An Effective Mechanism for Congestion Control in High Speed Networks

Ms.T.Sheela’ and **Dr.J.Raja’’**

Sathyabama University, SSN College of Engineering, INDIA

Summary
Due to the recent trends in Internet, for exchange of information in the form of pure data traffic and multi-media traffic, High-speed network is necessary. As there is a growing demand for high-speed networks, data transfer must take place without any congestion. In data networking and queueing theory, Network congestion occurs when a link or node is carrying so much data that its quality of service deteriorates. Typical effects include queueing delay, packet loss or the blocking of new connections. A consequence of these latter two is that incremental increases in offered load lead either only to small increase in network throughput, or to an actual reduction in network throughput. The Transmission Control Protocol (TCP) is one of the core protocols of the Internet Protocol Suite. TCP has not performed well on high-speed network because the standard TCP’s algorithm for congestion control may cause thousands of packet drops in one Round Trip Time (RTT) and the window size is halved at the time of congestion. So the utilization of the bandwidth and throughput is minimized. The researchers developed different TCP variants to improve the performance of the congestion control algorithms in high-speed network. In this paper we propose a sequence of algorithms based on window adjustment, feedback mechanism and buffer management to overcome the limitations for high-speed network. The congestion control mechanism that we adopt starts with Queue Management, which enables to modify and control the transmission queue. The Window Adjustment Mechanism alters the Congestion window size based on the Input load. The Feedback mechanism renders the optimum load that can be serviced currently when congestion had taken place. Also the feedback information is notified to the sender. This solution to control the congestion in the network achieves maximum throughput and bandwidth utilization, with minimum delay and drop probability. The performance of the congestion window, throughput, utilization of the bandwidth, delay are analyzed and presented.

Keywords: TCP/IP, Congestion Control, Window Management, Feedback Control, Queue Management, Explicit Control.

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1. INTRODUCTION

As with increase in the amount of data transfer across the various networks, to achieve low delay, maximum throughput and predictable performance on an end-to-end basis[1][5][9], a high bandwidth environment is required. An increase in bandwidth and data rate leads to congestion. So congestion has to be avoided while sending large volumes of data within short period of time.

Congestion window mechanism in TCP increases the window size to their threshold value and drastically decreases at the time of congestion in high-speed network[3][6][12]. This is because TCP increases and decreases its congestion window size too slowly with non-duplicate ACK packets [10][13][18]. If we require responses to be quicker, then a connection with large RTT must be present than a connection with smaller RTT, because increase of TCP is tied to each RTT.

If large amounts of data have to be sent then TCP sends all these packets based on the window size and which can increase the congestion problems, because the window size is halved [25]. In TCP, loss of packets by itself acts as a signal which indicates the sender to lower down the congestion window to limit the number of packets to be sent and thereby congestion can be reduced. Thus TCP initiates congestion control only if a packet is considered lost. To recover from the packet loss, congestion window size is reduced by half, and so the congestion window size is increased by 1 segment for every Round Trip Time. For example, in 10 Gbps connection, initially the link will operate at 10 bps connections and then gradually increase to 10 Gbps and hence by means of TCP it will take more time to transfer data on high-speed networks[12][13]. Due to the gradual increase and sudden decrease in high-speed network, utilization of bandwidth by TCP is very poor .

Another issue with TCP is the way it allocate bandwidth on networks with high bandwidth delay products and also the Queue size [8][14]. If we have a fast link (eg. Gbps), the TCP’s AIMD algorithm is not able to send large volume of data because the sending rate is very low. So automatically limited number of packets are transmitted when we have a high-speed environment. For example consider a high-speed network with Rtt=100 ms and packet size =1.500 bytes. By using of AIMD algorithm the sending rate is 15.00 bytes during the congestion time. So we can transmit only 10 packets/sec. So automatically the performance level degrades because of poor utilization and minimum throughput [15][19].

In the proposed work, the new protocol XCP operates between network layer and transport layer. The Window Management enables to dynamically adjust the congestion window size based on the input load. The Congestion window is adjusted dynamically to accommodate the incoming packets. Queue Management adopted in this protocol alters the queue size dynamically based on the required conditions. In this protocol, the feedback mechanism is useful and it enables to modulate sending rate based on the traffic and service rate of the network in order to avoid congestion. For each flow, the calculations were made and at the same time, sender knows the status of network capacity and the sending rate at the time of congestion. Here the congestion bit value is sent to the sender and the capacity of the network and window size is updated based on the arrival rate, service rate and traffic rate. So the sender is sending the data according to the feedback values in the reverse feedback field by the receiver. Due to the feedback mechanism, maximum number of packets are transmitted and it achieves high utilization, maximum throughout and low delay[16][17].

The rest of this paper is organized as follows. Section 2 explains the basics of XCP and its Overview. Section 3 describes the Mathematical model. Section 4 explains the proposed scheme based on Queue Management, Window adjustment and Feedback mechanism . Section 5 indicates the Simulation Results. Section 6 has the concluding remarks.

2.XCP BASICS

2.1 XCP overview

The per-flow product of bandwidth and latency increase leads to TCP becoming inefficient and prone to instability[2][4][7]. The new Explicit Control Protocol outperforms TCP and remains efficient, fair, scalable, stable and XCP generalizes Explicit Congestion Notification proposal. XCP is modeled and demonstrated as stable and efficient regardless of link capacity, round trip delay. XCP achieves fair bandwidth allocation, high utilization, small standing queue size, and near-zero packet drops with both steady and highly varying traffic. Additionally, XCP does not maintain any per-flow state in routers and requires few CPU cycles per packet, which makes it suitable for high-speed networks. This new eXplicit Control Protocol, XCP, generalizes the Explicit Congestion Notification proposal (ECN). In addition, XCP introduces the new concept of decoupling utilization control from fairness control. This allows a more flexible and analytically tractable protocol design and opens new avenues for service differentiation. Fig.1. represents the XCP overview.

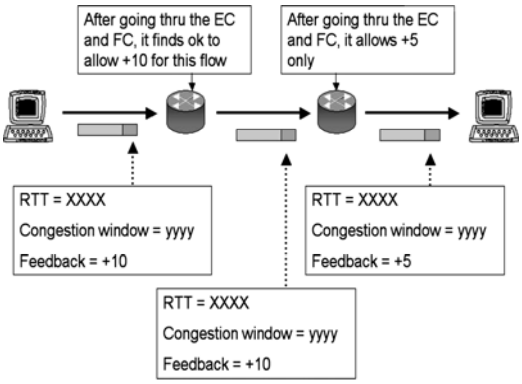

Fig 1- XCP Overview

3. MATHEMATICAL MODEL

The mathematical model renders the suitable parameters governing XCP's characteristics. It encompasses the,

- Input Model
- Output Model
- Queue Model
- Feedback Model

3.1 Input Model

The Input Model showcases the network parameters. They are as follows,

(1) Flow Rate : It may be defined as the total number of packets/sec sent by one sender at time ti and is given by λi

(2) Load on the network : It is defined as the number of flows per second
For n flows, Load is given by $\sum \lambda_i$ i= 1 to n where n is the number of flows

(3) Average Arrival Rate, arr_{avg} at time t_i is given by
$\sum\lambda_i$ / packet size

3.2 Output Model

The output model renders the output parameters of the network. They are referred to as Service Rate Parameters**.**

(1) Ts (Service Time)
The Service Time renders the time taken by the network to respond to a request.

T_S =Packet Length/Network Capacity

(2) Utilization Factor
The Utilization factor renders the maximum resources used by the network to render suitable performance

Utilization(ρ) = λ_iTs

(3) Service Rate
The Service Rate depicts the amount of packets serviced per second.

Service Rate(μ) =ρ * Router Capacity

3.3 Queue Model

(1) Queue Threshold
The parameter in the queue after which congestion is detected is defined as the queue threshold.
Queue Threshold (q_{th}) = 85% of queue size(q_{max})

(2) Queue Position
The position of the current pointer.

$$\text{Queue Position}(q_p) = \sum_{i=1 \text{ to } n} \lambda_i * T_w$$

where T_w is the waiting time.

(3) New buffer(q_{new}) = $q_{max}/2$
Whenever the queue position reaches the threshold value the new buffer is utilized.

(4) Throughput : It may be defined as the total number of packets transferred per second,

i.e., $\sum\lambda_i$ at time t_i i = 1 to n

(5) Delay
The delay is calculated based on the RTT value.

Delay = RTT * 0.02

3.4 Feedback Model

The feedback is calculated based on the input load and link capacity.

The feedback is given by,

For 0< load<89% ; Feedback = 2.8*(link capacity-load)
For load>90% ; Feedback = 10*(link capacity-load)

4. PROPOSED SCHEME

The Explicit Control Protocol adopts three major techniques to tackle Congestion. They are,

- Queue Management
- Window Adjustment
- Feedback Mechanism

The above stated techniques, their purpose and their mechanisms are explained in detail in the ensuing sections.

4.1 QUEUE MANAGEMENT

The Queue management procedure adopted in XCP is shown in the Fig.2. Congestion occurs commonly in many of the High speed networks. The occurrence of Congestion is mainly due to the presence of multiple senders transmitting volumes of data at varying arrival rates. The success of any networking protocol lies in its ability to counter this Congestion and render reasonable service rates. Also emergency packets are to be considered. Packet Loss is a major issue which is being encountered whenever congestion occurs. Packet loss can drastically affect the performance of any high speed network. The design of various networking protocols mainly deals with ways and measures of avoiding this factor. One approach which can be adopted is to utilize a queue. A queue may be considered as a temporary buffer to route the packets. All the packets are queued when transfer is initiated and are queued to complete the transfer.

In case of the traditional TCP protocol, the Random Early Detection (RED) queue management technique is adopted. Any protocol that runs on top of Internet Protocol (IP), such as Transmission Control Protocol (TCP), can detect packet drops and interpret them as indications of congestion in the network. In particular, a TCP sender will react to these packet drops by reducing its sending rate. This slower sending rate translates into a decrease in the incoming packet rate at the router, which effectively allows the router to clear up its queue.

Queues are used to smooth spikes in incoming packet rates and to allow the router sufficient time for packet transmission. When the incoming packet rate is higher than the router's outgoing packet rate, the queue size will increase, eventually exceeding the available buffer space.

Table 1:Queue Management Table

Time	Queue Position q_p $\Sigma\lambda_i.T_w$ MBps	Queue Size (.0033 * load) (in MB)	Q_{th} (85% of queue size)
T1	0.80	0.132	0.1122
T2	1.60	0.264	0.2244
T3	1.90	0.313	0.2660
T4	1.50	0.247	0.2099
T5	0.70	0.115	0.0977
T6	1.96	0.323	0.2745
T7	0.78	0.128	0.1088
T8	1.58	0.260	0.2210
T9	1.98	0.326	0.2771

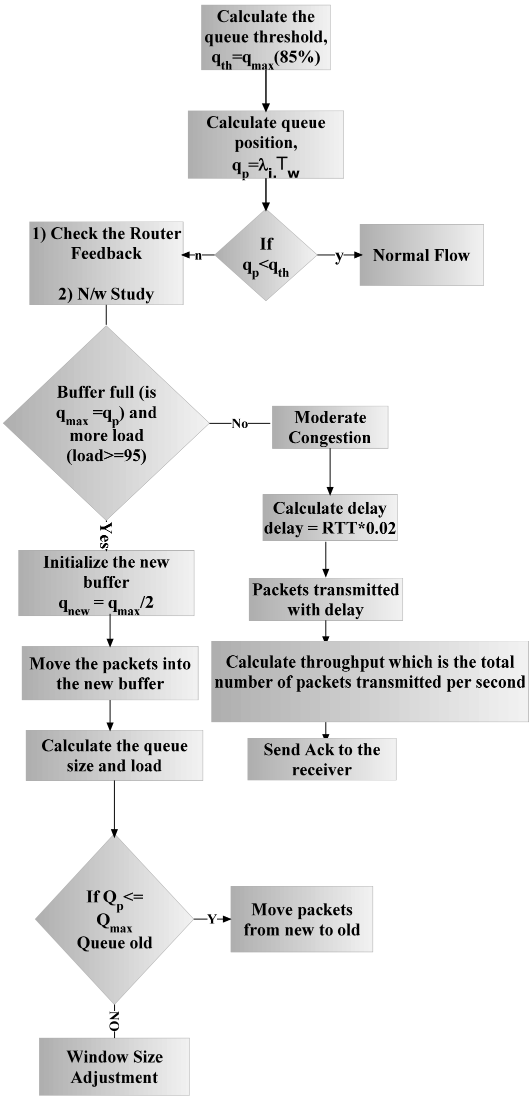

Fig 2 – The Queue management procedure

When the buffer is full, some packets will have to be dropped. In TCP's RED queue management, the problem is whenever the queue size reaches the q_{th} value packets are dropped abruptly. The congestion is detected early and the packets are dropped. The merit may be that congestion is detected early but the adverse effect is loss of packets. As a result the sender has to retransmit causing delay onto the network. This causes adverse effects on the network performance. In order to overcome this disadvantage a more refined approach is being adopted by XCP in order to cater to high speed networks. This approach is clearly depicted in the above flowchart. Initially the queue threshold is fixed to be 85% of maximum queue size. The Queue threshold is the maximum limit after which congestion is detected. Next the queue position is calculated. Based on the queue position, the router feedback is calculated in case the queue threshold exceeds the queue position. This feedback is a special feature of XCP. It informs the sender as to how to reduce the arrival rate whenever the congestion occurs. This enables to reduce delay reasonably. As a result the network performance would enhance reasonably. Whenever the queue buffer is full, a new buffer is initialized which is half the queue size. This would enable to counter congestion. If this measure fails, then XCP adopts Window adjustment procedure. For partially filled queue buffers, the delay is calculated based on the RTT value. The delay is 0.02 times the Round Trip Time which is reasonable when compared to TCP. Then the throughput is calculated before rendering acknowledgement to the Sender. The throughput renders the efficiency of the network. The queue size, queue position and queue threshold values are depicted in Table.1

Thus as explained the queue management procedure of XCP enables effective Congestion control in case of High-speed networks.

4.2 WINDOW ADJUSTMENT

The Window Adjustment is adopted next to the Queue management. The congestion window determines the number of bytes that can be outstanding at any time. This is a means of stopping the link between two places from getting overloaded by heavy traffic. The size of this window is calculated by the product of the bandwidth and the RTT value.When a connection is set up, the congestion window is set to the maximum segment size (MSS) allowed on that connection. Further variance in the collision window is dictated by an Additive Increase/Multiplicative Decrease approach. This means that if all segments are received and the acknowledgments reach the sender on time, some constant is added to the window size. The window keeps growing linearly until a timeout occurs or the receiver reaches its limit. In TCP,BICTCP,CUBIC TCP,HTCP and HSTCP the congestion window size is reduced drastically[20][21][22][23][24], but in XCP, the Congestion Window is calculated based on the load value and RTT.

Table 2 – The Window Adjustment table

Time	Load (in MBps)	Cwnd (in MB)	Window Size (in MB)
T1	40	0.0000	1.0000
T2	80	0.0126	0.9874
T3	95	0.0204	0.9796
T4	75	0.1004	0.8996
T5	35	0.0000	1.0000
T6	98	0.0211	0.9789
T7	39	0.0000	1.0000
T8	79	0.1652	0.8348
T9	99	0.0223	0.9777

The window size is calculated based on the bandwidth and RTT values Whenever congestion occurs, the cwnd value is calculated and the actual window size is modified as (Window Size-Cwnd value). The following information is fed back to the sender,informing him to alter his input load accordingly. The window adjustment values are represented in Table.2.

4.3 FEEDBACK MECHANISM

To avoid the congestion in high speed network, feedback mechanism is being used to know the status of the network at the time of data flow. So according to the receiver's feedback, the sender has to resize the window and also the additional parameter values are known by the sender. In the existing system, TCP's window size is changed which results in the control of the transmission rate. But the sender does not know how much amount of data to be transmitted when the congestion occurs and also the buffer is idle even though it has a greater capacity because TCP will gradually increase the window size, but when it detects a loss, the window size(W) is cut to W/2, So TCP is not allowed to send anything more until it has received W/2 acknowledgements.

In the proposed work, XCP's sender will send the data based on the feedback values given by the receiver and hence the sender knows the status of the network and the amount of data to be transmitted at the time of congestion state. As a result of this, the congestion is controlled in the network and there is a continuous flow of data in the high speed network and also the buffer is not idle and therefore there are always some packets in the buffer.

At the time of transferring the voluminous data in the network, the average arrival rate (*(λ)* is calculated based on the number of senders data rate and its time interval (t). Also, based on the arrival rate (λ) the service rate (***(μ)*** and traffic rate(Y) are calculated. For each flow, the calculations were made and at the same time, sender knows the status of network capacity, traffic rate and the sending rate at the time of congestion[16]. Here the congestion bit value is sent to the sender and the capacity of the network and window size is updated based on the arrival rate, service rate and traffic rate. The sender can thus send the data according to the feedback values in the reverse feedback field received by the receiver and the values are represented in Table.3.

Table 3 – The Feedback Mechanism Table

Time	Feedback	Throughput %	Delay (in µs)
T1	-	100	0.000
T2	56	92.38	0.005
T3	50	91.8	0.0063
T4	70	92.38	0.0049
T5	-	100	0.0000
T6	45	91.8	0.0065
T7	-	100	0.0000
T8	58.8	92.38	0.0052
T9	46.8	91.8	0.0065

5. SIMULATION RESULTS

5.1 NETWORK

The following Fig.3 represents the dumb bell topology in Omnet++.

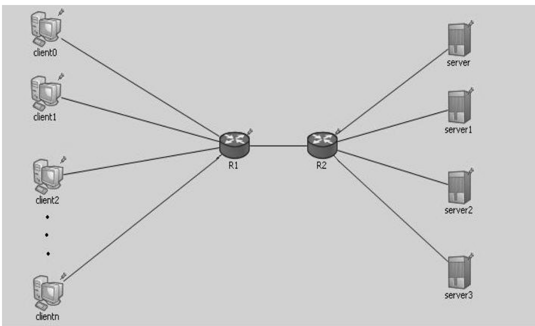

Fig.3-Network Topology

In this scenario the number of client nodes from 0 to n with maximum link capacity of 10 Mbps bandwidth and delay of 10 ms is considered in full duplex access link. The router link capacity is 100 MBps and delay is 10 ms. The parameters considered for comparison is number of packets per flow at time t.

5.2 ARRIVAL

The arrival rate of the clients at time t is defined as λi. The arrival rate is tested from 40 MBps to 95 Mbps for check the status of normal flow, Moderate congestion and severe congestion. The graph plotted between the time and the arrival rate is shown in Fig 4.

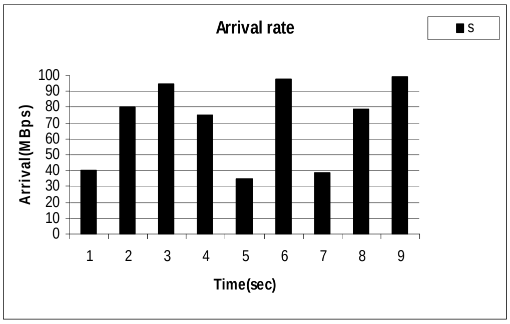

Fig 4-Arrival rate

5.3 THROUGHPUT

During the normal flow the throughput reached is maximum (100%). At the time of moderate congestion with an acceptable delay the throughput is maintained at 80 to 90% level. This is represented in the graph with respect to time and is represented in Fig.5..

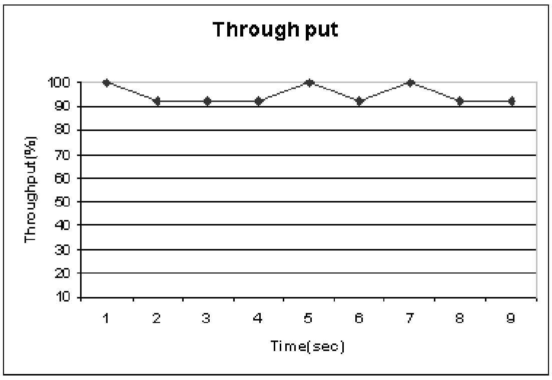

Fig.5. Time Vs Throughput

5.4 UTILIZATION.

Utilization of the bandwidth is compared with existing and proposed queue management scheme. This simulation is given in the following graph (Fig 6.). From this simulation it is concluded that the proposed queue management achieves the maximum utilization of the bandwidth when the congestion occurs.

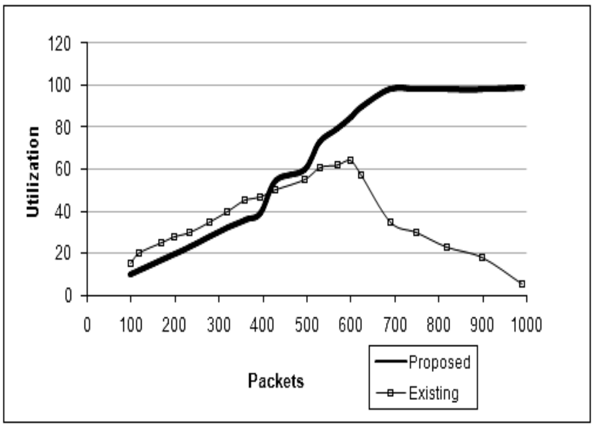

Fig.6. Bandwidth Utilization

Hence utilizing XCP, performance of high speed networks can be improved considerably.

5.5 DELAY

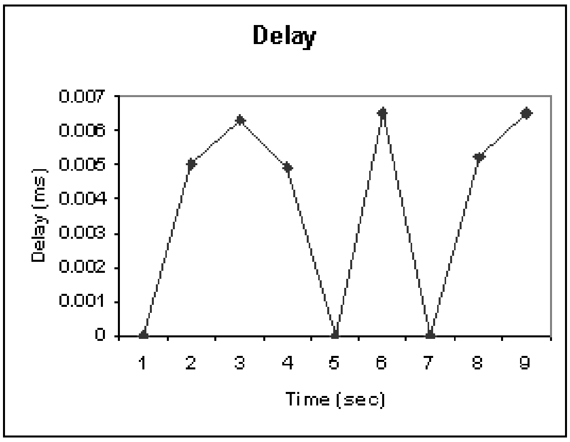

Fig 7 - Arrival Time Vs Delay

Whenever there is normal flow, there is no delay. At the time of moderate congestion, the delay ranges from 0.0049 to 0.0052 µs. Whenever severe congestion occurs, the delay value ranges from 0.005 to 0.0065µs.

5.4 WINDOW SIZE

Considering the various success of TCP such as HSTCP, BICTCP,HSTCP and cubic TCP which are considered as best in performance, the windows adjustment is achieved to an extent of 90% [14]. The results are best enunciated in the following graphs[20][21][22][23][24].(Fig 8a-8d)

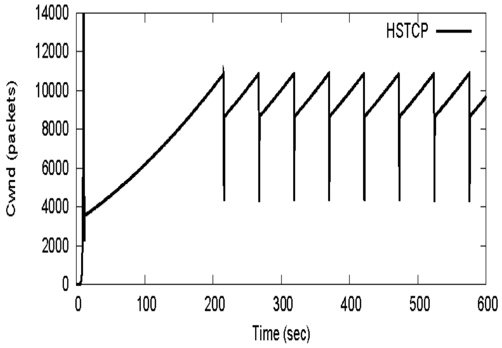

Fig.8a- Time Vs Cwnd(HSTCP)

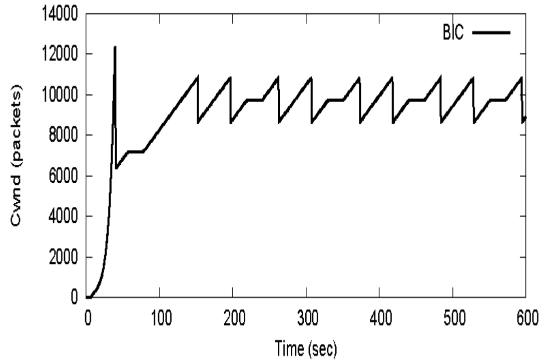

Fig.8b- Time Vs Cwnd(BICTCP)

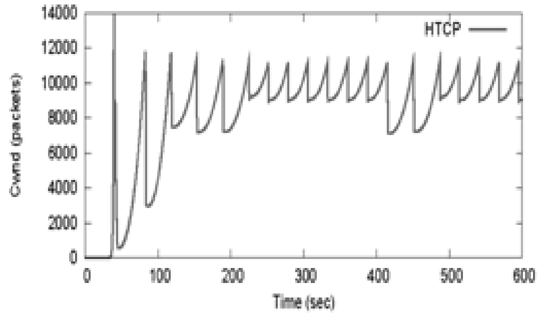

Fig.8c- Time Vs Cwnd(HTCP)

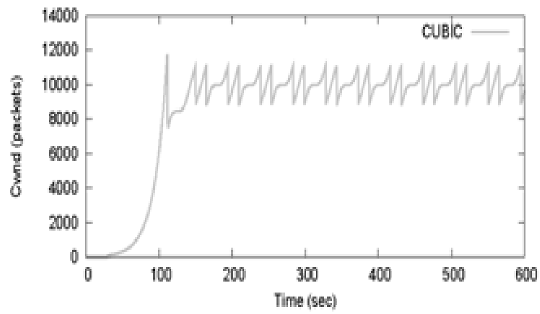

Fig.8d- Time Vs Cwnd(Cubic TCP)

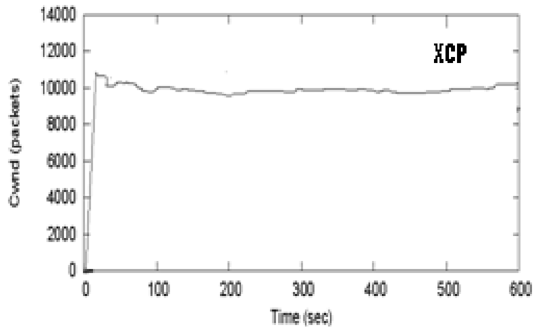

Fig.8e- Time Vs Cwnd(XCP)

Figure 8a depicts the performance of HSTCP. As in the Graph, When Congestion occurs, the entire performance of the network drips down indicating vagaries in performances. In HSTCP, 10,000 packets are transmitted during the time interval 0-200 sec. Then due to the window size adjustment there is drastic decrease and the throughput is minimized and at the same time the utilization of the bandwidth is decreased. The simulation results of HSTCP with respect to time and packets flow is shown. So the window size is updated based on the flow. In case of BICTCP(Figure.8b), it identifies target window size based on the binary search scheme. Here the pitfalls are reduced, but the time delay due to disruptions caused in Sender rate, produce the following Simulation Results The Simulation is more finer than HSTCP.In HTCP as shown in Figure 8c, the value experiences frequent pitfalls. A linear response is not achieved. Also in case of cubic TCP(Figure 8d),only a response which is slightly better than HTCP is achieved. But still the efficiency obtained is very low. Now comparing the above TCPs with XCP, the Windows Adjustment Procedure gives a result of 98% efficiency. The plot is almost linear. The Traffic is managed Effectively and its represented in Fig.8.e. The vagaries of HSTCP and BICTCP,HTCP are eradicated in XCP as shown in Figure (8a-8d) The Simulation Results show minimized drips in Cwnd values with increasing time. So it achieves more throughput because more number of packets are transmitted and the flow is normal at the time of congestion. If large number of packets are transmitted within the particular time interval then the utilization of the bandwidth is increased. The throughput and utilization are shown in Fig.5 and Fig.6 respectively So the network utilizes the full bandwidth and it achieves 98% efficiency.

6. CONCLUSION

The proposed scheme achieves the maximum throughput with minimum delay in high speed network with the help of the effective queue management scheme. In this paper, the algorithm for Window adjustment Procedure for XCP is proposed which achieves 98% efficiency when compared with High Speed Protocols like HSTCP,BICTCP,HTCP and cubic TCP. The window size is altered drastically based on the time and packet flow and the utilization of the bandwidth is reduced. But in XCP the procedure is based upon the bandwidth and feedback, where the input packets are being regulated by altering the source node and there by reducing the window size in unit step. So the sender can restrict the flow and can avoid congestion thereby resulting in effective Bandwidth utilization. Also the Feedback Mechanism in XCP enables to identify occurrence of congestion effectively. This mechanism is highly useful, particularly in Large Networks wherein many senders transfer continuously at varying data rates. Hence in comparison to the existing scenarios XCP provides a better mechanism for effective data transfer.This mechanism is highly useful, particularly in Large Networks wherein many senders transfer continuously at varying data rates. Hence in comparison to the existing scenarios XCP provides a better mechanism for effective data transfer. Also then by effectively calculating the Queue size, Window Size, Utilization and delay can still be minimized.

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T.Sheela obtained B.E degree in Computer Science & Engineering from Madurai Kamarajar University, Madurai TamilNadu, India, in 1989 and M.S Degree in Computer Science & Engineering from BITS – Pilani, Rajasthan, India, in 1993.She is a Professor in Information Technology in Sri Sai Ram Engineering College-Chennai. She has published research papers in international and national journals and seven papers in conference Proceedings. She authored books on Data Structure and Fundamentals of Computers. Her research work was selected for presentation at AICTE (Govt. of India), New Delhi.

Dr.K.Raja obtained Ph.D. from Department of Electronics and Communication Engineering at Anna University. He is a Professor and Head in Information Technology at SSN College of Engineering. Dr.J.Raja has supervised many Ph.D/M.E/M.Tech/M.S. thesis in the area of Digital communication and Networks. He has published research papers in international and national journals and conference proceeding with very high citation index of about 25 papers so far.