

Dynamic and Adaptive Media streaming over Heterogeneity network by Wireless Channel

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Abstract— Video streaming over wireless (networks) has gained popularity in recent times with the development of heterogeneity networks. The demand and market for video based services is expected to grow substantially in the future. In that network is used for video streaming, the inherent unreliability of frequently results in corrupted video packets and jitter during video playback. Such an aberration is especially pronounced in wireless networks due to the fading and multipath interference impediments in wireless channels. Protocol based video transmission provides reliability at the cost of severely increased delay arising from frequent retransmissions in case of packet error over wireless links and constraints from session and application layers for in-sequence delivery. These delays are substantially worsened in low SNR (signal-to-noise power ratio) wireless environments. Multiple Protocol connections, which efficiently utilize the bandwidth of the wireless channel, have been demonstrated to yield significantly improved quality of video streaming over wireless channels while enhancing the reliability of video delivery compared to protocols. In this paper we propose new scalable hierarchical adaptive bitrate video coding over heterogeneity network for reliable video transmission over wireless channels while reducing the delay and jitter of video streaming.

Key words: Bandwidth, Error Control, Internet, Overhead, PER, Prioritization, QoS, Video Streaming, Wireless Channel

I. INTRODUCTION

Video camera surveillance (VCS) has a growing presence in the public arena over the past several decades in stores, civic buildings and even on public streets. Video surveillance increases security and decreases liability [1]. However, the communication systems deployed for the transport of the video data of these outdoor systems are expensive [2]. Broadband WMNs allow real-time streaming of video footage from inside transit vehicles, providing first responders with the actionable intelligence they need to respond effectively to any emergency situation [1].

In this work, we exploit the possibility of path diversity to use OR of streams lost packets and use NC forwarding of multiple packets from different streams. This approach avoids streams contending in the most congested region of IGW and achieves packets delivery before playing deadline. Our performance metrics of interest is the average video streams delay, the video streams throughput, the video streams loss and the packet delivery ratio under proposed algorithm and shortest path routing (SPR).

II. RELATED WORK

Many approaches to using FEC algorithms have been extensively analyzed in literature. In [4] a method of using FEC codes called Linear Increase Multiplicative Decrease with History (LIMD/H) to stream Internet video is discussed.

The method uses FEC codes to achieve robust and efficient Internet video streaming and multicast consistent with the existing Internet architecture. This approach used history of packet losses to measure congestion levels in the network and incorporated FEC to formulate an algorithm that converts the prioritized stream of bits into an un-prioritized stream of bits [4].

From the simulation results presented in [4], it was observed that LIMD/H offers an approach that better tolerates variations of packet loss in multimedia transmissions over the Internet. However, because LIMD/H looks up the history of packet loss before adjusting the transmission rate in the network, this causes slowness in responding to changing network conditions. The approach described in this paper proposes a method that uses queue length at the Access Point as a measure for congestion thereby ensuring a faster response time while dynamically responding to changing network dynamics. Another approach that combines FEC with prioritization is that described in [5]. The authors investigated the effects of adapting interleaved packet-level FEC according to packet transmission properties to address issues of delay introduced during correction of errors in video streams at the wireless channel [5].

The original packets were further prioritized according to a packets assumed importance [5] and given additional FEC protection. The interleaving method introduces delays in the network because packets often have to be reordered. The proposed algorithm corrected packet errors due to wireless channel error using a Static FEC mechanism. This means there is a constant high FEC overhead in the network regardless of wireless channel conditions. Such a setup will perform poorly when the wireless network deteriorates.

In [6] the authors investigated the trade-off between the bandwidth consumed by FEC in terms of the redundant packets generated versus the bandwidth lost by TCP to make room for these redundancies. The authors efficiently compute the number of FEC packets so as to not hamper TCP performance in the network [6]. Although the authors in [6] focus mainly on the bandwidth trade-off between bandwidth consumed by FEC packets and that used by TCP, the adaptive FEC mechanism used has similarities to the proposed mechanism in this paper in ensuring fairness of network resource usage. However, the approach described in this paper not only makes the addition of FEC redundancies in such a way that the loss probability of packets in the network is reduced but also improves delivered video quality by adapting the level of FEC in the network based on current network channel conditions.

III. VIDEO STREAMING DESIGN SCHEDULING

In our work the video streams from n VCSs to the IGW is scheduled with the combination of three main components:

- To avoid flows contention near IGW, we propose different flows gathering to a certain MR distant from IGW.
- In case of packet forwarding failure, we use OR forwarding instead of packet direct retransmission.
- To reduce the overall delay of the n different streams, we use a NC forwarding of packets belonging to the n streams.

A. Flow Gathering Construction (FGC):

The main aim of FGC is to design an optimal throughput sub graph that avoids multiple throughput concentration around the IGW. The design of FGC is an algorithm that sequentially assigns the best streaming routes for the different flows to an aggregation MR called MRag. For each source VCS search the shortest path to IGW, we assume the maximum and minimum source path length to IGW is l_{max} and l_{min} , respectively. Search MRag such that the shortest path length of MRag to IGW lag satisfies the following conditions:

- $lag \geq l_{min}$
- Path length l of a VCS to MRag satisfies $l \leq l_{max} - l_{min}$
- If we have several MRs satisfying the above two conditions, the MR with greatest lag is elected. If there is no elected MR for the n VCSs, search an MRag for any of $n - 1$ VCSs to continue the recurrence over the rest of MRs. In the case that we find an MRag for some VCSs, construct a multiple unicast shortest path from these VCSs to their corresponding MRag. For an illustrating example see Fig.2b. If we have no elected MRag, the algorithm switches to the SPR construction.

B. Adaptive Frame Length Control:

This scheme is implemented at the data link layer in the network protocol stack. It is based on the following motivation. The probability of bit-error for transmission across wireless channels is significantly high. In such scenarios, high packet corruption rate leads to retransmission of entire packets, leading to wastage of bandwidth. Further, it is important to note that the probability of packet error increases with the packet length. As the packet length increases, the probability of retransmission increases, with loss of efficiency. Hence, packet length adaptation in sync with the signal strength of the fading wireless channel is key to efficient bandwidth utilization in wireless scenarios.

C. Dynamic Adaptive Streaming:

Dynamic Adaptive Streaming is an application layer technology with Protocol as the underlying transport layer mechanism. This scheme is designed to stream video by dynamically switching the streams according to the bandwidth available to the user [15]. This technology has several drawbacks as described below. Dynamic adaptive streaming requires multiple bit-rate video sequences already encoded and stored at the server side and hence cannot be used in live broadcast video streaming scenarios. Further, the storage of video sequences at several rates tremendously increases the storage space required at the streaming server. Since it is designed to wait for the next key frame, if the rate of the fading wireless channel is lowered momentarily, the playback at the client is halted, leading to jitter in the received video. Moreover, it switches between different bit-rate video sequences based on bandwidth, whereas the key criterion in

wireless channel is packet corruption resulting from burst errors and not network congestion. Hence, the efficiency of bandwidth utilization will degrade significantly leading to lower quality video at the user end.

D. Dynamic Congestion Control Protocol (DCCP):

DCCP [16] is designed for applications such as streaming which can tradeoff between the reliability and delay of the streamed video. It employs two types of congestion control namely: protocol-Friendly rate control (TFRC) [17] and protocol-like congestion control [18]. These schemes, which were primarily designed for wired networks, inherently have drawbacks similar to the conventional schemes and perform poorly in wireless conditions with high levels of noise and interference.

E. Adaptive Bit Rate Streaming:

Adaptive bit rate streaming is a technique used in streaming multimedia over computer networks. While in the past most video streaming technologies utilized streaming protocols such as RTP with RTSP, today's adaptive streaming technologies are almost exclusively based on HTTP and designed to work efficiently over large distributed HTTP networks such as the Internet.

It works by detecting a user's bandwidth and CPU capacity in real time and adjusting the quality of a video stream accordingly. It requires the use of an encoder which can encode a single source video at multiple bit rates. The player client switches between streaming the different encodings depending on available resources. "The result: very little buffering, fast start time and a good experience for both high-end and low-end connections."

More specifically, and as the implementations in use today are, adaptive bitrate streaming is a method of video streaming over HTTP where the source content is encoded at multiple bit rates, then each of the different bit rate streams are segmented into small multi-second parts. The streaming client is made aware of the available streams at differing bit rates, and segments of the streams by a manifest file.

IV. OVERVIEW OF PROTOCOL AND VIDEO OVER WIRELESS

Protocol based congestion control is especially suited for standard coaxial cable or fiber based networks, which have higher reliability compared to fading wireless links multiplexed over the shared frequency spectrum as explained in the previous section. As a result, packet error rate over a wireless link is significantly higher due to the burst error characteristic of the fading wireless channel. Several techniques such as ITCP (indirect protocol) [4], Link level retransmissions [5], Snoop protocol [6] have been proposed in literature to improve the performance of protocol over wireless networks. However, these schemes distort conventional protocol semantics. Other schemes such as Adaptive Frame length Control [7], Parallel protocol for packet transmission over wireless networks [8] are much more suited for practical implementation since they enhance the performance of protocol over wireless networks without violating the protocol semantics.

As shown in Fig.1, application of the 2-D Haar wavelet transformation [11] on the target video frame leads to a hierarchical base and enhancement layer decomposition of the frame. Also, the base layer contains energy compacted

information of higher significance compared to the enhancement layer, which contains high frequency component data of lower significance. Subsequent to the decomposition, the individual layers are compressed temporally through *motion estimation* followed by differential temporal encoding, *DCT* [12], *Quantization* [13] and *VLC* stages respectively. We employ the standard MPEG-2 encoding [14] quantization table during the quantization phase. On completion of all the above phases, the encoded video base layer comprises of the most significant video information and is transmitted over the TCP connection of lowest index at a higher SNR and the enhancement layer over the second TCP connection at lower SNR. At the receiver, on successful reception of the video frame packets, the video decoder repeats the above stages in reverse order to rebuild the original video frames using the received video layers. The proposed application employs a sequential number for each TCP connection so that the receiver is aware of all the packets that have been lost due to corruption. Towards this end the application layer sets a field for the start of each frame.

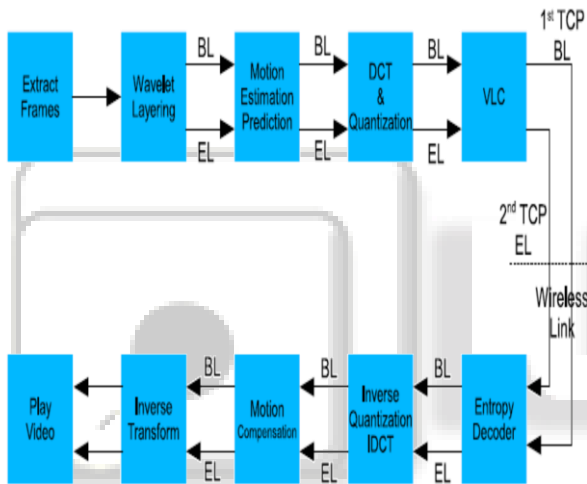


Fig. 1: Video transmission over heterogeneity network with adaptive bitrate streaming algorithm. EL-Enhancement Layer, BL-Base Layer

V. EXPERIMENTATION AND RESULTS

From the simulation data Fig.4 we can observe that using SPR, as the number of VCSs increases, the streams delay increases dramatically. But with proposed algorithm the video delivery increases slightly from 0.6s to 1s. From the two observe that when the the number of VCSs and the average hop length exceed 2, the SPR scheme causes a decrease in the aggregate throughput because more used links contend with each other. But adaptive bitrate streaming reduces contention between paths near the IGW.

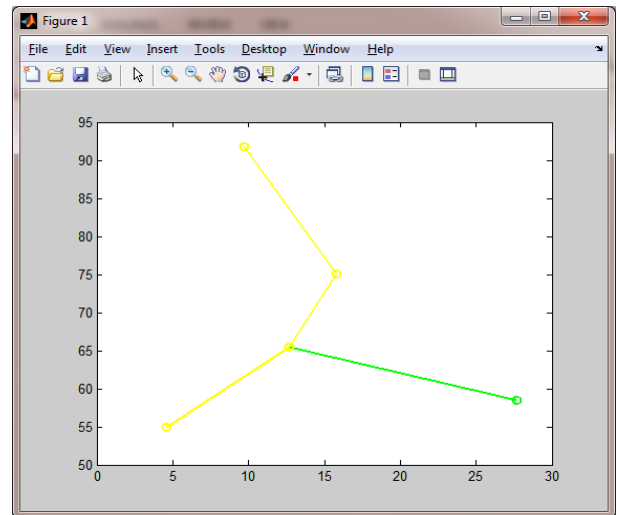


Fig. 2: Priority route of the network out of possible path identified by adaptive bitrate streaming algorithm.

Therefore adaptive bitrate streaming produces an advantage to increase the average throughput. The figures also show that an increase in the number of VCSs and the average hop length does not affect the throughput of adaptive bitrate streaming. Fig 2 shows the variation of priority route for streaming video throughout the network. As the bit rate increases the flow under SPR starts to suffer from major packet loss. But, because adaptive bitrate streaming can increase the capacity of the network with NC or can correct packet loss, adaptive bitrate streaming reduces losses due to the delay and link packet loss in the network.

VI. CONCLUSION

Several algorithms for streaming media traffic over internet have been implemented and tested in last few years. In this paper we presented a survey on recent trends and progression in the area of multimedia streaming in heterogeneity network. By discussing the adaptive video streaming algorithm based on the network awareness, various media content aware congestion control protocols with its pros and cons and finally presented the limitations in existing media streaming solutions for multimedia streaming was solved. With an evolving research area there are many issues to be solved in the area of multimedia streaming congestion control. There is no standard method for comparing the congestion control protocols. There is no benchmark for the protocols used in media traffic. It seems that at present there is no single protocol exists that can act as the standard protocol for multimedia streaming. An algorithm that can solve all the problems of congestion control in streaming applications is still in the stage as castle in the air. More research work is needed to solve these issues efficiently. Our future work is to develop a novel protocol that can reach the stage of maturity to be the standard protocol for media streaming.

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