Double-Buffer Traffic Shaper Modelling for Multimedia Applications in Slow Speed Network

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Abstract

Lack of end-to-end quality-of-service support, unbounded packet-data delay, poorly adapted transport protocol and slow-speed network are factors militating against the deployment of real-time video streaming across the Internet at slow-speed. This paper presents a Double-Buffer Traffic Shaper (DBTS) for rate adaption of real-time video transmission in a slow-speed network using Transmission Control Protocol (TCP) for data transportation. The scheme was tested in Matlab and it was found to reduce variable bit rate (VBR) traffic burstiness to its barest minimum and variability to almost nil.

Keywords: computer networks, multimedia communication, transport protocol, slow-speed network

Introduction

As the internet is experiencing rapid growth, Network-based multimedia applications such as video conferencing, video-on-demand, E-learning and Telemedicine has become a vital and increasing application on the Internet therefore attracting research activities. The lack of end-to-end quality-of-service (QoS) support in today's Internet has caused significant difficulties to the deployment of these applications (Lam et al., 2005). Therefore, the issue of QoS provisioning has become more essential than ever. QoS must support a bandwidth guaranteed transmission service, and provide an efficient communication path for transmitting data under acceptable delay, delay jitter, network throughput and error rate (JerYang & ChengChou, 2003). Bandwidth management and traffic control have become more complicated. Network control must support different types of traffic classes at their required QoS and must efficiently utilize the network resources (Liang, 2004). TCP is recently becoming most acceptable as the transport layer protocol used for real-time streaming instead of UDP because TCP is faster, efficient, more responsive to network conditions and also integrates flow and congestion control scheme which removes the need for loss recovery and flow control at the

application layer (Xiong et al., 2010). Transmission of real-time VBR video applications typically requires high bandwidth and low delay(Xiong et al., 2010; Xiong et al., 2009; Lam et al., 2005; Liang, 2004; JerYang & ChengChou, 2003; Won & Shim, 2002). These requirements have been identified as major challenges to the deployment of multimedia applications across the Internet in sub-Saharan countries particularly Nigeria. The reason is not far-fetched from the fact that these networks are slow-speed networks characterised by low speed and high delay as observed on end-user devices. In South-West Nigeria for example where this study was carried out, it was discovered; from informal interviews conducted for users on the campus on the average data rate that at peak period 5-10kbps was the available/guaranteed data rate. This is a far-cry from the experiences of average Internet users in Europe whose user-end data rate can be as high as 500Mbps described in literature as fast speed networks.

The focus of this work is to develop a rate adaptive scheme capable of guaranteeing the transmission of real-time VBR video across the Internet over slow-network at minimal delay and loss rate. A Double-Buffer Traffic Shaper (DBTS) for real-time VBR video traffic was formulated using network of queues. The scheme was modeled as a death and birth rate process. The contribution of this article is the proposition of a rate adaptive scheme for VBR Video traffic over slow-speed networks. The output result shows that the proposed scheme can successfully remove uneven distribution of video frames. The rest of the paper is organized as follow: Section two discusses related works; section three gives the description of the proposed scheme - Double-Buffer Traffic Shaper (DBTS). Section four presents the results while conclusions are given in Section five.

Related Works

Many researchers have worked on Real Time Variable Bit Rate (RT-VBR) traffic mobility scheme. Dinesh et al presented a delay jitter control algorithm that guarantees delay jitter in a packet-switched wide-area network (Dinesh et al., 1991). İbrahim introduced Virtual Scheduling Algorithm (VSA) for traffic shaping and achieved Lower loss rate (Ibrahim, 2000). de Cuetos et al and Lam et al designed network-adaptive video applications using MPEG-4 Fine-Grained Scalable (FGS) to adapt the coding rate of the streaming video to the variations of the available bandwidth (de Cuetos et al., 2002; Lam et al., 2005). Bajic et al propose an end-to end buffer management scheme for video flow for robust and efficient scalable video communication Congestion control and was able to reduce packet loss and produce high video quality (Bajic et al., 2003). Arne and Jirka, generates real rate adaptive MPEG-4 streaming traffic using the Shaped Variable Bit Rate (SVBR) for adjusting the sending rate quality of service (QoS) performance for VBR video traffic (Arne & Jirka, 2007). Xiong et al., propose a video frame schedule scheme for rate adaptive real-time video transmission over TCP to reduce end-to-end delay in real-time video (Xiong et al., 2009; Xiong et al., 2010). All of the above schemes were found unsuitable for the slow speed network. The reason being that design parameters and assumptions mainly as it concerns a link's minimum bandwidth were set to $\geq$ 1Mbps which is far greater the minimum obtainable in a typical slow-speed network.

The Proposed Scheme (DBTS) Description

Video Transmission over the Internet

The first step in the transmission of video is compression of the raw video using encoding coder-decoder scheme (CODEC) such as H.263, H.264, etc at the sender-end. The compressed video is passed to the sender-buffer within the application layer, thereafter passed to the TCP sender-buffer wherein they are fragmented into TCP data packets. From within the TCP sender-buffer the fragmented video frames are moved into the network. The video frames are later received into a TCP receiver-buffer wherein they are defragmented and decompressed. The decompressed

video frames are then sent into the receiver playout-buffer. Fig. 1 depicts the stage by stage transmission of video frames from the sender to the receiver across the internet.
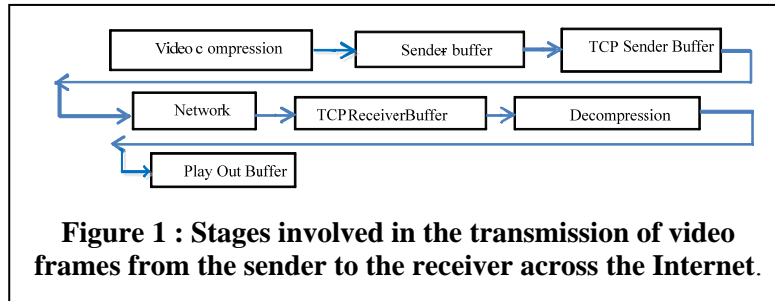

Figure 1 : Stages involved in the transmission of video frames from the sender to the receiver across the Internet.

In real-time video transmission, packet delay is experienced as video frames proceed from one stage to another. The cumulation of these delays result into an end-to-end delay (D_{END}). i.e

$D_{END} = D_{VC} + D_{SB} + D_{TSB} + D_N + D_{TRB} + D_D + D_P$ …………………..(1)

Where, D_{END} - End-to-end delay; D_{VC} – delay experienced during video compression;

D_{SB} – delay experienced at sender- buffer; D_N – delay experienced at network layer;

D_{TSB}– delay experienced at TCP sender-buffer; D_{TBR} – delay experienced at TCP receiver-buffer; D_D – delay experienced during video decompression; D_P – delay experienced at play out buffer.

Given that D_{VC}, D_D, D_{TRB} and D_P are commonly constant(Xiong et al.,2010), hence equation(1) becomes

$D_{END} = D_{SB} + D_{TSB} + D_N + C$ ……………………………………………… (2)

The delay experienced at the application sender-buffer, TCP sender-buffer and in the network are variable and of major interest to researchers (Xiong et al., 2009; Xiong et al., 2010).

The Double Buffer Traffic Shaper (DBTS)

Due to the variability in D_{SB}, D_{TSB}, and D_N, and the possibility to curtail them, we wish to introduce a rate-adaptation scheme called DBTS. DBTS will be located between the application sender-buffer and the TCP sender buffer because bursty compressed video frames arriving from the application will be regulated by the scheme before arriving into the TCP sender-buffer. DBTS is a traffic shaping technique developed upon network of queues principle. The first queue as a M/D/1/K and the second queues as a D/D/1/K. The flow of video frames is seen in fig. 2. The arrival rate into the first buffer is markovian (Balsamo, 2010)**,** its departure rate is deterministic at the feeding the second buffer, thereby ensuring that frame out flow from the second buffer is maintained at a constant rate. Video frames thereafter are sent into sender's TCP-buffer, hence achieving rate-adaptation. The frames flow dynamics can be expressed as a birth and death process. The transition is precipitated on H.263 encoder which sends an average of 10 frames per second. The arrival rate into the queue is Poisson, which implies that frames arrive in bursts (irregular spurts). The service rate is deterministic with mean departure rate, $\mu=\beta$ frames per second depending on the size of the queue. The inter-arrival rate in Figure 5 is inherited from the departure rate of the previous queue. There is a direct relationship between the two because the frame outflow of the M/D/1/K queue forms the inflow into the D/D/1/K queue.
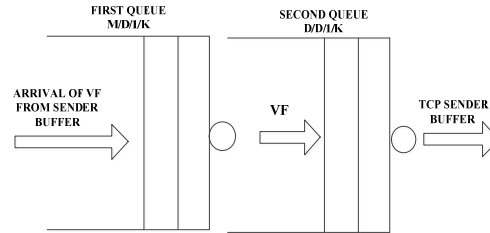
**Figure 2: Network of queues modeling of
the proposed scheme**

Compressed video frames from the application layer sender buffer arrives into the first queue at higher rate but the rate at which the frames flow into the second queue are as follows:

Case 1: When current first queue size, $a(t)$, approaches and above 75% capacity utilization, then current first queue outflow rate $r_y(t+1) = r_y(t) * \zeta$, where ζ belongs to the set of positive real numbers.

if $(a(t) \geq 0.75)$ then

$r_y(t+1) = r_y(t) * \zeta$

Case 2: When current first queue size, $a(t)$, operates below 25% capacity utilization, then current first queue outflow rate is halved i.e $r_y(t+1) = r_y * \gamma$, such that $0.0 > \gamma < 1.0$

Case 3: Otherwise, the flow rate is set at the default.

Assumptions:

Video frames inflow into First queue is markovian utilizing the H.263 Standard. Second queue size is assumed larger than the First queue i.e $(b \gg a)$.

List of terms

c_a- number of frames in first queue at time t

c_b-　　number of frames in second queue at time t

r_x:　　rate of Inflow to first queue

r_y:　　rate of Inflow to second queue

r_z:　　rate of Inflow to the TCP sender buffer

t:　　time(s)

a:　　size of First queue (number of video frames)

b:　　size of Second queue (number of video frames)

Input : uneven arrival of video frames, Output: even departure of video frames,

Initialization: $c_a, c_b, r_y, r_x,$ out : array $[1,\ldots,n]$.

$r_x \leftarrow$ H.263 trace data

begin

$c_a(i+1) = c_a(i) + r_x(i);$
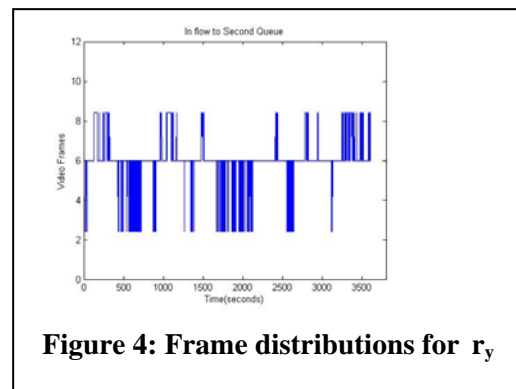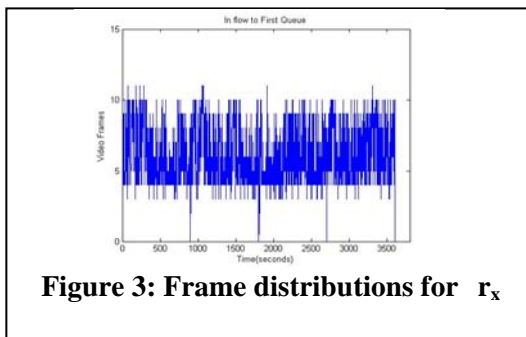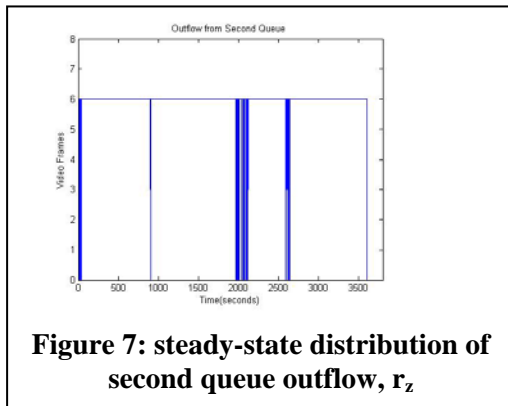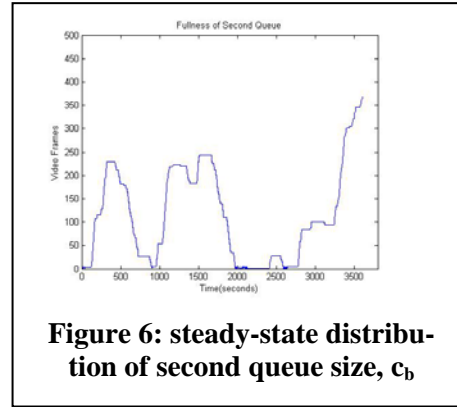
$if(c_a(i) > 0.75 * a)$

$r_y(i+1) = r_y(i) * \zeta;$

```
            elseif(c_a(i)< 0.25*a )
              r_y(i+1)= r_y(i)* γ;


          else
              r_y(i+1) = 10;
    end
    if(c_a(i+1)>0)
              c_a(i+1) = c_a(i+1) - r_y(i+1);
    end
    c_b(i+1) = c_b(i) + r_y(i+1);
      if(c_b(i+1)>r_z)
              out(i)=r_z;
              c_b(i+1) = c_b(i+1) - r_z;
    end
    continue
```
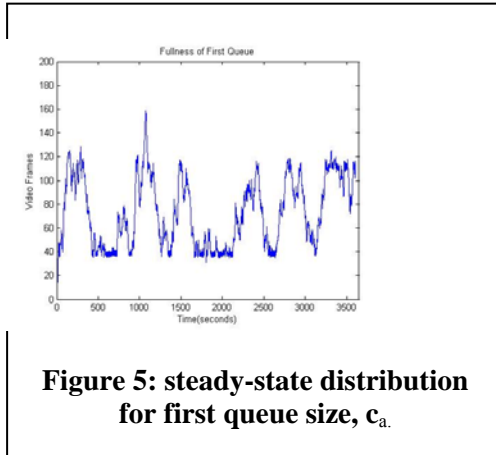
Results and Discussion

The algorithm was implemented in Matlab using H.263 video trace file obtained from (http://trace.kom.aau.dk/trace/tut/html). The data file is a one-hour video session in H.263 format composed of a total of 25,432 frames. Arrival rates into the first queue was set to $r_x \epsilon [1,…, 15]$. At steady-state, we observed that the first queue size oscillates between 32-160 video frames with ≈ mean of 6 frames and variance 3.5 as depicted in Figure 5. Frame arrival rate into the first queue was markovian, with a mean value of $r_x = 6$ and variance = 3.5 as shown in Figure 3, but the arrival rate into the second queue was found to be 6 and 1.4(mean and variance) for r_y respectively as depicted in Figure 4. The variability of the video data has been reduced by more 50%. Also, we observed that the second queue size oscillates between 0-250 frames for about 90% of the simulation period and peaked at 370 as shown in Figure 6. This result shows that the size of the second queue should be about twice the first queue for optimum performance. However, we are not unaware of the impact of mean packet delays which will be fully reported in subsequent papers.

Figure 3: Frame distributions for r_x

Figure 4: Frame distributions for r_y

Figure 5: steady-state distribution for first queue size, c_a.

Figure 6: steady-state distribution of second queue size, c_b

Figure 7: steady-state distribution of second queue outflow, r_z

The outflow of the first queue presents interesting results by curtailing the variability in the inflow into the second queue to a great extent. The simulation result gave a mean and variance for r_z as 6 and 0 respectively as shown in Figure 7. These results confirmed our hypothesis.

Conclusion

In this paper, a Double-Buffer traffic shaper for rate adaption of real-time video transmission in a slow-speed network over TCP was presented. The proposed scheme attempts to eliminate the uneven distribution of video frames arrival from the application layer sender buffer and produces even distribution before the video frames are injected into the TCP sender buffer. For further work, the birth and death process will be analysed for steady-state dynamics. A packet-level modeling and simulation will be carried out in NS3 with the aim to carry out performance analysis of average packet delay, jitter and bit-error rate.

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