Proficient Multicast Congestion Control Through Quality of Service

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Abstract: ***Computer network is the vital part of the networking. There are different approaches to transfer the data. When the data is transferred several scenarios i.e. congestion in the network, security issues, quality of service etc could take place. 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 an approach which provides both congestion control and quality of service. It provides the enhanced network routine for enhanced quality of data.***

Keywords: Multicasting, Congestion control, Quality of service etc.

1. Introduction

Computer network is one of the components of contemporary period. It uses a variety of methods to convey data from one node to a different. The spreading of a message from a sender to a receiver is called unicast. Communication of data from a sender to a group of receivers is called multicast. Multicast is used to launch information to various hosts over internet. It is mostly used in real time applications i.e. videoconferencing. It is same as the communication through Radio or TV where only concerned users accept the communication by selecting the exacting channel [1].

When the number of users enhances the extent of data also enhance. It generates the difficulty of congestion. The excellence of traffic is appreciably tainted when congestion occurs in the network. Congestion of the network can be condensed by varying the route, reducing and balancing the load. It can be evade by mounting the buffer, memory and link range of the available network [2].

There are several factors which creates congestion:

(a) **Connection breakdown:** If any connection stops functioning then all the packets over that particular connection will be vanished. The quantity of packet that should be conveyed depends upon the existing bandwidth and breakdown rate.

(b) **Bandwidth:** A precise bandwidth is selected in order that exacting quantity of data could be conveyed. If range of buffer is high in that case bandwidth should be lesser so that utmost delay could be granted. If the load is inferior and bandwidth is high in that case there may be situation of starvation.

(c) **Buffer Space:** The size of a buffer becomes limited at each node so that it could store limited information. Because it could provide the specific time period and delay.

(d) **Throughput:** When the sending rate is more than the receiving rate then the problem of congestion occurs [2][3].

2. Background

There are two scenarios for evading congestion control: open loop and closed loop congestion control. In the open loop congestion control, congestion is tackled earlier than it takes place while in the case of closed loop congestion control; congestion is tackled after it happened. First it is observed then based on the routine, exacting action is taken [3].The routine of a network is significant to both the service contributor and the end-user [2]. To get the information with reference to the quality of service parameters (Bandwidth, delay, jitter, losses etc) routine should be considered. To deliver the definite quality of service, resources should be kept in advance. Quality of service of a network is tainted due to the fault of network component or fewer resources [4].

The major idea of this work is to enlarge and relate approach for evaluate the routine by allowing for the quality of service parameters of subsisted application in the virtual system. As an example, Meeting through video application will be considered for estimation.

If data is sent by the sender to the number of receivers then the sending rate is accustomed by the slowest receiver. The feedback is given by the slowest receiver so that rate can be accustomed according to the existing metrics. Due to the rate amendment based on the slowest receiver, faster receiver will have to endure for it. It will degrade the routine of the system [7].

In another situation if the response is given by the representatitive of the group and transfer rate is accustomed based on feedback given by representative. In this case routine of the system will degrade due to delay (feedback given by representative) in the network [8][9].

Delay	Jitter	Loss	Quality Of Service
High	High	High	Poor
Medium	Medium	Medium	Average
Medium	Low	Medium	Average
Medium	Medium	Low	Average
Medium	Low	Low	Good
Low	Medium	Low	Good
Low	Medium	Medium	Average
Low	Low	Low	Good
Low	Low	Medium	Good

In the another case if there is congestion then the window size should be multiplitively decreases and if congestion is not there then sending rate is additively increased. Due to additive increase and multiplicatively decrease in the network, routine of the system degraded [6]. The routine of the system is resolute by taking the average of all system. To improve the routine of the system there is need to provide the quality of service. To provide the quality of service parameters is already chosen and users are divided based on the requirement.

If fixed bandwidth is chosen for the network and following scenarios can be taken place based on the network condition. There are three inputs (delay, jitter, losses) which define the nine rules. It provides the three type of services i.e. poor, average, good. In real time applications bandwidth is reserved in advance so that guaranteed quality of service can be provided. It can tolerate some losses and delay.

3. Proposed Work

In the following diagram Fig.1, Meeting through video is shown. In this each node works as a sender as well as receiver. There are nine nodes in the diagram. There are two nodes (profiles) i.e. application profile (node 0) and data profile (node 1) and other two work as interfacing (node 7) and as a server (node 8). In the application profile, applications i.e. email, file transfer chosen but in data profile bandwidth, delay, throughput is chosen. The role of every node changes according to the turn taking. When meeting takes place, a fixed packet size is chosen for the whole traffic and rate is adjusted according to the slowest receiver. The important parameters for providing quality of service in meeting through video are delay, jitter and packet loss.

4. Result and Analysis

We are going to simulate the result through OPNET.

4.1 Topology

First a video conferencing scenario is created as shown in fig.1 and interfacing is provided by the router.

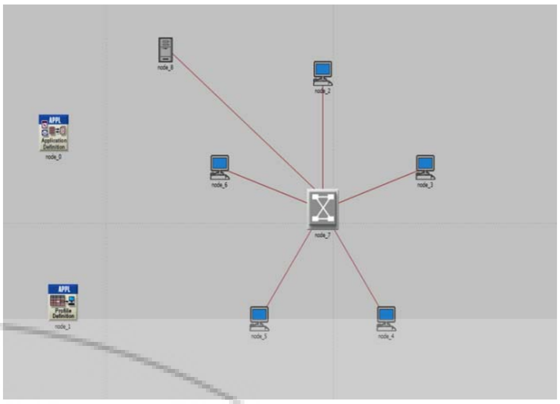

Figure 1: Meeting through video

It is shown that as the number of events (more users) increasing the simulation time also increasing. After a certain period of time when number of events (more users) become fixed then performance of the network increases.

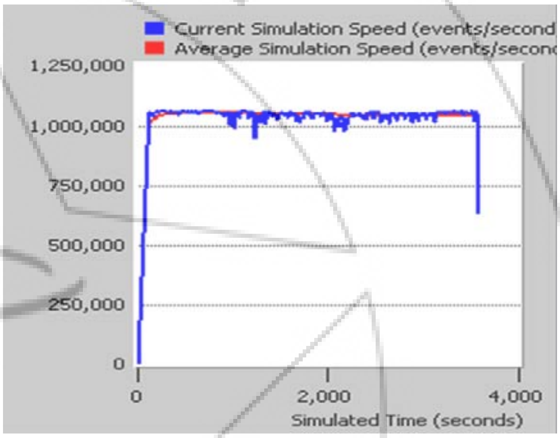

Figure 2: Performance evaluation

Fig.3 shows how memory space increases according to the increment of number of users but after a certain period of time when the number of users fixed the memory space used is also become fixed.

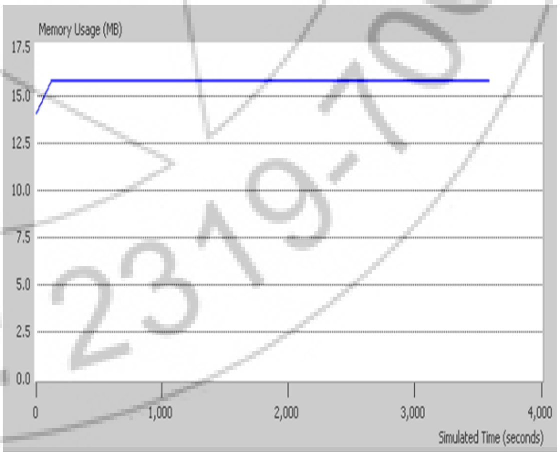

Figure 3: Memory space

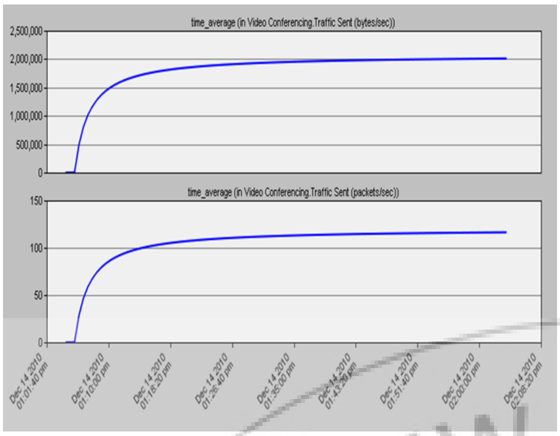

Figure 4: Average time

Fig 4 shows the performance of network in meeting through video. When the traffic is less and when the traffic increases and after a certain amount of data, performance becomes straight forward.

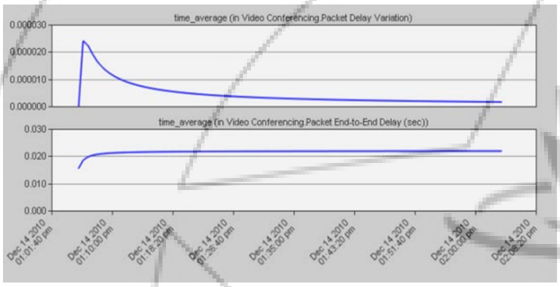

Figure 5: jitter and delay

Fig 5 shows the variation in the delay when the number of users is increasing then it suddenly goes to the peak point. It means jitter is high. It exponentially decreases and goes to the average result.
On the other hand end to end delay is average and does not change in quick way.

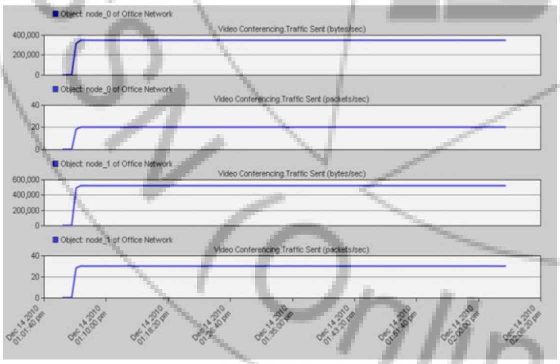

Figure 6: Traffic sent and received

It shows the how data traffic is increasing at node 0 and node 1 and based on that data and application profiles is decided and it varied according to the demand of the network.

5. Conclusion

We have proposed a new mechanism to provide the congestion control and quality of service which is based on delay, jitter, throughput and bandwidth. In this proposed approach the application and data profile adapts the network performance which is based on the receiver and increase or decrease traffic rate according to the situation. When delay, jitter and loss is more than quality of service is poor. To provide the better quality of data, jitter, delay and loss should be minimum. Results are simulating in OPNET and shows it provide the better quality, throughput and less packet loss.

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