Quality Adaptation For MPEG-4 FGS Video Streaming Over Internet

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ABSTRACT

A quality adaptation mechanism for MPEG-4 FGS video streaming is proposed. It decides the way FGS video is truncated at the server. The client receives the truncated video stream and decodes it. In any streaming application the system has to ensure that the streaming proceeds in a time synchronized maner, the quality adaptation algorithm tries to handle this misalignment and keep the client buffer filled to adequate level.

To accomplish this work, a new protocol TCP-Friendly Rate Control (TFRC) is used in the transport module in the server. The TFRC module continuously reports the available bandwidth by monitoring the packet loss rate on the path.the quality adaptation module truncates the video so that it fits into the avalable bandwidth. It also ensures that this truncation is done in such a way that the quality variation from frame to frame is minimal.

Keywords: *FGS VidioStreaming,TCP-Friendly Rate Control*

1. INTRODUCTION

Streaming audio and video application are becoming increasingly popular on the internet the most challenging part of multimedia communication is the real time transport of live video or stored video. This work deals with transport of the stored video. Real time transmission of stored video is called video streaming. There are two ways for transmitting stored video. In the first method, user downloads the entire video file and then starts playback of the video file. This procedure can take a long time and user machine should have enough space to store the large video file. Because of these two reasons a second method called streaming is often preferred. Here, instead of waiting for the full download to be over, as part of it arrives, it is played back. However, to support such streaming video over the Internet, several changing issues are to be addressed. Streaming application is generally delay sensitive and semi-reliable. The Internet in its present form dose not support QoS guarantees. The congestion in the network is controlled by employing appropriate congestion control algorithms on end systems.

Previous work has addressed transmission of stored VBR videos over networks were bandwidth reservation can be made [4,6]. In such schemes the receiver will have a buffer into which the server transmits smoothed video according to a pre-computed schedule. This smoothed video fits in to the reserved bandwidth. However, because of the fixed size of the buffer, and variable bit rate requirement of the video, smoothed video transmission may end up in buffer overflow or underflow. To avoid such situations the server renegotiate with the network for a different bandwidth. When a server renegotiate with the network for a different bandwidth, all the reservations along the path are to be changed, and this is a costly procedure. Many researchers have proposed optimal transmission scheduled algorithms, which tries to minimize the number of renegotiations. Most of them use dynamic programming to compute the optimal transmission schedule under the fixed buffer size constraint. All these works assume that bandwidth reservations can be made and such reserved bandwidth is constant till it is renegotiated. In the study reported here, we try to transmit video over networks that do not support bandwidth reservation.

2. BACKGROUND

In this section we review the video coding scheme, MPEG-4 FGS and the transport layer protocol TCP – Friendly Rate Control (TFRC) protocol we use in our study.

2.1 Video Coding

This subsection gives a review the relevent features and properties of the video coding scheme, MPEG-4 Fine Grained Scalability.

Since raw video consumes a huge amount of bandwidth , it is usually compressed befor transmission. There are two types of video compression, namely, sclable and non- sclable video coding. A non – sclable video encoder produces a single bit stream, which should be received in full for proper decoding. On the other hand, a sclable encoder [3] produces multiple streams (or layers), which can be decoded even if only parts of it are received [12]. In this study we use such a sclable coding scheme, called Fine Grained Sclable (FGS) coding [11].

FGS coding coding has recently been aceepeted as an amendment to the traditional non – sclable approch by MPEG -4 for the streaming viedo profile. FGS became an international standard in march 2001. An FGS encoder compress a raw video into two sub streams: a Bace Layer (BL) bit stream and an Enhancement Layer (EL) bit stream. Figure 1 shows the structure of a FGS coded video. The base layer is coded with an MPEG-4 compliant non- sclable coder and the enhancement layer consist of a single stream coded in a progressive maner . The enhancement bit stream is coded in a special way, called bitplane

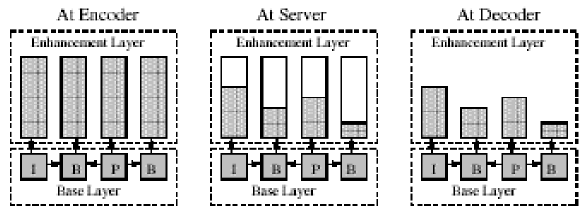

Fig. 1: Structure of FGS Video

coding. The advantage of this bitplane coding is that, the Enhancement bit stream can be truncuated at any level depending on the availablity of transmission capacity. At the receiver the base layer and the truncated enhancement layer can be combined and decoded.

Now, in order to ensure that the playback starvation does not occur at clients, rate adaptation of the video is to be done. This rate adaption is done in FGS by truncating the EL. The receiver will be able to decode the FGS video whose EL is truncated in any arbitrary way. However such an arbitrarily truncated FGS video will result in an output whose quality varies widely from frame to frame . It is known that differential sensitivity has significant impact on the human visual perception [14]. This wide variation in quility between adjacent frames is annoying to viewers. Thus, FGS video has to be truncated so that it meets both these requirments, namely, it does not cause playback starvation at clients, and quality variation among frames is minimal[8]. This is the problem that we try to address in this study.

FGS video consists of a non – sclable coded BL and bitplane encoded sclable EL. Since there is no motion compensation in the EL and the decoding of variable bit rate – base layer (VBR-BL) is supposed to yield constant quality, the server can send the same number of EL bits for each image within a scene. A scene here is defined as a temporal alignment of consecutive frames . fig 2 shows two ways of encoding the video signal and corresponding quality fluctuation for various bit allocations in BL and EL [7]. If the BL of the video is constant bit rate(CBR) coded, then we get variable decoded quility as shown in left half of the figure 2. on the other hand, if the BL is variable bit rate (VBR) coded,we get constant quility as shown in right half of the figure 2. the addition of the

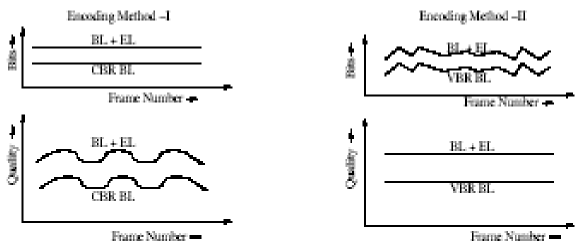

Fig. 2: Quality fluctuation for different bit allocations in Base Layer (BL) and Enhancement Layer (EL)

Enhancement layer to these two form of encoded base layer, continue to reflect the corresponding variation in quality .

When the BL has equal number of bits for all frames, the quality fluctuates widely. On the other hand, when the BL is coded with variable number of bits for different frames, constant quality is obtained. Figure 2 illustrates this problem and it is established by the analysis of a large library of MPEG-4 FGS videos in [13]. So , to obtain minimal variation in quality, the server should try to send the same number of EL bits from consecutive frames.

Every frames has some of its bits in the BL and remaining bits in the EL. We truncate the EL so that the frame fits into the available transmission capacity. If e_n bits are present in the EL of the n^{th} frame of video, and E_n bits are selected for transmission, the quality factor is defined as

$$\alpha_n = E_n / e_n$$

So, with this definition of quality factor, and the property of FGS that equal number of EL bits from adjacent frames yields approximatedly contant quality, our problem reduces to fixing the quality factors of adjacent frames as close as possible to each other.

2.2 TCP-Friendly Rate Control (TFRC) Protocol

Most best-effort traffic in the Internet uses Transmission Control Protocol (TCP) at its transport layer. However, some of the characteristics of TCP, such as retransmission, bursty transmission rate etc. are not well-suited to multimedia support. Any new protocol we introduce for the purpose, should share the bandwidth fairly with other ongoing TCP connections. The TCP-Friendly Rate Control (TFRC)[9] Protocol is one such protocol, which uses the TCP response function to determine the transmission rate.

The advantage of using TFRC, is that the sending rate is relatively steady, and the bandwidth share it occupies will be that of a TCP connection [15].

3. QUALITY ADAPTATION

In order to ensure that the streaming video does not cause congestion in the network and does not cause playback starvation at clients, rate adaptations of the video is to be done .this rate adaptation in FGS by truncating the EL. The receiver will be able to decode the FGS video whose EL is truncated in any arbitrary way. However such an arbitrarily truncated FGS video will result in an output whose quality varies widely. This wide variation in quality is annoying to viewers. Thus FGS video has to be truncated so that it meets both these requirements, namely it does not cause congestion in the network, and quality variation among frames is minimal. This is the problem that we try to address in this study.

Thus, quality adaptation is an application layer technique, which decides the way the enhancement layer of the FGS video, is truncated at the server. The client receives the truncated video stream and decodes it. In any streaming application, the system has to ensure that the streaming proceeds in a time synchronized manner. The quality adaptation algorithm tries to handle this misalignment and keep the client bugger filled to adequate level.

The above figure3 shows over end-to-end client server architecture. The transport module in the server uses TFRC Protocol. The TFRC module continuously reports the available bandwidth to the quality adaptation module. TFRC determines the available bandwidth by monitoring the packet loss rate on the path. The quality adaptation module truncates the video so that it fits into the available bandwidth. It also ensures that this truncation is done in such a way that the quality variation from frame to frame is minimal.

The basic idea of our quality adapted streaming is as follows. After the server has started streaming, at any instant there are three possibilities.

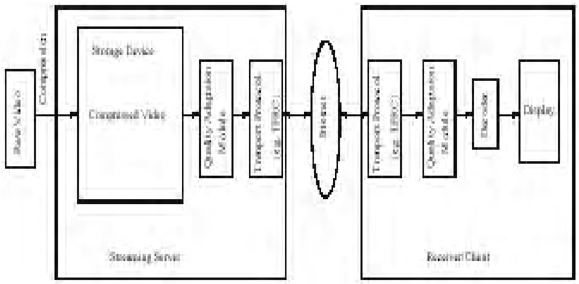

Fig. 3: End-to-End Architecure of Video Streaming

It has transmitted all the frames that it ought to have transmitted by this instant.

It is lagging by a few frames

It is leading by a few frames

It tries to compensate for any lagging or leading, while determining the number of frames(k) to be sent in next time slot. The length of the time slot(Tw) is fixed. Now it looks at the future video bit requirement, and determines the quality factor. The algorithm at the server is shown below:

start transmission at t_i

at instant t_k

Number of frames that must have been sent by t_k, θo = $(t_{k-} t_{i)/}$ time per frame number of balanced frames , θ_b = number of frames we ought to have sent , θ_o - number of frames actually sent, θ_a

If the θ_b>0 (θ_b < 0), it means the transmission is lagging (leading)

{ if it is lagging , we need to send this balance and a fixed number of frames in the next time slots (squeeze in this balance , in the time we could otherwise have used for a fixed number of frames)}

if (θ_b>0),number of frames to be sent in the next time slot , θ_k= fixed number of frames , θ_w + balance , θ_b

if (θ_b < 0), number of frames to be sent in the next time slot , $\theta_{k\,=}$ fixed number of frames , θ_w - balance , θ_b

if (θ_b= 0), number of frames to be sent in the next time slot , θ_k = fixed number of frames, θ_w

Determine the sum of base layer frame sizes falling in the next time slot, (B)

Determine the sum of enhancement layer frame sizes falling in the next time slot, (E)

Estimate the anticipated bit transmission capacity for sending the enhancement layer during the next time slot repeat until all frames are sent

$C_e = C * T_w - B$

Where c is the current instantaneous bit rate .

Quality factor = C_e/E {truncate the frame so that lagging /leading is minimized}

Our algorithm works by looking at the future video bit requirement and adjusting the quality factor accordingly. Consider the case when the bandwidth is decreasing. Now, if the future video bit requirement is less, we can continue to maintain the same quality. On the other hand, if the video bit requirement ahead is more, we start truncating at the earliest possible moment. The action is just the opposite when the bandwidth is increasing. The result of this is a narrower variation in quality factor, which in turn means a smoother quality video.

4. SIMULATION EXPERIMENTS

4.1 Simulation Methodology

We use ns-2[1] network simulator to obtain bandwidth traces of TFRC. A dumbbell topology was simulated in ns-2 and several TCP and TFRC connections were set up between the edge nodes. This is illustrated in figure 4. The bandwidth share of a particular TFRC connection in the presence of several other TCP and TFRC connections was computed from the ns-2 trace output. We use two VBR video frame size traces to represent FGS videos truncated for a quality factor of 1. In our simulation, we send quality-adapted video over a connection,

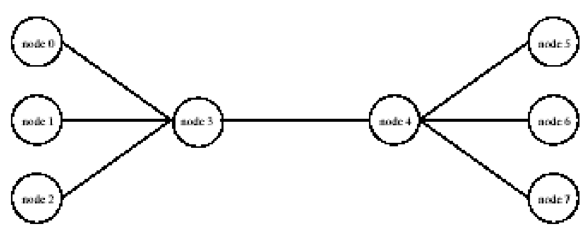

Fig. 4: Dumbbell topology simulated to obtain TFRC bandwidth share

whose bandwidth is computed from the ns-2 trace file.

4.2 Results

Figure 5 and 6 shows the variation of quality factor among a few consecutive frames. The bold one corresponds to the truncation using our quality adaptation algorithm. The dashed one is the quality factor variation resulting from truncating according to the bandwidth variation. Our results clearly show that the variation in quality among frames is less when the algorithm is used

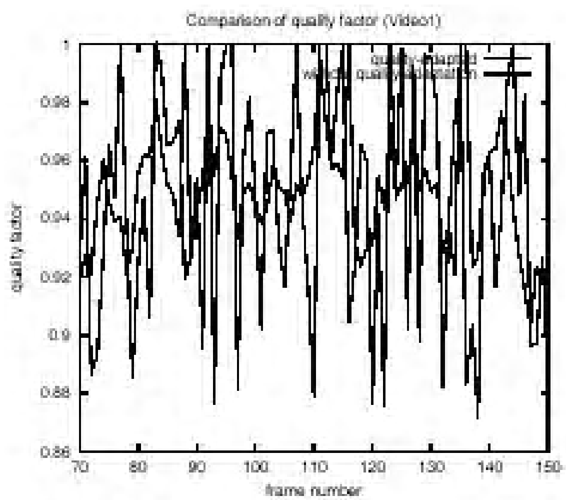

Fig 5: quality factor variation for quality-adapted stream and without quality Adaptation (Video1)

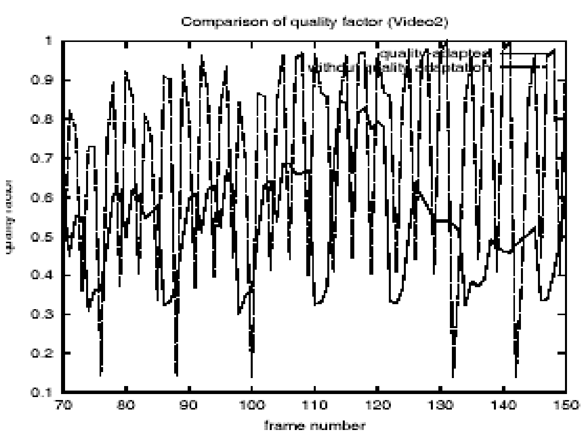

Fig. 6: quality factor variation for quality-adapted stream and without quality Adaptation (Video2)

Figure 7 and 8 shows the probability distribution of quality factor for the quality-adapted video, and video without quality adaptation. It shows that quality adapted case results in higher quality video. From the probability distribution curve, it might seem that the variation in quality is more for the quality-adapted case. However this is not true, as the probability distribution curve does not capture the quality variation between consecutive frames. This variation in quality between consecutive frames can only be observed from figures 5 and 6.

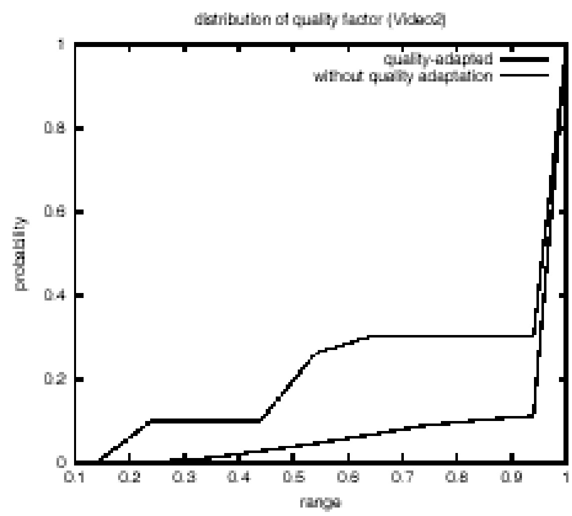

Fig. 7: Probability distribution of quality factor(Video1)

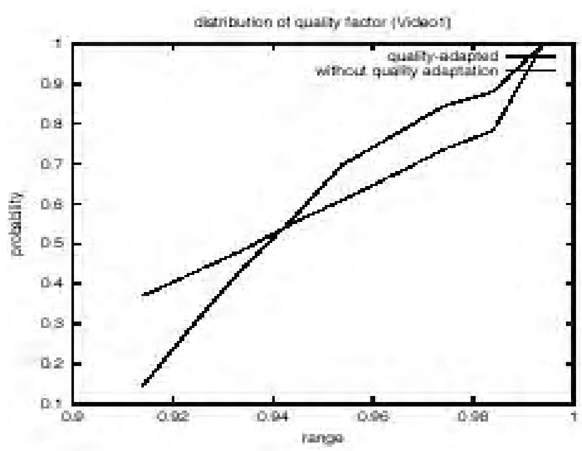

Fig. 8: Probability distribution without quality factor(Video1)

5. RELATED WORK

Unlike the field of video bit rate smoothing of non-scalable VBR video, where a large body of work is available, the field of FGS video streaming with smoother quality variation is relatively new and only a few published works are available. In this section we cover some of the important and closely related ones and highlight the differences.

Keith Ross et al. in [17] provide a theoretical framework for adaptive rate control of FGS video stream over wired network. The rate adaptation was performed for varying bandwidth through the wired network. It considers a single FGS video stream where both base layer and enhancement layer are CBR coded. The CBR coded video cannot be expected to give constant quality. Our optimal transmission polices are developed on the lines of this work but with important differences. Their heuristic algorithm for real time quality adaptation also differs from ours. Their heuristic considers only the buffered video and the availability of resources while fixing the new truncation level. In particular their heuristic gets aggressive when the buffered video is large and defensive otherwise. This perhaps is not a good strategy, as the future video bit requirement is also a deciding factor. For example consider a case, where the buffered video is large and the future video bit requirement is also large. As per the above heuristic which looks only at the buffered video, if we get aggressive by relaxing the truncation we might run into trouble. So we believe that a good heuristic should considered both the buffered video as well as the future bit requirement. Martin Reisslein et al. in [18] presents a prefetching protocol for media streaming over wireless enviorment. They have used non-scalable [VBR] videos, and their only goal was to minimize butter overflow. We use the same wireless environment as used by them. quality adaptation for layered was studied by Rejaei et al. in [16]. They used a TCP like protocol Rate Adaptation Protocol [RAP] at the transport layer, for streaming discrete layered videoed over wired network. As the transmission rate directed by RAP fluctuates widely and frequently, layered are added and dropped. To minimize such frequent additions and dropping, quality adaptation was performed.

6. SUMMARY

A quality adaptation mechanism for FGS video streaming is proposed. We evaluate the performance of our algorithm and compare it with FGS video streaming without quality adaptation. Our results show that quality adaptation results in smoother quality variation among frames and improved quality. Experimental verification of the algorithm remains to be done. The property of the proposed algorithm over other transport protocols is not yet investigated.

In this work we considered the quality adapted video streaming from a server to a single client over a wired Internet connection. In the next work we extend this idea, and investigate a streaming system where a server is streaming videos to multiple wireless clients.

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