Efficient Single Rate based Multicast Congestion Control

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ABSTRACT
Computer network is the essential part of the networking. Multicasting is used to send the information from one to group of receivers. It is a big issue because of more data demand of receivers is known as congestion. In this paper we are going to develop a single rate approach which provides congestion control. In this approach two congestion techniques are considered i.e. TFMCC (TCP Friendly Multicast Congestion control Technique). It has the problem of Feedback implosion. This problem was solved by AIMD (Additive Increase Multiple Decrease) approach. In this approach feedback is taken by the receiver which has more RTT (Round Trip Time). It is the slowest receiver approach. The number of packet loss is more and it does not provide the appropriate throughput. This problem is solved by LIMD (Logarithmic Increase and Multiple Decreased). In this approach feedback is provided by highest packet loss receiver. Two principal modifications are performed. First, each of the receiver (Highest packet loss) estimates its throughput based on a new equation derived according to the LIMD approach. Second, a hybrid rate-base preventive congestion control mechanism is implemented within the source. It improves in throughput and reduces the packet loss.

General Terms

Computer Network, Multicast Communication

Keywords

Congestion, Broadcast, Unicast, TFMC, LMD

1. INTRODUCTION
In the current scenario internet has been spread most all over the world. The number of users is increasing day by day. Transmission control protocol handles the 90% traffic over the internet [1, 15]. Video on demand as well as real time scenario applications has been increased from last few years [2, 13]. Video data consume more bandwidth in comparison to the audio as well as text data. If the data is transmitted from one sender to one receiver, it is called Unicast communication. On the other hand if the data is communicated from one sender to particular group of receivers then it is called multicasting. It is basically used to deliver the various information over the internet [3]. When the number of application increased then there is need of more bandwidth to handle the number of applications. When the number of transmitted packets exceeded the capacity of the receivers then the problem of congestion takes places. Due to the problem of congestion in the network, the performance of the network as well quality of data decreases. There are various approaches to handle the problem of congestion i.e. changing the route, increasing the bandwidth, changing the sending rate, increasing the buffer space etc [4].

There are various factors which creates the problem of congestion i.e. Bandwidth (it should be chosen so that whole data could be passed easily), Buffer space (the size of buffer should be appropriate so that limited amount of data could be stored to provide the particular delay), link failure (the whole data at that link could be lost), throughput (if the transmission rate exceed the capacity of receiving rate the it causes problem of congestion) [4] [5] [15].

On the other hand there are the problems of heterogeneity, scalability, efficiency and fairness. These are the problems which are solved by single rate based approach. It can work under different requirements. It provides the good network utilization as well solve the problem of starvation. It provides the efficiency i.e. good throughput and reduces the packet loss [6-9].

The paper is arranged in the following way: we begin in Section 2, providing the background of research work. In Section 3, we are proposing our research work. We present our results and discussion in Section 4 and finally conclude this paper in section 5.

2. BACKGROUND
There are two approaches to control the congestion in multicasting i.e. single rate and multi-rate approaches. In the case of single rate multicast congestion control all the receivers receive the data and the feedback is provided by the lowest rate receiver to adjust the sending rate. On the other hand in multi-rate approach different receivers receive the data at the different rate and feedback is provided based on different requirement of the different receivers. But between these two approaches single rate approach is better and simple but there is only drawback is that it doesn't fulfill the demand which are given in [2].

2.1 TCP-Friendly Multicast Congestion Control (TFMCC)
TFMCC [18] is the extend version of TFRC [17] (TCP friendly rate control). TFRC deals with the unicasting and TFMCC extends the features of unicasting into multicasting. It is basically a single rate based multicast congestion control technique. It is basically TCP friendliness i.e. if multicasting scenario deals with the bottleneck under TCP connections then at the receiver end all the data should reach with same delay and loss as TCP. Its main goal is to provide the responsiveness according to network changes. It basically improves the problem of feedback impulsion by selecting a current limiting receiver (CLR) which is the slowest, send the feedback to the source. Based on the feedback source adjust the sending rate. Basically throughput is calculated based on the TCP equation which is as follows:

$$X_{TCP} = \frac{S}{RTT\left(\sqrt{\frac{p}{3}} + \left(12\sqrt{\frac{3p}{8}}\right) p\left(1 + 32p^2\right)\right)}$$

Where p is loss ratio, S is the packet size, XTCP is throughput and RTT is round trip time. There are two problems with this approach, first one is, it is too slow to detect the CLR when the network condition changes. Another problem is if CLR is chooses wrongly then the performance of the whole system degraded [10][11].

2.2 Adaptive Increase Multiple Decrease (AIMD)

TFMCC [18, 16] is the extend version of TFRC (TCP friendly rate control). TFRC deals with the unicasting. It is basically an approach of ORMCC, which work on the principle of AIMD. In this throughput is Calculated when the congestion occurs. If there is congestion occurs then sending rate is multiplicatively decreases by the rate reduction factor β. The value of reduction factor β (0.65). It is an important parameter to avoid oscillations. If there is no congestion in the network then sending rate additively increases by the rate increasing factor α=S/RTT, where S is the packet size, and RTT is the roundtrip. Basically throughput is calculated based on the equation which is as follows:

$$x = \lim_{n \to \infty} x_n = \frac{\alpha}{p}\left(\frac{\beta + 1}{1 - \beta + p}\right)$$

These both are slow and complicated equation as well as difficult to provide scalability, fairness, heterogeneity and responsiveness. It doesn't provide good throughput and there is more packet loss [2][12].

3. PROPOSED WORK

In order to avoid the problems of TFMCC and AIMD approach, the feedback is provided by the highest loss receiver. If there is congestion in the network then the sending rate is logarithmically increased. Whenever, there is the congestion in the network then sending rate is multiplicatively decreased i.e. reduction factor β=0.65. It is basically multiplicative reduction factor. A new equation is developed to calculate the throughput to improve the performance of the system.

$$X = \beta \, LOG_2(1 + \alpha/\rho)$$

Where β: reduction factor, α=S/RTT, S: packet size, RTT: round trip time, ρ: loss event ratio, β= 0.65, ρ= 0.1 to 0.9.The second modification consists of monitoring delay, bandwidth and packet loss variations, in order to rapidly adjust the sending rate.3

Basically it is too complicated to develop the accurate method of congestion prevention. There are several approaches to measure the performance of the system and prevent the congestion. In this approach we are using the approach of congestion factor to prevent the congestion in the network.

To prevent the congestion, congestion factor is calculated based on following equation.

$$C_f = 1 - \Delta B/D$$

$$\Delta B = d_n - d_{n-1}$$

Where C_f: congestion factor, dn : delay for the current packet , dn-1 : delay for the current packet, d_{max}: maximum delay in the whole process, d_{min}: minimum delay in the whole process. Basically the source adjusts their sending based on the following rule.

$$D = d_{max} - d_{min}$$

$$X_{n+1} = X_n * C_f$$

X_{n+1}: throughput after congestion, X_n: throughput before congestion. If the value of ΔB is greater than zero it means that sending rate is decreased on the other hand if the value of ΔB is less than zero it means that there is improvement in the sending rate.

4. RESULT AND DISCUSSION

The research work has implemented in NS-2 [19] and simulation parameters are given in table 1. We have used multicast topology which is used to generate the result as shown in Fig. 1.1. There is a sender and four receivers. It is basically a wired connection topology. We have also test while using the network topology represented in Fig, described by links of varying capacities. All the simulation runs assumed in a multicast session.

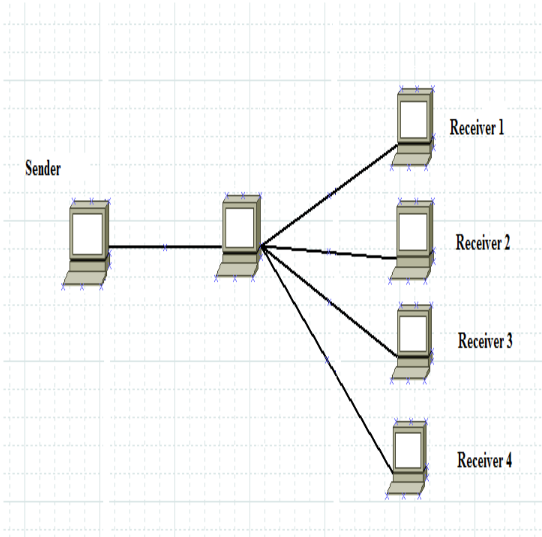

Fig. 1.1: Experiment Topology

Fig.1.2 shows TFMCC throughput performance for different values of packet loss ratio varying between 0.1 and 0.9.The time axis is in seconds. However, the magnitude of throughput fluctuations are higher with value of β=0.65.

Table 1: Simulation Parameters

Parameter	**Explanation**	**Value**
S	Packet size	100-600 (Bit)
p	Loss event ratio	0.1-0.9
β	Reduction factor	0.65-1.5
Queue	Queue type	RED, Drop-Tail
B.W	Bandwidth	1.5-10 (Mb)

W.S	window size	15 (Bit)
α	Increasing factor	0.5-0.9

Fig. 1.3 Shows AIMD throughput performance for different values of packet loss ratio varying between 0.1 and 0.9.The time axis is in seconds. However, the magnitude of throughput fluctuations are also higher with value of β=0.65.

Fig.1.4 shows LIMD throughput performance for different values of packet loss ratio varying between 0.1 and 0.9.The time axis is in seconds. However, the magnitude of throughput fluctuations is less with value of β=0.65.

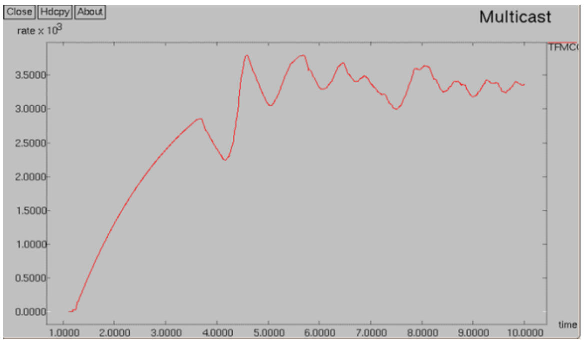

Fig. 1.2: Time Vs Rate (TFMC)

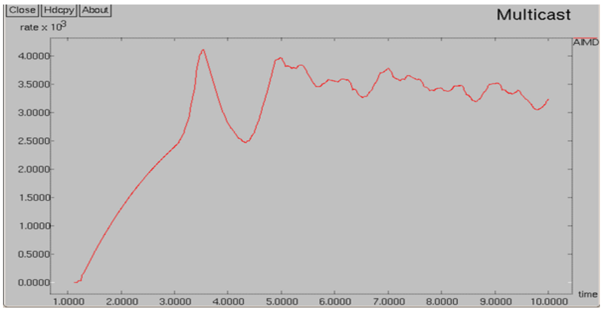

Fig. 1.3: Time Vs Rate (AMD)

Fig.1.5 shows the packet loss for the both cases i.e. AIMD and LIMD approach. AIMD is basically deals with CLR (current Limiting Receiver). It is clear that it provide the slow response to network state changes, slowest receiver.

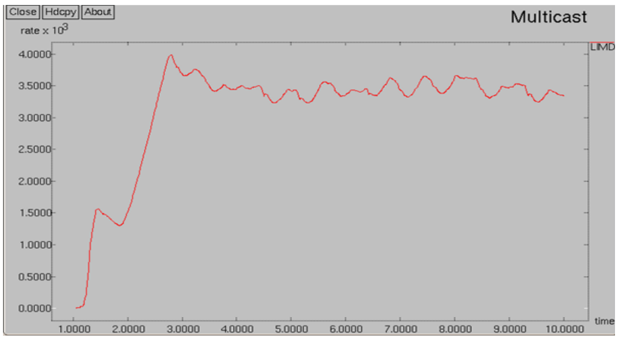

Fig. 1.4:Time Vs Rate (LMD)

There is more packet loss and less throughput the rate achieved by AIMD. Where as that of LIMD deals with the highest packet loss receiver. Further the shape shows that the magnitude of loss fluctuations of LIMD is considerably lesser compared to TFMCC. It is also considered that if the link of bottleneck becomes congested, losses increase as well as value of delay also increases.

Fig.1.2, it basically shows the comparison of steady-state throughput as anticipated by TFMCC, AIMD and LIMD for different values of packet loss ratio (p). It shows the impact of the loss-event ratio on the throughput equations of TFMCC, AIMD and LIMD. The sending rate depreciates extensively as the loss-event ratio raises.

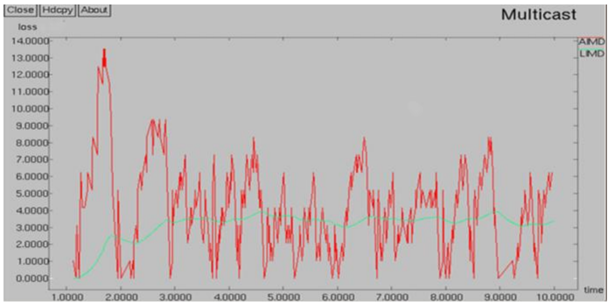

Fig. 1.5: Time Vs Loss (AMD and LMD)

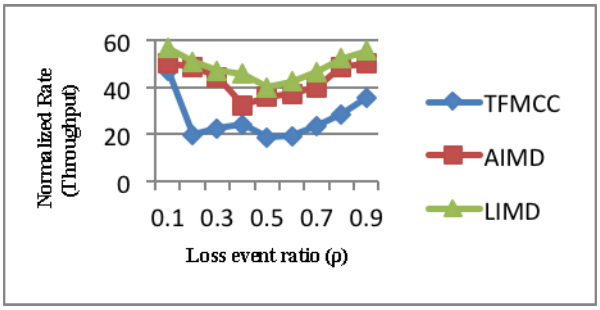

Fig. 1.6: Loss Ratio Vs Throughput

It is noticed that for the big values of p, it could afford a perceptible rate decrease and comparatively reduced throughput performance.

5. CONCLUSION

It is an improved version of the AIMD approach, which is an equation-based single-rate MCC technique. Two principal modifications are performed. First, each of the receiver (Highest packet loss) estimates its throughput based on a new equation derived according to the LIMD approach. Second, a hybrid rate-base preventive congestion control mechanism is implemented within the source. It improves in throughput; reduce rate fluctuations, RTT, and reduces the packet loss. It can be concluded that the totally throughput achieved in the case of LIMD approach is in the range of 40 to 57 kbps. But in the case of AIMD it is in the range of 32 to 50 kbps. Hence the throughput is good. On the other hand packet loss is very less. It basically 70% decreases as compared to AIMD approach.

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