A Novel Congestion Coǿtrol Technique for Mpeg - 4 Video streaming in wireless Networks

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Abstract

Transmission Control Protocol (TCP) is rarely chosen as the transport protocol for video streaming applications, mainly because of its poor congestion control capability. Recently, TCP Friendly Rate Control (TFRC) protocol and its variants are used for the congestion control in MPEG -4 video streaming. This paper proposes a novel congestion control algorithm that enhances the TFRC Veno protocol. It decouples the wireless loss from that of the congestion loss based on the queuing delay in the routing buffer. The queue length is estimated through the Round Trip Time (RTT) measurements. This algorithm is simulated under various congestion levels of WLAN. Significant improvement in Throughput, Packet delivery ratio, delay, jitter and fairness index are found.

Keywords: Congestion loss, loss differentiation, MPEG - 4 video streaming, TFRC, wireless loss

1. Introudction

Video has been an important media for communications and entertainment for many decades. Its transfer requires larger bandwidth to maximize its throughput. But a higher data rate of transmission creates congestion in the network and drops the network throughput drastically. Hence to provide better congestion control for MPEG-4 video transmission, a new mechanism has been proposed.

The proposed algorithm is an enhancement of the TFRC Veno protocol. It uses a technique called Loss differentiation (LD) to differentiate between wireless link loss and congestion loss [1]. It takes into account the number of packets in each connection and compares it with a threshold. The factor by which the Congestion Window (CW) size gets reduced depends on, whether the number of packets in each connection is greater or lesser than the threshold mentioned. The loss is differentiated by continuously monitoring the RTT and the Queue Size and making decisions based on the Queue size, whenever the packet losses occur. The MPEG-4 encoded data is transmitted over the network using the proposed transport protocol. This technique provides better Quality of Service (QoS) for MPEG-4 traffic [2][3]. The proposed scheme provides better throughput, fairness index and packet loss characteristics when compared to other available transport protocols.

2. Related Work

TCP Reno, the most widely used TCP variant has the fast recovery mechanism that enables the connection to quickly recover from the isolated segment losses. The fast retransmit is triggered upon the receipt of three duplicate ACKs. However, when a partial or new ACK is received, Congestion Window (CW) is set to Slow Start Threshold, causing the system to immediately enter the congestion avoidance phase. It also supports *header*

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prediction and *delayed acknowledgments* options. Therefore, the key advantage of TCP Reno over other TCP Protocols is that it maintains the clocking of new data with the duplicate ACKs and avoids initiating the slow start phase, whenever the transmission rate gets reduced [4]. However, Reno does not work well if multiple packets are dropped during a single round trip.

The other popular TCP variant, TCP Veno is a novel end-to-end congestion control scheme which can improve TCP performance quite significantly over heterogeneous networks, particularly when wireless links form part of such networks [5]. This scheme significantly reduces blind reduction of TCP window regardless of the cause of packet loss. Its main feature includes only simple modification at the sender side.

The congestion control mechanism generally used for unicast flows under best-effort environment is TFRC. It is reasonably fair when competing for bandwidth with TCP flows, but has a much lower variation of throughput over time compared with TCP. It is suitable for applications such as telephony or streaming media, where a relatively smooth sending rate is of importance. Moreover, TFRC is designed for applications that use a fixed packet size, and vary their sending rate in response to congestion. And TFRC is a receiver-based mechanism, with the calculation of the congestion control information (i.e., the loss event rate) in the data receiver rather in the data sender. Hence, it requires more memory and CPU cycles for computation at the receiver and large server with more number of concurrent connections at the sender. The variants of TFRC such as ECN based TFRC, Proxy-based TFRC, WM-TFRC and AED based TFRC also have congestion control mechanisms. TFRC Veno protocol makes use of TCP Veno equation to solve the TFRC wireless suffering problem with 300% improvement over TFRC at 10% loss rate. It needs to modify only the sender side protocol. The calculation of CW size and the modification of sending rate in TFRC Veno are similar to TCP Veno.

MPEG-4 is an object based open standard [6] [7] [8] that can easily deploy multi media content for all platforms. It advances audio and video compression by enabling the distribution of content and services from low bandwidths to high definition quality across broadcast, broad band, wireless and packaged media. All the coders in MPEG-4 are optimized for the appropriate data types. MPEG-4 encodes information about the scene composition in a separate stream with in the video bit stream.

3. Proposed Algorithm

The proposed algorithm is an extension of TFRC Veno protocol. It does not rely on any of the quality assessment functions of the routers such as *Random Early Detection* (*RED*) [9], *Explicit Congestion Notification* (*ECN*) [10] [11] or other *Active Queue Management* (*AQM*) mechanisms [12]. It employs a receiver centric technique and operates on top of any transport protocol. The receiver uses the control packets to send the feedback of reception statistics to the sender. These packets are effectively used to determine the bandwidth and RTT to regulate the transmission rate. The receiver understands the congestion either by packet loss or packet reordering. In the event of packet loss, the receiver observes the current RTT and subsequently determines the nature of loss. The RTT for the fully utilized bottle neck channel is given in Eqn. (1)

$$RTT = RTT_{min} + q_{delay} \qquad (1)$$

RTT_{min} = Round Trip propagation delay.

q_{delay} = Total queuing delay in all buffers across the network path.

When the queue copies the whole bottleneck buffer, then the maximum RTT is

$$RTT_{max} = RTT_{min} + q_{delaymax} \qquad (2)$$

From the equations (1) and (2) which are discussed in [13], it is understood,

$$q_{delay} = RTT - RTT_{min} \qquad (3)$$

$$q_{delaymax} = RTT_{max} - RTT_{min} \qquad (4)$$

The current value of RTT is monitored to calculate the queuing delay. Practically, if RTT is close to RTT_{min}, the loss is due to link error. On the other hand, if the measured RTT is larger than RTT_{min} and close to RTT_{max}, then it indicates the occurrence of congestive loss. Following these observations, the TFRC Veno's congestion control is complemented with the following Loss Differentiation algorithm (LDA).

Upon detection of packet loss, the transmission rate is decreased multiplicatively with the standard decrease ratio β, only when the following condition is satisfied.

$$(q_{delay} / q_{delaymax}) = (RTT - RTT_{min}) / (RTT_{max} - RTT_{min}) \geq q_{thresh} \qquad (5)$$

The threshold value q_{thresh} specifies the point of queue length, where the packet loss is considered to be congestion induced. This value is adjusted experimentally. Hence, when the queue occupies less than half of the buffer size, packet drops do not trigger congestion control mechanism. Certainly, this value can be adjusted differently in order to modify the protocol's error recovery strategy. The figure 1 shows the relation between RTT and q_{delay}.

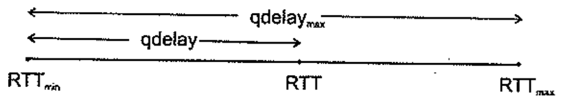

Figure 1: Relation Between RTT And Q_{delay}

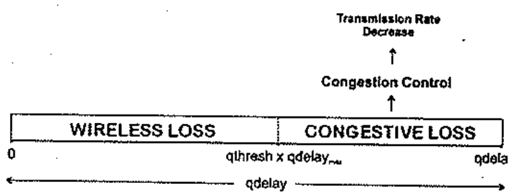

Figure 2 : Differentiation of Losses

2 shows the classification of type of loss with the values of qthresh and qdelay. And correspondingly congestion control can be performed by decreasing the transmission rate by a factor.

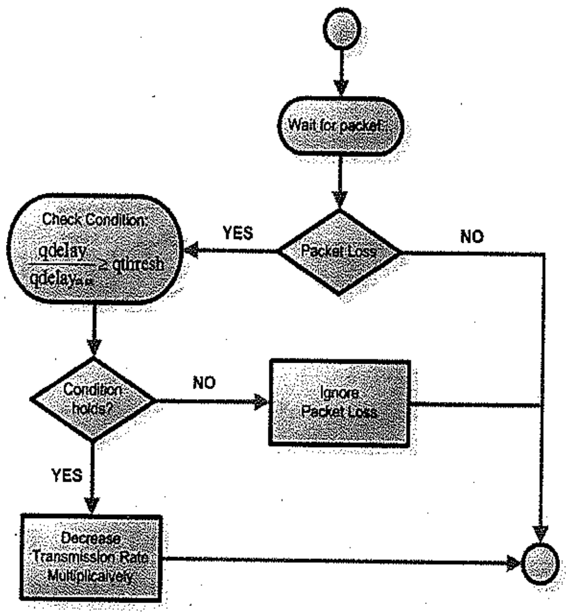

Figure 3 : Proposed Scheme

At sender: 1. MPEG4 Video packets are transmitted at a certain rate.
2. If acknowledgement is received for each packet then the rate is not modified.
3. If negative acknowledgement is received then determine the type of Packet loss that has occurred.
4. If packet loss is due to congestion reduce the rate, else, do not reduce the rate.

At Rcvr: 1. Receive the MPEG4 packets continuously.
2. Send acknowledgements for all the packets.
3. If packets are not received in sequence, wait for the missing packets for some time
4. If packets are lost, then send negative acknowledgement.

Figure 4: Pseudo Code for the Proposed Scheme

Figure 3 shows the flow diagram and figure 4 shows the pseudo code for the proposed scheme.

4. Performance Analysis

The proposed algorithm is simulated in Network Simulator (NS2:V2.31) [14] and its performance is analyzed with MPEG-4 traffic. The topology given in [15] is used for simulation and is shown in figure5.

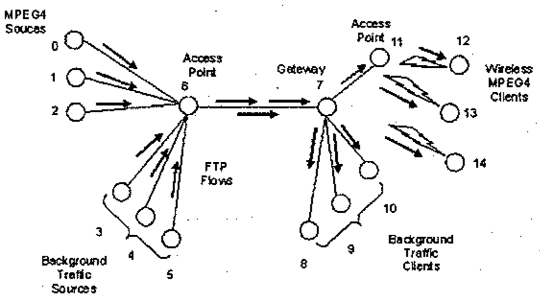

Figure 5 : Simulation Topology

The simulation parameters are shown in Table 1. The queue drops packet, when it is full. FTP traffic is generated to create congestion in the network and is made available throughout the simulation time of 500 mS. the increase in the number of FTP clients and servers increase the congestion and reduce the Throughput of video traffic. The frame rate of the MPEG – 4 traffic is by default set to 30 frames per second and it can be controlled manually. The value of qthresh is computed through experiments as 0.8. More over, MPEG-4 Flow 1 is the traffic from Node 0 to Node 12. MPEG-4 Flow 2 is the traffic from Node 2 to Node 14. MPEG-4 Flow 3, the traffic from Node 1 to Node 13.

Table1 : Simulation Parameters

PARAMETER	VALUE
Adhoc Routing	DSDV
Radio Propagation model	Two way ground model
Interface Queue type	Drop tail/ Priority Queue
Interface Queue Length	50 packets
Link Layer type	Traditional Link Layer (LL)
Antenna Model	Omni-directional (Unity gain)

A. Throughput Analysis

The figure 6 shows the throughput Vs time for various threshold values. For both the extreme values, the throughput is moderate. It was found that for q_{th} =0.8, the throughput was maximum. Therefore, with that value of q_{th} =0.8, the throughput has been analyzed for different

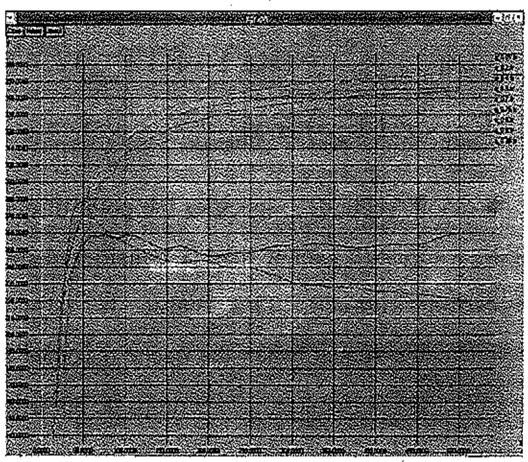

Figure 6 : Throughput Vs Time For Various Threshold For MPEG-4 FLOW 3

algorithms (TFRC, TFRC Veno, Proposed) and are compared in figure 7.

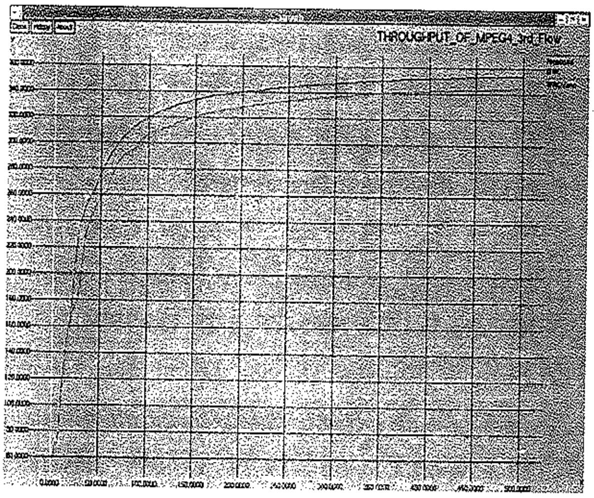

Figure 7 : Analysis of Throughput Vs Time For MPEG-4 FLOW 3

B. Delay and Jitter Analysis

Figures 8 shows the comparison of Jitter values among the proposed and other TCP friendly protocols. Similarly figure 9 shows the comparison of delay values of the same. It is clear that the proposed algorithm has relatively less jitter and delay than the other protocols.

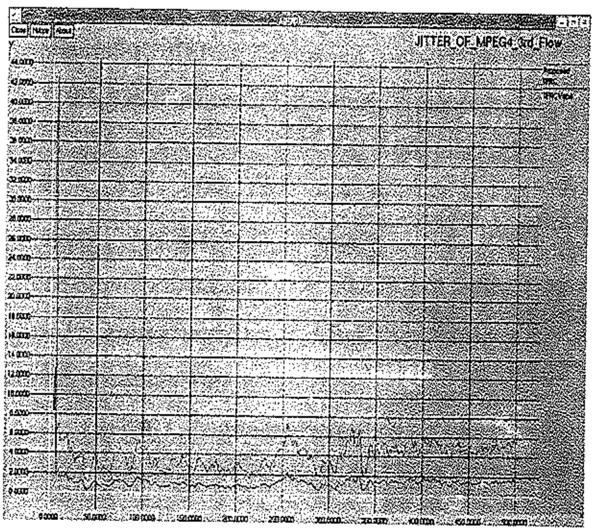

Figure 8 : Analysis of Jitter Vs Time For MPEG-4 FLOW 3

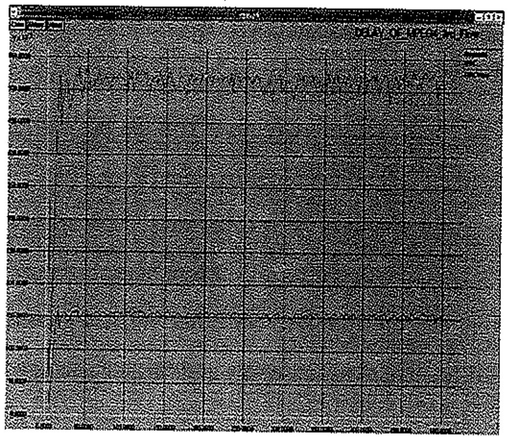

Figure 9 : Analysis of Delay Vs Time For MPEG-4 FLOW 3

C. Fairness Index Analysis

The table 2 shows the comparison of fairness Index for the TCP friendly protocols along with the proposed algorithm for different number of MPEG -4 servers. The fairness index of the proposed algorithm is better than other protocols.

Table 2 : Comparison Of Fairness Index With Various MPEG-4 Flows

No. of MPEG-4 SERVERS	FAIRNESS INDEX		
	Proposed	TFRC Veno	TFRC
2	0.93604	0.91722	0.6652
3	0.85527	0.80649	0.6566
4	0.70052	0.68915	0.6195
5	0.40242	0.39808	0.4043
6	0.40084	0.40624	0.3868

5. Conclusion

The proposed congestion control protocol is simulated for a specific topology of WLAN and the results are compared with the basic protocols of TFRC and TFRC Veno. The simulated results prove that, by incorporating the RTT to be a factor to increase or decrease the sending rate, there is a significant improvement in throughput of MPEG -4 video stream. And also the delay and jitter show a significant improvement. But the protocol does not show any significant improvement over TFRC Veno under wired networks. This is because, there are no link error losses in wired networks. More over, in future this protocol can be simulated for other WLAN topologies and Ad-hoc wireless networks and further analyzed.

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