An Investigation of Improving the Traditional TCP over MANETs

by

Ping-Huan Ho

A thesis submitted to

Auckland University of Technology

in partial fulfillment of the requirements for the degree

of

Master of Computer and Information Sciences

2011

School of Computing and Mathematical Sciences

Primary Supervisor: Associate Professor Nurul Sarkar

Content

Attestation of Authorship

I hereby declare that this submission is my own work and that, to the best of my knowledge and belief, it contains no material previously published or written by another person nor material which to a substantial extent has been accepted for the qualification of any other degree or diploma of a University or other institution of higher learning, except where due acknowledgement is made in the acknowledgements.

Signature: ______________________________

Date: ___________22/10/2012______________

Ping Huan Ho

Acknowledgements

I would like to thank the people who have supported me during this process. I would like to thank my supervisor Associate Professor Nurul Sarkar for the kind guidance, encouragement, comments. I am also grateful to my classmate Kang Wook Byeon. His suggestions and discussion led to a viable contribution to the simulation experiments. Thank you for your kindness support.

Abstract

Transmission control protocol (TCP) is not originally design for use in the mobile ad hoc network (MANET) environments, and therefore it raises a serious performance issues. The transport layer issues such as link failure, network congestion and wireless channel error degrade TCP performance. A research on developing a method of improving TCP performance in MANETs is required to assist an efficient design and deployment of such systems. TCP-WELCOME (Wireless Environment, Link losses, and Congestion packet loss ModEls) is found be a better TCP variant suitable for MANETs. However, TCP-WELCOME has weaknesses especially it deployed the original congestion control mechanism of TCP New Reno to handle packet losses due to network congestion. The proposed TCP offers the most adequate recovery strategy corresponds to each identified packet loss by combining features of both TCP-WELCOME and TCP-AW (Adaptive Westwood), which improves the TCP performance especially when recovering from network congestion related packet losses.

This research aims to improve the performance of TCP by redesigning TCP-WELCOME's loss recovery algorithm and the resulting improvement is called 'Enhanced TCP-WELCOME'. The performance of the proposed TCP is evaluated by extensive simulations using OPNET Modeler. This thesis investigates the effect of varying network sizes, node mobility, traffic loads and wireless channel conditions on the system performance. Empirical results obtained have shown that the proposed Enhanced TCP-WELCOME can offer higher throughputs and packet delivery ratio, and lower end-to-end delay and retransmission attempt than the existing TCP-WELCOME for medium to large size networks. An exhaustive comparative analysis of the proposed method and TCP-WELCOME is also presented in this thesis.

List of Abbreviations and Acronyms

ACKs	Acknowledgment packets
CWND	Congestion Window
Enhanced TCP-WELCOME	Enhanced TCP- Wireless Environment, Link losses, and Congestion packet loss ModEls
ERE	Eligible Rate Estimation
FTP	File Transfer Protocol
IP	Internet Protocol
MAC	Medium Access Control
MANET	Mobile Ad Hoc Network
MSS	Maximum Segment Size
N	Number of Node
NAV	Network Allocation Vector
NS	Node Speed
PL	Packet Length
QoS	Quality of Service
RED	Random Early Detection
RTO	Retransmission Timeout
RTT	Round Time Trip
TCP	Transmission Control Protocol
TCP-ABSE	TCP-Adaptive Bandwidth Share Estimation
TCP-AW	TCP- Adaptive Westwood
TCP-WELCOME	TCP- Wireless Environment, Link losses, and Congestion packet loss ModEls
ToS	Type of Service

List of Figures

List of Tables

Chapter 1

Introduction

MANET connects a set of mobile nodes by using wireless links. These nodes do not require a central facilitated infrastructure to communicate with each other. The nodes do not stay stationary on MANET environment hence the network topology would be changing frequently due to node mobility. TCP provides the reliable data transmission mechanisms and compatibility with most other protocols and applications. It is the most commonly used transport layer protocol in wired networks today. However, the standard TCP does not perform well on MANET environments because the behaviour of data packet loss in wireless environments is different from wired networks [1]. Contrary to wired network, TCP faces several challenges due to the nature of the wireless environments including, intermittent disconnection, high error rates, lower throughput and prone to packet losses. These factors affect the wireless channels lead to different levels of channel errors. The frequent change of network topology increases the complexity of routing and connection management [2], and hence would cause network partitioning and link failures and then degrades TCP connection performance. Thus, TCP does not have the ability to distinguish whether the packet loss is due to channel errors, link failure, or network congestion. A better TCP for MANETs is required to assist an efficient design and deployment of such systems.

TCP-WELCOME [3] was developed to recognize efficiently the common causes of data packet loss (i.e. channel errors, link failure, or network congestion) and take the appropriate action for recovery by using Loss Differentiation Algorithm and Loss Recovery Algorithm. Loss differentiation algorithm is designed to identify the cause of data packet losses for TCP within a MANET, and loss recovery algorithm takes the most suitable recovery mechanism according to the identified packet loss model. TCP-WELCOME is able to improve energy consumption and TCP average throughput significantly; it also outperforms other TCP variants in the most cases. However, TCP-WELCOME used the normal congestion control mechanism of TCP New Reno to handle network congestion related packet losses. The network congestion recovery algorithm needs to be developed in order to enhance the performance of TCP-WELCOME. To improve the loss recovery algorithm of TCP-WELCOME, TCP-AW is adopted in this project [4]. It is a rate and delay based congestion control algorithm that improves network utilization substantially and co-exist with TCP-New Reno without significant harm to the latter.

1.1 Objectives of this thesis

The objective of this research was to investigate a method of improving the traditional TCP so that it can be efficiently used in wireless environments such as in MANETs. An improved TCP, named Enhanced TCP-WELCOME has been proposed in this thesis. The performance of the proposed TCP has been evaluated by simulation experiments. Comparative evaluation of this protocol with the original TCP-WELCOME has also been included.

In this thesis the following research question was addressed:

How to improve the performance of TCP-WELCOME?

To approach this research question, this thesis designed and evaluated a new class of TCP called Enhanced TCP-WELCOME developed as modification of the original TCP-WELCOME. The proposed TCP offers combined features of both TCP-WELCOME and TCP-AW. The loss recovery algorithm of TCP-WELCOME is redesigned so that TCP can use the most adequate recovery strategy corresponds to each classified data packet loss and achieves the better TCP performance especially when handling network congestion related packet losses. In particular, the factors that may influence the TCP performance in MANET environment is considered in the experiment designs and investigation of this study, including network size, traffic load, node mobility and wireless channel conditions.

1.2 Thesis Structure

This thesis is organized as follows: The related literature and work are reviewed in Chapter 2. Chapter 3 presents the research methodology which is used to conduct the performance results in this study. Chapter 4 proposes an improved TCP variant called Enhanced TCP-WELCOME. The proposed method is also described in this chapter. Chapter 5 presented the simulation modeling and configuration of the proposed system. Chapter 6 presents simulation results and analysis. Finally, conclusion and future research directions are discussed in Chapter 7.

Chapter 2

Literature Review

2.1 Introduction

The background material relevant to MANET and TCP is provided in this chapter. MANET characteristics are discusses in section 2.2. In section 2.3, TCP performance degradation issues over MANET are discussed. Section 2.4 discusses some contemporary TCP variants introduced in the related research. The intended research is presented in section 2.5, and finally, the summary of this chapter is in section 2.6.

2.2 Mobile Ad Hoc Networks

MANET connects a set of mobile nodes by using wireless links. These MANET nodes are self-configuring and do not require a central facilitated infrastructure to communicate with each other. The nodes do not stay stationary on MANET environment hence the network topology would be changing frequently due to node mobility. Mobility can be configured into individual node mobility or group mobility and the routes is randomly selected or pre-defined. However, the network performance may degrade since the location of the mobile nodes is required to be learnt each time after the movement. The MANET nodes reply on the battery that only has limited power and the nodes are not able to generate power generally, and therefore, when traffic traverse through multiple hops prior to reach destination, it need to be considerate of the energy conservation and re-use of spectrum.

Figure 2.1 represents that each MANET node is centered at its transmission range circle and can communicate with any node within its transmission range. For instance, node 1 and node 2 are able to send and receive data with each other because they are within each other's transmission ranges. However, because node 2 and node 3 are not within each other's transmission range, so node 2 can send data to node 3 only if node 1 is willing to forward the packet. If there is any MANET node wants to send data to node 1, 2, or 3, but it is in none of the transmission ranges of all these nodes, then it is impossible communicate with node 1, 2, or 3 [5].

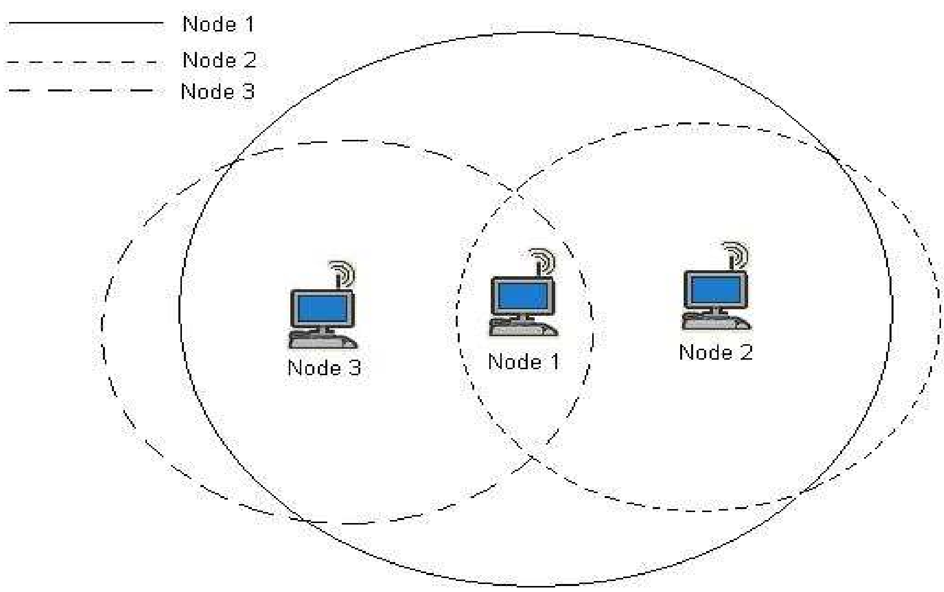


Figure 2.1: Mobile Ad Hoc Network with Three Nodes

2.3 Transmission Control Protocol

Transmission Control Protocol (TCP) is the most common transport layer protocol used in the network environment. TCP uses the port number to provide the process-to-process communication [8]. TCP is a connection-oriented protocol because it requires two communicating end nodes to establish a virtual connection for data transmission before communication starts. TCP provides flow control, error control, and congestion control mechanisms to ensure the reliability at the transport layer, so it adds reliability and connection-oriented features to Internet Protocol (IP) service. This research is mainly focusing on the recovery algorithm of packet losses due to the network congestion and loose communication channel, so only the error control and congestion control are discussed [6].

2.3.1 Error Control Mechanism

Error control includes error detection and error correction mechanisms. Error detection is used to recognize if there is any segment is corrupted, lost, duplicated or arriving at receiver side out of order. TCP detects and corrects the error been identified through three TCP parameters, which are checksum, acknowledgement and timeout.

Checksum is used to check if the segment is corrupted, if it is, then the corrupted segment would be dropped at receiver side and diagnosed as segment loss. The data segment receipt is confirmed by using acknowledgement segment that is sent from the receiver to the sender in TCP. Once the error been detected, TCP retransmits the segments not been acknowledged yet. There are two situations would trigger the retransmission, expiry of retransmission timer and three duplicate acknowledgement segments received at sender side. For the retransmission after Retransmission Timeout (RTO), it would be triggered due to the segments have been sent but not yet acknowledged. The value of RTO is updated based on Round Trip Time (RTT), so it would be dynamic since RTT value is changeable. RTT is the time taken for a segment to reach the destination from the source and for an acknowledgement segment arrive at sender side. For the retransmission after three duplicate acknowledgement segments received at sender side, it retransmits the segments that arrive at receiver side out of order or the segments are discarded. This type of retransmissions would resend the lost segment immediately, and therefore it is also as know as Fast Retransmission[6].

2.3.2 Congestion Control Mechanism

When the congestion is recognized within the network, TCP would start congestion control mechanism which includes three phases: slow start, congestion avoidance and congestion detection. In the beginning of slow start phase, the congestion window is cut down to one Maximum Segment Size (MSS) and increases one MSS each time when an acknowledgement segment is received. The congestion window size would increase exponentially until it reaches the slow start threshold defined by TCP, and hence the slow start phase would end. In the congestion avoidance phase, the congestion window size would only increase additively each time when every segment in the congestion window is acknowledged. When TCP requires retransmitting a segment, the congestion detection mechanism would act accordingly. If the retransmission is due to RTO, the slow start threshold would be set as one half of the current size of congestion window, the size of congestion window decreases to one segment, and then the slow start phase would start again. If the sender receives three acknowledged segments and trigger the retransmission, the slow start threshold would also be set as one half of the current size of congestion window, the congestion window size would be set same as the slow start threshold, and then begins the congestion avoidance phase. TCP would remain in congestion avoidance phase till another RTO happens or it attempts to retransmit another lost segment[6].

2.3.3 TCP Performance Issues on MANET Environment

TCP is a reliable connection-oriented protocol which provides important features of connection management, flow control, error control, and congestion control for data transmission at the transport layer [10]. However, TCP was originally designed for wired network environment so there are some issues appear to make TCP suffers from performance degradation when implemented in wireless network such as MANET [16]. TCP faces several challenges due to the nature of the wireless environments including, loose wireless channel, intermittent disconnection, high error rates, lower throughput, node mobility and prone to packet losses. For instance, the behaviour of packet loss in wireless environments is different from wired networks and the complexity of routing and connection management increases due to the frequent change of network topology [7]. TCP interprets all these random packet losses is due to network congestion and trigger congestion control algorithm to decrease data sending rate by reducing the size of congestion window multiplicatively, and consequently it would result in further TCP performance degradation [7].

2.4 TCP Variants

Various modifications to the existing TCP variants have been developed and reported in the literature. Several recent proposed TCP variants relevant to MANET environment are focused in this section. This section firstly introduces the loss differentiation algorithm and loss recovery algorithm of TCP-WELCOME, and then concentrates on the selected literatures that focus on the performance improvement which recovers from the packet loss due to link failure and network congestion.

2.4.1 TCP-WELCOME

TCP- Wireless Environment, Link losses, and Congestion packet loss ModEls (TCP-WELCOME) [3] is an implicit, end-to-end TCP variant that combines loss differentiation and recovery algorithms facing different kind of the packet loss models. It does not require any intermediate nodes to support the networks. The parameters that differentiate the cause of packet losses are differed by the end-to-end solutions. Contrary to other TCP variants that need the co-operation from the receiver side, TCP-WELCOME estimates the required measurements at the sender-side, and therefore the synchronism between sender and receiver sides is not necessary in this case. Since the introduced loss differentiation and recovery algorithms are also the end-to-end and sender-side solutions, it would decrease the TCP overheads when executing algorithms and improve the overall performance when interacting with other nodes within the network.

Loss Differentiation Algorithm

TCP-WELCOME implemented the loss differentiation algorithm that is able to identify the reason of data packet loss correctly for TCP within MANET environments. It should differentiate between the packet loss models including wireless channel error, network congestion and link failure. The implemented loss differentiation algorithm helps TCP to notice the packet loss and identify the causes of packet loss at the sender side by observing the evolution of RTT samples history of sent packets over the connection, RTO, and if three duplicated acknowledgement packets are received (i.e. the triggers of packet loss).

The wireless channels established between the MANET nodes are considered as the unreliable communication medium. There are several factors that may affect the wireless channel and then provide the unreliable data transmission, such as the conditions of the weather, interferences from any other wireless devices or networks. In order to identify the packet loss is wireless related or congestion related, the evolution of RTT samples is observed. The RTT value increases with time if the packet loss is due to network congestion; this is because the queuing delay is increased at the intermediate nodes. Therefore, the packet loss is not network congestion related if the evolution of RTT samples is almost constant. The consequence of link failure is similar to network congestion because both of them would cause the burst packet losses. However, it is also able to differentiate if the packet loss is due to link failure or network congestion by observing the evolution of RTT samples. In the case of link failure related packet losses, the value of RTT evolution is almost stable, but the value of RTT evolution increases with time in the latter case. To distinguish between link failure and wireless related packet losses, RTO and duplicated acknowledgment packets are monitored. It can be detected through RTO if the communicate connection or the intermediate channel within the route between two end nodes is failed. On the other hand, wireless related packet loss is not generally recognized through RTO but duplicated acknowledgement packets because burst packet loss is not its nature. Figure 2.2 presents the loss differentiation algorithm proposed in TCP-WELCOME.

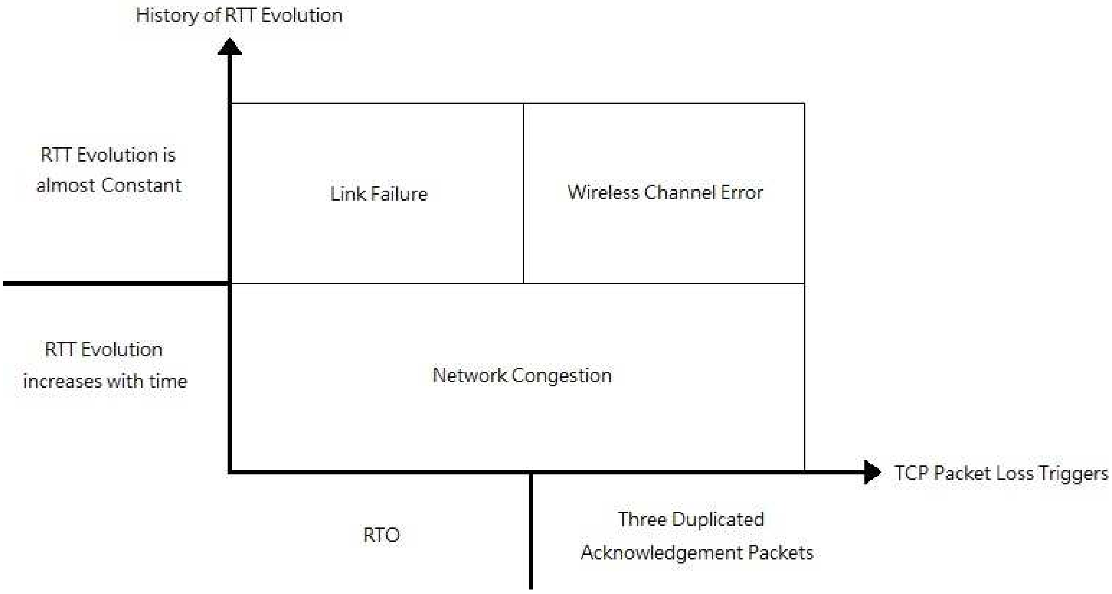


Figure 2.2: Loss Differentiation Algorithm of TCP-WELCOME

Network Congestion Related Packet Loss Classification

If RTT samples evolution of sent packets increases with time, TCP-WELCOME classifies the packet loss is due to network congestion. This is because when network congestion occurs, the buffers of the mobiles nodes are filled gradually as the queuing delay would increase with time, thus, the evolution of RTT samples is also increased at the sender side. Therefore, it is regardless of duplicated acknowledgement packets or RTO with respect to network congestion related packet loss identification.

Wireless Related Packet Loss Classification

If RTT samples evolution of sent packets remains at an average value and does not highly vary with time and three duplicated acknowledgement packets is received at sender side, then the data packet loss is classified as a consequence of wireless channel error. Both queuing delay and network propagation would not be affected and changed over the connection even there is any unreliable channel within the route between the mobile nodes; hence the evolution of RTT samples would remain almost constant rather than increasing with time. Moreover, if there is any channel error on the route which is valid between the sender and receiver mobile nodes, the sender can still always receive the acknowledgment packets by forwarding the packets through other channels or intermediate nodes.

Link Failure Related Packet Loss Classification

If RTT samples evolution of sent packet does not vary with time and stays almost constant but RTO is expired, then the packet loss is recognized as due to link failure in TCP-WELCOME. Link failure is generally occurs within the route that is used to forward the data packets towards to the destination. When link failure happens within the network, the ad hoc routing protocol (AODV, Ad-Hoc On-Demand Distance Vector) requires some time to look up for the new route to forward the data packets. If the time taken for discovering a new route is longer than RTO value defined by TCP-WELCOME and the evolution of RTT samples is relatively constant, then TCP will classify the packet loss is due to link failure. When the sender receives duplicated acknowledgment packets before RTO expires, then TCP will classify the packet loss incorrectly since it will be identify as a wireless related loss. However, this will have limited impact on TCP performance because the reaction would still be more appropriate compare traditional TCP.

Loss Recovery Algorithm

Once the packet loss model is identified by loss differentiation algorithm, TCP-WELCOME uses the most suitable actions to retransmit the lost packet immediately according to the diagnosed packet loss model in order to optimize the fast retransmission and fast recovery performance. RTO expiration and three duplicated acknowledgement packets are used to recognize and identify the cause of packet loss in loss differentiation algorithm, and therefore, the adjustment of RTO value and the TCP packet transmission rate needs be also considered in the packet loss recovery process. Figure 2.3 presents the loss recovery algorithm proposed in TCP-WELCOME.

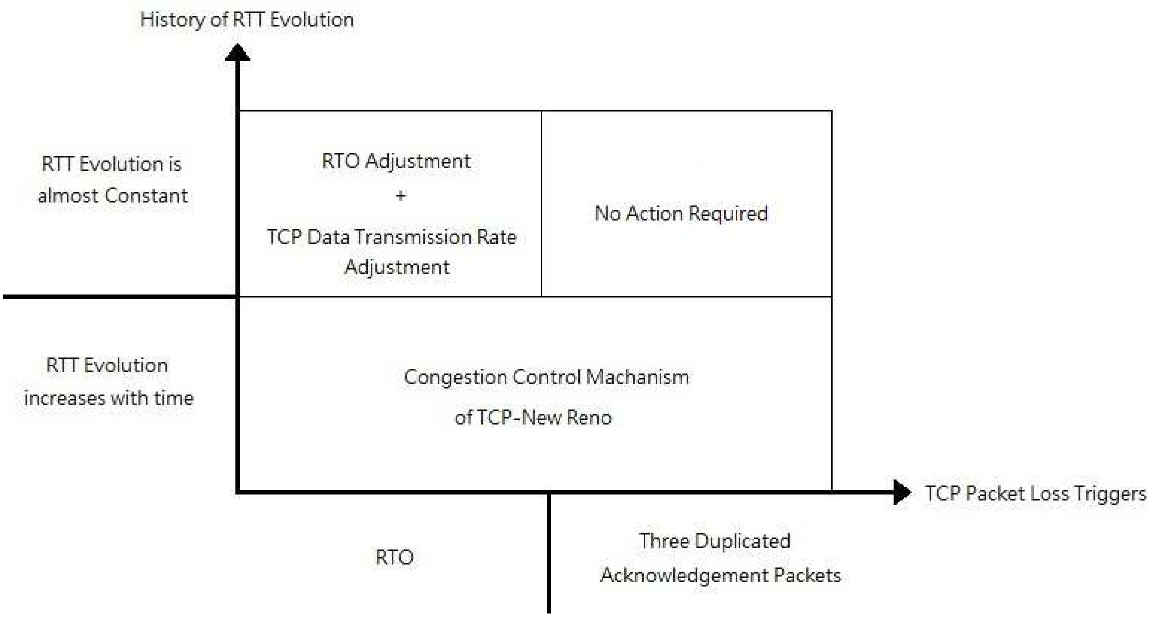


Figure2.3: Loss Recovery Algorithm of TCP-WELCOME

Network Congestion Related Packet Loss Recovery Algorithm

TCP-WELCOME uses the congestion control algorithm of TCP-New Reno for recovery if the packet loss model is identified as network congestion related.

Wireless Related Packet Loss Recovery Algorithm

Since the conditions of the network is not supposed to be varied, and therefore the packet transmission rate of TCP and RTO are not necessary to be updated.

Link Failure Related Packet Loss Recovery Algorithm

1) Adjustment of RTO value

After the new route is discovered by AODV routing protocol, the RTO value is required to be updated based on the queuing delays and propagation of the new route. The queuing delays and propagation could affect and be represented by RTT value, and it can also directly reflect the channel conditions over the connection, therefore the recalculation of RTO value is depending on the RTT values. The new value of RTO is recalculated as follows.

$$\mathrm{RTO_{new}} = \left(\frac{\mathrm{RTT_{new}}}{\mathrm{RTT_{old}}}\right) \times \mathrm{RTO_{old}} \tag{2.1}$$

Where RTTold is the RTT value over the old route before it failed, and RTTnew is the RTT value over the new route that is discovered. The new RTO value is calculated after collecting a number of RTT samples to ensure the accuracy of the estimation.

2) Adjustment of TCP Data Transmission Rate

The conditions of packet buffering and queuing within the end nodes and the path capacity are the main parameters that require to be considered when adjusting the transmission rate. The data transmission rate is estimated according to the characteristic of new route and it is calculated differently under different circumstances. Before any packet loss is notified within the network, TCP can use the actual rate of data transmission and its own congestion control mechanism to handle packet queuing and buffering. The related literature [29] proposed the new solution that updates the congestion window size based on the characteristic of the new route once the packet loss is detected. The RTT value is used in here again because it is able to reflect the capabilities of the transmission links. The new TCP data transmission rate is calculated as follows.

$$\mathrm{CWND_{new}} = \left(\frac{\mathrm{RTT_{old}}}{\mathrm{RTT_{new}}}\right) \times \mathrm{CWND_{old}} \tag{2.2}$$

where CWNDold is the size of the congestion window used for the previous failed route. CWNDnew is the new congestion window size that will be used over the new discovered route.

2.4.2TCP-Adaptive Westwood

TCP-AW implemented a delay and rate based congestion control algorithm which combines the best features of TCP-Adaptive Bandwidth Share Estimation (TCPW-ABSE) [8] and TCP-Adaptive Reno. [4] TCP-AW was designed to improve network utilization substantially and co-exist with TCP-New Reno without significant harm to the latter. TCP-AW estimates the eligible transmission rate and uses the information of packet loss and delays simultaneously in the congestion control mechanism, and therefore it is able to provide superior round trip time fairness, bandwidth efficiency and friendliness compare with other loss based or delay based TCP models. TCP-AW introduces fours steps to approach the TCP performance enhancement.

1) Safe Slow Start

TCP-AW adopted the Eligible Rate Estimation (ERE) algorithm introduced in TCP-ABSE to predict the impending congestion and avoid packet loss in order to reduce retransmission attempts and achieve better efficiency. ERE is based on ACKs arriving interval and used to estimate the capacity of receiver buffer to decelerate the congestion window increment during the slow start. The proposed the idea of create a pause or skip increment to slow down the congestion window increasing rate is given below[4].

IF(Slow Start AND (ERE * (RTT – RTT min)) > 0)
then IF ((CWND mod 2) == 0)
CWND = CWND + 1
else
Create a pause or skip increment

2) Incipient Congestion Detection

Generally when congestion occurs in the network, it would take some time to build up the queue, and therefore it cannot detect the congestion immediately due to the RTT value would not increase promptly. Similar to the popular TCP variant, TCP-Westwood, the researchers [4] used the rate estimation to detect the network congestion before RTT value increasing. The modified RTT value, RTT^c, was used for the congestion detection which is described as follows.

$$RTT^c = RTT + \max\left\{\left(\frac{CWND/RTT}{ERE} - 1\right)\gamma, 0\right\} \tag{2.3}$$

For the best TCP performance in non-congested network, the value of CWND / RTT supposes to be equal to the value of ERE, hence RTT^c would be equal to RTT. Since the arriving interval of ACKs would change rapidly after network congestion, the ERE value would change immediately as well. In this case, ERE value would become small before the value of CWND / RTT actually gets small. Therefore when RTT^c is greater than RTT, that shows the network is congested and the congestion control mechanism should react promptly.

3) Light – Weight ERE Calculation

The ERE calculation introduced in TCP-ABSE requires enormous ACKs history which may degrade the performance. Accordingly, TCP-AW proposed a new rate estimation algorithm which does not require a large ACKs history and it would be more suitable in the high-speed network environment. The calculation is described as follows.

$$\mathrm{ERE_{sample}} = \frac{\sum_{T_j > (t_{k-1} - T_k)} d_j}{T_k} \tag{2.4}$$

$$\delta = \frac{4\mathrm{CWND} - \sum_{T_j > (t_{k-1} - T_k)} d_j}{4\mathrm{CWND} + \sum_{T_j > (t_{k-1} - T_k)} d_j} \tag{2.5}$$

$$\mathrm{ERE_k} = \delta\, \mathrm{ERE_{k-1}} + (1 - \delta)\mathrm{ERE_{sample}} \tag{2.6}$$

Where dj is the amount of data report by ACK j, and tk-1 is the time instance at which k-1th ACK is received at the sender side. Tk is the time taken when is EREsample calculated using the arriving interval of ACK k, and Tj is the time taken when is EREsample calculated using the arriving interval of ACK j. δ would be the time-vary coefficient at Tk.

4) Final Tunings

In TCP-AW, the congestion window increment is based on the packet loss interval time in order to ensure RTT fairness during the congestion avoidance. The proposed mechanism is given below [4].

$$\mathrm{CWND_{dif}} = \left(\alpha L - \mathrm{CWND}\frac{\mathrm{RTT} - \mathrm{RTT_{min}}}{\mathrm{RTT}}\right)\frac{\mathrm{RTT_{min}}}{\mathrm{RTT_{ref}}} \tag{2.7}$$

$$\mathrm{CWND} = \mathrm{CWND} + \mathrm{CWND_{dif}}, \qquad \text{if } \mathrm{CWND_{dif}} \leq 0 \tag{2.8}$$

$$\mathrm{CWND} = \mathrm{CWND} + \mathrm{CWND_{dif}}e^{-c}, \qquad \text{if } \mathrm{CWND_{dif}} > 0 \tag{2.9}$$

Where α is the coefficient of congestion window increment, and the value is assigned as 0.2 based on non-congestion time interval, which is able to increase friendliness and gain more network utilization. L is the time period between two network congestion related packet losses occur. c is the packet loss time interval, which is able to reflect the congestion level in long term, and value of c would be less than 0.4 so it is more accurate for congestion level estimation. RTTref is the gain factor for effect of each error on RTT.

2.4.3 E-TCP

The reason of best effort is generally assigned to the Type of Service (ToS) at the application layer automatically, the best effort traffic are starved and superior timeout would be more susceptible in Quality-of-Service (QoS)-aware MANET environment, especially when TCP facing long delays but handling the data generated by delay sensitive application. The proposed Enhanced TCP (E-TCP) [9] is a cross-layer model which adjusts the retransmission timer and prioritizing TCP ACKs to avoid retransmission of superior timeout and traffic starvation when processing the high priority traffic like voice connection. E-TCP achieves better retransmission efficiency, delay, good-put and utilization of communication channel. The introduced approaches for TCP performance enhancement are discussed below.

1) Prioritization of TCP ACKs

Contrary to other control packet, TCP ACKs are not prioritized and considered as the data packets. If an ACK packet is delayed and the time taken to reach the source node is longer than the defined retransmission timeout, then TCP assumes the ACK packet is lost and retransmits the packets again. For prioritizing TCP ACKS, the fifth access category was defined in E-TCP and assigned as the highest priority. The idea of assigning the highest priority for TCP ACKs is to ensure the packets would reach the source node before the transmission timer expires. In an uplink and downlink traffic coexist network, it would be able to lead to the best effort traffic prioritization because the data packets piggyback some TCP ACKS during the transmission. However, it will only prioritize the TCP ACKs which are data-less otherwise it would degrade the performance of delay sensitive applications in this case. The TCP performance degradation may caused by the ToS is assigned as best effort at the transport layer, and therefore, ToS is assigned as network control (reserved value 224) to the data-less TCP ACKs to address this issue and optimize the data transmission efficiency.

2) Superior Timeouts Reduction

E-TCP optimizes the resource utilization by adjusting the RTO value to limit the attempts of retransmission due to the medium contention. The purpose of introducing the RTO adjustment is to allow the TCP ACKS are able to reach the sender node before the retransmission timer actually expires. E-TCP freezes the retransmission timer countdown when the medium is sensed busy and the timer resumes retransmission countdown once the medium is idle. Even it would add the additional delays on RTO values by halting the retransmission timer; however, this additional delay reflects the status of medium contention. The additional delay is added to RTO every time the medium is sensed busy and generated continuously until the retransmission timer expires or the TCP ACKS is received at the sender side.

E-TCP relies on the collision avoidance mechanism provided in the Medium Access Control (MAC) layer to monitor the status of the medium contention. The MAC layer notifies the transport layer of medium availability based on the value of Network Allocation Vector (NAV) parameter, which informs and allows the node to access the medium for a specific period of time. If the value of NAV is greater than zero, the medium is considered as busy, and the medium is sensed as idle otherwise. This status information of medium contention would then inform to the transport layer from the MAC layer.

2.4.4 RED Tuning Approach for TCP Performance

The related literature [10] introduced the approach of tuning one of TCP congestion control mechanism, Random Early Detection (RED), to handle the packet loss due to link failure and optimize the TCP performance on MANET environment. Unlike other related work focus on updating congestion window size and RTO value, the proposed scheme modified the parameters of RED, which is one of the congestion control variables and it also can influence the TCP performance. Generally, the early congestion detection in TCP helps to notify the sender if there are any randomly discarded packets through RED, but RED cannot identify the packet is dropped due to network congestion or link failure when implemented on MANET. Furthermore, the size of average queue length grows rapidly and even exceeds the minimum threshold while TCP is looking up for new routing path, and once the average queue length size if greater than the maximum threshold, the arriving data packets would be discarded at sender side and results in TCP performance degradation. And therefore the related RED parameters, the average queue length size, minimum and maximum thresholds, are updated to enhance the performance in the proposed scheme. In order to handle the link failure due to the MANET node mobility, the adjustment of RED parameters are described as follows.

$$\text{MaximumThreshold} = \text{MinimumThreshold} \times \text{x} \tag{2.10}$$

$$\text{AverageQueueLength} = (1 - \text{Weight}) \times \text{AverageQueueLength} + \text{Weight} \times \text{SampleLength} \tag{2.11}$$

For the equation of adjusting maximum threshold, the value of x is set as 3. For the equation of calculating the average queue length, it can compute a low-pass filter which can avoid the average queue length increases rapidly in the early network congestion phase. The range of the weight value is assigned between one to zero. If the value of weight is assigned too small, then it is not able to reflect the actual queue length. Also if the value of weight is set too big, then early network congestion cannot be filtered.

The researchers [10] also introduced a self-configuration algorithm for tuning the RED parameters dynamically to handle the node mobility for the further performance improvement. The value of minimum threshold was updated to temporarily accommodate the increasing of data packet amount generated during the new route look-up. The RED algorithm is given below.

```
IF (Next Packet == Previous Packet) then
        Counter = Counter + 1
        IF (Counter > Threshold Value) then
            IF (Minimum Threshold < Maximum Threshold) then
                Minimum Threshold += β * (Maximum Threshold – Minimum Threshold
                initial value)
ELSE IF (Counter > 0) then
        Counter = Counter – 1
        IF (Minimum Threshold > Minimum Threshold initial value) then
            Minimum Threshold −= β * (Maximum Threshold – Minimum Threshold initial
value)
```

If the first packet on the queue is not delivered and retain on the queue, then increment of the counter would be additively. The communication channel is considered as failed if this situation remains the same and the value of counter is greater than the threshold value. In this case, RED requires adjusting the minimum threshold and assigns a higher value into it to temporarily tolerate the generated packets during the establishment of the new link. Otherwise, the minimum threshold value would be updated to its original value gradually. The amount of packets to decrease or increase is β * (Maximum Threshold – Minimum Threshold initial value), where the value of β is the range of 0 to 1.

Table 2.1: Key researchers and their main contributions for the performance improvement of TCP

Researchers	**Contribution**	**Key concepts/description**
Henna. 2009	TCP-Vegas, TCP-Tahoe, TCP-Reno, TCP-New Reno, TCP-SACK, TCP-Veno, TCP-Westwood, TCP-Westwood New Reno and TCP-New Jersey [7]	Investigated the performance of TCP variants under random packet loss rate
Kim et al., 2006	TCP-Vegas and TCP-Reno [11]	Investigated the suitable TCP variant for smooth integration by evaluating the performance
Mbarushimana and Shahrabi. 2009	E-TCP [9]	Proposed a cross-Layer TCP enhancement that can avoid the unnecessary retransmission and traffic starvation
Wu et al., 2007	TCP-HO [12]	Proposed the method that can minimize the handoff bandwidth delay by estimating the bandwidth of new link
Seddik-Ghaleb et al., 2009	TCP-WELCOME [3]	Investigated the reasons for packet losses and then developed an appropriate recovery mechanism.

2.5 Intended Research

This research aims to improve the performance of TCP over MANETs by modifying the existing TCP-WELCOME. After an in-depth literature review, TCP-WELCOME is found to be better suited for MANETs compared to other variants of TCP. It enhanced the performance of TCP by adopting loss differentiation algorithm and loss recovery algorithm which can identify the packet loss models more accurately and trigger the proper recovery process accordingly. However, TCP-WELCOME has weaknesses especially with its loss recovery algorithm. This thesis attempts to redesign TCP-WELCOME's loss recovery algorithm to improve TCP performance called Enhanced TCP-WELCOME. The detailed description of Enhanced TCP-WELCOME is presented in Chapter 4.

Since TCP-WELCOME was originally designed for small network with low speed MANET nodes, so the scenarios on network size focus on the small, small-medium and medium networks in this experiment. A small network with less than 10 nodes, the number of nodes in a small-medium network would be greater than 10 but less than 20 and the node size of a medium network would be defined with 20. The mobility of the MANET nodes would be considered in all scenarios in this experiment. The low node speeds would be used to represent the people movement, for instance, walking and running in the office. Traffic load considers general traffic load, medium load and heavy load. Furthermore, the varying wireless channel conditions would influence the TCP performance, and therefore the data transmission over the perfect and noisy wireless channels would be considered in the scenarios. The conducted experiment design and investigation are discussed in detail in Chapter 5.

2.6 Summary

This chapter provides the background information of MANET environment, and several literatures of recently proposed TCP variants are reviewed and discussed. Precisely, a closer look at the introduced TCP performance improvement approaches in the relevant variants provides the significant information of the differences on the adjustment of the performance related parameters. The objective of the research is to modify the existing TCP variant, TCP-WELCOME, to achieve the better performance enhancement. The scenarios would focus on the influence of network size, traffic load, node mobility and conditions of communication channel impact on TCP performance on MANET environment. A simulation approach will be used to evaluate the TCP performance with different defined scenarios; the detail of research methodology is presented in Chapter 3.

Chapter 3

Research Methodology

3.1 Introduction

This chapter outlined the research methodology used for evaluating and analyzing the proposed Enhanced TCP-WELCOME. Section 3.2 discusses the advantage and disadvantage of the three methodologies been generally used in the related research areas. The strengths and weaknesses of the popular simulators used for TCP variants evaluation and performance analysis are discussed in section 3.3. Section 3.4 summarises this chapter.

3.2 Research Paradigm

For network modeling and performance evaluation in the related areas, the methodologies used generally include analytical modeling, computer simulation and direct experiment. Analytical analysis involves the development of new information based on the spatial analysis and the mathematical computation. Analytical analysis is difficult to operate and control the protocol [16] even it may be able to define a new TCP variant. Besides, due to the node mobility and scalability of MANETs, it is challenging to configure and reconfigure the nodes in the large networks with the frequently changed topology.

The acquired modeling and evaluation results of direct experiment would be more accurate since the experiment is carried out in the reality. The information of the TCP behaviour and the performance related parameters can be easily observed if direct experiment is used to model the network. However, it is not easy to implement the MANET network if using direct experiment in this study due to the complexity of the MANET environment and the cost of the required resources and efforts.

The system based simulation experiment is able to construct the network model and evaluate the model with consists of defined scenarios using a simulator (a computer program) to acquire the solution and approach the research goal. Simulation experiment can build a model like a real world system and be able to observe and predict the variant behaviour in the system, it can reduce the cost mentioned in direct experiment, and also, the MANET implementation complexity issue would not be arisen when using the simulator for network modeling. Simulation experiment not only can generate the results of defined measurement to prove the analytical model, but also can evaluate the new variant performance and compare with the performance of existing variant. However, simulation experiment requires the user to have good background knowledge about the simulator, or the evaluation results and the obtained solutions may not be representative or accurate in the otherwise. If a simulator is easy to use, able to generate the valid and appropriate analysis results and also has the flexibility when developing or modifying the model, then the simulator is considered as a good simulator [16].Apart from using a good simulator, the credibility of the findings acquired from the simulation experiment also relies on the result verification and validation process to ensure the viability, which including the validation of parameters and event, operational graphics, and comparison of the obtained results and valid historical data [17] [18].

Simulation methodology is commonly used to evaluate the proposed model and carry out the solution to approach the related research question because only few assumptions is required and it can behave like a real world system. The TCP modification complexity would need the features provided by a simulator such as flexibility of model development and validation, and performance evaluation. Therefore, to address the research question (i.e. *How to improve the performance of TCP-WELCOME),* the simulation methodology is adopted to examine the impact on TCP performance on MANET by constructing the proposed TCP variant, evaluating the performance, and comparing the results with the performance of the existing variant. Various defined scenarios will be used to study the impact of proposed TCP variant performance in MANET environment based on OPNET Modeler. The simulation modelling and scenarios are discussed in detail in Chapter 5.

3.3 Optimized Network Engineering Tool (OPNET) Simulator

There are several existing simulators that can be used to compare the network performance. The most popular and commonly used simulators to evaluate the impact on performance of TCP variants over wireless networks are NS2 and OPNET Modeler.

NS2 was originally designed for the wired network simulations, and now is also able to support the simulations over wireless networks. It is a discrete event network simulator and core programming language of NS2 is C++. NS2 is one of the simulators that can build the model like a network system in the real world, and it has the open source license, which is another reason to make it become a popular. There are some drawbacks when using NS2 as the simulator in the experiment, which includes the limited documentation, the inconsistency of the programming environment across different released versions. Furthermore, it is lack of tool to describe the scenarios used for simulation and most of tools are in the scripting language and therefore it would be difficult to debug the code and learn the background knowledge of the simulator [19] [20].

OPNET Modeler is a discrete event and object-oriented simulator which is able to perform the simulation for general networks. The modeling environment provided by OPNET Modeler can achieve the parameter and event validation, ease the implantation and construction of the model of the scenarios, and it offers the simulation results comparison and analysis of the network performance. Contrary to NS2, OPNET ensures the consistency of programming environment across the released versions, user friendly interface and it is embedded with model library, results analysis tool, and different types of editors. It is used for simulation experiment widely to evaluate and analyze the network model performance, and it is capable to model the TCP variant over MANET environment. OPNET employs the three tier hierarchical modeling architecture. The first tier is used to design the network topology for the model, the second tier consist the data traffic flow, and the third tier handles the data flow control by using the process editor. This three tier hierarchical architecture is significant when modeling, and evaluating and revising the TCP variant on MANET environment.

There are some disadvantages to use OPNET Modeler as a simulation tool. OPNET Modeler is a commercial package which the cost would be relatively expensive and heavily controlled. On another hand, running the simulation on OPNET Modeler requires a large amount of CPU usage, so it would cost higher computing power. However, it does not only offer the useful embedded tools, but also provides the user friendly graphical interface, and the output evaluation result is also graphical and can be customized, and therefore, OPNET Modeler would be used for the simulation experiment in this research.

3.4 Summary

This chapter described three methodologies been generally used for modeling the networks and evaluating the performance. There are some drawbacks when using analytical and direct experiment to model TCP variant on the complex MANET environment. The simulation experiment methodology is chosen to model and evaluate the TCP variant in this research. It is able to construct the model like a system in the real world. OPNET Modeler is able to simulate the defined scenarios easily and generate the valid result to represent the performance of TCP variant more accurately. There are some drawbacks in OPNET Modeler, however, it can still consider as a good simulator because of its strength, so it would be suitable to use OPNET Modeler for the simulation experiment in this research.

Chapter 4

Enhanced TCP-WELCOME

4.1 Introduction

In this chapter, an Enhanced TCP-WELCOME is proposed to improve the overall performance of TCP in MANETs. The algorithm of the proposed Enhanced TCP-WELCOME is presented in Section 4.2. The proposed system offers the combined features of TCP-WELCOME and TCP-AW to identify the packet loss model correctly and performs the recovery process accordingly. Section 4.3 summarises this chapter

4.2 Proposal Algorithm

Enhanced TCP-WELCOME implements the loss differentiation algorithm and loss recovery algorithm for congestion detection and act appropriately according to the identified packet loss model. It also combines the features of TCP-AW [4] to improve the early congestion detection process and handle the network congestion related packet loss model more efficiently rather than just use the traditional congestion control mechanism of TCP-New Reno proposed in TCP-WELCOME. Figure 4.3 presents the flow chart of Enhanced TCP-WELCOME algorithm.

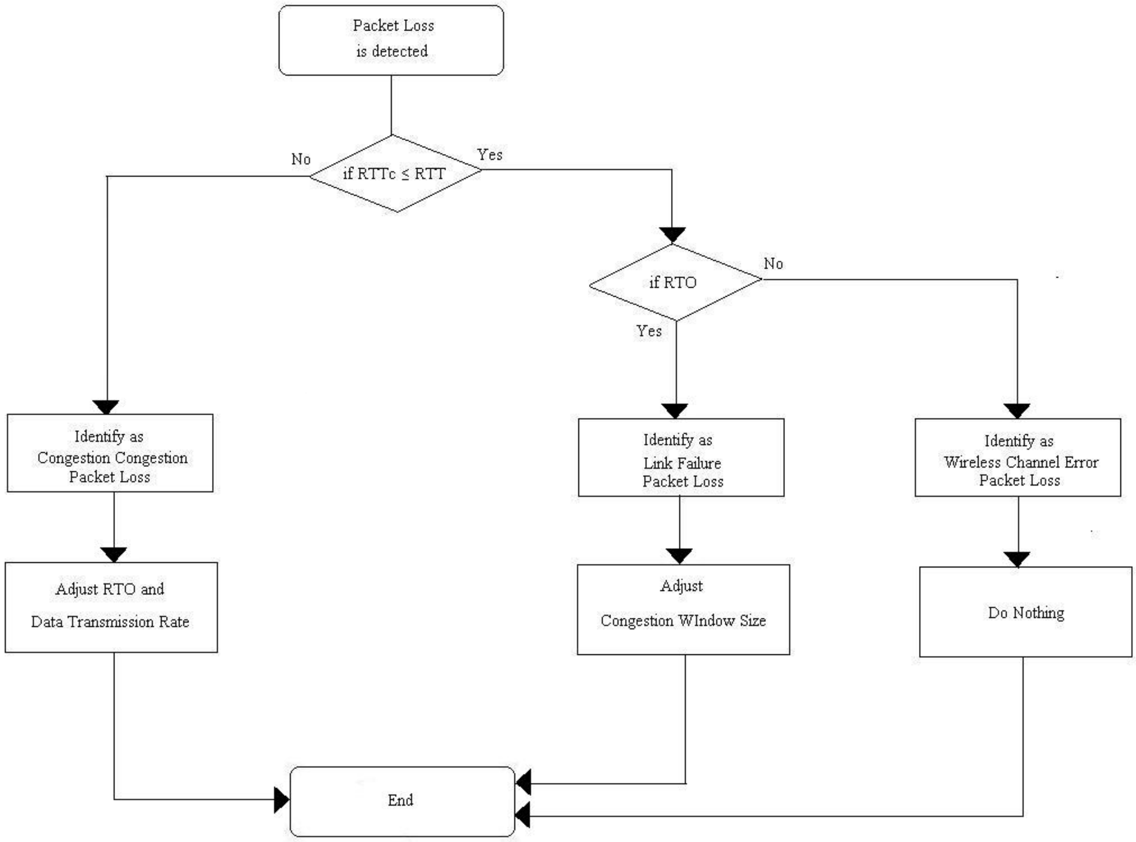


Figure 4.1: Proposed Enhanced TCP-WELCOME Algorithm

4.2.1 Loss Differentiation Algorithm

The loss differentiation algorithm implemented in Enhanced TCP-WELCOME is used to identify the reason of packet discarded during the data transmission accurately. The general cause of packet loss includes network congestion, link failure and wireless channel error. The loss differentiation algorithm is able to detect the packet discarded and identify the packet loss model at the sender side. In order to detect the packet loss, the parameters related to the packet loss trigger are observed during the transmission, including RTO, three duplicated ACKs, and the RTT value.

Contrary to TCP-WELCOME requires monitoring a history of evolution of RTT samples, Enhanced TCP-WELCOME adopted the Incipient Congestion Detect mechanism introduced in TCP-AW which does not need a large amount of RTT history and can detect the congestion before RTT value actually increases. Detect the congestion in the network through RTT value would need to take longer time to wait for the queue to be built up and results in the RTT value, hence it is not able to detect the congestion immediately when the data packets starts to be discarded. The detail of modified RTT value introduced in incipient congestion detection is described as follows.

$$\mathrm{RTT}^{c} = \mathrm{RTT} + \max\left\{\left(\frac{\mathrm{CWND/RTT}}{\mathrm{ERE}} - 1\right)\gamma, 0\right\} \tag{4.1}$$

The light-weight ERE calculation algorithm proposed in Enhanced TCP-WELCOME resembles TCP-AW but does not require ACK history.

$$S_{rate} = \frac{CWND}{RTTmin} \tag{4.2}$$

$$T_k = RTT \times \frac{(S_{rate} - ERE_{prev})}{S_{rate}} \tag{4.3}$$

$$ERE_{sample} = \frac{Acknowledged\_bytes}{T_k} \tag{4.4}$$

$$\delta = \frac{4CWND - Acknowledged\_bytes}{4CWND + Acknowledged\_bytes} \tag{4.5}$$

$$ERE_k = \delta\, ERE_{k-1} + (1 - \delta) ERE_{sample} \tag{4.6}$$

where Srate is the instantaneous sending rate, and EREprev is the generated ERE value used for the previous transmission. Tk is the time taken when is EREsample calculated using the arriving interval of ACK k, and Tj is the time taken when is EREsample calculated using the arriving interval of ACK j. δ would be the time-vary coefficient at Tk.

To identify the packet loss is link failure related or network congestion related, the value of RTT^c is observed. If the packet loss is as a consequence of link failure, then the value RTT^c would be less or equal to the RTT value, but if the RTT^c value is greater than the RTT value, then the packet loss is diagnosed as network congestion related. It can also differentiate if packet loss is caused by wireless channel error or network congestion by observing the RTT^c value, when the value of RTT is less than RTT^c value, then the packet loss model is identified as a consequence of network congestion, otherwise, it would be considered a the wireless channel error related packet loss. To distinguish between link failure and wireless related packet losses, the parameters related to the triggers of the retransmission are monitored. It can be recognized through RTO if the data transmission link or the intermediate link within the route between sender and receiver nodes is down. Besides, wireless related packet loss is generally detected through three duplicated ACKs because it is not its nature if burst packet loss occurs. Figure 4.1 illustrates the loss differentiation algorithm proposed in TCP-WELCOME.

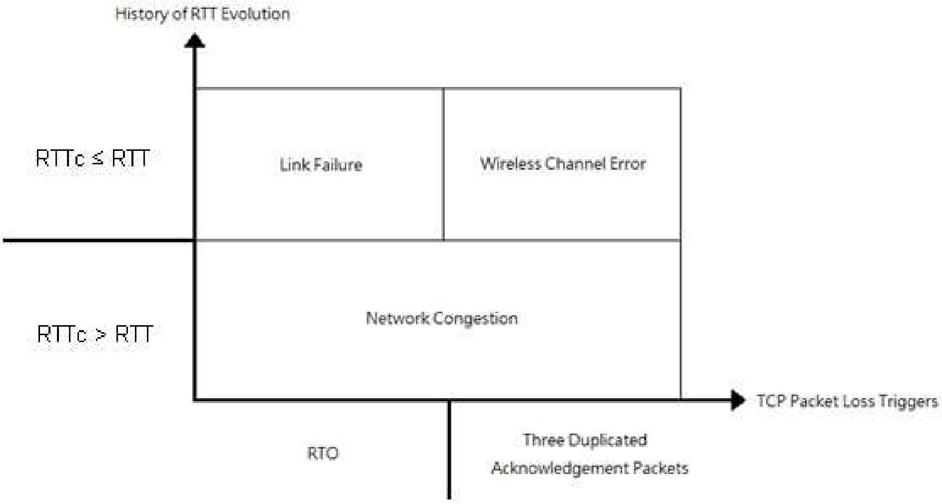


Figure 4.2: Loss Differentiation Algorithm in Enhanced TCP-WELCOME

Network Congestion Related Packet Loss Classification

If RTT^c value is greater than the RTT value, Enhanced TCP-WELCOME classifies the packet loss is due to network congestion. This is because when network congestion occurs, the arrival time of ACK would be change promptly and results in the value of ERE changes immediately as well. Therefore, it is regardless of three duplicated acknowledgement packets or RTO with respect to network congestion related packet loss identification.

Wireless Related Packet Loss Classification

When the value of RTT^c is less or equal to the RTT value and three duplicated acknowledgement packets is received at sender side, then the data packet loss is classified as a consequence of wireless channel error. Both queuing delay and network propagation would not be affected and changed over the connection even there is any unreliable channel within the route between the mobile nodes. Moreover, if there is any channel error on the route which is valid between the sender and receiver mobile nodes, the sender can still always receive the ACKs by forwarding the packets through other paths or intermediate nodes.

Link Failure Related Packet Loss Classification

Link failure is generally occurs within the route that is used to forward the data packets towards to the destination. When link failure happens within the network, the ad hoc routing protocol (AODV) requires some time to look up for the new route to forward the data packets. If the time taken for discovering a new route is longer than RTO value defined by Enhanced

TCP-WELCOME and the RTT^c value is equal or less than RTT value, then TCP would classify the packet loss is due to link failure. When the sender receives three duplicated ACKs before RTO expires, then TCP would classify the packet loss incorrectly since it will be identify as a wireless related loss. However, since the reaction would still be more suitable compare with traditional TCP, so it would only have limited impact on TCP performance.

4.2.2 Loss Recovery Algorithm

In order to optimize the fast retransmit and fast recovery processes in congestion control mechanism, loss recovery algorithm need to perform the most appropriate action to recover the packet loss accordingly. Since the timer of RTO and duplicated ACKS are used to classify the model of packet loss, and therefore, it would use the most suitable algorithms to update and adjust the RTO value and congestion window size to improve the congestion recovery process. Figure 4.1 presents the loss recovery algorithm proposed in TCP-WELCOME.

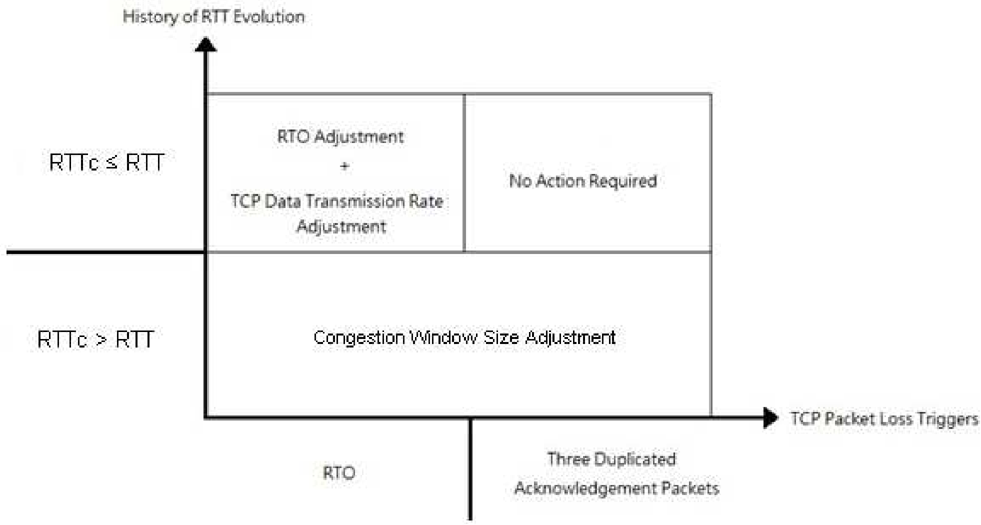


Figure 4.3: Loss Recovery Algorithm in Enhanced TCP-WELCOME

Network Congestion Related Packet Loss Recovery Algorithm

Enhanced TCP-WELCOME uses the congestion control algorithm introduced in TCP-AW for recovery if the packet loss model is identified as network congestion related. The congestion window increment is based on the packet loss interval time in order to ensure RTT fairness. The adjustment of congestion window size is given below.

$$\mathrm{CWND_{dif}} = \left(\alpha L - \mathrm{CWND}\,\frac{\mathrm{RTT} - \mathrm{RTT_{min}}}{\mathrm{RTT}}\right)\frac{\mathrm{RTT_{min}}}{\mathrm{RTT_{ref}}} \tag{4.7}$$

$$\mathrm{CWND} = \mathrm{CWND} + \mathrm{CWND_{dif}}, \qquad \text{if } \mathrm{CWND_{dif}} \leq 0 \tag{4.8}$$

$$\mathrm{CWND} = \mathrm{CWND} + \mathrm{CWND_{dif}}e^{-c}, \qquad \text{if } \mathrm{CWND_{dif}} > 0 \tag{4.9}$$

Where α is the coefficient of congestion window increment, and the value is assigned as 0.2 based on non-congestion time interval, which is able to increase friendliness and gain more network utilization. L is the time period between two network congestion related packet losses occur. c is the packet loss time interval, which is able to reflect the congestion level in long term, and value of c would be less than 0.4 so it is more accurate for congestion level estimation. RTTref is the gain factor for effect of each error on RTT.

This congestion control mechanism was originally designed for the coexistence with TCP-New Reno to achieve better fairness and can reduce the delay generated over the connection. The idea of implementing this mechanism in Enhanced TCP-WELCOME is to improve the TCP performance by overcome the issues existing in congestion control algorithm of TCP-New Reno and handle the packet loss retransmission due to the network congestion.

Wireless Related Packet Loss Recovery Algorithm

Since the conditions of the network is not supposed to be varied, so the parameters that may influence the TCP performance, congestion window size and RTO value, are not necessary to be adjusted.

Link Failure Related Packet Loss Recovery Algorithm

The recovery process of the packet loss due to link failure would remain the same mechanism as proposed in TCP-WELCOME.

1) Adjustment of RTO value

The RTO value is required to be recalculated based on the queuing delays and propagation of the new discovered path. Since both of the queuing delays and propagation could influence and results in RTT value and it is able to reflect the channel conditions over the connection directly, so the adjustment of RTO value is based on the RTT values. The calculation of new RTO value is described as follows.

$$\mathrm{RTO}_{\mathrm{new}} = \left(\frac{\mathrm{RTT}_{\mathrm{new}}}{\mathrm{RTT}_{\mathrm{old}}}\right) \times \mathrm{RTO}_{\mathrm{old}} \tag{4.10}$$

Where RTTold is the RTT value over the old route before it was down, and RTTnew is the RTT value over the new route that is discovered. The new RTO value is updated after collecting a certain amount of RTT samples to ensure the estimation would be accurate.

2) Adjustment of TCP Data Transmission Rate

The conditions of packet buffering and queuing within the end nodes and the path capacity are the main parameters that require to be considered when updating the size of congestion window. The rate of data transmission is estimated according to the characteristic of new discovered path and it is calculated differently under different circumstances. TCP can use the actual rate of data transmission and its own congestion control mechanism to handle packet queuing and buffering before any packet loss is notified within the network. The RTT value is used in here again because it is able to reflect the capabilities of the transmission links. The calculation of the new TCP data transmission rate is given below.

$$\mathrm{CWND_{new}} = \left(\frac{\mathrm{RTT_{old}}}{\mathrm{RTT_{new}}}\right) \times \mathrm{CWND_{old}} \tag{4.11}$$

where CWNDold is the size of the congestion window used for the previous failed route. CWNDnew is the new congestion window size that will be used over the new discovered route.

4.3 Summary

This chapter outlined the new TCP variant proposal of this study. Enhanced TCP-WELCOME is combined with the features of TCP-WELCOME and TCP-AW. The proposed variant implemented the loss differentiation algorithm and loss recovery algorithm to identify the packet loss model correctly and performance the recovery process appropriately. Contrary to TCP-WELCOME, the proposed variant uses the early congestion detection mechanism to detect if there is any packet been discarded, furthermore, the new congestion control mechanism resembles TCP-AW is proposed to handle the packet loss due to network congestion to improve the TCP performance and fairness.

Chapter 5

Experiment Design and Investigation

5.1 Introduction

This chapter provides the detailed description of the experimental design used for the simulation to model the network and evaluate the impact on Enhanced TCP-WELCOME performance. Section 5.2 outlines the performance metrics used for measure the performance of Enhanced TCP-WELCOME. The design of the network model is presented in section 5.3, and description of the scenarios is outlined in Section 5.4. Section 5.5 is the summary of this chapter.

5.2 Performance Metrics

Since this research is the performance study, the performance metrics is necessarily to be used in the simulation as the measurement for the TCP performance evaluation. The metrics which been generally used to evaluate the proposed mechanisms and models in the research area relevant to TCP performance over MANET are throughput, end-to-end delay, packet delivery ratio and the retransmission attempts, and the description of each performance metrics are provided as follows:

5.2.1 Throughput

Throughput is the average rate of the delivered packets which been successfully acknowledged by the receiver. Throughput is the most significant and commonly used metric for the performance measurement and it is normally measured in the total number of bit are sent through the communication channel per second. As used in the related studies [21, 9, 22-25], the obtained throughput efficiently represents the performance of TCP by showing the amount of data have been successfully delivered from a source to the destination. However, throughput includes the retransmitted packets which may be discarded as the consequence of the overheads of TCP and link layer protocol. Furthermore, the node mobility would decrease if throughput increases, and therefore, in order to show the TCP performance enhancement in MANET, the evaluation measurement cannot rely on throughput.

5.2.2 End-to-End Delay

End-to-End delay is the average time taken for a packet travels across the network from the source to destination and be acknowledged by the receiver. End-to-End delay is combined from the different type of generated delay since the packet is ready to be forwarded till the packet is acknowledged, which including the time taken for the router to process the packets, the time taken for the packets queuing on the sending buffer, the time taken to put the packets on the transmission medium and the time taken for the packets to reach the destination.

End-to-end delay shows how efficiency the network resource has been used by TCP. If end-to-end delay is relatively high then it represents the medium is busy and in lack of flow control [9]. Therefore, minimizing the end-to-end delay can represent as reduce the possibility of congestion occurrence and achieve the resource utilization.

5.2.3 Retransmission Attempts

Retransmission Attempts is the source sends out the discarded packets again to the destination. Loss of packets is not only due to the medium contention, but also due to the link failure which may be caused by the node mobility [23]. Retransmission can enhance the TCP performance by reducing the cost of the route failure [21]. But as the number of times of retransmission is increased, the power consumption is increased [24] as well.

5.2.4.Packet Delivery Ratio

Packet delivery ratio is the ratio between the number of packets received by the receiver and the number of packets generated by the sender. It does not include the retransmitted data packets and the protocol overheads. Most of the researchers of the previous studies [21, 9, 23, 25] used the simulation results of packet delivery ratio as the statistical evidence to prove the TCP performance enhancement.

Throughput and end-to-end delay are the most important metrics to measure the best effort traffic of proposed TCP variant, and retransmission attempts is used to measure the performance of proposed congestion control mechanism. These performance metrics are not absolutely independent to each other, for instance, the throughput would decrease and the retransmission count increases relatively if the end-to-end delay is increasing due to the overhead generated from TCP and MAC protocol. In the other words, if the generated end-to-end delay is relatively small over the communication channel, then the TCP throughput increases and retransmission attempts is reduced; hence the TCP variant performs well in the model.

5.3 Assumptions

The assumptions of this investigation are, 1) the battery life is not set on the MANET node because this attribute does not exist in OPNET Modeler; 2) the background traffic exists in MANET is not considered so it is assumed to be the same all the time; 3) the nodes do not generate any traffic at anytime; 4) all the mobile nodes are moving within the network with defined node speed; 5) not all the nodes are in the movement at a given time.

5.4 Network Modeling

Figure 5.1 presents the experimental scenarios. The two TCP variants, Enhanced TCP-WELCONE and TCP-WELCOME are implemented in 802.11b standard network with the mobile nodes that only move within the defined propagation range. These experiments are aiming to evaluate the TCP variants under different circumstances and compare the simulation results to identify if Enhanced TCP-WELCOME is able to achieve better TCP performance over MANET.

The investigations of the simulation experiment in this study are categorised into network size, traffic load, node mobility and wireless channel conditions. In the investigation of network size, Enhanced TCP-WELCOME is implemented into small, small-medium and medium networks, and the traffic load node mobility and wireless channel conditions are also investigated with the network size.

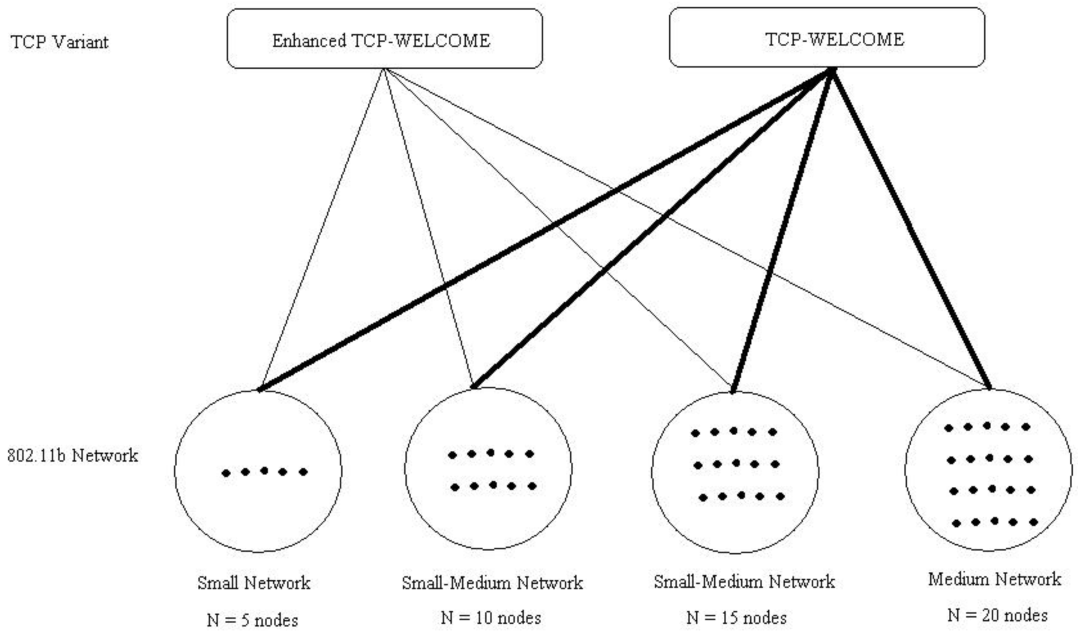


Figure 5.1: Simulation Scenarios

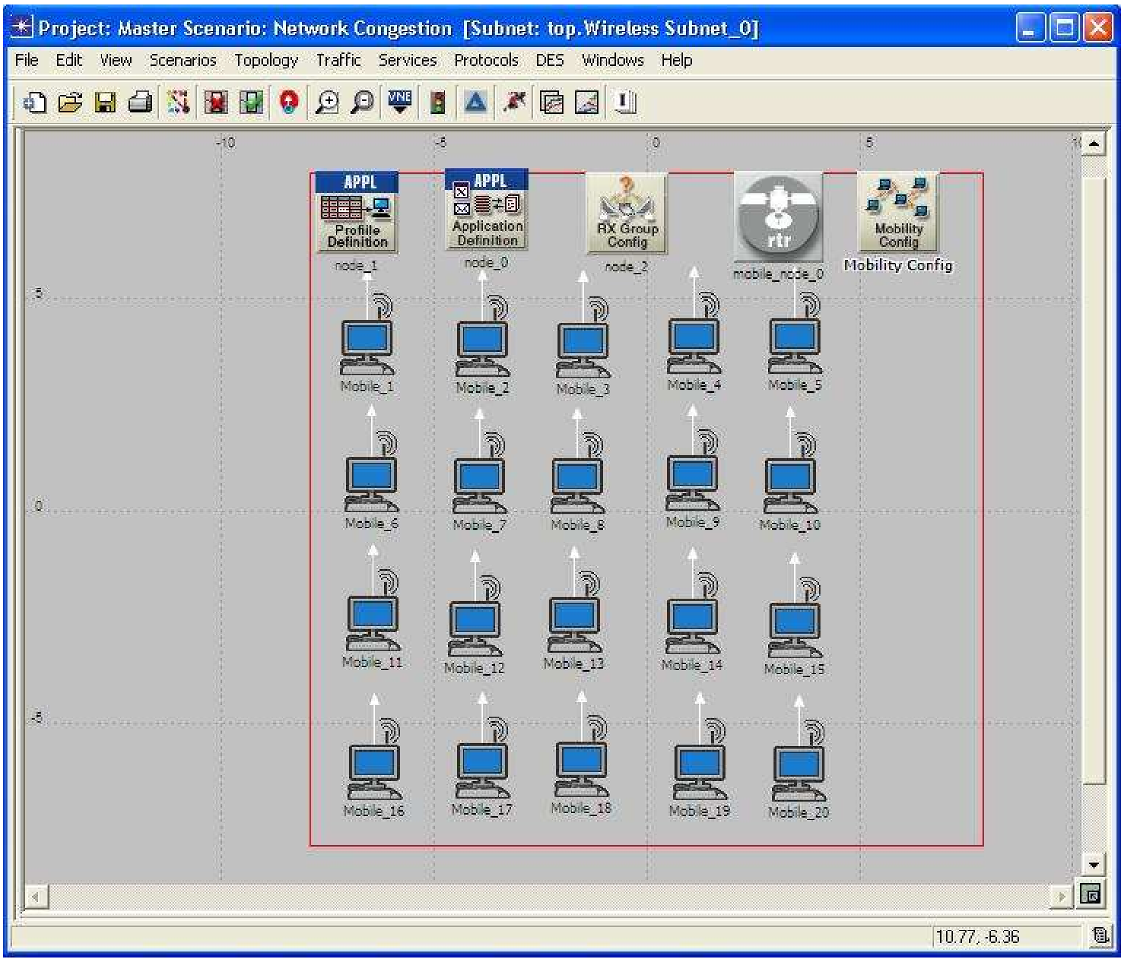


Figure 5.2: OPNET representation of 802.11b model (Node Size = 20)

Figure 5.2 illustrate the constructed MANET mode in OPNET Modeler simulation environment with the network size of 20 nodes. The objects in the constructed model are MANET mobile nodes, MANET gateway, the configuration objects of RX group and mobility model, and the profile and application definitions objects. The name of each mobile node is edited according to the node number it presented in the network. For example, mobile_5 represented the node number 5 in the network. The MANET gateway is a wireless local area network based router with one Ethernet interface. This object is used to connect the mobile nodes to IP and support the transmission of the generated FTP traffic by forwarding the received packet towards to the destination. The configuration of routing protocol on all the mobiles nodes and gateway are the same.

The mobility configuration object is used to define the node movement based on the configured mobility related parameters that individual MANET nodes reference to model mobility. The RX group configuration object allows all the mobile nodes in the network move within the allocated moving area, and if the traffic in generated outside of propagation range from any node will be discarded.

The application definitions object provides the different types of applications available in OPNET Modeler, the corresponding attributes in the process can be specified and create new applications. For instance, a high load FTP application indicates the application performing high loading traffic transmission using FTP. The specified name of defined application is used when creating a new user profile on the profile definitions object. The profiles definitions object is used to create the user profile that can be implemented on the nodes and generate the application layer traffic within the network. The application is requires to be defined first before using the profile definitions object. The traffic pattern can be specified by either application or profile definitions objects.

The general parameters configured in the constructed model for all the experiments are showing in Table 5.1, and the simulation parameters of AODV and TCP protocols used for evaluation of Enhanced TCP-WELCOME performance are carefully designed and presented in Table 5.2 and 5.3. In order to observe the impact on Enhanced TCP-WELCOME over MANET in all possible directions, there would only be one control factor is varied at a time in all defined scenarios. The control factors used to reflect the impact on performance of Enhanced TCP-WELCOME are node speed and packet length.

Table5.1: General Parameters used in simulation

Parameters	**Value**
Area	250 x 250 square meters
Network Size	5, 10, 15, 20
Mobility Model	Random waypoint (Auto Create)
Data Rate	11Mbps
Transmission Power	0.005 watt
Packet Size	5000, 10000, 15000, 20,000, 25,000 bytes
Data Type	FTP
Mobility Speed	3, 4, 5, 6, 7 meter/second
Noise Figure	1, 5
Simulation Duration	1000 seconds

Table 5.2: AODV Parameters used in simulation

Parameters	**Value**
Active Route Timeout	3
Hello Interval	Uniform(1,1.1) second
Allow Hello Loss	2
Net Diameter	35
Node Traversal Time	0.04
Route Error Rate Limit	10
Timeout Buffer	2

Table 5.3: Main TCP parameters used in simulation

Parameters	**Value**
Version	New Reno
Receive Buffer	8760 bytes
Receive Buffer Adjustment	None
Maximum ACK Delay	0.2 seconds
Maximum ACK Segments	2
Slow Start Initial Count	2 MSS
Fast Retransmit	Enable
Duplicate ACK Threshold	3
Fast Recovery	New Reno
Initial RTO	3.0 seconds
Minimum RTO	1.0 seconds
Maximum RTO	64 seconds
RTT Gain	0.125
Deviation Gain	0.25
RTT Deviation Coefficient	4.0

The moving area of the mobile nodes within the network in the conducted experiment is defined as 250 square meters. The nodes size used to represents the small-medium and medium sized networks are 10, 15, and 20, and these network sizes reflect the small-medium and medium enterprises. Since the traffic pattern are not investigated in this study, so instead of using the Random Waypoint (Record Trajectory) mobility model, Random Waypoint (Auto Create) mobility model is used for configuration of the node speeds. The node mobility related parameters are defined by OPNET Modeler automatically for the MANET nodes. The node speeds used in this simulation are 3, 4, 5, 6 and 7 meters per second (m/s) to reflect the people movement in the office. The node speed of 3, 4 and 5 m/s reflect the people walk slowly and quickly in the office, and the node speeds of 6 and 7 m/s reflects the people running in the office. The generated traffic type within the network is File Transfer Protocol (FTP). In the MAC layer, the rate of data frames transmission via physical layer is 11Mbps. For traffic load investigation, the lengths of the packet used in this study are 5,000, 10,000, 15,000, 20,000 and 25,000 bytes, where 5,000 bytes packet length reflects the general traffic load, the packet length with 10,000 and 15,000 bytes represents the medium traffic load, and 20,000 and 25,000 bytes packet length reflects the heavy traffic load. The good wireless channel and noisy wireless channel are used for the investigation of performance impact on wireless channel conditions. For each scenario, five runs of simulations have been conducted for one setting. The simulation duration of all the scenarios is 30 minutes.

5.5 Description of Scenarios

The experiment is duplicated and the TCP variants would be changed for the performance comparison and analysis purpose. The evaluation experiments on both TCP-WELCOME and Enhanced TCP-WELCOME are placed in a scenario.

Network Size under Good Wireless Channel

Scenario 1

Scenarios 1 consists of eight experiments, four experiments implemented TCP-WELCOME and Enhanced TCP-WELCOME respectively. The network size with the node number of 5, 10, 15 and 20 are used to investigate the TCP performance under the good wireless channel impact on small, small-medium and medium network sizes. The node speed is configured as 5 m/s on each mobile node, and the generated FTP traffic is with packet length of 5,000 bytes. Increasing the network size would higher the possibility of network congestion occurrences. The performance results will be measured and analyzed through throughput, retransmission counts and end-to-end delay to identify the TCP variant that achieves better performance optimization with varying network sizes.

Node Mobility under Good Wireless Channel (Scenario 2 to 4)

Scenario 2

Scenario 2 consists of eight experiments four experiments implemented TCP-WELCOME and Enhanced TCP-WELCOME respectively. The network implemented with 10 mobile nodes to presents the small-medium network. The first two experiments each with a different variant are configured with the node speed of 3 m/s, the seconds two experiments are configured with 4 m/s for the node speed on each node, the third two experiments are configured with the node speed of 6 m/s and the remaining experiments use the node speed of 7 m/s. The packet length of the data traffic is defined as 5,000 bytes. This scenario intends to acquire the evaluation results of both variants and identify the one that can optimize TCP performance under good wireless channel in the small-medium sized network through the comparative analysis.

Scenario3

Scenario 3 also consists of eight experiments and four experiments implemented TCP-WELCOME and Enhanced TCP-WELCOME respectively. To investigate the TCP performance impact on varying node mobility under good wireless communication channel in the slightly bigger small-medium sized network, the node size of 15 nodes is implemented in the network. The first two experiments each with a different variant are configured with the node speed of 3 m/s, the seconds two experiments are configured with the node speed of 4 m/s, the node speed of the third two experiments are configured as 6 m/s and the remaining experiments use the node speed of 7 m/s. The packet size of the FTP traffic is kept constant at 5,000 bytes. This scenario aims to identify the TCP variant that is able to achieve better performance through the comparative analysis based on the simulation results.

Scenario 4

There are also eight experiments consist in Scenario 4 and four experiments implemented TCP-WELCOME and Enhanced TCP-WELCOME respectively. The medium sized network is with the node size of 20 nodes, and the packet length of FTP traffic is kept constant at 5,000 bytes. The first two experiments each with a different variant are configured with the node speed of 3 m/s, the node speed of the seconds two experiments are configured as 4 m/s, the third two experiments are configure with the node speed of 6 m/s and the remaining experiments use the node speed of 7 m/s. This scenario intends to distinguish the variant that can improve better TCP performance with varying node speed under good wireless communication channel in the medium sized network, the simulation results would be evaluated and analyzed through the performance metrics.

Traffic Load under Good Wireless Channel (Scenario 5 to 7)

Scenario 5

Scenario 5 consists of eight experiments; four experiments implemented TCP-WELCOME and Enhanced TCP-WELCOME respectively. The small-medium sized network is presented by employing 10 nodes in the network, and the node speed is configured as 5 m/s. The first two experiments each with a different variant are configured with the packet length of 10,000 bytes, the packet length of the second two experiments uses 15,000 bytes, the third two experiments are set with the packet length of 20,000 bytes and the remaining experiments are configured with the packet length of 25,000 bytes. This scenario intends to identify the variant that can outperform under good communication channel in the small-medium network with vary traffic load through the performance analysis based on the simulation results.

Scenario 6

Similar to Scenario 5, there are eight experiments consists in this scenario. The slightly bigger small-medium network is implemented with the node size of 15 nodes, and the node speed is configured as 5 m/s constantly on each node. The first two experiments each with a different variant are configured with the packet length of 10,000 bytes, the second two experiments uses 15,000 bytes for the packet length, the packet length of third two experiments are set as 20,000 bytes and the remaining experiments are configured with the packet length of 25,000 bytes. This scenario aims to distinguish the variant that can optimize the performance under good communication channel in the small-medium sized network.

Scenario 7

Scenario 7 also consist eight experiments similar to Scenario 5. The network implemented with 20 nodes to present the medium sized network, and the each node is configured with the node speed of 5 m/s. The packet length of the first two experiments each with a different variant are configured as 10,000 bytes, the second two experiments sets the packet length with 15,000 bytes, the packet length of third two experiments used is 20,000 bytes and the remaining experiments are configured with the packet length of 25,000 bytes. This scenario intends to identify the variant that can outperform under the good wireless channel in the medium sized network.

Network Size under Noisy Wireless Channel

Scenario 8

Scenarios 8 consists of eight experiments, four experiments implemented TCP-WELCOME and Enhanced TCP-WELCOME respectively. The network size with the node number of 5, 10, 15 and 20 are used to investigate the TCP performance under the good wireless channel impact on small, small-medium and medium network sizes. The node speed is configured as 5 m/s on each mobile node, and the generated FTP traffic is with packet length of 5,000 bytes. Contrary to Scenario 1, this scenario aims to identify the variant that can achieve better performance under the noisy wireless channel with varying network sizes through the comparative analysis based on the experimental results.

Node Mobility under Noisy Wireless Channel (Scenario 9 and 10)

Scenario 9

Similar to Scenario 8, there are eight experiments consist in Scenario 9. The small-medium sized network is presented by implementing 10 nodes in the network, and the packet length of FTP traffic is kept constant at 5,000 bytes. The first two experiments each with a different variant are configured with the node speed of 3 m/s, the seconds two experiments are configured with the node speed of 4 m/s, the node speed of the third two experiments are configured as 6 m/s and the remaining experiments use the node speed of 7 m/s. The only difference compare with Scenario3 is that the wireless channel is assumed to be noisy in this scenario. This scenario intends to distinguish the variant that can outperform under the noisy communication channel with varying node speed.

Scenario 10

Scenario 10 consists of eight experiments; four experiments implemented TCP-WELCOME and Enhanced TCP-WELCOME respectively. There are 15 nodes implemented in the network to show the slightly bigger small-medium network, and the transmission data packet is configured as 5,000 bytes constantly. The first two experiments each with a different variant are configured with the node speed of 3 m/s, the node speed of the seconds two experiments are configured as 4 m/s, the third two experiments are configured with the node speed of 6 m/s and the remaining experiments use the node speed of 7 m/s. Contrary to Scenario 4, this scenario aims to obtain the performance results under the noisy wireless channel with varying node speed and identify the variant that is able to optimize the performance through the comparative analysis.

Traffic Load under Noisy Wireless Channel (Scenario 11 and 12)

Scenario 11

Scenario 11 consists of eight experiments; four experiments implemented TCP-WELCOME and Enhanced TCP-WELCOME respectively. The node size implemented in the network is 10 nodes to presents the small-medium sized network, and the FTP packet length is kept constant at 5,000 bytes. The first two experiments each with a different variant are configured with the packet length of 10,000 bytes, the packet length of the second two experiments uses 15,000 bytes, the third two experiments are set with the packet length of 20,000 bytes and the remaining experiments are configured with the packet length of 25,000 bytes. Different from Scenario 7, this scenario intends to investigate the performance under the noisy communication channel with varying traffic load, and identify the variant that can achieve better performance through the comparative analysis based on the simulation results.

Scenario 12

There are also eight experiments consist in Scenario 12. There are 15 nodes implemented in the network to show the slightly bigger small-medium network, and the packet length of FTP traffic is 5,000 bytes. The first two experiments each with a different variant are configured with the packet length of 10,000 bytes, the second two experiments uses 15,000 bytes for the packet length, the packet length of third two experiments are set as 20,000 bytes and the remaining experiments are configured with the packet length of 25,000 bytes. Contrary to Scenario 8, this scenario aims to distinguish the variant that can outperform under the noisy communication channel with varying traffic load according to the comparative analysis conducted from the acquired experimental results.

5.6 Summary

The experimental designed and the parameters specified are discussed in detail in this chapter. The performance metrics used for measuring the evaluation results are throughput, end-to-end delay and retransmission attempts. In order to investigate the TCP performance impact on network size, traffic load, node mobility and wireless channel conditions, the scenarios are designed careful where only one control parameter is changed in each scenario, including the number of nodes, packet length, node speed and noise figure. The experimental results and performance analysis are presented in Chapter 6.

Chapter 6

Results and Analysis

6.1 Introduction

This chapter provides the experimental results and TCP performance analysis of the specified scenarios presented in the Chapter 4. Section 6.2 outlines the evaluation results and comparative analysis based on the defined performance metrics. In Section 6.3, the overall observation and TCP performance are presented. Section 6.4 includes the results validation discussion, and Section 6.5 summarises this chapter.

6.2 Experimental Results

The evaluation of experimental results are classified and categorised into varying node numbers, packet length and node speed. For the data transmission under the good wireless channel, the varied numbers of nodes in small, small-medium and medium sized networks present the results obtained from Scenario 1. The varied node speeds in small, small-medium and medium sized networks present the results collected from Scenario 2 to 4. The varied packet length in small-medium and medium sized networks present the results acquired from Scenario 5 to 7. For the data transmission under the noisy wireless channel, the varied numbers of nodes in small, small-medium and medium sized networks present the results obtained from Scenario 8. The varied node speeds in small, small-medium and medium sized networks present the results collected from Scenario 9 and 10. The varied packet length in small, small-medium and medium sized networks present the results collected from Scenario 11 and 12.

The summary of investigation is shown in Figure 6.1. Is categorised into three different sections, including are performance metrics, scenario and the experiment figure number. The scenarios are grouped into the control parameters of the investigation, including number of nodes (N), packet length (PL) and node speed (NS). The evaluation results can be easily mapping out through the performance metrics and the figure results is listed in the order of performance metrics respectively. For instance, the retransmission attempts results of the two TCP variants with varying node size as defined in Scenario 1 is presented in Figure 6.5.

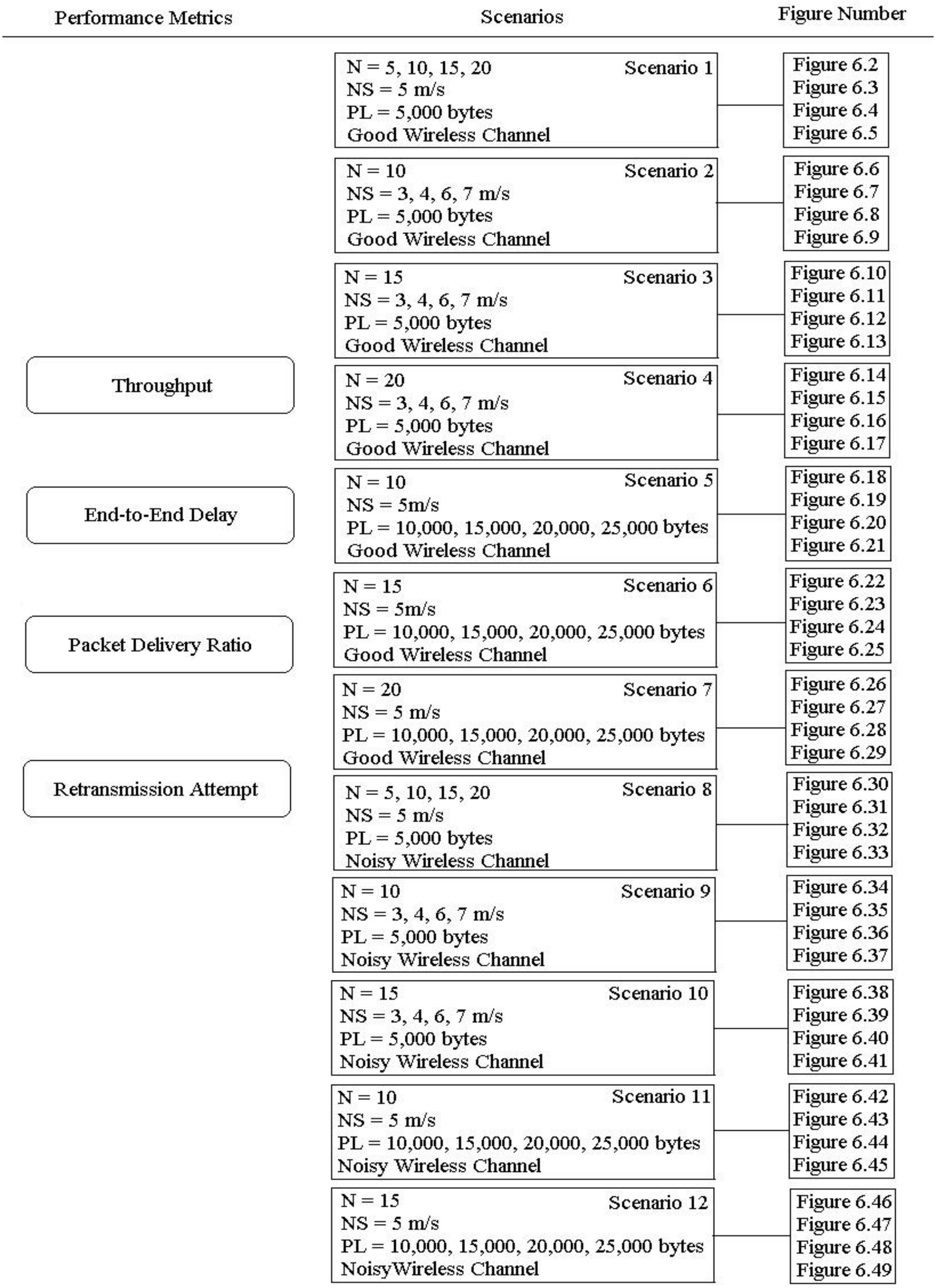


Figure 6.1: Summary of Investigation

6.2.1 Varying Network Size under Good Wireless Channel

The section outlines the experimental results acquired based on Scenario 1. This scenario represents the experimental results of TCP performance impact on varying network sizes under good wireless channel, and the scenario is presented with the different number of nodes (5, 10, 15, 20 mobile nodes), presenting small, small-medium, and medium networks.

Scenario 1 (N = 5, 10, 15 and 20, NS = 5m/s, PL = 5,000 bytes)

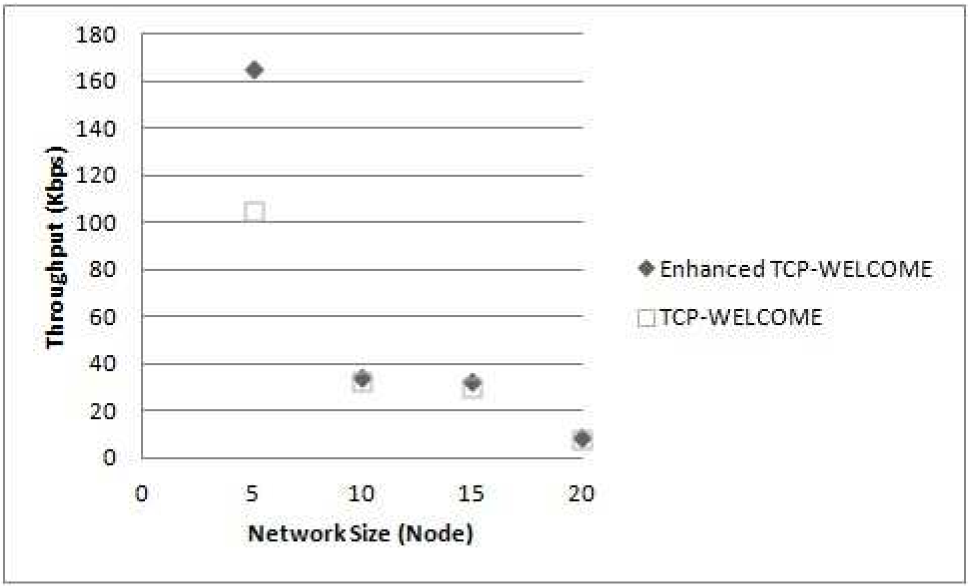


Figure 6.2: Throughput versus varying Network Size

Figure 6.2 compares the throughputs of both the Enhanced TCP-WELCOME and TCP-WELCOME under the good channel in varying network sizes. Since the possibility of congestion occurrences would increase as the number of nodes increasing in the network, so the throughput generated decrease as the network size enlarged. Enhanced TCP-WELCOME was able to offer 20% higher throughputs compare with TCP-WELCOME overall. However, as the node size increases, the differences of generated throughputs are reduced between Enhanced TCP-WELCOME and TCP-WELCOME.

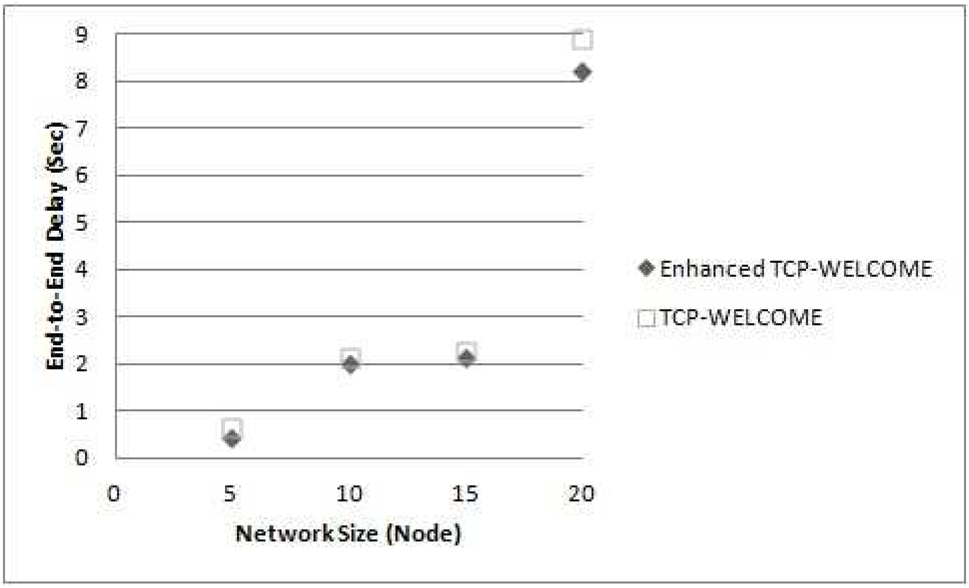


Figure 6.3: End-to-End Delay versus varying Network Size

Figure 6.3 shows the end-to-end delay products of both variants under the good channel. As the number of nodes implemented in the network increases, the network congestion would occur more often hence TCP would introduce longer end-to-end delay during the data transmission. Enhanced TCP-WELCOME achieves 14% lower end-to-end delay than the existing TCP-WELCOME.

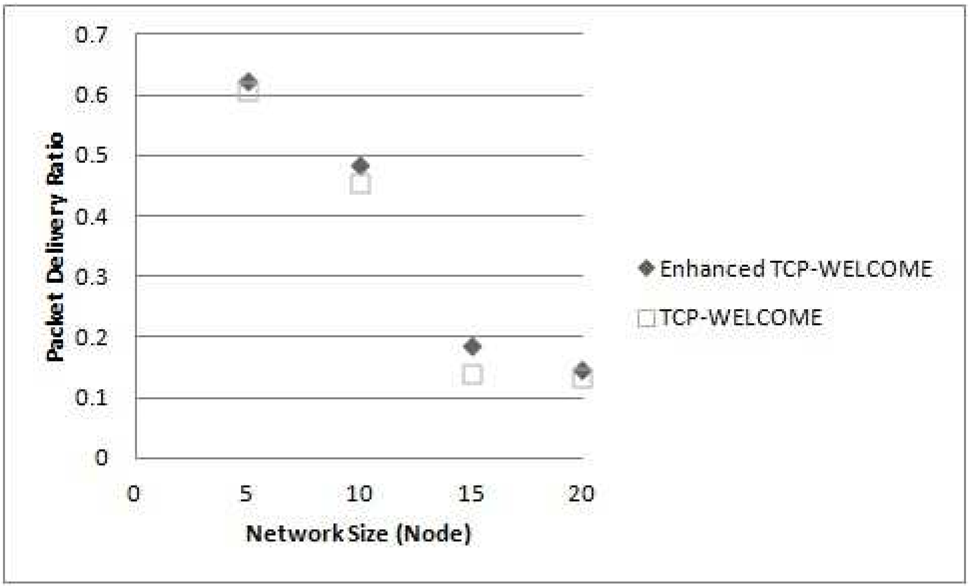


Figure 6.4: Packet Delivery Ratio versus varying Network Size

Figure 6.4 compares the packet delivery ratios of the Enhanced TCP-WELCOME and TCP-WELCOME under the good channel with varying network size. Enhanced

TCP-WELCOME can successfully receive more packets that generated at the application layer in the small-medium and medium sized networks. Enhanced TCP-WELCOME can achieve 13% higher packet delivery ratio. As the node sizes increase, the packet delivery ratio drops due to the congestion and end-to-end delay.

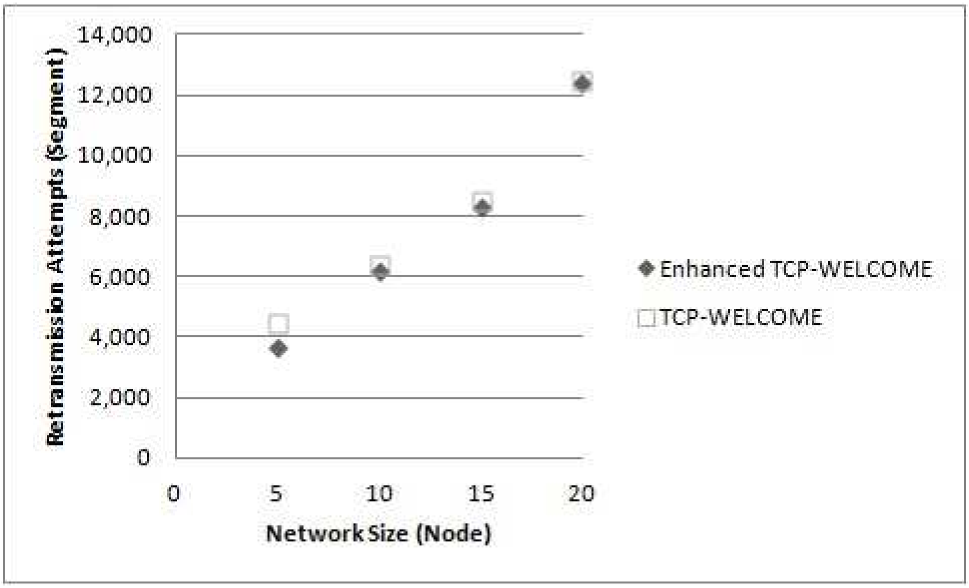


Figure 6.5: Retransmission Attempt versus varying Network Size

Figure 6.5 compares the retransmission attempts of both variants under the good channel in varying sized networks. The packets are retransmitted due to the congestion and Enhanced TCP-WELCOME sent out 6% less discarded packets again to the destination. Therefore, Enhanced TCP-WELCOME is able to outperform in terms of data retransmission with varying network size.

6.2.2 Varying Node Speed under Good Wireless Channel

The section outlines the experimental results acquired based on Scenario 2 to 4. These scenarios represent the experimental results of TCP performance impact on varying node speeds under good wireless channel, and each scenario is presented with the different node speed (3, 4, 5, 6 and 7 m/s), presenting the people movement in the office.

Scenario 2 Small-Medium Network (N = 10, NS = 3, 4, 5, 6 and 7 m/s, PL = 5,000 bytes)

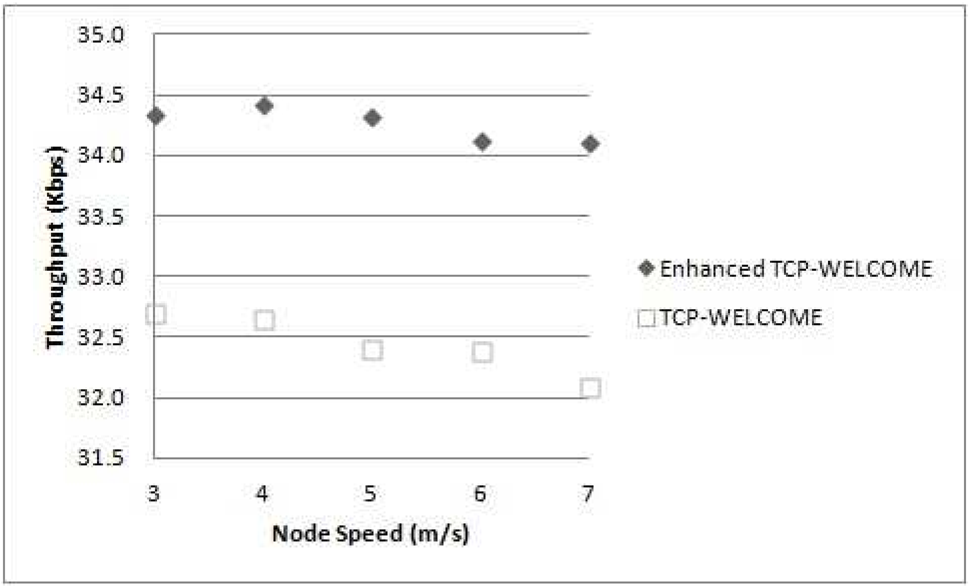


Figure 6.6: Throughput versus varying Node Speed in Small-Medium Network (N = 10)

Figure 6.6 compares the throughputs of the Enhanced TCP-WELCOME and TCP-WELCOME under the good channel in the small-medium network with node size of 15 nodes. Enhanced TCP-WELCOME offers 6% higher throughputs than existing TCP-WELCOME. As shown in Figure 6.6, the significant differences between the generated throughputs shows that Enhanced TCP-WELCOME is able to optimize the data transmission process during the communication in this scenario.

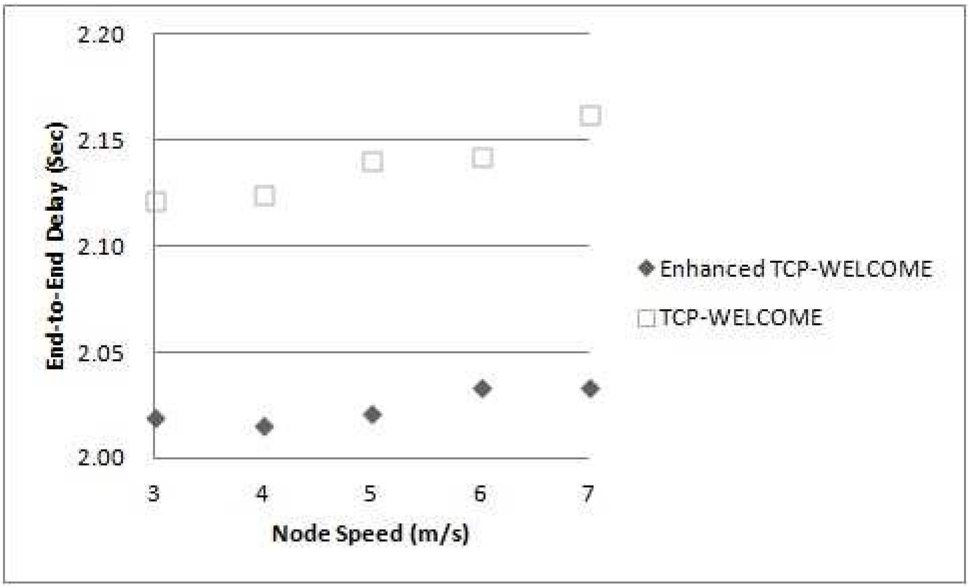


Figure 6.7: End-to-End Delay versus varying Node Speed in Small-Medium Network (N = 10)

Figure 6.7 compares the end-to-end delays of both variants under the good channel in the small-medium network. Enhanced TCP-WELCOME offers 5% lower end-to-end delay than the existing TCP-WELCOME. The lower end-to-end delays achieved by Enhanced TCP-WELCOME are decreasing as the node speed increase. This is because link failure would occur frequently in the high speed node environment, and it would increase the time taken to re-establish a new route for the source to reach the destination which results in longer latency.

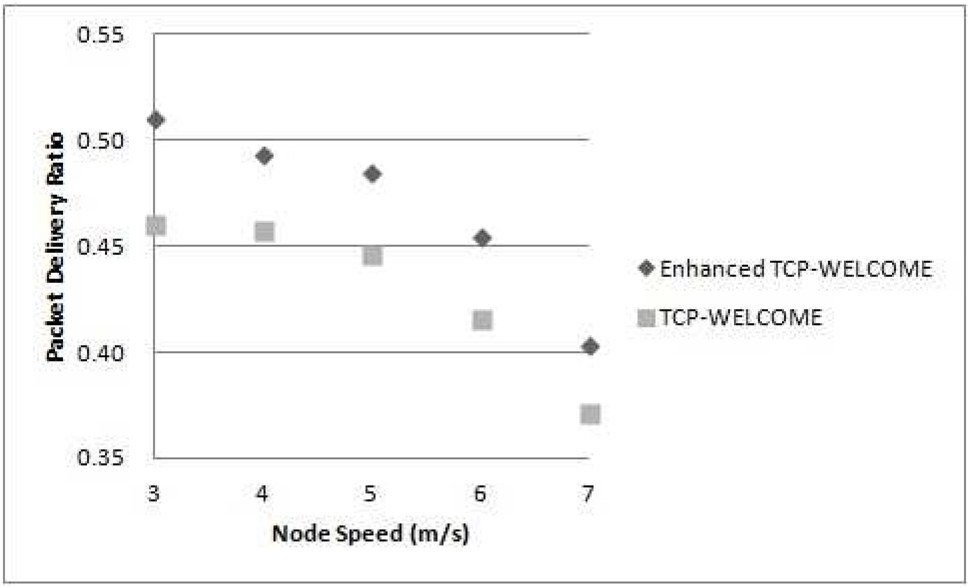


Figure 6.8: Packet Delivery Ratio versus varying Node Speed in Small-Medium Network (N = 10)

Figure 6.8 compares the packet delivery ratios of both variants under the good wireless in the small-medium sized network. Enhanced TCP-WELCOME can successfully receive more amounts of data that generated at the application layer without retransmission required than the existing TCP-WELCOME. Enhanced TCP-WELCOME achieves 9% higher packet delivery rates. The delivery rates decreased as the node speed increases is because the link failure due to the higher node mobility would cause more random packet loss hence the packet delivery ratio would be influenced and reduced.

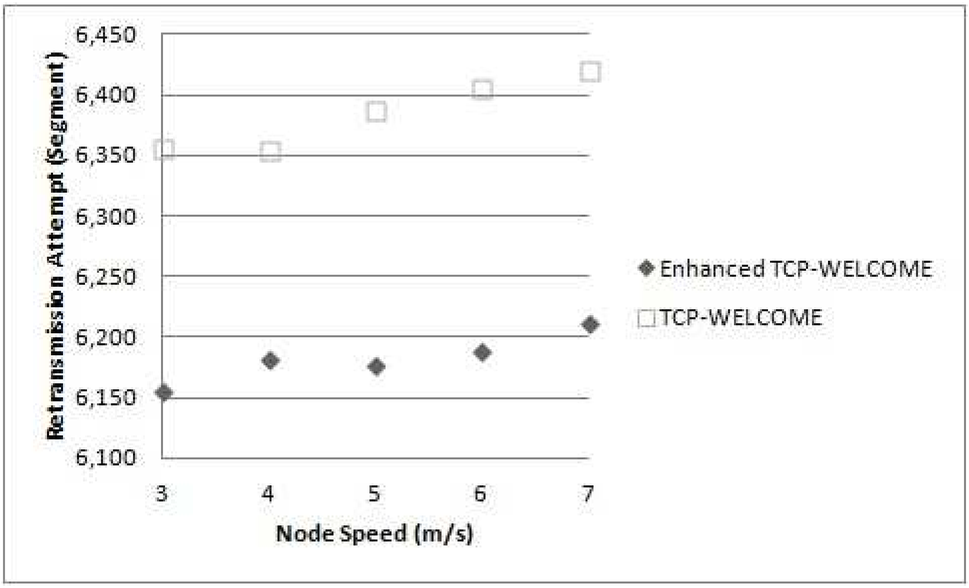


Figure 6.9: Retransmission Attempt versus varying Node Speed in Small-Medium Network (N = 10)

Figure 6.9 compares the data retransmission attempts of the Enhanced TCP-WELCOME and TCP-WELCOME under the good channel with varying node speeds in the small-medium network. Enhanced TCP-WELCOME offers 3% lower retransmission attempt than the existing TCP-WELCOME. Since the link failure would occur more frequent as the node speed increases, Enhanced TCP-WELCOME would require more data retransmission to recover from the packet loss.

Scenario 3 Small-Medium Network (N = 15, NS = 3, 4, 5, 6 and 7 m/s, PL = 5,000 bytes)

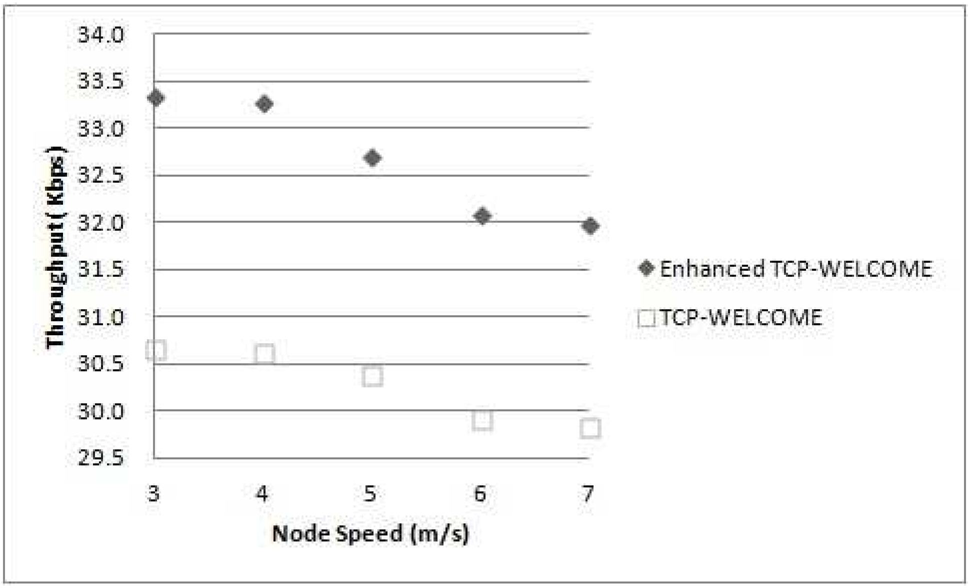


Figure 6.10: Throughput versus varying Node Speed in Small-Medium Network (N = 15)

Figure 6.10 compare the throughputs of both Enhanced TCP-WELCOME and TCP-WELCOME under the good channel with different node speeds in the small-medium network with node size of 15 nodes. Enhanced TCP-WECLOME allows 8% more amount of delivered packets been acknowledged successfully by the receiver than the existing TCP-WELCOME. The link failure would happen more often when the node speed increases, and therefore the throughputs is dropped accordingly.

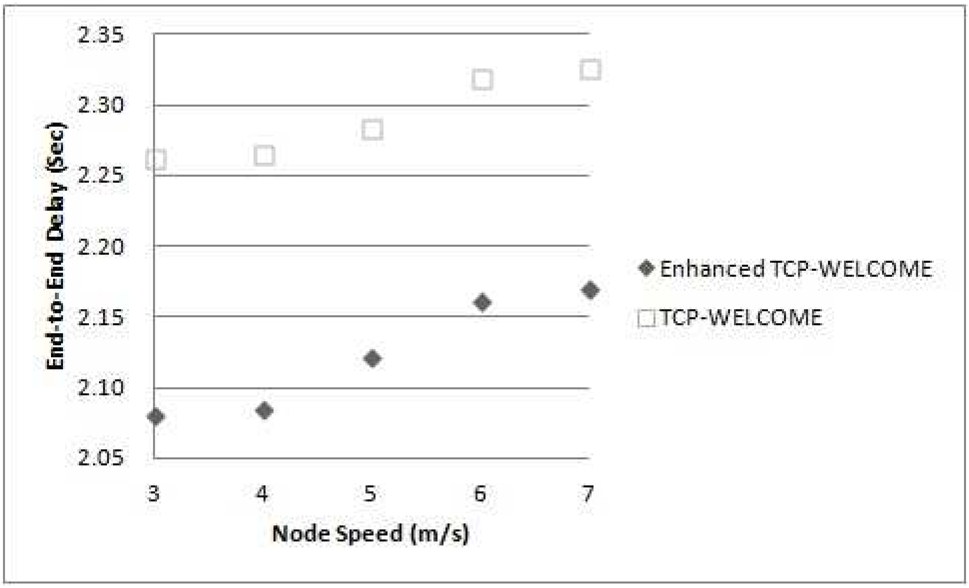


Figure 6.11: End-to-End Delay versus varying Node Speed in Small-Medium Network (N = 15)

Figure 6.11 compares the end-to-end delays of both variants under the good channel in the small-medium network. Enhanced TCP-WELCOME achieves 7% lower end-to-end delay during the data transmission across the network and outperforms TCP-WELCOME. The end-to-end delay increase with the node speed is due to the longer queuing delay caused by link failure.

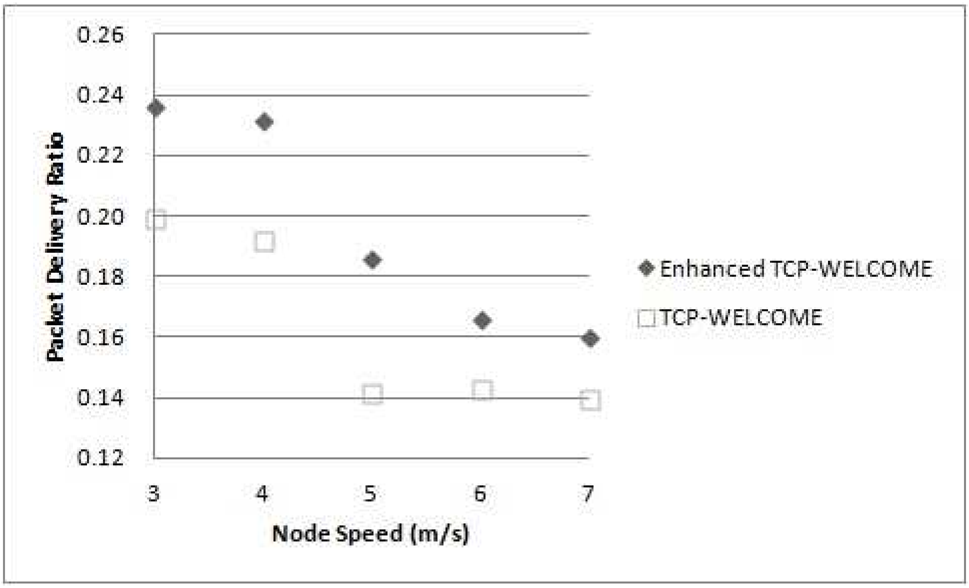


Figure 6.12: Packet Delivery Ratio versus varying Node Speed in Small-Medium Network (N = 15)

Figure 6.12 compares the packet delivery ratios of both variants under the good channel in the small-medium network. Enhanced TCP-WELCOME allows the destination node receives more amount of data generated from the application successfully and achieves 20% higher packet delivery rate than the existing TCP-WELCOME. The higher node mobility results in link failure occurs more during the communication, hence the packet delivery ratio decreases due to packet loss.

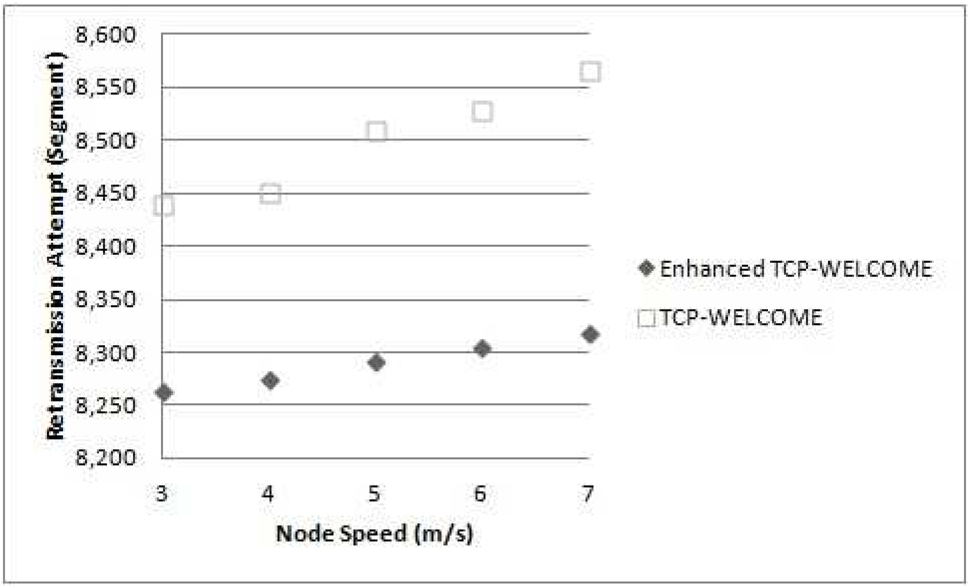


Figure 6.13: Retransmission Attempt versus varying Node Speed in Small-Medium Network (N = 15)

Figure 6.13 compares the retransmission attempts of the Enhanced TCP-WELCOME and TCP-WELCOME under the good channel. Enhanced TCP-WELCOME offers 2% lower retransmission requests to handle the discarded segments. The retransmission attempts increase with the node speed is because the packet drop occurs more often due to link failure.

Scenario 4 Medium Network (N = 20, NS = 3, 4, 5, 6 and 7 m/s, PL = 5,000 bytes)

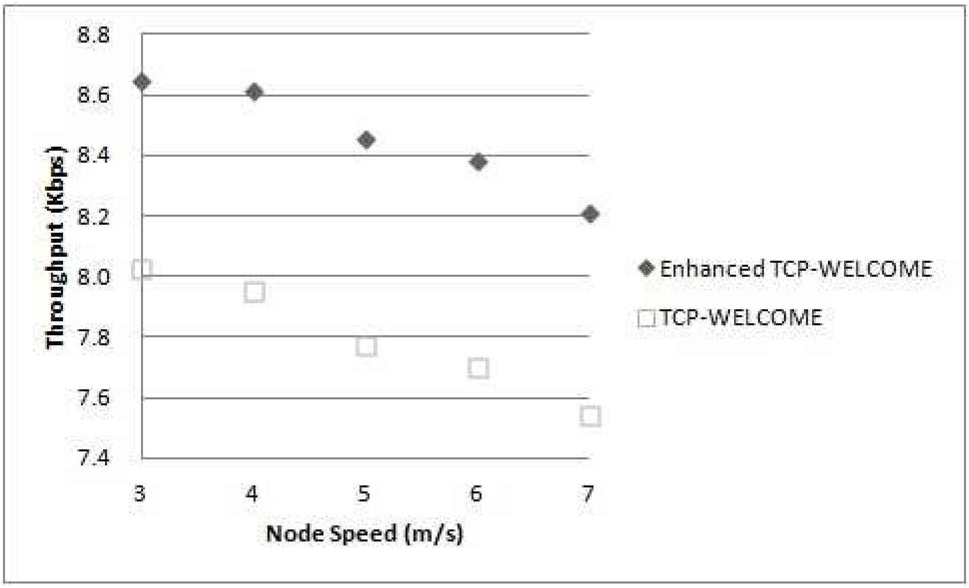


Figure 6.14: Throughput versus varying Node Speed in Medium Network (N = 20)

Figure 6.14 compares the throughputs of both variants under the good channel in the medium network with node size of 20 nodes. Enhanced TCP-WELCOME offers 9% higher throughputs than the existing TCP-WELCOME with varying node speed. Since the link failure frequently occurs in the high node mobility network, and the bigger network size would introduce more traffic load during the data transmission and cause the network congestion, and therefore the throughput generated would decrease with the increasing node speed.

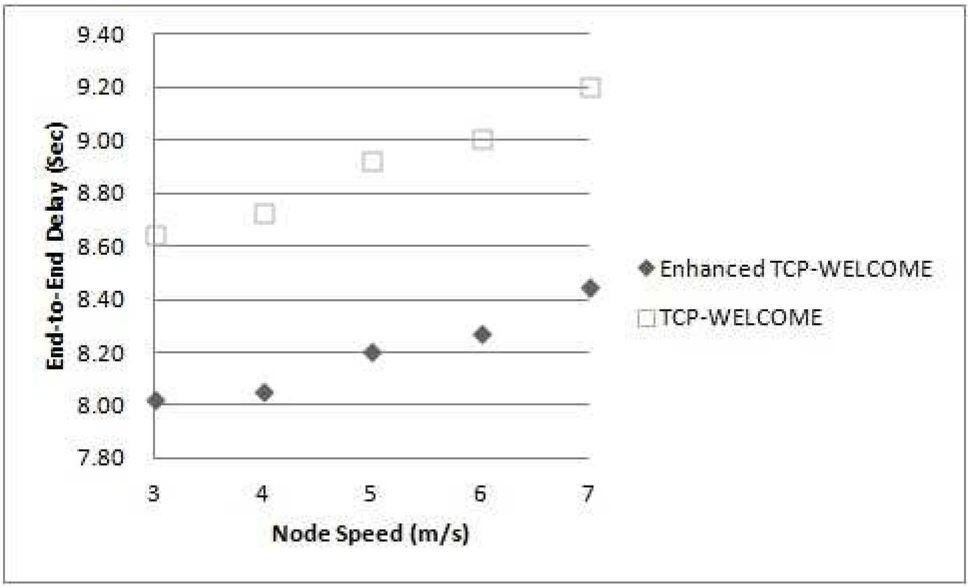


Figure 6.15: End-to-End Delay versus varying Node Speed in Medium Network (N = 20)

Figure 6.15 compares the end-to-end delays of both variants under the good channel with varying node speed in the medium sized network. Enhanced TCP-WELCOME offers 8% lower end-to-end delay overall than the existing TCP-WELCOME. Because the link failure and network congestion would occur more often in the bigger sized network, the end-to-end delay produced with higher node speed would be longer than with lower node speed.

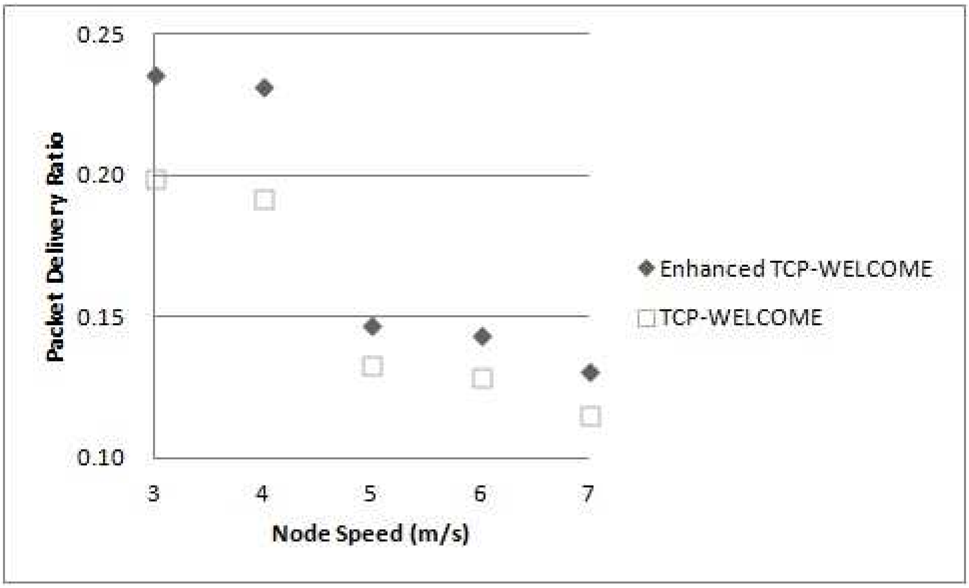


Figure 6.16: Packet Delivery Ratio versus varying Node Speed in Medium Network (N = 20)

Figure 6.16 compares the packet delivery ratios of the Enhanced TCP-WELCOME and TCP-WELCOME under the good channel in the medium network. Enhanced TCP-WELCOME can successfully receive 15% more data generated at application layer than the existing TCP-WELCOME. The achieved packet delivery ratio decreases with the increasing node speed is because the link failure happens more often in the high mobility environment and this results in more packet loss during the data transmission.

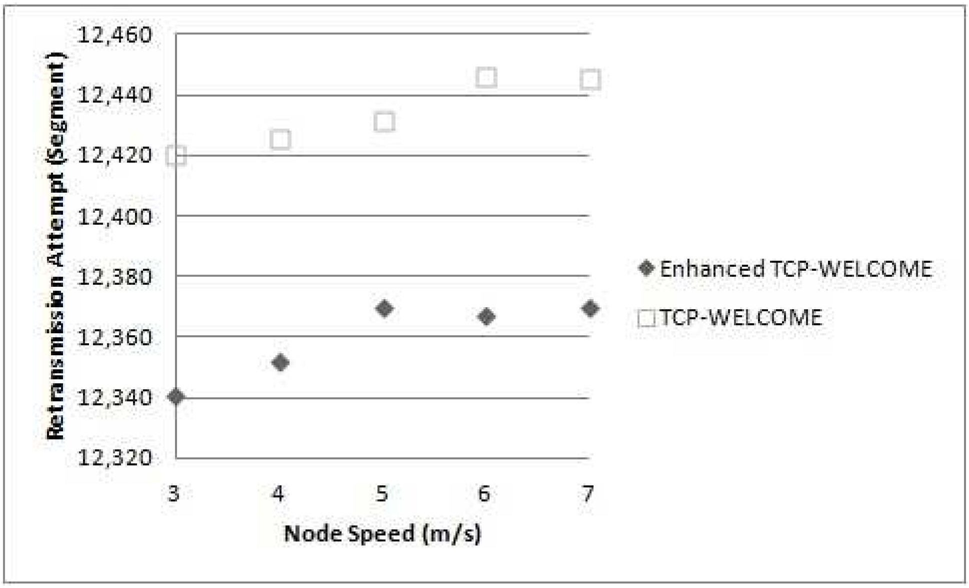


Figure 6.17: Retransmission Attempt versus varying Node Speed in Medium Network (N = 20)

Figure 6.17 compares the retransmission attempts of both variants under the good channel in the medium sized network. Enhanced TCP-WELCOME offers 1% lower retransmission attempt to recover from the discarded packets. Since the link failure and network congestion happens frequently in the bigger sized network with high node mobility, TCP would require more data retransmissions as the node speed increases.

6.2.3 Varying Packet Length under Good Wireless Channel

The section outlines the experimental results acquired based on Scenario 5 to 7. These scenarios represent the experimental results of TCP performance impact on varying traffic load under good wireless channel, and each scenario is presented with the different packet lengths (5,000, 10,000, 15,000 20,000 and 25,000 bytes), presenting the general, medium and heavy traffic load.

Scenario 5 Small-Medium Network (N= 10, NS= 5 m/s, PL= 5,000, 10,000, 15,000, 20,000, 25,000 bytes)

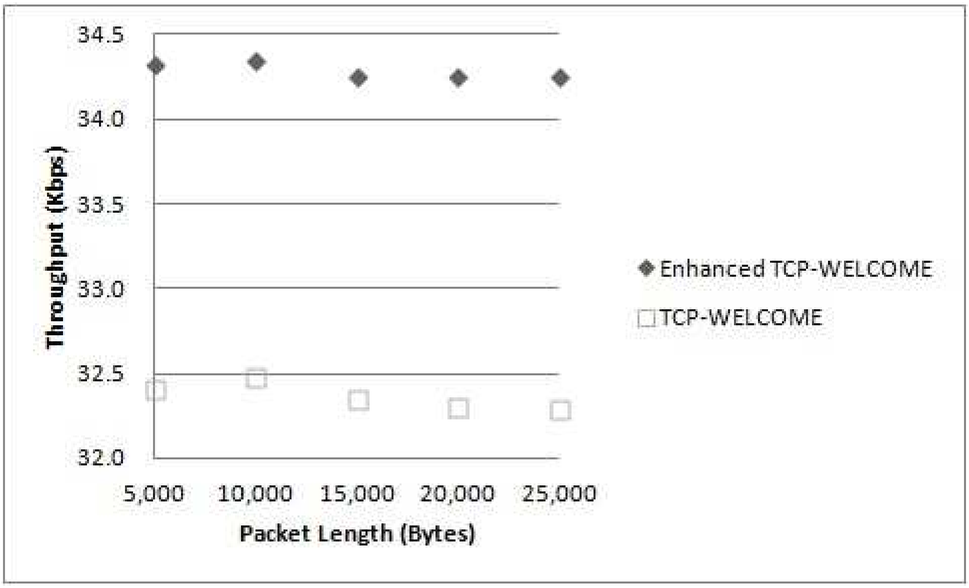


Figure 6.18: Throughput versus varying Packet Length in Small Network (N = 10)

Figure 6.18 compares the throughputs of the Enhanced TCP-WELCOME and TCP-WELCOME under the clean channel with varying packet length in the small-medium network with node size of 10 nodes. Enhanced TCP-WELCOME achieves 6% higher throughputs compare with existing TCP-WELCOME.

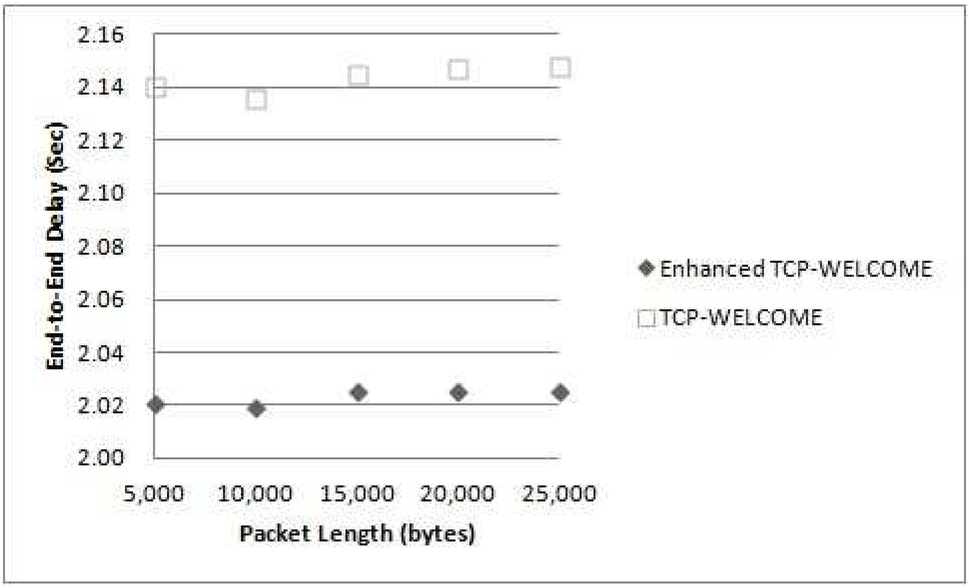


Figure 6.19: End-to-End Delay versus varying Packet Length in Small Network (N = 10)

Figure 6.19 compares the end-to-end delays of both variants under the good link with varying packet lengths. Enhanced TCP-WELCOME offer up to 6% lower end-to-end delay than the existing TCP-WELCOME.

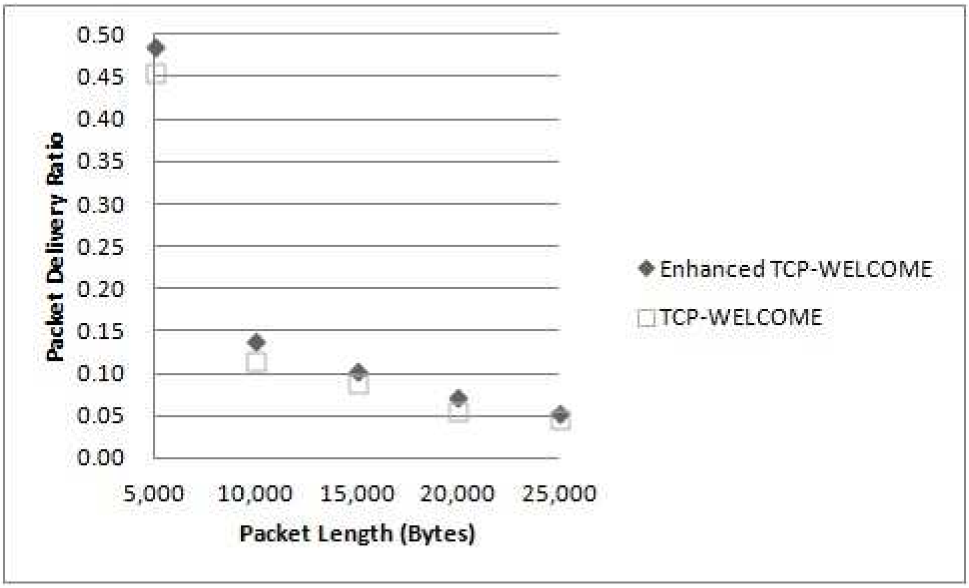


Figure 6.20: Packet Ratio Delay versus varying Packet Length in Small Network (N = 10)

Figure 6.20 compares the packet delivery ratios of both variants under good channel with varying packet length. Enhanced TCP-WELCOME achieves 18% higher delivery ratio which the destination can successfully receive the data generated at the application layer without retransmission required. The larger packet length would introduce more traffic load during the communication and the network congestion occurs more frequently, hence the packet delivery ratio decreases as the packet length increases

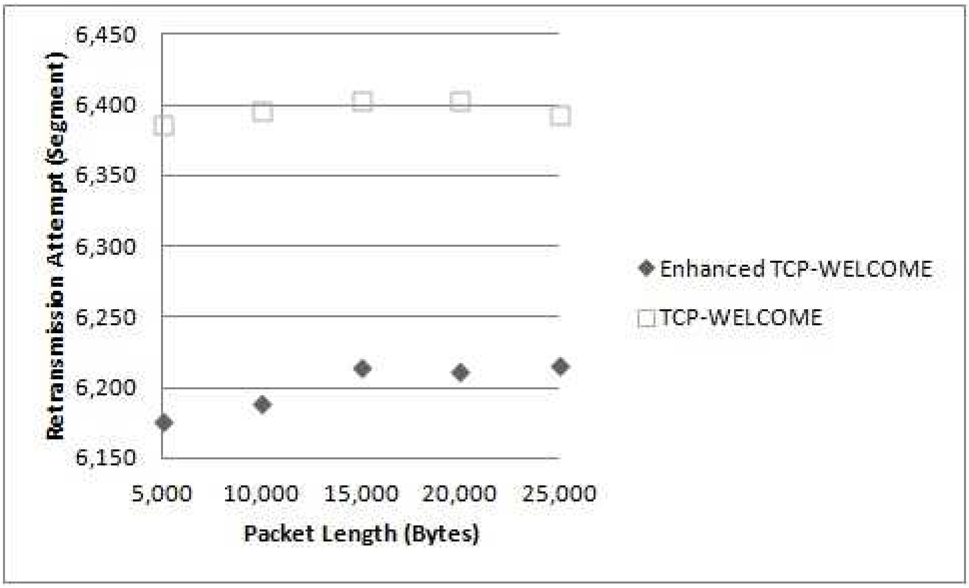


Figure 6.21: Retransmission Attempt versus varying Packet Length in Small Network (N = 10)

Figure 6.21 compares the retransmission attempts of both variants under the clean channel with varying of packet length. Enhanced TCP-WELCOME offers 3% lower retransmission to recover from the discarded packet than the existing TCP-WELCOME. The improvement of retransmission attempt confirms that the proposed recovery mechanism is able to optimize the fast recovery process when handling the packet loss due to network congestion.

Scenario 6 Small-Medium Network (N= 15, NS= 5 m/s, PL= 5,000, 10,000, 15,000, 20,000, 25,000 bytes

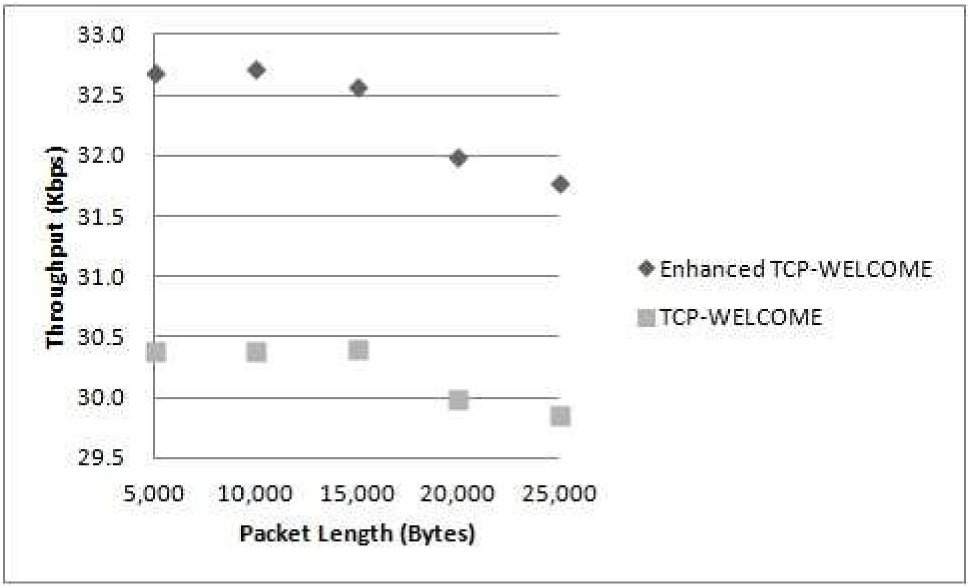


Figure 6.22: Throughput versus varying Packet Length in Small Network (N = 15)

Figure 6.22 compares the throughputs of the Enhanced TCP-WELCOME and TCP-WELCOME under the good channel with varying packet length in the small-medium network with node size of 15 nodes. Enhanced TCP-WELCOME offers 7% higher throughput and outperforms TCP-WELCOME in terms of throughput. The larger packet size would generate more traffic load during the data transmission, hence the the network congestion would occurs more often and degrade the TCP performance.

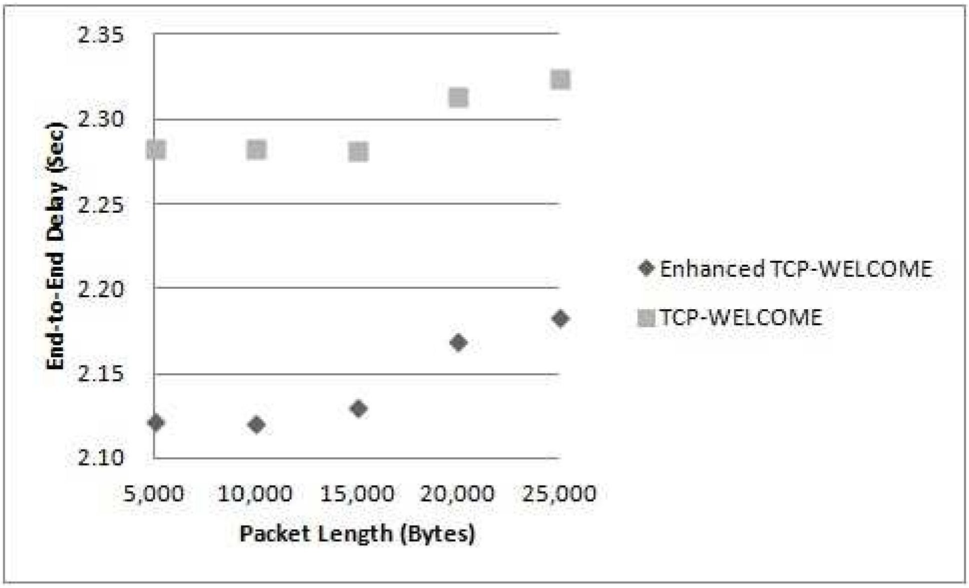


Figure 6.23: End-to-End Delay versus varying Packet Length in Small Network (N = 15)

Figure 6.23 compares the end-to-end delays of both variants under the good channel with varying packet lengths. Enhanced TCP-WELCOME offers 7% lower end-to-end delay than the existing TCP-WELCOME. The end-to-end delay is increased with the packet length is because the network congestion introduce the longer queuing delay at the sender buffer.

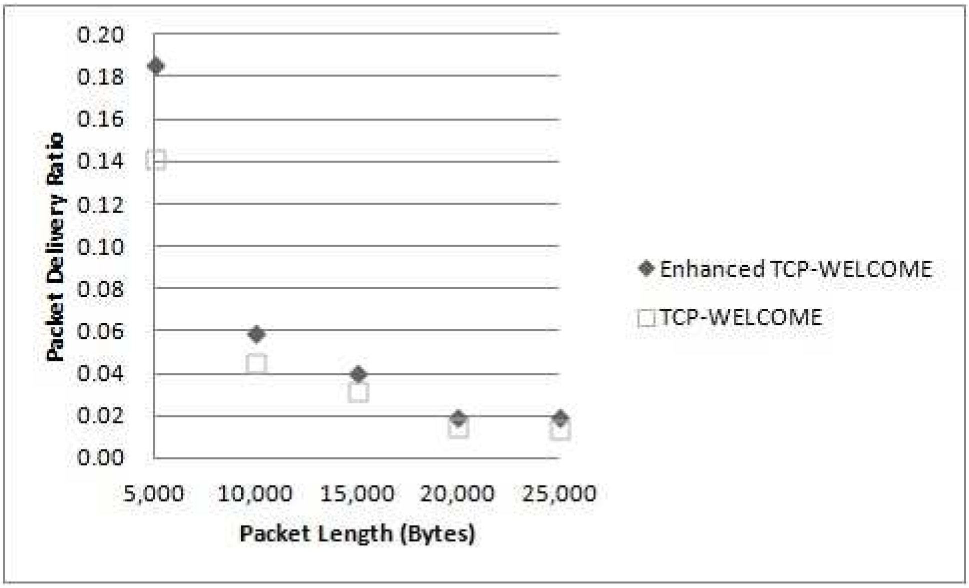


Figure 6.24: Packet Delivery Ratio versus varying Packet Length in Small Network (N = 15)

Figure 6.24 compares the packet delivery ratios of both variants under the good link with varying packet length. Enhanced TCP-WELCOME offers 33% higher packet delivery ratio overall. However, since there would be more traffic load generated over the connection with larger packet size and cause more network congestion, so the packet delivery ratio decreases as the packet length increases.

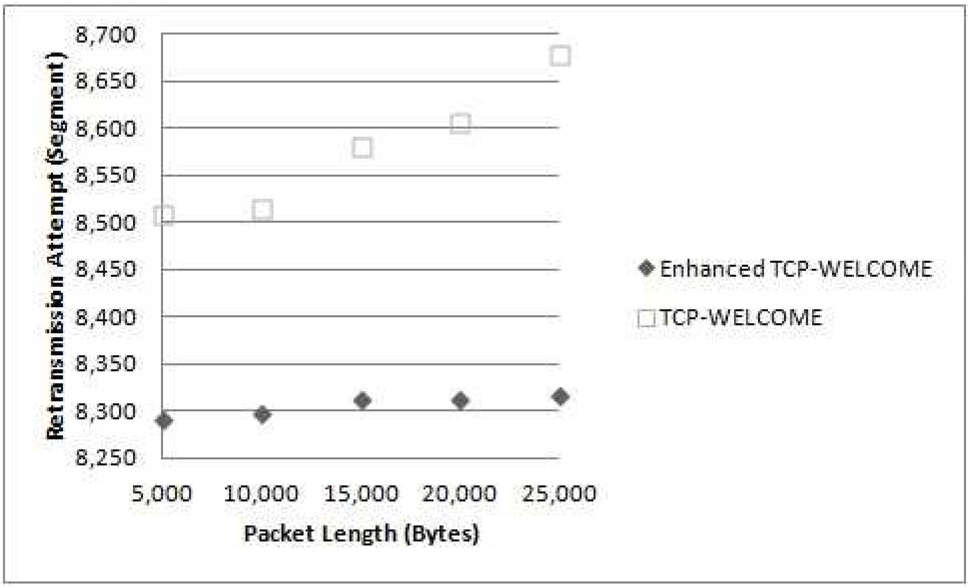


Figure 6.25: Retransmission Attempts versus varying Packet Length in Small Network (N = 15)

Figure 6.25 compares the retransmission attempts of both variants under the good channel with varying packet lengths in the small-medium network. Enhanced TCP-WELCOME achieves 3% lower retransmission attempts to handle the discarded segments during the communication and outperforms TCP-WELCOME. The heavier traffic load with larger packet size would make the network becomes more congested, and therefore it requires more retransmission for the lost data.

Scenario 7 Medium Network (N= 20, NS= 5 m/s, PL= 5,000, 10,000, 15,000, 20,000, 25,000 bytes)

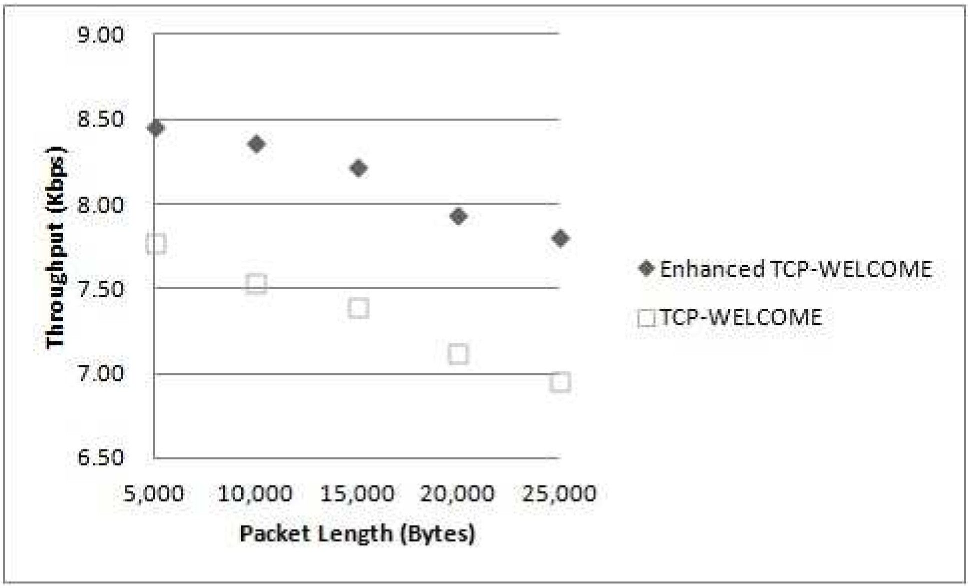


Figure 6.26: Throughput versus varying Packet Length in Small Network (N = 20)

Figure 6.26 compares the throughputs of the Enhanced TCP-WELCOME and TCP-WELCOME under the clean channel with varying packet length in the medium size network with node size of 20 nodes. Enhanced TCP-WELCOME offers 11% higher throughputs and achieves the better performance. The bigger network size and the longer packet length would introduce more traffic load over the connections, consequently the network is congested more frequently and hence the generated throughputs would decrease as the packet length increases.

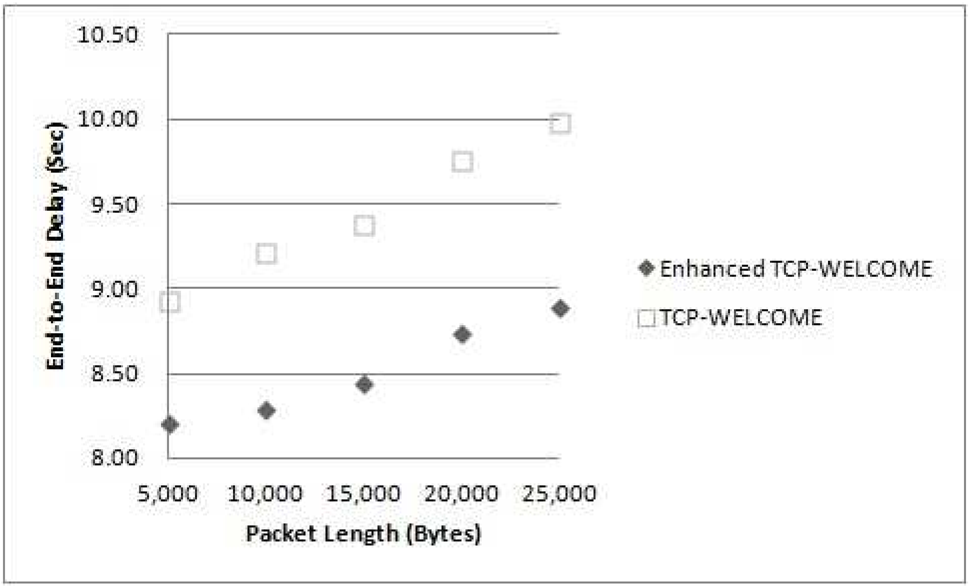


Figure 6.27: End-to-End Delay versus varying Packet Length in Small Network (N = 20)

Figure 6.27 compares the end-to-end delays of both variants under the good channel with varying lengths of packet. The end-to-end delay of TCP-WELCOME is 10% lower than the existing TCP-WELCOME overall. The network congestion occurs more frequently is the consequence of the bigger network size and packet length. Therefore, the end-to-end delay of TCP-WELCOME increases with the packet length increment as shown in Figure 6.27.

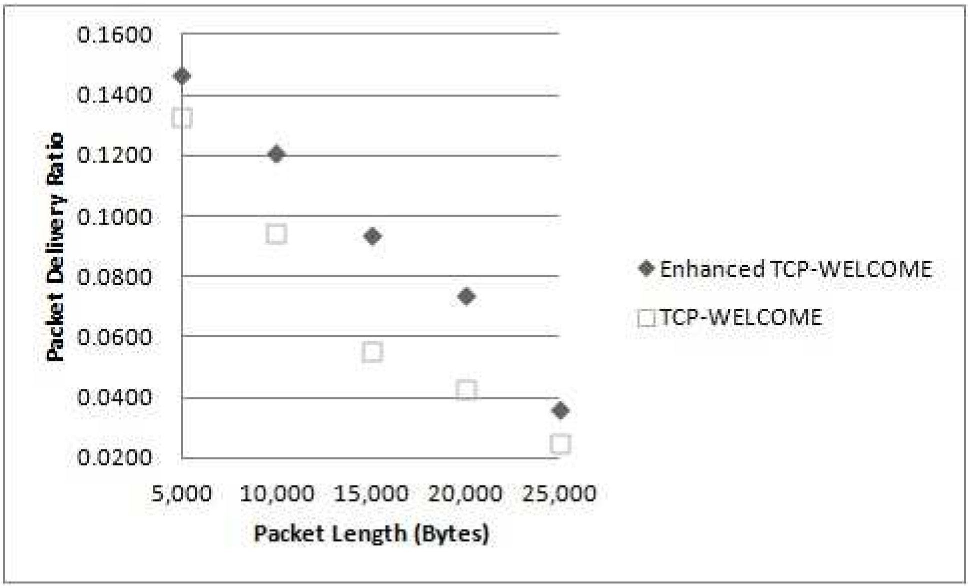


Figure 6.28: Packet Delivery Ratio versus varying Packet Length in Small Network (N = 20)

Figure 6.28 compares the packet delivery ratios of both variants under the clean channel with varying packet lengths. Enhanced TCP-WELCOME achieves 45% higher packet delivery ratio and allows more amount of data been acknowledged by the receiver without retransmission required. However, because of more traffic load generated over the communication channel in the medium sized network, and therefore the packet delivery ratio decreases with the packet length increment.

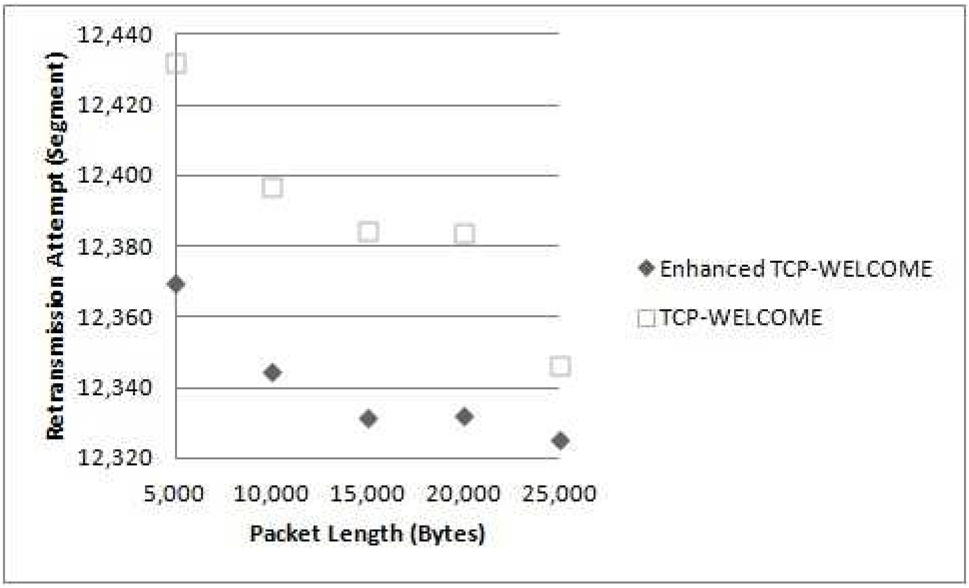


Figure 6.29: Retransmission Attempt versus varying Packet Length in Small Network (N = 20)

Figure 6.29 compares the retransmission attempts of both variants under the good link with varying packet length. Enhanced TCP-WELCOME offers 0.4% lower data retransmission to handle the discarded packet averagely. The decreasing of the retransmission attempt with packet size increment shows that the proposed loss recovery mechanism is able to take more appropriate action to recover from the packet loss due to network congestion.

6.2.4 Varying Network Size under Noisy Wireless Channel

The section outlines the experimental results acquired based on Scenario 8. This scenario represents the experimental results of TCP performance impact on varying network sizes under noisy wireless channel, and the scenario is presented with the different number of nodes (5, 10, 15, 20 mobile nodes), presenting small, small-medium, and medium networks.

Scenario 8 (N = 5, 10, 15 and 20, NS = 5m/s, PL = 5,000 bytes)

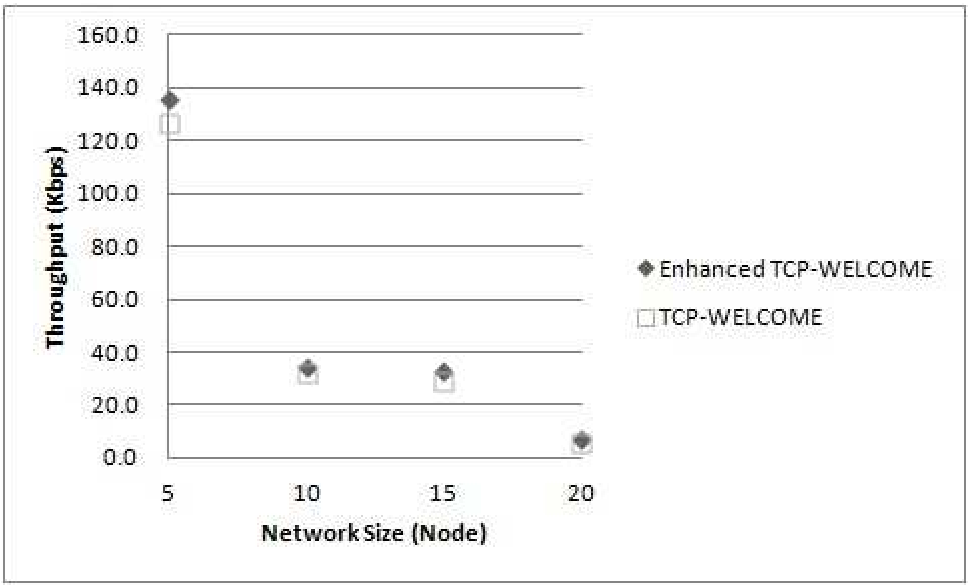


Figure 6.30: Throughput versus varying Network Size under noisy wireless channel

Figure 6.30 compares the throughputs of the Enhanced TCP-WELCOME and TCP-WELCOME under the noisy channel with varying network sizes. Enhanced TCP-WELCOME offers 11% higher throughput than the existing TCP-WELCOME. Since the network congestion occurs more frequently due to more traffic load generated through the communication channel in the bigger network size, therefore the generated throughputs decreases as the network size increases.

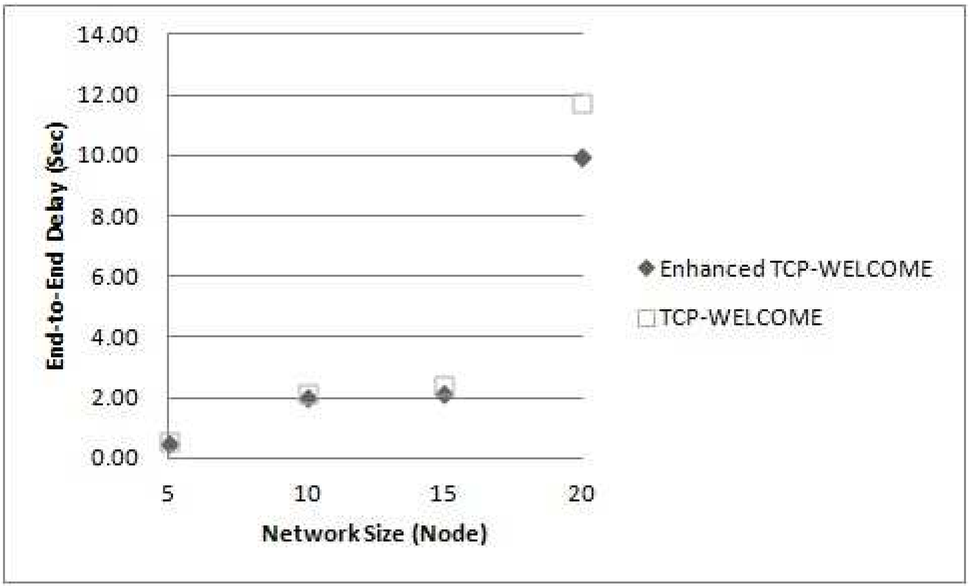


Figure 6.31: End-to-End Delay versus varying Network Size under noisy wireless channel

Figure 6.31 compares the end-to-end delays of both variants under the noisy channel with varying sized networks. Enhanced TCP-WELCOME offers 10% lower end-to-end delay compare with existing TCP-WELCOME. The end-to-end delay increases rapidly is because of the congestion occurs in the network would require longer queuing delay for waiting the packet to be sent out at the buffer.

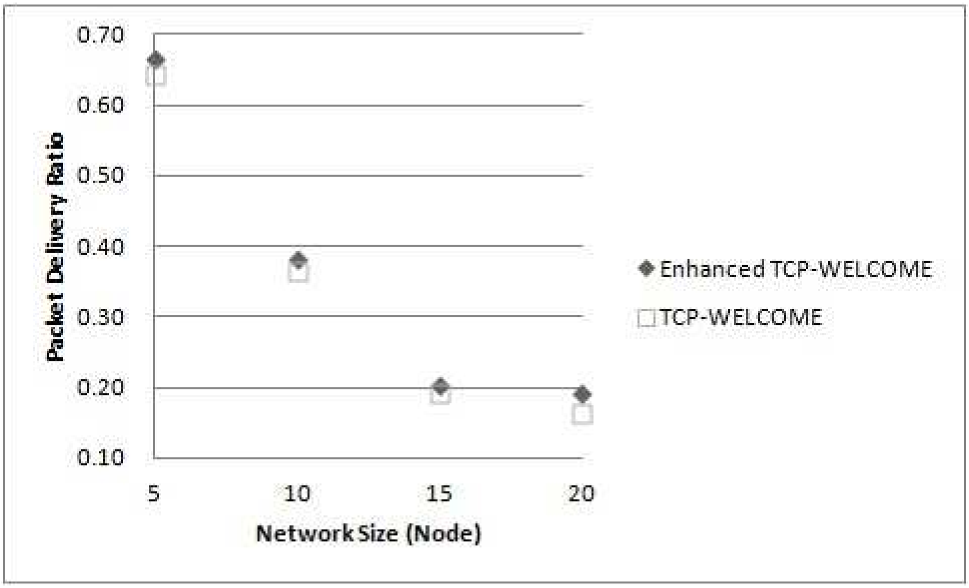


Figure 6.32: Packet Deliver Ratio versus varying Network Size under noisy wireless channel

Figure 6.32 compares the packet delivery ratios of both variants under the noisy link with varying network sizes. Enhanced TCP-WELCOME achieves 8% higher delivery ratio which allows the destination nodes can receive more data generated from application without retransmission required. The packet delivery ratio is influenced and decreases as the network size increases because of the noisy channel and larger network size would generate more traffic load over the connection and discards the packets due to network congestion.

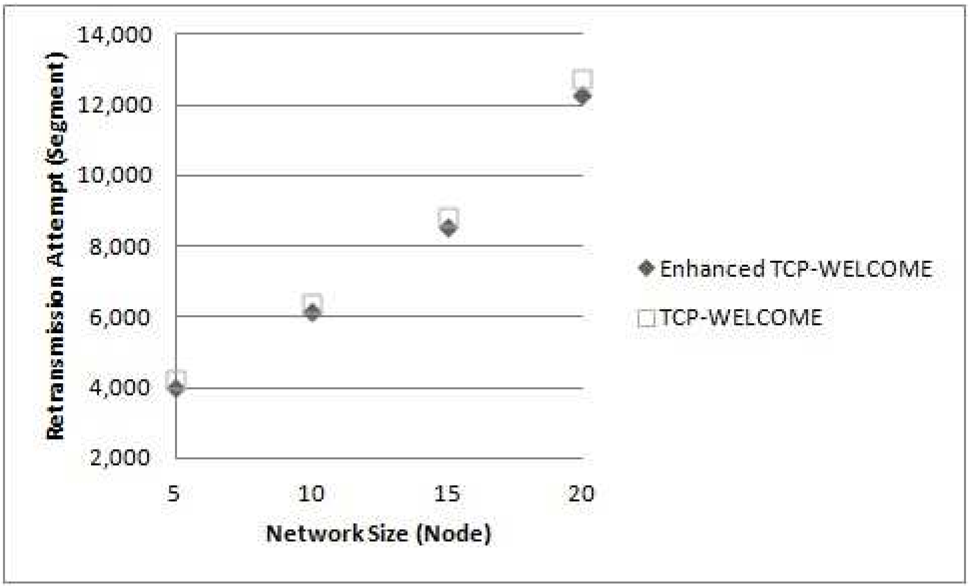


Figure 6.33: Retransmission Attempt versus varying Network Size under noisy wireless channel

Figure 6.33 compares the retransmission attempts of the Enhanced TCP-WELCOME and TCP-WELCOME under the noisy channel in varying network sizes. Enhanced TCP-WELCOME offers 4% lower data retransmission to recover the packet loss due to network congestion. The heavy traffic load due to the noisy channel and bigger network size would results in more network congestion, and therefore the retransmission attempt increases rapidly as the network size enlarges.

6.2.5 Varying Node Speed under Noisy Wireless Channel

The section outlines the experimental results acquired based on Scenario 9 and 10. These scenarios represent the experimental results of TCP performance impact on varying node speed under noisy wireless channel, and each scenario is presented with the different node speeds (3, 4, 5, 6 and 7 m/s), presenting the people movement in the office.

Scenario 9 Small-Medium Network (N = 10, NS =3, 4, 5, 6 and 7 m/s, PL = 5,000 bytes)

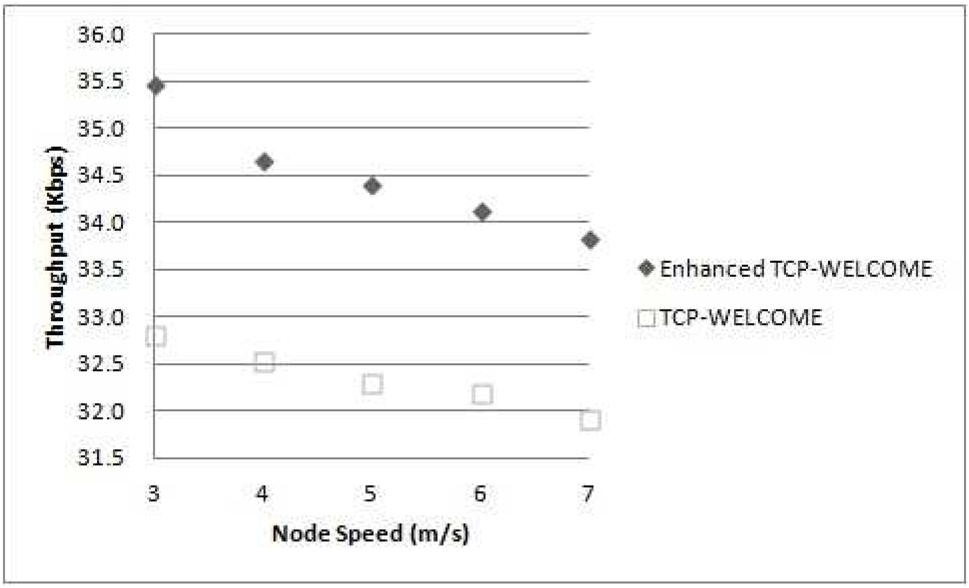


Figure 6.34: Throughput versus varying Node Speed under noisy wireless channel (n = 10)

Figure 6.34 compares the throughputs of the Enhanced TCP-WELCOME and TCP-WELCOME under the noisy channel with varying node speeds. Enhanced TCP-WELCOME achieves 7% more throughputs with significant outperformance as shown in Figure 6.34. Since the link failure would occur more frequent in the higher node speed network environment and the noisy channel would cause the network becomes congested, hence the throughputs decrease with the increasing node speed.

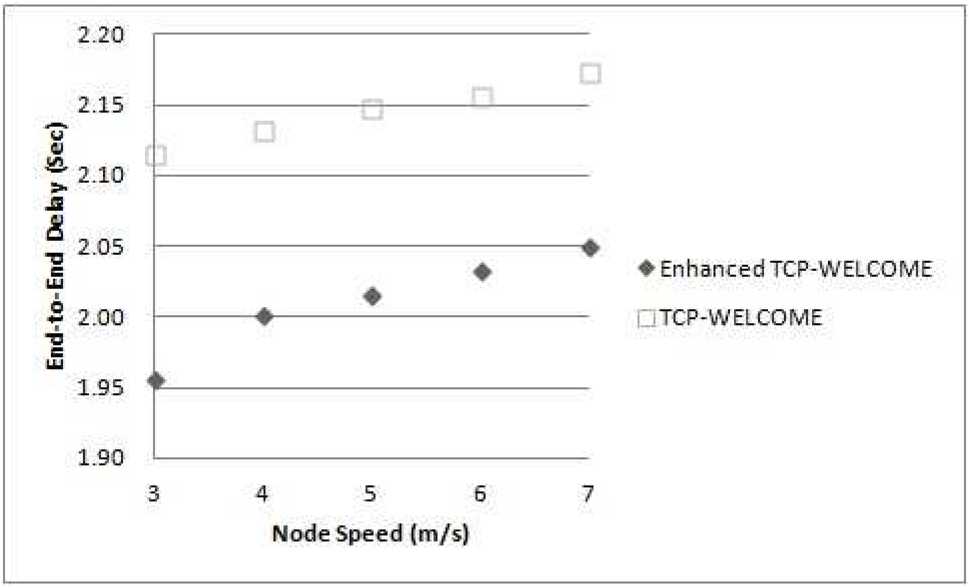


Figure 6.35: End-to-End Delay versus varying Node Speed under noisy wireless channel (n = 10)

Figure 6.35 compares the end-to-end delays of both variants under the noisy channel with varying node speeds. Enhanced TCP-WELCOME offers 6% lower end-to-end delay. The link failure and network congestion would occur more often because of the higher node speed and noisy channel. Consequently, the queuing delay is increased and takes longer for the packet waiting to be sent out at the buffer; hence the end-to-end delay is affected and increased as the node speed increases.

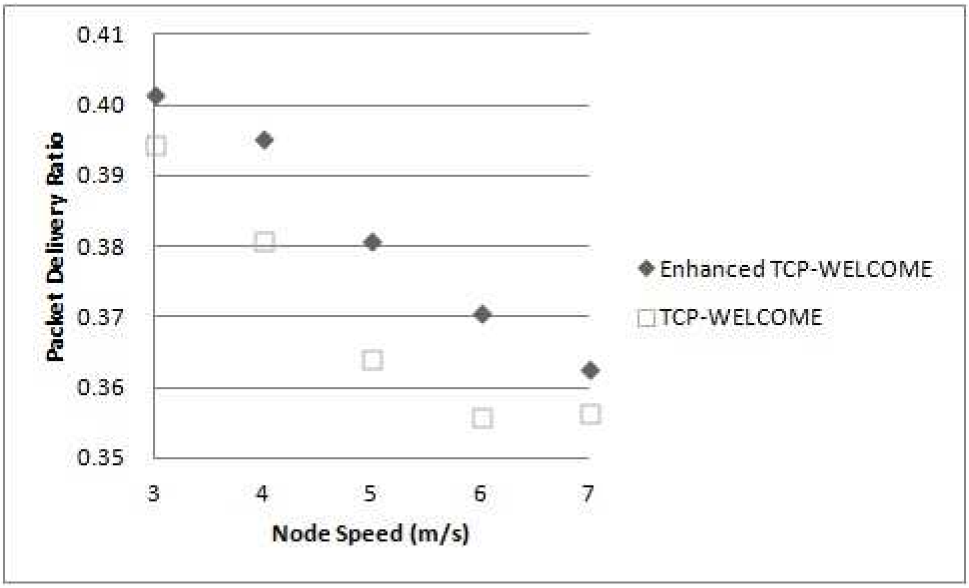


Figure 6.36: Packet Delivery Ratio versus varying Node Speed under noisy wireless channel (n = 10)

Figure 6.36 compares the packet delivery ratios of both variants under the noisy link with varying node speeds. Enhanced TCP-WELCOME achieves 3% higher packet delivery ratio which allows the destination node receiving more data generated from application at the destination nodes successfully without retransmission required. The link failure and network congestion caused by higher mobility and noisy communication link would result in more random packet loss, so the packet delivery ratios decrease with the increasing node speed.

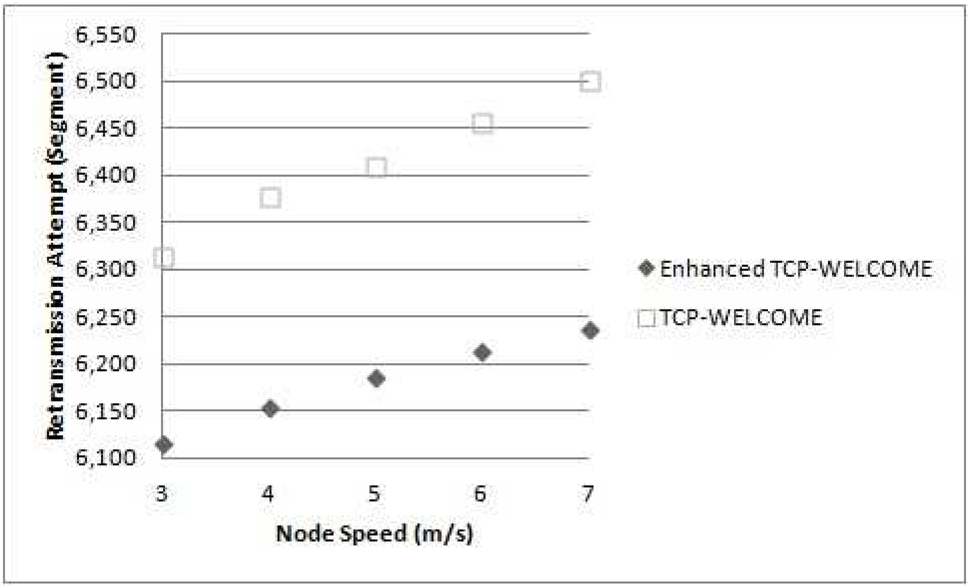


Figure 6.37: Retransmission Attempt versus varying Node Speed under noisy wireless channel (n = 10)

Figure 6.37 compares the retransmission attempts under the noisy link with varying node speeds. Enhanced TCP-WELCOME achieves 4% lower data retransmission attempt to recover from the packet loss. As shown in Figure 6.37, the significant outperformance of Enhanced TCP-WELCOME proves that the proposed loss recovery algorithm takes more appropriate action to handle the packet loss due to link failure and network congestion and improve the performance with respect to retransmission attempt.

Scenario 10 Small-Medium Network (N = 15, NS = 3, 4, 5, 6 and 7 m/s, PL = 5,000 bytes)

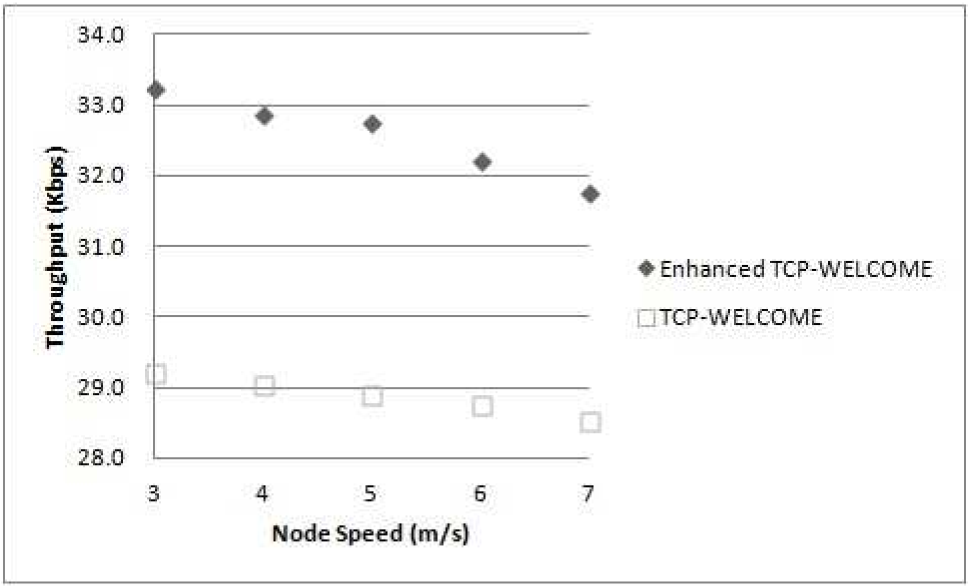


Figure 6.38: Throughput versus varying Node Speed under noisy wireless channel (n = 15)

Figure 6.38 compares the throughputs of the Enhanced TCP-WELCOME and TCP-WELCOME under the noisy channel with varying node speeds in the small-medium sized network with node size of 15 nodes. Enhanced TCP-WELCOME offers 13% higher throughput than the existing TCP-WELCOME. The higher node mobility and noisy wireless channel would cause the link failure and network congestion occur more frequently during the communication, and therefore the generated throughputs decreases as the node speed increases.

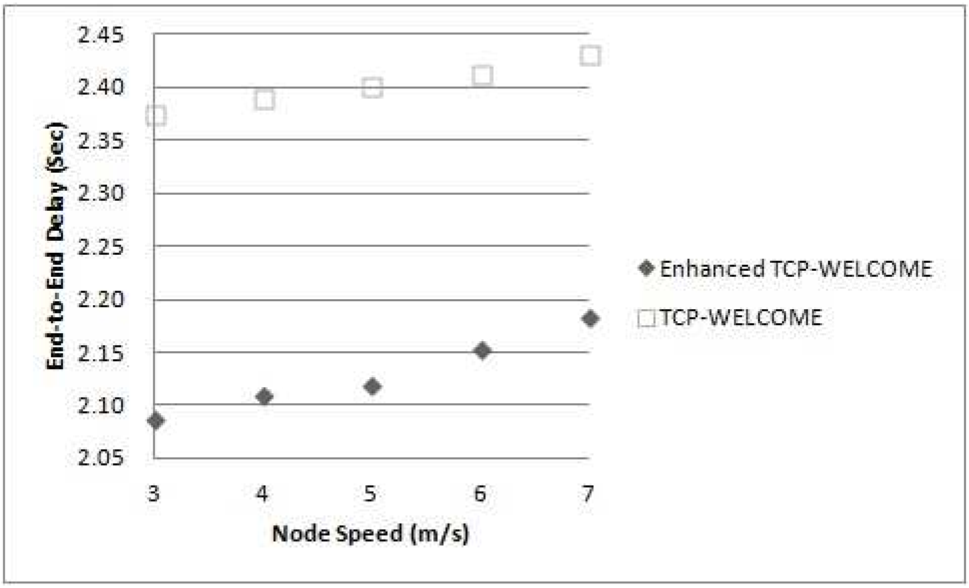


Figure 6.39: End-to-End Delay versus varying Node Speed under noisy wireless channel (n = 15)

Figure 6.39 compares the end-to-end delays of both variants under the noisy wireless channel with varying node speeds. The end-to-end delay of Enhanced TCP-WELCOME is 11% lower than the existing TCP-WELCOME. The link failure and network congestion due to the higher node speed and noisy wireless channel would results in longer queuing delay for the packet waiting to be forwarded at the buffer, and therefore the end-to-end delay increases with the increasing node speed.

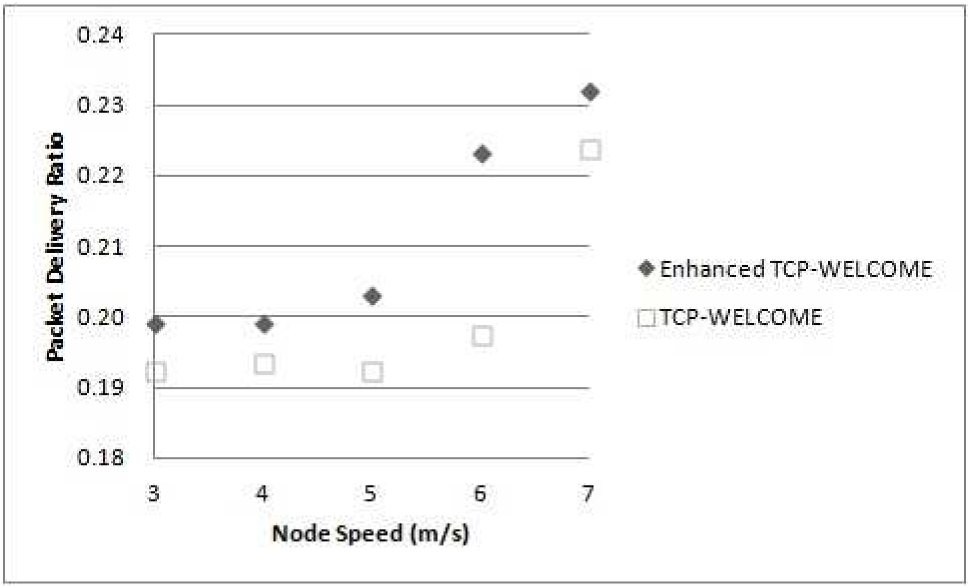


Figure 6.40: Packet Delivery Ratio versus varying Node Speed under noisy wireless channel (n = 15)

Figure 6.40 compares the packet delivery ratios under the noisy link with varying node speeds. Enhanced TCP-WELCOME offers 6% higher packet delivery ratio which the destination node can receive more data generated at application layer without retransmission required than the existing TCP-WELCOME. Moreover, the achieved packet delivery ratio increases with the increasing node speed; this proves that the proposed loss recovery mechanism of link failure and network congestion can perform better with higher node mobility under noisy channel.

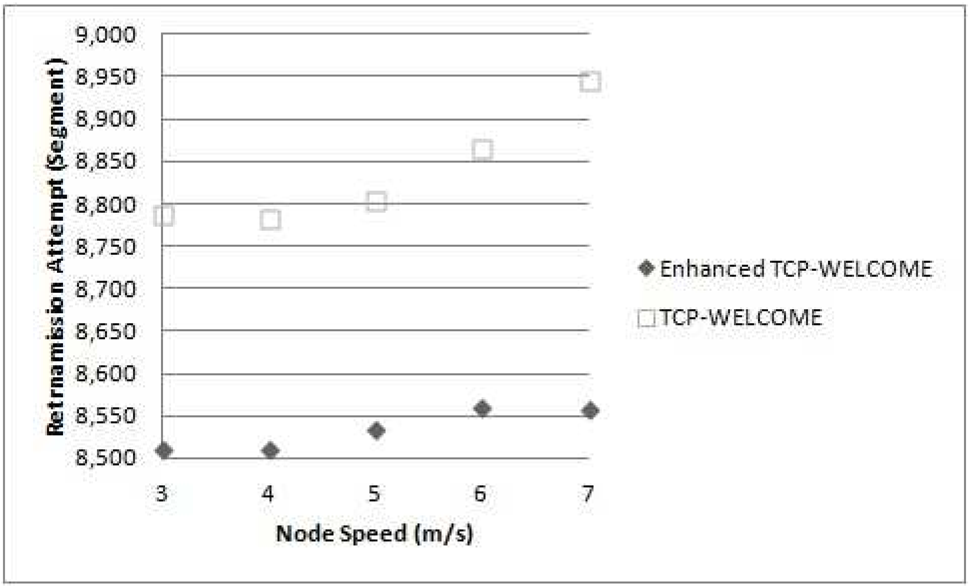


Figure 6.41: Retransmission Attempt versus varying Node Speed under noisy wireless channel (n = 15)

Figure 6.41 compares the retransmission attempts of both variants under the noisy channel with varying node speeds. Enhanced TCP-WELCOME offers 3% lower data retransmission to handle the discarded packet during the communication. The significant outperformance of Enhanced TCP-WELCOME shown in Figure 6.41 confirms that the proposed loss recovery algorithm acts more appropriate to recover from the packet loss due to link failure and network congestion.

6.2.6 Varying Packet Length under Noisy Wireless Channel

The section outlines the experimental results acquired based on Scenario 11 to 12. These scenarios represent the experimental results of TCP performance impact on varying traffic load under good wireless channel, and each scenario is presented with the different packet lengths (5,000, 10,000, 15,000 20,000 and 25,000 bytes), presenting the general, medium and heavy traffic load.

Scenario 11 Small-Medium Network (N= 10, NS= 5 m/s, PL=5,000, 10,000, 15,000, 20,000, 25,000 bytes)

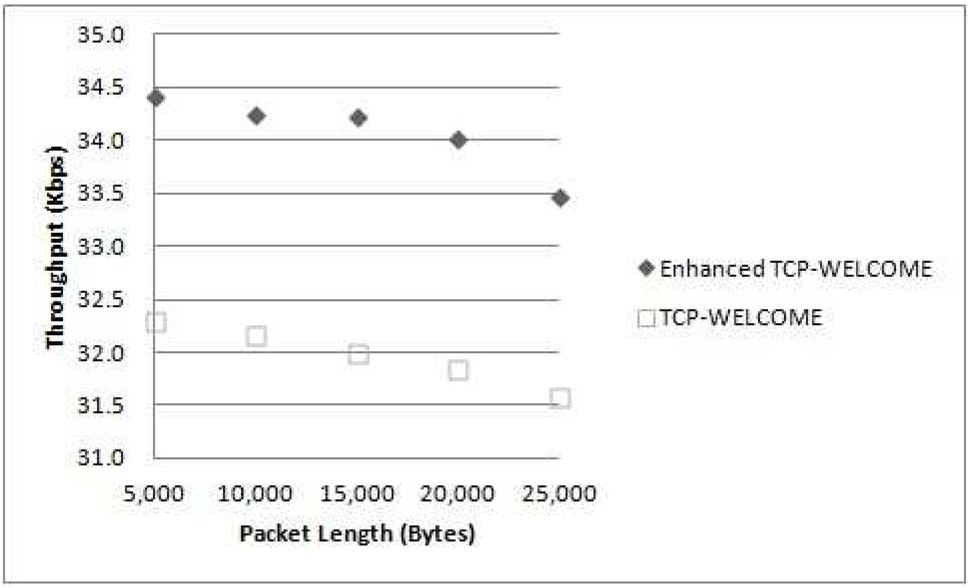


Figure 6.42: Throughput versus varying Packet Length under noisy wireless channel (n = 10)

Figure 6.42 compares the throughputs of the Enhanced TCP-WELCOME and TCP-WELCOME under the noisy channel in the small-medium sized network with nodes size of 10 nodes. Enhanced TCP-WELCOME offers 7% higher throughputs with significant outperformance as shown in Figure 6.42. The noisy channel and the increasing packet length would introduce more traffic load over the connection and cause the packet loss due to network congestion; hence the throughput decreases with the increasing packet length.

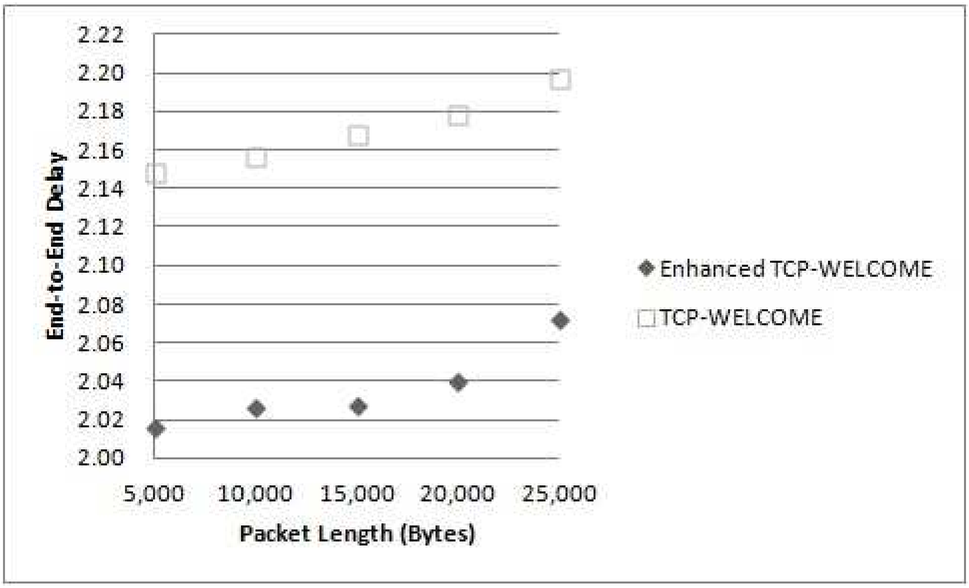


Figure 6.43: End-to-End Delay versus varying Packet Length under noisy wireless channel (n = 10)

Figure 6.43 compares the end-to-end delays of both variants under the noisy channel with varying packet lengths. Enhanced TCP-WELCOME offers 6% lower end-to-end delay overall. The network congestion would occur more often since the heavy traffic load is generated by the noisy channel and increasing packet length. The network congestion introduces more queuing delay at the buffers during the data transmission, and therefore the end-to-end delay increases as the packet size increases.

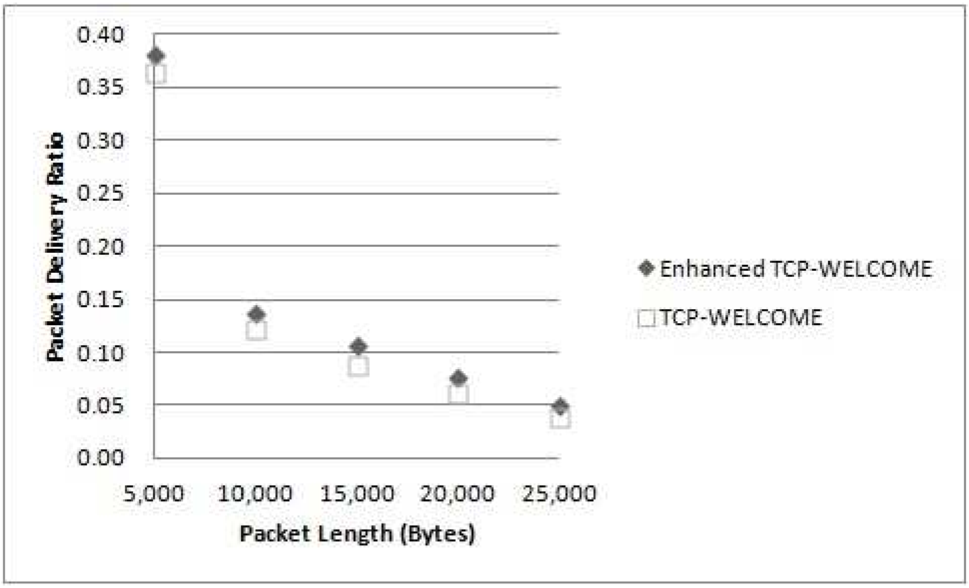


Figure 6.44: Packet Delivery Ratio versus varying Packet Length under noisy wireless channel (n = 10)

Figure 6.44 compares the packet delivery ratios of both variants under the noisy link with varying packet length. Enhanced TCP achieves 18% higher packet delivery ratio that allows the receiver can successfully receive more generated data from application layer without retransmission required. The noisy channel and increasing packet length would generate heavy traffic load over the connection which cause the network congestion occurs more frequent during the communication. The packet is discarded due to the network congestion, so the packet deliver ratio decreases as the packet length enlarges.

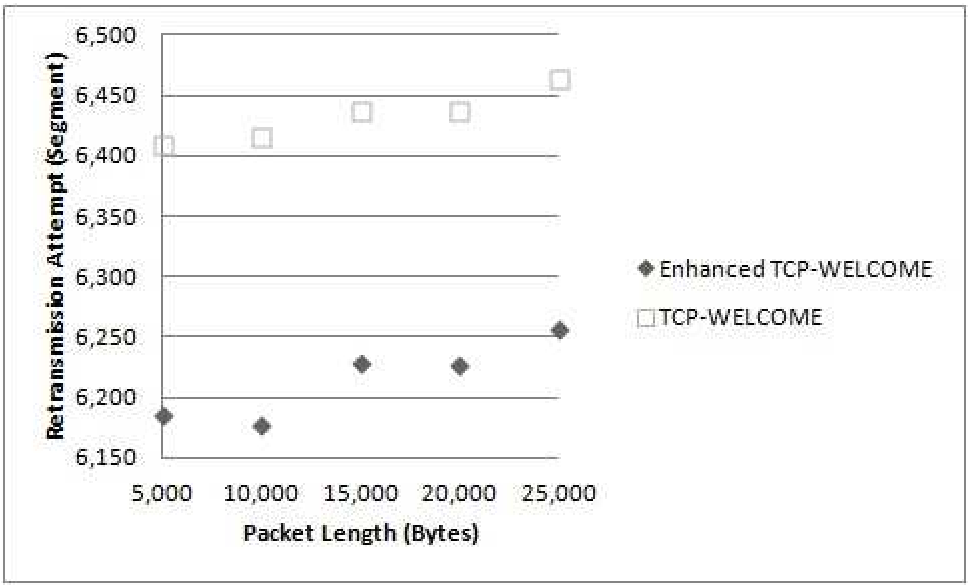


Figure 6.45: Retransmission Attempt versus varying Packet Length under noisy wireless channel (n = 10)

Figure 6.45 compares the retransmission attempts of both variants under the noisy links with varying packet length. Enhanced TCP-WELCOME achieves 3% lower data retransmission required to handle the discarded packet due to network congestion. The performance improvement shown in Figure 6.45 confirms that the proposed loss recovery mechanism of network congestion is able to take more suitable actions to recover the packet loss.

Scenario 12 Small-Medium Network (N= 15, NS= 5 m/s, PL=5,000, 10,000, 15,000, 20,000, 25,000 bytes)

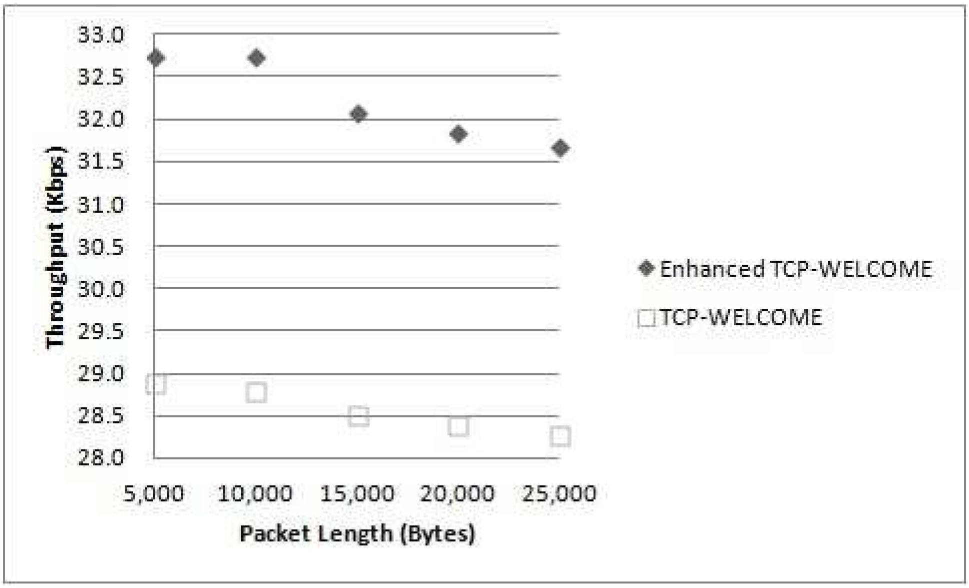


Figure 6.46: Throughput versus varying Packet Length under noisy wireless channel (n = 15)

Figure 6.46 compares the throughputs under noisy channel in small-medium sized network with node size of 15 nodes. Enhanced TCP-WELCOME offers 13% higher throughput than the existing TCP-WELCOME. The noisy channel and the increasing packet length would produce more traffic load over the connection and introduce more packet loss due to network congestion. Therefore, the throughput decreases as the packet length is enlarged.

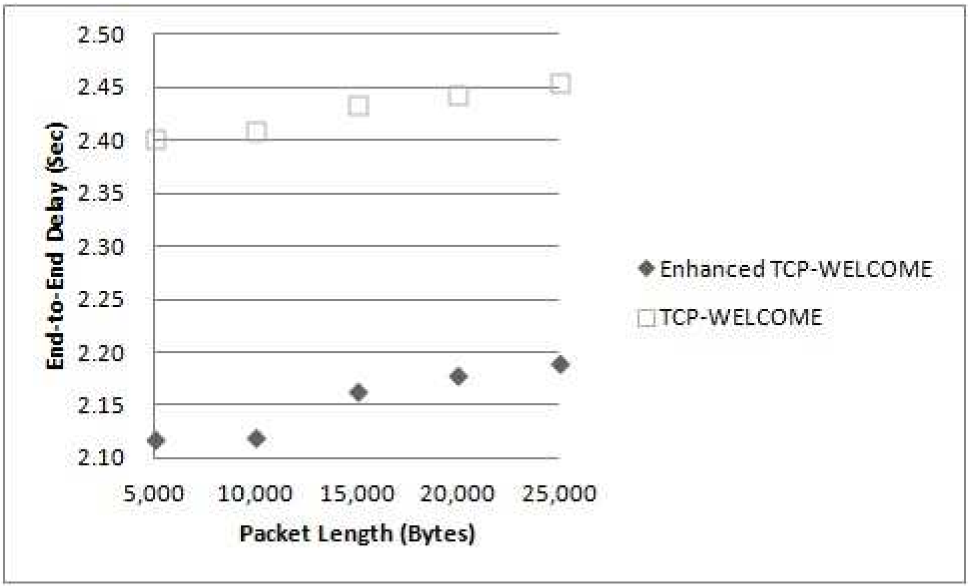


Figure 6.47: End-to-End Delay versus varying Packet Length under noisy wireless channel (n = 15)

Figure 6.47 compares the end-to-end delays of both variants under noisy channel. Enhanced TCP-WELCOME successfully achieves 11% lower end-to-end delay than the existing TCP-WELCOME. The network congestion caused by heavy traffic load would increase the time taken for the packets queuing on the sending buffer; hence the generated end-to-end delay increases with the increasing packet length.

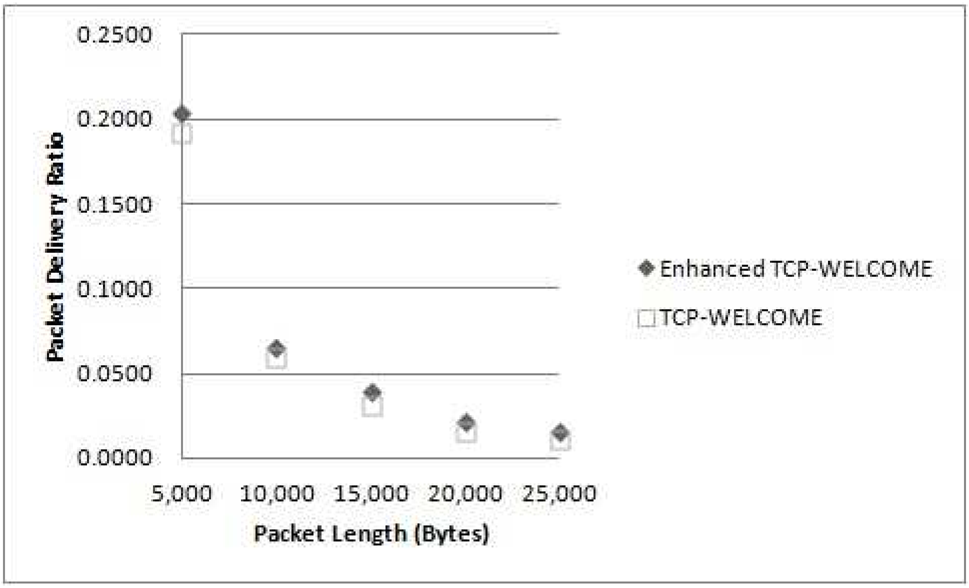
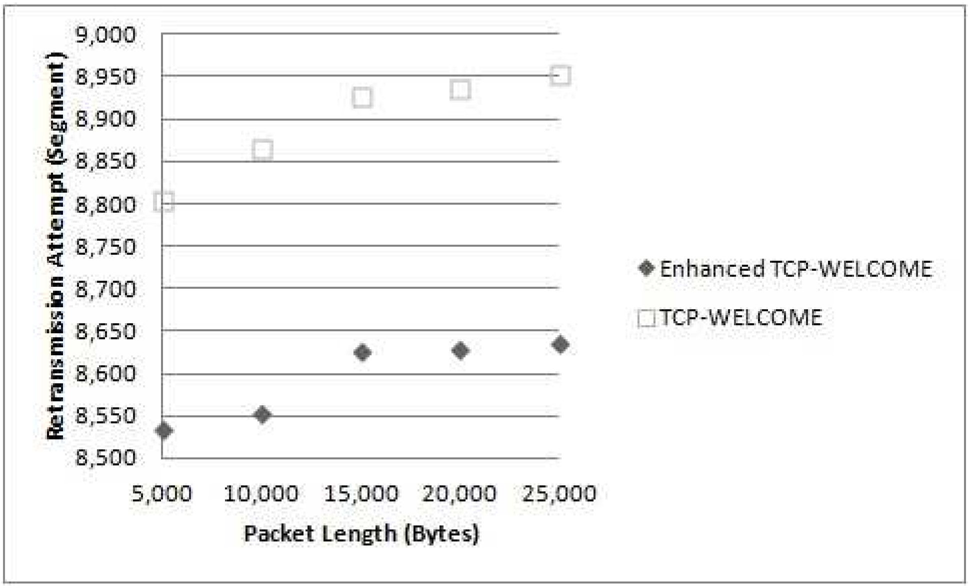


Figure 6.48: Packet Delivery Ratio versus varying Packet Length under noisy wireless channel (n = 15)

Figure 6.48 compares the packet delivery ratios of both Enhanced TCP-WELCOME and TCP-WELCOME under the noisy link. Enhanced TCP-WELCOME achieves 25% higher packet delivery ratio. The heavy traffic load over the connection causes more packet loss due to network congestion, so the packet delivery ratio decreases with the increasing packet length.



sFigure 6.49: Retransmission Attempt versus varying Packet Length under noisy wireless channel (n = 15)

Figure 6.49 compares the retransmission attempts of both variants under noisy link. Enhanced TCP-WELCOME offers 3% lower the retransmission attempt with varying packet length to recover from the packet loss due to network congestion. The significant outperformance shown in Figure 6.49 confirms that the proposed loss recovery algorithm can handle the discard packet during the communication with more appropriate actions.

6.3 Overall Observation and Performance

According to the simulations results, the varying network size, traffic load, node mobility and the conditions of wireless channel have impacted the performance of TCP variants. The different congestion control mechanisms implemented in the variants has led to a variety of differences in TCP performance. The simulation results and discussion presented in Section 5.1 lead to the different effects, which are the effect of network sizes, the effect of packet length, the effect of node speed and the effect of wireless channel conditions. The analysis for the effect on network size, packet length and node speed are on based on the performance under the good wireless channel, and for the effect on wireless channel conditions, it analyzes the performance under the noisy channel error.

6.3.1 Effect on Network Size

The implemented loss differentiation algorithm and loss recovery algorithm can react more appropriately as the network size increases. The number of nodes increases in the network would higher the possibility of the network congestion related packet losses. Since the proposed loss recovery algorithm is mainly designed to enhance the performance of data loss recovery due to network congestion, Enhanced TCP-WELCOME is able to achieve the performance optimization with varying network sizes.

According to the simulation results based on Scenario 1, Enhanced TCP-WELCOME outperforms TCP-WELCOME under the good channel in the small and small-medium sized networks. Enhanced TCP-WELCOME is able to offers up to 24% higher throughputs and 16% lower end-to-end delay by using the proposed network congestion related recovery mechanism to handle to discarded segments. Especially in the small-medium sized network with 5 nodes, Enhanced TCP-WELCOME is able to achieve up to 57% higher throughputs and 36% lower end-to-end delay compare with existing TCP-WELCOME. Furthermore, Enhanced TCP-WELCOME allows the destination nodes receive 13% more data successfully without retransmission required in the small and small-medium sized networks, and it also achieves 8% less retransmission attempt as shown in Figure 6.5. The performance improvement confirms that Enhanced TCP-WELCOME can not only classify the packet losses due to network congestion correctly and react accordingly, but also the proposed network congestion recovery mechanism can improve the performance compare with the TCP-New Reno recovery mechanism which adopted in TCP-WELCOME.

6.3.2 Effect on Node Speed

Traditional TCP suffers from node mobility due to the costs of subsequent route discovery and link breakage. Invalid route and stale route would cause the link failure as the mobile node is not within the propagation range or the node has the expired routes. Once the link failure occurs, the update message packets would increase hence some additional traffic load would be generated over the connection and degrades the performance with varying node speed. Moreover, retransmission is required to recovery the random lost data due to the link failure, and the retransmission accumulation would increase the overhead of network traffic load which results in less throughput and performance degradation. Link failure would occur more often with the higher node speed configured in the network.

Enhanced TCP-WELCOME offers up to 6% higher throughputs and 5% lower end-to-end delay in the small-medium sized networks with 10 nodes. Moreover, the packet delivery ratio is improved by up to 9% higher, hence it achieves 3% lower data retransmission attempt to recover the packet loss due to link failure. The lower retransmissions can reduce the overhead of network traffic, so it is able to offer higher throughputs and achieve lower end-to-end delay over the connections.

In the small-medium sized network with 15 nodes, Enhanced TCP-WELCOME offers lower queuing delay at the buffer and achieves up to 7% lower end-to-end delay, so the throughputs is improved by 8% higher. The destination node is able to receive 20% more data generated from application layer and therefore it offers 2% lower retransmission required during the communication.

In the medium sized network with node size of 20 nodes, Enhanced TCP-WELCOME achieves 0.6% less retransmission attempt and 15% higher packet delivery ratio. Since the retransmitted traffic is reduced, the end-to-end delay is also lowered by 8%, and therefore the throughput generated within the network is 9% higher.

Enhanced TCP-WELCOME can successfully optimize the performance with varying node speed comparing with existing TCP-WELCOME under good wireless channel. It can also achieve further performance enhancement with increasing node speed. Enhanced TCP-WELCOME has the ability to differentiate the packet losses due to link failure and take the most appropriate reaction to recover from the loss which leads to the better performance compare with TCP-WELCOME. The adjustment of data transmission rate is based on the characteristics of the new discovered route helps to maximize the throughput.

6.3.3 Effect on Packet Length

Packet length represents the traffic load handled in the network and it is an effective parameter towards scalability of TCP variants. It would generate more overhead of traffic load if the packet length is increased; hence it would cause the data losses due to the network congestion. Moreover, if the packet length increases, it would make the buffer filled up quickly with the queuing segments and slow down the transmission process. Therefore, the queuing delay at the buffers would increase the end-to-end delay and reduces the throughput which results in further performance degradation.

In the networks implemented 10 nodes, Enhanced TCP-WELCOME is able to shorten the queuing delay as the packet length increases, so the end-to-end delay is lowered by 6% and offers up to 6% higher throughput. The proposed algorithm is able to update the congestion window size efficiently based on RTT value which can discard fewer segments randomly and achieves 3% lower retransmission attempt. It also offers 18% higher packet delivery ratio and allows more packets can reach the destination node successfully without retransmission required.

In the small-medium sized networks with node size of 15 nodes, Enhanced TCP-WELCOME offers up to 7% lower end-to-end delay which helps to achieve 7% higher throughputs over the connections. The retransmission attempt is also lowered by 3% because the data transmission rate is adjusted based on the characteristics of the route which is able to update the sending rate more appropriately and efficiently. Enhanced TCP-WELCOME also achieves up to 33% higher packet delivery ratios which performs outstandingly compare with TCP-WELCOME.

In the medium sized network with 20 nodes, Enhanced TCP-WELCOME offers up to 10% lower end-to-end delay; hence it improves throughput by 11% higher during the. Furthermore, the receiver is able to receive 45% more traffic generated at application layer without retransmission required. However, the network congestion occurs more frequently because of the increasing packet length and bigger network size would introduce more overhead of the traffic over the connection. Therefore, it only can offer 0.4% lower retransmission attempt.

Enhanced TCP-WELCOME can successfully optimize the overall performance with varying packet length outperforms the existing TCP-WELCOME under the good wireless channel. Moreover, it can offer further performance improvement as the traffic load increases. The performance improvement again confirms that Enhanced TCP-WELCOME can identify the packet loss due to the network congestion correctly and the proposed loss recovery algorithm is able to take more appropriate actions to recover from the loss. The adjustment of congestion window is based on the characteristics of the new discovered route and ACKs that helps to maximize the throughput and optimize the performance improvement.

6.3.4 Effect on Wireless Channel Conditions

The wireless channel is generally assumed to be perfect communication channel for the data transmission in the simulation experiment. However, the noisy wireless channel would be more close to the general communication link in the real world. For instance, the noise can be generated by other wireless devices within the overlapping propagation ranges. The noisy wireless channel would create the interference and cause the packet discarded and lead to the performance degradation.

With varying network sizes under the noisy channel, Enhanced TCP-WELCOME achieves up to 11% higher throughputs and 10% lower end-to-end delay. The packet delivery ratio is improved by 8% higher and the retransmission attempt is lowered by 4%, so the receiver can receive more data without retransmission required. Moreover, as shown in Figure 6.30 to 6.33, Enhanced TCP-WELCOME is able to achieve further performance optimization as the network size enlarges.

With varying node speed under the noisy communication channel in the small-medium sized networks with the network size of 10 nodes, Enhanced TCP-WELCOME can reduced the time taken for the packet waiting at the buffers and achieves up to 6% lower end-to-end delay and 7% higher throughputs. The packet delivery ratio is improved by 4 % and it requires 3% less data retransmission to handle the discarded packets. When the network is implemented with 15 nodes, it achieves 11% lower end-to-end delay by reducing the queuing delay at the buffer, so the throughput is improved by 13% higher. Moreover, if offers 6% higher the packet delivery ratio in the small-medium sized network and the data retransmission recovers from the packet loss due to link failure is lowered by 3%. The significant outperformance of Enhanced TCP-WELCOME confirms that it can also optimize the performance with varying node speed under the noisy wireless channel.

With varying packet length under the noisy communication channel in the small-medium sized networks, Enhanced TCP-WELCOME achieves 7% higher throughputs and 6% lower the end-to-end delay in the network size with 10 nodes. The amount of discarded packet is dropped by 3% and it’s able to offer 18% higher packet delivery ratio. Furthermore, Enhanced TCP-WELCOME performs outstandingly in the network size of 15 nodes by offering up to 11% higher throughputs and 13% lower end-to-end delay that allows the packet travel across the network with shorter latency and then more amount of delivered packets which been successfully acknowledged by the receiver. The packet delivery ratios are improved by 25% higher in the small-medium sized networks, which the receiver is able to successfully receive 3% more traffic without retransmission required.

TCP suffers from the noisy communication channel, but the statistical results shows that Enhanced TCP-WELCOME can successfully achieve performance improvement significantly with vary network sizes, node speeds and packet lengths under the noisy wireless channel and it can even offer better performance enhancement than using the clean wireless channel.

Enhanced TCP-WELCOME is able to offer further performance improvement with increasing node speed and packet length under the noisy wireless channel. Enhanced TCP-WELCOME can classify the packet loss model accurately and the proposed loss recovery algorithm acts more appropriately, hence it is still able to ensure certain performance optimization with varying node speed and packet length.

6.4 Validation of Simulation Results

The graph comparison and the output of result from the previous results in literature were used in validating the acquired results of simulation [26]. The simulation results presented in graphs is to ease the illustration and comparison of the proposed and existing TCP variants. In order to ensure the accuracy of the simulation results, each experiment is replicated and ran multiple times and the data collected from each run was recorded and compared. The graphical results of each replication can provide the similar or same illustration and therefore the analysis and interpretation of the experimental results could lead to the same conclusion.

As discussed in literature [3], TCP-WELCOME was implemented in NS-2 Simulator, and the comparative analysis of simulation results showed that TCP-WELCOME outperformed some of the existing TCP variants such as TCP-New Reno, TCP-SACK, TCP-Vegas and TCP-Westwood. Therefore, in order to evaluate TCP-WELCOME and Enhanced TCP-WELCOME and conduct the comparative analysis, both variants were implemented in OPNET Modeler for the simulation experiments.

The values of the general parameters implemented in the experiments were referred to TCP-WELCOME. Moreover, the network size used for TCP-WELCOME evaluation in the literature [3] was 5 nodes and configured with a low node speed environment. Therefore, the performance evaluations of Enhanced TCP-WELCOME impact on varying network size and node mobility were concentrated. The varying network sizes were relatively small with the node size of 5, 10, 15 and 20 nodes. The node mobility with the node speed of 3, 4, 5, 6 and 7 m/s were used to present the slow movement. Moreover, the performance impact on varying packet length and wireless channel conditions are also focused in this research. The packet length with 5,000, 10,000, 15,000, 20,000 and 25,000 bytes were configured to present general, medium and heavy traffic load. The noisy and good wireless communication channels were implemented for investigating if Enhanced TCP-WELCOME can still optimize the performance when there is any other interference during the communication, and therefore only the small-medium sized networks were used for this investigation.

The network size, node mobility, traffic load and conditions of wireless channel have a good impact on the performance of TCP-variants, the experimental results show the effect of node speed and packet length have the impact on determining the performance of Enhanced TCP-WELCOME and TCP-WELCOME under good and noisy communication channels in small-medium and medium networks.

6.5 Summary

The chapter presented the simulation results of the experiments with varying number of nodes, packet lengths and node speed as designed in Chapter 5. The obtained results show that the network size, traffic load node mobility, and conditions of wireless channel are the factors that affect the TCP variants performance. The simulation results was measured and analyzed based on the performance metrics, including throughput, end-to-end delay, retransmission attempts and packet delivery ratio. According to the performance analysis, Enhanced TCP-WELCOME can outperform TCP-WELCOME in the small-medium and medium sized network, and in order to improve the performance of Enhance TCP-WELCOME in the high speed environment, the further study on node mobility is required in the future.

Chapter 7

Conclusions and Future Work

A class of TCP variant called Enhanced TCP-WELCOME is designed and reported in this thesis to improve the performance of TCP over MANETs. Enhanced TCP-WELCOME is developed by modifying TCP-WELCOME's loss recovery algorithm. The proposed TCP performance was evaluated through several simulation experiments based on the defined scenarios, and the evaluation results were measured through the performance metrics. An in-depth understanding of the impact of TCP variants on a MANET performance has been acquired in this study. The experimental results help the performance comparative analysis between Enhanced TCP-WELCOME and the existing variant, and also an insight into the parameter that influence the TCP performance.

The background material relevant to MANET and TCP are provided. MANET characteristics and TCP performance degradation issues over MANET are discussed. Some contemporary TCP variants in the related research are introduced. TCP-WELCOME can perform adequately to handle the packet loss due to link failure, congestion. TCP-WELCOME implemented with the original congestion mechanism of TCP-New Reno to recover from the network congestion packet dropped. In order to improve the TCP performance over MANET, the literature related to the loss recovery algorithm of handling the congestion packet loss is reviewed and discussed in this paper.

Analytical modeling, computer simulation and direct experiment have been generally used for modeling the networks and evaluating the performance, and a simulation approach will be used to evaluate the TCP performance with different defined scenarios by using the OPNET Modeler. OPNET Modeler is able to simulate the defined scenarios easily and generate the valid result to represent the performance of TCP variant more accurately.

The new TCP class, Enhanced TCP-WELCOME, is proposed in this study. It is combined the features of TCP-WELCOME and TCP-AW. The proposed variant implemented the loss differentiation algorithm introduced in the existing TCP-WELCOME to identify packet loss due to link failure, network congestion and wireless channel error correctly, and then performs the recovery process more adequately with the new loss recovery algorithm proposed in this paper. The recovery mechanism for network congestion packet loss is modified and implemented with the features of TCP-AW to approach the further TCP performance improvement.

The performance metrics used for measuring the evaluation results are throughput, end-to-end delay, packet delivery ratio, and retransmission attempts. To investigate the TCP performance impact on network size, traffic load and node mobility, the scenarios are designed careful where only one control parameter is changed in each scenario, including the number of nodes, packet length and node speed. Moreover, the good and noisy channels are used to investigate the performance impact on wireless channel conditions.

The simulation results of the experiments with varying number of nodes, packet lengths and node speed under the good and noisy wireless channels. The simulation results was measured and analyzed based on the performance metrics. The results comparison and performance analysis shows that the proposed loss differentiation algorithm can identify the reason of the packet loss accurately and the introduced loss recovery algorithm can take more appropriate actions to recover from the loss due to network congestion and link failure. The proposed loss differentiation and recovery algorithms can react accordingly with increasing network size. The increasing number of nodes over MANET would introduce more network congestion during the data transmission. The performance improvement based on the statistical results confirms the proposed network congestion recovery mechanism can outperforms the recovery mechanism of TCP-New Reno which implemented in TCP-WELCOME. Traditional TCP suffers from node mobility due to the costs of subsequent route discovery and link breakage. The route update message packets would increase additional traffic load and it is required to recovery the random lost data due to the link failure, and the overhead of network traffic load which results in less throughput and performance degradation. Enhanced TCP-WELCOME can successfully optimize the performance with varying node speed comparing with existing TCP-WELCOME under good wireless channel. It can also achieve further performance enhancement with increasing node speed. Enhanced TCP-WELCOME has the ability to recover from the link failure loss which leads to the better performance compare with TCP-WELCOME. The adjustment of data transmission rate is based on the characteristics of the new discovered route helps to maximize the throughput. Packet length is an effective parameter towards scalability of TCP variants since it represents the traffic load handled in the network. The increasing packet length would generate more overhead of traffic load and slow down the transmission process; hence it would cause the data losses due to the network congestion. Enhanced TCP-WELCOME can successfully optimize the overall performance with varying packet length outperforms the existing TCP-WELCOME under the good wireless channel. The performance improvement shows the proposed loss recovery algorithm is able to take more appropriate actions to recover from the loss. The adjustment of congestion window is based on the characteristics of the new discovered route and ACKs that helps to maximize the throughput and optimize the performance improvement.

The experiments of noisy wireless channel in this paper represent the general communication link in the real world. The noisy wireless channel would create the interference and cause the packet discarded and lead to the performance degradation. The performance analysis shows Enhanced TCP-WELCOME is able to offer further performance improvement with vary network sizes, node speeds and packet lengths under the noisy wireless channel.

The performance evaluation of Enhanced TCP-WELCOME is based on the defined scenarios and metrics, and the following performance limitations are expected, 1) performance under the large network size with more than 20 mobile nodes. 2) Handling the real-time traffic, such as online video streaming. 3) Performance under high mobility network environment, for instance, node speeds over 7 m/s. 4) Performance under the wireless network with heavy interferences. 5) Recovering the packet loss due to wireless channel errors.

Chapter 1 introduces the research basis, outlined the motivation and aims of this study, it also showed the structure of this paper. Chapter 2 provides the background information of MANET environment, and several literature of recently proposed TCP variants are reviewed and discussed. The intended was discussed in the end of this chapter. Chapter 3 describes the methodologies used for network modelling and performance evaluation. In this study, the simulation approach is used for TCP performance evaluation by using the OPNET Modeler. Chapter 4 presents the new TCP variant proposal of this study. Enhanced TCP-WELCOME is implemented the loss differentiation algorithm introduced in the existing TCP-WELCOME to identify the packet loss model correctly and performs the modified loss recovery algorithm recovery process appropriately. In Chapter 5, the network modelling and defined scenarios are discussed in detail. The performance metrics includes throughput, end-to-end delay, packet delivery ratio, and retransmission attempts. The simulation environment and protocol parameters are also discussed. Chapter 6 presents the simulation results comparison of the experiments with varying number of nodes, packet lengths and node speed under the good and noisy wireless channels. The performance comparison were measured and analyzed according to the performance metrics.

7.1 Future Work

Performance Evaluation with Various Applications

The experimental research of this study was mainly concentrate on the control parameters, which are network sizes, traffic load and node mobility. FTP was the application data traffic used in the simulation. However, other types of application traffic may have the impact on TCP performance. Therefore, other application data traffic such as HTTP and VoIP are possibly included in the future research. The following areas are considered as possible research in the future.

Performance Study with Higher Node Mobility

According to the obtained results, the proposed TCP variant suffers from higher speed network, so it needs to be improved in order to handle the high speed MANET environment with large network size. The recovery mechanism of link failure is required to be revised to handle the varying node mobility. The further performance improvement of Enhanced TCP-WELCOME is considered as one of the future research. The simulation results will be acquired by using OPNET Modeler, and the performance comparison and comparative analysis would still be used to carry out in this research.

References

[1] Casetti, C., Gerla, M., Mascolo, S., Sanadidi, M. Y., & Wang, R., *TCP westwood: end-to-end congestion control for wired/wireless networks* Wireless Networks 2002. **8**(5): p. 467 - 479.

[2] El-Sayed, H.M., *Performance evaluation of TCP in mobile ad hoc networks*, in *Second International Conference on Innovations in Information Technology.* 2005: Seoul, South Korea.

[3] Seddik-Ghaleb, A., Ghamri-Doudane, Y., & Senouci, S.-M., *TCP WELCOME TCP variant for Wireless Environment, Link losses, and COngestion packet loss ModEls*, in *First International Communication Systems and Networks and Workshops*. 2009: Bangalore. p. 1 - 8

[4] Marcondes, C., Sanadidi, M.Y., Gerla, M., & Shimonishi, H., *TCP Adaptive Westwood- Combining TCP Westwood and Adaptive Reno: A Safe Congestion Control Proposal*, in *IEEE International Conference on Communications*. 2008: Beijing p. 5569 - 5575.

[5] Ahmad, A., *Wireless and Mobile Data Networks*. 2005, Hoboken, NJ, USA: John Wiley & Sons, Inc.

[6] Forouzan, B.A., & Fegan, S.C., *Data communications and networking*. 2007: McGraw-Hill.

[7] Henna, S., *A Throughput Analysis of TCP Variants in Mobile Wireless Networks*, in *Third International Conference on Next Generation Mobile Applications, Services and Technologies*. 2009: Cardiff, Wales p. 279 - 284.

[8] Wang, R., Valla, M., Sanadidi, M.Y., & Gerla, M., *Using adaptive rate estimation to provide enhanced and robust transport over heterogeneous networks*, in *10th IEEE International Conference on Network Protocols*. 2002. p. 206 - 215.

[9] Mbarushimana, C., & Shahrabi, A.; , *E-TCP: Enhanced TCP for IEEE802.11e Mobile Ad Hoc Networks* in *15th International Conference on Parallel and Distributed Systems* 2009: Shenzhen. p. 632 - 639.

[10] Yang, S.J., & Lin, Y. C., *RED tuning for TCP performance on the mobile ad hoc networks* in *11th International Conference on Parallel and Distributed Systems*. 2005. p. 412 - 418.

[11] Kim, D.K., Cano, J. C., Manzoni, P. & Toh, C. K, *A comparison of the performance of TCP-Reno and TCP-Vegas over MANETs*, in *3rd International Symposium on Wireless Communication Systems*. 2006: Valencia p. 495-499.

[12] Wu, X.C., Choon, M., & Ananda, A. L, *TCP HandOff: A Practical TCP Enhancement for Heterogeneous Mobile Environments*, in *IEEE International*

Conference on Communications. 2007: Glasgow p. 6043 - 6048

[13] Tung, L.P., Sun, Y. S., Huang, T. L., & Chen, M. C. *TCP-sleek channel access scheduling for multi-hop mesh networks*. in *Proceedings of the 2009 international Conference on Wireless Communications and Mobile Computing: Connecting the World Wirelessly*. 2009. Leipzig, Germany

[14] Lien, Y.N., & Yu, Y. F. , *Hop-by-Hop TCP over MANET*, in *IEEE Asia-Pacific Services Computing Conference*. 2008: Yilan, Taiwan. p. 1150-1155.

[15] Gajjar, S., & Gupta, H.M. , *Improving performance of adhoc TCP in Mobile Adhoc Networks*, in *Annual IEEE India Conference*. 2008: Kanpur p. 144-147.

[16] Liu, J., Yougu, Y., Nicol, D.M., Gray, R. S., Newport, C. C., Kotz, D., & Perrone, L.F., *Simulation Validation using direct execution of wireless ad-hoc routing protocol*, in *18th Workshop on Parallel and Distributed Simulation*. 2007. p. 7-11.

[17] Carson, J.S., *Introduction to modeling and simulation*, in *Winter Simulation Conference*. 2005. p. 16-23.

[18] Sargent, R.G., *Verification and validation of simulation models*, in *37th conference on Winter simulation*. 2005: Orlando, Florida. p. 130-143.

[19] Cavin, D., Sassion, Y., & Schiper, A., *On the Accuracy of MANET simulators*, in *Second ACM international workshop on Principle of mobile computing*. 2002: Toulouse, France. p. 38-43.

[20] Hogie, L., Bouvry, P., & Guinand, F., *An overview of manets simulation*. Electronic Notes in Theoretical Computer Science. Vol. 150. 2006. 81-101.

[21] Chen, Q., & Niu, Z. H. , *A delayed adaptive retransmission scheme for false route failure in MANET*, in *The 2004 Joint Conference of the 10th Asia-Pacific Conference on Communications and the 5th International Symposium on Multi-Dimensional Mobile Communications Proceedings*. 2004.

[22] Raisinghani, V.T., Singh, A.K., & Iyer, S. , *Improving TCP performance over mobile wireless environments using cross layer feedback*, in *Paper presented at IEEE International Conference on Personal Wireless Communications*. 2002.

[23] Selvam, A., Kathiravan, K., & Reshmi, R. . *A Cross-Layer TCP Protocol with Adaptive Modulation for MANETs*. in *International Conference on Signal Processing, Communications and Networking*. 2008. Chennai.

[24] Seok, Y.H., Park, Y. S., & Choi, Y. H. , *Vertical and horizontal flow controls for TCP optimization in the mobile ad hoc networks*, in *2003 IEEE 58th Vehicular Technology Conference*. 2003.

[25] Sun, D., & Man, H. , *Performance comparison of transport control protocols over mobile ad hoc networks*, in *12th IEEE International Symposium on Personal, Indoor and Mobile Radio Communications*. 2001: San Diego, CA.

[26] Balci., O. *Principles and techniques of simulation validation, verification and testing*.

in *Simulation Conference Proceedings*. 1995.

[27] Kim, D., Bae, H., & Toh, C. K, *TCP-Vegas Performance Over MANET Routing Protocols*, in *IEEE Transactions on Vehicular Technology* 2007. p. 372-377.

[28] Mbarushimana, C., & Shahrabi, A., *Resource Efficient TCP: Reducing Contention-Induced Spurious Timeouts in QoS -Aware MANETs*, in *22nd International Conference on Advanced Information Networking and Applications*. 2008: Okinawa, Japan.

[29] Miyake, M.I., H., *Enhancing Fast Recovery Algorithm using Loss Recovery for NewReno TCP*, in *IEEE 16th International Symposium on Personal, Indoor and Mobile Radio Communications*. 2005: Berlin p. 1555 - 1559

[30] Park, M.Y., Chung, S. H., & Sreekumari, P. , *Estimating Rate of Queue Usage to Differentiate Cause of Packet Loss in Multi-hop Wireless Networks*, in *33rd Annual IEEE International Computer Software and Applications Conference*. 2009: Seattle, WA p. 500 - 508.

[31] Park, M.Y., & Chung, S. H. , *Analyzing effect of loss differentiation algorithms on improving TCP performance*, in *The 12th International Conference on Advanced Communication Technology* 2010: Phoenix Park p. 737 - 742.

[32] Pawlikowski, K., Jeong, H.-D. J., & Lee, J.-S. R, *On credibility of simulation studies of telecommunication networks*, in *IEEE Communications Magazine* 2002. p. 132-139.

[33] Qureshi, B., Othman, M., & Hamid, N.A.W., *Progress in various TCP variants (February 2009)* in *2nd International Conference on Computer, Control and Communication*. 2009: Karachi. p. 1 - 6

[34] Sarkar, N.I., & Halim, S. A., *Simulation of computer networks: simulators, methodologies and recommendations*, in *5th International Conference on Information Technology and Application*. 2008: Cairns, Australia. p. 420-425.

[35] Sarkar, N.I., & K. W. Sowerby *The effect of traffic distribution and transport protocol on WLAN performance*, in *IEEE Australasian Telecommunications Networks and Applications Conference (ATNAC '09)*. 2009: Canberra, Australia. p. 1-6.

[36] Seddik-Ghaleb, A., Ghamri-Doudane, Y., & Senouci, S.-M., *TCP computational energy cost within wireless Mobile Ad Hoc Network*, in *IEEE/ACS International Conference on Computer Systems and Applications*. 2009: Rabat. p. 955 - 962.

[37] Shimonishi, H., & Murase, T. , *Improving efficiency-friendliness tradeoffs of TCP congestion control algorithm*, in *IEEE Global Telecommunications Conference*. 2005.

[38] Shimonishi, H., Hama, T., & Murase, T., *TCP-AdaptiveReno for improving efficiency-friendliness tradeoffs of tcp congestion control algorithm*, in *PFLDnet*.

2006. p. 87-91.

[39] Wong, A.H., Neve, M. J., & Sowerby, K. W. *Performance analysis for indoor wireless systems employing directional antennas*. in *IEE Proceedings Communications* 2005.

Appendices

This section presents the extra figures showing OPNET configuration parameters and the modification of Enhanced TCP-WELCOME, and the tables present the average results of all experimental simulation scenarios.

Appendix A: OPNET simulation configurations

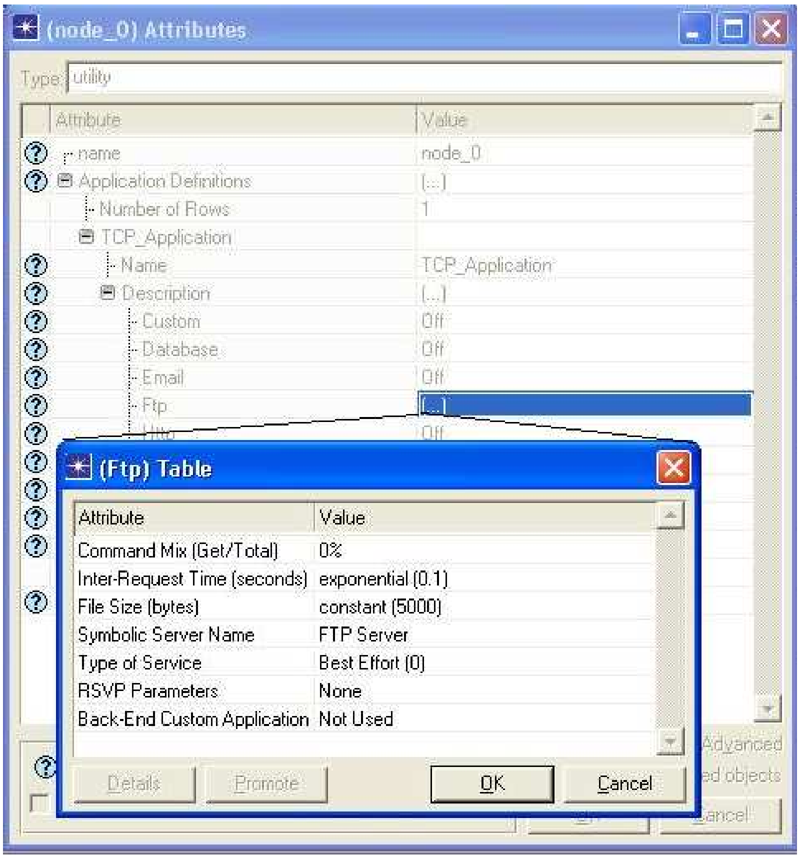


Figure A.1: Application Configuration

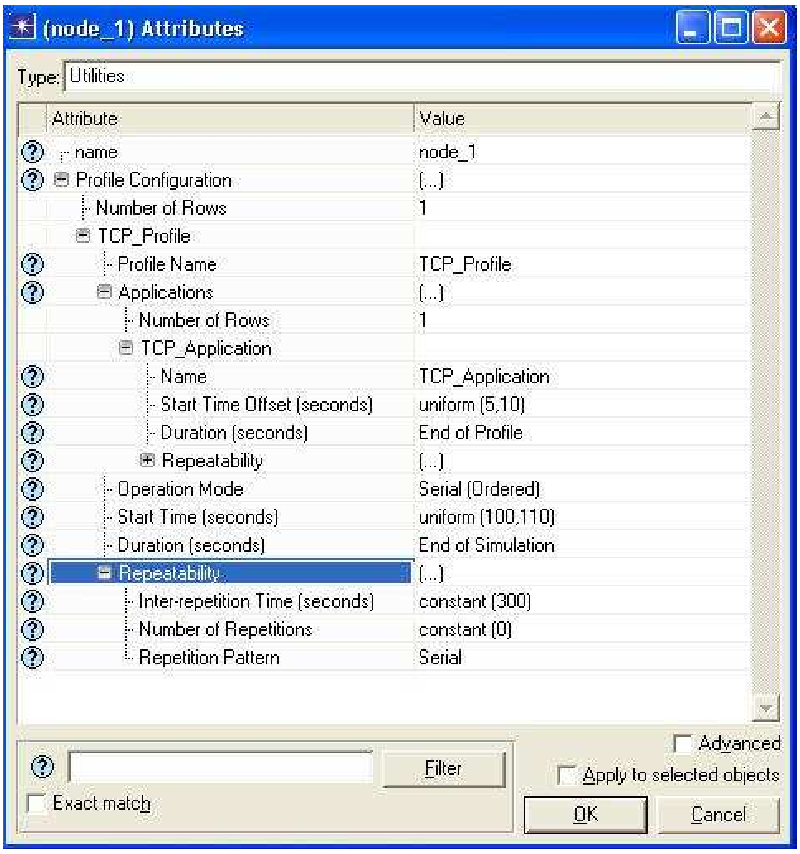


Figure A.2: FTP Profile Configuration

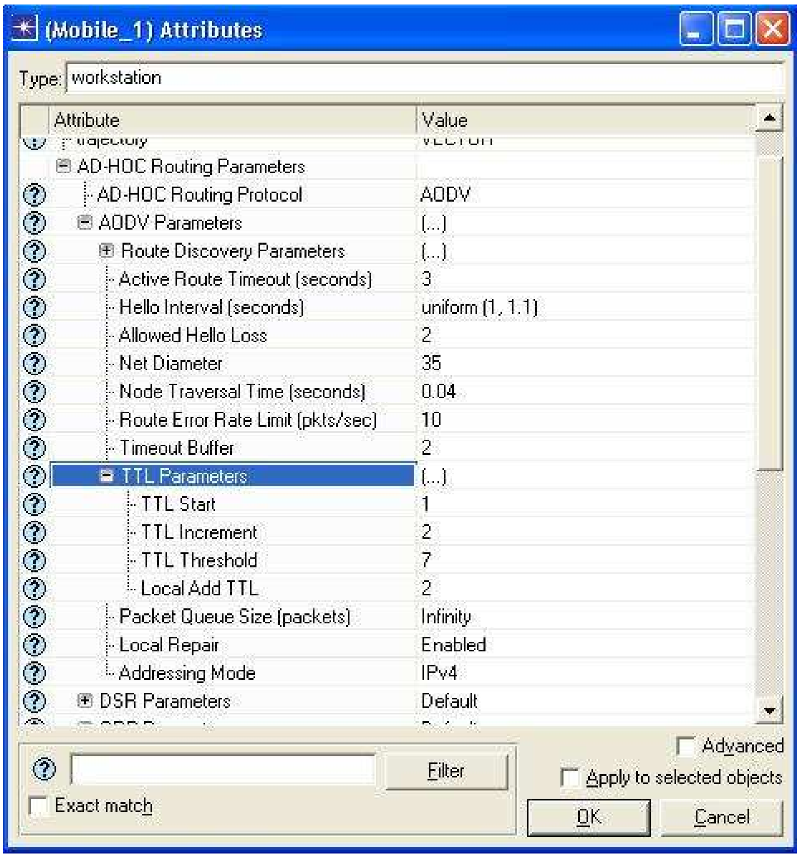


Figure A.3: AODV Routing Protocol Configuration

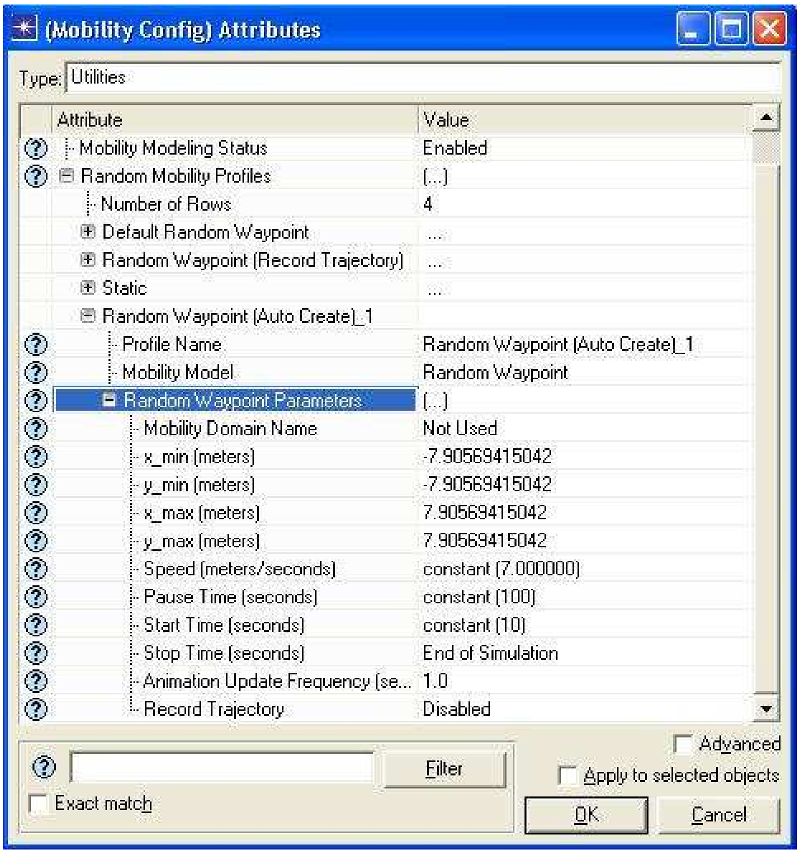


Figure A.4: Mobility Configuration

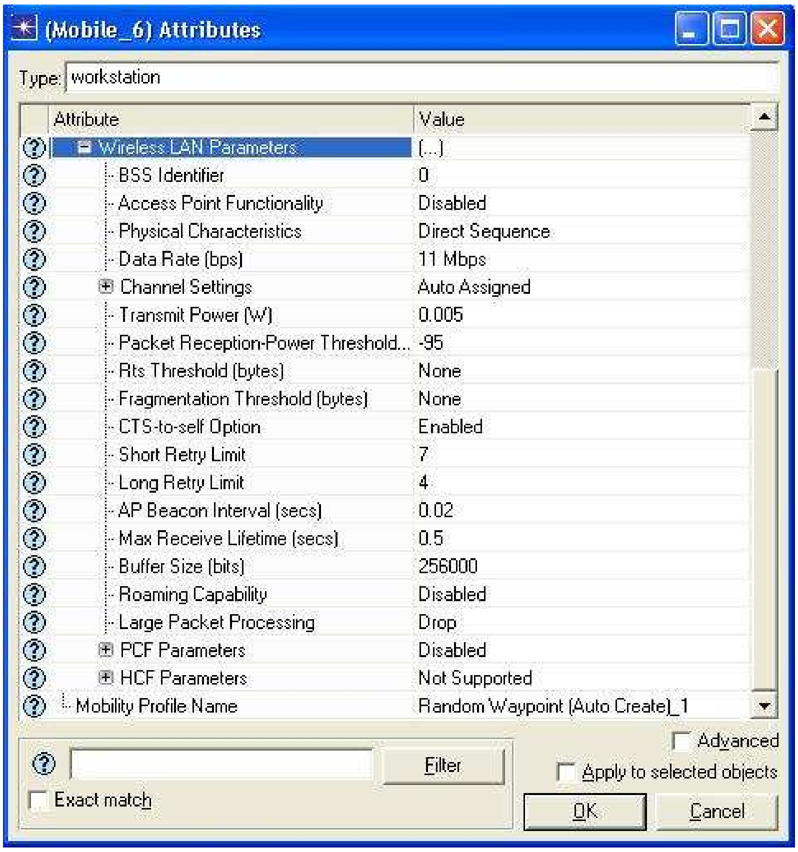


Figure A.5: Wireless and Mobility Configuration

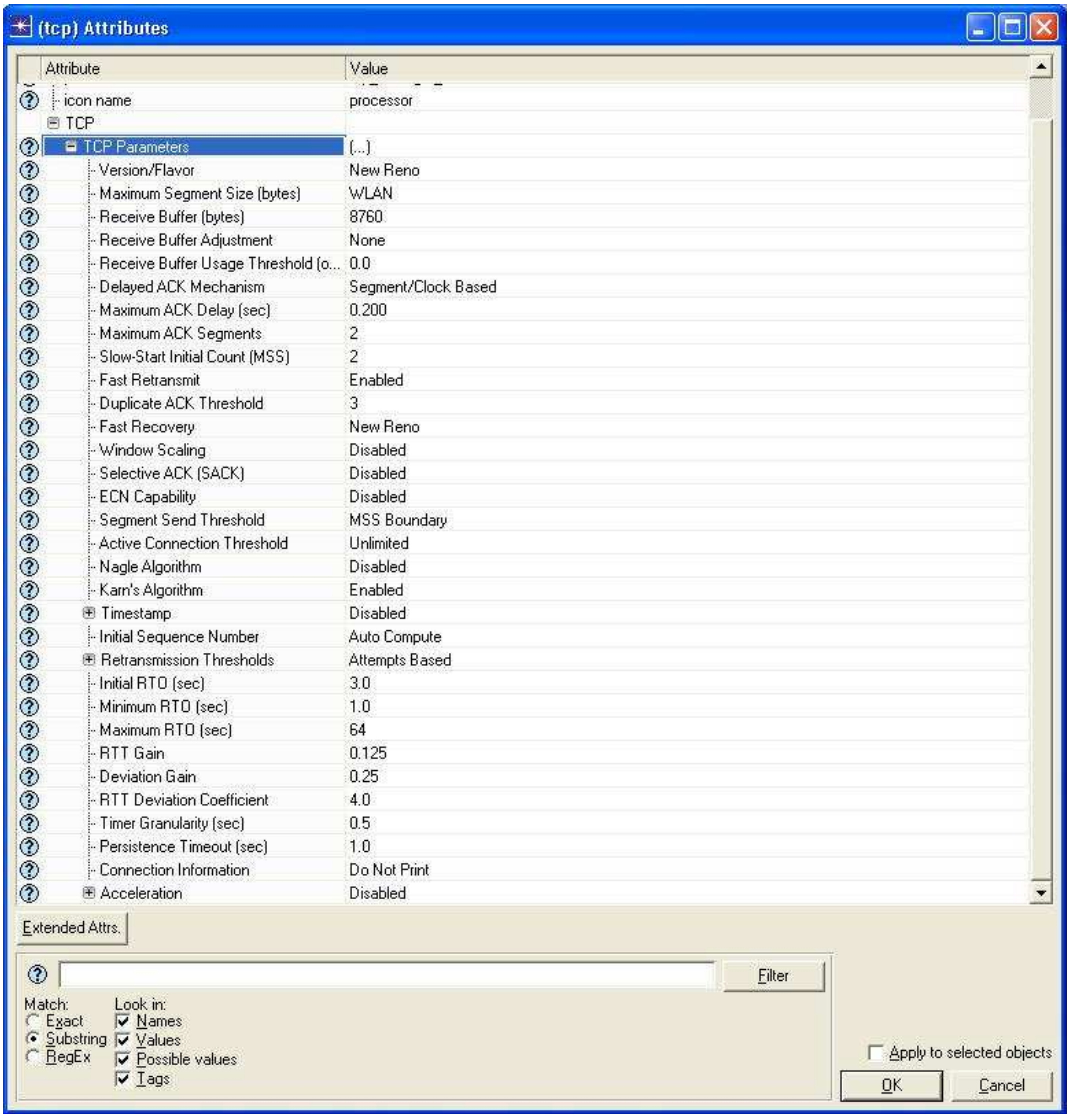


Figure A.6: TCP Protocol Configuration

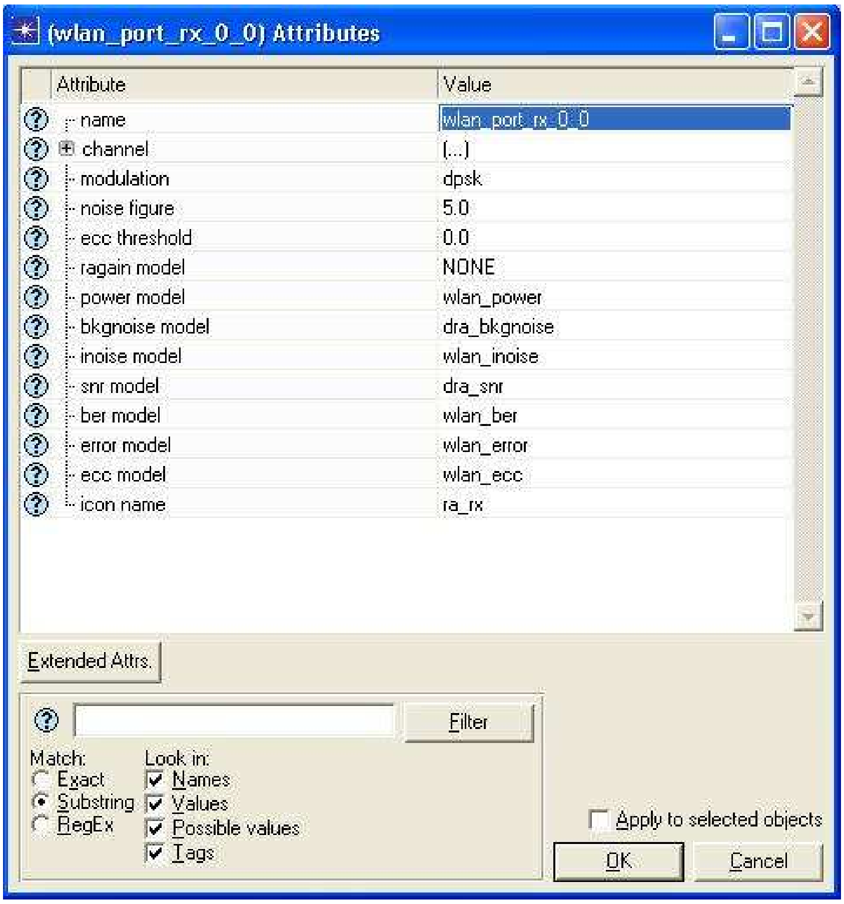


Figure A.7: WLAN Noisy Channel Configuration

Appendix B: Enhanced TCP-WELCOME Algorithms

```
//@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@

// RTO Calculation - Link Failure Handling introduced in TCP-WELCOME

if (rtt_c < retrans_rtt){

    //Calculate the current RTO
    current_rto = (retrans_rtt / previous_rtt) * current_rto;

    // restrict it to within limits.
    if (current_rto > rto_max){
        current_rto = rto_max;
    } else if (current_rto < rto_min){
        current_rto = rto_min;
    }
}else{
    current_rto *= 2;
    current_rto = MIN (current_rto, rto_max);

}

//@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@
```

Figure B.1: Modification of Link Failure Recovery Algorithm

```
// Network Congestion Handling (RTO Expiration)
}else if (rtt_c > retrans_rtt  && tcp_flavor == TcpC_Flavor_New_Reno){

    // e is the mathematical constant (Euler's number)
    double  e = 2.7182818284590452353602874713526624977572470936999 5;

    // non-congested time period
    double c;

    cwnd_dif = ((alpha * (current_time - transmission_start_time)) - (cwnd*((retrans_rtt - rtt_min)/retrans_rtt))) * (rtt_min/rtt_gain);

    if(cwnd_dif < 0 || cwnd_dif == 0){

        cwnd = cwnd + cwnd_dif;

    }else if(cwnd_dif > 0){

        if((current_time - transmission_start_time) < 0.4){

            c = current_time - transmission_start_time;
        }else{

            c = 0.3;
        }

        cwnd = cwnd + (cwnd_dif * pow(e, -c));
    }

}else{
    cwnd = snd_mss;
}

//@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@
```

Figure B.2: Modification of Network Congestion Recovery Algorithm (RTO Expiration)

```
//@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@

// Network Congestion Handling (Three Duplicated ACKs)
if (rtt_c > retrans_rtt  && tcp_flavor == TcpC_Flavor_New_Reno && dup_ack_cnt == 3){

    // e is the mathematical constant (Euler's number)
    double  e = 2.71828182845904523536028747135266249775724709369995;

    // partial non-congested time period defined in TCP-AW
    double c;

// get the current time
    current_time = op_sim_time ();

    cwnd_dif = ((alpha *(current_time - transmission_start_time)) - (cwnd*((retrans_rtt - rtt_min)/retrans_rtt))) * (rtt_min/rtt_gain);

    if(cwnd_dif < 0 || cwnd_dif == 0){

        cwnd = cwnd + cwnd_dif;

    }else if(cwnd_dif > 0){

        if((current_time - transmission_start_time) < 0.4){

            c = current_time - transmission_start_time;
        }else{

            c = 0.3;
        }

        cwnd = cwnd + (cwnd_dif * pow(e, -c));
        }
}else{

    cwnd = ssthresh;

}

//@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@@
```

Figure B.3: Modification of Network Congestion Recovery Algorithm (3 Duplicated ACKs)

Appendix C: Additional Results for Chapter 6.

Table C.1: Simulation Results of Varying Network Sizes under Good wireless Channel

Scenario 1: Varying Network Sizes under Good Wireless Channel			
Performance Metrics	**Node Number**	**Enhanced TCP-WELCOME**	**TCP-WELCOME**
Throughput (Kbps)	5	165.2743	104.9521
	10	34.3192	32.4051
	15	32.6889	30.3830
	20	8.4537	7.7717
End-to-End Delay (Seconds)	5	0.4197	0.6609
	10	2.0210	2.1404
	15	2.1218	2.2829
	20	8.2047	8.9247
Packet Delivery Ratio	5	3,687	4,467
	10	6,176	6,386
	15	8,292	8,508
	20	12,370	12,432
Retransmission Attempt (Segments)	5	0.6225	0.6083
	10	0.4846	0.4555
	15	0.1856	0.1413
	20	0.1469	0.1327

Table C.2: Simulation Results of Varying Node Speed under Good wireless Channel in Small-Medium Network (n=10)

Scenario 2: Varying Node Speed under Good Wireless Channel in Small-Medium Network			
Performance Metrics	**Node Speed (m/s)**	**Enhanced TCP-WELCOME**	**TCP-WELCOME**
Throughput (Kbps)	3	34.3428	32.7011
	4	34.4155	32.6480
	5	34.3192	32.4051
	6	34.1139	32.3758
	7	34.1038	32.0852
End-to-End Delay (Seconds)	3	2.0196	2.1210
	4	2.0154	2.1245
	5	2.0210	2.1404
	6	2.0332	2.1423
	7	2.0338	2.1617
Packet Delivery Ratio	3	0.5105	0.4597
	4	0.4928	0.4568
	5	0.4846	0.4459
	6	0.4541	0.4154
	7	0.4031	0.3711
Retransmission Attempt (Segments)	3	6,156	6,356
	4	6,182	6,353
	5	6,176	6,386
	6	6,188	6,405
	7	6,211	6,420

Table C.3: Simulation Results of Varying Node Speed under Good wireless Channel in Small-Medium Network (n=15)

Scenario 3: Varying Node Speed under Good Wireless Channel in Small-Medium Network			
Performance Metrics	**Node Speed (m/s)**	**Enhanced TCP-WELCOME**	**TCP-WELCOME**
Throughput (Kbps)	3	33.3328	30.6633
	4	33.2772	30.6213
	5	32.6889	30.3830
	6	32.0861	29.9204
	7	31.9699	29.8268
End-to-End Delay (Seconds)	3	2.0808	2.2620
	4	2.0843	2.2651
	5	2.1218	2.2829
	6	2.1617	2.3182
	7	2.1695	2.3254
Packet Delivery Ratio	3	0.2360	0.1988
	4	0.2312	0.1915
	5	0.1856	0.1413
	6	0.1658	0.1427
	7	0.1601	0.1394
Retransmission Attempt (Segments)	3	8,264	8,438
	4	8,274	8,449
	5	8,292	8,508
	6	8,305	8,528
	7	8,318	8,565

Table C.4: Simulation Results of Varying Node Speed under Good wireless Channel in Small-Medium Network (n=20)

Scenario 4: Varying Node Speed under Good Wireless Channel in Medium Network			
Performance Metrics	**Node Speed (m/s)**	**Enhanced TCP-WELCOME**	**TCP-WELCOME**
Throughput (Kbps)	3	8.6476	8.0232
	4	8.6101	7.9510
	5	8.4537	7.7717
	6	8.3846	7.7032
	7	8.2116	7.5384
End-to-End Delay (Seconds)	3	8.0207	8.6449
	4	8.0556	8.7234
	5	8.2047	8.9247
	6	8.2723	9.0040
	7	8.4466	9.2009
Packet Delivery Ratio	3	0.2360	0.1988
	4	0.2312	0.1915
	5	0.1469	0.1327
	6	0.1436	0.1288
	7	0.1309	0.1149
Retransmission Attempt (Segments)	3	12,341	12,420
	4	12,352	12,426
	5	12,370	12,432
	6	12,367	12,446
	7	12,370	12,445

Table C.5: Simulation Results of Varying Packet Length under Good wireless Channel in Small-Medium Network (n=10)

Scenario 5: Varying Packet Length under Good Wireless Channel in Small-Medium Network			
Performance Metrics	**Packet Length (Bytes)**	**Enhanced TCP-WELCOME**	**TCP-WELCOME**
Throughput (Kbps)	5,000	34.3192	32.4051
	10,000	34.3490	32.4746
	15,000	34.2467	32.3423
	20,000	34.2493	32.3034
	25,000	34.2461	32.2927
End-to-End Delay (Seconds)	5,000	2.0210	2.1404
	10,000	2.0193	2.1358
	15,000	2.0253	2.1446
	20,000	2.0252	2.1471
	25,000	2.0253	2.1479
Packet Delivery Ratio	5,000	0.4846	0.4555
	10,000	0.1389	0.1149
	15,000	0.1014	0.0876
	20,000	0.0718	0.0560
	25,000	0.0528	0.0454
Retransmission Attempt (Segments)	5,000	6,176	6,386
	10,000	6,189	6,396
	15,000	6,214	6,403
	20,000	6,211	6,403
	25,000	6,216	6,393

Table C.6: Simulation Results of Varying Packet Length under Good wireless Channel in Small-Medium Network (n=15)

Scenario 6: Varying Packet Length under Good Wireless Channel in Small-Medium Network			
Performance Metrics	**Packet Length (Bytes)**	**Enhanced TCP-WELCOME**	**TCP-WELCOME**
Throughput (Kbps)	5,000	32.6889	30.3830
	10,000	32.7148	30.3852
	15,000	32.5664	30.3983
	20,000	31.9828	29.9836
	25,000	31.7699	29.8446
End-to-End Delay (Seconds)	5,000	2.1218	2.2829
	10,000	2.1201	2.2827
	15,000	2.1298	2.2817
	20,000	2.1687	2.3133
	25,000	2.1832	2.3240
Packet Delivery Ratio	5,000	0.1856	0.1413
	10,000	0.0591	0.0448
	15,000	0.0399	0.0313
	20,000	0.0189	0.0144
	25,000	0.0189	0.0131
Retransmission Attempt (Segments)	5,000	8,292	8,508
	10,000	8,297	8,514
	15,000	8,311	8,580
	20,000	8,313	8,606
	25,000	8,317	8,678

Table C.7: Simulation Results of Varying Packet Length under Good wireless Channel in Medium Network (n=20)

Scenario 7: Varying Packet Length under Good Wireless Channel in Medium Network			
Performance Metrics	**Packet Length (Bytes)**	**Enhanced TCP-WELCOME**	**TCP-WELCOME**
Throughput (Kbps)	5,000	8.4537	7.7717
	10,000	8.3633	7.5270
	15,000	8.2170	7.3956
	20,000	7.9395	7.1151
	25,000	7.8057	6.9488
End-to-End Delay (Seconds)	5,000	8.2047	8.9247
	10,000	8.2934	9.2148
	15,000	8.4410	9.3786
	20,000	8.7361	9.7483
	25,000	8.8858	9.9816
Packet Delivery Ratio	5,000	0.1469	0.1327
	10,000	0.1207	0.0946
	15,000	0.0937	0.0556
	20,000	0.0740	0.0427
	25,000	0.0358	0.0251
Retransmission Attempt (Segments)	5,000	12,370	12,432
	10,000	12,345	12,397
	15,000	12,332	12,384
	20,000	12,332	12,384
	25,000	12,325	12,346

Table C.8: Simulation Results of Varying Network Sizes under Noisy wireless Channel

Scenario 8: Varying Network Sizes under Noisy Wireless Channel			
Performance Metrics	**Node Number**	**Enhanced TCP-WELCOME**	**TCP-WELCOME**
Throughput (Kbps)	5	136.0168	126.3620
	10	34.4078	32.2956
	15	32.7451	28.8865
	20	6.9796	5.9119
End-to-End Delay (Seconds)	5	0.5099	0.5489
	10	2.0158	2.1477
	15	2.1182	2.4011
	20	9.9375	11.7322
Packet Delivery Ratio	5	0.6644	0.6414
	10	0.3807	0.3639
	15	0.2031	0.1923
	20	0.1907	0.1619
Retransmission Attempt (Segments)	5	4,003	4,241
	10	6,186	6,409
	15	8,535	8,802
	20	12,288	12,750

Table C.9: Simulation Results of Varying Node Speed under Noisy wireless Channel in Small-Medium Network (n=10)

Scenario 9: Varying Node Speed under Noisy Wireless Channel in Small-Medium Network			
Performance Metrics	**Node Speed (m/s)**	**Enhanced TCP-WELCOME**	**TCP-WELCOME**
Throughput (Kbps)	3	35.4739	32.7959
	4	34.6619	32.5309
	5	34.4078	32.2956
	6	34.1217	32.1782
	7	33.8343	31.9163
End-to-End Delay (Seconds)	3	1.9552	2.1149
	4	2.0010	2.1321
	5	2.0158	2.1477
	6	2.0327	2.1555
	7	2.0500	2.1732
Packet Delivery Ratio	3	0.4015	0.3945
	4	0.3951	0.3807
	5	0.3807	0.3639
	6	0.3705	0.3556
	7	0.3626	0.3563
Retransmission Attempt (Segments)	3	6,116	6,313
	4	6,154	6,378
	5	6,186	6,409
	6	6,213	6,456
	7	6,236	6,500

Table C.10: Simulation Results of Varying Node Speed under Noisy wireless Channel in Small-Medium Network (n=15)

Scenario 10: Varying Node Speed under Noisy Wireless Channel in Small-Medium Network			
Performance Metrics	**Node Speed (m/s)**	**Enhanced TCP-WELCOME**	**TCP-WELCOME**
Throughput (Kbps)	3	33.2347	29.2070
	4	32.8700	29.0426
	5	32.7451	28.8865
	6	32.2174	28.7580
	7	31.7648	28.5305
End-to-End Delay (Seconds)	3	2.0870	2.3748
	4	2.1101	2.3882
	5	2.1182	2.4011
	6	2.1529	2.4118
	7	2.1835	2.4311
Packet Delivery Ratio	3	0.1991	0.1924
	4	0.1990	0.1934
	5	0.2031	0.1923
	6	0.2232	0.1973
	7	0.2320	0.2239
Retransmission Attempt (Segments)	3	8,510	8,787
	4	8,510	8,783
	5	8,535	8,802
	6	8,559	8,864
	7	8,557	8,944

Table C.11: Simulation Results of Varying Packet Length under Good wireless Channel in Small-Medium Network (n=10)

Scenario 11: Varying Packet Length under Noisy Wireless Channel in Small-Medium Network			
Performance Metrics	**Packet Length (Bytes)**	**Enhanced TCP-WELCOME**	**TCP-WELCOME**
Throughput (Kbps)	5,000	34.4078	32.2956
	10,000	34.2342	32.1626
	15,000	34.2182	31.9959
	20,000	34.0047	31.8401
	25,000	33.4694	31.5729
End-to-End Delay (Seconds)	5,000	2.0158	2.1477
	10,000	2.0260	2.1565
	15,000	2.0270	2.1678
	20,000	2.0397	2.1784
	25,000	2.0723	2.1968
Packet Delivery Ratio	5,000	0.3807	0.3639
	10,000	0.1370	0.1214
	15,000	0.1068	0.0868
	20,000	0.0755	0.0616
	25,000	0.0501	0.0388
Retransmission Attempt (Segments)	5,000	6,186	6,409
	10,000	6,177	6,416
	15,000	6,229	6,437
	20,000	6,227	6,438
	25,000	6,256	6,463

Table C.12: Simulation Results of Varying Packet Length under Good wireless Channel in Small-Medium Network (n=15)

Scenario 12: Varying Packet Length under Noisy Wireless Channel in Small-Medium Network			
Performance Metrics	**Packet Length (Bytes)**	**Enhanced TCP-WELCOME**	**TCP-WELCOME**
Throughput (Kbps)	5,000	32.7451	28.8865
	10,000	32.7294	28.7927
	15,000	32.0753	28.5126
	20,000	31.8406	28.3907
	25,000	31.6663	28.2591
End-to-End Delay (Seconds)	5,000	2.1182	2.4011
	10,000	2.1192	2.4089
	15,000	2.1624	2.4326
	20,000	2.1784	2.4431
	25,000	2.1903	2.4544
Packet Delivery Ratio	5,000	0.2031	0.1923
	10,000	0.0652	0.0589
	15,000	0.0397	0.0311
	20,000	0.0216	0.0154
	25,000	0.0160	0.0114
Retransmission Attempt (Segments)	5,000	8,535	8,802
	10,000	8,553	8,864
	15,000	8,626	8,926
	20,000	8,629	8,936
	25,000	8,635	8,952