LOSS CONTROL THROUGH THE COMBINATION OF BUFFER MANAGEMENT AND PACKET SCHEDULING

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Abstract: Conventional Quality of Service (QoS) control techniques are designed for achieving network-level QoS objectives. Due to the large differences between network-level and application-level QoS properties, these techniques cannot provide desirable QoS for video users. Previous work has been conducted to design a packet scheduling approach where application requirements and network-level QoS objectives are addressed simultaneously. In this paper, the packet scheduling approach is integrated with a buffer management technique for increasing the numbers of video users with QoS satisfaction. The effectiveness of the proposed technique is demonstrated through simulations.

1 INTRODUCTION

Quality of Service (QoS) can be achieved through the use of packet scheduling techniques. Majority of the packet scheduling methods has been designed for allocation of a minimum bandwidth to each flow that crosses a link and provision of throughput and a delay bound [Zhang, 1995]. For example, Weighted Fair Queuing (WFQ) [Parekh and Gallager, 1993] assigns a weight to each flow. The weight logically specifies how many bits to transmit each time the router services that particular flow; this controls the percentage of the link's capacity for that flow. The weight depends on the request rate of each flow and how many other flows are sharing the link. Each request rate is assured if the sum of all the request rates is smaller than the link capacity; otherwise, the allocated rate is the ratio of the request rate of the flow to the sum of the request rate of the backlogged flow at that instant. Another representative scheduling mechanism is Class-based Queuing (CBQ) [Floyd and Jacobson, 1995]. CBQ is a hierarchical link-sharing mechanism. It partitions network bandwidth among the different traffic classes, in which a higher percentage of bandwidth is allocated to the important traffic class.

These scheduling mechanisms are very efficient from a network perspective. However, they are inadequate from the viewpoints of application users, such as video users. The reason is that they are not designed to provide application-level QoS but QoS from the standpoint of a network. However, for video applications, there are performance gap between application-level QoS and network-level QoS. Specifically, there is no linear relationship between visual quality and bit-rate. Some bits may be more important than others. In other words, perceived video quality will generally be dependent on the data rate and the data content. Current packet scheduling techniques do not use content information in the data; rather, they treat all the data in the same manner. Another common problem with existing scheduling techniques is that they consider dropping data that misses the deadline and do not consider data loss due to buffer overflow [Dovrolis and et.al.]. To overcome these limitations, we proposed a new packet scheduling scheme, called MPAPS [Bai and Ito, 2003]. The idea of MPAPS is to make enough available buffer space before incoming packets arrive by appropriately scheduling queued packets to exit the router. Thus, incoming packets will be admitted into the router and the packet loss due to buffer overflow will be reduced. Also, MPAPS considers the characteristics of video data and users' requirements, as well as network QoS parameters in the control of service time and

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service order of the queued packets. Consequently, the perceived QoS of video users is improved.

In MPAPS, when the input buffer at a router is full, arriving packets are dropped. Thus, it is likely to introduce arbitrary loss distribution between videos and different parts of a video. As a result, the numbers of video users with QoS satisfaction is reduced because the locations at which the losses occur can have a significant effect on the QoS perceived by the user. For example, a loss of several consecutive packets in a frame may be imperceptible to the user whereas the same loss rate distributed over different video frames can largely degrade visual quality [Ito and Bai, 2002]. Therefore, an Enhanced MPAPS (E-MPAPS) is needed. In the E-MPAPS, a new buffer management at the input buffer is integrated to the original MPAPS in order to control loss distribution at the input buffer.

This paper investigates the performance of E-MPAPS scheme. Section 2 describes the scheme. Section 3 presents the simulation results, and Section 4 concludes the paper.

2 SCHEME DESCRIPTION

E-MPAPS targets MPEG video due to the abundance of existing videos of MPEG format. It is composed of packet scheduling and buffer management (Figure 1).

The buffer management at the input buffer of a router is designed to maintain high number of videos with QoS satisfaction. Therefore, it accomplishes the following two criteria:

1) Selection of which video to be rejected when congestion occurs such that every video achieves a loss performance at a level commensurate with its individual expectations. It other words, no overservicing or underservicing of a particular video stream occurs. Thus, network utilization is maximized.

2) Determination of how much data from the selected streams should be discarded during the periods of congestion in order to maximize the video quality in the presence of packet loss.

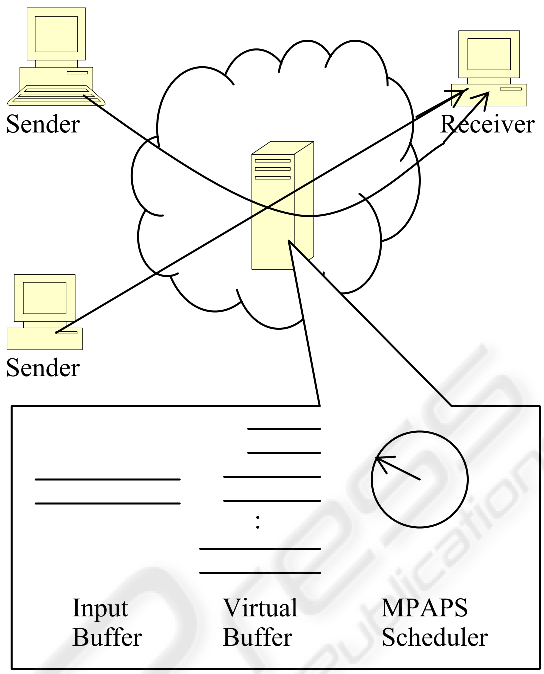

Figure 1: The E-MPAPS Scheme

The pseudo code of buffer management is listed in Figure 2.

```
/*
LOW:  the buffer length threshold at
which B-packets start being dropped.
HIGH:  the buffer length threshold at
which P-packets start being dropped.
Len: the buffer length.
Size: the buffer size.
E-frame: a partially discarded frame.
PLRi: the acceptable packet loss ratio
of video stream i.
Wi: weight parameter for video stream
i, it is set inversely proportional to
its packet loss tolerance.
Δ: the difference between initial and
update values of LOW.
*/

if (packet == E-frame)
   drop();
if (Len == Size)
   drop();
else if (Len > HIGH ){
         if((packet == first P-packet )
|| (packet == B-packet)){
              drop();
           }
         else
              accept();
}else if (Len > LOW){
```

```
        if ((packet == B-packet) ||
(Len > Wi*Size ))
            drop();
        else
            accept();
}else   accept();

/* In the drop () procedure, LOW
increases by one when an I- or P-packet
gets accepted subject to Δ remaining
greater than zero. */
```

Figure 2:.Pseudo Code of Buffer Management at Router Input Buffer

Packet Scheduling is based on the original MPAPS scheme. It is designed for per-flow queue router structure. The E-MPAPS scheme thus constructs virtual buffers in order to use MPAPS with the above buffer management method. Each virtual buffer holds a video stream. The virtual buffer size (B*v*) is computed according to the following equation.

$$Bv = W_i * Size \quad (1)$$

Due to the use of virtual buffer E-MPAPS has one distinguished feature: it does not provide strict isolation between different videos and video sources could use the buffer reserved to others when load level at the router is low.

MPAPS first maps the videos into the groups with different transmission priorities based on their upcoming packet type and current loss performance (SL), and then a specific transmission schedule for a video stream in a selected group is adaptively set to respond instantaneously to needs based on the Adaptive Priority Index (API). Here, the SL is defined as ratio of the actual packet loss and the maximum allowable packet loss, and the API is defined as the product of SL and the normalized length of an virtual input buffer holding stream i. The rationale behind MPAPS is that the drop probability of I packets will be lower than that of P packets, which in turn, will be less than that of B packets. Moreover, the videos that have a worse loss performance than expected receive expedited and more servicing, whereas videos that have satisfactory or even better loss performance receive slow and less servicing. Therefore, all the videos will be transmitted with more acceptable loss targets. Further details can be referred in [Bai and Ito,2003].

3 RESULTS AND DISCUSSION

To demonstrate the advantages of E-MPAPS, we compare its performance, including packet loss rate and I-frame error rate with that of the original MPAPS, and first-come-first-serve scheduling (FCFS) schemes. FCFS is widely used in the current Internet routers due to its simple implementation. In FCFS, the incoming packets are accepted in order of arrivals. The simulation details are presented in the following.

Sources: real MPEG-1 video traces where the number of bits per frame used by the MPEG coder is described [http://www3.informatik.uni-wuerzburg.de/MPEG/].

Parameters used in the simulation:

1) Fixed:

- Packet Size: 1500 bytes or less
- Simulation Time: 40 minutes
- Video Starting Interval: 60 seconds
- Output Link: 100 Mbps
- *Size*: 150KB
- *LOW*: 0.90 [a threshold value of 0.90 means that the buffer length threshold is 90% of the buffer size (in packets)]
- *HIGH*: 0.95
- *PLRi:* 3% for first half of videos and 6% for the others

2) Variable:

- The number of background sources varies in order to change the load level at the router.
- The starting sequence of a video stream was randomly selected in each run. The results presented in this section show the final values of the average of different runs.
- The test-scenarios featured varying degrees of load level and various traffic patterns. The presented test results here are an illustrative comparison of the differences of the three schemes.

Table 1 and Figures 3 and 4 show the results obtained for different number of transmitted videos, namely, 10, 20 and 30 videos.

We see in Table 1 that E-MPAPS produces the lowest number of videos whose actual packet loss rate is greater than their maximum allowable packet loss rate for all the cases.

Looking at the standard deviation of packet loss difference for each of the three cases, we find that the smallest value appears in the E-MPAPS scheme. It indicates that the loss differences fluctuate slightly between videos when E-MPAPS is applied. In other words, for many videos, the actual loss of each stream is controlled to nearly match the specified allowable values. Conversely, relative high and too much loss variation from expected packet loss rate appears in MPAPS and FCFS. This gives the reason why E-MPAPS decreases the number of videos with packet loss beyond the expected values.

The above results can be explained as follows. FCFS does not have any loss distribution control mechanism. While MPAPS adaptively adjusts the probability of transmissions that individual streams receive: those stream queues that are in danger of a violation of loss requirements receive servicing sooner and more frequently. Conversely, those stream queues that have a lower loss than expected receive servicing that is delayed and less frequent. When streams are delivered to a network node quickly enough to make MPAPS impossible to adjust the servicing sequence and frequency timely, unexpected packet loss would occur. It results in an inequitable loss distribution between videos.

E-MPAPS, except the adaptation mechanism done by MPAPS, the buffer management allocates buffer occupancy between videos in a fair manner: buffer occupancy is inversely proportional to their loss constraints. Therefore, the losses distribution among the streams is further enforced, just matching their individual loss tolerance.

Table 1: Comparison of Packet Loss

Number of videos		**10**	**20**	**30**
Number of violating stream[1]	E-MPAPS	4	6	6
	MPAPS	6	9	12
	FCFS	7	11	18
Standard Deviation of Δ^2	E-MPAPS	0.00091	0.00085	0.00170
	MPAPS	0.00102	0.00245	0.00241
	FCFS	0.00614	0.00787	0.00771

[1]Violating stream: actual packet loss rate is greater than their maximum allowable packet loss rate.

[2]Δ: the difference between achieved packet loss and maximum allowable packet loss for each individual video.

Figures 3 to 4 plot I-frame error rate for the three schemes when the number of transmitted video (N) is 10, 20 and 30, respectively. I-frame is defined as an error frame if one packet in the frame is lost. From the figures, we obtain the following. For all the three cases, the I-frame error rate in E-MPAPS is no more than 0.3%, while around 2-3% in MPAPS. In FCFS, I-frame error rate is largely increased, approximately 20% with N=30.

This can be explained as follows. E-MPAPS includes buffer management mechanism. It detects a congestion condition by observing when the router's input buffer occupancy is close to crossing, or has crossed, a specified threshold and determines how best to allocate the available buffer to reduce the I-packet loss. Instead, MPAPS discards packets arbitrarily, therefore, most likely distributing packet loss over all frame types during the congestion episode. In particular, a lost packet could belong to an I-frame.

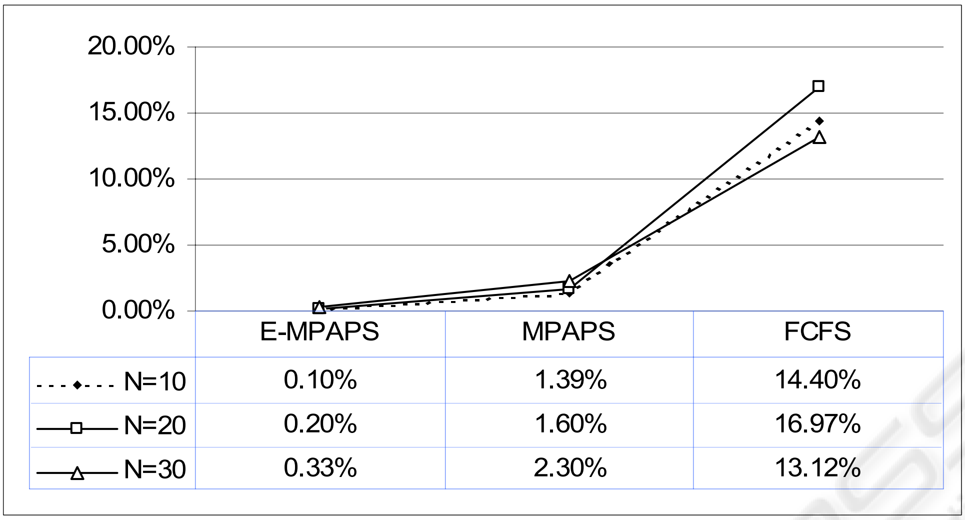

Figure 3: I-frame Error Rate for the First Half of Videos

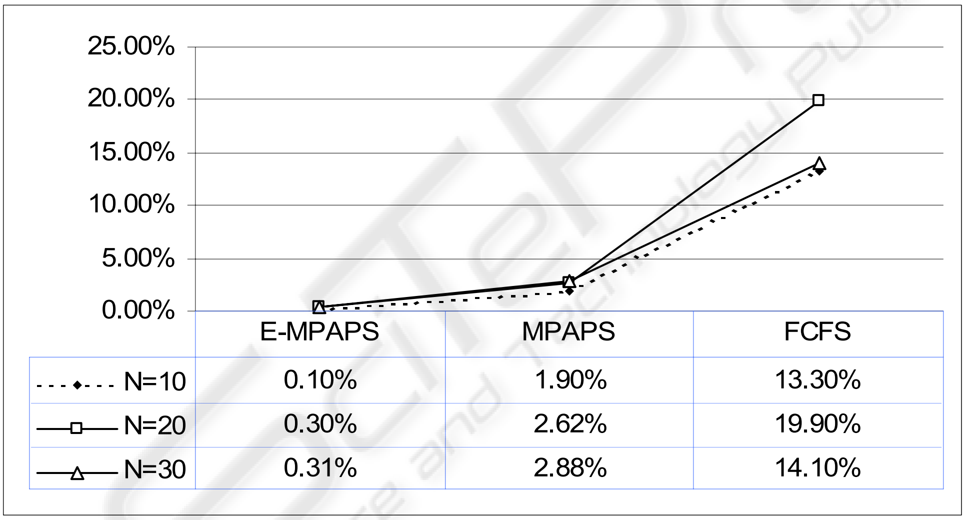

Figure 4: I-frame Error Rate for the Second Half of Videos

Adding buffer management in E-MPAPS largely decreases I-frame error rate, giving a great increase in the visual quality of video. It suggests that the number of video users with QoS satisfaction is increased. The reason is that the I-frame error has significant adverse effects on the perceived video quality. Each video frame is very large in size, and is thus segmented into a sequence of IP packets when delivered through an IP network. The loss of a packet may cause the errors in a video frame. Furthermore, MPEG video possesses a frame independent nature. The I-frame is coded independently. The P-frame and B-frame are coded by using the closest past I- or P-frame, and the closest past and future I- or P-frames, respectively. Therefore, the loss of an I-packet distorts the whole GOP, which is equivalent to affecting half a second of video.

4 CONCLUSION

In this paper, a modification of the MPAPS scheme (E-MPAPS) is proposed. E-MPAPS introduces an application-aware buffer management in MPAPS at the input buffer of a router. It improves the loss distribution between different parts of a video stream and the contending videos. Simulation results have shown that E-MPAPS significantly improves the number of videos with high visual quality over MPAPS and the conventional packet scheduling method, without the use of data content. E-MPAPS, however, has a relatively high computational complexity due to the use of application requirements in the buffer management.

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