Analysis on Differential Router Buffer Size towards Network Congestion

A Simulation-based

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Abstract—**Network resources are shared amongst a large number of users. Improper managing network traffic leads to congestion problem that degrades a network performance. It happens when the traffic exceeds the network capacity. In this research, we plan to observe the value of buffer size that contributes to network congestion. A simulation study by using OPNET Modeler 14.5 is conducted to achieve the purpose. A simple dumb-bell topology is used to observe several parameter such as number of packet dropped, retransmission count, end-to-end TCP delay, queuing delay and link utilization. The results show that the determination of buffer size based on Bandwidth-Delay Product (BDP) is still applicable for up to 500 users before network start to be congested. The symptom of near-congestion situation also being discussed corresponds to simulation results. Therefore, the buffer size needs to be determined to optimize the network performance based on our network topology. In future, the extension study will be carried out to investigate the effect of other buffer size models such as Stanford Model and Tiny Buffer Model. In addition, the buffer size has to be determined for wireless environment later on.**

Keywords – OPNET, network congestion, bandwidth delay product, buffer size

I. INTRODUCTION

Router plays an important role in switching packet over a public network. A storage element called as buffer is responsible to manage transient packets in a way of determining its next path to be taken and deciding when packets suppose being injected into network. Several studies [1-3] have agreed that the single biggest contributor to the uncertainty of Internet is coming from misbehavior of router buffer per se. It introduces some queuing delay and delay-variance between flow transitions. In some cases, packets are potential to be lost whenever buffer is overflow. Oppositely, it is wasteful and ineffective when buffer is underutilized. As a result, it shows some degradation in the expected throughput rate.

The main factor to increase the network performance is to seize the optimal size of router buffer. Currently, it is set either a default value specified by the manufacturer or it is determined by the well known "Bandwidth-Delay Product" (BDP) principal that has been invented by [4]. This rule is aiming to keep a congested link as busy as possible and maximize the throughput while packets in buffer were kept busy by the outgoing link. The BDP buffer size is defined as an equal to the product of available data link's capacity and its end-to-end delay at a bottleneck link. The end-to-end delay can be measured by Round-Trip Time (RTT) as presented in Equation (1). The number of outstanding packets (in-flight or unacknowledged) should not exceeds from TCP flow's share of BDP value to avoid from packet drop[5].

$$BDP\ (bits) = Available\ Bandwidth\ (bits/sec) \times RTT\ (sec) \quad (1)$$

In ideal case, the maximum packets carrying in a potential bottleneck link can be gain from a measurement of BDP_UB where there is no competing traffic. The BDP_UB or Upper Bound is given in Equation (2) as stated below:

$$BDP\_UB\ (bits) = Total\ Bandwidth\ (bits/sec) \times RTT\ (sec) \quad (2)$$

When applied in the context of the TCP protocol, the size of window sliding should be large enough to ensure that enough in-flight packets can put in congested link. To control the window size, TCP Congestion Avoidance uses Additive Increase Multiple Decrease (AIMD) to probe the current available bandwidth and react against overflow buffer. The optimal congestion window size is expected to be equal to BDP value; otherwise packet will start to queue and then drop when it "overshoots".

Today, several studies have been conducted to argue the realistic of BDP such as Small buffer which also known as Stanford Model [6] and Tiny Buffer Model [7]. They keep try to reduce number of packets in buffer without loss in performance. Larger buffers have a bad tradeoff where it increases queuing delay, increase round-trip time, and reduces load and drop probability compared to small buffers which have higher drop probability [8]. However, applications able to protect against packet drop rather than recapture lost time.

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The goal of this paper is to study the effectiveness of BDP on a simple network topology. This will be demonstrated on a group of users from a range of 5 until 1000 users. A simulation study is carried out with OPNET Modeler 14.5 [9].

The rest of the paper is organized as follows. Section II reviews the term of congestion from several aspects and briefly explain about a well known buffer sizing model, BDP. Section III describes the network model and evaluation metrics for the simulation. In Section IV, we analyses simulation results. Section V concludes the present paper and discusses some possible extensions of our work.

II. BACKGROUND STUDY

A. Congestion

In [10] stated that network congestion was related to the buffer space availability. For normal data transmission, the number of packet sent is proportional to the number of the packets delivered at destination. When it reaches at saturation point and packets still being injected to network, a phenomenon called as Congestion Collapse will be occurred. In this situation, the space buffer considers limited and fully occupied. Thus, the incoming packets need to be dropped. As a result, a network performance has been degraded.

Most previous studies [11-13] emphasized that the key of congestion in wired network is from network resources limitation. This limitation is including the characteristics of buffer, link bandwidth, processor times, servers, and forth. In a simple Mathematical definition, congestion occurred once there are more demands exceed the available network resources as represented by Equation (3).

$$\sum Demand > Available\ Resources \qquad (3)$$

In [13], the congestion problem has been widely defined from different perspectives including Queue Theory, Networking Theory, Network Operator and also Economic aspect. However, it still emphasizes on buffer-oriented activity and capability to handle unexpected incoming packets behavior. For instance, the access rate exceeds the service rate at intermediate nodes.

B. Rule of thumb

Most routers in the backbone of the Internet have a Bandwidth-Delay Product (BDP) of buffering for each link. This rule has been concluded based on an experimental of a small number of long-lived TCP flows (eight TCP connections) on a 40 Mbps link. The selection of TCP flows has been proved by [14, 15] that more than 90 % of network traffics is TCP-based. Meanwhile, the value of BDP that more than 10^5 bits (12500 bytes) is applicable for Long-Fat Network (LFN). In this case it refers to Satellite Network [16].

III. METHODOLOGY

In this study, a proper methodology has been designed to get an expected output. This can be referred to the following work flow depicts in Figure 1.

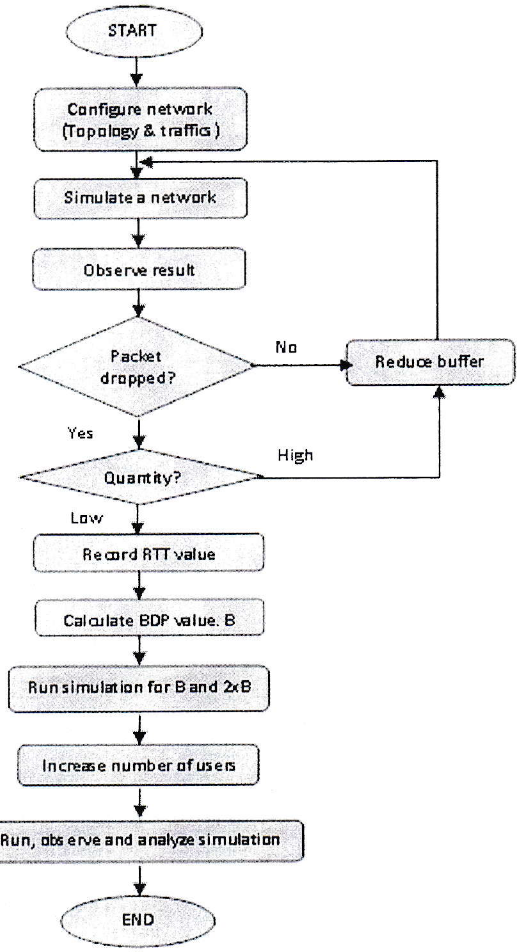

Figure 1: Methodology to be used

The first step demands for defining the value of the Round-Trip Time (RTT). This value can be set based on a normal data transmission where there no packets drop yet. To achieve it, the network needs to be configured based by using a default setting that available in simulation tool. Then, the memory size at router need to be adjusted until last configuration where there a small number of packets dropped appear. Once RTT has successfully estimated, a current buffer size will be recorded and then need to adjusted base on BDP model.

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The next step is to compare the effect of different buffer size as mentioned previously in Section I. There are two scenarios created to represent Scenario 1 (Small Buffer **B**) and scenario 2 (Large Buffer **2xB**). Both scenarios will be tested for a different range of users from 5 to 1000. Several parameters will be observed and then analyzed more detail later. This simulation will be run for 900 seconds.

IV. EXPERIMENTAL APPROACH

A. Network Environment Setup

In this section, a simple network topology which also known as dumb-bell topology was designed as illustrated in Figure 2. This topology is a typical model used by researcher to study congestion issues as stated in [17]. The network consists of three servers, LAN users, two intermediate routers and links interconnecting between them. For both links between servers/LAN users, the data rate is given as 100 Mbps. Meanwhile, routers are connected using Point-to-point Protocol (PPP) 1.544 Mbps.

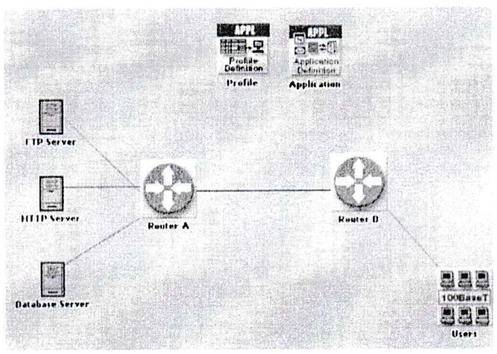

Figure 2. Proposed system network

For application configuration, TCP-based services such as File Transfer Protocol (FTP), Database and web browsing traffic (HTTP) were defined. Table 1 shows the traffic definition that used in our simulation.

TABLE I TRAFFICS DEFINITION FOR SIMULATION

Services	Description	Value
FTP	Command Mix (Get/Total) :	50%
	Inter-Request Time (seconds) :	360
	File size (bytes):	1000
Database	Transaction Mix (Queries/ Total Transaction) :	100%
	Transaction Interarrival Time (seconds) :	12
	Transaction Size (bytes) :	32768
HTTP	HTTP Specification :	HTTP 1.1
	Page Interarrival Time (seconds) :	10

B. Evaluation Metrics

In this study, the behavior of the packet once passing throughout Router B was observed. This study assumed that router maintains a single FIFO queue, and drop packets from the tail when the queue is full. This action is known as Drop-tail which is the most widely deployed scheme today. We collect some useful information such as number of packet dropped, retransmission count, end-to-end TCP delay, queuing delay and link utilization. This selection based on possible output to represent a possible picture of congestion phenomenon in the network topology.

V. SIMULATION RESULTS & ANALYSIS

In this section, simulation result for the impact of changing buffer sizes on network performance was presented. Simulations were run for Bandwidth-Delay Product (BDP) model. Based on Equation (1), we used two values of buffer sizes which are **B = 2000 bytes**, referred as the "small buffer" and another is given as **B = 4000 bytes**, referred as the "large buffer". This BDP values were calculated to show the differences buffer space availability towards network congestion.

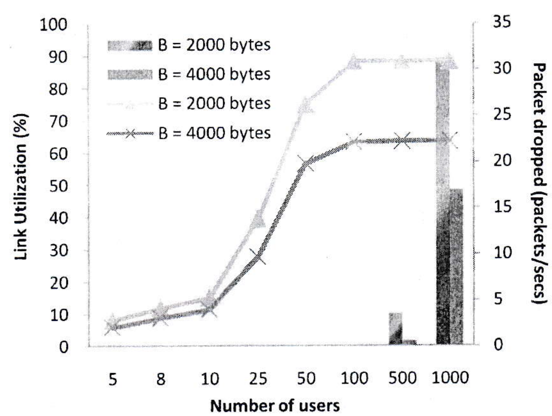

Figure 3: The influence buffer size to link utilization and packet drop

Figure 3 shows the influence buffer size to link utilization and packet drop when the number of users N is changed. To be clear, the line graph represents link utilization meanwhile the bar chart represents packet drop activity. For both graphs, it can be seen that "small buffer" always obtained high link utilization and high packet drop compared to "large buffer". To analyze this simulation result, we divide users into three grouping: Group A, Group B and Group C as shown in Table 2.

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TABLE 2. USERS GROUP

	Group A	Group B	Group C
Users	5-10	11-100	101-1000

For link utilization, we found that small number of user in Group A for only occupy backbone link less than 20%. Meanwhile Group B which has medium number of users keeps increases its link usage until 60-90% from the available link. However, the link utilization for Group C has remained at almost 60% (large buffer) and 90% (small buffer). This link saturation caused by buffer space limitation in Router B for both cases when it considered as fully occupied. As a result, the incoming packets start to be dropped.

For the bar chart information, the packet discarded obviously in Group C particularly when users count more than 500. The higher packet dropped was slightly 30 packets/second for "small buffer" and slightly 15 packets/second for "large buffer". It can be conclude that buffer space is still available and no packet drop when users is in Group A and Group B for BDP model.

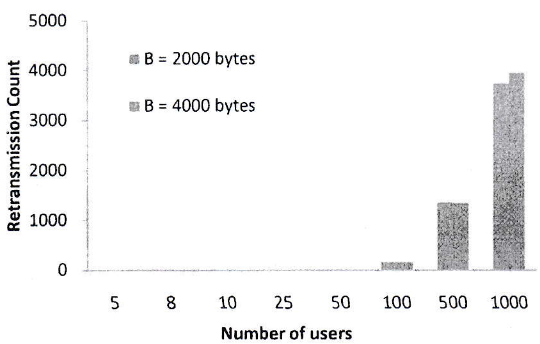

Figure 4: Packet retransmission

Figure 4 depicts the number of packet retransmission when the number of users N is changed. It can be seen that the retransmission activity has been detected started when the user reached 50 for "large buffer" and 100 for small buffer size. Based on TCP Congestion Control specification [18], each delivered packets must be acknowledged in time. If timeout or packets delay, sender will automatically do packet retransmission. By default, retransmission attempts are allowed not more than 3 times in sequence. If exceeds, the packet is assumed to be lost and then TCP Congestion Control mechanism will start to halve congestion window and reduce sending rate.

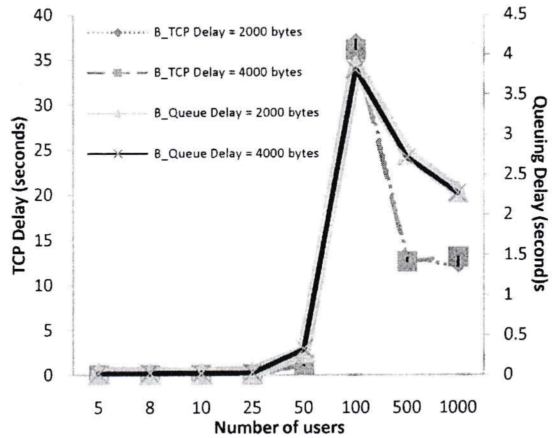

Figure 5: End-to-end TCP delay and Queuing Delay for different users

Figure 5 shows the End-to-end TCP delay and Queuing delay when the number of users N is changed. For both delays, it kept to increase rapidly when user between a range of 50 to 100. However, these delays start to drop when the link between routers started to be saturated. This action result from TCP congestion control that applies rate adaptation once network congested.

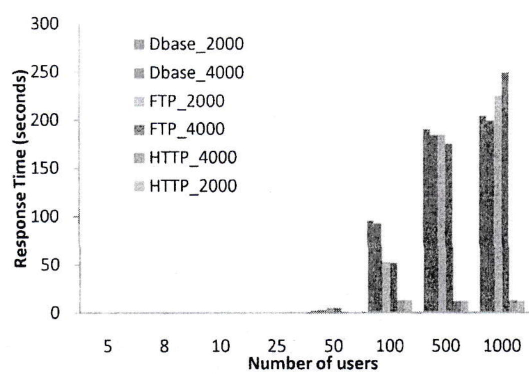

Figure 6: The influence of buffer size on the Application Response time

Figure 6 illustrates the influence of the buffer size on the applications response time when the number of users N is changed. For both buffer sizes, it can be seen that FTP and Database applications has higher response time compared HTTP services.

In summary, the determination of buffer size based on the Bandwidth-Delay product (BDP) gives a value of small buffer (B = 2000 bytes) and large buffer (B = 4000 bytes) to be used in understanding of their effects on network performance. By taking consideration on the influence of the growth of users in network, the packet behavior has

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been observed correspond to the availability of router buffer space such as link utilization, packet dropped, retransmission count, end-to-end TCP delay, queuing delay and application's response time.

From the simulation result discussed above, the buffer start to be congested when the user reach to 500. This assumption was based on situation where there are higher link utilization and higher packets dropped. The symptom of near-congestion situation can be observed from activities such as packets retransmission, end-to-end TCP delay, queuing delay and application response time. This symptom occurred when users are between 25 and 50.

VI. CONCLUSION

In this paper, the effect of router buffer size based on Bandwidth-Delay Product (BDP). Through a simulation, the value of small buffer is important element rather than large buffer in order to have better network performance. This also depends on number of users and applications running on a network. In the future, we plan to investigate the effect of other buffer size models such as Stanford Model and Tiny Model. Furthermore, the buffer size has to be determined for in wireless environment later on.

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