Indian Journal of Science and Technology Vol.2 No 3 (Mar. 2009) ISSN: 0974- 6846

Maximization of throughput by effective queue management scheme for high speed network

T. Sheela[1] and J. Raja[2]
[1]*Sathyabama University;* [2]*Department of IT, SSN College of Engineering, Chennai, India*
sheela_saiit@yahoo.com, rajaj@ssn.edu.in

Abstract: A new algorithm is proposed to control the congestion in the network based on the minimum drop ratio. Explicit Control Protocol (XCP) is adopted for high speed network, where the queue threshold is assigned to be 85% of the queue size. At the time of congestion, whenever the queue size reaches the threshold value, a new buffer is initialized which is half the queue size. This mechanism thus prevents the drop of packets and hence the drop ratio is minimized and throughput is maximized. Performance of bandwidth utilization, throughput and drop ratio is presented.
Keywords: Network, XCP, drop ratio, throughput, queue.

Introduction

Due to the recent trends in Internet, exchange of information in the form of pure data traffic and multi-media traffic, a high speed networks is necessary to meet the requirements. In such a network, the occurrence of congestion is mainly due to the presence of multiple senders transmitting large volume of data at varying arrival rate. To control the congestion an active queue management mechanism is required. A queue may be considered as a temporary buffer to route the packets. In TCP the queue mechanisms have a significant drop probability ratio, because whenever the queue size meets the desired threshold value, the arrival of the incoming packets will automatically drop at the time congestion. This leads to minimum throughput, low utilization of bandwidth and increase in the drop probability ratio. But the proposed

The Explicit Control Protocol (XCP) is mainly for congestion control. To control the congestion in the high speed network, the queue management is very essential (Yongguang & Lang, 2005; Peng Wang & Mills, 2006). Exchange of information in the form of pure data traffic and multi-media traffic demands higher bandwidth and the service quantities of the current network. Hence, the high speed network is necessary. At the time of transmitting the large volume of data with seamless flow, the queue management algorithm is essential because it uses a temporary buffer to route the packets (Mehdi & Haeri, 2005).

The incoming packets are transferred to the destination node through the router at the time of normal flow. When congestion occurs, the status of the queue size has to be checked since all the incoming packets are stored in the temporary buffer and forwarded to the destination node. In the RED queue management technique of TCP, a queue threshold value is maintained (Sanjeeva & Low, 2001; Steven H. Low, 2003; Huajun, 2005). At the time of congestion, the incoming packets are stored in the queue and when it reaches the threshold value, the upcoming packets are dropped. So the packet drop rate is high and that automatically reduces the throughput.

In XCP depending upon the header feedback, the receiver sends the network status information to the sender in the reverse feedback field (Kyle *et al.*, 2004; Nishanth *et al.*, 2005; Yongguang & Lang, 2005). In this paper, new algorithm is proposed to control the congestion in the network based on the minimum drop ratio. Here, the incoming packets are stored in the temporary buffer and when it reaches the queue threshold value, the algorithm will not drop the incoming packets. Instead it introduces the new buffer which is half the queue size. So the incoming packets are moved in to the new buffer and after the queue position is less than the queue threshold ($q_p < q_{th}$) value, the new buffer content is transferred to the old buffer. Hence the packet drop rate is minimized and the algorithm maintain 95 % throughput.

XCP basics

XCP overview

The per-flow product of bandwidth and latency increase leads to TCP becoming inefficient and prone to instability (Sherali & Zhan, 2004; Fernando & Low, 2005; Steven L. Low & Achlan L.H. Andrew, 2005). The new XCP outperforms TCP by means of remaining efficient, fair, scalable, stable and generalizes Explicit Congestion Notification proposal. The 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. The overview of the XCP is shown in Fig.1.The sender's information is sent to the router and the router is performing the data flow through the EC (Efficiency Controller) and FC (Fairness Controller) (Sally & Fall, 1998; Ramesh & Tang, 2001; Nishan & Lam, 2005).

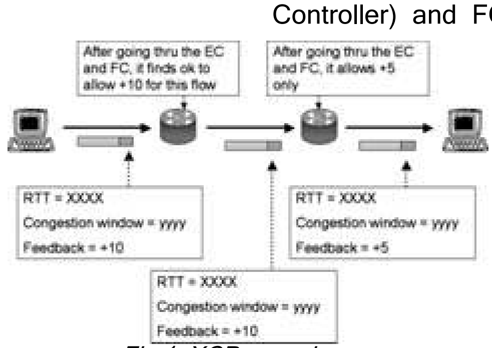

Fig.1. XCP: overview

XCP header

The XCP's or congestion header is present between IP and transport headers and it is 32 bits in size. XCP is an end-system-to-network communication (Fernando & Low, 2003) The Header format of the XCP is shown in Fig.2.

The XCP congestion header

Indian Journal of Science and Technology Vol.2 No 3 (Mar. 2009) ISSN: 0974- 6846

consists of the following fields:

Version (4 bits): This field indicates the version of XCP that is in use.

Format (4 bits): This field contains a code to indicate the congestion header format.

Protocol (8 bits): This field indicates the protocol to be used in data level portion of the packet.

Length (8 bits): This field indicates the length of the congestion header, measured in bytes.

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|version|format |   protocol    |    length     |    unused     |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                              rtt                              |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                           throughput                          |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                        delta_throughput                       |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                        reverse_feedback                       |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Fig.2. The XCP header structure depicting various fields

Unused (8 bits): This field is unused and must be set to zero in this version of XCP.

Rtt (32 bits): This field indicates the round-trip time measured by the sender, in fixed point format with 28 bits after the binary point, in seconds.

Throughput (32 bits): This field indicates the inter-packet time of the flow as calculated by the sender, in fixed point format with 28 bits after the binary point, in seconds

Delta Throughput (32): This field indicates the desired or allocated change in throughput.

Reverse Feedback (32): This field indicates the value of Delta Throughput received by the data receiver. The receiver copies the field Delta throughput into the Reverse Feedback field of the next outgoing packet in the same flow.

In a high speed network, network congestion may cause more losses due to the fact that congestion windows may grow to very high values. But in the proposed scheme window adjustment procedure is to control the congestion in the network based on the feedback given from the receiver end. The features that make explicit control protocol (XCP) usable in high speed networks are, the control information carried in congestion header and the generalized explicit congestion notification proposal (Satyanarayan & Reddy, 2008). The extent of congestion is sent to the sender by the receiver or router. Then the sender can increase or decrease its sending window in response to the network state.

XCP layer

XCP is a joint design of end systems and routers and it differs from the end-to-end design of TCP and the hop-by-hop design of IP (Peng Wang & David Mills, 2006). The Position of XCP in the protocol suite is shown in Fig.3.This protocol is placed between the TCP and IP and it is a transport level protocol.

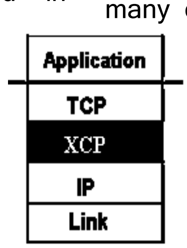

Fig.3. XCP layer

Mathematical model

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

a. Input model

The Input Model showcases the network parameters such as: (1) Flow Rate: It may be defined as the total number of packets/sec sent by a 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 number of flows, Load is given by $\sum \lambda_i$ i= 1 to n. (3) Average Arrival Rate, arr_{avg} at time ti is given by $\sum\lambda_i$ / packet size.

b. Output model

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

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

T_S =Packet Length/Network Capacity

(2) Utilization factor which renders the maximum resources used by the network to render suitable performance. Utilization (ρ) = $\lambda_i Ts$

(3) Service Rate which depicts the amount of packets serviced per second. Service Rate (μ)=ρ *Router Capacity

c. 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: of the current pointer.

Queue Position (q_p) = $\sum\lambda_i$ * Tw for i=1 to n

Where T_w is waiting time

(3) New buffer (q_{new}) = $q_{max}/2$

Whenever the T_w waiting time current queue reaches the threshold value, the new buffer is utilized.

(4) Throughput: 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: is calculated based on the RTT value.

Delay = RTT * 0.02

Proposed scheme

The queue management procedure adopted in XCP is shown in the Fig. 4. 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 dequeued 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. When the buffer is full, some packets will have to be dropped. But with respect to TCP's RED queue management, the problem is the buffer size is reduced to half and the 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

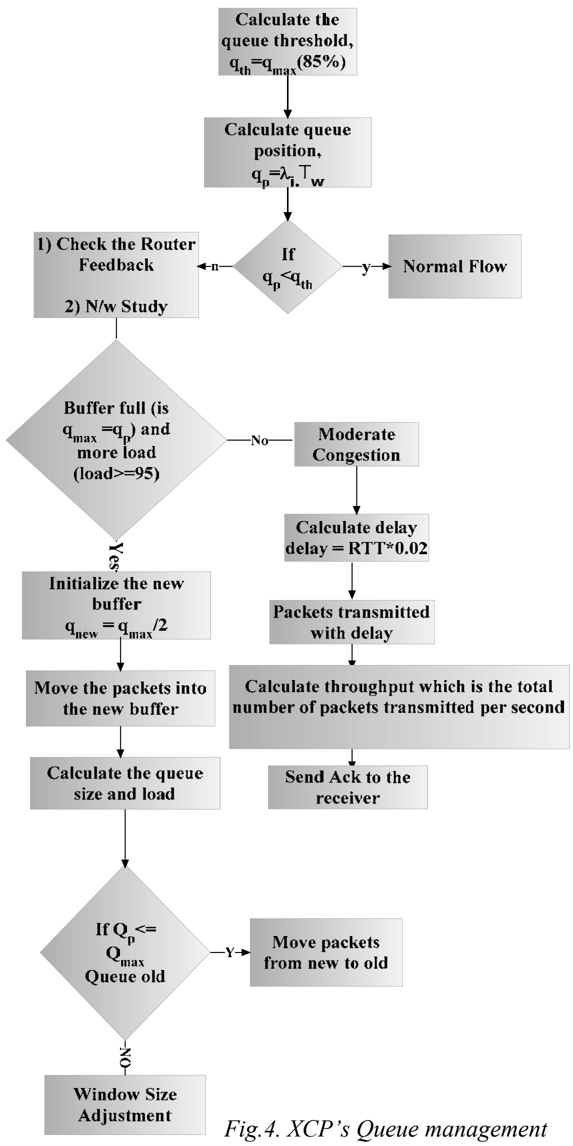

Fig.4. XCP's Queue management

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.

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

Simulation results

Network

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.

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. 6.

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 (Fig. 7).

Utilization

Utilization of the bandwidth is compared with existing and proposed queue management scheme. This simulation is given in the following graph (Fig. 8). From this simulation it is concluded that the proposed queue management achieves the maximum utilization of the bandwidth when the congestion occurs.

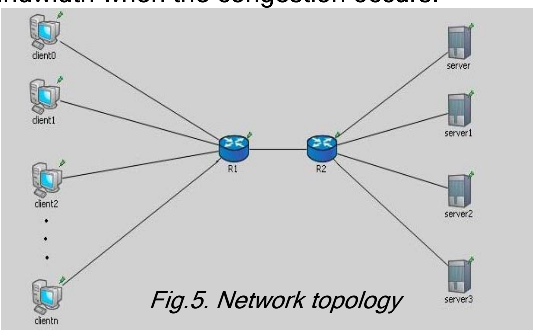

Fig.5. Network topology

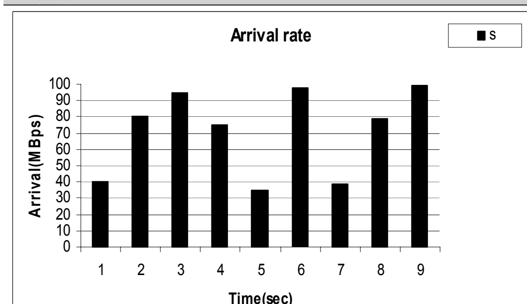

Fig.6. Arrival rate

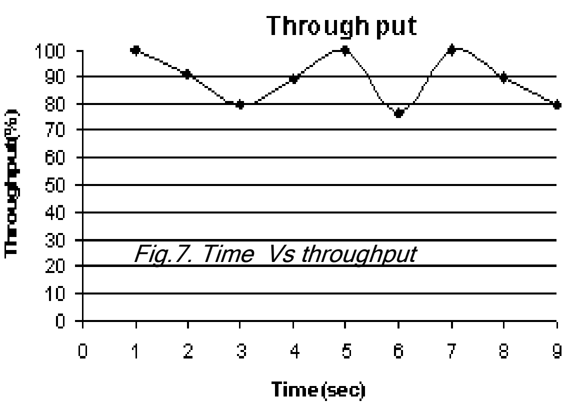

Fig.7. Time Vs throughput

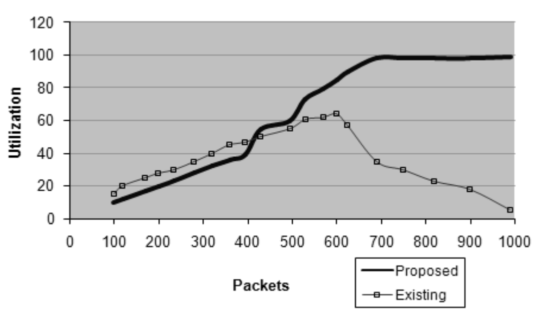

Fig.8. Bandwidth utilization

Conclusion

The proposed scheme achieves the maximum throughput with minimum delay in high speed network with the help of the effective queue management scheme. Here the bandwidth is shared and utilized effectively. By managing the QOS between the networks properly maximum utilization can be achieved in real time data transfer in the high speed network.

References

1. David X (2006) BIC TCP: motivation, architecture, algorithms, performanc. *IEEE/ACM Trans. Networking*. 14 (6), 1246-1259
2. Fernando Paganini, Steven H. Low (2005) Congestion Control for high performance, stability and fairness in General Networks. *IEEE/ACM Trans. Networking*. 13 (1), 43-56.
3. Huajun Liu (2005) Research of the queue management algorithms application to the network processor. *J. Commu. Computer*. 2 (5), 112-117.
4. Kyle Halliday, Andrew Hurst and Jerom Nelson (2004) Analysis of next generation TCP. *Lawrence Livermore Natl. Lab*. 12-13.
5. Li-song Shao, He-ying Zhang and Wen-hua Dou (2006) General Window based Congestion Control: Buffer occupancy, network efficiency, packet loss. *Proc. 4th Ann. Commu. Networks Services Res. Conf.(CNSR'06)*. 2, 195-201.
6. Mehdi Farokhian Firuzi and Mohammad Haeri (2005) Active queue management in TCP networks bared on self tuning control approach. *Proc. 2005 IEEE Conf. Control Algorithms*. 3, 904-909.
7. Nishan R.Sastry and Simon S.Lam (2005) CYRF: A theory of window based unicast congestion control. *IEEE/ACM Trans. Networking*. 13, 330-342.
8. Peng Wang and David L. Mills (2006) Simple analysis of XCP equilibrium performance. *Proc. 40th IEEE Ann. Conf.* 2, 585-590.
9. Ramesh Johari and David Kim Hong Tang (2001) End-to-end congestion control for internet: delays and stability. *IEEE/ACM Trans. Networking*. 9 (6), 818-832.
10. Sally Floyd and Kevin Fall (1998) Promoting the use of end-to-end congestion control in internet. *IEEE/ACM Trans. Networking*. 7 (4), 1-15.
11. Sanjeeva Athuraliya and Steven H.Low (2001) REM: active queue management. *J. IEEE Network*. 15, 48-53.
12. Satyanarayan Reddy K and Lokanatha C. Reddy (2008) A survey on congestion control mechanisms in high speed networks. *IJCSNS Intl. J. Computer Sci. Network Security*. 8 (1),, 208-212.
13. Satyanarayan Reddy K and Lokanatha C. Reddy (2008) A survey on congestion control protocols for high speed network. IJCSNS Intl. J. Computer Sci. Network Security. 8 (7), 44-52.
14. Sherali Zeadally and Liqiang Zhan (2004) Enabling gigabit network Access to end users. *IEEE Proc. IEEE*. 92 (2), 340-353.
15. Steven H. Low (2003) A duality model of TCP and queue management algorithms. *IEEE/ACM Trans. Networking*. 11 (4), 520-546.
16. Steven L. Low, Achlan L.H. Andrew (2005) Understanding XCP: equilibrium and fairness. *Proc. IEEE Infocom*. 2, 1-12.
17. Yongguang and Lang (2005) An implementation and experimental study of the explicit control protocol (XCP). *Proc. IEEE 24th Ann. Joint Conf. of the Computer & Commu. Soc*. 2, 1037-1048.