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Optimization of Round-Trip Time (RTT) on New Reno TCP Wireless Network Performance (Case Study: Banda Aceh City Coffee Shop)

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Abstract

The TCP protocol is one of the transport agents on the communication line, with the Quality of Service (QoS) parameter as a fixed indicator of network performance measurement including throughput, time delay, packet loss, and fairness. The use of wireless network systems often occurs disruption. The existence of wifi in a coffee shop in the city of Banda Aceh has become a modern lifestyle as the main facility, the better the internet connection, the more communities and people who come. thus motivating the author to conduct further research on the methods that have been applied before. The research results are expected to contribute to the increase in wireless network throughput with a strategy to reduce the Round-Trip Time (RTT) so that traffic will be better and this model can be applied to coffee shops in Banda Aceh City. The research objective is to simulate and analyze existing algorithms developed previously using the NS-3 application, which affects the measurement of RTT, cwnd, and throughput as well as to evaluate the performance of TCP New Reno based on topology and research parameters so that it is obtained specifically how to optimize RTT and throughput on wireless networks. Banda Aceh City coffee shop. The research method utilizes the literature and simulates the development of the TCP New Reno algorithm on wireless network coding so that the results can be applied and studied directly. The approach taken is to describe the various stages or techniques in increasing throughput, controlling the congestion window (Cwnd), and reducing RTO. The results of the study concluded that the performance of TCP New Reno is better than other TCP models when the Round-Trip Time (RTT) is smaller and the probability of Bursty Loss is lower. Then the simulation based on NS-3 produces an average value of RTO less in the point-to-point multi-hop adhoc topology than single-hop, meaning that the interference on the network that occurs is smaller. Meanwhile, if the point-to-point network is compared to the grid, there will be less disruption to the grid network.

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Keywords:*Wireless, TCP New Reno, RTT, Cwnd, Throughput.*

I. Introduction

The TCP protocol is part of the transport agent on the communication line, has a Quality of Service (QoS) parameter which is a fixed indicator in measuring network performance such as throughput, time delay, packet loss, and fairness. The many variations of New Reno that are implemented have been discussed in [1,3], and the so-called impatience variation is highly recommended [1]. The development of the TCP algorithm affects the QoS indicator [3]. The four bottlenecks in TCP are related to congestion control algorithms such as slow start, congestion avoidance, fast retransmit, and fast recovery [4,5]. It is explicitly possible to modify the algorithm [6], and modification affects "partial acknowledgments (ACKs)" to new data, but not all missing data is detected without SACK [5]. Return timeouts [7] are used as a last resort to recover lost packets (packet loss).

TCP New Reno is a development of TCP Reno, which can only handle one segment of lost data packets so that it can handle more than one packet of lost data packets in one window without decreasing ssthresh many times because it will not leave the fast recovery phase before all packets are in one. window in ACK all [1]. On certain internet connections, the large number of packets is usually controlled by the TCP window size. During the congestion avoidance phase, the congestion window increases by 1 packet in general for each RTT calculation. This means that in each second, the throughput of the node typically can be increased by 1 / RTT packet/second. This means that if the RTT is smaller, there will be a higher throughput increase.

This research is based on research conducted by KaiyuZhoua, Kwan L. Yeung, and Victor O.K. Li uses NS-2 [1], as well as that of other researchers [2-3]. Based on the literature study that has been done, it motivates the author to conduct further research on the methods that have been applied before, including evaluating and the reasonableness of the RTT value on the New Reno TCP algorithm wireless network. In addition, the results of this study can contribute to an increase in wireless network throughput with a strategy of reducing RTT so that traffic will be better, this will also affect throughput and reduce packet loss and time delay. The analysis and evaluation of the reasonableness of the RTT use the Network Simulator (NS) application, a software that is able to display how the communication process takes place in a simulation. NS simulates communication via cable and wireless. NS-3 is a network simulator application that has the ability to model networks that can implement various protocols and topology forms with various scenarios. The C ++ programming language is also supported in NS-3 applications which can facilitate the development of models ranging from topology to nodes to detailed protocol mechanisms [8].

Coffee shops have become a modern lifestyle in various regions in Indonesia (Said, 2017), as well as in the city of Banda Aceh which is known as a million coffee shops, where each coffee shop has free wireless facilities with the aim of getting customer interest. Speed and stable connections are the main choices of visiting customers (Panuju, 2017), especially for young people (Herlyana, 2014). The existence of free wifi in coffee shops gives a new color to the community and communities around the city of Banda Aceh (Hayati, 2019). With the public's need for behavior using the internet in the City of Banda Aceh is of particular interest in conducting this research, in addition, the results of this study can contribute to increasing wireless network throughput with a strategy of reducing Round Trip Time (RTT) so that traffic will be better, this is also will affect throughput and reduce packet loss and time delay so that the resulting network model can be applied to coffee shops in the city of Banda Aceh.

II. RESEARCH METHODS

The research subject focused on developing a wireless network coding algorithm with the goal of achieving a better optimization of QoS indicators, based on this the object or target that determines the optimization of RTT values in a wireless network architecture. Figures 1 and 2 show the steps taken in the study, related to the basic research data, the network model under study, the sequence and method of testing, and the comparison of the results achieved in the study, especially the use of TCP New Reno. The flow of research activities can be explained as follows:

a. Simulate existing algorithms developed

July-August 2020
ISSN: 0193-4120 Page No. 4465 - 4476

previously using the NS-3 application, which affects the measurement of RTT, cwnd, and throughput. Achievements are made through analysis and evaluation of 3-stage simulations, including:

- Time Out Simulation (ERTT, RTO, and SRTT); to obtain and evaluate the optimization value of the RTO.
- cwnd simulation; to obtain and evaluate the value of congestion (congesti), congestion avoidance, Slow Start Threshold (ssthresh), and packet loss.
- Simulation of throughput; to obtain and evaluate the quantity value of data transferred successfully (real bandwidth condition).

b. Analyze and evaluate the performance of TCP New Reno based on topology and research parameters in order to obtain how the RTT optimization affects RTT, cwnd, and throughput on wireless networks.

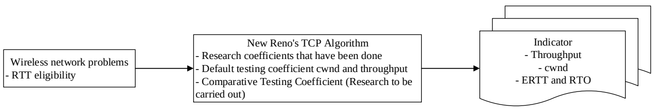


Figure 1. Problems and research objectives

In general smaller, geographically localized networks tend to use broadcast networks, while larger networks generally use point-to-point and grids. Based on this, the study used 2 topologies, Adhoc Point-to-Point (P2P) and Grid. These two topologies will be analyzed and evaluated based on the parameters used in order to obtain a better topology design for use in wireless networks in relation to the feasibility of the RTT value.

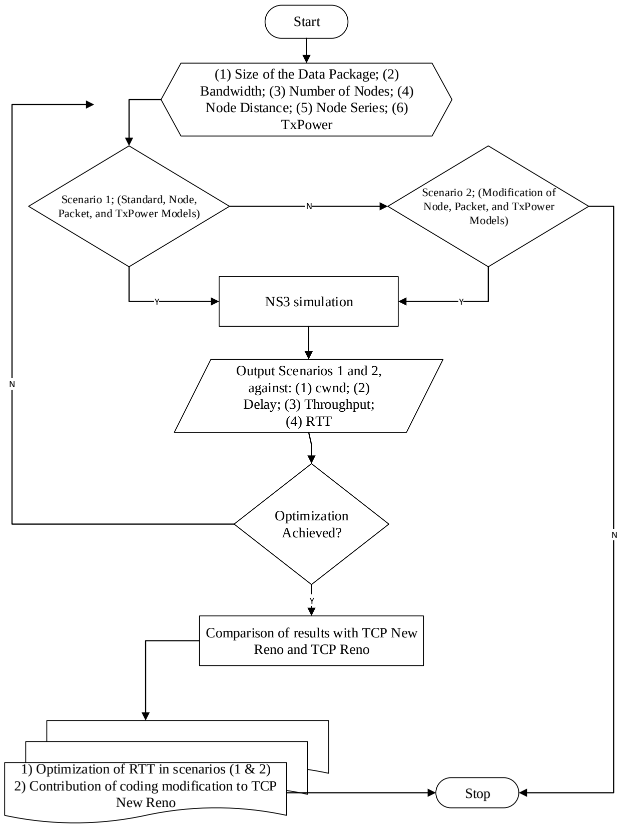


Figure 2. Research Evidence Procedure

2.1. Point-to-Point Adhoc Topology

a. Single Hop

- *Single Flow*

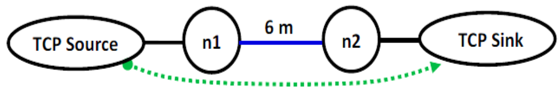


Figure 3. Ad Hoc P2P Single Hop Single Flow Topology

- *Multi Flow*

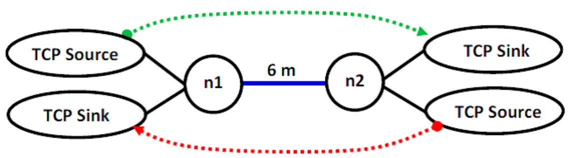


Figure 4. Adhoc P2P Single Hop Multi Flow Topology

b. Multi Hop

- *Single Flow*

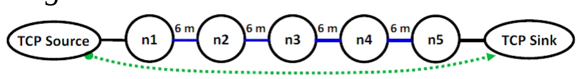


Figure 5. Ad Hoc P2P Single Hop Single Flow Topology

July-August 2020
ISSN: 0193-4120 Page No. 4465 - 4476

- *Multi Flow*

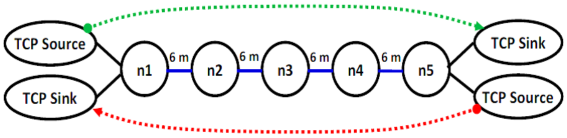


Figure 6. Ad Hoc P2P Single Hop Single Flow Topology

2.2. Adhoc Grid Topology

a. Single Flow

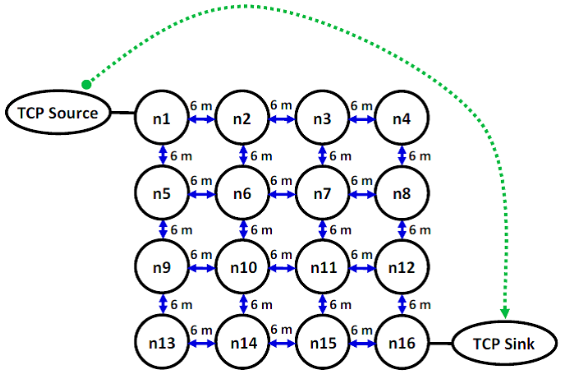


Figure 7. Adhoc Grid Single Flow Topology

b. Multi Flow

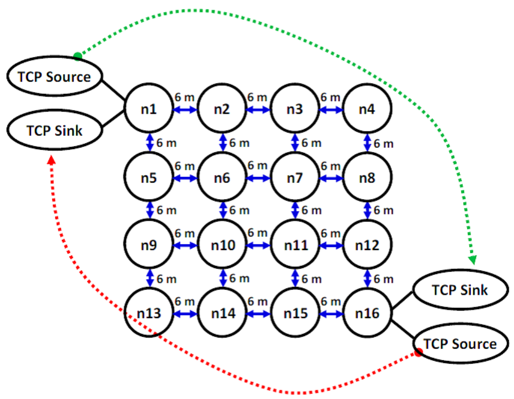
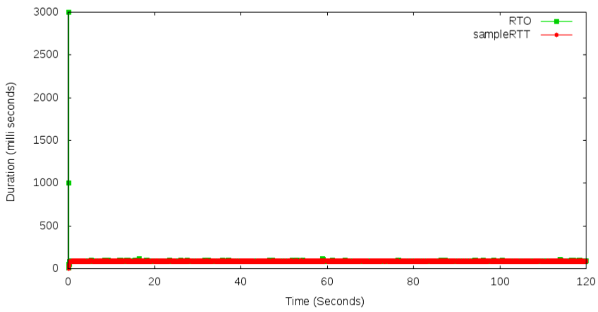


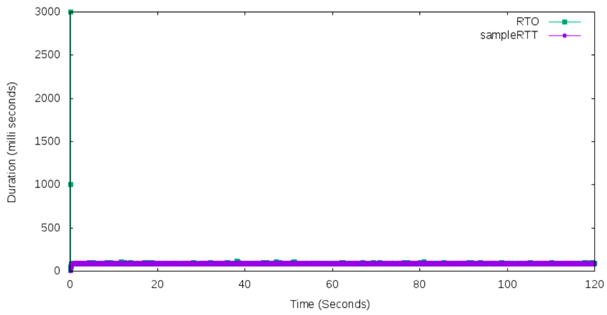
Figure 8. Multi Flow Grid Adhoc Topology

III. DISCUSSION

3.1. *Round Trip Time (RTT) and Retransmission Time Out (RTO).*

3.1.1. Single Flow.





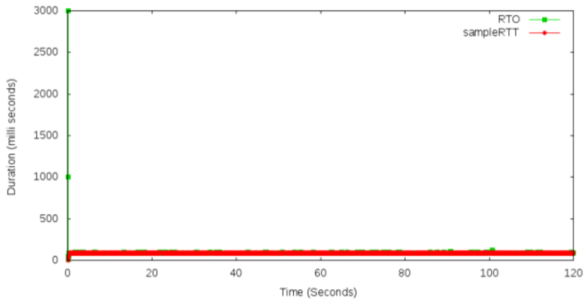
(a) **Distance of 6 meters**
(b) **Distance of 3 meters**

Figure 9. SRTT, ERTT and RTO Single Hop P2P Adhoc Network.

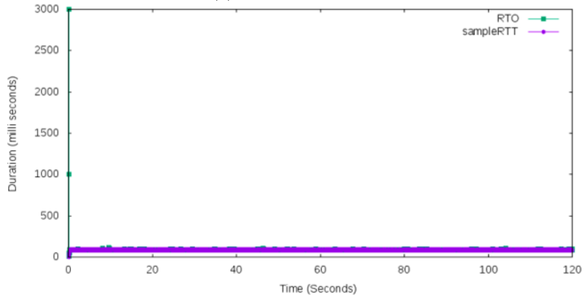
The simulation results in Figure 9 (a and b) are a scenario on a single-hop network that has 2 nodes with different distances (6 m and 2 m) showing the SampleRTT value (the value of the amount of time between when the segment is sent and the ACK for the segment received without retransmitting) changes slightly and updates continue at the beginning of time until a certain time the SampleRTT value becomes stable and is maintained by TCP that value in the EstimateRTT calculation (the average value of SampleRTT). On the other hand, the RTO value requires a greater duration than SampleRTT, the retransmit process time is faster at the beginning with the large RTO value. The duration will be greater when the length of time, in other words, the occurrence of interference on the TCP network.

Packet drop is marked by an increase (the higher) the value of the RTO so that the network is interrupted (link error), then the ACK is sent beyond the time limit when the RTO occurs frequently (high value), meaning that when the connection breaks, TCP performs RTO. The packet sending failed marked with ACK and the transmitted data stream did not reach the recipient and affected the RTT, meaning that the RTT calculation affected the RTO value.

July-August 2020
ISSN: 0193-4120 Page No. 4465 - 4476



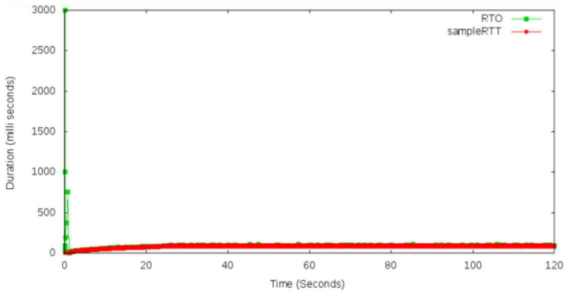
(a) **Distance of 6 meters**



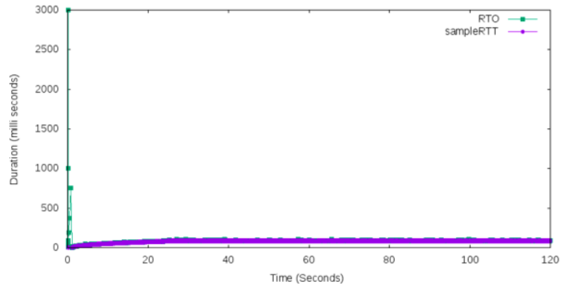
(b) **Distance of 3 meters**

Figure 10. STT, ET and RTO Multi Hop P2P Ad Hoc Network.

The simulation results in Figure 10 (a and b) are a scenario on a 5 node multi-hop network with different distances (6 m and 2 m), not much different from Figure 4.1 in that the SampleRTT value also changes slightly (fluctuates) and at The initial time continues to update, until a certain time the SampleRTT value becomes stable and TCP maintains that value in the EstimateRTT calculation (the average value of SampleRTT). Meanwhile, the RTO process requires a smaller duration than Figure 9, meaning that the retransmit process is faster and there are fewer disturbances. On the other hand, the RTO value requires a smaller duration than SampleRTT and the retransmit process time is faster than Figure 9 so that the disturbance occurs less.



(a) **Distance of 6 meters**



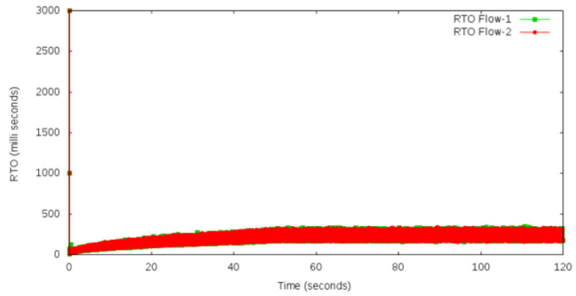
(b) **Distance of 3 meters**

Figure 11. Along with, ARTI and RTO Multi Hop Grid Ad Hoc Network.

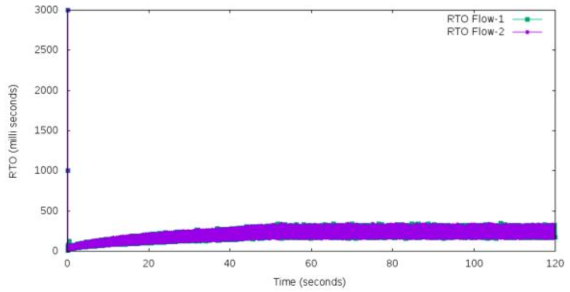
The simulation results in Figure 11 (a and b) are a scenario on a grid network (4x4) of 16 nodes with different distances (6 m and 2 m), very different from Figures 9 and 10, the duration for the SampleRTT value is faster and more stable at the specified time including the calculation of EstimateRTT starting at 24 seconds. Between the value of RTO and SampleRTT are similarly proportional, which indicates that RTO has less effect than single hop (Figure 9) and multi-hop (Figure 10). Except at the beginning of the packet/data transmission, the duration of the RTO value is too large, up to 187-750 milliseconds at 0.29-0.86 seconds, meaning that significant disruption to the network occurs at the beginning of time.

3.1.2. *Multi Flow*

Then in multi flow, the analysis is carried out based on the simulation results between flow-1 and flow-2 related to RTO. Unlike single flow, multi flow compares between 2 (two) the same indicators, namely RTO.



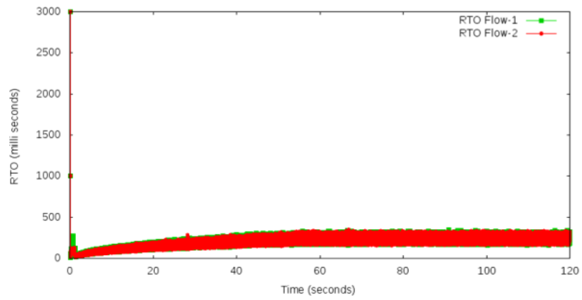
(a) **Distance of 6 meters**



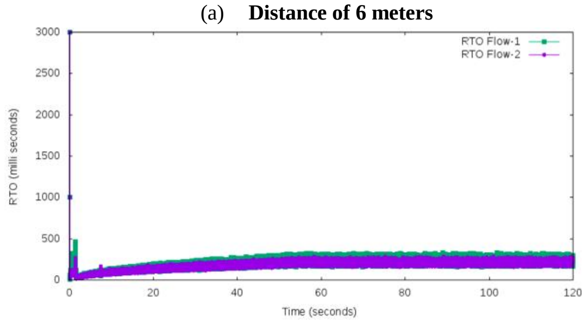
(b) **Distance of 3 meters**

Figure 12. RTO of Ad Hoc P2P Single Hop Multi Flow Network.

The simulation results in Figure 12 are a single hop 2 node network scenario in multi flow showing the duration of the RTO flow-2 value is smaller than flow-1. This indicates that at the beginning of the RTO flow-2, the retransmit process time is faster than flow-1. The duration will be greater for both flows (1 and 2) when the length of time required for packet delivery means that there is significant disruption to the TCP network.



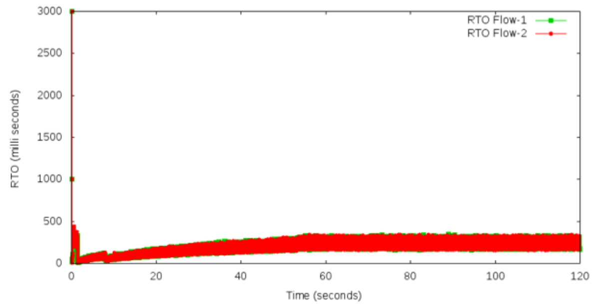
(a) **Distance of 6 meters**



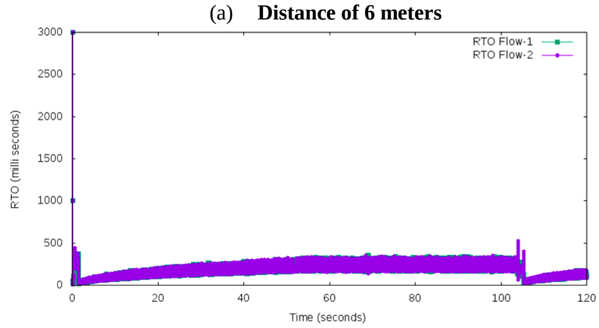
(b) **Distance of 3 meters**

Figure 13. RTO of Multi Hop Multi Flow Adhoc P2P Network.

The simulation results in Figure 13 are a multi-hop 5 node network scenario on multi-flow which shows that it is not much different from Figure 12, namely the duration of the RTO flow-2 value is smaller than flow-1. This indicates that at the beginning of the RTO flow-2, the retransmit process time is faster than flow-1. The duration will be greater in both flows (1 and 2) when the length of time required for packet delivery means that there is a significant disruption to the TCP network. The difference in the result is that there is greater interference in multi-hop than a single hop for both RTOs (flow-1 and flow-2) due to the higher RTO value.



(a) **Distance of 6 meters**



(b) **Distance of 3 meters**

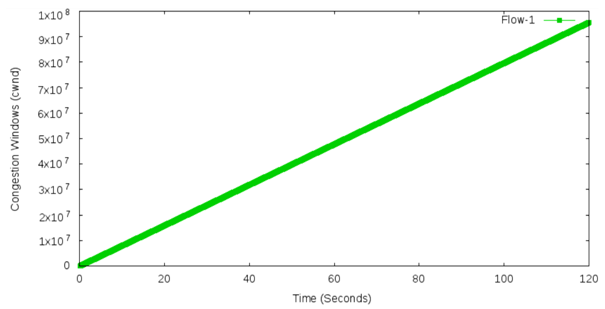
Figure 14. RTO of Ad Hoc Grid 4x4 Multi Flow Network.

Figure 14 differs from the simulation results with Figures 12 and 13, the 4x4 grid network scenario in multi-flow with 16 nodes shows that the duration of flow-1 RTO value is smaller than flow-2. This is because the amount of RTO flow-1 at the beginning indicates that the retransmit process time is faster than flow-2. However, in contrast to point-to-point (2 nodes and 5 nodes), the longer the grid takes, the smaller the duration for both flows (1 and 2), indicating a significant reduction in disruption to the TCP network. This means that when compared to Figure 12 with Figures 13 and 14, both RTOs (flow-1 and flow-2) in multi-hop have less interference (smaller RTO values).

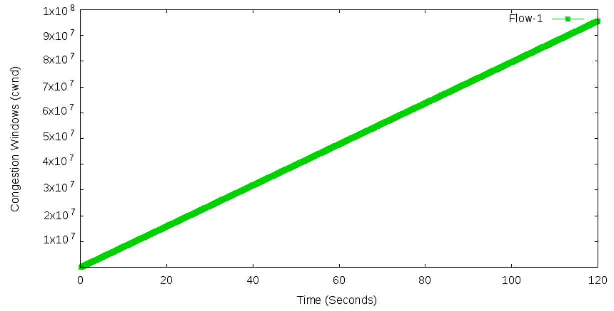
3.2. *Congestion Windows (cwnd)*

3.2.1. Single Flow

July-August 2020
ISSN: 0193-4120 Page No. 4465 - 4476

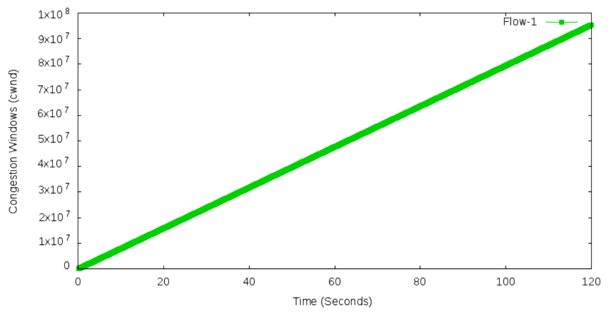


(a) **Distance of 6 meters**

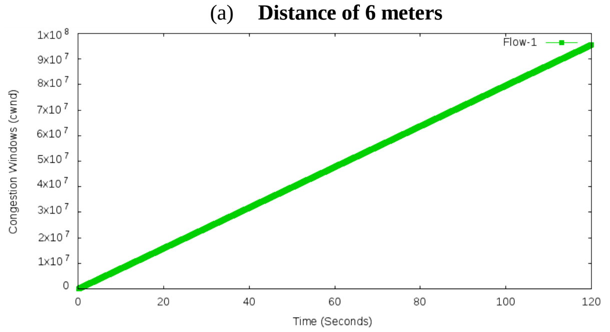


(b) **Distance of 3 meters**

Gambar 15.1.*Point-to-Point Single Hop.*

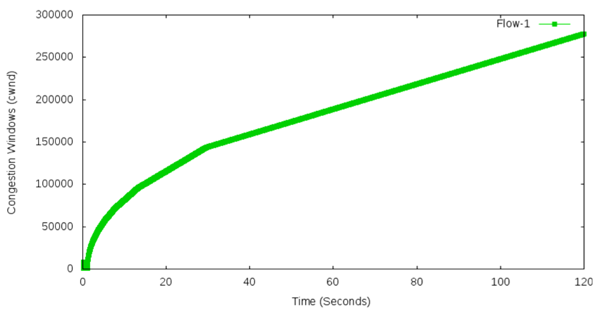


(a) **Distance of 6 meters**

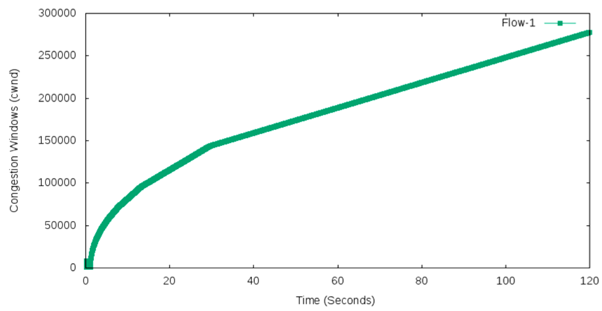


(b) **Distance of 3 meters**

Figure 15.2. Point-to-Point Multi Hop.



(a) **Distance of 6 meters**



(b) **Distance of 3 meters**

Figure 15.3. 4x4 Multi Hop Grid.

Figure 15. Cwnd on the Single Flow Adhoc Network.

Figures 15 (15.1, 15.2, and 15.3) are simulation results that show the initiation of RTT, sending the cwnd bytes of data to the connection allowing problems to occur. At the end of the RTT, the ACK for data is received by the sender. In other words, the estimated rate of sending by the sender is cwnd / RTT in bytes/second. Sending a number of data is controlled by the receiver (flow control) based on the level of congestion (congestion) in a single flow network.

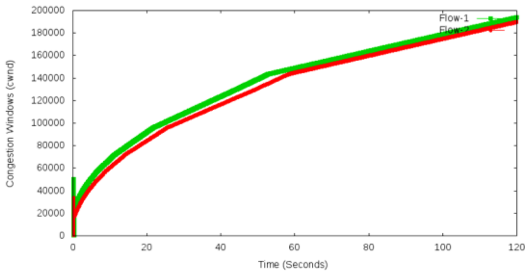
Figure 15.1 has a different flow control (flow-1 and flow-2), the value of cwnd flow-2 does not change, while flow-1 continues to increase compared to flow-2 and has no effect on cwnd and time. The size of the cwnd affects the delivery rate, causing loss events, including the relatively slow arrival time of the ACK (for example, at the end of the line the amount of bandwidth is reduced). Although TCP does not experience a retransmit phase at flow-1, it shows poor control techniques, meaning that it indicates packet loss or ACK restrictions received.

The increase in the Cwnd value is smaller in Figure 15.2 than in 15.1, but the working principle of the difference in flow control (flow-1 and flow-2) is not much different. The cwnd flow-2 value has no effect on cwnd and time, while the cwnd flow-1 value continues to increase. Same as Figure 15.1, although TCP does not experience a retransmit phase at flow-1, it shows that the control technique is not good, meaning that it indicates packet loss / ACK restrictions received.

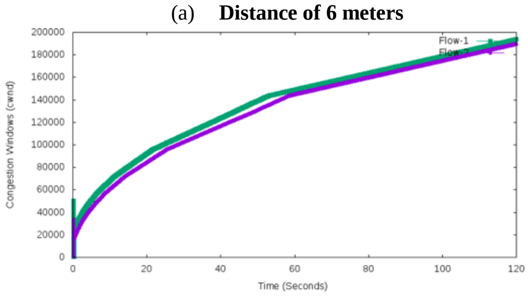
July-August 2020
ISSN: 0193-4120 Page No. 4465 - 4476

Figure 15.3 differences in flow control (flow-1 and flow-2), TCP experiences a retransmit phase at flow-1 in data transmission, which is seen in flow-1 (cwnd = 0.123125 bytes; time = 8040 seconds), it indicates packet loss or the accepted ACK limits. TCP enters the retransmit phase at flow-1 (cwnd = 0.134845 bytes; time = 536 seconds), meaning that the slow start phase starts again where each cwnd size will increase twice every RTT for each ACK received by the sender, effectively doubling (slowly data added) windows size RTT (time). This means that flow-1 in Figure 15.3 is better than 15.1 and 15.2, because it shows a good control technique on flow-1 which reduces packet loss or limits the ACK received (smaller cwnd value).

4.2.2. Multi Flow

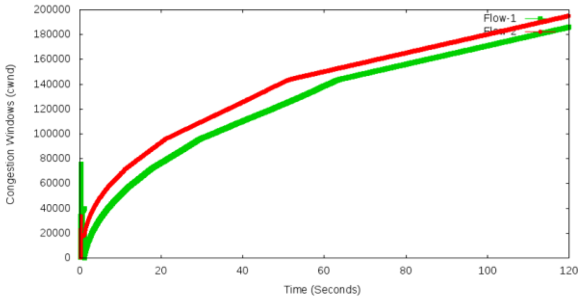


(a) **Distance of 6 meters**

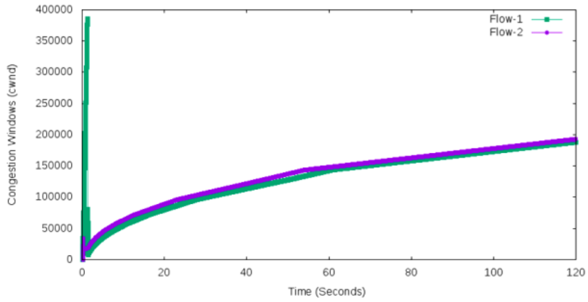


(b) **Distance of 3 meters**

Figure 16.1. Point-to-Point Single Hop.

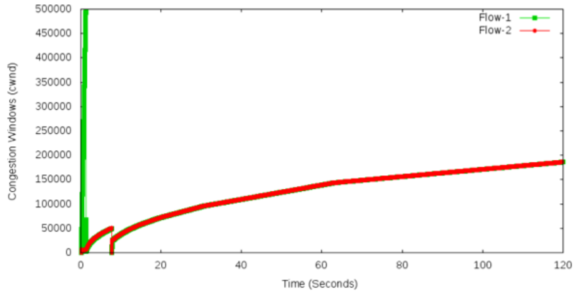


(a) **Distance of 6 meters**

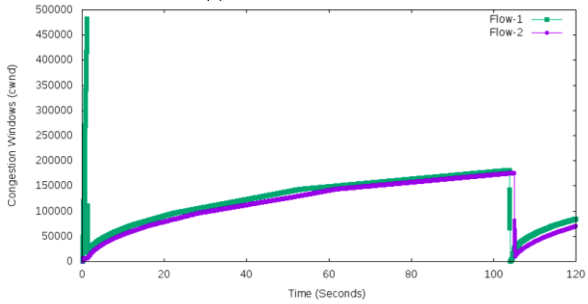


(b) **Distance of 3 meters**

Figure 16.2. Point-to-Point Multi-Hop.



(a) **Distance of 6 meters**



(b) **Distance of 3 meters**

Figure 16.3. 4x4 Multi Hop Grid.

Figure 16. Cwnd on the Multi Flow Adhoc Network.

For each RTT start in Figure 16 (16.1, 16.2, and 16.3), the sending of the cwnd byte of data to the connection may be hampered. At the end of the RTT, the sender receives an ACK for the data. This means that the estimated sending rate is cwnd / RTT in bytes/second. The amount of data transmission is controlled by the receiver (flow control) based on the level of congestion on the multi-flow network.

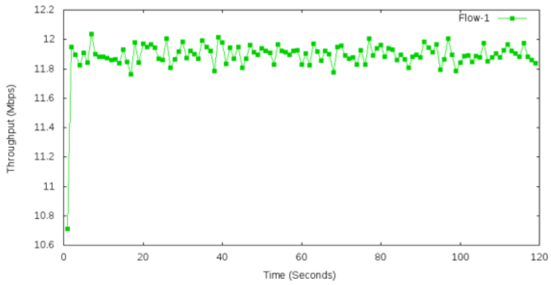
In Figure 16.1, there is a difference in flow control (flow-1 and flow-2), the cwnd value of flow-2 is lower than flow-1 even though the increase in cwnd is almost the same between flow-1 and flow-2. Flow-2 shows a better control technique in ensuring the sending entity to the receiving entity (no data accumulation occurs). The amount of data transfer length (buffer) is specifically allocated by the receiving entity, at the same time as the data transmission (time = 0 seconds) the slow start phase occurs. Then it can be seen that flow-1 and flow-2 indicate packet loss or the ACK limit received. The retransmit phase experienced by TCP in flow-1 and flow-2, namely the start of the slow start phase, each cwnd size increases twice for each RTT for each ACK received by the sender, cwnd will effectively double (slowly data is added) the RTT windows size (time).

The difference in flow control (flow-1 and flow-2) in Figure 16.2 is inversely proportional to Figure 16.1, the cwnd value of flow-1 is lower than flow-2. meanwhile, a significant difference between flow-1 and flow-2 was seen when the Cwnd increased slowly. Flow-1 shows a better control technique than Flow-2 in guaranteeing the sending entity to the receiving entity. At the beginning of data transmission, the slow start phase occurred and it was also seen that flow-1 and flow-2 indicated packet loss or ACK limits were received. The retransmit phase experienced by TCP at flow-1 and flow-2, which indicates that the slow start phase begins again, cwnd will effectively double (slowly add data) the size of the windows RTT (time).

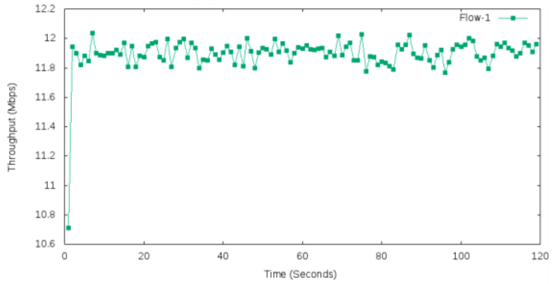
Figure 16.3 is inversely proportional to Figure 16.2 due to the difference in flow control (flow-1 and flow-2), the cwnd value of flow-2 is lower than flow-1 and there is also a difference between flow-1 and flow-2 when the cwnd increases slowly. Flow-2 shows a better control technique than Flow-1 in guaranteeing the sending entity to the receiving entity. At the beginning of data transmission, the slow start phase occurred and it was also seen that flow-1 and flow-2 indicated packet loss or ACK limits were received. The retransmit phase is experienced by TCP on flow-1 and flow-2 which indicates that the slow start phase starts again, cwnd will effectively double (slowly the data is added) the size of the window RTT (time).

3.3. ***Throughput***

3.3.1. Single Flow

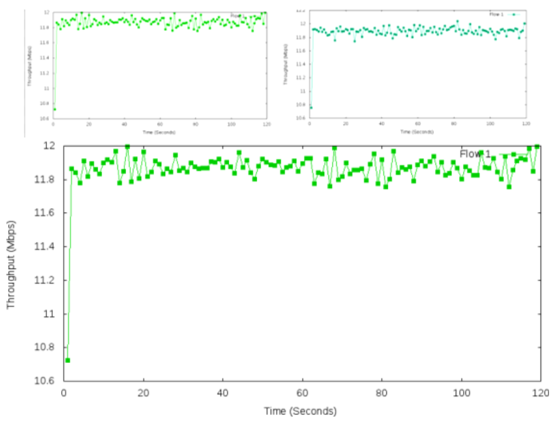


(a) Distance of 6 meters

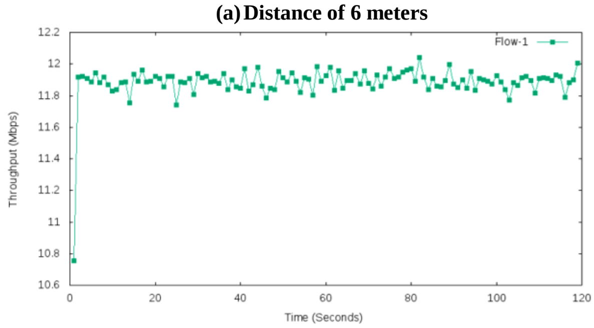


(b) Distance of 3 meters

Figure 17.1. Point-to-Point Single Hop.

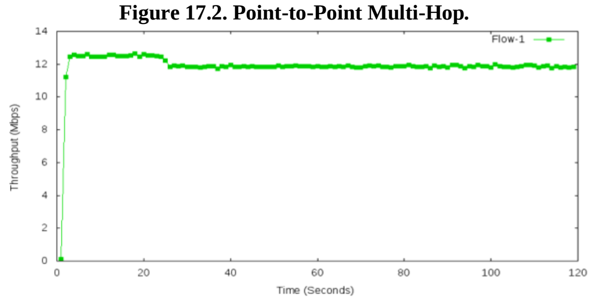


(a) Distance of 6 meters



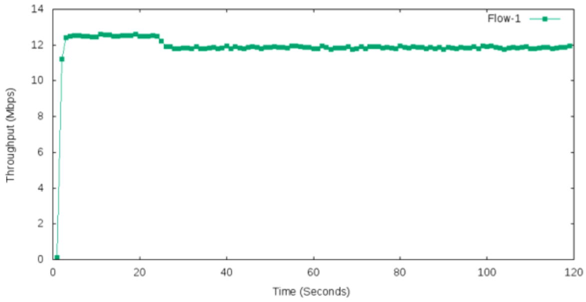
(b) Distance of 3 meters

Figure 17.2. Point-to-Point Multi-Hop.



(a) Distance of 6 meters

July-August 2020
ISSN: 0193-4120 Page No. 4465 - 4476



(b) Distance of 3 meters
Figure 17.3. 4x4 Multi Hop Grid.
Figure 17. Throughput on a Single Flow Adhoc Network.

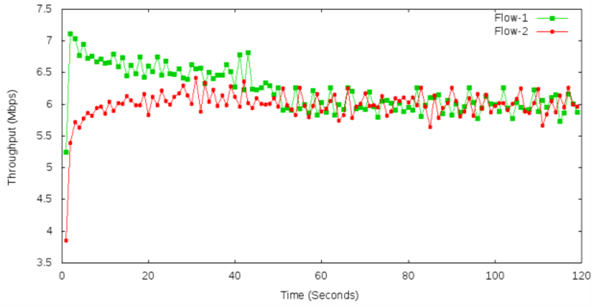
Figure 17.1 on a single hop (2 node) ad-hoc network only at the beginning of the increase in throughput, the rest until the end of the data packet transmission time, throughput instability occurs.

Figure 17.2 on a multi-hop adhoc network (5 nodes) is also only at the beginning of the increase in throughput, if compared to single-hop (Figure 17.1) there is less increase, the rest until the end of the data transmission time is less significant than single hop due to throughput instability.

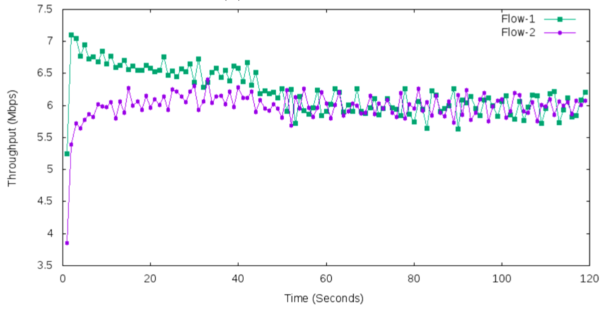
On the other hand, Figure 17.3 of the adhoc grid network (16 nodes) is inversely proportional to Figures 17.1 (single hop) and 17.2 (multi hop), lacks obstacles, and is not always moving (static) so that data transfer is much smoother. Noise is less on the network and throughput does not fluctuate frequently, so data transmission is smooth or uninterrupted. In contrast to Figures 17.1 (single hop) and 17.2 (multi hop), noise is greater on the network and throughput often fluctuates, until data transmission is interrupted or experiencing disruption (up and down window size).

3.3.2. Multi Flow

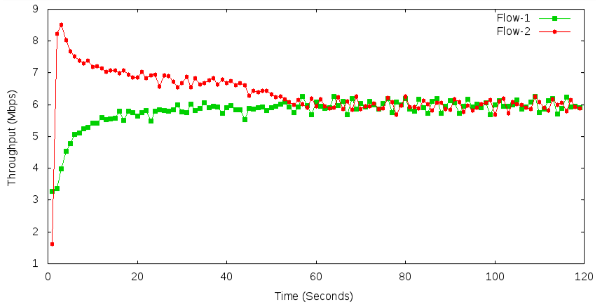
While Figure 18 (18.1, 18.2 and 18.3) describes the comparison between 2 flows (flow-1 and flow-2) or also called multi-flow about increasing throughput linearly with the load offered, throughput decreases because the load offered reaches a certain maximum point. The better the throughput, the better the sending and receiving of data packets.



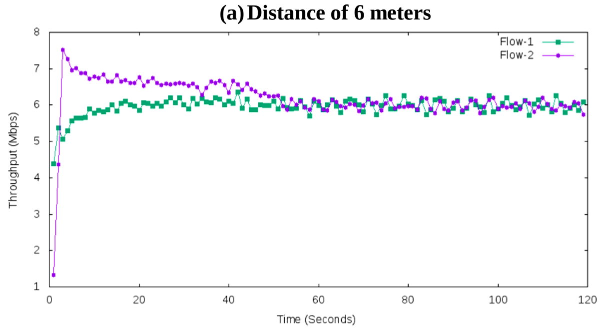
(a) Distance of 6 meters



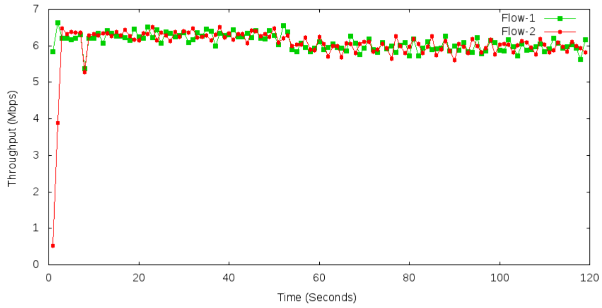
(b) Distance of 3 meters
Figure 18.1. Point-to-Point Single Hop.



(a) Distance of 6 meters

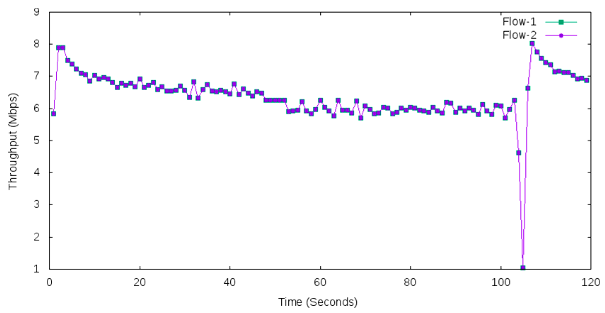


(b) Distance of 3 meters
Figure 18.2. Point-to-Point Multi-Hop.



(a) Distance of 6 meters

July-August 2020
ISSN: 0193-4120 Page No. 4465 - 4476



(b) Distance of 3 meters

Figure 18.3. 4x4 Multi Hop Grid.

Figure 18. Throughput on a Multi Flow Adhoc Network.

Figure 18.1 only at the beginning of the adhoc single-hop network (2 nodes) there was an increase in throughput (flow-1 and flow-2), the rest until the end of the data packet transmission time throughput instability occurred. The two figures show the comparison of flow-2 throughput values is smaller than flow-1.

Figure 18.2, only at the beginning of the multi-hop adhoc network (5 nodes) there was an increase in throughput (flow-1 and flow-2), the rest until the end of the data packet transmission time throughput instability occurred. The two figures show the comparison of flow-1 throughput values is smaller than flow-2.

Figure 18.3 illustrates an adhoc grid network (16 nodes) where data transfers are much smoother due to the lack of barriers (when compared to single-hop and multi-hop). Noise is less on the network and throughput does not fluctuate frequently, so data transmission is smooth or uninterrupted. The two figures show the comparison of flow-2 throughput values is smaller than flow-1. In contrast to figures 18.1 (single hop) and 18.2 (multi hop), noise is greater on the network and throughput often fluctuates, until data transmission is interrupted or experiencing disruption (up and down window size).

IV. CONCLUSION

The results evidenced from the research are:

a. The use of the TCP New Reno algorithm on a wireless network on the Adhoc Grid topology (16 nodes) produces a better value than Point-to-Point (single-hop and multi-hop). The TCP New Reno algorithm has a retransmit phase (at flow-1 and flow-2) which indicates that the slow start phase again starts, cwnd will effectively double (slowly add data) the size of the window RTT (time) thereby reducing packet loss of data.

b. Based on the parameters of Estimate Round Trip Time (ERTT) and Retransmission Time Out (RTO), Congestion Windows (cwnd), and Throughput in general, it shows that the Grid topology (16 nodes) is a better control technique than Point-to-Point (single-hop 2 nodes and multi-hop 5 nodes).

c. Based on the Throughput parameter on the Grid topology (16 nodes) single flow, it shows that the network control technique is better than Point-to-Point (single-hop 2 nodes and multi-hop 5 nodes). This is evidenced by less noise on the TCP network and throughput not often fluctuating, so data transmission is smooth or uninterrupted and throughput is greater.

d. Based on the Throughput parameter on the Grid (16 nodes) multi-flow topology, it shows that flow-1 is a better network control technique because the noise is less on the TCP network and throughput does not often fluctuate, so data transmission is smooth or uninterrupted and the throughput value greater than.

V. ACKNOWLEDGMENTS

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