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Extending the Real Rime Congestion Control Mechanism for Multicast Applications over 3G Wireless Networks

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Abstract— In this paper we introduce a real time congestion control mechanism to multicast the multimedia applications. The key challenge in the design of this mechanism is to calculate the rate of feedback messages by various receivers; appropriate feed back suppression, scalable round trip time measurements & ensuring that feedback delays in the control loop do not adversely affect fairness towards competing flows. Major contributions are the feedback mechanism and provide an end-to-end multicast congestion control schemes. We improve the well-known approach for multicasting, by using exponentially weighted random timers by biasing feedback for low-rate receiver and preventing the response implosion. We evaluate the design using simulation and demonstrate the results for multicast transmission of multimedia data.

Keywords- GECM; ROTT; Congestion; ECN; TFMCC; TFRC; RBRC.

I. INTRODUCTION

Congestion control is an integral part of any internet data transport protocol. Congestion control over network, for all types of media traffic, has been an active area of research in the last decade. A variety of network applications exist for streaming the media either in real time or on demand. The numbers of users for these network applications is continuously growing hence resulting in congestion. As the development of wireless networks increases more and more applications based a wireless multimedia networks are increased. By increasing the multimedia traffic on networks traffic on networks there is a need to increase the performance of wireless networks. A number of researches have been made to increase the efficiency of these networks and reduces the congestion control for improving the speed of data transmission. TFMCC (TCP friendly multicast congestion control) is an equation based multicast congestion control mechanism, an extended version of TFRC protocol from the unicast to multicast domain. As the growth of wireless networks increased more and more applications based a wireless multimedia are increased. For congestion control an explicit Congestion Notification (ECN) [7] based congestion control algorithm controls sending rate by marking the IP Packets [4]. ECN is not so much feasible because when a successive packet loss happen it cannot control the loss rate properly. The other congestion control algorithm is end – system based source algorithm. It detects the nature of loss and identifies the packet loss types. This way it improves the efficiency and performance of network by relative one-way trip-time (ROTT) [5] or packet inter-arrival times [6]. All of these congestion control algorithms cannot work for sudden link blockages when the throughput drops heavily due to channel degradation or other wireless errors. The performance of all these algorithms depends upon network topologies and the number of flows. So if the sudden blockage or congestion occurs in wireless networks then these congestion control algorithms are insufficient to reduce the congestion loss and maintain the network performance. In multicast approach it is necessary that all recipients get the data without any loss. The existing congestion control algorithm for multicast applications is TFMCC, which belongs to the class of single rate congestion control schemes. We introduce a novel approach for congestion control in multicast applications our aim is to design a rate control mechanism that can adjust the sender’s behavior as per the congestion detection. When sudden congestion happens we can detect it and control the sending rate as per the congestion notification. By this we can better utilize the networks, increase the network efficiency, decrease the packet loss and increase the delivery rate of packets in the wireless networks.

In this paper, we propose a model which detects the sudden losses by compare the current RTT with an average one constantly. According to the congestion level and the current RTT value the sending rate is adjusted accordingly. We include the Gilbert-Elliot channel model for adjust the current sending rate-and reduce the congestion in network.

II. THE GILBERT- ELLIOTT CHANNEL MODEL

In wireless network there is always a sudden blockage or congestion occurs and this is not depending on the sending rate of the sender. The GECM model [2] use a RTT (round trip time) based rate control (RBRC) schemes, in which the sender can detect the sudden losses [1]. In GEC Model it is well known that the transmission errors occur in bursts, this means there is correlation between consecutive errors. The Gilbert Elliot Model [2] takes into account this correlation. This model is a two-state one order discrete time Markov chain shown in figure 1.

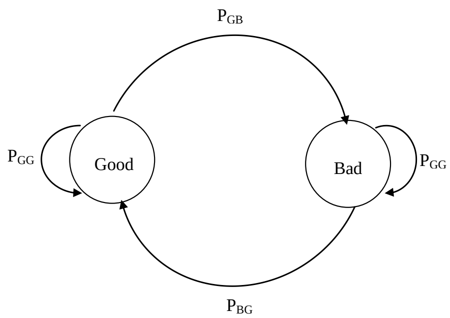

Figure 1: Gilbert Elliot Model

One state represents a good channel conditions and the other one bad channel conditions. Each state is assigned a constant Bit Error Rate (BER) probability, Pg in good state and PB in bad state. It is assumed that the bit errors occur independently from each other. The packet loss probability by the wireless channel is determined by the transition of states. PG determined the losses occur with lower probability when in “Good” state and PB determined the losses occur with higher probability. When the packet losses is less then the probability of transforming from a “Bad” state to “Good” state is PBG and PGB is the transition from “Good” state to “Bad” state. It the system remain in same sate then the steady state probability to remain in same “Good” or “Bad” state is PGG and PBB respectively. The GEC Model produces an average packet loss rate by equation (i).

When a channel is in Good state πG denotes the probability for it and πB denotes the probability of a channel in a “Bad” State. Equation (ii) and (iii) denotes this channel probability.

$$P_{avg} = P_B \pi_G + P_G \pi_B \quad (i)$$

$$\pi_G = \frac{P_{BG}}{P_{BG} + P_{GB}} \quad (ii)$$

$$\pi_B = \frac{P_{GB}}{P_{BG} + P_{GB}} \quad (iii)$$

III. DETERMING AN ACCEPTABLE SENDING RATE

We introduce o new approach to decide the sending rate of a sender according to the congestion notification of the network. In a network when the data flow over the network there is sudden blockage or congestion occur. This congestion occurs due to various reasons, such as channel error or a sudden link blockage [1]. So this proposed algorithm used for the long congestions loss and also short lasted congestions. These two congestions are differently treated. First we concentrate on long congestions, when congestion occurs we deal by very basic congestion control mechanism TFRC [9]. TCP friendly rate control (TFRC) is equation based [10] end to end rate control mechanism, it uses an equation as shown in equation (iv)

$$R_L = \frac{S}{RTT\sqrt{\frac{2P}{3}} + RTO(3\sqrt{\frac{3P}{8}})\, P(1+32P^2)} \text{........}(iv)$$

In this equation S is the s1ze of packet, RTT is the round rip time, P is the packet loss rate, and RTO is retransmission time. Here we perform a little modification, we are not calculate the packet loss rate P in equation (iv), instead of this P we calculate the loss event rate K. the loss event rate K is calculated indirectly by an averag0e loss interval $\hat{s}$. this $\hat{s}$ denotes the number of received packets between two immediate loss events and k=1/ $\hat{s}$. The control equation for throughput used by TFRC & TFMCC is derived as equation (v).

$$T_{TCP} = \frac{S}{t_{RTT}(\sqrt{\frac{2P}{3}} + (12+\sqrt{\frac{3P}{8}})P(1+32P^2)} \text{.......}(v)$$

This equation (v) is use for long term TCP through put in bytes / sec [12]. The expected throughput TTcp of a TCP flow is calculated as a function of the steady – state loss eventuate p, round trip time tRTT and the packet size S. If the sender does not exceed this rate for any receiver then it should be TCP friendly.

IV. ADJUSTING THE SENDING RATE

In multicast network sender continuously send the data to multiple receivers. To detect the congestion in network the sender continuously receive the feedback form the receivers. If the sender receive the feedback that indicates a rate that is lower than it’s current rate, the sender immediately reduce the sending rate for multicast network a message or data sent to multiple recipient in the network and the recipient send the feedback message to sender. So a large number unnecessary message got by the sender. In order to eliminate large number of unnecessary message the receiver not send the feedback message unless their calculated rate is less then sending rate. But how can the sender increase the transmission rate without feedback. So the current limiting receiver (CLR) used [3]. To get smoother sending rate certain modification perform in multiplicative increase and multiplicative decrease (MIMD) [11]. This MIMD play an important part over here in rate control process. We take congestion point as a new beginning of a congestion cycle (one more RTT time), the transmission end holds the sending Rate (R). If there is no congestion packet loss occur during the congestion cycle then the sending rate end's rate will be (a+1) × R and if congestion occur during the cannot cycle then the sending end's rate is (1-b) × R.

Every time the sending end's rate is depend on the congestion control cycle with different increase or decrease sending rate. When channel state is good then the sending rate increase quickly when we get full information about channel state we increase or decrease the sending rate accordingly. So generally we increase the sending rate when no congestion happens during 3RTT time while decrease the sending rate when congestion occur during a RTT.

So the rate control mechanism totally based on the congestion level. We use the channel state "Good" and "Bad" for congestion notification and the modified MIMD (a, b) congestion control mechanism. We set a = 1/5 and b=1/10 in simulation. If there is a packet loss, then by using equation (iv) of the TFRC congestion control algorithm we calculate the throughput. This throughput is used according to certain rule to calculate another sending rate.

V. TRANSMISSION CONTROL MECHANISM BASED ON RTT LEVEL FOR MULTICAST NETWORK

In multicast networks when sudden congestion occur. The current RTT is much larger than the normal level and it is at most about 4RTT time for multicast network we use the strategy of rate control mechanism [1]. In this mechanism the packet loss is first check the sender that is sent by the CLR. (Current limiting receiver.)

Sender record the RTT and update the queue. This RTT is checked will previous PTT of the current RTT is not much larger than average than there is not a packet loss and the transmission rate increase by (1+a) × R. If there is packet loss then the current transmission rate is decreased by (1-b) × R. This is shown in figure 2.

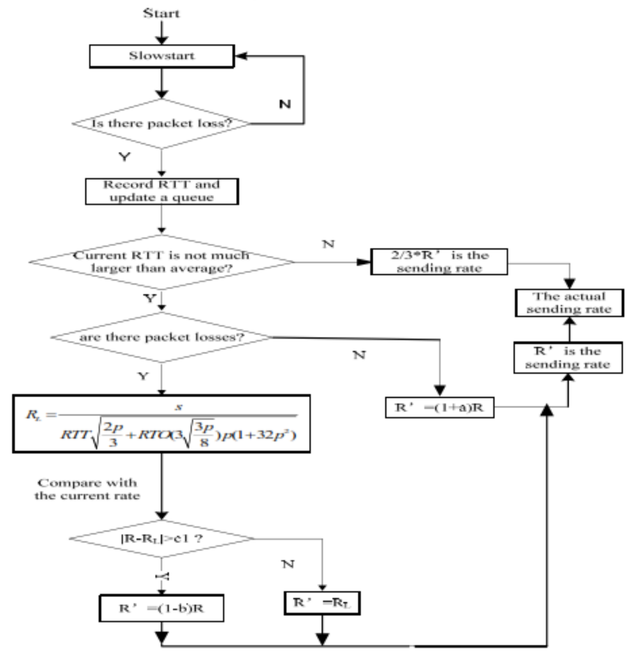

Figure 2: Flow chart for Transmission Control Mechanism

VI. SIMULATION RESULTS

We implemented TFMCC in the ns2 network simulator to get the simulation result. For sudden blockage or congestion we investigate the behavior of network under controlled conditions. In this paper we change the transmission rate according to the behavior of the network and the packet loss. We can only try to implement the new transmission rate at the time of congestion we use a drop tail queue at the routers to ensure acceptable behavior in the current wireless network.

A TCP flow is analyzed using the well-known single bottle neck topology for multicast network shown in figure 3.

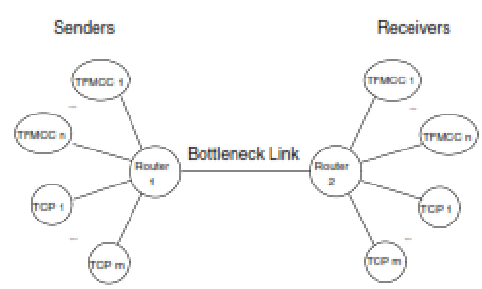

Figure 3: Topology

In this topology a number of sending nodes are connected to a number of receiving nodes through a common bottleneck. The throughput of a TFMC flows shows in figure (4). The average throughput of TFMCC closely matched the average TCP throughput. TFMCC achieve a smoother rate rather then TCP throughput.

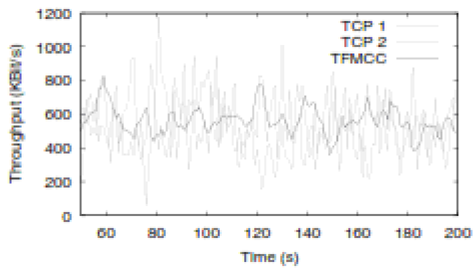

Figure 4: One TFMCC flow and 15 TCP flows over a single bottleneck

TCP sends the data packet back to back whether it can send multiple packets. This makes the TCP more sensitive to nearly full queues and for these full queues a drop-tail queue management is used, we use a wireless last hop network topology in which there are 3 wired nodes 2 wireless nodes and 1 base station node that spread over than 500 ×500 m2 . The defaults routing protocol is DSDV and the simulation runs over 500 seconds. We implement the Gilbert Elliot channel model in multicast network topology. In figure (5) we show the nodes connected by wired and wireless topology.

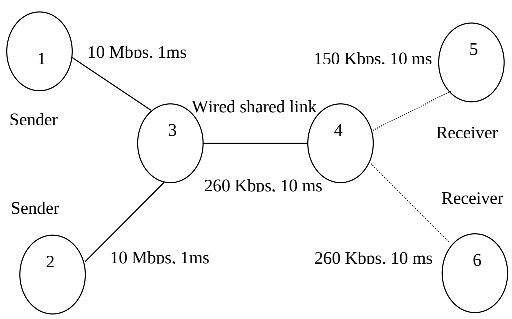

Figure 5. The wireless last hope network topology

The dotted line shows the wireless topology and other lines show the wired topology. The setting same as in [14] in our simulation.

We evaluate the network performance. It dynamically adjust the sending rate according to the congestion occurred in the network, achieve good network throughput and reduce the packet loss probability.

VII. CONCLUSION

In this proper we reduce the congestion and sudden blockage occur in the multicast network. It is more difficult to reduce the congestion in multicast network due to multiple receivers that get the packets simultaneously. So for better congestion control we implement the Gilbert Elliot model with MIMD implementation to reduce the packet loss in the network due to congestion. This reduces the packet loss in the network. In UMTS networks this sudden blockage or congestion happens time to time so this strategy is more useful for these networks. The congestion is easily detected by the current RTT with the average RTT to detect the packet loss. This average RTT got by the updating queue. According to the received RTT the sending rate of the sender is changed. The NS2 simulation shows the results that the proposed mechanism calculates the packet loss and detects the congestion effectively and adjusts the sending rate according to the congestion detection and improves the throughput.

VIII. FUTURE WORK

In this paper we implement the GCM [2] with Mimd [11] strategy. These both are implemented for the multicast networks. In multicast network it is critical to get the feedback from all the receivers and feedback is necessary to increase the sending rate of the sender. So a CLR is used to get the feedback from a slowest congested path. By this feedback the sender adjust the sending rate. In future we have to smooth the sending rate and also improve the strategy to calculate the RTT for the effective congestion.

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