Comprehensive Exploration for Proposing Hybrid Scheme upon Congestion Avoidance Algorithms

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ABSTRACT

Congestion free services are ultimate preference of every network consumer and service providers. Variety of parameters like packet dropping rate, latency, jitter, throughput, bandwidth, fair response of resources, link utilization and queue length are responsible to fabricate or reduce congestion. Current TCP model for high speed networks is unstable and ineffectual due to slow response, large window size and fairness issues. The ideal and positive utilization of indicated factors can reduce congestion up to ideal strength with enhanced fairness. These entire factors cannot be handled with single congestion handling technique but a joint committee of congestion techniques can manage all these constraints. We considered packet loss as a primary congestion and fairness metric that differs with already conveyed hybrid congestion techniques that utilize delay as primary metric. We reviewed several congestion algorithms to find out most essential parameters to negate congestion in packet switched networks among the above mentioned parameters. We proposed a hybrid congestion handling technique after performing sufficient comparison with already conveyed hybrid congestion management techniques. Our propose hybrid congestion management technique (ECN + IFRC) is empirically superior to exiting hybrid congestion management techniques in some extents.

General Terms

Network Communication, Congestion and Flow Control, Networks Protocols, Network Services and measurements

Keywords

Congestion Avoidance, QoS, Latency, Jitter, Hybrid Congestion Schemes

1. INTRODUCTION

Most of applications require heavy contents with rapid transfer rate under the requirement of bulky bandwidth. In order to manage bandwidth and fair response there is need to manage congestion. The information regarding the congested situation collected by the sender may not be accurately reflecting the actual situation of congestion. This type of non-accurate information may caused worst situation because the network behaves like a black box for newly joined source (network node or machine), therefore in order to get fresh network situation the newly joined node initiates first request with small sending rate and increases the sending rate in next subsequent requests. The newly joined node may require the issuance of many requests in order get complete network situation. These types of network requests creates extra load on networks that may lead to congestion. The solution of this kind of congestion issue requires to judge the load on the behalf of hop counts and RTT rate ratio as discussed by Ijaz A. Shoukat and Iftikhar M. According to their opinions, network path is substantially amended with sufficient increment in hop counts and RTT values when congestion occurs [1]. Congestion management is reliant on four mutual algorithms (Efficient Retransmission, sluggish Start, Congestion Avoidance, Quick Recovery) and all these algorithms can be implemented under generic congestion handling protocols [2] : (1) *Buffer based Congestion Protocol* - in which every node sends the packet to its downward node close to it if and only if the receiving node has some buffer capacity. (2) *Rate Based Congestion Protocol* – in which transmission rate is measured through both incoming and outgoing packet streams among the neighboring nodes by calculating the weight function. (3) *Priority Based Congestion Protocol* [3] – it deals with measuring hope's congestion severity (through packet arrival rate) and priority index of node (depending upon the weight of fairness).

Congestion investigation and control management can be done in three ways: (1) Detection, (2) Notification and (3) Adjustment of transmission rate [3]. Any control protocol that deals only with either index (delay) based indicator or loss based indicator cannot perform ideally against real time streaming video applications in a high speed network because for real time video streaming delays are not tolerated by users. Standard TCP mechanism and its several relatives like TCP New-Reno, TCP Illinois etc., are not sufficiently enough to deal with streaming application [4]. Our concern is with QoS of remote servers in network environment that mostly get engaged with congestion. We proposed a hybrid congestion handling technique that employs the ideal utilization of all congestion parameters to get enhanced result avoid congested situation.

2. LITERATURE REVIEW

The fixing of equalized throughput to each node does not mean ideal fairness. In 2006 the authors of study [3] reported that congestion is necessarily based on performance that can be improved by reducing the packet loss rate and fairness (Faire throughput utilization on each node). Furthermore they claimed that fairness itself is dependent on scheduling of packets. In 2011, the authors of study [5] proposed linear Matrix Inequality (LMI) based approach to deal with congestion situation. They studied the congestion occurrence under delay and link capacity parameters and they claimed that there approach is able to enhance performance in closed loop environment. But there study is only limited to judge two congestion constraints (delay and link capacity) that are not enough to get enhanced congestion free performance.

An ideal Packet switched network relies on routing decision and congestion free linkage. Efficient path decision making is done under routing strategy and congestion is the greatest hurdle in efficiency of transmission of remote queries. Present design of IP (Internet Protocol) can resolve single path routing with sluggish time degree and TCP (Transmission Control Protocol) requires end to end packet delivery under the management of fixed transmission rate through congestion window size limits that is not an effective solution for getting enhanced results [6]. The congestion window limits actually causes the increment in Round Trip Time (RTT) that severely effects the transmission rate[6]. HTCP(Hamilton TCP) [7] works on adaptive back-off strategy to attain a better efficiency and improved the performance over responsiveness in high speed networks. But it can't tackle the congestion state because of the rapid increment/ decrement in their window sizes with high flow traffic can results throughput degradation. BIC-TCP(Binary Increase Congestion TCP) [8] scheme was developed to overcome this drawback by executing a linear increment of window size at initial phase and then amplifying its value logarithmically towards the reference point. But this also undergoes the same RTT unfairness problem. According to the authors of study [9] the use of individual congestion control scheme cannot predict all type of congestion parameters to reflect actual loaded or congested situation. The authors of study [10] analyzed the joint effect of Binary Congestion Notification (BCN) and Transport Control Protocol (TCP) with heterogeneous traffic that can trigger the performance by reducing multiple packet losses. Compound Transport Control Protocol (CTCP) [21] is a hybrid approach which includes a scalable delay based constituent into some loss based component like TCP Reno congestion avoidance scheme. When network path is underutilized, the delay based component increases sending rate rapidly and once the path utilized or bottleneck queue built, it automatically becomes nonaggressive i.e. it reduces the sending rate decently since delay based schemes have a feature of automatic adjustment of its aggressiveness on basis of link utilization[12] [13]. CTCP performs well in case of single network flow/link but perform poor when dealing with shared network links i.e. their performance measures degrade and poor fairness achieved [11].

Many other congestion handling schemes as well as hybrid congestion management schemes have already been reported as discussed in Table (1 and 2). Each congestion handling scheme either individual or hybrid has its own affirmative and feeble characteristics with a reality that these all schemes cannot handle all congestion parameters in ideal way. Therefore, there should be a hybrid approach through which all type of required congestion parameters can be judged in optimal way.

3. COMPARISON OF CONGESTION TECHNIQUES

We technically performed the comparison of several congestion handling algorithms in the presence of important congestion parameters to analyze their discrepancies and affirmative effects on QoS as discussed in Table 1.

Table 1: Comparison of Congestion Algorithms

AIMD (Additive Increase and Multiplicative Decrease) [14],[15],[16]	
Evaluation Factors	**Detail and discussion**
Packet Dropping Probability (PDP)	Packets are lost when throughput rate becomes equal to buffer capacity.
Latency	Latency will be large as the size of queue will be larger
Jitter	Dynamic increase and decrease causes jitter [17]. Its buffer size is quite large that can create jitter situation too. <variable>
Throughput	It has the capability to decrease the buffer size additively to get optimal throughput. [18] [19]. Its throughput is dependent on its window size. <variable>
Link/ Channel Capacity (Bandwidth)	Channel capacity is severely affected due to its capability of multiplicative decrease. So it is not a good solution for high bandwidth links.
Fairness (Fair System Response)	It has the capability to achieve fairness by allocating the resources in increasing and decreasing fashion. [20].
Link utilization (achieved throughput)	High [18] [20] because it additively increases, so it is not an effective decision to use in High Bandwidth Delay Product (HBDP) networks.
Queue length	it uses round robin model to serve these flows. [17]
RED(Random Early Detection)[21],[22],[23]	
Evaluation Factors	**Detail and discussion**
Packet Dropping Probability (PDP)	As soon as queue size increases the packets loss occurs. There is chance to drop packets when queue size grows enormously. [24]
Latency	Large buffer size causes latency as it is directly proportional to buffer delay [24]. <variable>
Jitter	-
Throughput	Throughput is dependent on traffic load. <variable>
Link/ Channel Capacity (Bandwidth)	It operates well with routers having high bandwidth capacity. [24]
Fairness	Fairness can be achieved by increasing the queuing delay since more bandwidth-delay product is allotted for each flow and packet dropping for each flow reduces thereby reducing the overall congestion. [25]
Link utilization	If the buffer size is small the link utilization is reduced drastically and the queuing delays may get short [26] Higher the queue occupancy value will be the better link utilization but it causes more queuing delays. [24] <variable>
Queue	Packets loss occurs on average increment in queue size.
BCN(Backward Congestion Notification)[27], [28]	
Evaluation Factors	**Detail and discussion**
Packet Dropping Probability (PDP)	It depends upon the number of switches involves in congestion area.

	In normal network condition probability of dropping is nearly zero. [29] Queue has a limit. When limit is overflowed packet loss occurs. <variable>
Latency	Latency is dependent on distance metric. Latency is also dependent in retransmission [29]. <variable>
Jitter	???
Throughput	(TCP-BCN) combination is better for good throughput.
Link/ Channel Capacity (Bandwidth)	
Fairness	TCP-BCN combination is better for good Fairness. Unfairness is dependent on long distance.
Link utilization	TCP-BCN combination is better for getting optimal link utilization
Queue	Queue has a limit. When limit is overflowed packet loss occurs.
CPT(Choke Packet technique)[30]	
Evaluation Factors	**Detail and discussion**
Packet Dropping Probability (PDP)	If Avg. queue Size > Max. allocated-Value, Then packets begins to drop.
Latency	Latency depends on the processing delay requires for Choke condition. If both CPT-UDP are combined in parallel then latency is short. <variable>
Jitter	Jitter is more in case of CPT used with UDP.
Throughput	In case of TCP, it gives higher throughput. In case of UDP, it gives minimum throughput. [31]
Link/ Channel Capacity (Bandwidth)	Channel capacity is not fully utilized in this algorithm. [32]
Fairness	
Link utilization	Good in case of responsive sources.
Queue	
ECN(Explicit Congestion Notification) [33],[34]	
Evaluation Factors	**Detail and discussion**
Packet Dropping Probability (PDP)	ECN is effective because it has low packet dropping rather to TCP. [33],[34] ECN – RED also have low rate of packet dropping. [33],[34] But in case of dealing with satellite networks packet get lost due to buffer overflow. [35]
Latency	ECN possesses low latency when it is combined with TCP. But high latency occurs in case of satellite networks as these networks causes high propagation delays resulting in late congestion notification. [35]
Jitter	ECN has low rate of jitter because it has lower packet lose rate which makes the congestion window more consistent. [36].
Throughput	Higher throughput can be achieved in case of small gateway buffer size values in comparison to drop tail and RED algorithm but in opposite case it doesn't so. [36]
Link/ Channel Capacity (Bandwidth)	ECN bandwidth allocation is very good.
Fairness	Resources are utilized fairly in ECN as it has low packet lose ratio [36].
Link utilization	ECN – TCP provides 100% link utilization even in presence of queuing delay and propagation.
Queue	Average size of queue is noted for getting the level of congestion.
EDD(Earliest due Date) [37],[38],[39],[40]	
Evaluation Factors	**Detail and discussion**
Packet Dropping Probability (PDP)	Packets may get drop when sum of local relative deadlines is more than the end to end deadline of connection i.e., throughput suffers less when deadlines monitored carefully.
Latency	Depends on network nodes. <variable>
Jitter	It depends on the degree of traffic / huge traffic (bursty traffic). <variable>
Throughput	Throughput is optimal. [40].
Link/ Channel Capacity (Bandwidth)	Fair among intermediate nodes and it utilizes the channel capacity fairly well.[38]
Fairness	Resources are fairly utilized in successful condition but unfair in case of un-successful condition. [37].
Link utilization	-
Queue	-
CAB(Congestion Avoidance Bit) Scheme [41],[42]	
Evaluation Factors	**Detail and discussion**
Packet Dropping Probability (PDP)	Approximately zero. [42]
Latency	Low latency in presence of round trip delay metric. It is a fast network protocol for users.
Jitter	Dependent on the change of window size. [41]
Throughput	Routers operate efficiently, when traffic operates below knee that results fair throughput.
Link/ Channel Capacity (Bandwidth)	In optimal case, it is good.
Fairness	The binary bit is fairly allocated to all resources by using global optimality concept.[FAIR]
Link utilization	-
Queue	Average queue size is used as a metric to decide the load condition. [42].

TFRCP(TCP Friendly Rate Control Protocol)	
Evaluation Factors	**Detail and discussion**
Packet Dropping Probability (PDP)	Packet lose ratio is dependent on Round Trip Time (RTT) as much as RTT increases, the packet lose will be more. [43]. <variable>
Latency	Latency is dependent on RTT if RTT is low the latency will be lower [44]. <variable>
Jitter	Jitter is dependent on queuing delay and RTT, as more RTT or delay as will be the jitter. [43]. <variable>
Throughput	Throughput is dependent on the quality of signals as high quality signal have more throughput. [44]. <variable>
Link/ Channel Capacity (Bandwidth)	Bandwidth / channel capacity is poorly utilized due to RTT delays.
Fairness	It is relatively fair to TCP because it re-computes its connection rate only after every " re-computation" time unit. [FAIR] [44].
Link utilization	Not good. [45].
Queue	-
XCP(Explicit Control Protocol)	
Evaluation Factors	**Detail and discussion**
Packet Dropping Probability (PDP)	Approximately zero due to good resources utilization. [46].
Latency	The average feedback value for all the links exceeds 0, link capacity will be fully shared and the queuing delay (Latency) will be minimum i.e., nearly to zero.[47].
Jitter	Less as compared to TCP [46].
Throughput	The performance or throughput may degrades in shared access media environment like radio/satellite communication. [32].Throughput increases proportionally to the increase in value of average feedback. [46]
Link/ Channel Capacity (Bandwidth)	Fully optimized [46].
Fairness	It is good in terms of fairness compare to other (end to end schemes)TCP schemes [48].
Link utilization	Good. [31]. Link utilization is full when average feedback value is greater than zero, and this scheme works efficiently. [33]
Queue	Queue length remains constant [48].
DCCP (Datagram Congestion Control Protocol)	
Evaluation Factors	**Detail and discussion**
Packet Dropping Probability (PDP)	Packet dropping probability is non-zero because this scheme is intended for real-time traffic data and prefers the timely delivery instead of reliable or in-order delivery of data. [49]
Latency	Minimum because, This scheme focus on timeliness delivery of data and not required reliability and timing constraint [49].
Jitter	Jitter is maximized as compare to UDP, especially in case of multiple hope counts [50].
Throughput	Maximized because, This scheme focus on timeliness delivery of data and not required reliability and timing constraint [49].
Link/ Channel Capacity (Bandwidth)	This method is suitable for varying bandwidth/channel capacity because it consists of various transport layer type protocols for its operation. [49]
Fairness	Provide well fairness [51].
Link utilization	-
Queue	In this scheme data are not queued for final delivery to applications unlike TCP and other protocols instead the queued data may go forward only after the accordance of featured values and options. [52]
RCP(Rate Control Protocol)	
Evaluation Factors	**Detail and discussion**
Packet Dropping Probability (PDP)	Packet losing depends upon the change of networking condition and RTT [53]. <variable>
Latency	Lower Latency as router offers uniformly flow with small delay. [52].
Jitter	Jitter only occurs when total flow is larger than link capacity [52].
Throughput	It gives better throughput even in multiple bottlenecks rather to TCP [53].
Link/ Channel Capacity (Bandwidth)	Fully utilized [54].
Fairness	Resource sharing is fair [54].
Link utilization	Links are fully utilized [53].
Queue	Length is quite large [55].
BPT(Back Pressure Technique)	
Evaluation Factors	**Detail and discussion**
Packet Dropping Probability (PDP)	Packet dropping probability can be reduced by ADPCM(Adaptive differential Pulse code Modulation technique) [56]
Latency	Large queues are maintained in ICN(Intermittent connected Network) so delays are longer than normal and high latency achieved due to maximized rate controller utility and operating close in proximity of capacity region. [57]
Jitter	Propagation of backpressure is too slow causing the fluctuation in flows of packet since

	network state changes frequently that caused jitter. [47]
Throughput	Optimal [57] [58].
Link/ Channel Capacity (Bandwidth)	Network capacity is highly utilized. [57] [58].
Fairness	There is a tradeoff between fairness and delays, fairness improved with the increasing cost of latency otherwise fairness is not achieved. [58] <variable>
Link utilization	Links are fairly utilized as they are scheduled regularly and their rates are computed at each time by controller. [58]
Queue	Large queues maintained only at intermittently connected nodes, rest nodes have small queues [57].
IFRC(Interference aware fair rate control) Scheme	
Evaluation Factors	**Detail and discussion**
Packet Dropping Probability (PDP)	Packet dropping probability is almost zero even having small buffer sizes because it's rate adaptation quality is much effective. [59]
Latency	When IFRC is implemented for retransmission on link level, it uses software ACKS that may cause delays and increase the latency thereby reducing throughput. [60]
Jitter	It is an adaptive rate control scheme which means, rate lies between a minimal difference, hence less jitter is experienced in this scheme. [61]
Throughput	Overall, it gives a higher throughput for protocols that use link-quality metrics in order to establish the flow routing tree otherwise it provides minimum throughput in order to reduce number of dropped packets. [59] <variable>
Link/ Channel Capacity (Bandwidth)	Capacity is fairly utilized through all the nodes by estimating the transmission time for packets [62].
Fairness	It employs a distributed rate adaptation technique to achieve fairness and support weighted fair allocation. [59]
Link utilization	Link utilization is maximized and buffer dropping rates is almost zero which is the property of this scheme. [62][61]
Queue	Average queue lengths are monitored to find out the emerging congestion [59].
PBS(Partial Buffer Sharing) Scheme	
Evaluation Factors	**Detail and discussion**
Packet Dropping Probability (PDP)	Packet dropping probability decreases when threshold position increases but in case of *delay tolerant traffic* it continuous to increase. [63] <variable>
Latency	Latency is high because, response time increases for both delay tolerant and delay sensitive traffic streams when threshold position increases. [63]
Jitter	It employs space priorities for different multimedia traffic classes in order to avoid the jitter [63].
Throughput	Throughput depends upon delays [64].
Link/ Channel Capacity (Bandwidth)	Buffers are highly utilized for high priority traffic data. [64]
Fairness	It uses a threshold value that limits the access to buffer space against each higher/lower priority traffic, and hence it cannot achieve better fairness among the resources, but with CBS (Complete buffer sharing), better fairness can be achieved. [65].
Link utilization	Links are also utilized fairly in case of large threshold values. [63].
Queue	The average queue length plays an influential part for allocating the threshold positions. [63].
CTCP(Compound TCP)	
Evaluation Factors	**Detail and discussion**
Packet Dropping Probability (PDP)	Packet dropping is zero when dealing with single flow but in shared network, it loses the packets. [13] <Average or variable>
Latency	Some network latency is experienced even dealing with single network flow but slightly lower than CUBIC TCP.[13] <variable / average>
Jitter	Jitter is minimized through the use of synergized approach.
Throughput	With single or shared network connection severe degradation in throughput is observed.[66] [67].
Link/ Channel Capacity (Bandwidth)	Efficient due to scalable window management rule. [12]
Fairness	Good in case of single link utilization but poor incase of shared link utilization. [11]
Link utilization	It achieves efficient link utilization by having a rapid increase rule in its delay based components e.g. multiplicative increase. [12]
Queue	Queue sizes are relatively shorter since it uses delay based approach as primal index of congestion and hence window size is degraded before the happening of congestion.[13]

To get optimized congestion free situation the throughput and link utilization should be maximized. Furthermore, the queue length should be ideal, packet loss ratio and latency should be considerably minimized. The implementation of single congestion handling approach may not get the actual values of all parameters.

4. DISCUSSION AND ANALYSIS

Web traffic is progressively growing; the size of indexed web contents was estimated more than 25 billion of web pages in January 2011 by the authors of study [70]. Growing volume of web is causing bulky network traffics. Complex network architecture and large number of internet traffic creating extra load on communication networks. In loaded situation, packet dropping is occurred at router's end that causes substantial increment in round trip times as well as in hop counts and this situation is designated as congestion [1]. Happening of latency in any remote service is caused by large incremental variations in round trip times because latency itself is the sum of all packet's round trip delays [71], therefore handling of load and congestion are the backbone to acquire quality oriented remote services in communication networks. Several hybrid congestion control schemes have been proposed that utilize the delay as primary indicator of congestion and packet loss as secondary indicator to control congestion. our analysis differs with the selection of delay as a primary metric and support to select packet loss as an essential metric to get optimal results. We summarized our results in Table 2 on the behalf of previously discussed comparison of congestion techniques in Table 1. We combined the two congestion techniques (ECN + IFRC) in order to satisfy most important constraints (Packet dropping, link utilization) because the ideal satisfaction of other remaining congestion constraints is dependent on the satisfaction of these two constraints. We compared our proposed hybrid congestion management technique (ECN +IFRC) with prior hybrid congestion techniques (CPT-UDP [9] , TCP-BCN [8] , C-TCP [21] , ECN – TCP [10] [11] , ECN – RED [10] [11]) as summarized in Table 2.

Table 2 : Comparison results of Hybrid Congestion Techniques

Hybrid Techniques	**Packet Dropping Probability (PDP)**	**Latency**	**Jitter**	**Throughput**	**Link/ Channel Capacity (Bandwidth)**	**Fairness** (Fair System Response)	**Link Utilization**	**Queue length**
CPT-UDP	Variable	Short	High	Minimum	Fully utilized	???	Good	???
TCP-BCN	Variable	Variable	???	Good	???	Good	Optimal	Define Limits
C-TCP	Average or variable	variable OR average	Min.	severe degradation	efficient	Good	Efficient	Shorter
ECN – TCP	Effective (it means not low and not zero)	Low	Low	High	Very Good	Fair	100%	Average
ECN – RED	Low	Low	Low	High	Very Good	Fair	Variable	Average
ECN + IFRC ***Proposed***	Zero	Low	Low	High	Good	Fair	Maximum	Average

According to our analysis, Packet dropping is the key metric rather to select delay as primary metric. Our suggested primary metric (packet loss) play a vital role in congestion occurrence because when packet starts to drop then it means there is a measureable latency (end to end delay) with jitter, limited throughput, poor bandwidth, unfair node response, limited link utilization and large queue length. Hence, this realistic trait clearly invokes that in suggesting hybrid congestion handling technique, packet dropping probability should be equivalent to zero as in the case of our suggested hybrid congestion technique (ECN + IFRC). In case of previously proposed hybrid congestion technique (ECN + TCP) packet dropping is effective (non-zero) and link utilization is 100% therefore; it possesses the more chances of congestion rather to our suggested hybrid congestion handling technique (ECN + IFRC). Furthermore, the other prior hybrid technique (ECN + RED) cannot provide better link utilization and packet dropping capability as compared to our proposed scheme. ECN-RED has low packet dropping rate (Non-Zero) and link utilization is ambiguous because it depends on the buffer size and if buffer size is small the link utilization is poor and if buffer is fully occupied then it increases the delay. Hence, proposed hybrid congestion scheme (ECN + IFRC) is superior to prior schemes.

5. CONCLUSION

Packet dropping is a prime metric rather to delay metric because the ideal control of packet loss ratio means ideal utilization of delay. Congestion escaping can enormously be improved with a joint committee of congestion handling schemes having packet dropping rate equivalent to zero with maximum link utilization because these ideal values mean all other congestion parameters (Latency, Jitter, Throughput, bandwidth, Fairness and Queue length) are ideally be satisfied. Minimized or equivalent to zero percent packet dropping rate lies under maximum link utilization and these both parameters are further dependent on all other congestion parameters. Our proposed hybrid congestion handling scheme [Explicit Control Notification (ECN) + Interference aware fair rate control (IFRC)] possesses equivalent to zero percent packet dropping rate with maximum link utilization as compared to prior conveyed hybrid congestion management schemes (ECN-RED , ECN-TCP). Hence, our convey approach is superior to the existed congestion management schemes in terms of all discussed congestion parameters, therefore, we confidently advised the researchers to utilize our proposed congestion managing scheme to avoid congestion in packet switched networks.

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