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Congestion Control for Internet Media Traffic

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***Abstract*—**In this paper we investigated a number of the Internet congestion control algorithms that has been developed in the last few years. It was obviously found that many of these algorithms were designed to deal with the Internet traffic merely as a train of consequent packets. Other few algorithms were specifically tailored to handle the Internet congestion caused by running media traffic that represents audiovisual content. This later set of algorithms is considered to be aware of the nature of this media content. In this context we briefly explained a number of congestion control algorithms and hence categorized them into the two following categories: i) Media congestion control algorithms. ii) Common congestion control algorithms. We hereby recommend the usage of the media congestion control algorithms for the reason of being media content-aware rather than the other common type of algorithms that blindly manipulates such traffic. We showed that the spread of such media content-aware algorithms over Internet will lead to better congestion control status in the coming years. This is due to the observed emergence of the era of digital convergence where the media traffic type will form the majority of the Internet traffic.

***Keywords*—**Congestion Control, Media Traffic.

I. Introduction

THE emerging digital convergence era, with its applications, imposed an increase in the media traffic over Internet represented in audiovisual content traffic. In the last few years a number of congestion control algorithms have been developed to handle the problem of Internet congestion resulting from the increase in the media and in other traffic over Internet. Upon our survey on the congestion control algorithms we found a number of well-known algorithms based on modifying the Transmission Control Protocol TCP such as TCP Reno [1], TCP New Reno [2], TCP Vegas [3], TCP Tahoe [4], TCP Westwood [5], H-TCP [6], HS-TCP [7], FAST TCP [8], Scalable TCP [9], and CUBIC [10]. Besides, another group of algorithms which are based on TCP modification, but are also less mature, such as TCP Hybla [11], TCP Fusion [12], TCP YeAH [13], and TCP Illinois [14]. On the other hand another set of algorithms that handle the problem of congestion control as well but taking into consideration the nature of the media traffic that is increasingly occupying the Internet bandwidth. These algorithms are considered as tailored for such traffic and has passed through some research enhancement to reach maturity namely as SCTP [15], TFRC [16], MTFRCC [17], PQAM [18], and SSVP [19]. We also added some other trial optimizations to control congestion for media streaming applications. This later set of algorithms is content-aware and does not deal blindly with all Internet traffic as a bulk of consequent packets.

In this paper an investigation is made to these algorithms and a classification accordingly as well. This is followed by a recommendation for the usage of a named category of these algorithms rather than the other category in order to efficiently control congestion over Internet.

The rest of this paper is organized as follows: Section II shows our investigation made to the mentioned congestion control algorithms. Section III is the categorization section for these algorithms, and section IV concludes and states the future work.

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II. Congestion Control Algorithms Investigation

In this section we briefly investigate a number of the congestion control algorithms. Then we categorize these algorithms into content-aware algorithms and common algorithms based on the investigation.

A. TCP Reno and New Reno

TCP Reno algorithm is based on four main mechanisms of work that are namely as follows: i) slow start. ii) Congestion Avoidance. iii) Fast Retransmit. iv) Fast Recovery.

Slow Start is to solve the compression problem through manipulation of congestion window (cwnd). *Congestion Avoidance* handles the problem of lost packets. Congestion is normally detected when the percentage of lost packets, that have not been received though sent, typically exceeds 5% [20]. *Fast Retransmit* works when a packet is received that is out of sequence, the receiver sends duplicate ACK, this can occur upon losing a packet as well. *Fast Recovery* is active when the receiver receives three packets and the lost packet is recognized, hence (cwnd) sends this packet again.

TCP New Reno is obviously an enhancement of TCP Reno algorithm by introducing the duplicate acknowledgement counting (DAC) as the loss recovery mechanism. DAC has a problem of sending unauthorized transmission of segments that may exceed the requirements of TCP congestion control. DAC does not provide the appropriate congestion control for fast recovery also.

B. TCP Vegas , TCP Tahoe, and TCP WestWood

The most distinguished difference between TCP Reno, New Reno and TCP Vegas lies in the later congestion avoidance mechanism. TCP Vegas utilizes a sophisticated mechanism of varying its (cwnd) for bandwidth estimation, while TCP Reno and TCP New Reno rely on the packet loss detection.

TCP Tahoe is also one of the congestion control algorithms that is window based. It also increases and decreases the (cwnd) as a consequent of packet loss. TCP Tahoe uses the AIMD mechanism, but yet in a different way than used by TCP Reno and TCP New Reno.

TCP Westwood TCPW functions on the basis of a novel congestion control window mechanism named as eligible rate estimation (RE) as well as bandwidth estimation BE. BE is done in TCPW via measuring the inter-arrival intervals, this measure directly controls the (cwnd).

C. FAST, HS-TCP, H-TCP, Scalable TCP, and CUBIC

FAST is a congestion control algorithms designed for high-speed networks. FAST implements a self clocking in streaming individual packets mechanism in addition to another mechanism to increase window size smoothly. Burstiness in FAST is reduced by limiting the number of packets that can be sent when an ACK pushes the congestion window by a large amount. A window pacing methodology specifies how to increase the (cwnd) over the idle time of a connection to its predetermined target. An adequate amount of scheduling overhead is used in this burstiness reduction. FAST can respond to queuing delay and packet loss. The (cwnd) is updated periodically in FAST according to the average round trip time RTT and average queuing delay.

High Speed-TCP HS-TCP is another congestion control algorithm for high-speed networks. HS-TCP is built on the basis of changing the TCP response function. It can achieve higher per connection throughput without demanding unrealistic low packet loss rates. It can reach a high throughput in a speedy manner and in slow start. Reaching high throughput in FAST is done over long delays and upon recovering from multiple retransmit timeouts, or when ramping-up from a period of small congestion windows.

H-TCP is a third congestion control algorithm for high speed networks. H-TCP focused on large (cwnd) that is associated with high bandwidth delay products (BDP) paths. The increase in the network speeds imposes the prevalence of these BDP paths. H-TCP proposed a change in the AIMD algorithm to suit long-lived flows. H-TCP made no major changes in slow-start mechanism and was also keen on applying minor modifications on the existing TCP paradigm.

Scalable TCP introduced the multiplicative increase multiplicative decrease concept (MIMD). Scalable TCP improves the network performance, especially when high available bandwidth exists on long haul routers. It can be easily implemented in current TCP stack and incrementally deployed on the current network devices without any need to change them. It is considered as a modification of HS-TCP and it benefited from all previous congestion control algorithms.

CUBIC is a high speed congestion control algorithm. It is a modification on an earlier version, it simplifies the window control, improves TCP friendliness and RTT fairness. A cubic function in terms of the elapsed time since the loss event is used to handle the CUBIC window growth. This function provides good stability and scalability. It has real-time nature that preserves the window growth rate independent of RTT. This feature enables the protocol to be TCP-friendly under both short and long RTT paths.

D. TCP Hybla, TCP Fusion, TCP YeAH, and TCP Illinois

TCP Hybla is an enhancement of TCP for heterogeneous networks. It stems from the analysis and evaluation of the (cwnd) dynamics of the TCP standard versions namely; Reno, New Reno, and Tahoe. The modification made by Hybla removes the RTT performance dependence. TCP Hybla advantage was evident in the performance over satellite links. It managed to reduce the aggressive penalization of such links type.

TCP Fusion is a hybrid congestion control algorithm for high speed networks. It combines the loss-based concept and the delay-based algorithms features. It tries to make use of the residual capacity effectively without affecting concurrent flows that will be using the widely deployed TCP-Reno most probably. To accomplish this task TCP Fusion mingles the mechanisms used by TCP Reno, TCP Vegas, and TCPW in congestion avoidance. The cwnd in TCP Fusion is not drastically decreased and is increased in a smart way according to the congestion estimated using RTT. TCP Fusion showed superiority in the TCP-Reno friendliness characteristic.

TCP YeAH stands for yet another high speed TCP. It has two modes of operation; fast and slow. The cwnd is incremented aggressively during the fast mode, while in the slow mode TCP YeAH acts like TCP Reno. The mode is determined based on the number in the bottleneck queue. TCP YeAH tried to exploit the network capacity efficiently, reduce the stress induced to the network by TCP Reno, and be friendly to it meanwhile. TCP YeAH design goals also included being internal and RTT fair, and not to be hindered from optimum performance due to small link buffers.

TCP Illinois utilizes AIMD where the increase and decrease parameters are controlled according to the estimates of queuing delay and buffer size. When no queuing delay is detected the rate of additive increase is fixed at 10 packets per RTT; when the estimated queuing delay increases the additive increase mechanism is gradually decreased to reach 0.3 packets per RTT. This decrease is done once the estimated queuing delay is at its maximal value and the network buffers are expected to be full. If the RTT is close to its maximum value, then the loss is deemed as buffer overflow. As RTT gets smaller, then loss is considered as packet corruption. TCP Illinois shows poor performance as path BDP increases.

E. SCTP, TFRC, MTFRCC, PQAM, and SSVP

Streaming Control Transmission Protocol SCTP is a general purpose transport protocol for IP data communications networks. It works in the transport layer just like TCP and User Datagram Protocol UDP. SCTP mainly provides reliable end-to-end transmission over IP networks. It also supports multi-streaming and multi-homing. Many institutes and universities paid a lot of attention to SCTP regarding both research and development. SCTP accomplishes the transmission task between two SCTP end-points and it is connection-oriented. One SCTP association can have multiple streams. The multi-streaming feature allows for multiple streams to co-exist in an association. SCTP utilizes a Streaming Sequence Number SSN that is assigned to each stream to maintain their order. This mechanism permits SCTP to be reliable and secure. SCTP multi-streaming can tolerate some packet loss in a stream without affecting the other co-existing streams. SCTP uses U flag to distinguish the ordered data chunks from the unordered ones. The U flag is simply a binary value that can be set to one to flag an unordered group of segments. This flag can enable the receiving terminal to reduce its waiting time taken to receive the packets in order, and wait for the lost data as well. The multi-homing feature in SCTP makes it possible for it to handle more than one transport address as a destination address for the given data, which is something common in the Internet media traffic.

TCP Friendly Rate Control TFRC is a version of Data Congestion Control Protocol DCCP that uses the Congestion Control ID number three CCID3. TFRC is based on TCP Reno’s throughput equation. TFRC was primarily designed for serving unicast media streams over wired Internet. TFRC was observed to highly degrade in performance over wireless networks and an effort has been done in enhancing its performance in the wireless environment [21]. TFRC leads to a smoother rate based congestion management than the abrupt changes that AIMD leads to.

Media and TCP-Friendly Rate based Congestion Control MTFRCC is mainly designed to serve scalable video streaming over Internet. MTFRCC made use of the mechanism known as utility-based model that uses the rate-distortion function as the application utility measure for optimizing the overall video quality. It also applies a two-timescale approach of rate averages (long-term and short-term) to satisfy both media and TCP-friendliness. MTFRCC has shown superiority in smoothness in different congestion levels. MTFRCC has been tested also regarding its responsiveness and aggressiveness in the situations of sudden changes in the available bandwidth. MTFRCC had lower oscillations in the sending rate during transitional states. MTFRCC above all was concerned with the overall video quality received and viewed.

Priority Queue Algorithm for Media traffic PQAM is built basically built on the concept of giving support to more users who are placing requests in the Internet and hence reducing congestion. PQAM classifies data into text, audio, and video. It also assigns a separate queue for each of these data types and allocates the time to service and the available bandwidth to each data type accordingly. PQAM may enable receiving a number of text data requests that are assigned a bandwidth and time to service equal to one video data request. This methodology enables PQAM to allow for maximizing the number of users being served at a given time interval and minimizing congestion meanwhile.

Scalable Streaming Video Protocol SSVP is an end-to-end protocol that works as a payload on UDP. SSVP employs the AIMD mechanism and controls the sending rate by tuning the inter-packet gap IPG. Handling AIMD and IPG in SSVP is done with great care of smoothness and oscillation reduction to this rate, SSVP keeps the TCP-friendliness feature in mind also. SSVP managed to respond the network vagaries and was successful in real-time video transmission with remarkable performance. In cases of awkward network conditions that affect the perceptual video quality, SSVP applies a layered adaptation mechanism using the receiver buffering capability to adapt the video quality to the long-term variations in the available bandwidth. SSVP sends a refinement layer based on the status of the receiver buffer and the available bandwidth, avoiding unwanted layer changes that have an adverse result on the viewer-perceived video quality. SSVP succeeded to transmit a visually useful video under limited bandwidth constraints.

F. Optimization Trials

In [22] an optimization is made to AIMD algorithm to suit media streaming applications. This optimization is done in the frame of bandwidth efficiency, smoothness, and inter-protocol fairness. It is assumed in this context that SSVP is the underlying congestion control algorithm. It is observed that despite of the fact that multiplicative decrease is essential to accomplish fairness, it does not have to sacrifice throughput. The proposed optimization presents the congestion control parameters that are adaptable to the current network status. It protects the system from operating below the knee where a residue of the available bandwidth is not occupied and smoothness is traded as throughput fluctuates.

An optimized version of SCTP is proposed in [23] to transmit and receive video data. It tries to benefit from the feature of partial reliability, partial order, and multiple streams in SCTP. This optimized version attempts to fairly share the bandwidth, it resolves the network congestion via the early detection of it and adapting the transfer rate of the video data accordingly. It makes use of the encoding mechanism of this video data and its different frame types, with different degrees of importance, to achieve its mission.

Another optimization is found in [24], it deals with congestion across a video dominated Internet tight link and seeks resolving it. It was demonstrated in this context that when controlled flows of different types compete across a tight link it is possible for TFRC to exceed the available bandwidth leading to high packet loss and consequently poor video quality at the receiver side. It was shown that fuzzy-logic control is more flexible in cases of video streams domination.

We refer also to [25] and [26] as adequate surveys, approaches, and steps towards appropriate evaluation that are related to the field of Internet congestion control.

III. Congestion Control Algorithms Categorization

We hereby present our proposed categorization, which is based on the investigation of the previous section, to the above mentioned algorithms of congestion control over Internet (see Table I). The table states the algorithms vs. the feature of the media content-awareness where the word "Yes" denotes that this algorithm is media content-aware while the word "No" denotes that the algorithm is not.

The table shows that SCTP, TFRC, MTFRCC, PQAM, and SSVP are media content aware congestion control algorithms. It also shows that TCP Reno, TCP New Reno, TCP Vegas, TCP Tahoe, TCP Westwood, H-TCP, HS-TCP, FAST TCP, Scalable TCP, CUBIC, TCP Hybla, TCP Fusion, TCP YeAH, TCP Illinois are of the other common congestion control algorithms type, and are not media content aware. The table also draws the attention to the fact that the media content-aware algorithms form a minority among the total number of the algorithms developed for the congestion control purpose in the last few years.

We generally recommend the deployment of one of the media content-aware algorithms over Internet to handle congestion. Our recommendation is based on the statistics made that show the invasion media traffic to Internet over the last few years. This invasion mandates, in our point of view, the biasing of the Next Generation Networks NGN protocols towards the media traffic.

TABLE I
CONGESTION CONTROL ALGORITHMS CATEGORIZATION

No.	Algorithm	Media Content-Awareness
1	TCP Reno	Not Media Content-Aware
2	TCP New Reno	Not Media Content-Aware
3	TCP Vegas	Not Media Content-Aware
4	TCP Tahoe	Not Media Content-Aware
5	TCP Westwood	Not Media Content-Aware
6	H-TCP	Not Media Content-Aware
7	HS-TCP	Not Media Content-Aware
8	FAST TCP	Not Media Content-Aware
9	Scalable TCP	Not Media Content-Aware
10	CUBIC	Not Media Content-Aware
11	TCP Hybla	Not Media Content-Aware
12	TCP Fusion	Not Media Content-Aware
13	TCP YeAH	Not Media Content-Aware
14	TCP Illinois	Not Media Content-Aware
15	SCTP	Media Content-Aware
16	TFRC	Media Content-Aware
17	MTFRCC	Media Content-Aware
18	PQAM	Media Content-Aware
19	SSVP	Media Content-Aware

IV. Conclusion and Future Work

Several congestion control algorithms for Internet traffic have been developed and tested in the last few years. A group of these algorithms deals with the Internet traffic as a train of packets and lacking the awareness of its content nature even if it is of the media type. The other group of these algorithms is media content-aware; and pays attention to the nature of the media data type and its transfer requirements. The most popular congestion control algorithms were classified into two categories according to this concept in this paper. It is recommended, in our point of view, to use a content-aware congestion control algorithm to handle the Internet congestion problem. This is due to the obvious prevalence of the media data type over Internet in the last few years and the expected increase of its invasion in the near future as a result of the digital convergence implications. Our future research will include the testing of each of these content-aware algorithms performance utilizing various kinds of media content. We will try these algorithms on news, sports, entertainment, and educational videos and see if an algorithm performs for a given content type better than the others.

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