Streaming video using dynamic rate limiting and TCP congestion control

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Abstract-

We present a new Proposed Algorithm for streaming real time video on today's Internet, based on dynamic rate limiting and TCP congestion control.

Dynamic rate limiting is a technique that adapts the rate of compressed video (MPEG-1, MPEG-2, H.26x) to dynamically varying bandwidth constraints. This provides an interface (or filter) between the source and the network, with which the encoder's output (Either live or stored) can be perfectly matched to the network's available bandwidth.

Keyword: Quality of service, congestion control, report only congestion losses, streaming

1. Introduction

Video streaming refers to the real time transmission of stored video. There are two ways to deliver the video over a packet switch network i) File download ii) Streaming.

With file download the entire video is downloaded to the user’s terminal before the playback commences. The advantage of file download is that it is relatively simple and ensures the high video quality. The drawback of file download is the large response time, typically referred to as start-up-delay. With video streaming, on the other hand, playback commences before the entire file is downloaded to the user’s terminal. In video streaming typically only a small part of the video ranging from a few video frames to several hundreds or thousands of frames are downloaded before the streaming commences. The remaining part of the video is transmitted to the user while the video playback is in progress.

2.Quality of service

In the fields of packet switched and computer networking, the traffic engineering term **Quality of Service** (**QoS**, pronounced "que-oh-ess") refers to the probability of the telecommunication network meeting a given traffic contract, or in many cases is used informally to refer to the probability of a packet succeeding in passing between two points in the network.

The current internet which offers best effort service does not offer any quality of service (qos) guarantees to streaming video. Two approaches have emerged to tackle this problem. One approach is to design the new protocols and router scheduling disciplines to provide the desired performance guarantees. But these mechanisms are not expected to be available in very near future. Another approach is to make the application adapt the packet rate according to the state of the network, the objective being to limit the packet rate to capacity of the network. This is achieved by adjusting the output rate of the video coder through the adjustment of parameters inside the video coder. The above mechanism is known as “rate control mechanism” [3].the objective of the “rate control” is to avoid congestion and to maximize quality in the presence of packet loss.

3. Dynamic Rate Limiting-Rate Control is employed to adapt the output rate of the video coder based on the estimated available bandwidth in the network.
Computer Network Rate limiting is the function of controlling the maximum rate of traffic sent or received on a network interface. PROPOSED Rate limit control is dynamic rate control system.
In PROPOSED Rate limit control, sender will send the packet to the network according to the PROPOSED Algorithm. This algorithm is based on the principle of fuzzy system i.e. increase or decrease in packet sending rate will occur smoothly. Here we will take L as limiting value of the Rate with which we can send the packet from sender on the network. Initially, we take the value of L as the half of the available bandwidth (BW).here we use p (Packet loss event ratio experienced by connection) in making adaptation decision.

4. Effect of Loss Rate on Rate Control

Source-based rate control mechanisms employ feedback to adjust the output rate of the video coder according to the state of the network. The Internet infrastructure typically doesn't provide sources of traffic with explicit feedback information. The only easily available information is implicit information such as measures of losses and/or round-trip delays. Experiments and simulations have shown that control schemes which use packet delays as feedback information can't compete with TCP like traffic which use loss-based feedback [5]. Hence, most rate control algorithms (like AIMD, TFRC, etc.,) choose packet losses measured at the receivers as feedback information.

In probe-based rate control schemes like AIMD, if the fraction of packets during a feedback interval p is above some tolerable limit, the maximum output rate of the coder is decreased by a constant factor β otherwise the maximum output rate is increased by adding a constant α. Thus, the control algorithm is as follows:

```
If (p > tolerable_loss)
        Max rate = max rate – maxrate / β
Else
        Maxrate = maxrate + α
```

Here, we have developed the Rate control algorithm (PROPOSED rate control algorithm),which uses packet losses measured at the receivers as feedback information.

This algorithm works on the principle of Fuzzy system i.e. Increase and decrease in packet sending rate will occur smoothly. It assumes that if network was not congested in the previous interval, then the chances of loss rate to become very high in immediate interval is very less.

Here, L is the output rate of the video coder and it is a variable quantity.

Initially we take, $L = BW/2$

The loss ratio p will vary from 0 to 1. Here 0 represents no loss and 1 represents 100% packets loss. We divide this range into ten equal parts. We change the value of L according to the value of p received in the previous interval. Thus, the PROPOSED control algorithm is as follows:

Proposed_Algorithm (p)

```
{
p=p*10
Switch ( ┌ p ┐ )                    // here we take the ceiling value of p
{
   Case 0: L=L + (L/10)
           Break ( )
   Case 1: L=L - (L/10)
           Break ( )
   Case 2: L=L - (2*L/10)
           Break ( )
   Case 3: L=L - (3*L/10)
           Break ( )
   Case 4: L=L - (4*L/10)
           Break ( )
   Case 5: L=L - (5*L/10)
           Break ( )
   Case 6: L=L - (6*L/10)
           Break ( )
  Case 7: L=L - (7*L/10)
           Break ( )
  Case 8: L=L - (8*L/10)
           Break ( )
  Case 9: L=L - (9*L/10)
           Break ( )
  Case 10: L=L - (10* L/10)
           Break ( )
}
```

5. Overview of the Problem

In a heterogeneous wired/wireless network scenario where streaming server is located on the wired network and the client/receiver is located on wireless network, due to the different packet loss characteristics of the two networks, the loss rate reported by the receiver may not be correct indicator of the congestion in the network. Hence, rate control schemes which employ the loss rate as their principle feedback parameter inaccurately estimate the state of the network and respond by decreasing their output rate assuming congestion. This effect the quality of the video delivered to the client. Especially during bad wireless channel conditions, when the error rate is high due to bursty errors, the loss rate reported will be high and will drastically affect the quality of video. The above problem occurs due to two prime reasons.

1. The inability of the receiver to distinguish between packet losses due to congestion in the network and wireless channel errors. As a result of this, the receiver reports the total loss rate experienced which may include both congestion and wireless losses.
2. The sender side rate control relies mainly on the loss rate reported by the receivers which may not be accurate in a heterogeneous network environment.

6. The Solution Schemes(s)

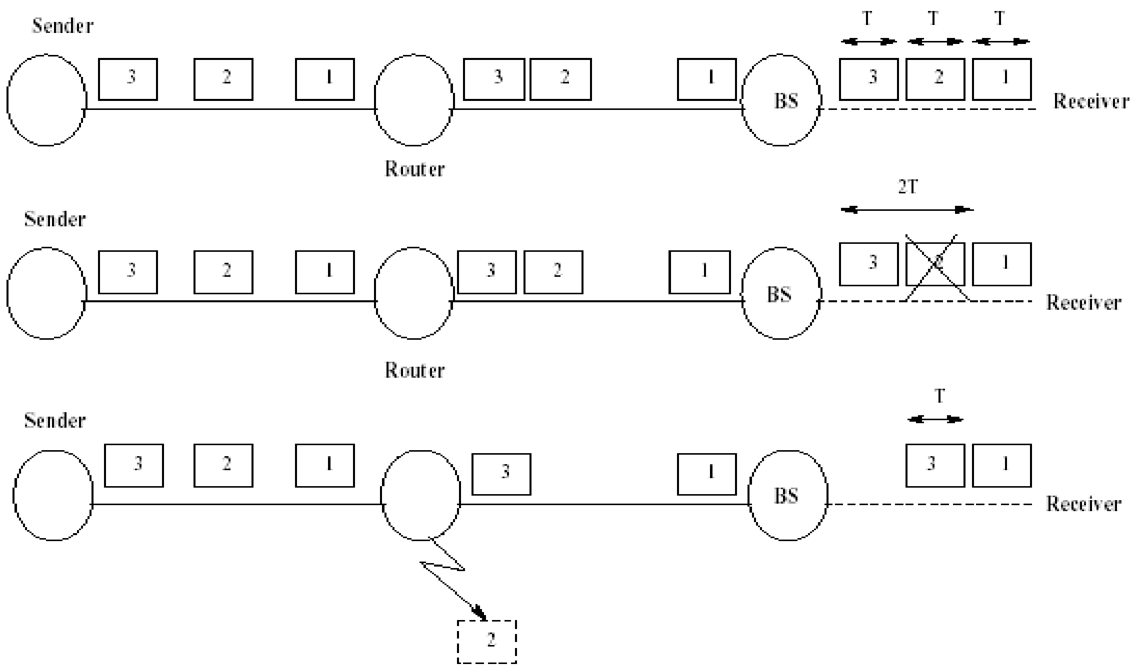
The main objective of rate control is congestion control. In order to do this, a class of rate control schemes employs the packet loss rate reported by the receiver as an indicator of congestion in the network. The packet loss experienced by the receiver may be due to congestion in the network and/or bad channel conditions. Since the receiver cannot distinguish between them, the loss rate reported may not be an exact indicator of network congestion. This is the primary reason which leads to the problem defined. Two schemes which are proposed to alleviate this are:

1. Report only congestion losses (ROCL)
2. Report correlation of loss and delay (RCLD)

Here we describe only ROCL and try to show that if we combine ROCL with PROPOSED algorithm, its performance will be improved.

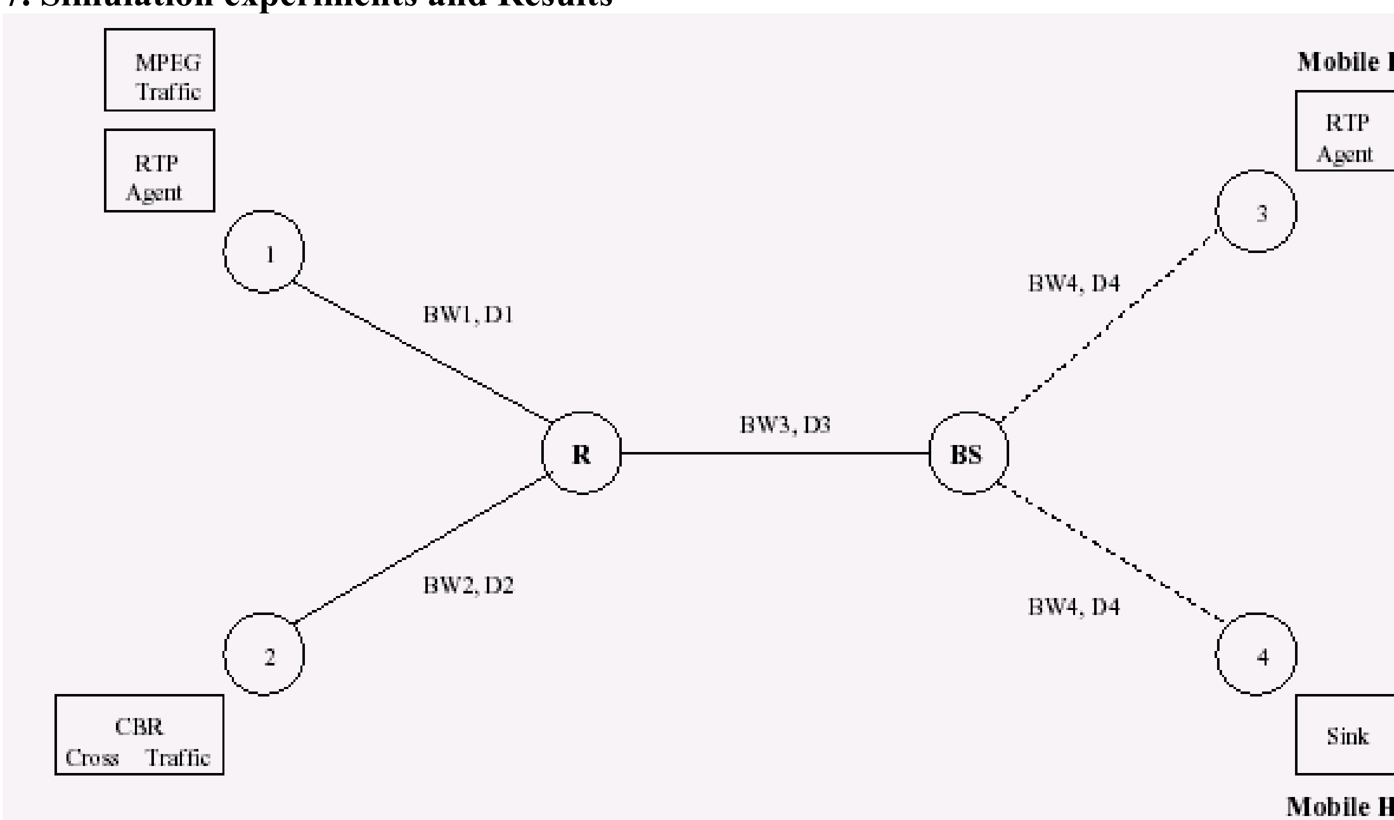
ROCL-

Rate control algorithm at the sender will now start responding only to congestion in the network which is desirable. In this scheme, the receiver side algorithm used for detecting packet losses is altered to enable it to distinguish the packets lost due to congestion and wireless transmission errors. Saad Biaz and Nitin H. Vaidya have proposed a heuristic in [2] to discriminate congestion losses from wireless losses at the receiver.



(From Nitin Vaidya et al, Discriminating Congestion Losses and Wireless Losses Using Inter−arrival times at the Receiver)

7. Simulation experiments and Results



Simulation model

In the experiment, the network is set in an uncongested state so that the only losses which occur are due to wireless channel errors. Accordingly, the bandwith the propogation delay of the links have been initialixed to the following values:

BW_1 = BW2 = 1Mbps, $D_1 = D_2$ = 2ms
BW_3 = 256kbps, D_3 = 10ms
BW_4 = 64kbps, D_4 = 1ms

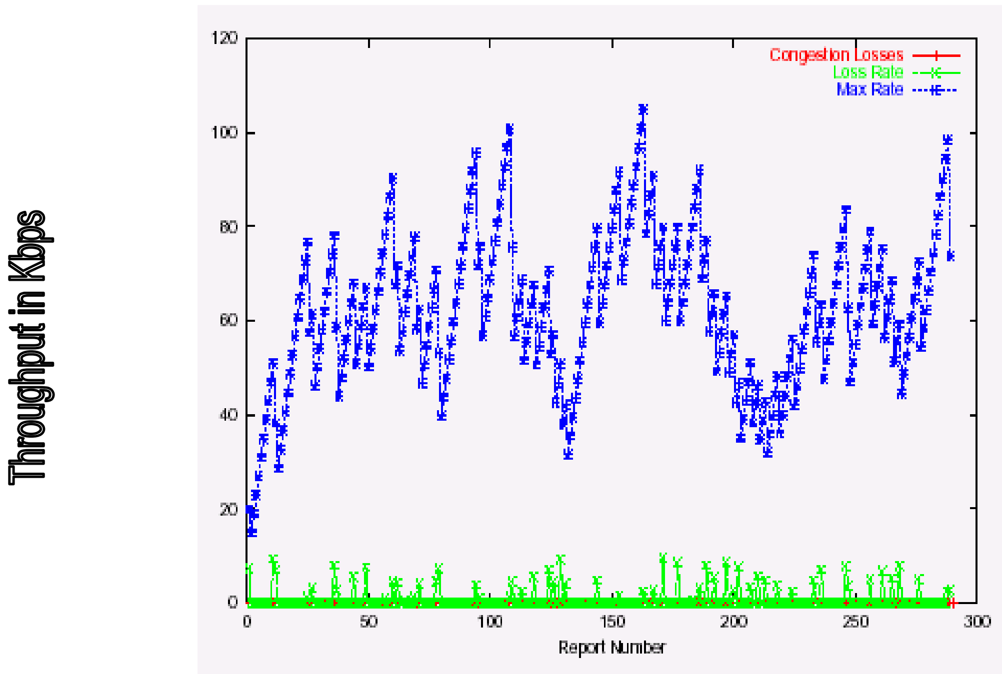


Figure 1: Original Scheme without Modification

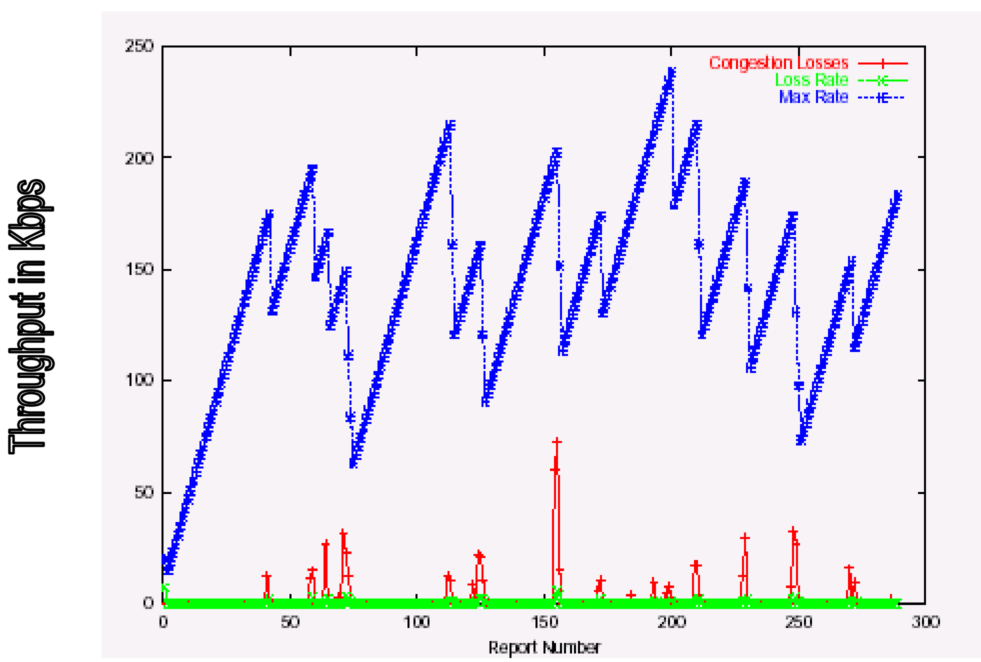


Figure 2: With ROCL Modification

8. Performance graph of PROPOSED Algorithm

Suppose we take the bandwidth BW= 256 Kbps.

So, L = BW/2= 128 Kbps.this chart shows that value of L (i.e. rate of sending the packet) is changing smoothly. If we combine the PROPOSED Algorithm with ROCL, the graph of Max rate will not fluctuate very much. Consequently loss rate reported by wired network will also be reduced.

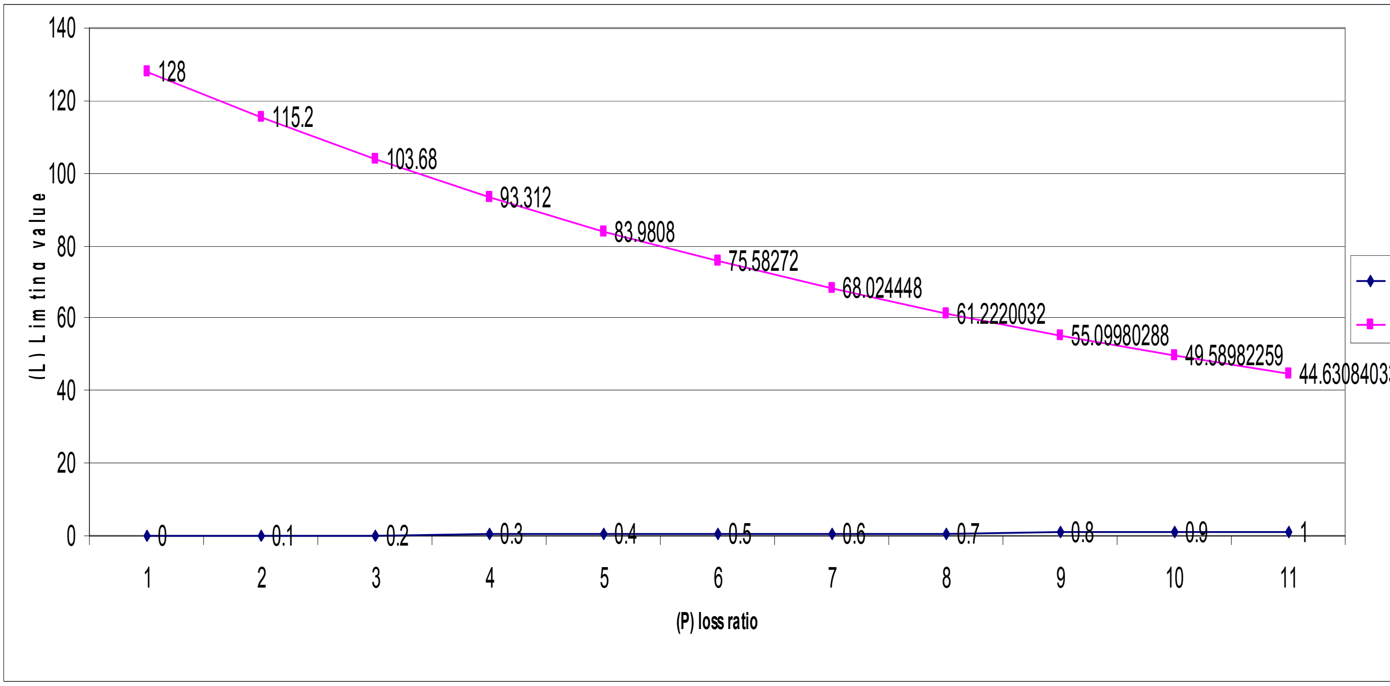


Fig 5.9: performance with PROPOSED modification

The following table shows the value of L calculated by PROPOSED Algorithm.

P	L
0	128
0.1	115.2
0.2	103.68
0.3	93.312
0.4	83.9808
0.5	75.58272
0.6	68.02445

0.7	61.222
0.8	55.0998
0.9	49.58982
1	44.63084

9. Future Work

Model based schemes like TFRC use equation, $\lambda = 1.22 * MTU / (RTT*\sqrt{P})$ to adjust their output rate. In a wired/wireless heterogeneous network, the loss event rate p may not be a good parameter to use. Hence, one possible direction of future work is to investigate appropriate function to replace loss event rate p in model based schemes.

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