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MINIMIZING PACKET LOSS USING CONGESTION CONTROL SCHEME FOR VIDEO STREAMING

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Abstract

In the Multimedia streaming the transferring of data is continuous streaming. Before the actual transmission the frames were divided into packets. For the creation of efficient communication the receiver side must receive all the packets. For the video streaming if there is any packet is dropped it is useless.in order to overcome this problem, we use Selective frame discard analyse their performance by means of competitive analysis. The QoS of a video stream is measured in terms of a cost function, which takes into account the discarded frames. But in this method due to congestion or heterogeneous nature of the network, data loss may be occurring. So, in order to overcome this problem, introducing an innovative technique called buffer management algorithms in specific environments. In such environments, the algorithm used to decide which packet to drop in case of buffer overflows, to avoid goodput degradation. This paper present a model which captures such interpacket dependencies, and build algorithms for performing discarding the packet. Traffic consists of an aggregation of multiple streams, each of which consists of a sequence of interdependent packets. So, by using this buffer management algorithm we effectively reduce the packet loss and maximize the Quality-of-Service.

Keywords*:* Buffer management; inter packet dependencies;

1. INTRODUCTION

Wireless network refers to any type of computer network that utilizes some form of wireless network connection. It is a method by which homes, telecommunications networks and enterprise installations avoid the costly process of introducing cables into a building, or as a connection between various equipment locations. Wireless telecommunications networks are generally implemented and administered using radio communication. This implementation takes place at the physical level of the OSI model network structure.
The emergence of high-speed networks facilitates many multimedia applications that rely on the efficient transfer of compressed video. Such applications include streaming video broadcasts, distance learning, shopping services, etc. However, compressed video, especially variable-bit-rate (VBR) [4][8] video, typically

exhibits significant burstiness on multiple time scales, owing to the encoding schemes and the content variation between and within video scenes. This burstiness complicates the design of efficient transport mechanisms for such media. In a network where resources such as bandwidth and buffering capacity are constrained there is a need for an efficient video delivery system that can achieve high resource utilization and maximize the QoS perceived by the user.

A simple strategy called Frame-Induced Packet Discarding, in which upon detection of loss of a threshold number of packets belonging to a video frame, the network attempts to discard all the remaining packets of that frame. In the problem of optimizing the quality of the transmitted video for a given cost function has been considered with leaky bucket constraints. Our work differs from theirs in that we are trying to optimize the QoS perceived by the user, rather than minimizing loss in general. In offline algorithms for optimal selective frame discard have been considered. The notion of selective frame discard at the server has been introduced and the optimal selective frame discard problem using a QoS-based [5][7] cost function has been defined.

The major problem for packet loss in the networks is buffer overflows due to congestion. In case where the traffic has interpacket dependencies, dropping packets due to overflow may result in very poor performance. For streaming data with packet dependencies, it must differentiate between the throughput and goodput [10]. Throughput is the amount of data delivered in terms of packets, and the goodput, is the amount of data decoded effectively at the receiving end. These two measures were different, e.g., the throughput can be high, while its goodput can be low. As an example of this scenario considers the case where a single packet is dropped from every frame, that will be result in zero goodput, even the overall packet-level throughput will be high. The steps to decide which packet to drop in case of overflow is important to system performance, while considering that such a decision might affect other packets which have already been forwarded, or packets that have not yet arrived. The aim of this paper is to maximize the goodput.

This paper proposes and develops buffer management algorithms in specific environments, namely, those employing a FIFO scheduling buffer. We use FIFO scheduling buffer because: 1) it is simple, 2) Incoming traffic arrival order maintained, so that packet reordering is maintained, and

1.1 Video compression

The video compression a video frame that can be compressed using different algorithms with different advantages and disadvantages, video frames algorithm are called **picture types** or **frame types**. The three main frame types are **I**, **P** and **B**.

- **I-**frames are the least compressible but it does not require any other video frames to decode. It is the main frame
- **P-**frames use data from previous frames to decompress and largely compressible than I-frames.
- **B-**frames can use both before and after frames for the reference of data to get the largest amount of data compression.

Here are three types of *pictures* (or frames) used in video compression: I-frames, P-frames and B-frames. The Fig 1.1 shows how the frames are arranged in an order.

- An **I-frame** is an 'Intra-coded frame', in effect a fully specified picture, it is a static image file. P-frames and B-frames hold only some of the image information, so they need less space to store than an I-frame and thus improve video compression rate.
- A **P-frame** is a 'Predicted frame' contains only the changes in the image from the before frame. For example, in a scene where a car moves across a stationary background, only the car's movements need to be encoded. The encoder need not to store the unchanging background pixels in the P-frame, thus saving space. P-frames are also known as *delta-frames*.
- A **B-frame** is a 'Bi-predictive frame' saves even high space by using differences between the present frame and both the preceding and following frames to specify its content.

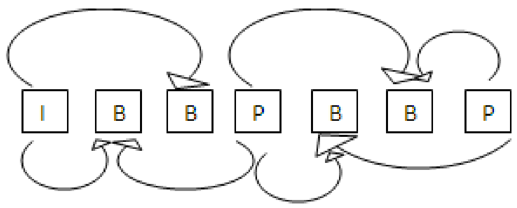

Fig.1.1 Bidirectional prediction

1.2 Related work

In case if the buffer overflows the packet were not discarded, the Frame were discarded, they follow the various scheme to discard the frame, In **the selective frame discard** [1][2][3] policy the server pre-emptively discards the frame by taking the network bandwidth and client QOS into an account. Here the end to end delay is reduced but complexity of optimal algorithm is high.

In paper propose a buffer management policy that takes packet dependencies into consideration, an entire frame is dropped too several packets of the frames are dropped. As a rule for dropping packets, they suggest a **latest-frame-first rule**. This rule states that the frame to be dropped that the smallest amount range of packets are delivered [9].

2. Proposed work

To overcome the issue of packet losses in the multimedia streaming applications we providing two guidelines for developing the buffer management algorithm. Actually in the wireless network, before transmission takes place across the network the data frames are split into n number of smaller sized packets. In the video streaming applications the packets are dependent and belong to each other. The receiving side can make use of the data only if it receives all packets of a frame. The problem of ensuring that all packets of a frame arrive at the destination is critical one. When considering real-time traffic, such as streaming multimedia traffic, where retransmission of missing packets is not possible due to delay constraints by the application and also to increase the cost.

We suggest and extend the buffer management algorithms. Our focus on FIFO scheduling is:

It is too simple; it maintains the receiving order of incoming traffic, hence avoiding the need for mechanisms that deal with packet reordering

Contribution

The main contributions are as follows:

We provide two design guidelines for algorithms, namely,

- No-regret: Once a frame has a packet came into the buffer that should make every attempt to deliver the complete frame.
- Ensure-progress: Deliver a complete frame as soon as possible.
- proposing a buffer management algorithm, WEIGHTPRIORITY that follows these guidelines. We analyse the performance of our algorithm, and show that for any arrival traffic the ratio between its performance and that of an optimal algorithm is always bounded.

2.1 System Model

Consider a collection of N streams in traffic model of small-sized packets, designated by $T_1,\dots,T_n$. Each stream T_n is observed as a sequence of frames, r_i^n , each consisting of a sequence of exactly k packets, $s_1^{n,i},\dots,s_k^{n,i}$. A packet $s_j^{n,i}$ is referred to as the j-packet of framer_i^n, and its arrival time is denoted by$b(s_j^{n,i})$. When referring to packets, we will sometimes omit the frame index i, and use the notation $\{s_j^n\}_{j=0,1,\dots}$ when referring to the sequence of packets corresponding to streamSx_m, where s_j^m denotes the jth packet of streamSx_m, and the $(j mod k)$ packet of frame $\frac{j}{k} \vee f^m$ (i.e., the $\frac{j}{k} \vee$th frame of streamSx_m). The packets of a stream arrive in order, i.e., $a(s_j^{ni}) \leq a(s_{j+1}^{n})$ for all j. The above notation gives that the structure of incoming packet in a stream T_m consisting of r_m frames

2.2 Buffer Model

When the packets arrive in FIFO buffer that can store up to B ≥ k packets and which can send out one packet. At the beginning the buffer is empty. Every cycle consists of two steps. The first step is called the delivery step if the buffer is nonempty the packets are transmitted on the link. The second step is called the arrival step. In this step the packets arrives at the buffer. At the discretion of the buffer management algorithm, some packets will be dropped, while the other packets are stored in the buffer.

2.3 Weight priority algorithm

In the weight priority algorithm we have to calculate the rank for the frame and packet, The rank of the frame and the rank of the frame We present the main algorithm called WEIGHTPRIORITY, which follows the design criteria. In the beginning of the arrival step of any cycle t, and for every frame r_i^m define its rank at t by

$ra_t(r_i^m)$ =($w_t(r_i^m, sxm)$)

For completeness, we also define a tie breaking rule for frames of the same stream, where given any two such frames corresponding to the same stream, while considering the frame with the lower index of the frame as having the highest rank.

3 Conclusion

The buffer management technique is used for avoiding the packet loss. Because in case of the streaming application packets depends on other packets. So, if there is a loss [6] in a packet the whole sequence is said to be waste. In order to avoid these problem we use buffer management schemes to discard some packet that depends on the ranking by providing the guidelines for the design of such algorithms,. We provided guarantees on its performance under any traffic conditions by proving it has a bounded competitive ratio.

Securing multimedia data has become of utmost importance especially in the applications related to military purposes. Using innovative encryption algorithms for video sequences are necessary for protect the data. This can be done in future work.

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A Brief Author Biography

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