P.Praveen Kumar, Dr.K. Rameswaraiah / International Journal of Engineering Research and Applications (IJERA) ISSN: 2248-9622 www.ijera.com
Vol. 3, Issue 3, May-Jun 2013, pp.917-920

# Improving The Performance Of Congestion Control In Wireless Networks

**P.Praveen Kumar[1], Dr.K. Rameswaraiah[2]**

[1]Asst Prof ,Dept of CSE, Samskruthi College Of Engineering & Technology, Ghatkesar, A.P, India
[2]Professor, Dept of CSE, ST.Peters Engineering College, Maisammaguda, Hyderabad, A.P, India

## ABSTRACT

**The Transmission Control Protocol (TCP) provides an end-to-end, ordered transmission of data over a logical connection which links two applications during their communication.TCP is a mainly concerned with moving data between applications and it uses the routing services provided by IP. The most widely used Internet protocol in the present days is TCP. It should have the knowledge of locating the applications and it need not know about the topology of the network. TCP actually takes the data from the source application and delivers to the destination application.TCP is mainly designed for using with unreliable systems such as IP, which only gets the data from one host to another and perform less error checking. The important feature of TCP with respect to its performance is congestion control algorithm.TCP mainly uses a number of techniques for achieving high performance and avoiding congestion collapse, which makes the network performance to fall by several levels of magnitude. The TCP congestion avoidance algorithm is the primary basis for the control of congestion.**

**Keywords:** Congestion Control, TCP, Internet Protocol.

## 1. INTRODUCTION:

The main protocol of the Internet protocol suite is Transmission Control Protocol (TCP) which is one of the two main components of the suite, and the other is Internet Protocol (IP) and hence it is commonly referred to as TCP/IP [1] [2] [3]. Most of the internet applications use TCP protocol for remote administration and file transfer. The performance of the Internet depends to a great extent as the TCP carries most internet traffic [4]. A particular version of TCP is defined by the congestion control algorithm as it employs. To establish the connection between machines, TCP uses a lower-level communications protocol and the Internet Protocol [5] [6]. An interface that allows streams of bytes to be sent and received converts the data into IP datagram packets which are provided by this connection. Datagram cannot guarantee about the arrival of the packets at their destination but TCP guarantees the delivery of bytes of data [7] [8]. Even if the network errors prevents the delivery, but TCP handles the implementation problems such as resending packets, and alerts the programmer in case if there is no network host or a connection is lost [9]. A socket represents the virtual connection between two machines and allows the data to be sent and received. TCP sockets are connected to a single machine and allow transmission of data through input and output streams. User Data gram Protocol (UDP) sockets are connected to multiple machines and only can send and receive packets of data [10] [11] [12].

## 2. ADVANTAGES OF TCP

Automatic Error Control: Data transmission through TCP is more reliable than transmission of packets through UDP. In TCP, data packets are sent through a virtual connection and they make sure that they have not been corrupted and guarantee the delivery of data and the lost packets are retransmitted [13]. In TCP, if the timer is disabled when the recipient sends an acknowledgment and if it is not received before the time, the packet is retransmitted. Reliability: The datagram packets will arrive out of order when the two machines participating in a TCP connection is transmitted by IP datagram Fig1. For any program reading information from a TCP socket the order of the byte stream will be disrupted and become unreliable [13] [14].

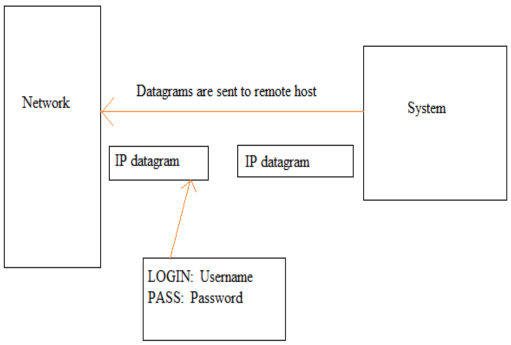


Fig 1 IP datagram for transport over the network.

The data will be passed to the interface of the socket. Ease of Use: To Store the information in datagram packets is not the efficient way of communication between the systems which creates complexity for the task of designing and creating software within a limited time for programmers [15]. TCP makes the programmers to think in a different way i.e. by making data packaged into datagram packets, it is otherwise treated as a continuous input and outputs.

## 3. AN OVERVIEW OF CONGESTION CONTROL AND THE ALGORITHMS:

TCP sockets follows the same of Unix programming, where communication is treated as file input and output where the developer writes to a network socket, a data structure, or a file [16] [17] [18]. Communication through TCP sockets is simpler than communicating through datagram packets. TCP sockets performs different operations such as TCP can establish a network connection to a remote host , it can Send the data to a remote host and can receive data from a remote host and can close the connection is shown in fig 2. There is a special socket providing the service of binding to a specific port number is found in TCP [19] [20]. This socket is generally observed only in servers, and performs the operations such as Binding to a local port, Accepting incoming connections from remote hosts and Unbinding from a local port. The two sockets are combined into different categories and they are used by a client or a server. TCP will use a sequence number in identifying each data byte. The sequence number of TCP helps in identifying the order of the bytes sent from each system and it helps in reconstruction of data when the data is lost during the process of transmission or fragmentation process. In the cumulative acknowledgment scheme of TCP, in the segment data field, the first byte of payload was set as the sequence number by the sequence number field and the receiver will send an acceptance by specifying the sequence number of the next byte for which they are expecting to receive.

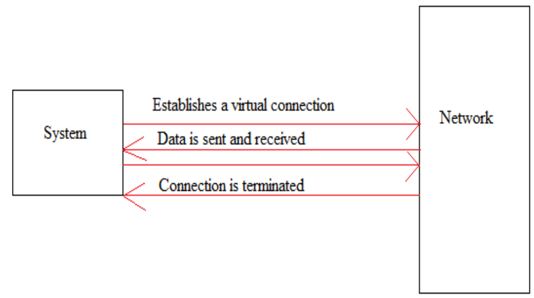


Fig 2. TCP virtual connection to transmit data

TCP uses an end-to-end flow control protocol in order to avoid the process of making the sender to send the data much fast for the TCP receiver to receive the data and process it. Congestive collapse is a condition where no communication occurs because of congestion and attains a packet switched computer network Congestion collapse is observed at choke points in the network, where the complete incoming traffic exceeds the outgoing bandwidth to a node. The local area network and a wide area network connecting points are choke points. When the network is in congestive collapse state, the packet delay and the loss of data are at high levels and traffic demand also high and poor service quality. When additional data packets were send more than required by the intermediate routers, makes additional packets discarded and expect the end points for retransmission of the information to the network. In the earlier times TCP implementations had worse retransmission performance. When the packet was lost then the end points sent extra packets of data which repeated the information which was lost, and doubled the data rate sent, exactly the reverse of what has to be done during congestion and this pressed the entire network into a congestion collapse making most of the packets loss. Congestion control controls the traffic entry into a telecommunication network, for preventing congestive collapse by avoiding over subscription of the processing capabilities of the intermediate networks and by reducing the rate of sending data packets. There are different ways for classification of congestion control algorithms: Based on the type and amount of feedback received from the network. Based on the feature of performance it is aiming to improve. Based on the equality criterion it uses. Two main components are required for the prevention of network congestion. They are a system in routers for reordering or dropping packets under overload, End-to-end flow control system which is designed into the end points for giving response to congestion. The correct end point performance usually should repeat the dropped information, but gradually repeat the information slowly. When all end points perform this, the congestion lifts and makes the good use of the network, and all the end points obtains a fair contribution of the available bandwidth. The mechanisms used for preventing congestion collapses are fair queuing ,scheduling algorithms, and random early detection (RED), in which packets are dropped randomly and slowed down the transmission process by proactively triggering the end points before the occurring of congestion collapse. Fair queuing mechanism is useful for mostly in the choke points routers having a small number of connections passing through them. A larger router mainly depends on RED. Some protocols such as TCP behave better under congested conditions where as UDP does not congestion control mechanism. Protocols building atop UDP is independent of congestion and transmits at a fixed rate should be able to handle the congestion in their own manner. Congestion should be kept out at the periphery position of the network in pure datagram networks, where the above described mechanisms are going to handle it. Congestion is very difficult to deal with the Internet backbone and instead of that low priced fiber-optic lines have made the cost cheaper in the Internet backbone with sufficient bandwidth for keeping congestion at the periphery. For implementing connection oriented protocols, such as TCP protocol, generally observes packet errors, delays for adjusting the speed of transmit. There are different congestion avoidance processes available such as TCP congestion avoidance algorithm which is the primary basis for control of congestion in the Internet. TCP uses a number of systems for achieving high performance and avoiding congestion collapse, which makes the falling of network performance by several orders of magnitude. These mechanisms control the congestion collapse by controlling the rate of data inflow into the network, maintaining the data flow below the rate at which trigger collapses. Senders use the acknowledgments for sending the data and for interfering network circumstances between sender and receiver of TCP. TCP senders and receivers can change the performance of the data flow by coupling with timers and is commonly referred to as network congestion avoidance. Recently the research is ongoing in the areas of TCP reliably handling loss, minimizing errors and managing congestion. In the present system, without the intermediate station the sender sends the packet data.

## 4. PROPOSED SYSTEM

Based on the consideration of traffic rates and buffer sizes, a router detection protocol was designed for the number of congestive loses of packet data. A foundation was built for all presently known host to host algorithms by the proposal of host to host congestion control. It includes: The basic principle of probing the available network resources and to estimate the congestion state in the network a Loss and delay-based techniques are used and the Techniques to detect packet losses quickly. A sliding window based flow control is specified by the TCP standard. This control has several mechanisms. The buffered data are packetized into TCP packets which contains a sequence number of the first data byte in the packet. The prepared data packets are transmitted to the receiver using the IP protocol. It transmits a new portion of packets as soon as the sender receives delivery confirmation for at least one data packet. Until the receiver clearly confirms the delivery of the block, the sender is responsible for a data block. Finally, the sender may decide that a particular data block has been lost and start recovery procedures. The receiver forms an ACK (acknowledgement) packet that carries one sequence number and several pairs of sequence numbers by admitting the data delivery. The data blocks which having smaller sequence numbers have already been delivered is defined by a collective ACK (acknowledgement). The sequence numbers of delivered data packets are clearly indicated by a selective ACK. However, we will discuss a notion of ACK packets as a separate entity without loss of generality.

## 5. CONCLUSIONS:

In this paper, we have discussed and built a foundation for host-to-host congestion control principles. Based on the consideration of traffic rates and buffer sizes, we have designed and implemented a router detection protocol for the number of congestive loses of packet data. In this paper, we have described the various approaches to TCP congestion control. The basic problem of eliminating the congestion collapse phenomenon to problems of using available network resources effectively in different types of environments are surveyed. In this paper, we have discussed about congestion control proposals which are focused on environments in which the data packets are reordered regularly by using this proposal efficiency can be improved. New challenges for TCP congestion control has introduced with the technology advances.

## REFERENCES:

[1] J. Postel, “RFC793—transmission control protocol,” *RFC*, 1981.
[2] C. Lochert, B. Scheuermann, and M. Mauve, “A survey on congestion control for mobile

ad hoc networks," *Wireless Communications and Mobile Computing*, vol. 7, no. 5, p. 655, 2007.

[3] J. Postel, "RFC791—Internet Protocol," *RFC*, 1981.

[4] A. Al Hanbali, E. Altman, and P. Nain, "A survey of TCP over ad hoc networks," *IEEE Commun. Surveys Tutorials*, vol. 7, no. 3, pp. 22–36, 3rd quarter 2005.

[5] J. Widmer, R. Denda, and M. Mauve, "A survey on TCP-friendly congestion control," *IEEE Network*, vol. 15, no. 3, pp. 28–37, May/June 2001.

[6] H. Balakrishnan, V. N. Padmanabhan, S. Seshan, and R. H. Katz, "A comparison of mechanisms for improving TCP performance over wireless links," *IEEE/ACM Trans. Netw.*, vol. 5, no. 6, pp. 756–769, December 1997.

[7] K.-C. Leung, V. Li, and D. Yang, "An overview of packet reordering in transmission control protocol (TCP): problems, solutions, and challenges," *IEEE Trans. Parallel Distrib. Syst.*, vol. 18, no. 4, pp. 522–535, April 2007.

[8] S. Low, F. Paganini, and J. Doyle, "Internet congestion control," *IEEE Control Syst. Mag.*, vol. 22, no. 1, pp. 28–43, February 2002.

[9] G. Hasegawa and M. Murata, "Survey on fairness issues in TCP congestion control mechanisms," *IEICE Trans. Commun. (Special Issue on New Developments on QoS Technologies for Information Networks)*, vol. E84-B, no. 6, pp. 1461–1472, June 2001.

[10] M. Gerla and L. Kleinrock, "Flow control: a comparative survey," *IEEE Trans. Commun.*, vol. 28, no. 4, pp. 553–574, April 1980.

[11] J. Nagle, "RFC896—Congestion control in IP/TCP internetworks," *RFC*, 1984.

[12] C. A. Kent and J. C. Mogul, "Fragmentation considered harmful," in *Proceedings of the ACM workshop on Frontiers in computer communications technology (SIGCOMM)*, Stowe, Vermont, August 1987, pp. 390–401.

[13] S. Floyd and K. Fall, "Promoting the use of end-to-end congestion control in the Internet," *IEEE/ACM Trans. Netw.*, vol. 7, no. 4, pp. 458–472, August 1999.

[14] V. Jacobson, "Congestion avoidance and control," *ACM SIGCOMM*, pp. 314–329, 1988.

[15] Z. Wang and J. Crowcroft, "Eliminating periodic packet losses in 4.3– Tahoe BSD TCP congestion control," *ACM Computer Communication Review*, vol. 22, no. 2, pp. 9–16, 1992.

[16] V. Jacobson, "Modified TCP congestion avoidance algorithm," email to the end2end list, April 1990.

[17] M. Allman, V. Paxson, and W. Stevens, "RFC2581—TCP congestion control," *RFC*, 1999.

[18] S. Floyd and T. Henderson, "RFC2582—the NewReno modification to TCP's fast recovery algorithm," *RFC*, 1999.

[19] S. Floyd, T. Henderson, and A. Gurtov, "RFC3782—the NewReno modification to TCP's fast recovery algorithm," *RFC*, 2004.

[20] M. Mathis, J. Mahdavi, S. Floyd, and A. Romanov, "RFC2018—TCP selective acknowledgment options," *RFC*, 1996.

**BIOGRAPHY:**

***P.Praveen Kumar*** obtained his B.Tech graduation from JNTUH in 2008, M.Tech degree from JNTUH in 2011.Having 2 years of Industry experience and 2 years of teaching experience, working as Assistant Professor in Samskruthi College Of Engineering & Technology, Ghatkesar, Ranga Reddy, AP-India, Presented 2 international papers at various journals and attended workshops at various institutions. Interested areas of research are: Data Mining and Ware housing, Network Security Software Engineering, Information Security.

***Dr.K.Rameshwaraiah*** obtained his B.Tech graduation from JNTUH in 2001,M.S degree from UK in 2004and Ph.D from SBU,UK in 2011.Having 12 years of teaching experience , working as professor in ST.Peters Engineering College, Maisammaguda, Hyderabad, AP-India ,Presented more than dozen international and national papers at various journals and attended more than 20 conferences and workshops across the globe. Interested areas of research Software Engineering, Image Processing, cyber terrorism&Security ,Network Security ,Information Security Data Mining and Ware housing and MANETS..