A SURVEY ON CONGESTION CONTROL MECHANISMS

Mahammad Mastan[1], P.Suresh Verma[2], Mohammed Ali Hussain[3]

[1]*Associate Professor in CSE, Malla Reddy Engineering College, Telangana, India*
[2]*Professor in CSE, Adikavi Nannaya University, Andhra Pradesh, India*
[3]*Professor in ECM, KL University, Andhra Pradesh, India*

Abstract

Congestion Control performs a very important role in Computer Networks, Modern Telecommunication, Internet and both wired and wireless communications are being intended for high-speed communication of large amounts of data. Due to lack of proper Congestion control mechanism the congestion collapse of such networks would become highly complex. A network with Streamed media traffic is a challenge for Congestion control because of sensitivity. Many researchers have motivated and challenged over the last decade to increase more congestion control protocols and mechanisms that go well with the traffic and provides fair maintenance mutually for communications in unicast and multicast. We survey a major congestion control mechanisms, TCP-Friendliness and categorization characteristics and a most recent for congestion control mechanisms designed for network and estimate their characteristics in this paper.

***Keywords**: Congestion, Metrics of Congestion Control, UDP Traffic and TCP-Friendliness*

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1. INTRODUCTION

Congestion control over network, for all types of media traffic, from the last decade it has been dynamic area of research [1]. This is because of promising increase in the audiovisual traffic of digital convergence. In a network variety of applications are existed and rise on its ability of media with streaming equally on demand or in real-time for example voice over IP (VoIP) video streaming and conferencing, and (VoD) video on demand. In an amount of users for these network application is regularly increasing therefore resulting in congestion.

The complete networks applications do not use TCP and therefore do not allow fair allocation with the existing bandwidth. Therefore, the result of inequality of the non-TCP applications did not have much impact because of mainly traffic with network uses TCP-based protocols. Nevertheless, the quantity of streaming audio/video applications while Internet video conferencing, audio and video players, and corresponding category of real-time applications is frequently increasing and soon expected as there have a tendency for raise in the proportion in non-TCP traffic. Here views of the fact that the applications frequently carry out no merging mechanisms of TCP compatible congestion control, network applications consider challenging TCP flows in an awkward manner. The TCP flow decrease their data rates and try to break up congestion, where a non TCP flows maintains on the way to transmit at their unique rate. It is highly biased condition will lead to starvation of TCP-traffic i.e.., congestion fall down [2], [3], It set out the disagreeable circumstances where the network bandwidth is accessible almost totally in use of packets which are leftover due to the congestion prior to reach their destination. Due to, this is popular to identify appropriate congestion control mechanisms for well matched i.e., for non-TCP traffic through the mechanism like rate adaptation TCP. The non TCP applications will become TCP friendly by using these mechanisms, hence guide to a fine allocation of bandwidth. Unicast is a discussion type of communication in networks where multicast will be one to many. Multicast is beneficial more than unicast particularly in bandwidth reduction, however unicast is until extensively widen communication variety network.

2. CONGESTION CONTROL SYSTEM THEORY

A Congestion control concerns in controlling the network traffic in a telecommunications network, to prevent the congestive subside by trying to keep away from the unfair allocation of several processing or capabilities of the networks and making the proper resource dropping steps by dropping the rate of packets sent.

Goals that are taken for the assessment procedure for algorithms of congestion control:
1). It accomplishes more bandwidth consumption.
2). headed for congregate to efficiently and fairness quickly.
3). headed for decrease oscillations related to amplitude.
4). headed for maintain a more responsiveness.
5). headed for coexist moderately and well-matched through long recognized broadly used protocols.

The Metrics [24] that have been set for Congestion control are:
i. The Convergence speed - The Convergence speed estimates time accepted to reach the equilibrium state.
ii. Smoothness - The Smoothness reflects the oscillations magnitude through multiplicative reduction and it rely on oscillations range.
iii. Responsiveness - The Responsiveness is calculated through the round trip times (RTTs) to attain equilibrium.

The discrepancy among Convergence Speed and Responsiveness is associated towards a particular flow and the convergence is related to the method.
I. Efficiency - The Efficiency is the standard flow throughput per step or round trip time, as the method is in stability.
II. Fairness: The fairness classifies the equitable allocation of resources among the flows in a mutual traffic jam link.

3. CONGESTION CONTROL ALGORITHMS CLASSIFICATION

The Congestion Control Algorithms are classified mainly based on the below criterion:
i. Can be classified by the category and size of the response received from the network
ii. Can be classified by increasing the deploy capability on the network. Simply the sender needs for the alteration (or) sender and receiver require only the router needs for the modification (or) tall the three: sender, receiver and routers needs for the modification.
iii. Can be classified by the aspect of performance. To make improvements in performance: high bandwidth networks, lossy links, fairness, advantage to short flows, variable-rate links
iv. Can be classified by the fairness criterion it uses minimum potential delay, proportional, max min.

Algorithms of congestion control are able to use network awareness as a standard. Following are three categories for the congestion control mechanisms. The Black box consists of a collection of algorithms based on the concept that reflects on black box network type, affected of no awareness of its condition much than the binary response leading to congestion. A Grey box is grey group approximate that use the dimensions to evaluate accessible bandwidth and the contention level or still the provisional features of congestion. Because of the opportunity of incorrect assessment and measurement dimensions, a network is examined as grey box. A Green box contains bimodal congestion control through which it can calculate explicitly the fairs hare, also the network assisted limitation, as a network transfer through its transport layer. So, it is considered as green box.

3.1 Black Box

A black box is a classified congestion control and it is also known as the Blind Congestion Control method and this methodology uses the AIMD (Additive Increase Multiplicative Decrease) algorithm. AIMD implements TCP window adjustments. Stability is achieved in set of circumstances with algorithms everywhere the authoritative ordinance of emulate flows exceeds the available bandwidths of the channel. Congestion control mechanism is conventional TCP and is predicated on the fundamental plan of AIMD. The TCP-NewReno, TCP-Tahoe and TCP-Sack, preservative rise in stage and is assumed exactly as in AIMD, where protocols and mechanisms are within the congestion control phase. In packet drop case, as an alternative of the multiplicative reduction, an extra conservative method is employ in TCP-Tahoe. Protocol mechanisms enter again the slow-start phase and congestion window reset. However, in TCP-sack and TCP-NewReno, as soon as sender collects 3 DACKs, when both windows and slow-start threshold phase is applied a multiplicative reduction is used. The protocol mechanism remains at the Congestion control phase in such case. It goes through the slow start segment as in TCP Tahoe, as soon as retransmission timeout expires. A Highspeed-TCP change with the reaction function in environment through high interruption bandwidth product, increases congestion window extra violently ahead getting the acknowledgment, furthermore it reduces window extra quietly ahead a failure occurrence. BIC (Binary Increase Congestion Control Protocol) TCP use a hollow raise rate a reference, following all congestion events awaiting the window is equivalent to that previous to the occurrence, to maximize the utilization time of the network. CUBIC TCP - It is not as much of powerful and further organized BIC derivative, everywhere the window is a cubic task of time because of the final congestion event, by means of modulation point put to the window former to the occurrence. A current advancement of Additive Increase Multiplicative Decrease (AIMD) by means of Fast Convergence is not found on a recent algorithm, although an optimization of AIMD and the system of convergence permit the algorithm to congregate quicker and attain superior efficiency. In nonlinear congestion control algorithms Binomial Mechanisms forms a new class and identified as algorithms of type binomial congestion control. It is also called as binomial since the control mechanism that is found on the contribution of two added algebraic conditions with diverse exponents. TCP friendly non linear congestion control algorithm is SIMD which controls the congestion by utilizing the past information. General AIMD Congestion Control induces congestion control mechanism of AIMD with parameterizing, ‘α’ additive raise value and ‘β’ multiplicative reduce ratio.

3.2 Grey Box

Grey box is also called as congestion control with measurement based. Principles of TCP confide on packet losses as virtual signal of congestion signal as congested links. There are a number of reasons for indicating the congestion one of the common reasons is the loss of packet and random bit corruption is the main cause for the packet loss and is caused when bandwidth is still accessible. At the sender side acknowledgement-based loss detection can be changed by the interweave traffic on the invert path. The loss of packet, as a binary response, cannot specify the stage of contention earlier than the amount of congestion. So, a well-organized window alteration approach should replicate diverse network situation, which cannot all be apprehend simply by packet drops. Some measurement form transport protocols collect in sequence on present network conditions. The queuing delay is approximate by TCP Vegas. To make a regular number of packets per flow the window is linearly increased and decreased in the network.

The FAST TCP achieve the same stability as vegas, however uses relative control as a substitute of linear raise, and deliberately scales the increase behind as the bandwidth increase with the aim of ensure stability. The loss cause the window to be retune to the sender's estimation of the bandwidth interruption product in TCP-Westwood which is the minimum calculated round trip times the experimental pace of getting acknowledgement. TFRC is based on the rate based congestion control mechanism, which expects to economically contend for bandwidth with flows during network.

3.3 Green Box

A Green box contains bimodal congestion control mechanism by which it can calculate explicitly the fairs hare of the system flow in the network. Bimodal congestion prevention and Control mechanism for each flow the fair share of the whole bandwidth that have to be allocated is calculated at any point through the execution of the method flow. A RED (Random Early Detection) packet is randomly dropped in fraction to the router's queue size, triggering multiplicative reducing in several flows. In ECN (Explicit Congestion Notification) routers are enabled to probabilistic mark a bit in the IP header instead of dropping packets, near intimate the end-hosts of imminent congestion when the length of the queue exceeds a threshold [21]. The VCP (variable structure congestion control protocol) uses two ECN (Explicit Congestion Notification) bits to clearly get the feedback of the network status of congestion.

4. CONGESTION CONTROL ALGORITHMS

4.1 RED (Random Early Detection) Algorithm

RED Algorithm B. Braden et al., had been proposed to be primarily used in the performance of AQM (Active Queue Management) [4]. The average queue size is calculated upon the arrival of each packet, by means of the Exponential Weighted Moving Average (EWMA) [5]. The calculation of the standard queue size is differentiating by means of minimum and maximum threshold to create after that accomplishment.

4.2 Choke Algorithm

This algorithm was proposed by Konstantinos Psounis et al., [6, 7], every time the emergence of a new packet take place at congested gateway router, randomly a packet is drawn from the FIFO buffer, and the drawn packet is then differentiate through the arriving packet. But in cooperation together belong to the same flow in the network subsequently both are dropped, also the packet that was chosen randomly be kept integral as well as the fresh arriving packet is admit to the buffer through a possibility depending going on the phase with congestion. It will be the computation of possibility is the similar as RED. This is stateless and easy algorithm where no particular data structure is required. Though, this algorithm be not present fit while amount of flows is huge when compare to the buffer space.

4.3 Drop Tail Algorithm

Drop Tail (DT) algorithm was deliberates by F. Postiglione et al., [15] have a great accuracy, easy and generally make use of the algorithm in the present networks, as the packets drops from the full queue buffer tail. This algorithms major advantage is suitability, effortlessness to its decentralized nature and heterogeneity. Though, this algorithm also has some severe disadvantage, such as no security alongside the mischievous or non responsive flows, lack of fairness and no comparative QoS (Quality of Service). QoS is of scrupulous apprehension for constant transmission of multimedia information and high- bandwidth video [15]. This type of transmitting the content is complicated in the current Internet and network through DT.

4.4 REM (Random Exponential Marking) Algorithm

REM as specified by Debanjan Saha is a fresh method meant for congestion control, as it focus to accomplish more consumption of link scalability, capability, delay and minor loss. Its major limits are it does not give reason to cooperative sources and accurately considered and rigid value of φ have got to be famous internationally [8].

4.5 VQ (Virtual Queue) Algorithm

The VQ algorithm is an essential method deliberates by Gibben and Kelly [12]. A virtual queue is maintained in this scheme and a link with similar approaching pace as real queue. Though, the ability of the implicit queue is minor than the capability of an actual queue. As soon as the packets are dropped virtual, after that all packets by now enqueued in real queue and every new incoming packet are noticeable awaiting the virtual queue become vacant again.

4.6 Fair Queuing Algorithms

The Fair queuing algorithms was deliberates by Alan Demers et al., [9] Stochastic Fair Queuing Algorithms [10] are primarily use in the multimedia incorporated services networks meant for their interruption bounding in the flow and fairness. The frame establishes a class of FQ is known as weighted round robin, everywhere a router queue arrangement system is used in queues are service in fashion like round robin in fraction to a weight assign for every queue [11].

4.7 Adaptive Virtual Queue Algorithm

The Adaptive virtual queue algorithm was deliberates by R.J. Gibben et al., the ability of the link and the needed consumption maintains a virtual queue at the link. The aptitude and buffer size of the virtual queue is the identical as that of the real queue. On the arrival of each packet, the virtual queue capacity is updated. The adjustment of virtual queue algorithm does not suitably follow the varying traffic model at flow in the network, and it is also FIFO base approach [13].

5. CATEGORIZATION OF CONGESTION CONTROL PROTOCOLS

Congestion control protocols were classified as four main groups according to an amount of characteristics in their method of work [20]. The subsequent shows the applicable category of arrangement.

5.1 Window Based Congestion Control

The window based protocols are build based on the technique of mechanism related to congestion window based, and it is used at the dispatcher or recipient side [23]. The gap to facilitate window is held in reserve for every packet as soon as the sent packet is acknowledged to be arriving the slot becomes free and allows transmission only when free slots are valid. In absence of congestion the size of window increases and decreases when congestion occurs in the network [14].

5.2 Rate-Based Congestion Control

Rate-Based protocols are built based on the adaptation of their speed of transmission following to several incorporated advice algorithm that intimates regarding congestion when it exists. The rate based algorithms is separated in to easy mechanisms and congestion control. The marks of saw tooth throughput form are used and this kind of scheme frequently is not completely compatible with the streaming media applications on which the Simple schemes are based. The current research tends to create the modification rate mechanisms ensuring the equitable antagonism among TCP and non TCP flows equally in the network.

5.3 Single Rate Congestion Control

Single rate congestion control mechanisms are usually acquired by every unicast congestion control protocols. Transmissions in unicast have single one receiver, hence transfer rate is adapted in accordance to the receiver position. Multicast broadcast can assume the single rate approach also, everywhere the sender streams the data among similar rate to everyone recipients of the multicast group in the network.

5.4 Multi Rate Congestion Control

The covered multicast move towards in multi rate congestion control, because multi-layering enables to separate data of the sender into dissimilar layers to be sent to diverse multicast groups. All receivers join the major feasible amount of groups allowed by the traffic jam in the mode to dispatcher. While data value to be send to the receiver becomes high while union additional multicast groups. As the characteristic is mainly evident in multicast video periods where more the class that the recipient accept in, is additional layers that the recipient encounter, and also more improved feature of video is temporarily, designed for previous mass data, the transmit time is decreased by additional layers [19]. By the usage of this mechanism, congestion control is attained absolutely through the group managing and routing mechanisms of the primary multicast procedure.

5.4 TCP Friendliness

The unicast protocol which is connection-oriented yield consistent data transfer with congestion control and flow. As TCP maintain a congestion window which control the amount of exceptional unidentified data packets in the network. The sender can send packets only as long as free slots are available because the data send will consume slots of the window. As soon as an acceptance intended for exceptional packets is arriving, the window is deviated so that the acknowledged packets can depart the window and the identical number of free slots becomes accessible for the upcoming data. TCP performs slow start, and the rate roughly doubles each round-trip time (RTT) to quickly increase its fair share of bandwidth. In its steady state, TCP uses an additive increase, multiplicative decrease mechanism to react to congestion by the detection of additional bandwidth. TCP increases the congestion window by one slot per round-trip time when there is no sign of loss. During loss of packet it is specified by a timeout and the congestion window is minimized to one slot and TCP re-enters the slowstart phase.TCP-friendliness can be calculated during the consequence of a non TCP flow going on challenging TCP flows below the similar circumstances about throughput and previous parameter. The non TCP unicast stream can be TCP friendly but it do not control the extensive term throughput for any of the synchronized TCP flows by a issue that is further than that prepared by a TCP flow under the same circumstances. A multicast flow is assumed to be TCP friendly if it separately views for every sender receiver pair of the multicast flow TCP-friendly [24].

6. FUTURE STUDY

As a developing research area, a number of unsolved issues remain. A particular problem is the deficit of comparison congestion control protocols standard methods. A test background that investigates dissimilar important aspects like as equality and scalability of the flow, united with method to directly compare the protocol performance [18] would be very useful which also provides standardized suite of test scenarios. Even as such a test background is not enough to walk around all details of a particular protocol, it would offer a sensible source for more objective comparisons of the protocols.

In numerous cases, the imitation scenarios offered for a protocol concentrate on a few broad-spectrum scenarios and are frequently too simple to capture behaviour and various characteristics of protocol in non-standard situations. Traffic circumstances in the network are getting too composite to be formed in all the conditions by a network simulator, building it significant to estimate the protocols also below real-time applications. As we discussed the various features and behaviour of single-rate and multi rate congestion control. It is possible that diverse forms of congestion control are practical possibly with router maintain and does not show signs of the disability of these methods. While TCP-friendliness is a practical fairness measure in today's network, it is also possible that future network architectures will agree to or necessitate dissimilar interpretations of fairness. Moreover the fairness denotation for multicast and many methodologies are still issues to investigate. We offered one feasible factors and methods to overcome and also momentarily addressed a dissimilar structure where multicast flows are allowable to use a high percentage of bandwidth than the unicast flows are, but these can be by no means the only capable fairness definitions. In advance area of research is the enhancement of the models for TCP network traffic that are used for some of the rate based congestion control mechanisms. Existing TCP formulae are based on some assumptions that are frequently not met in real-time circumstances. A feature of congestion control mechanism is, that is not explicitly related to the traffic examined in this paper but extremely appropriate to congestion control in common is how to care for the short-lived flows that consists of only some data packets. The TCP congestion control, as well as the congestion control strategy presented in this paper, requires that flows persistence for a definite quantity of time period. If not those forms of congestion control are insignificant.

7. CONCLUSIONS

In this paper, we presented a survey on current trends and advancements in the part of TCP-Friendly congestion control. We discuss the necessity for TCP-friendly congestion control for together non-TCP based unicast traffic and multicast communication and thus provided a general idea of the plan for such congestion control mechanisms. We briefly survey of various congestion control algorithms. It shows that at present there is no single algorithm that can resolve every problems of congestion control on computer networks and the Internet. Further research work is needed in this direction. It is also to note that not almost all of the surveyed papers have employed any statistical techniques to verify their simulation results. The above discussed are the theory of congestion its goals and merits and the most common factors for the occurrence of congestion and the methods to overcome the congestion collapse. This paper in brief discusses the congestion control algorithms based on the network awareness and various common congestion control algorithm used and its protocols. The paper also discusses the TCP- friendliness and the characteristics of the TCP and non-TCP flows and also the discussed issues that remain to be solved.

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BIOGRAPHIES

Mastan Mahammad was working as an Associate Professor in CSE Department, Malla Reddy Engineering College, Hyderabad, Telangana, India. He had published several papers in National and International Conferences & International Journals. His research interests Computer Networks, Wireless Networks with specialization in Quality of Service (QoS) in IEEE 802.11 Wireless LANs.

P.Suresh Verma working Dean CDC & Professor Department of CSE, Adikavi Nannaya University, Rajahmundry, Andhra Pradesh , India. He had published several papers in National and International Conferences & International Journals. His research interests Computer Networks, Wireless Networks He is a member of IACSIT and ISTE.

Mohammed Ali Hussain working as Professor Department of ECM, KLU, Guntur, Andhra Pradesh, India. He had published several papers in National and International Conferences & International Journals. His research interests Computer Networks, Wireless Networks with specialization in Quality of Service (QoS) in IEEE 802.11 Wireless LANs. & Ad-Hoc Networks. He is a member of IACSIT and ISTE .