

# Correlation Analysis of Data Rate and Router Buffer Size on TCP Performance using OPNET Simulator

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## ABSTRACT

TCP/IP is a dynamic window based protocol that is used over the Internet. Its widely use is mainly based on its high dynamic nature of adaptability to any kind of network capacity. Since the development of TCP as an internet standard, special attention has been drawn to its behaviour over different network environment. Many TCP performance studies have been conducted on single TCP connection and multiple TCP connections through a sequence of finite router buffer capacity and data rates. However, the result of these works on TCP performance behaviour cannot be generalized because performance of TCP is dependent of the network environment. Each network environment has set of network elements, and application traffics generated at a particular period of time. Therefore, it becomes necessary to study the performance of TCP behaviour in any kind of network environment. Particularly, in a network environment of multiple TCP connections with UDP traffic flows. The paper studied the state of TCP behaviour in a network environment that establishes multiple TCP connections with UDP traffic flows, over a limited router buffer size and wireless link data rate. Using selected test cases, performance matrix of interest are observed, the reason for the unwanted behaviour are identified and estimate alternatives to avoid the problematic behaviour were identified.

## Keywords

Simulation, TCP traffic flows, UDP traffic flows, and TCP performance.

## 1. INTRODUCTION

TCP/IP is a dynamic window based protocol that is used over the Internet. Its usage is mainly based on its high dynamic nature of adaptability to any kind of network capacity, from a few bits to several gigabits per second. TCP is a reliable connection-oriented protocol that includes a built-in capability to regulate the flow of information, a function referred to as flow control. The flow of information is managed by TCP, by increasing or decreasing the number of segments that can be outstanding at any point in time. For example, under periods of congestion when a station is running out of available buffer space, the receiver may indicate it can only accept one segment at a time and delay its acknowledgment to ensure it can service the next segment without losing data. Conversely, if a receiver has free and available buffer space, it may allow multiple segments to be transmitted to it and quickly acknowledge the segments [5]. Each segment of memory waits to be

encapsulated with an IP header to form a datagram for transmission.

The datagram set for transit are structured in three slides and a “window” is placed over the slides by Gilbert [5]:

1. data transmitted and acknowledged
2. data transmitted but not yet acknowledged, and
3. data waiting to be transmitted.

Because this “window” slides over the three types of data, the window is referred to as a sliding window. Widmer *et al* pointed out that TCP congestion control mechanism using a smaller packet size than TCP sliding window size, would only achieve a fraction of the throughput justifiable according to its resource usage, if no corrective measure were taken [7]. Another scheme is to make active queue management such as RED drop packets in direct proportion to their size [4].

On the Internet, the TCP application occupies a higher and higher proportion in the total Internet traffic. However, the current TCP protocol does not work very efficient; it introduces extra overhead on the network, wasting the network’s capacity [3]. More importantly, takes a long time to fetch all the information that the user requires. This is especially obvious in network environment where the bandwidth capacity cannot cater for the total application traffic requirements.

## 2. TCP BEHAVIOUR IN AN ENVIRONMENT OF RELIABLE CONNECTIONS

It is the explosive growth of multiple reliable connections (MRC) in networks that causes severe traffic congestion problems and is common to see internet gateways drop a good percentage of the incoming packets because of local buffer overflows limited bandwidth. Bandwidth in the context of Internet access point is refers to the amount of information that can be passed through a communications channel in a given amount of time; that is, the capacity of the channel [6]. It is often expressed in bits per second, or alternatively bytes per second, leading to lots of confusion. Bandwidth figures are used to express the capacity of a network link or interface (e.g., a 100Mbps Ethernet interface) or the rate of information transfer.

Much research has been conducted on the performance of TCP including those by Bacceli and Hong [1], and Ott *et al* [9]. However, among the bottleneck resource is bandwidth; flows may use packets of different sizes. Resource sharing then depends on packet size and relies on transport protocol implementations. The amount of Internet Bandwidth that is available to each application traffic requirement on the shared link is therefore a factor that determines the degree of packets drop and services delivery of the entire network. When there are multiple TCP connections on the shared link, the queuing delay at the link affect the behaviour of the TCP more than it does for a single TCP connection. It is observed that in a network environment with multiple parallel connections, the queuing delay increases and cannot be predictable because the TCP connections tend to interfere with each other [8]. When a TCP connection gets its packets in the router, the other TCP connections will have to wait, starving from an unexpectedly long queuing delay.

Another study by Floyd and Jacobson revealed that when the router buffer is full, packets are dropped, much more from several parallel connections [4]. The number of dropped packets due to buffer overflow is expected to be higher when several parallel TCP connections are in transit, because the last-hop router is likely to be more congested. This causes all the TCP connections that suffered from packet loss, decreasing their congestion window at the same time, which causes a sudden decrease in the amount of traffic outstanding in the network until the congestion window for different TCP connections is increased again.

Whenever one of the parallel TCP connections ceases transmission, the other TCP connections uses the available bandwidth on the link and fill the router buffer with their packets. When the ceased TCP connection starts transmitting, it is possible that the router buffer will be full of packets queue, and the router drops the packet transmitted. Even when the retransmission timer expires, the router buffer may still be occupied by the packets of the other the TCP connections and the retransmission triggered by the retransmission timeout will be dropped again at the router. Therefore the TCP sender will have to wait for another retransmission timeout in order to retransmit the segment again. According to Braden *et al*, this kind of situation could continue repeating several times, which causes substantial unfairness among the parallel TCP connections [2].

In the other way round, the nature of congestion control of TCP connections cause unfairness, and whenever one of the TCP connections drops a packet, the dropped TCP congestion window reduces its transmission and causes the other TCP connections which did not have its packet dropped get a larger share of the available bandwidth [2]. If the packet loss cannot be recovered using fast retransmit, the sender has to wait for the retransmission timeout. This allows the link capacity to be used by the competing TCP connections and might lead to substantial unfairness as describe above.

### **3. UDP BEHAVIOUR IN A NETWORK ENVIRONMENT**

The User Datagram Protocol (UDP) is a connectionless, best-effort, non-error checking transport protocol [5]. There is no handshaking between sending and receiving transport layer entities before sending a segment. UDP was developed in recognition of the fact that some applications may require small pieces of information to be transferred. The use of a connection-oriented protocol would result in a significant overhead error checking to the transfer of data. Therefore, UDP

transmits a piece of information referred to as a UDP datagram without first establishing a connection to the receiver. The protocol is also referred to as a best-effort protocol [5]. To ensure that a series of UDP datagram are not transmitted into a black hole if a receiver is not available, the higher layer in the protocol suite using UDP as a transport protocol will wait for an acknowledgement. If one is not received within a predefined period of time, the application can decide whether to retransmit or cancel the session.

## **4. SIMULATION ANALYSIS APPROACH**

Performance evaluation is a critical component of systems research that allows the evaluation of new ideas, identification of problems and bottlenecks and optimization of existing systems. There are three general approaches to performance evaluation:

1. Prototyping: build it (or a scaled down version of it) and see how it works.
2. Analytical modelling: build a mathematical model of it and use it to analyze the system.
3. Simulation: build a software model of the system.

Prototyping is often not feasible or time consuming especially for large scale systems. It also provides limited control and observation. Similarly, analytical modelling cannot capture highly complex systems. Thus, simulation has emerged as an attractive alternative that is heavily used in performance evaluation of computer systems.

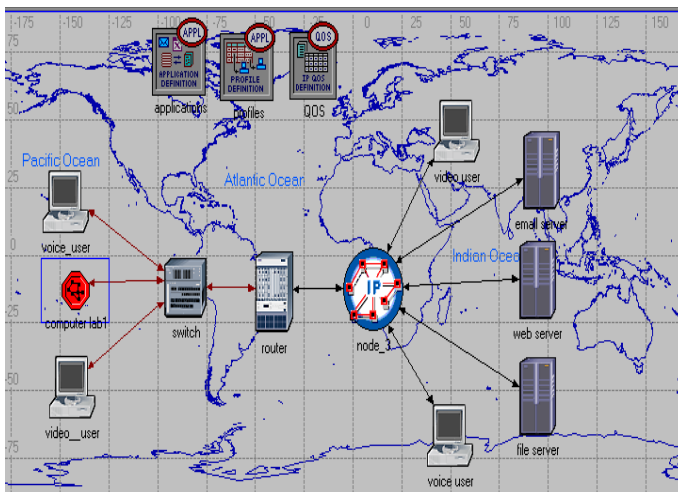
In addition, there are many obstacles in the process of performing some studies. The most well know among others is the building real test-bed for performance analysis. The cost is sometimes not effective or even not feasible. The intensity of the problem increases, as the need for different configuration parameters arises. For these reasons, it becomes necessary to build a simulation model and study it as a surrogate for an actual system. In this study, effort is made to build a model for simulating the traffic flows and TCP behaviour using one among widely acceptable network simulator, Optimum Network Engineering Tools (OPNET). OPNET is a graphic network simulator that provides a set of tools for network modelling, displaying statistics. Many predefined types of nodes are present and almost all widely used protocols and technologies are supported. The supported operating system is Windows only.

The use of network simulator to emulate an existing network environment has many merits. With the network simulator, the wireless link and buffer properties, and the emulated network architecture under TCP can be controlled. The predefined test configurations may also run automatically, which makes the execution of the performance measurements more convenient.

### **4.1 The Network Environment**

The target environment of study is a local area network (LAN) network generating UDP and TCP traffic flows. The network environment consists of ten client hosts each communicating with one of the three different server hosts via a wireless link and a last network element. The study concentrated on the behaviour of a client TCP in the LAN, in the presence of multiple TCP and UDP traffic flows. The LAN has bandwidth of 100 megabits, wireless link data rate of 64,000 bps and router buffer of 2MB. The network is assumed to provide traffic flows significantly faster than what the wireless link can

transmit. Figure 1 illustrates the elements which are emulated from the target network environment.



**Figure 1: Emulation environment.**

OPNET software was used to emulate the target network environment. In the emulation environment, performance tests were made using ten client hosts on the network and two different hosts were set to generate voice and video streams. Each of the ten hosts generates TCP traffic. Traffic shaping was not imposed in the network set up. The traffic flows are forwarded based on Best-effort scheme.

The network hosts were assumed to be running the Windows operating system. Each of the client hosts and the server hosts are the TCP endpoints communicating with each other using TCP connection. The TCP enhancements that are used in the performance tests are implemented only in the endpoint hosts.

The wireless link is emulated by issuing appropriate data transmission rate and propagation delay. Link layer retransmissions are emulated by causing an error delay for a packet. During the error delay no packets are released from the link receive buffer. The last-hop router queue is emulated using the input queue. The traditional first-in-first-out scheduling is used on the input queue. If there is no room for incoming packet in the input queue, emulator router discards the packet.

### 4.2 Network Environment Models

Two application traffics models are selected for the purpose of this study. The application traffic models are described below.

*Reliable connections environment without UDP traffics:* in this model, multiple TCP connections workload for transfer of only TCP traffic is setup. Each TCP traffic connection is transferring a bulk data from the fixed host on the Internet to a client host on the subnet of the fixed network.

*Reliable connections environment with UDP traffics:* in this model, multiple TCP and UDP connections are established. The TCP traffic connections are engaged in transfer of bulk data from a fixed host on the Internet to a client on the subnet of the fixed network. While UDP traffics are flows of point-to-point interactions. These flows could be video conference or voice over IP (VOIP) stream.

Number of tests using these workload models were selected and tested. Having multiple-parallel application connections transmitting over a bottleneck link has remarkable implications

on TCP. However, this research work is concern with TCP behaviour in the model, in the presence of UDP traffic flows.

### 4.3 Simulation Process Configuration

As described earlier, the target network path contains a wireless link and last-hop router. The main objective is to inspect the performance of TCP in a multiple reliable connections with the presence of UDP traffic flows over wireless link and router of a limited buffer memory, and to inspect certain details of TCP behaviour more closely.

Buffer size of 8mb and data rate of 64,000 kb were used as baseline for the performance analysis. However, three different router buffer sizes are combined with three different data rates scenarios were also analysed. The primary interest of the study is not to find exactly the optimal buffer size and data rate for the different scenarios. Hence the selection of the three different buffer sizes and data rates are to limit the number of test runs. Table 1 shows the data rates and buffer sizes selected for the performance tests.

**Table 1: Selected Data Rates and Router Buffer sizes**

Data rate (bps)	Buffer size (mb)
128,000	16
256,000	32
512,000	64

SACK scheme was used as the baseline technology for establishing a reliable connection. Using the same modelled environment, a number of TCP implementations were tested and compared to see which among give better performance. The selected TCP technologies are Reno and New Reno. The TCP attribute configuration for client and server hosts are summarised in table 2.

**Table 2: TCP Attribute Configuration**

Attribute	Value
Maximum segment size (bytes)	512
Receive Buffer (bytes)	8760
Delay ACK mechanism	Segment/clock based
Maximum ACK delay (sec)	0.2
Slow start initial count (MSS)	1
Duplicate ACK threshold	3

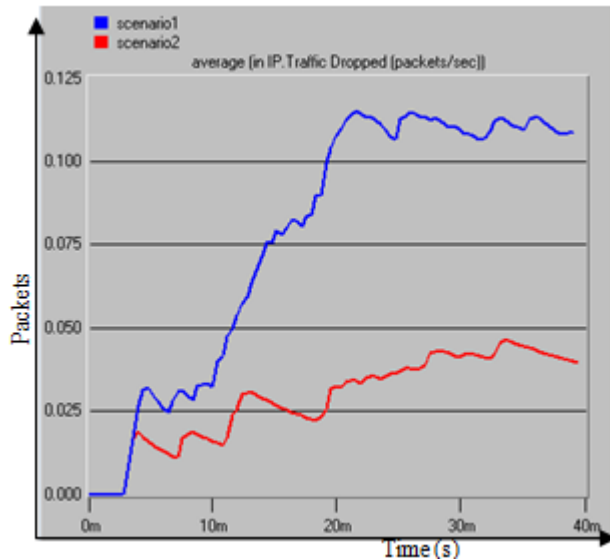
## 5. ANALYSIS OF TCP PERFORMANCE IN A NETWORK ENVIRONMENT OF RELIABLE CONNECTIONS WITH UDP TRAFFIC FLOWS

The results of tests for reliable connections environment workload with User Datagram Protocol (RCEU) traffic flows are analysed as follows. In addition to the model discussed and analysed, two client host each established UDP connection. One of the client hosts is actively involved in video conference transaction and the other involved in VOIP communication. Figure 2 and 3 showed the results of baseline configuration.

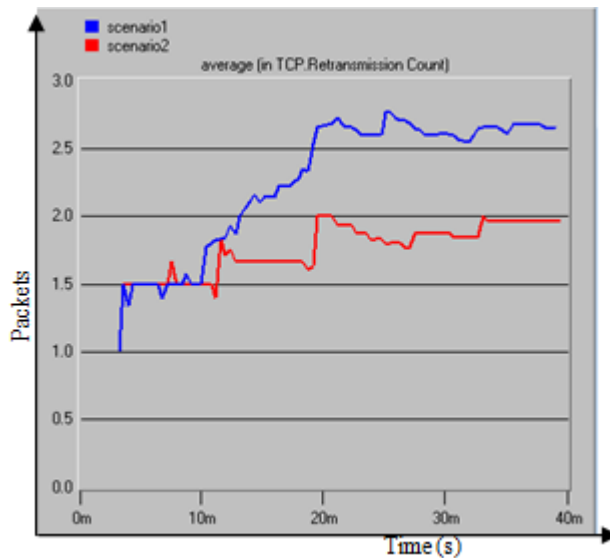
The results pointed out the performance of TCP in a multiple TCP connections with UDP traffic connections involved.

Observing the graph pattern of how the packets are dropped at the router, it is obvious that the presence of UDP traffic interfered with TCP behaviour, inducing jitter and packet loss. The presence of UDP traffic flows in the multiple reliable connections environment caused an average increase packet loss difference of up to 8.56% on the TCP behaviour

and the packet dropped reached up to 11.9%. The results indicated that TCP and UDP contend for both buffers and bandwidth.



**Figure 2: RCEU Dropped Packets**



**Figure 3: RCEU Retransmission Count**

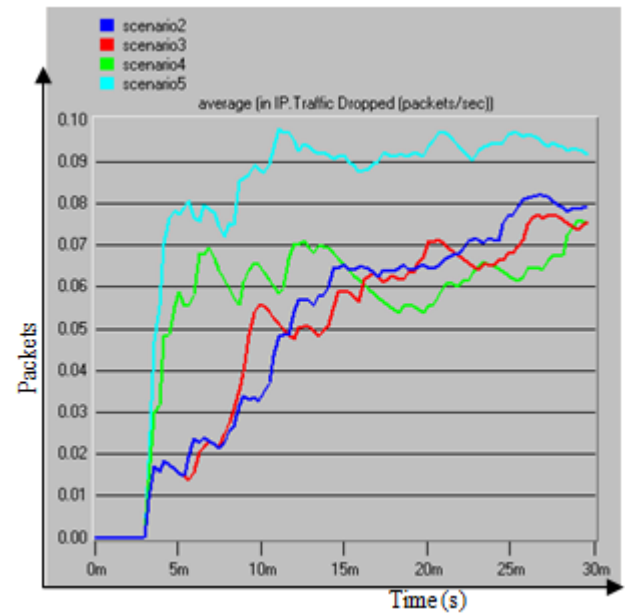
The pattern of the retransmission shows that UDP traffic is favoured and significantly lower TCP performance. This is because in a shared link, TCP suffers from significant loss. The number of packets that are processed by the forwarding processor and the forwarding rate are the parameters that significantly results to the retransmission behaviour.

### 5.1 Effect of Data Rate on TCP Behaviour Using Baseline Configuration (EDTB)

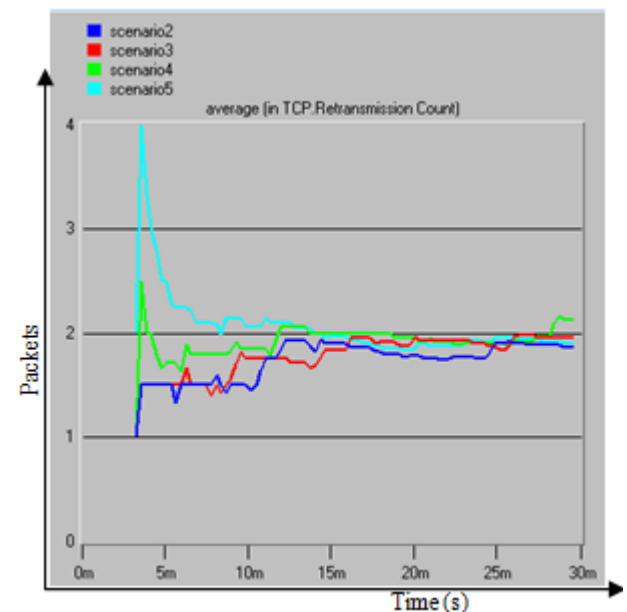
Since link capacity is mentioned among problems causing poor performance of TCP, using the selected data rate in table 1 four simulation scenarios were tested. Each scenario used a different wireless link data rate. In scenario 2, data rate of 64,000bps was configured, 128,000bps configured in scenario 3, 256,000bps configured in scenario 4, and 512,000bps configured in scenario 5. The modelled scenarios used the

same emulated environment over a router buffer size of 16mb. The objective of these simulations is to see the effect of data rate on the performance of TCP over the same router buffer size and find other possible causes of the change in behaviour. The graph pattern of the simulation result in figures 4 and 5 showed that different data rate significantly yield different packet dropped and retransmission behaviour.

With the router buffer size of 16mb, 64,000kps and 128,000kps offered better TCP performance. This is because the pattern of the graph in figure 4 indicated low packet drop for in 3-to-8 minutes of simulation.



**Figure 4: EDTB Dropped Packets**



**Figure 5: EDTB Retransmission Count**

Within the same period of simulation, scenario 4 with data rate of 256,000 experiences significant packets dropped difference of 5.5% compared to scenario 2 and 3. But scenario 5 with data rate of 512,000bps experiences 6.5%. This means that router buffer size or data rate only cannot optimise the performance of TCP. Therefore, in the performance of TCP data rate and router buffer size are strongly related. After 10 minutes of simulation,

TABLE 3  
SIMULATION CONFIGURATION FOR EDTDR

Simulation1: over 16mb		Simulation2: over 32mb		Simulation3: over 64mb	
Scenarios	Data rate	Scenarios	Data rate	Scenarios	Data rate
1	64,000bps	1	64,000bps	1	64,000bps
2	128,000bps	2	128,000bps	2	128,000bps
3	256,000bps	3	256,000bps	3	256,000bps
4	512,000bps	4	512,000bps	4	512,000bps

the graph pattern of packet dropped gradually increased because of the TCP window size adjustment, adapting to available link resources. The graph pattern of retransmission count in figure 5 shows the presence of link noise. However, it is observed there is change in the TCP behaviour when the four different data rates were tested. Hence, data rate significantly has effect on performance of TCP.

### 5.2 Effect Of Data Rate On Tcp Behaviour Using Different Router Buffer Size

The selected data rates and router buffer sizes shown in table 1 were tested the same emulated environment. For each of the router buffer size, the selected data rates were tested to observe among the simulation scenarios which give the best TCP performance. Table 3 summarises the data rate and router buffer size configurations for the simulations.

After running the three simulations, the best scenario from each simulation were selected and simulated again. The best scenarios are scenarios1 from simulation1, scenario2 from simulation2, and scenario3 from simulation3. The best scenarios were tested together to identify the degree of effect on the performance of TCP. Scenario2 with data rate of 128,000bps over 32mb of router buffer size reflected low packets dropped followed by scenario1 with data rate of 64,000 bps over 16mb with slight fluctuating packets dropped difference of between 0.1% to 1.0%.

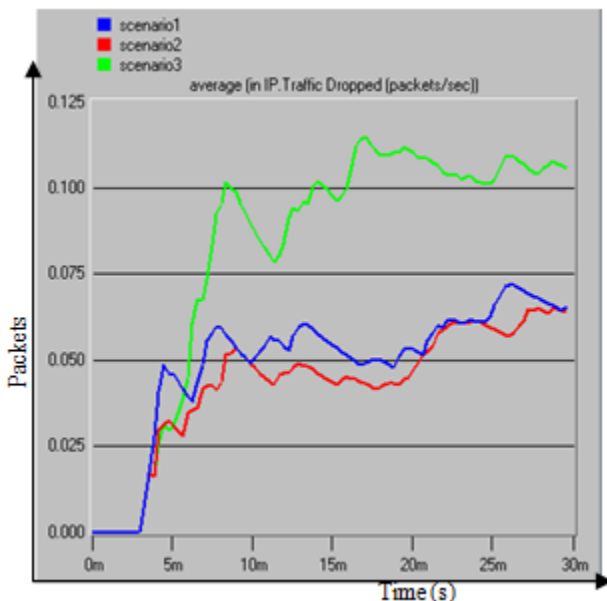


Figure 6: EDTDR Dropped Packets

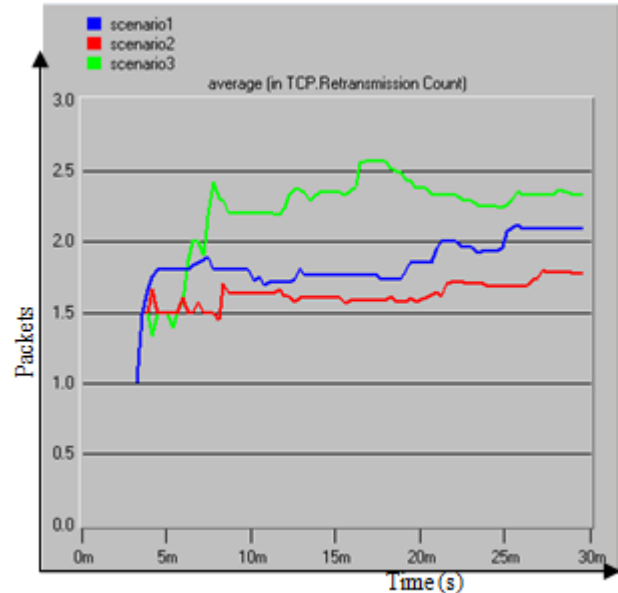


Figure 7: EDTDR Retransmission

The TCP retransmission behaviour appeared almost to have the same degree of behaviour with packets dropped as shown in figure 7. Scenario3 with data rate of 256,000bps over 64mb yielded worse performance.

The simulation for the effect of selected different data rate and router buffer size on the performance of TCP in the presence of UDP traffic reflected similar behaviour. Hence, router buffer size and data rate are strongly correlated in determining requirements for optimum performance of TCP in any kind of network environment.

### 5.3 Summary Of The Various Scenarios

In the various scenarios that determines the performance of TCP in a network environment of multiple TCP connections with UDP traffic flows, it is obvious that the presence of UDP traffic interfered with TCP behaviour, inducing jitter and packet loss. The presence of UDP traffic flows in the multiple reliable connections environment caused an average increase packet loss difference of up to 8.56% on the TCP behaviour and the packet dropped reached up to 11.9%. The results indicated that TCP and UDP contend for both buffers and bandwidth. The pattern of the retransmission shows that UDP traffic is favoured and significantly lower TCP performance.

The performance of TCP behaviour with selected data rate showed that the router buffer size results in several packets dropped. The simulation reflected that Scenario2 with data rate of 128,000bps over 64mb of router buffer size reflected low packets dropped followed by scenario1 with data rate of 64,000

bps over 32mb with slight fluctuating packets dropped difference of between 0.1% to 1.0%.

This increases the rate of retransmission when the data rate is high. A larger router buffer of allows a larger average congestion window, which results in a larger number of unnecessary retransmissions. Using the small router buffer increases the probability of having a retransmitted packet dropped when retransmitting after a retransmission timeout, which results in another retransmission timeout. In the other hand, use of the medium router buffer size does not cause as low TCP performance as use of the small router buffer, because the larger router buffers avoid several packets drop problem which was present when using the small router buffer.

#### 5.4 Summary Of The Finding

Based on the research work using the selected cases, the performance of the TCP has not remained the same rather varied along with changes in the data rate and router buffer size in the emulation environment. From the analysis of results of the simulation, having then known that UDP flows do not have any congestion avoidance mechanisms and they do not slow down when their packets are dropped at the routers, the baseline TCP implementations in the presence of UDP is at a disadvantage in shared link. On the other hand, UDP sent data faster than TCP. In the presence of UDP flows, TCP probes the data path by increasing its congestion window until loss occurs. In the case of the emulated environment, the TCP connections back off and not use more bandwidth than what is left after the UDP has fetch enough bandwidth smooth flow of its traffics. Therefore, multiple reliable connections are left with competition for the left over bandwidth. Hence, the performance of TCP becomes poor.

#### 6. CONCLUSION

This study has attempted to contribute to the body of work on simulation, modeling, analyzing and improving TCP and UDP protocol. This research also help towards modeling a network environment to have an idea of what happen to applications under TCP transport when any user of UDP application is allowed. This study contributes in the development of a network services, and may form a useful simulation for utilization by other researchers in order to assess, study, and test the TCP behavior in presence of UDP. This research also provides guide to TCP and UDP researchers whom thinking about improving TCP or UDP performance.

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