Performance Measures for Congestion Control Techniques in a Wireless Sensor Network

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Abstract- Internet communication this day's houses different types of data transmitted concurrently and as such requires a well-organized and unfailing transmission method which is crucial to achieve high performance in networking environment. Congestion happens when data to be transmitted by the network is more than the strength available for the network to accommodate. Limited capacity of the network results to a long time waiting for these data to be processed while in transient and transmitted to the receiver. Data are lost as a result of prolonged waiting and as such could not get to their respective destinations. It is noted that congestion during data transmission is as a result of the way protocols are implemented, tight transportation routes, size of packets, speed of the sender and receivers machine which brings about packet lost. During congestion, the amount of data passing through the network may reduce to nothing and data waiting to be processed may become very high. A technique for controlling congestion will help a network recuperate from its blocked status. Control techniques are recovery tools for a congestion territory. Various techniques are in existence for managing a network congested territory. Our focus is to analyze congestion control techniques using standard measures and choose the best amongst the selected technique to perform control as many researchers have proposed techniques for controlling it.

Key Words- Congestion, Packet Loss, Throughput, Link Utilization,

I. INTRODUCTION

Packet loss is a very big issue in networking environment as users struggle to access same properties concurrently. Therefore, it is eminent to avoid extraordinary rate of loss during transmission of data from senders to receivers. More so, once packets are lost before getting to its required destination, resources put in place are wasted. Congestion happens when the space to be occupied by the data for processing before sending to their respective destinations is minute to carry out this task in a network. Data loss is as a result of but not limited to poor signal strength at the terminus, normal or human intrusion, unwarranted noise, hardware disaster, software exploitation, or overtaxed network nodes, protocol in use [9]. Control can be successfully attained by allocating the signal for processing across several connections in a network. In linkages, overcrowding brings about all-inclusive
channel standards to reduce and loss rates to increase, which results in buffer drops and increased delays. Node-level congestion and link-level congestion are basically the two congestions that could take place. Buffer overflow is a type of node-level congestion that brings about packet loss and long waiting time for data to be processed [8]. To mention but a few, the outcome of packet loss are severe defacement of acknowledged data, fragmented pictures, partial deficiency of received data [9]. Network resource management and traffic control are general ways of dealing with congestion. They extend resources to quieten bottleneck when it happens. Power control and multiple radio links are ways to expand signal processing and incapacitate congestion.

II. CONGESTION

Congestions happens in a communication network when applied load is more than the space the load can accommodate. Congestion control are techniques and scheme used to manage overloading when it occurs and keeps the load below its expected range. Congestion is unavoidable in a network because the devices like routers and switches have spaces that hold data and after it has been processed. Once an incoming packet arrives the router, the packet in there undergoes 3 different stages before it is finally sent to its terminus.

1). The queue in the router houses the arrived packet while waiting to be checked.
2) Packet is processed.
3) Processed packet remains in the queue and waits for its turn to leave and be transmitted.

We also need to know that if the rate at which packets arrives the queue is more than the rate packets are processed, arriving packets will have to wait in the queue for a longer time before it be processed. Also, if the rate at which the processed packets leave the queue is slower than the processing rate, packets in queue waits longer before circulation.

III. PACKET LOSS

Packet loss is said to occur when one or more packets transmitting in a network could not get to their respective destinations. This could be as a result of overcrowding. Packet loss is inevitable in communication links in as much as the routers used has a
particular queue size and once the queue is full and can no longer accommodate more packets, definitely there will be loss.

IV. TWO GENERAL METHODOLOGY FOR TRAFFIC CONTROL IN A NETWORK

A. Hop-By-Hop

It is a control system that reciprocates fast to actions. Habitually, it is challenging to regulate the packet-forwarding rate at in-between nodes.

B. End –to-End

This type of control system can inflict precise level modification at source node and streamline the strategy at in-between nodes. It consequences are dawdling actions and depends immensely on round-trip time (RTT).
Inevitable congestion in networks happens when data submitted is more than space available for its processing. Congestion causes channel standards to reduce and loss rates increase. This brings about packet drops, enlarged interruptions, squandered drive, and requires re-transmissions and in a great deal lessen the production and networks lifespan. Also, networks have restrictions on drives, retention and information measures. Therefore, dynamic and well-organized data broadcast protocols are essential to lessen bottle neck emanating from dwindling stations and additional load.

V. SOURCES AND CAUSES OF CONGESTION

Linkages comprises of nodes dispersed within a region by means of individual or added sinks. As traffics produces, such nodes increase in size, the provided data overshoot accessible volume in addition the system turn out to be overloaded. The foremost bases of congestion consist of buffer overflow, passage conflict, intrusion, packet accidents, etc. Buffer overflow happens when the volume of sent packets is more than offered space. Conflict happens between separate flows and separate packets. Intrusion occurs when there is concurrent transmissions along several paths within corporal closeness [2]. Packet accidents points out beneath side by side bottleneck which progresses to packet loss. As such, congestion brings about the deterioration of passage performance and loss rate increases. Throughput reduces as a result of the harmful nature of Congestion. This situation furthermore causes waste of assets, energy, as well as event detection consistency.

VI. SELECTTED TECHNIQUES FOR CONTROLLING NETWORK CONGESTION

Congestion control monitors the procedures of overseeing the overall inflow of data to keep movement points on a satisfactory rate. It is all about directing entries to escape congestive breakdown, endeavoring to side step excess subscription. Congestion control takes into consideration, size and use necessities of the network [5]. A number of methods were projected to address these encounters. The commonly used congestion control techniques remain:

A. Drop Tail

Current internet routers commonly use this technique because of its simplicity. It is a passive queue management (PQM) algorithm that applies first-in-first out (FIFO) based queue of restricted size, that humbly drops arriving data when queue is filled. Due to its simple and decentralized nature, its implementation is stress-free, it is appropriate to heterogeneity. It offers better link consumption and it fascinate busty traffic.
Current internet uses TCP Reno router on Drop tail technique. It has two main weaknesses as discussed in [1] which is, its lock-out behavior and full queue phenomena. Lock–out behavior includes the exploitation of accessible bandwidth by a particular or a limited sources whose end results to worldwide harmonization [7]. When a queue is full, it is referred to as full queue. This has its major downside and subsequently outcomes to bulky end-to-end delays. Approaches like Drop front or Indiscriminate drop resolves problem of lock out behavior nonetheless are incapable to resolve the problems of full queues. To handle these problems, early congestion detection matters a lot and consequently to accept the causes that brings about congestion through a notice earlier before queue gets filled up.
Drop tail is best used to implement network schedulers in network equipment with limited size. Drop tail does not distribute buffer space fairly since there is no differentiation of traffic from different sources. Sources with higher traffic volume will take more buffer space. Hence it is not suitable for networks with multiple TCP connections because a buffer overflow will cause TCP global synchronization, which reduces the network throughput and utility significantly.

B. Additive Increase/Multiplicative-Decrease

This applies feedback control algorithm. It is majorly used to implement TCP window adjustment as discussed in [3]. It exhibits fair behavior with bulk data transfer. It is best used for TCP Reno and Tahoe Routers. When congestion occurs, AIMD in a straight line increases the congested space with rapid reduction. This technique increases the congestion space by 1 maximum segment size (MSS) every round trip time (RTT) up to the discovery of packet loss. Upon discovery of packet loss, there is multiplicative decline in space size.
As a feedback control method best known for its use in TCP congestion control, AIMD is best used for networks with several TCP connections. It is appropriate for applications such as bulk data transfer. On the contrary, all of its flows have the same RTT and its network response arrive same time to all users even when they have same RTT.

C. Partial Buffer Sharing

This technique plays a crucial role in controlling congestion in routers. It meritoriously pedals the distribution of buffer to numerous traffic sessions according to their delay limitations. Its enthusiasm is to meet the different requests of Quality of Service (QoS) which can be succeeded by refining the loss performance of high precedence traffic while degrading the performance of the low precedence traffic. Packets get into the queue in descending order and a checker is installed to check the delay limitation as to know is of higher precedence.

PBS Algorithm works as follows:

1). Step 1: Set descending sequence of threshold to Ni (Ni >0, i=1, 2… R) corresponding to Q with finite capacity & single

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server having R priority classes

2). Step 2: To meet desired QoS demands adjust threshold values under different load conditions accordingly

3). Step 3: The highest priority jobs of class can join the queue subject to space availability in Q

4). Step 4: Jobs with lower priority class i (i = 2… R) can only join the Q if total jobs in Q, N< Ni (Ni ≤ Ni-1)

5). Step 5: Once the number of jobs waiting for service reaches Ni, all jobs with lower priority will be lost on arrival but higher priority jobs can still join the queue until it reaches threshold value, Nj (j = 1,…, i-1).

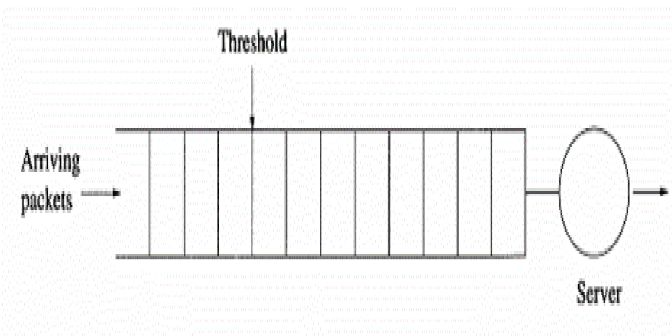

Fig-1: Threshold based PBS technique [7].

The figure above adopts a buffer using single FIFO queue. Resolution is centered on Maximum Entropy Approach for an unwavering GE/GE/1/N censored queue with a particular server, finite size and several sessions of traffic in PBS technique in relations of the mutual collection ME queue length distribution {P(n), n ∈ Ω} is given by [7],

$$P(0) = \frac{1}{Z}$$

$$P(k) = \sum_{i=1}^{R} \Pr ob\left(Q_{i;k}\right)$$

$$= \frac{1}{Z}\left(\prod_{j=1}^{R} x_j^{k_j}\right)\sum_{j=1}^{R} k_j \left(\frac{\left(\sum_{i=1}^{R} k_i - N_j\right)!}{\prod_{i=1}^{R}\left(k_i - N_j\right)!}\right) g_j y_j^{\delta(k)}$$

The external busty movement in addition to service interval have remained demonstrated by means of the generalized exponential (GE) circulation. The overcrowding or packet loss possibility is given as [7]:

$$\pi_i = \sum_{k=0}^{N} \delta_i(k)\left(1-\sigma_i\right)^{[N_1-k]^+} P_{N_1}(k)$$

Where

$$\delta_i(k) = \begin{cases} \dfrac{r_i}{r_i(1-\sigma_i)+\sigma_i}, & k = 0 \\ 1, & \text{otherwise} \end{cases}$$

PBS is best used when there is need for reservations or different services that provide QoS.

VII. COMPARISM OF CONTROL TECHNIQUES

This section tends to compare the methods for network congestion control discussed in section 4. The comparim is based on the analysis using some standard performance measure as follows.

A). Throughput (T)

In data transmission, it refers to the volume of data moved positively starting from one abode to another in a specified time frame. It is measured in megabits per second (Mbps) or gigabits per second (Gbps).

B). Packet Loss Probability (PLP)

Packet loss occurs when packets flop to get to their terminus. The loss ratio of each flow should be very close to loss ratio of the combined traffic. Then it can be concluded that the estimated loss possibility of aggregated traffic can remain as individual packet loss probability.

C). End-to-End Delay or Latency (EEDL)

End-to-end delay or one-way delay (OWD) is the time taken for a packet to be transmitted from source to destination.

EEDL= processing time + queuing time + transmission time + propagation time.

Table 1: Comparison of Drop tail, AIMD and PSB

METHODS	T	PLP	EEDL
Drop Tail	Once buffer is full, it reduces throughput.	A packed queue status results to packets dropping.	Full queue results transmission delay.
AIMD	Once there is congestion and packet loss at hand, window size reduces.	Upon congestion, packets are lost and congestion space is improved by 1 maximum segment size (MSS) every round trip time (RTT)	End-to-end delay intensifies as window size drops.

PBS	Figure 2 indicates throughput values for delay sensitive traffic and delay tolerant streams which varies by increasing the threshold position.	Figure 3 showcases that once there is an increase in threshold location, the packet loss probability for delay sensitive traffic reduces.	Figure 4 shows that, there is an increase in response time as threshold positions increases

VIII. RECOMMENDATIONS

To successfully control the distribution of buffer to different circulations, the Partial buffer sharing methods is best as it has to do with the application of delay constraints. We recommend this technique for congestion control considering the important role it plays towards its successful management of network congestion. PBS addresses the demand of QoS by refining the loss show of the high precedence traffic while degrading the show of the low precedence traffic.

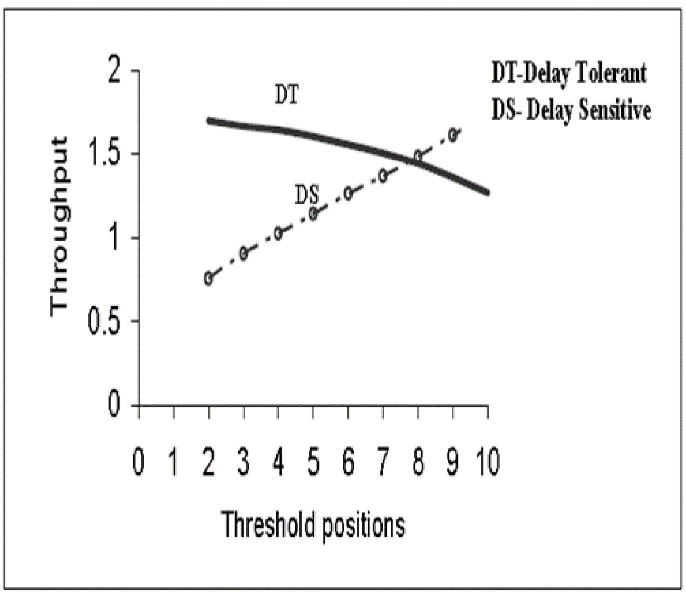

Fig-2: Outcome of threshold positions on throughput [7]

It is noted that, once threshold point intensifies, the packet loss probability for delay sensitive traffic drops although it rises in instance of delay tolerant traffic as shown below.

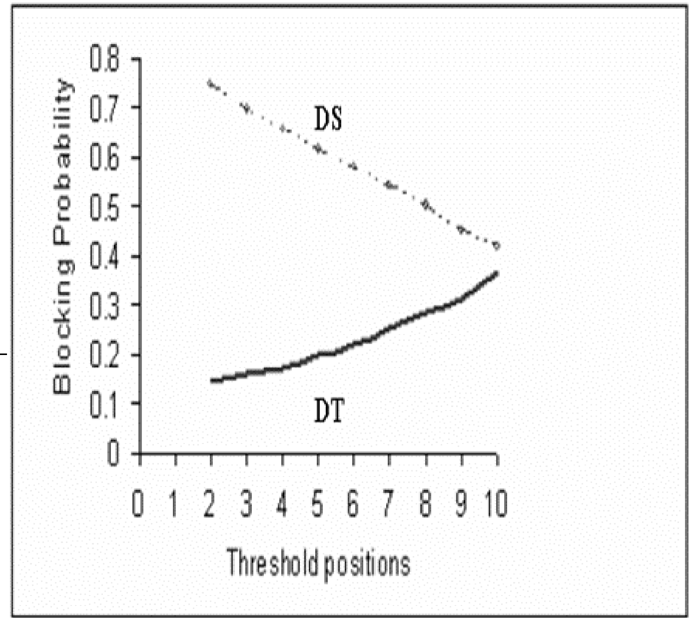

Fig- 3: Outcome of threshold positions on packet loss Possibilities [7]

The figure below indicates rise in mean response time and a rise in threshold locations together in event of DS and DT traffic streams.

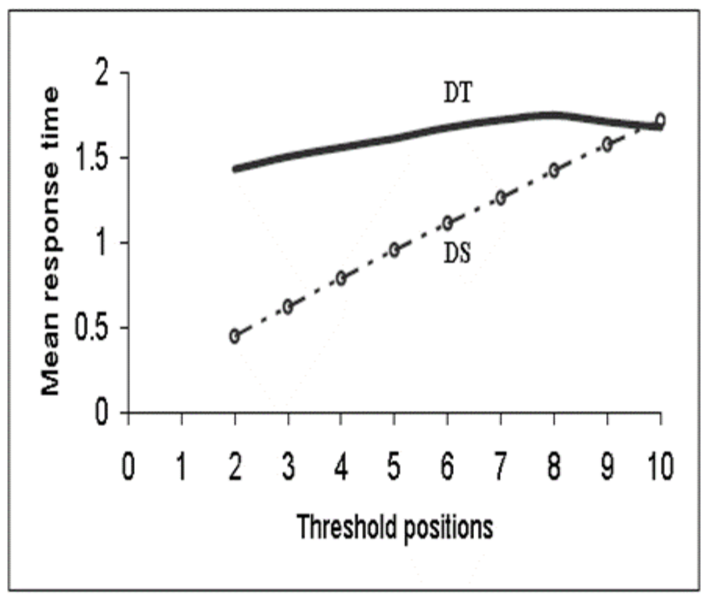

Fig.4. Effect of threshold positions on mean response time [7]

IX. CONCLUSIONS

In this work we have presented a fair study of congestion control techniques using more or less key performance measures including Throughput (T), Packet Loss Probability (PLP), End-to-End Delay or Latency (EEDL). It is perceived that no particular congestion control technique is a one fix all scheme owing towards the comprehensive quantity of considerations that partake on system's performance. Observations show that currently, high speed network and its nature of congestion is not well-known and cannot characterize the diverse intensities of congestion alongside with facts such as; what is an extreme condition of congestion? How long it is lost and what is the ratio of fallen packets? Thus, for this study's drive, putting into consideration results on our analysis we recommend Partial Buffer Sharing (PBS) method for Congestion Control.

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