A Survey on Congestion Control and Congestion Control Mechanism-A Study

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Abstract-

Nowadays, by accelerated growth of number of devices connecting to the Internet and emergence of IoT, the internet traffic is rapidly increasing. In these environments the probability of congestion is constantly rising. In the traditional networks, the end devices are responsible for controlling the congestion, but since these devices do not have a global and comprehensive view of the network, they cannot predict the congestion to prevent the packet loss. Hence the overall network throughput is reduced. this paper proposes a survey on congestion control mechanism in SDN. Telecommunication, Computer Networks and both wired and wireless communications including the Internet, are being designed for fast transmission of large amounts of data, for which Congestion Control is very important. Without proper Congestion control mechanism, the congestion collapse of such networks would become highly complex. Congestion control for streamed media traffic over network is a challenge due to the sensitivity of such traffic towards. This challenge has motivated the researchers over the last decade to develop a number of congestion control protocols and mechanisms that suit the traffic and provides fair maintenance for both unicast and multicast communications. This paper gives out a brief survey of major congestion control mechanisms, categorization characteristics, elaborates the TCP-friendliness concept and then a state-of-the-art for the congestion control mechanisms designed for network. The paper points the pros and cons of the congestion control mechanism, and evaluates their characteristics. To compensate packet loss some protocols, use fast retransmission scheme but suffer problem of Congestion collapse. Congestion collapse means the bandwidth is available but the network becomes under-utilize. When a network is in such condition, since the network is under-utilizing and the demand is high, both lead to worst throughput. The network in this situation experience high level of loss and delayed packet. This results in need for better algorithms, protocol implementations scheme(s) and efficient network devices with sufficient space (storage), because the reason behind the Network Congestion problem lies in transport protocol implementations. The obvious looking way to implement a Protocol sometimes results in wrong response to the network. The algorithms in this category are rooted in the idea that the condition of packet conservation must not violet. This paper is a survey of congestion control algorithms in packet switched networks.

Keywords- Congestion. Congestion Control. TCP-Friendliness, Goals, and Metrics of Congestion Control and UDP Traffic.

INTRODUCTION;

Wireless Network Wireless network refers to any type of computer network that is implemented without the use of wires between intermediate nodes and having features like fast deployment , cost reduction, vulnerability, easy sharing of data, fast communication, security and scalability. The applications of wireless networks include cellular telephony, wireless energy transfer, Wi-Fi, security systems and the problems involving the wireless network are high setup cost, limited range coverage, high susceptibility to interference, interception by unauthorized user, slow speed.Congestion control over network, for all types of media traffic, has been an active area of research in the last decade. This is due to the flourishing increase in the audiovisual traffic of digital convergence. There exists a variety of network applications built on its capability of streaming media either in real-time or on demand such as video streaming and conferencing, voice over IP (VoIP), and video on demand (VoD). The number of users for these network applications is continuously growing hence resulting in congestion.Congestion Control Congestion occurs when the number of packets being transmitted through the network approaches the packet handling capacity of the network. Network congestion is caused by variety of factors, including network topology, bandwidth and usage patterns. Faulty hardware, network broadcasts and expansion also slow the networks. The measures for avoidance of network congestion require two major components i.e. a router mechanism to reorder or drop packets under overload and end-to-end flow control mechanisms designed into the end points which respond to congestion and behave appropriately.

I. THEORY OF CONGESTION CONTROL SYSTEM;

Congestion control concerns in controlling the network traffic in a telecommunications network, to prevent the congestive collapse by trying to avoid the unfair allocation of any of the processing or capabilities of the networks and making the proper resource reducing steps by reducing the rate of packets sent.

A. Goals and Metrics of Congestion Control; Goals that are taken for the evaluation process of a congestion control algorithm are: i. To accomplish a high bandwidth utilization. ii. To congregate to fairness quickly and efficiently. iii. To reduce the amplitude of oscillations. iv. To sustain a high responsiveness. v. To coexist fairly and be compatible with long established widely used protocols.

The Metrics that have been set for Congestion control are:

i. Convergence Speed - The Convergence speed estimates time passed to reach the equilibrium state.
ii. Smoothness - The Smoothness reflects the magnitude of the oscillations through multiplicative reduction and it depends on the oscillations size.
iii. Responsiveness - The Responsiveness is measured by the number of steps or the round trip times (RTTs) to attain equilibrium.

The discrepancy between Responsiveness and Convergence Speed is that the responsiveness is related to a single flow and the convergence is related to the System. I. Efficiency - The Efficiency is the standard flow throughput per step or round trip time (per RTT), when the system is in equilibrium. II. Fairness: The Fairness characterizes the fair allocation of resources between the flows in a shared bottleneck link.

II. CLASSIFICATION OF CONGESTION CONTROL ALGORITHMS;

The Congestion Control Algorithms are classified mainly based on the below criterion: i. Can be classified by the type and size of the feedback received from the network ii. Can be classified by increasing the deploy ability on the network. Only the sender needs for the modification (or) sender and receiver need modification (or) 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: max-min, proportional, "minimum potential delay"

A. Classification of Congestion Control by Network;

Congestion control algorithms can be categorized using network awareness as a criterion. The following are the three categories for the congestion control mechanisms. The Black box consists of a collection of algorithms based on the concept that reflects on the network as a black box, pretentious of no knowledge of its state much other than the binary feedback upon congestion. The Grey box is grey group approaches that use the measurements to estimate accessible bandwidth and the level of contention or even the provisional characteristics of congestion. Because of the opportunity of wrong estimations and measurement dimensions, the network is considered as a grey box. The Green box contains the bimodal congestion control through which it can calculate explicitly the fairs hare, also the network-assisted control, whereas the network communicates through its transport layer. Hence, this is considered as green box.

i. The Black Box;

The black box classified congestion control is also called the Blind Congestion Control method and this methodology uses the Additive Increase Multiplicative Decrease (AIMD) algorithm. The AIMD implements the TCP window adjustments. Stability is achieved with these algorithms in situations where the demand of competing flows exceeds the available bandwidths of the channel. The congestion control mechanism in the conventional TCP is based on the fundamental idea of AIMD. In TCP-Tahoe, TCP-New Reno and TCP-Sack, the preservative increase phase is adopted exactly as in AIMD, where the protocols mechanisms are in the congestion control phase. In case of a packet drop, instead of the multiplicative reduction, a more conservative method is used in TCP-Tahoe. The congestion window resets and the protocol mechanisms enter again the slow-start phase. On the other hand, in TCP-New Reno and TCP- Sack, when the sender receives 3 DACKs, a multiplicative reduction is used for the both windows and slow-start threshold phase is applied. In such case, the protocol mechanism remains at the Congestion control phase. When the retransmission timeout expires, they enter the slow-start phase as in TCP-Tahoe. high-speed-TCP - High-speed-TCP modifies the response function in environments with high delay-bandwidth product, increases the congestion window more belligerently upon getting an acknowledgment, and reduces the window more gently upon a loss event. BIC-TCP - Binary Increase Congestion Control Protocol uses a hollow raise of the sources rate following each congestion event until the window is equivalent to that before the event, to maximize the utilization time of the network. CUBIC TCP - It is a less aggressive and more systematic derivative of BIC, where the window is a cubic function of time because of the final congestion event, with the modulation point set to the window former to the event. AIMD-FC - A current advancement of AIMD is Additive Increase Multiplicative Decrease with Fast Convergence is not based on a new algorithm, but on an optimization of AIMD and the convergence procedure that enables the algorithm to congregate faster and attain higher efficiency. Binomial Mechanisms - Binomial Mechanisms form is a new class for the nonlinear congestion control algorithms named Binomial Congestion Control Algorithms. These algorithms are called binomial because of the control mechanism that is based on the contribution of two additional algebraic terms with different exponents. SIMD Protocol - SIMD is a TCP-friendly nonlinear congestion control algorithm that that controls the congestion by utilizing history information. GAIMD - General AIMD Congestion Control generalizes congestion control mechanism of AIMD by parameter zing the additive increase value α and multiplicative decrease ratio β.

ii. **The Grey Box;** The Grey Box is also called as Measurement-based Congestion Control. Standard TCP relies on packet losses as an implicit congestion signal from congested links. There are a number of reasons for indicating the congestion one of the common reasons is the packet loss: Random bit corruption is the main cause for the packet loss and is caused when bandwidth is still available. Acknowledgement-based loss detection at the sender side can be affected by the cross-traffic on the reverse path. Packet loss, as a binary feedback, cannot indicate the level of contention before the occurrence of congestion. Therefore, an efficient window adjustment tactic should reflect various network conditions, which cannot all be captured simply by packet drops. Several measurement-based transport protocols gather information on current network conditions. TCP Vegas -- The queuing delay is estimated by TCP Vegas. To make a constant number of packets per flow the window is linearly increased and decreased in the network.

FAST TCP -- FAST achieves the same equilibrium as Vegas, but uses proportional control instead of linear increase, and intentionally scales the gain down as the bandwidth increases with the aim of ensuring stability. TCP-Westwood -- A loss causes the window to be reset to the sender's estimation of the bandwidth-delay product in TCP-Westwood which is the minimum measured round trip times the experimental rate of getting acknowledgement. TFRC -- TFRC is based on the rate-based congestion control mechanism, which intends to efficiently compete for bandwidth with flows in the network. TCP-Real -- TCP-Real mechanism is based on a receiveroriented and measurement-based congestion control mechanism that improves the overall performance of TCP over heterogeneous both wired or wireless networks and over asymmetric paths. TCP-Jersey -- TCP-Jersey is also based on the TCP scheme that focuses on the competence of the transport mechanism in the network. iii.

The Green Box

The Green box contains the bimodal congestion control mechanism by which it can calculate explicitly the fairs hare of the system flow in the network. Bimodal Mechanism -- Bimodal Congestion Avoidance and Control mechanism for each flow the fair-share of the total bandwidth that should be allocated is measured at any point during the execution of the system flow. Random Early Detection – In Random Early Detection (RED) packets are randomly dropped in proportion to the router's queue size, triggering multiplicative reducing in some flows. Explicit Congestion Notification – In Explicit Congestion Notification (ECN) routers are enabled to probabilistically mark a bit in the IP header instead of dropping the packets, to intimate the end-hosts of imminent congestion when the length of the queue exceeds a threshold. VCP -- The variable-structure congestion control protocol (VCP) uses two ECN (Explicit Congestion Notification) bits to explicitly get the feedback of the network state of congestion.

IV. CONGESTION CONTROL ALGORITHMS;

A. Drop Tail Algorithm

F. Postiglione et al., discussed that the drop Tail (DT) algorithm has a great accuracy, simplest and most commonly used algorithm in the current networks, which drops packets from the tail of the full queue buffer. The main advantages of this algorithm are simplicity, suitability to heterogeneity and its decentralized nature. However, this algorithm also has some serious disadvantages, such as lack of fairness, no protection against the misbehaving or non-responsive flows (i.e., flows where the sending rate is not reduced after receiving the congestion signals from gateway routers) and no relative Quality of Service (QoS). QoS is of particular concern for the continuous transmission of high- bandwidth video and multimedia information . This type of transmitting the content is difficult in the present Internet and network with DT. B.

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) . On the arrival of each packet, the average queue size is calculated by using the Exponential Weighted Moving Average (EWMA) . The computation of the average queue size is compared with the minimum and the maximum threshold to establish the next action.

C. Choke Algorithm Konstantinos Psounis et al., proposed CHOKe algorithm , 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. It is a simple and stateless algorithm where no special data structure is required. However, this algorithm is not present well when the number of flows is huge when compared to the buffer space.

D. 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 .

E. Random Exponential Marking Algorithm

According to Debanjan Saha the Random Exponential Marking Algorithm (REM) 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 φ must be known globally.

F. Fair Queuing Algorithms Alan Demers et al., proposed the Fair Queuing Algorithms and Stochastic Fair Queuing Algorithms 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 , 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.

G. Virtual Queue Algorithm The Virtual Queue Algorithm (VQ) is a radical technique proposed by Gibben and Kelly . 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.

H. Adaptive Virtual Queue Algorithm R.J. Gibben et al., discussed in the Adaptive Virtual Queue algorithm 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.

V. TCP-FRIENDLINESS TCP is a connection-oriented unicast protocol provides reliable data transfer with flow and congestion control. TCP maintains a congestion window, which controls the number of exceptional unacknowledged 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. When an acknowledgment for exceptional packets is received, the window is shifted so that the acknowledged packets can leave the window and the same number of free slots becomes available 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. In case of packet loss is indicated by a timeout, the congestion window is reduced to one slot, and TCP reenters the slowstart phase.

TCP-friendliness can be measured through the consequence of a non-TCP flow on the competing TCP flows under the same conditions regarding throughput and other parameters.

A non-TCP unicast flow can be TCP-friendly if it does not influence the long-term throughput for any of the synchronized TCP flows by a factor that is more than that done by a TCP flow under the same conditions. A multicast flow is said to be TCP-friendly if it separately views for each sender-receiver pair of the multicast flow TCP-friendly.

A. TCP-Friendliness Vs UDP Traffic One of the grave drawbacks of FIFO-based queue management is that there is no way to homogenize the connections which send more than their bandwidth share and are non-responsive or very slow in response to congestion collapse indication. In order to present, a fair share of accessible bandwidth to all TCP-friendly connections that is amenable to the congestion collapse indication and the misbehaving in connections should be successfully synchronized by a queue management algorithm. One possible methodology is to solve the above consequences is to use per-flow queuing to discriminate against the non-TCP-friendly connections and to present fair bandwidth share to connections. It is also possible to provide an inducement to TCP-friendly connection in terms of financial benefits. Another possible method is to append a new concept of service i.e., differentiated services to connections. Thus, the differentiated services are being studied by the Differentiated Services Working Group in the IETF .

VI. CLASSIFICATION OF CONGESTION CONTROL PROTOCOLS Congestion control protocols are classified into four major categories according to a number of features in their mechanism of work . The following shows the valid categories of classification. A. Window-Based Congestion Control Window-Based protocols are built based on the technique of congestion window-based mechanism, and the congestion window is used at the sender or receiver side . A slot in that window is reserved for each packet, when the sent packet is acknowledged to be received 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 . B. Rate-Based Congestion Control Rate-Based protocols are built based on the adaptation of their rate of transmission according to some incorporated feedback algorithm that intimates about congestion when it exists. Rate-based algorithms can be subdivided into simple mechanisms and Congestion control. The results of saw-incentive to cooperative sources and a properly calculated and fixed value of φ must be known globally.

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C. Single-rate Congestion Control Single-rate congestion control mechanisms are usually adopted by all the unicast congestion control protocols. Transmission in unicast has only one recipient, so sending rate is adapted in accordance to the recipient‘s status. Multicast transmission can adopt the single-rate approach also, where the sender streams the data with same rate to all recipients of the multicast group in the network.

D. Multi-rate Congestion Control Multi-rate congestion control uses the layered multicast approach, because multi-layering enables to divide data of the sender into different layers to be sent to different multicast groups. Every receiver joins the largest possible number of groups permitted by the bottleneck in the way to sender. The quality of data to be sent to this receiver becomes high when joining more multicast groups. This feature is most evident in multicast video sessions where more the groups that the recipient subscribes in, is more layers that the recipient receives, and also more better the quality of video is. Meanwhile, for other mass data, the transfer time is decreased by additional layers . By the usage of this mechanism, congestion control is achieved absolutely through the group management and routing mechanisms of the primary multicast protocol.

VII. AREAS OF FUTURE RESEARCH As in the case with an evolving research area, several unsolved issues remain. One particular problem is the lack of comparison congestion control protocols standard methods. A test background that investigates different important aspects such as fairness and scalability of the flow, combined with measures to directly compare the protocol performance would be very handy which also provides standardized suite of test scenarios. While such a test background is not sufficient to walk around all details of a precise protocol, it would provide a sensible basis for more objective comparisons of the protocols. In many cases, the imitation scenarios presented for a protocol concentrate on a few broad-spectrum scenarios and are frequently too simple to capture behavior and various characteristics of protocol in non-standard situations. Traffic conditions in the network are getting too complex to be modeled in all the aspects by a network simulator, making it significant to estimate the protocols also under real-time applications. We already discussed the various characteristics and behavior of single-rate and multi rate congestion control. It may well be possible that different forms of congestion control are practical maybe with router support that do not show signs of the disadvantages 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 different definitions of fairness. Also the fairness definitions for multicast and many methodologies are still subject to research. We presented one possible factors and methods to overcome and also briefly addressed a dissimilar form where multicast flows are allowable to use a higher percentage of bandwidth than the unicast flows are, but these can be by no means the only promising fairness definitions. A further 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 several assumptions that are often not met in real-time conditions. One feature of congestion control mechanism is, that is not openly related to the traffic discussed in this paper (i.e., streaming media traffic) but highly relevant to congestion control in common is how to treat the short-lived flows that consists of only a few data packets. The TCP congestion control, as well as the congestion control schemes presented in this paper, requires that flows persistence for a certain quantity of time period. If not those forms of congestion control are insignificant.

VIII. CONCLUSION:

In this paper we have proposed a model and mechanism for congestion control using window size, Queue length and link capacity with accelerating effect over capacity based wireless network. We also tries to identify the cases for reasons of congestion with different values of link capacity, processing capacity of nodes with overall effect of accelration mechanism and tries to provide solution for them by proposed algorithm with illustration of three phases and their steps. In this paper, we presented a survey on current trends and advancements in the area of TCP-friendly congestion control. We discussed the necessity for TCP-friendly congestion control for both non-TCP based unicast traffic and multicast communication and thus provided an overview of the design space for such congestion control mechanisms. This paper briefly surveys of various congestion control algorithms. It seems that at present there is no single algorithm that can resolve all of the problems of congestion control on computer networks and the Internet. More 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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