A User-Centered Approach to Enhance QoS for Networked Video

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Abstract

Buffer management plays an important role for enhancing Quality of Service (QoS) for video streaming over IP networks. However, most existing buffer management techniques have been developed according to network’s viewpoints; consequently, QoS requirements from users’ perspectives are not well satisfied. This paper proposes a novel buffer management scheme based on user’s expectations. Simulation results that the advantages of the scheme in terms of video quality, service fairness and network efficiency.

I. Introduction

Buffer management plays an important role in enhancing Quality of Service (QoS) for video streaming over IP networks. It adjusts buffer occupancy to prevent congestion at routers, thus decreasing packet loss and improving video quality. The most common buffer management technique is Tail Drop (DT) in which packets are dropped when the buffer is full. Recent research have proposed proactive discard approaches to ensure the queues will not actually reach their full discard thresholds. A representative technique is Random Early Detection (RED)[2]. In RED, an arriving packet is randomly discarded with a probability proportional to the average queue length of a router when a preset threshold is exceeded. RED is not a stand-alone mechanism and relies on joint use of rate control techniques in order to reduce the packet loss for networked video [3, 5]. Furthermore, neither DT nor RED use video-related information in making packet discard decisions. They manage issues such as packet loss rate and congestion. However, loss distribution, which has a significant impact on video quality, cannot be effectively controlled.

Our previous paper [4] proposed an alternative buffer management approach, called FDDT, which focuses on improving user-expected video quality, rather than just aiming at reducing packet loss. FDDT uses three control thresholds in a router output buffer to always ensure enough free space for queuing the more important video packets and the entire set of packets of each accepted video frame. This scheme prevents error propagation and minimizes the number of frames in error. As a result, degradation of perceived video quality during congestion episodes is reduced and network efficiency is improved. However, FDDT does not have a mechanism to ensure that the packet loss ratio of each video is within its acceptable range. Aggressive video streams may transmit too many packets, leading to a lower packet loss than expected, and vice versa; therefore, it is not fair. A new scheme, called FairFDDT overcomes this limitation. It provides selective treatment not only for different parts of video traffic but also for different video streams.

The rest of this paper is organized as follows: Section II describes the fairness criteria and the design of FairFDDT. Section III evaluates FairFDDT. Finally, Section IV gives conclusions.

II. The FairFDDT scheme

This section presents the FairFDDT scheme for a router that handles streaming MPEG video. MPEG video was chosen due to the abundance of existing videos of MPEG format. The objectives of the FairFDDT scheme are to assure high quality video streaming in the presence of loss and to provide fairness among multiple video streams. The classical notion of fairness refers to bandwidth allocation fairness and buffer allocation fairness [6, 7]. The fairness measure used in this paper is based on the concept of service fairness. Service fairness means fair allocation of service levels to different traffic such that all traffic receives service at a level commensurate with their individual expectations. In other words, service fairness tries to guarantee that all traffic with
the same quality requirements perceives similar QoS; traffic with higher QoS requests obtains more quality, while traffic with lower QoS requests receives lower quality. Clearly, service fairness is a more rational objective since it reflects user perceived service quality that a network endeavors to offer, while fairness in resource allocation is just a means of achieving a network service goal. Specifically, the definition of fairness here refers to a fair loss distribution among contending video streams such that video streams exhibit different loss ratios matched to the loss tolerance.

In considering the dual objectives of different parts of the same video stream as well as for different video streams, the FairFDDT scheme carries out buffer occupancy control among video streams, in addition to managing buffer allocation between different parts of data within the same video stream as performed by FDDT. The new contribution of FairFDDT is presented as follows.

FairFDDT classifies a video stream into different classes based on loss tolerance. It then applies queue-limits to these classes by calculating the "weight" of each class. The queue-limits determine the number of packets of each video that are allowed in the router buffer when the total buffer occupancy exceeds the predefined buffer length threshold. If the number of packets is larger than the queue limit, then the packet is dropped. The weight of each class matches the class criteria. This means that low loss tolerance traffic has priority over high loss tolerance traffic. Specifically, if two classes of videos are considered, where class $r$ has lower packet loss tolerance than class $s$, the weight parameters to each are $\omega_r$ and $\omega_s$, and the numbers of video within each are $n_r$ and $n_s$, therefore a relationship will be defined as:

\[
\omega_r n_r + \omega_s n_s = 1
\]

subject to
\[
\omega_r > \omega_s
\]

The main advantages of the above operation are twofold: firstly, buffer space is allocated depending on loss tolerance, thus providing a loss performance matched to the loss tolerance. Therefore, packet loss among all videos is equally distributed. Moreover, an efficient use of the buffer to serve as many streams as possible can be achieved. Secondly, it is used only when the network is highly congested; otherwise, the buffer is completely shared by all streams. This can lead to high utilization of the buffer and high network throughput. The FairFDDT scheme operates as described below:

For each arriving packet from stream $i$:

/* New contribution of Fair FDDT */

1) If (Buffer Length >LOW) and ($R_i > 1$), drop the packet. Here, LOW is the initial buffer length threshold at which B-packets start being dropped, and $R_i$ is the ratio between the actual buffer occupancy, $Q_i$, of stream $i$, and the allocated buffer share which is given by product of weight $\omega_i$ and the buffer size $B$. Thus,

\[
R_i = \frac{Q_i}{\omega_i B}
\]

where $\omega_i$ is calculated according to equation (1).

Next, do the following:

/* FDDT */

2) If Buffer Length >LOW, and the packet is the first packet of a B-frame, drop the packet.

3) If Buffer Length >HIGH, and the packet is the first P-packet or B-packet, drop the packet. Here, HIGH is the buffer length threshold at which P-packets start being dropped. If the dropped packet is a P-packet, LOW decreases by one. Let the difference between initial and update values of LOW be $\Delta$, LOW increases by one when an I- or P-packet gets accepted subject to $\Delta$ remaining greater than zero.

4) If the arriving packet(s) belongs to a partially discarded frame, drop the packet.

5) If buffer is full, drop the packet.

Next, perform:

/* update related parameters */

6) If the packet is accepted, increase $Q_i$ and Buffer Length each by one, respectively.

7) For each departing packet from stream $i$, decrease $Q_i$ and Buffer Length each by one.

III. Performance evaluation

Extensive simulations were conducted to study the effectiveness of the FairFDDT scheme using a streaming MPEG video application running over simulated IP networks. Simulator is implemented in a C program, and simulates the behavior of an output packet buffer in a router with one of the FairFDDT or FDDT or DT schemes, respectively. In the simulation, real video data from several MPEG traces [8] were used. The scheme was evaluated by new parameters that represent user-perceived QoS. They are Frame Error Rate, Effective Throughput, and Fair Index. The definitions are given below.

- Frame Error Rate (FER) (for each video stream) FER is the fraction of frames in error for each video stream. A "Frame in Error" is defined as follows: if one packet in a frame is lost then this whole frame and its propagated frames (P and or B) are considered to be frames in error. A low frame error rate represents high perceived quality of a video.

- Effective Throughput (ET) (for all video streams) ET is the fraction of usable data over all the video streams. "Useable data" is the video data that belongs to a successfully delivered frame. High effective throughput means high network efficiency.
Fair Index (FI) (for each video stream) FI is the ratio of actual packet loss ratio (PLR) over the acceptable PLR for each video stream. A value of one or less than one indicates satisfactory loss performance and a value of greater than one indicates unsatisfactory loss performance. Therefore, the higher the percentage of streams with the Fair Index equal to or less than one, the higher the level of fairness.

A. Performance of FairFDDT

Figure 1 shows the simulation results of FairFDDT when six videos were used. Videos 1-3 are “Soccer” traces with a packet loss constraint of 3%, and videos 4-6 are “Talk” traces with a packet loss constraint of 6%. LOW and HIGH are set to 0.80 and 0.90, respectively. Here, the weights for videos 1-3 and videos 4-6 are set to 0.20 and 0.13, respectively. As shown in Figure 1 (a) the FairFDDT scheme provides lower FER than the DT scheme. FairFDDT attempts to admit packets belonging to completely correct video frames and discards packets from partially corrupted ones. Moreover, it uses a preventative strategy to discard less important packets before the buffer is full in order to protect more important packets (1 and P-packets) from being discarded. Since the loss of I and P-packets is reduced, there is less error propagation between the frames, further decreasing FER as compared to DT. Conversely, DT discards packets arbitrarily causing more or less even distribution of packet loss, yielding a large FER because each lost packet may belong to a different frame. A lost packet could belong to an I or P-frame, and consequently, a small packet loss would affect a large number of consecutive frames due to error propagation. FER correlates with the users’ perceptions of the video quality, and therefore, DT results in significantly more video quality degradation than FairFDDT.

FairFDDT tries to deliver only correct video frames and thereby produces a higher ET than DT, leading to much better network efficiency (See Figure 1(b)). Also from Figure 1(c), the level of fairness provided by the FairFDDT scheme is much higher than that provided by the DT scheme. It can be seen than most of the time FI in the FairFDDT scheme is less than or closer to one. This is because the FairFDDT scheme sets the weight in the buffer sharing mechanism according to the acceptable PLR. Videos with a higher acceptable PLR receive a lower share of buffer and vice versa. Therefore, when the FairFDDT scheme is applied, less buffer space is allocated to streams 4-6 than streams 1-3. In turn, every stream achieves loss performance at a level commensurate with expectations. This results in an equitable loss distribution among the video streams with different loss tolerances. Thus, losses are arbitrarily distributed among video streams. This is shown by the following two facts. Firstly, the streams with same packet loss tolerance, i.e., streams #4 and #5, exhibit significantly different FI, meaning that unfair service is provided to both streams. Secondly, most values of FI in the DT scheme are greater than one. In general, the DT scheme provides satisfactory service only for a small percentage of the video streams in terms of loss performance. A large fraction of streams cannot meet their loss constraints, and thus, the DT scheme does not provide fairness of service.

Experiments using a number of videos N = 2 to 10 exhibit similar trends as those presented here. DT is inferior to FairFDDT in all the cases. The effects of varying threshold values and weights and changing traffic patterns on FairFDDT performance were also studied. The results show the robustness of the FairFDDT scheme in terms of video quality and network efficiency, regardless of network load and the variety of video, although a slight tuning of the parameters, such as thresholds and weights for network conditions and various videos is necessary. Additional results on the impact of thresholds, traffic patterns, load level and choices of weights on FairFDDT will be presented in a more detailed technical report.

B. Comparison of FairFDDT and FDDT

The comparison of FDDT and FairFDDT in terms of FI is shown in Figure 2. Out of 6 video streams, the acceptable PLR of the first half is 3% and the second half is 6%. Figure 3 shows that both DT and FDDT have a lower level of fair service than the FairFDDT. This is, again, the result of the loss-based buffer sharing mechanism built into the FairFDDT. Recall that in this scheme, the buffer usage among video streams is fairly distributed in accordance to the loss constraints. The losses among the streams are equitably distributed to match their individual loss tolerance. On the other hand, both DT and FDDT lack a loss-aware buffer allocation mechanism for the contending streams. ‘Greedy’ streams may use the buffer space more than the weighted average, forcing other streams to use less buffer space than needed. As a result, the ‘greedy’ streams exhibit much lower loss ratios than they should, while other streams experience higher losses. Therefore, a random loss distribution among video streams occurs in DT and FDDT. However, the improvement on services fairness from FairFDDT comes with a small cost: the FairFDDT needs to know the application requirement and to compute the corresponding buffer share.

IV. Conclusions

This paper proposes a novel buffer management scheme, called FairFDDT. The goal of FairFDDT is to
minimize the degradation of video quality in the presence of packet losses, rather than just reducing packet loss as in traditional buffer management schemes. This new technique also provides fair distribution among contending video streams, based on service quality, in contrast to the traditional fairness objectives dealing with resource allocation. Simulation experiments using actual MPEG video traces have been carried out to test the performance of the FairFDDT scheme. The experiments show that the FairFDDT scheme not only significantly increases the viewing quality of individual video streams, but also provides a superior level of fairness among competing video streams. Also, high network efficiency is achieved.

FairFDDT relies on video-specific functions inside the network. Implementation in real networks is feasible with the help of Active Networking technology (AN) [1], because AN-based techniques make networks allow recognition of application and process-specific procedures, including video-specific processing. It seems clear that application-specific control within the network is almost inevitable, and the FairFDDT is a good example of this approach. Future work will extend the FairFDDT scheme to multi-hop routers by incorporating a loss assignment scheme.

References