 Analysis of Issues in conventional FEC.

$$\sum_{i=n-k+1}^n n C_i P_{loss\_fec}^i (1 - P_{loss\_fec})^{n-i} \quad (3)$$

The EPLR is defined as the loss ratio of the lost packets that cannot be recovered even after using FEC. **Figure 4** shows the EPLR of conventional FEC with different percentages of real-time traffic to the total traffic, according to the network load. Here, the (6,5) SPC code is applied as an example.  $K$  is set to 30 and its results are presented because the improvements of the proposed method compared to conventional methods are almost constant regardless of different values of  $K$ . Figure 4 shows that the EPLRs of all methods increase with the network load. The conventional FEC methods increase EPLRs with the percentage of real-time traffic, and most of them are higher than EPLR without FEC. In other words, the FEC overhead degrades the QoS of streaming compared to the streaming without FEC. Here, we define the point at the intersection of a line given by No FEC with lines given by conventional FEC methods as the maximum percentages where FEC can improve the packet loss ratio. When the network load is less than this threshold, FEC is considered as a suitable scheme to recover packet losses. When it is more than this threshold, Retransmission is considered as a suitable scheme because the conventional FEC degrades the performance compared to No FEC situation and therefore Retransmission is effective to solve the network congestion.

#### 4.2 Evaluations for Unicast Communications

Streaming media can be categorized into live and on-demand streaming contents. Therefore, the percentage of real-time traffic can be written as  $\alpha = \alpha_{Live} + \alpha_{Ondemand}$ , where  $\alpha_{Live}$

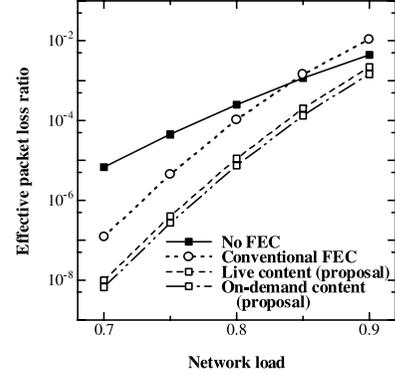


Fig. 5 Comparison of RLR to conventional FEC methods for unicast communication.

and  $\alpha_{Ondemand}$  are the percentages of live and on-demand streaming traffic to real-time traffic respectively. In the RLR scheme, the FEC overhead is caused by only the live streaming contents. Therefore, the realistic network load of the RLR scheme minimizing the adverse effect of the FEC overheads,  $\rho_{rlr}$  and its packet loss rate during transmission are given by;

$$\rho_{rlr} = \rho_0 + \alpha_{Live} \alpha \rho_0 \left( \frac{1}{R_{cr}} - 1 \right) \quad (4)$$

$$P_{loss\_rlr} = \frac{1 - \rho_{rlr}}{1 - \rho_{rlr}^{K+1}} \rho_{rlr}^K \quad (5)$$

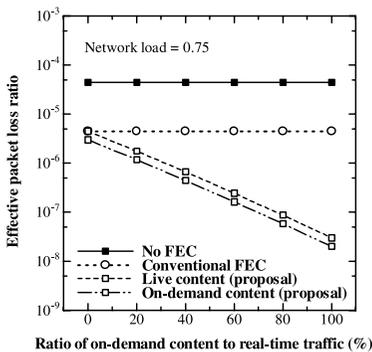
It is clear that the packet loss rate given by Eq. (5) is smaller than that given by Eq. (2) because the adverse effect of the FEC overhead is minimized. As this packet loss rate during transmission is applied to both live and on-demand streaming contents, the QoS of both contents is improved. The EPLRs of live and on-demand streaming contents using RLR are given by Eqs. (6) and (7) respectively.

$$\sum_{i=n-k+1}^n n C_i P_{loss\_rlr}^i (1 - P_{loss\_rlr})^{n-i} \quad (6)$$

$$\sum_{i=n-k+1}^k k C_i P_{loss\_rlr}^i (1 - P_{loss\_rlr})^{k-i} \quad (7)$$

**Figure 5** shows the EPLR of RLR compared to conventional FEC for live and on-demand streaming contents and without FEC. For simplicity and better comparability with Fig. 4, the (6,5) SPC code is applied and  $K$  is set to 30. Here, the percentage of real-time traffic is set to 50% as an example since real-time traffic consumes much of the bandwidth and the number of sessions of such traffic increases rapidly with the development of broadband networks. Also, the percentage of on-

demand streaming traffic is set to 50% of the real-time traffic as an example. In conventional FEC, the performances of live and on-demand streaming contents are the same because the same FEC method is applied. The proposed method greatly reduces the EPLRs of both contents compared to the conventional method although it is applied for only the on-demand streaming contents, as shown in Fig. 5. The on-demand streaming contents slightly reduce the EPLR compared to the live streaming contents in the RLR scheme. From this evaluation, the improvements of RLR compared to conventional methods can be considered as almost constant, regardless of the network load. Then, the performance of RLR is evaluated according to the various ratios of on-demand streaming contents to real-time traffic when the network load is fixed. The results of different network loads indicate the same pattern and therefore, the result of network load with 0.75 is presented here. **Figure 6** shows that the EPLR of RLR reduces with the increase of the ratio of on-demand streaming. Although the demand for on-demand streaming contents increases as described in Section 2.1, its ratio depends on the network situations. Then we evaluated the performance of RLR using P2P streaming presented in the next section for the various ratios of the on-demand streaming contents in order to determine the design examples of RLR. From these evaluations, it is observed that the performance improvements for P2P streaming according to the ratio of on-demand contents are based on the results of Fig. 6. For simplicity, 40% and 70% of on-demand contents are selected as examples and their results are presented in the subsequent evaluations.



**Fig. 6** Comparison of RLR to conventional FEC methods according to different ratio of on-demand streaming content.

### 4.3 Evaluations for P2P Streaming

Application Level Multicast (ALM)<sup>10</sup> is used as an example of P2P streaming and its characteristics are evaluated by a 3-level binary tree. Let M be the level of binary tree, the EPLRs of all the methods at each level in the binary tree are generally given by Eq. (8).

$$1 - \prod_{j=1}^M \left\{ 1 - \sum_{i=n_j-k_j+1}^Z z C_i P_{loss}^i (1 - P_{loss})^{Z-i} \right\} \quad (8)$$

where  $n_j$  and  $k_j$  are block and data lengths used at each level respectively. The EPLR of conventional FEC is given by Eq. (8) when  $P_{loss}$  and Z are  $P_{loss\_fec}$  and  $n_j$  respectively. In RLR, EPLR of live content is given by Eq. (8) when  $P_{loss}$  and Z are  $P_{loss\_rlr}$  and  $n_j$  respectively, and EPLR of on-demand content is given by Eq. (8) when  $P_{loss}$  and Z are  $P_{loss\_rlr}$  and  $k_j$  respectively. **Table 1** shows 4 proposed methods with the adoption patterns of delivery of base and ext layers at each level in the 3-level binary tree.

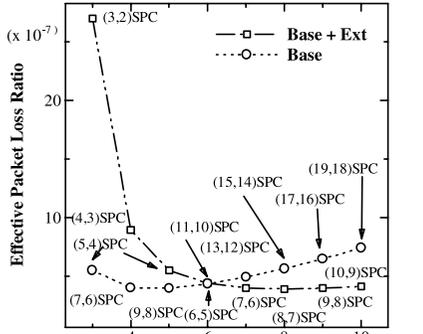
Suitable FEC codes for base and ext layers are determined by evaluating the EPLR of each SPC code as a design example of RLR. **Table 2** shows the possible combinations of different FEC codes for base and ext layers. For example, when (3,2) SPC code is selected as the FEC code using both base and ext layers, the (5,4) SPC code must be the FEC code of the base layer. **Figure 7** shows the EPLR of each combination of FEC codes for base and ext layers. In Fig. 7(a), when the combination IV is used, the performances of each code are almost the same. When the block length for both base and ext layers is less than 6, the performance of the base and ext layers is worse

**Table 1** Delivery way of base and ext layers.

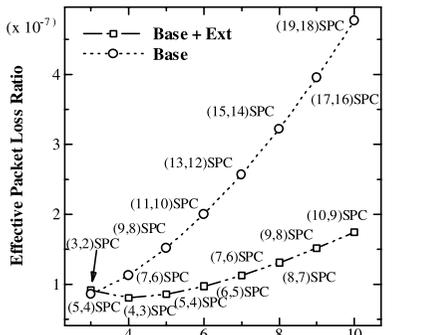
Delivery Method	Level in binary tree		
	1	2	3
1	Base+Ext	Base+Ext	Base+Ext
2	Base+Ext	Base+Ext	Base
3	Base+Ext	Base	Base
4	Base	Base	Base

**Table 2** Examples of possible combinations of FEC codes for layered coding.

Layer	Combinations of SPC code			
	I	II	III	IV
Base	(5,4)	(7,6)	(9,8)	(11,10)
Base+Ext	(3,2)	(4,3)	(5,4)	(6,5)



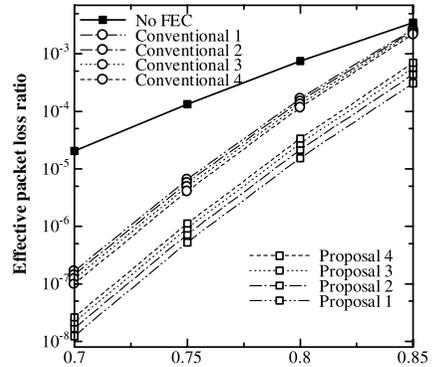
Block length of FEC code using base&ext layers (packet)  
(a) 40% of on-demand streaming content



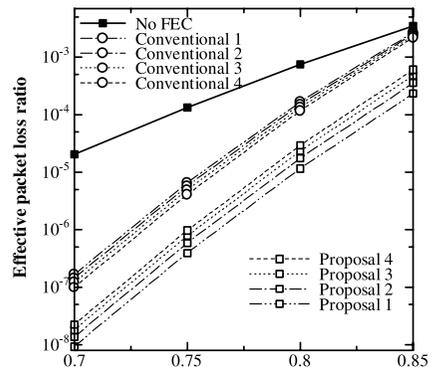
Block length of FEC code using base&ext layers (packet)  
(b) 70% of on-demand streaming content

Fig. 7 Evaluations of each combination of layered coding.

than that of the base layer because of the FEC overhead. However, when it is longer than 6, the performance of the base and ext layers is better than that of the base layer. This is because the adverse effect of the FEC overhead gets smaller and the performance is improved according to the FEC recovery ability. In contrast, in Fig. 7 (b), the performance is improved according to the FEC recovery ability except for the combination I. This difference between Figures 7 (a) and 7 (b) is due to the different ratio of on-demand contents. Here, the (15,14) SPC and (8,7) SPC codes are selected for the FEC contents of base layer and both base and ext layers respectively as a design example of the RLR. In the conventional FEC method, the applied patterns of FEC codes at each level in the tree are the same as compared in Table 1. **Figure 8** shows the EPLRs of all the methods at level 3 for 70% of on-demand streaming contents. The proposed methods improve EPLS of both live and on-demand streaming contents compared to conventional methods. From these results, it is observed that in RLR the perfor-



(a) Live streaming contents

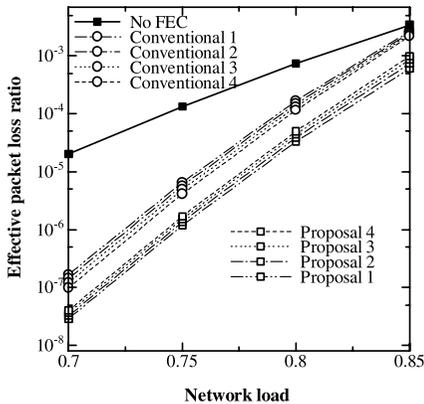


(b) On-demand streaming contents

Fig. 8 Effective packet loss ratio at level 3 for 70% of on-demand streaming contents.

mance of proposal 1 is better than any other proposals although in conventional FEC methods it is vice versa. It is considered that since the RLR can minimize the adverse effect of the FEC overhead, proposal 1 using the FEC code with the higher recovery ability applied for all levels provides the best performance. However, in the conventional method 1, its overhead causes the degradation, which results in the worst performance. It is also observed that the improvements of live and on-demand contents in Fig. 8 indicate the same pattern compared to conventional methods. Therefore, the result of only the on-demand content of which the ratio is 40% is presented for comparison. **Figure 9** shows that the performance of RLR indicates the same pattern as Fig. 8, but the improvements of EPLR compared to conventional methods and among the different delivery patterns of the base and ext layers are small compared to Fig. 8.

From these, the proposed RLR, which can re-



**Fig. 9** Effective packet loss ratio of on-demand contents at level 3 for 40% of on-demand contents.

cover lost packets at each end host minimizing the degradation due to FEC overheads, is suitable for P2P streaming to provide a higher media quality in wide area networks.

#### 4.4 FEC Content Size

FEC content size depends on the media content length, the bit rate of media stream and the applied FEC code. Here, the ITU G.726 ADPCM with 32 kbps is adopted as an audio encoding. The on-off model is used as the voice source traffic since human speech consists of talk-spurts and silence gaps that are known as on-off patterns. The holding time in the on and off periods is assumed to be exponentially distributed with mean values of 1.004 s and 1.587 s, respectively<sup>11</sup>. The silence suppression is also applied. Then, the (7,6) SPC and (4,3) SPC codes are applied as the FEC codes of base and ext layers respectively, to reduce the block delay since the generation of audio traffic is periodic and is not bursty. For the video source, a constant bit rate traffic with 1 Mbps average rate is used for the evaluation. The (15,14) SPC and (8,7) SPC codes are applied as the FEC codes of base and ext layers respectively. **Table 3** shows the FEC content size of each base and ext layers for the audio and video sources according to the different play out time of the media. Their sizes are increased with the increase of their play out time. However, all content sizes are small. Moreover, if a technique of Bit Torrent<sup>12</sup> or Parallel Download<sup>13</sup> is used to make the best use of P2P technology, its time will be shorter because all end hosts that have already downloaded the FEC content can be servers of this technique.

To summarize all the evaluations, the effective ranges where each conventional FEC and

**Table 3** FEC content size (MB).

Media play out time (min.)	Audio (G.726)		Video (1 Mbps)	
	Base	Ext	Base	Ext
5	0.076	0.076	2.679	2.679
10	0.151	0.151	5.357	5.357
15	0.227	0.227	8.036	8.036
20	0.303	0.303	10.71	10.71
25	0.378	0.378	13.39	13.39
30	0.454	0.454	16.07	16.07

**Table 4** Summary of 4 proposed methods.

Method	EPLR	FEC content size
Proposal 1	Best	Good
Proposal 2	Much better	Better
Proposal 3	Better	Much better
Proposal 4	Good	Best

Retransmission scheme works well is clarified since the conventional FEC degrades EPLR with the increase of real-time traffic. The proposed RLR can minimize the adverse effect of FEC overheads and highly improve EPLRs of both live and on-demand streaming contents although it is applied for only the on-demand streaming contents. Its EPLR is improved as the ratio of the on-demand streaming contents increases. The evaluations to determine suitable FEC codes for the layered coding in RLR shows that there is a combination of FEC codes for base and ext layers where the performance of only the base layer is almost the same as that of both base and ext layers because of the FEC overheads. This specific combination varies according to the change of ratio of on-demand contents. In RLR using P2P streaming, there are 4 patterns of delivery of the base and ext layers for a 3-level binary tree. **Table 4** summarizes the performances of all the proposed methods in terms of EPLR and FEC content size. Proposal 1 is the best performance of EPLR but increases the FEC content size as both base and ext layers are delivered to each end host. Proposal 4 is vice versa to proposal 1. Proposals 2 and 3 are intermediate performances between proposal 1 and 4. Proposal 2 is better and worse than proposal 3 for the EPLR and the FEC content size, respectively. For example, when the ratio of the on-demand content is 70%, proposals 1, 2 and 3 reduce EPLR by about 40%, 60% and 80% of Proposal 4 respectively but need about 200%, 167% and 133% of FEC content size compared to proposal 4 respectively. Thus, this method can provide flexible loss recovery for the users, and each method is selected according to the requirements of the

users and the applications.

Multicasting is basically unsuitable for on-demand streaming contents although it can effectively reduce the traffic. To solve this, the Asynchronous Multicasting (AM) technique is discussed<sup>20)</sup>. In AM, the on-demand content is sent by multicasting to users whose requests are made at about the same time. It is called shared flow. The initial data of the content that subsequent receivers cannot obtain is delivered to the subsequent receivers. It is called patch flow. One receiver receives a shared flow only, and the other subsequent receivers receive both shared and patch flows. Thus, it makes the multicasting of on-demand contents effective. The AM has been developed for IP multicast. However, it is possible to apply it to ALM and P2P streaming because the difference from IP multicast is that the function of the multicasting is implemented in the application layer. Also, joining and leaving the multicast tree can be done according to the function of ALM. Therefore, the AM using P2P streaming with RLR is effective in providing an on-demand streaming service with a higher QoS.

### 5. Consideration for Implementation

In conventional media streaming, the FEC packets are usually sent as a separate RTP stream, on a different UDP port but to the same destination address, to make FEC methods backward-compatible<sup>14)</sup>. Therefore, the synchronization between FEC and media packets is important. In Ref. 3), the FEC packet format of RTP is described. Its header has the sequence number base and mask fields that are used for the synchronization. The sequence number base field indicates the minimum sequence number of the original media packets composing this FEC packet. In other words, this field indicates the first media packet in the FEC block length. The mask field is a bit mask indicating which of the packets following the sequence number base are included in the FEC operation. These 2 fields can detect the appropriate media packets so that each FEC packet can be applied correctly. Moreover, they can detect the correspondent FEC packets between base and ext layers. In RLR, the sequence number base fields of the corresponding packets in both base and ext layers indicate the same number. However, the value of the mask field in the base layer packets is longer than the ext layer packets. Thus, it is possible to detect the

appropriate combination between FEC packets in base and ext layers. The difference between RLR and conventional streaming is the location of FEC packets. The FEC stream generated from the FEC content already at the receiver ends in the RLR can be considered as a separate RTP stream in conventional streaming. Therefore, the RLR is feasible with the same packet format as Ref. 3) and P2P software.

### 6. Conclusion

This paper proposes a novel packet loss recovery method to provide a guaranteed QoS for P2P streaming. In the proposed method, the adverse effect of the FEC overhead is minimized and the quality of streaming for both live and on-demand streaming contents is highly improved. By implementing of layered coding of FEC, it can provide a different QoS according to the user requirements and equipment. Packet losses are recovered at each end host and do not affect the next-hop streaming. Thus, it can provide a higher quality of streaming in wide-area networks by repeating this loss recovery process. The FEC content sizes to download during start-up delay time are small and we believe this start-up delay is within tolerable limits. This scheme does not need the feedback information and is a suitable scheme for P2P streaming in the future broadband Internet.

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