# 고속 전송을 위한 적응형 FEC 및 전송률 제어

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# Adaptive FEC and Rate Adaptation for High-speed Transport

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## 요 약

본 논문은 적응형 오류제어 기법을 바탕으로 신뢰성 있는 UDP 기반의 미디어 고속 전송을 제안한다. 제안된 적응형 전송기법은 대역폭의 변화에 효과적으로 대처하기 위해서 네트워크 모니터링을 기반으로 잉여 데이터의 양을 제어한다. 수신측 피드백은 패킷 손실의 유형 전송률 등의 수신 상황을 송신측이 인지하도록 하여 앞으로 발생될 네트워크 상황을 예측하고 이를 바탕으로 전송률과 적응형 FEC 코드 조합을 적응적으로 제어함으로써 신뢰성 있는 전송을 가능하게 한다. 제안된 시스템의 성능을 측정하기 위한 고속 네트워크에서의 전송 실험은 수백 Mbps의 전송 속도를 보이며 향상된 신뢰성을 보여준다


Key Words : Adaptive FEC, rate control, high speed, reliable media transport, estimation of network condition.


## ABSTRACT

In this paper, we propose a reliable high-speed UDP-based media transport with an adaptive error control. The proposed adaptive transport scheme controls the amount of redundancy by monitoring the network in order to adapt to network fluctuations efficiently. The feedback of receiver enables the sender to be aware of current reception status (i.e., rate and type of packet loss) and to estimate the expected network status. Based on this, the proposed transport attempts to enable reliable transport by adaptively controlling the amount of both whole sending rate and the ratio for adaptive FEC code. Experiment with high-speed network has been conducted to verify the performance of the proposed system that demonstrates the enhanced reliability of the proposed transport at the speed of up to several hundred Mbps.


## Ⅰ. Introduction

To provide a reliable transport service for massive continuous media over high-speed networks, packet-level FEC has been studied to overcome the randomized bursty packet loss, i.e., packet erasure. Together with retransmission schemes that trade the bandwidth efficiency with additional delay, the FEC-based transport schemes are known to be effective, especially when the latency is limited. However, when deployed in open loop by simply fixing the redundancy level for the given estimated loss rate, FEC-based schemes will end up with wasting precious network bandwidth. It is thus very important to be able to react to the network fluctuation by adaptively controlling the

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amount of redundancy. To cope with packet losses as well as to avoid bandwidth waste, several adaptive FEC-based transports have been introduced to control the amount of redundancy adequately based on the monitored network feedback [1][2]. However, most existing ideas for proactive packet-level FEC adaptation have been limited to low to medium speed ranges (e.g., up to several Mbps) and little attention has been paid to the issue of high-speed transport at the speed of up to Gbps. However, with the advent of optical-based high-speed networks, various efforts on the transport and application layers have been made over the years to improve the throughput limitation caused by the congestion control. From the protocol side, there are XCP, Striped TCP, and HS-TCP [3, 4]. HS-TCP, Scalable TCP modify the congestion response function to allow high bandwidth product flows through the rapid probing of available bandwidth. XCP achieves maximum link utilization separating the efficiency and fairness policies of congestion control. Note that some of them require the modification of network infrastructure and it is not expected deployed in near future. From the application layer, SABUL and Tsunami are rate-based controls combining UDP transport and TCP control [5][6]. These works, however, focus only on retransmission-based reliability control such that the receiver requests the retransmission immediately when it detects the loss and sender run independent thread for sending data and control of the feedback. However, its target is large file transfer, as in the case of interactive continuous media streaming, they may not be best solution in sustaining packet loss rate below the user- specified target with only small-size receiver-side buffering. Also, when there exist persistent packet losses, the proactive error control will be natural choice.

In this paper, we propose an adaptive FEC-based UDP transport/control with network monitoring and reliability adaptation, targeting the reliable real-time delivery of immersive massive continuous media. Based on the feedback of network monitoring (i.e., loss rate and loss burstiness), the proposed adaptive control guides the reliability of media streaming. It controls the amount of redundancy in face of network fluctuations while considering the limit on the sending rate (i.e., like the available bandwidth). More specifically, the proposed scheme includes 1) the estimation of network condition based on the receiver feedback of network monitoring, 2) the rate adaptation about the available bandwidth considering current network status estimate, and 3) the decision to the proposed FEC code ratio to be applied. Focusing on the impact of high-speed aspects to the proposed FEC adaptation, the performance of the proposed transport is evaluated over a high performance networking testbed with network emulation and real-time video delivery. The outline of the paper is as follows. After discussing the details of the proposed transport control in Section 2, we discuss its performance from evaluation over a networking testbed in Section 3. Finally, we wrap up with the conclusion of Section 4.

## Ⅱ. PROPOSED ADAPTIVE FEC TRANSPORT CONTROL

The best effort Internet does not guarantee the sufficient QoS. Thus, successful media transport requires the coordinated networking support guaranteeing the factor related to the QoS - delay, loss and jitter. That is, the transport system has to adjust the sending rate, applying various error controls adaptively responding to the dynamic fluctuation of underlying networks. In this paper, we are concerned with transport of massive media data reliably with high performance in real-time. To satisfy the user specified QoS, we have diverse requirements as following.

**Time constrained media transport**: The data to transmit is interactive continuous media with the limit of time to deliver. Thus the transport should be aware of the characteristics of media.

**High performance transport**: The network we target is optical-based high-speed network capable of bandwidth speeds of 100M ~ 10Gbps. As of today, it is believed that high-speed transport of Gbps range is only possible under a high-performance networking environment where the QoS (quality of service) is provided to certain level [7][8]. That is, certain level of guarantee is required to realize transport approaching Gbps range. So much attention has to be paid to the issue of high performance transport with efficient utilization of bandwidth.

**Adaptivity for network fluctuation**: To provide reliability to overcome the factors affecting the quality of transmission, the transport should adapt the dynamic fluctuation of underlying network. While minimizing the probability of loss occurrence with rate control, we have to adopt the end-to-end error control according to the network state.

Thus, to satisfy these requirements, the proposed adaptive transport adjusts the redundancy of UDP packets based on the control packets. As depicted in Fig.1, it is divided into three major components: the monitoring and estimation module for network condition, the rate adaptation module, and the adaptive FEC decision module. First, the network status is monitored to capture the packet loss rate, loss type, and others. For this, by calculating the standard deviation within a suitable adaptation window, we grasp the current network status. For the rate adaptation module, the total sending rate is controlled to meet desired quality level while avoiding bulky loss and delay variation, which are mainly linked with the estimated available bandwidth. The FEC decision module makes tradeoff between error resilience and coding efficiency according to the fluctuation of current network. Special attention is also paid to mitigate the bandwidth overhead and the FEC encoding/decoding complexity that are undesirable for the high-speed transport.

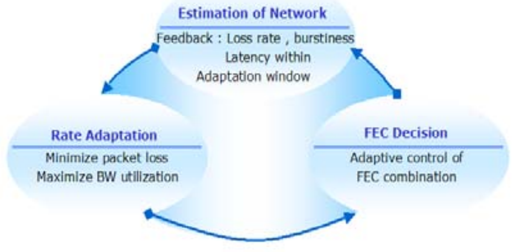



Fig 1. Three main component modules of the proposed adaptive transport.

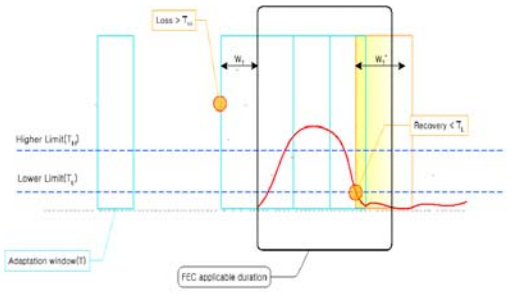



Fig 2. Network estimation with the adaptation windows.

### 2.1 Network Monitoring and Estimation

To capture the status of underlying networks, we monitor at the receiver such parameters as loss rate $L(t,T)$, and burst length $B(t,T)$[1]. Here, $t$ stands for the sampling instant and $T$ means the period of feedback. Based on the regular feedbacks, the sender applies an adaptation window with duration $WT$ (set to multiples of $T$), as shown in Fig. 2, to estimate the network status smoothly. Also, we specify two fixed thresholds - *lower limit*($TL$) and *higher limit*($TH$) thresholds for packet loss. At first, when the loss begins to exceed $TH$[2], we enable the network estimation process at the sender. Then we calculate the standard deviation of loss $\sigma(L(t, WT))$ and burst length $\sigma(B(t, WT))$ within the adaptation window. Based on these, we distinguish the network status into three cases - stable, steady random loss, and burst loss cases as depicted in Fig. 3. Note that $\sigma LH$ is the threshold for random loss standard deviation, $\sigma LB$ is the threshold of burst loss standard deviation, and $\sigma BH$ is the threshold for burst length.

1) The burst length denotes the number of consecutively lost packets.
2) For the sake of simplicity, the sender doesn't apply the adaptation window when the current loss is maintained lower than the higher threshold.

Finally, let us comment more on the adjustment of adaptation window shown in Fig. 2. If some level of losses is persisted during the adaptation window, the FEC scheme can be effectively applied to combat these persistent losses. When the FEC is applied, recovery rate starts to increase while loss rate is decreasing. Similarly, the recovery rate is decreasing when the loss terminates. Thus, when the recovery rate meets *TL*, we apply longer window *WT*'(>*WT*) to allow time to converge. Note that, like this, the proposed adaptation window at the sender helps us to check the persistence of network status change and avoid self-induced fluctuation resulting from oscillatory FEC adaptation[3].

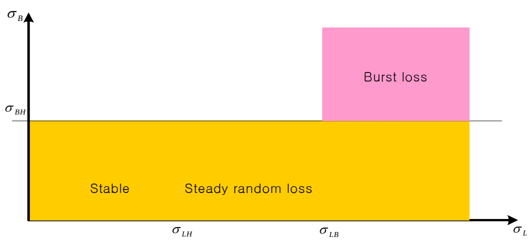



Fig 3. Three network states employed to differentiate adaptations.

### 2.2 Rate Adaptation

In designing the high-performance transport, it is also very important to consider the rate control principle. The transport has to adjust the sending rate by discovering the available bandwidth so that we can minimize the overall packet loss by avoiding the network congestion[4] . However, with the adaptive FEC, we are adding redundancy to the flow within the limit of available bandwidth. The consequence of this redundancy increase means that some portion of data should be omitted from the transport so that we can effectively reduce the required bandwidth. For the video case, increased redundancy can reduce residual packet loss at the cost of video quality. Especially at high-rate loss with bursts, we should reduce the total sending rate while increasing the protection power at the same time.

Thus we adopt a rate-based gradual adjustment of sending rate to minimize the effect of fluctuation. That is, switching of rate does not happen instantly even though the distinct change of loss condition is predicted. The selected sending rate *(R(t))*, which is calculated by dividing the total amount of data with the whole time period spent, should be in the range of *Bmin* ~ *Bavailable*, where *Bavailable* is available bandwidth[5] and *Bmin* is a minimum guarantee bandwidth that is highly dependent on the type and bandwidth flexibility (i.e., adaptivity) of target real-time applications. In addition, the change of sending rate is related to the loss rate of underlying networks. Thus, measured loss rate after $\tau$ from time $t$ can be expressed as a function of *R(t)* as described in Eq. (1).

$$L(t+\tau) = f(R(t)), \quad Bmin < R(t) < Bavailable. \tag{1}$$

The main role of function *f(R(t))* is not only to guarantee the loss rate under user-specified level but also to maximize the bandwidth utilization. Thus, if there exists steady random losses, *R(t)* is decreased reflecting the difference of current loss and the target limit(*TH*). On the other hand, when the loss is decreased, *R(t)* is recovered linearly back to the *Bavailable* in order to maximize the bandwidth utilization. However if *R(t)* gets close to the bandwidth limit, the loss rate is oscillating around *TH* and can cause the unnecessary variations of sending rate. To solve this, *R(t)* has to be increased only up to *Bavailable* − △. Also, to adapt to the network state in a gradual manner, we use *L(t, WT )* and *B(t, WT )*. The aug-

3) Note, however, that current version of network monitoring is still lacking in catching up with the rapid variations in the high-speed networks. In this work, we are only trying to explore the possibility of simplified monitoring as the base of adaptation.

4) Note that this issue is heavily related to the TCP-friendly congestion control. However, in this paper, we are barely touching this issue for the sake of simplicity.

5) Note that, according to the popular equation-based TCP-friendly congestion controls, the available bandwidth is usually determined with loss rate, RTT, and MTU size. However, at current stage, $B_{available}$ is controlled according to the loss rate only in this paper.

mented rate adaptation algorithm[6] is given by Eq. (2).

$$f(R(t)) = \begin{cases} R(t) - \alpha(L(t,W_T) - T_H), & \text{if } L(t,W_T) > T_H \\ R(t){+}{+}, & \text{if } R(t) < B_{available} - \Delta \ \& \ L(t,W_T) < T_H \end{cases} \quad (2)$$

### 2.3 FEC Decision

Given the constraint of a sending rate, the choice on the FEC can be made by attempting to optimize both reliability and bandwidth utilization. Based on the network status, different procedures are applied as depicted Fig. 4. In case of random loss, the amount of redundancy should be able to cover the loss burst. That is, the number of redundant packets $h(=n-k)$[7] has to be greater than $B(t,T)$. Assuming to find $k$ with given $h$, the efficiency of bandwidth, thus, depends on $k$. That is, as the increase of $k$ when $h$ is fixed, the bandwidth efficiency is also increased. However, increase in $k$ causes the reduction of protection ability as well as the increase of processing overhead. The processing overhead can also cause the throughput deterioration if it starts to affect the sending rate. The processing throughput is calculated by

$$Th_{proc} = \frac{Amount\ of\ data\ for\ n\ packets}{((C_s + C_m)\times n + (C_{e1}\times k + C_{e2}\times h))}. \quad (3)$$

where $Cs$ is sending cost in time that is related to the packet size and the packet scheduling, $Cm$ is the cost of memory copy of massive data, and $Ce$ is the encoding overhead that changes with $n$ and $k$.

Meanwhile, in the context of packet-level FEC, the size of all packets including parity packets is fixed to equal size based on the MTU(maximum transfer unit) of the underlying network(i.e., around 1500bytes for most practical situation). With this equal packet-size assumption, protection efficiency for bandwidth is normalized by $k/n(\%)$. As an example, if 1 parity packet is generated over 3 packets (i.e., $(n,k)$ = (4,3)), protection efficiency is 75%. For the efficient utilization of bandwidth, the choice of FEC combination has to be determined to maximize the protection efficiency only under the condition that it satisfies user specified quality level. Therefore, by keeping that available FEC processing capability, denoted as *Thproc*, is above the sending rate and current loss is under *TH*, we increase $k$ linearly.

For the burst loss case that causes rapid decrease in total sending rate, $h$ needs to be increased while $k$ is decreasing to minimize the loss burstiness. When the FEC combination change is decided, the information $(n,k)$ that is included the FEC header affects the buffer size and other processes at the receiver in order to apply newly set FEC combination.

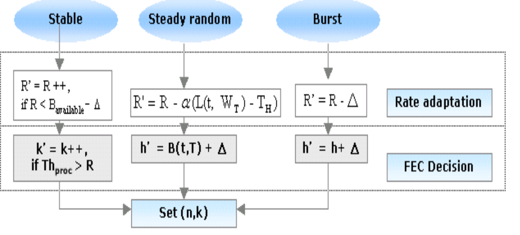



Fig 4. Proposed adaptive control reactions (rate adaptation & FEC decision) based on the type of loss.

## Ⅲ. EXPERIMENTS and RESULTS

### 3.1 Adaptive Transport System and Test Condition

In Fig. 5, the building blocks of proposed reliable high-speed transport system are depicted. At the sender side, source data are grouped into a block of $k$ packets and varying number of parity packets are attached by the FEC encoder. To enable high-speed transport, we exploit two independent threads for sending and adaptive control. From the feedback, we estimate the network condition using the statistics calculated from feedback values in the adaptation window. Then,

6) This kind of aggressive rate adjustment should be improved further in order to provide friendliness to TCP traffics.
7) In general, we put the FEC combination (n,k) - (number of total and source packets in a block, respectively) [11].

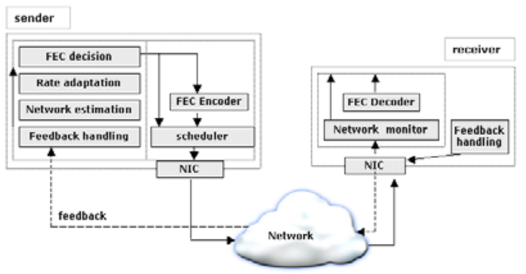



Fig 5. Prototype architecture of the proposed adaptive transport system.

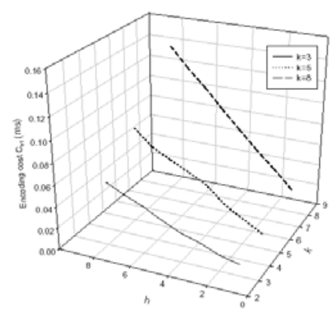
according to the each condition, different strategies are applied at rate adaptation and FEC decision part. The receiver checks whether there are lost source packets by performing gap-based loss detection. Through this monitoring, the receiver sends back to the sender the required feedback. If source packets are lost, it waits until sufficient number (at least $k$) of packets arrive and then forwards them to the FEC decoder for reconstruction.

The prototype of high-speed transport system is implemented using high-performance PC's. Dell Workstation with dual Xeon™ 1.8GHz CPU, 1G memory, 915Mbps system bandwidth and Dell Workstation with Xeon™ 2.4GHz CPU, 1G memory, and 955Mbps system bandwidth are used. In addition, the performance of the proposed transport is evaluated by transmitting over the real-world Internet. The WAN path of KOREN/KREONET between GIST (Gwangju) and KISTI (Daejeon) includes three hops in each direction and has an RTT of approximately 3.2ms. The video is also transmitted over Gbit LAN testbed with network emulator[12]. The channel model we emulate in this paper is a burst loss model called as Gilbert-Elliot model, where two-state Markov chain is used to switch between "active" and "idle" states. If a packet is transferred at time $t$, then no loss occurs at active state. On the other hand, the packet is lost in idle state. Thus, using the emulator, we can approximate congestion situation over high performance network by controlling loss probability and its burst length.

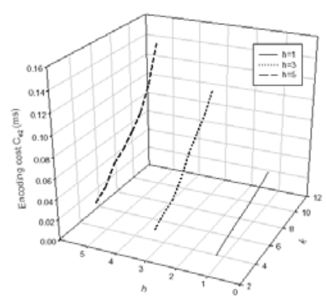
### 3.2 Experiment Results.

#### 3.2.1 Processing overhead

Transmission with FEC faces the various processing costs. Fig. 6 shows the encoding cost ($Ce1$, $Ce2$) relating to the $k$ and $h$. These are compared with respect to source and parity packet combinations. We can see that the change of $h$ has more cost than the change of $k$. And then, by fixing the number of parity packet to one, we can observe the impact of sending rate on the processing overhead caused by the FEC. Fig. 7 shows the ratio of the sending cost ($Cs$) to the total cost as percentage. According to the speed up with sending rate, the standard deviation of cost is increasing. Thus the time to spend to send packets is increasing while the minimum interval per packet required by given sending rate is decreasing. Therefore, the ratio of sending cost is proportionally increasing up to the 100% which is the point affecting the throughput. So current implementation has the limitation of throughput around 430Mbps.



(a)



(b)

Fig 6. Encoding cost comparison w.r.t. source ($k$) and parity ($h$). (a) $Ce1$ and (b) $Ce2$.

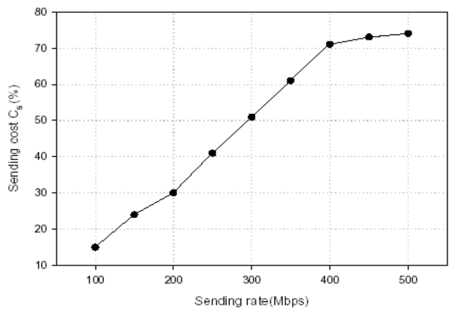
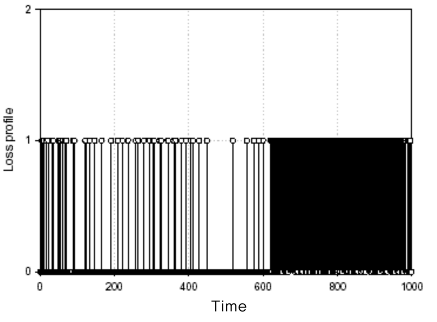


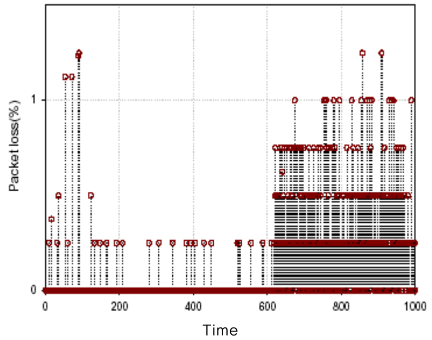

Fig 7. Sending cost (Cs).

#### 3.2.2 Adaptive FEC verification over emulated LAN testbed

Fig. 8 shows a scenario where the relative advantage of adaptive FEC comparing static FEC. Assume that we are given an application that can tolerate a loss rate of $L(t, T)$ = 0.5%. The network was configured as shown in Fig. 8 (a). The emulated loss probability is changed from 1% to 5% after 600$T$ and burst length is fixed to 2. Using a prior information from static FEC experiment, the redundancy of static FEC at $h$ = 1 would suffice to achieve the desired loss rate of 0.5%. Fig. 8 (b) is $L(t, T)$ of static FEC with combination of $(n, k)$ = (4,3) measured during 1000$T$. The loss rate is maintained below 0.5% during time interval [0,600] when one redundant packet can recover the loss with burst length 1. However, the loss rate starts to fluctuate with increasing the gap of loss occurrence. Therefore, it fails to guarantee the desired loss rate. We can see that it acts the weakness of applying FEC statically over the real Internet with dynamic perturbations of loss. Fig. 8 (c) shows the trace of adaptive FEC run for the same configuration as above. It shows improvement after change of emulated loss. The overall loss rate converge below 0.5% due to the adaptive change of $h$. In this experiment, $h$ is changed to 2 from 1 during the interval of 5% loss insertion. Thus, it shows that adaptive FEC is able to achieve desired QoS minimizing the effect of network fluctuation. The reason of frequent loss occurrence at initial time [0,100] of both case, relies on the time of applying FEC with start of application.




(a)

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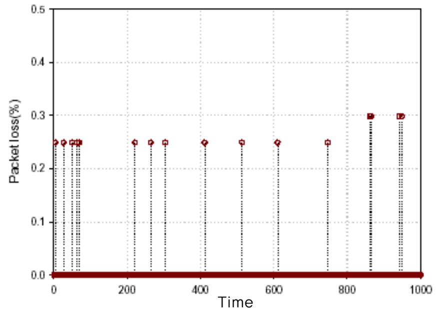


(c)

Fig 8. Static vs. adaptive FEC performance: (a) loss profile, (b) static FEC, and (c) adaptive FEC.

#### 3.2.3 Adaptive transport over WAN testbed

**Transport over WAN**: Fig. 9 shows the burst length as a function of time that is measured between GIST and KISTI. It is measured at daytime with the period of [0,1000000] sec. Most of loss periods involve from one to five packets and it is confirmed by looking at the frequency distribution in Fig. 9(b), which shows the number of loss occurrences. The burst length above 10 packets is randomly occurred. The slope of the distribution decreases from the origin. We have done several experiments by increasing the sending rate over the

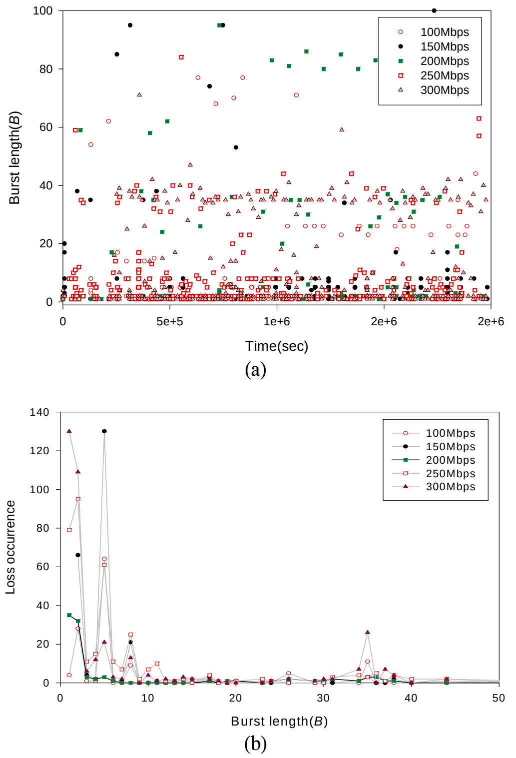


Fig 9. Loss profile over WAN test. (a) Burst length and (b) Loss occurrence.

the same connection. The frequency of loss occurrences is increasing with the increase in the sending rate, and all cases have similar distribution as depicted above.

**Rate adaptation with adaptive FEC**: Rate adaptation with FEC depicted in Fig. 10 shows the variation in the sending rate and feedback parameter during time interval [1800, 2200] time unit (*WT*), which is interval of adaptation window. Depicted parameters are measured after classifying the network states. In this experiment, we set *WT* as 1 sec, bandwidth offset Δ as 30Mbps, and reduction factor is set to 1. To simulate network loads, we insert background traffic of 100Mbps at 1925 *WT*. Initially, we start with the FEC combination (5,5) and assume *TH* is set to 1%. Until 1925 *WT*, the sending rate $(R(t))$ is maintained at 400Mbps with small residual losses. Note that, to recover from these losses, *h* is set to 1. After inserting the background traffic, increased $L(t,\ WT)$ is decreasing if we reduce the total sending rate. The degree of reduction follows the difference amount of $L(t,\ WT)$ and *TH*. As the result of reconstruction, the recovery rate is also increasing. When the sending rate is saturated to about 295Mbps, $L(t,\ WT)$ gets converged below 1% with less fluctuation. Like this, the FEC is applied when the loss is first detected as significant and the redundancy is controlled as depicted in Fig. 11. Over the time, the redundancy of FEC is adjusted automatically by reflecting the feedback parameters.

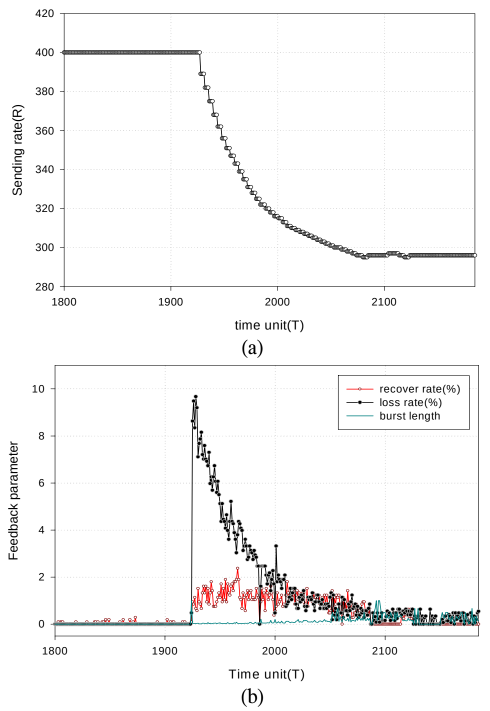


Fig 10. Adaptive transport performance: (a) Sending rate and (b) Loss rate.

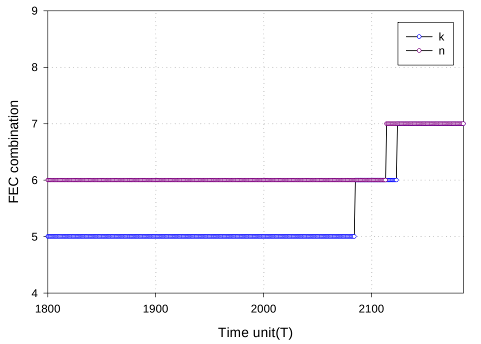


Fig 11. Variations in the FEC combination $(n,k)$ over time.

## Ⅳ. Conclusion

We have presented a reliable high-speed UDP-based transport and its adaptive FEC control. Based on the feedback, the adaptive FEC control can improve the quality of delay-constrained immersive media streaming while mitigating the impact of network fluctuation and system limitations over the high-speed networks. We are currently undergoing several refinements to address the limitations discussed in the current version of paper.

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