Survey of Different Congestion Control Techniques in Wired Networks

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ABSTRACT
Congestion is very common in wired networks, but classical congestion control techniques do not suit well to the resource constrained environments. Congestion has been considered as one of the basic issues in networks. The ultimate motto of congestion control remains very same for also type of communication networks. But the workings of congestion control techniques differ with respect to communication technologies. The difference between congestion avoidance and congestion control are explained. Congestion avoidance schemes like choke packet, probe packet, increase in cost, piggyback and acknowledgement schemes are studied in this paper. Efficiency, responsiveness, convergence, fairness, robustness and simplicity are discussed as a part of performance metrics. Importance of decision function, increase and decrease algorithms are discussed as congestion avoidance schemes. These schemes are demonstrated with shared paths, heterogeneous routers, and different workloads of data. The main goal of this paper is to make an overview of the existing congestion control techniques for constrained environment and propose a need for the development of new flexible technique. It has addressed all restrictions set by the environment and at the same time be generic enough.

Key Words: Congestion, wired network, AQM, TCP.

INTRODUCTION
All A router output queue could suffer congestion conditions when it receives more traffic than its line capacity. Uncontrolled congestion could cause high queue build up resulting in increased queuing delays and packet loss [1,3]. One potential solution is to allocate more resources, i.e., provide higher line capacities. Unfortunately this is an expensive solution, especially in access networks. A carefully designed congestion
control mechanism is more cost-effective. It also provides opportunities for differential treatment of packets under congestion. Active queue management of router queues attempts to provide congestion control by monitoring the congestion state of a router queue and pro-actively dropping [2] or “marking” packets before any impending congestion could cause reduced performance. Several schemes have been proposed for active queue management. These schemes have been modeled and evaluated for stability, throughput and delay performances [4]. Our goal in this paper is not to propose a scheme that outperforms any of these existing schemes. Rather, we would like to examine a scheme that performs reasonably well compared to the existing scheme but we believe is more robust and scalable with varying system and network parameters including buffer size, link capacity and network load.

Those characterized data traffic as fluid used a set of differential equations to describe the Active Queue Management (AQM) policy and the router queuing process in [5] the end-to-end congestion control mechanisms were employed in transmission control protocol (TCP) flow control.

In this paper, we first study the applicability of various load measures of a router queue and justify the choice of using acceptance rate and departure rate.

The continuous increasing demand for data transfer through the internet is exceeding the available network resources. This is mainly due to different applications that run concurrently and require high transfer data rates. TCP protocol which is the most important and popular protocol for data exchange through the internet, controls network congestion via measured packet loss rate. Congestion control in current Internet still is a critical issue. The number of Internet users is rapidly increasing and therefore the amount of data to be carried also increases. The main problem of Congestion occurred when arrival rate to a router is greater than its departure rate. Each router in the network uses queue management (QM) and scheduling as two classes of algorithms that are related to congestion control. A queue management system is used to control queues. Queues of people form in various situations and locations in a queue area. The process of queue formation and propagation is defined as queuing theory. Queues exist in two main forms. The QM algorithms try to control the length of packet queues by dropping packets when appropriate. Scheduling algorithms on the other hand, determine which packet to drop next and which is to send and also used to manage the allocation of bandwidth among flows. Buffer management schemes at routers decide when to drop a packet and which packet to drop. The simplest queue management scheme is drop-tail, where each packet is treated identically. With drop tail, when the queue is filled to its maximum capacity, the newly arriving packets are dropped until the queue has enough room to accept incoming traffic. Once a queue has been filled, the router begins discarding all additional datagram’s, thus dropping the tail of the sequence of datagram. The loss of datagram causes the TCP sender to enter slow start. It’s worth nothing that most of the Internet today still runs drop-tail gateways.
One of the key requirements of any congestion control scheme is to obtain a robust measure of system load. The offered and processed load constitutes the system load. The offered load is a measure of new load on a system whereas the processed load is what the system has already seen and processed. In a congestion control scheme, offered load measurement is necessary to quickly respond to a sudden burst of traffic but not sufficient. It is possible that the offered load in a given time is not very high but due to backlog from load offered at earlier instants the processed load can still be high. Hence a combination of both offered load and processed load should be used for robustly controlling congestion under different traffic arrival patterns.

We now discuss the applicability of some of the well known system load measures.

**Queue Length** - Queue length is one of the most widely used measure of load. It is used naturally in drop tail schemes where incoming packets are dropped when the queue reaches its capacity. It is also used in RED and several of its variants. At any instant, the queue length provides a measure of the load remaining in the system, i.e., the load arriving at the queue minus the processed load. Hence it seems the most natural measure of congestion. The problem with queue length is that it is not robust to changes in the system. If an AQM scheme sets target queue length(s), these targets have to be changed with changes in line capacity. A router operator might be willing to operate with longer queues when capacity increases. This issue also arises in situations where the available line capacity for best-effort traffic changes due to any co-existing bandwidth reservation mechanisms, especially in networks supporting multiple service classes. In this scenario, it is not possible to determine the available capacity a priori.

**Delay** - Queuing delay offers a more robust measure. It is essentially a measure of the queue length normalized by the line capacity. Any queuing delay thresholds need not be changed with changes in line capacity. The problem with using queuing delay is an appropriate choice of an operational value. A typical network consists of several hops (and at times there are multiple cards a packet has to go through per hop). The choice of an operating delay at each hop is difficult because a smaller value could result in under utilization and a high value could result in large delays. There is no simple way to budget an end-to-end delay requirement across hops or in some cases, even across multiple components in the same hop.

**Departure Rate** - Departure rate is defined as number of bits transmitted or serviced in a given time interval. This rate provides a very good measure of processed load. Under steady state conditions, processed load could be a good measure of the system load. When there are sudden changes in the system load, just measuring processed load is not sufficient because the offered load might be much higher than the processed load. Congestion control based on departure rate only could result in slow response under a sudden burst of traffic. Line occupancy is departure rate normalized by capacity. Since this is dimensionless, controls based on this measure are robust against capacity changes.

**Acceptance Rate** - Acceptance rate is the number of bits in packets arriving at the router queue minus the number of bits in dropped packets in a given time interval. It provides an accurate measure of offered load.
CONGESTION CONTROL MECHANISM

The congestion control mechanism of various versions of TCP provides better throughput in a wired network. Best effort find in wired network and deals with congestion effectively, where the packet loss is mainly due to congestion at various nodes and routers.

Network Congestion occurs when the aggregate demand for bandwidth exceeds the available capacity of a link and when the arrival rate to the router is greater than its departure. To solve this problem, it is necessary that the router must implement effective queuing algorithm that governs how packets are buffered while waiting to be transmitted. According to the dynamic of input packets and available link bandwidth, queue management becomes very complex. For this reason it is better to use an intelligent algorithm.

**Congestion avoidance system control**

Congestion Control and Congestion Avoidance are two known solutions which address the problem given below. Congestion results occurred when the aggregate demand for resources as bandwidth exceeds the capacity of the link. Congestion is characterized by delay and loss of packets in delivery. In TCP, congestion is said to have occurred when the sender receives three duplicate acknowledgments (dupacks) or when a timeout (packet loss) occurs, resulting in wastage of resources. In congestion control, system controls the network parameters after realizing congestion (reactive) whereas, in congestion avoidance, system controls the network parameters before congestion (proactive).

**Congestion Control in Wired Network**

TCP was designed for use in wired networks. It becomes imperative to ask whether the same mechanism will also work in the case of wireless networks. In TCP, the window size Wireless is reduced whenever packets are lost assuming that the loss is due to traffic congestion along the path of the flow under consideration. This assumption is not valid in wireless networks since the channel may have been in a poor state thereby causing the packet to be lost at the physical layer (OSI Model) itself. The statistics of this type of loss may be very different from a packet dropping at a router queue due to traffic congestion.

**The adaptive window based congestion control**

This type of adaptive window based congestion control used by TCP for wired network may not be appropriate for wireless network. This is due to the time varying nature of a wireless channel and interference due to other nodes causing packet loss, which is different from packet loss due to congestion. But, TCP’s congestion control mechanism does not discriminate packet loss due to congestion and that due to bad channel or interference, rather apply the same congestion control mechanism for both.

**Adaptive Window Management traffic control**

Most of the traffic in the Internet is TCP traffic. TCP’s congestion control in wired network is based on Adaptive Window Management technique. In this technique, congestion window (cwnd) increases or decreases based on packet drops and duplicate acknowledgements.
CONGESTION CONTROL TECHNIQUE

Cross layer
Cross layer Congestion control technique provides performance improvement In terms of throughput and window size variations. Instead of usual congestion control technique, we propose a cross layer technique involving TCP and MAC (Medium Access Control) layer. TCP layer performs the Windowing flow control and MAC layer varies transmission Power of wireless nodes depending on the channel condition and interference.

Per-flow for controlling the Window
Congestion control is performed independently in slotted time of RTT (Round Trip Time) on each connection, which is also the control latency in its feedback loop .well controlling the receive window is challenging: The receive window should be small enough to avoid unicast congestion, when we use per flow and also large enough for good performance.

Random Early Detection Algorithm
B. Braden et al., discussed that the Random Early Detection Algorithm (RED) had been proposed to be mainly used in the implementation of AQM (Active Queue Management) [6]. On the arrival of each packet, the average queue size is calculated by using the Exponential Weighted Moving Average (EWMA) [7, 25]. The computation of the average queue size is compared with the minimum and the maximum threshold to establish the next action.

Choke Algorithm
Konstantinos Psounis et al., proposed CHOKe algorithm [8][9], whenever the arrival of a new packet takes place at the congested gateway router, a packet is drawn at random from the FIFO buffer, and the drawn packet is then compared with the arriving packet. If both belong to the same flow in the network then both are dropped, else the packet that was chosen randomly is kept integral and the new incoming packet is admitted into the buffer with a probability depending on the level of congestion. This computation of the probability is the same as in RED. space.

Blue Algorithms
Rong Pan et al., discussed the basic idea behind the RED queue management system is to make early detection of the incipient congestion and to feed back this congestion notification and allowing them to decrease their sending rates accordingly. The RED queue length gives very less information about the number of contending connections in a shared link of the network. BLUE and Stochastic Fair Blue Algorithms (SFB) were designed to overcome the drawbacks of the problems caused by the RED techniques, the TCP flows are protected by using packet loss and link idle events against non-responsive flows. SFB is highly scalable and enforces fairness using an enormously miniature amount of state information and a small amount of buffer space. The FIFO queuing algorithm identifies and limits the non-responsive flows based on secretarial similar to BLUE [9].
Random Exponential Marking Algorithm

According to Debanjan Saha the Random Exponential Marking Algorithm (REM) [10] is a new technique for congestion control, which aims to achieve a high utilization of link capacity, scalability, negligible loss and delay. The main limitations of this algorithm are: it does not give incentive to cooperative sources and a properly calculated and fixed value of \( \phi \) must be known globally.

Fair Queuing Algorithms

Alan Demers et al., proposed the Fair Queuing Algorithms [11] and Stochastic Fair Queuing Algorithms [12] are mainly used in the multimedia integrated services networks for their fairness and delay bounding in the flow. The frame-based class of FQ is called Weighted Round Robin [13], where a router queue scheduling method is used in which queues are serviced in round robin fashion in fraction to a weight assigned for each flow or queue.

Virtual Queue Algorithm

The Virtual Queue Algorithm (VQ) is a radical technique proposed by Gibben and Kelly [14]. In this scheme, a virtual queue is maintained in link with the same arrival rate as the real queue. However, the capacity of the virtual queue is smaller than the capacity of a real queue. When the packets are dropped virtual, then all packets already enqueued in the real queue and all new incoming packets are marked until the virtual queue becomes empty again.

Adaptive Virtual Queue Algorithm

R.J. Gibben et al., discussed in the Adaptive Virtual Queue algorithm [15] the capacity of the link and the desired utilization maintains a virtual queue at the link. The capacity and buffer size of the virtual queue is the same as that of the real queue. At the arrival of each packet, the virtual queue capacity is updated. The adaptation of virtual queue algorithm does not suitably follow the varying traffic pattern at flow in the network, and it is also FIFO based methodology.

Related Work

This paper [16] presents performance evaluation of RED with ECN Support using Standard RED as the evaluation baseline with respect to their abilities of improving performance of congested routers by keeping packet drop rate low. The performance is evaluated for FTP-like bulk data TCP flows and UDP based traffic such as Constant Bit Rate(CBR) based on average throughput results of TCP sources. Although ECN leads to fewer packets drops, it does not necessarily lead to improved throughput for TCP transfers. On the other hand, in no case does ECN seem to lead to an actual degradation in TCP performance. In summary, the TCPIECN sender has a competitive advantage over ECN unaware senders because it reacts faster to incipient congestion and can thus avoid unsuccessful segment transmissions. Nevertheless, TCPIECN has the same mechanisms as standard TCP for detecting improved network conditions, and, therefore, is unable to deliver guaranteed improved good put.

In this paper [17] studies the performance and the efficiency of TCP under constant congestion when using those methods and shows that ECN actually improves the efficiency of TCP without harming its
performance. Random Early Detection (RED) and other AQM mechanisms has been subject of criticism regarding their ability to improve the behavior of bottleneck links. This paper is an experimental study of Drop-Tail, RED and RED+ECN (Explicit Congestion Notification) when operating over a constantly congested link.

In this paper [18] a simple modification to the drop tail algorithm in which a generic queue management controls methodology in TCP/IP networks, that case we dynamic change queue length in our wired network. Queue length (varying maximum Queue limit for drop tail policy) while the buffer size of the input port is fix gives the good performance. The proposed scheme shows the improvements as compare to drop tail, and RED algorithm in terms of drop rate ,and buffer utilization in dynamic network environment (such as topologies and traffic condition).

Now in [19] author proposed a delay based uni directional delay jitter based TFRC with end-to-end semantic over wired wireless networks. This scheme provide smooth sending rate and TCP friendly characteristics like standard TFRC, even it also increase the throughput by estimating the available bandwidth in wired-wireless networks with bursty nature of background traffic. Simulation results show performance improvement without intrusiveness issue and even if background traffic is bursty over wired-wireless networks.

The work [20] extends AQM control design for single network systems to large-scale wired network systems with time delays at each communication channel. A system model consisted of several local networks is first constructed. The stability condition guaranteeing overall stability is subsequently derived using Lyapunov stability theory [21].This research has modeled and analyzed stability of large-scale wired networks under control where the AQM strategy uses RED to fulfill the queue management. The problem of feedback control has been solved for the LWNS with delayed perturbations in the interconnections and a new condition ensuring the overall loop stability is presented.

The traditional technique used to control traffic congestion has been Drop Tail [22]. Drop Tail tends to penalize bursty connections by dropping all the incoming packets only when the buffer is full. This will continue until the number of packets in the queue is below the maximum queue length and congestion is eliminated. This method has two main drawbacks: ‘Lock–Out’ which is a result of global synchronization, and ‘Full Queue’ which results in high packet delay [23].

**PROBLEM STATEMENT**

In This paper we focus number of different congestion control using queue system but all are feasible for TCP transmissions but we further design dynamic drop tail queue mechanism and provide fairly serve TCP as well as UDP data under wired communication.
CONCLUSION

In this paper an overview of congestion control techniques for constrained environment is made, scenarios of network performance, when congestion can appear and become a catastrophic are underlined in previous works. This article has discussed the performance and the behavior of different congestion control mechanisms and investigated the effects of each congestion control technique. Also it provided an analysis to some TCP variants and explanation to the new TCP’s that developed to support new different networks applications. TCP Tahoe and TCP Reno are mostly applied over many wired and wireless applications because of the effective congestion control mechanisms. These mechanisms provide varying in size of congestion window depending on ACK status, thus when packets acknowledged the window size is increased and decreased when detect lost in packets.

In future we propose an idea of improving congestion control mechanism by applying varying queue length method to improve performance of network.

REFERENCES


