Evaluation of Transport Layer Protocols for Wireless Multi-hop Networks

by

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DECLARATION

This dissertation represents the author's own work and has not been submitted in any form to another University for degree purposes. All sources used in this dissertation have been duly acknowledged.

S M Ncanana

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DEDICATION

My sons, Thobani and Mthobisi

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ABSRACT

Wireless Multi-hop Network (WMN) is a collection of wireless nodes that dynamically form a network without an infrastructure support. It is a promising technology for several interesting applications such as broadband home networking, community and neighborhood networks, coordinated management, intelligent transportation system. WMN is gaining significant attention as a feasible way for Internet providers (ISPs) and other end-users to establish robust and reliable wireless broadband service access at a reasonable cost.

Quality of Service (QoS) requirements and network services provided by WMNs vary from reliable file transfer to real-time. Multiple protocols for QoS such as routing protocols, topology controls, medium access control (MAC) and transport layer protocol (TLP) have been proposed for WMNs. The focus of this study is on the transport layer protocol (TLP) as the mechanism provides QoS in WMNs. Performance of Traditional TLP (such as TCP) deteriorates in WMNs. Traditional TLP designed for wired network and assumes that all packets losses are due to network congestion, whereas in WMNs congestion is not always the cause of packet losses. As result, several TLPs proposed to tailor for traditional TLPs in WMNs.

Despite the large number of TLPs proposed for WMNs, it is not clear which TLP performs better for WMNs, since they are not compared in a consistent manner. Therefore, we have designed literature analysis framework which enabled us to categorize the TLPs into four classes. This framework was also used to select the TLPs for evaluation. We selected eight (two from each category) TLPs plus two traditional

TLPs (i.e. TCP and UDP). Using simulation, we evaluated and compared the performance of the selected TLPs in WMN scenarios. In our evaluations and comparisons of the selected TLPs, the QoS performance metrics such as throughput, delay, and packet delivery ratio were considered. Two congestion and reliability related performance metric namely: number of packet retransmission (PR) and round trip time (RTT) were also considered. These performance metrics were varied with network size, number of flows, distance between nodes and simulation time.

The evaluations and comparisons studies enabled us to choose TLP suitable for WMNs. Our results show that, among the four TLP categories, Hybrid performs better than all TLP categories. Finally, we made recommendations for an ideal TLP applicable to WMNs that would improve efficiency of the TLP over WMNs.

CHAPTER ONE

INTRODUCTION

1.1. Preamble

A wireless multi-hop network (WMN) is a network of devices (nodes) which are connected by wireless links (De Couto, 2004). In some circumstances, each wireless link has a limited communication range and several pairs of nodes are unable to communicate directly, therefore, the nodes must forward data to each other through one or more cooperating intermediate nodes. WMN is a promising technology for several interesting applications such as broadband home networking, community and neighborhood networks, coordinated management and intelligent transportation systems (Gungor *et al*, 2006).

This technology is gaining significant attention as a feasible way for Internet providers (ISPs) and other end-users to establish robust and reliable wireless broadband service access at a reasonable cost. WMNs are divided into three classes as shown in Figure 1-1, namely: i. Wireless Ad hoc Networks, ii. Wireless Sensor Networks and iii. Wireless Mesh Networks. This study tends to provide network for a small rural village community with the potential to grow more in the future. The growth can be through the number of users in the network or the physical growth of the village. Therefore, with the ability of self-organization and self-configuration in which the components parts (nodes) can all connect to each other via multiple hops, Wireless Mesh Network can be deployed

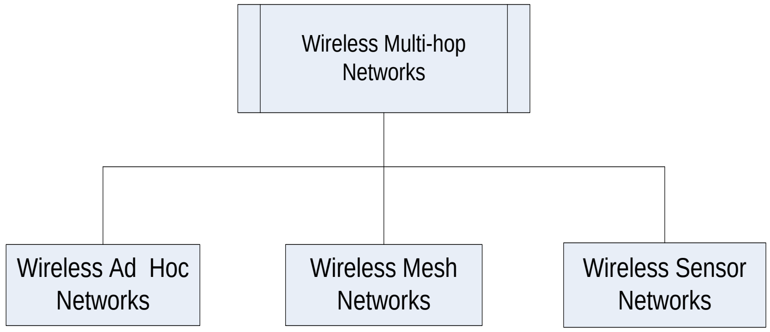

Figure 1-1: Types of wireless multi-hop networks

incrementally, one node at a time, as needed in a small and growing village. As more nodes are installed, the reliability and connectivity for the users increase accordingly. The static multi-hop wireless mesh network mode suits this rural village network, since when there is a physical growth you can only add new nodes and the network will configure and organize itself to the rest of the network.

Despite the advantages of WMNs quality of service (QoS) is still a challenge, due to the shared nature of the wireless medium (Feistel and Stanczak, 2007). Multiple mechanisms or protocols for QoS have been proposed for WMNs namely: transport layer protocols (TLP), routing protocols (RP), load balancing (LB), medium access control (MAC) and topology control (TC) (Feistel and Stanczak, 2007), (Jones *et al*, 2005).

The focus of our research is on evaluations and comparisons of the existing transport layer protocols (TLPs) applicable to WMNs. This chapter gives the background for TLPs that are applicable to WMNs. The problem of the existing TLPs applicable to WMNs is defined in the chapter as well as approach followed to solve the problem.

1.2. Background

Quality of Service (QoS) is the collective effect of the service performance which determines the degree of satisfaction of a user of the service (Masip-Bruin *et al*, 2006). Wireless multi-hop networks as an emerging network service, requires specialized QoS functionalities in order to perform and meet application requirements. The mechanisms that provide QoS in WMNs are usually not cross-layer aware (does not allow knowledge sharing between all layers of the network stack) (Jones *et al*, 2005).

As a result, the performance of the TLPs can degrade due to the faulty of other network layers a not transport layer which is main focus. Routing protocols have been used to support QoS as they determine the best path to use for traffic flow and can maintain this path for the duration of the flow (Du, 2004). Routing protocols can be utilized also to provide load-balancing in order to improve congestion, maximize network throughput and extending lifetime of network by distributing network traffic evenly (Gao *et al*, 2006).

Medium access control (MAC) regulates access to resources in the shared network environment (Reddy *et al*, 2007). TLPs are defined in the transport layer of the Open System Interconnection (OSI) model. The transport layer is responsible for QoS provisioning through flow control, congestion control, end-to-end connection setup, and end-to-end reliability. TLPs are responsible for connection establishment and attempt to ensure that all data are transmitted from source to target destination. Several transport layer protocols (TLPs) exist to date. The most widely used transport layer protocols for Internet applications are the Transmission Control Protocol (TCP) and the User Datagram Protocol (UDP). TCP is a reliable byte-stream connection-oriented, data ordered delivery and bi-directional transport layer protocol, whilst UDP is a connectionless, unreliable and unordered transport layer protocol, and it is designed for unicast multimedia applications that prefer timeliness of data to reliability. UDP has the advantage of not introducing additional delays due to retransmission as in TCP (Mujica *et al*, 2004). These traditional (TCP and UDP) TLPs were designed for wired networks and their performance has been found to be extremely poor in WMNs (Calagaz *et al*, 2004), (Gerla, *et al*, 1999), (Ludwig, 2000), (Balakrishnam *et al*, 1997), (Balakrishnam *et al*, 1996), and (Fu *et al*, 2003). Most of wireless TLPs were derived from wired TLPs. TLPs that are applicable to WMNs can be categorized into the following manner: i. TCP variants, ii. UDP variants, iii. Hybrid and iv. Entirely New TLPs (ENTs).TCP variants and UDP variants are extensions of TCP and UDP respectively.

WMNs are emerging and used as backbone network for accessing the Internet as well as for community networking. The traffic in WMNs originates from numerous applications which use different TLPs such as reliable and unreliable. As real-time applications coexist with TCP traffic (non real-time), therefore, improving only the performance of TCP protocol may not improve the overall performance of the WMNs (Franklin *et al*, 2008). Hybrid protocols such as LLAP and LLE-TCP were, therefore, developed to improve the overall performance of WMNs. Entirely New transport layer protocols have been proposed to avoid fundamental problems that exist within traditional TLPs (TCP and UDP) (Akyildiz *et al*, 2005) and their variants and also to tailor the requirements of the specific type of the network e.g. WMNs. The four categories (Akyildiz *et al*, 2005) of TLPs (TCP variants, UDP variants, Hybrid and ENT) as are not compared in a consistent manner. Thus, it is not clear which TLP, from which category is optimal for use with wireless multi-hop networks.

1.3. Statement of the problem

The performance of TLPs can be affected by different parameters and factors such as network congestion, MAC contention, interference, load imbalance and data flow in WMNs. There are several TLPs proposed in the literature to date. We were unable to identify an optimal TLP that is most suitable for use with wireless multi-hop networks (WMN), since they are not compared in a consistent manner. The fact that they are not compared in a consistent manner, disallows us to tell how much improvement has been done as far as the TLPs applicable to WMNs is concerned, and it is not clear which one is the best.

There is a problem in identifying an optimal performing TLP, since there is no standard comparison of them. It is therefore, necessary to come up with recommendations or design criteria for an ideal TLP through comparing the performance of the existing protocols. These recommendations should be considered by anyone who would like to design a new TLP applicable to WMNs in future.

1.4. Research Questions

Apposite to the study three research questions are posed:

1. How can we create a classification framework for existing protocols?

2. What factors must be considered when comparing TLPs for wireless multi-hop networks?

3. What features should an ideal transport protocol possess?

1.5. Rationale of the study

The design features for ideal TLP suitable for WMN will provide a starting point for researchers who are concerned with implementing TLPs to improve the WMN performance. There are many TLPs existing already, therefore, we are comparing the existing TLPs in order to make recommendations or design features of an ideal TLP suitable for WMNs, can alleviate problems such as network congestion, data flow, MAC contention, network interferences, network bottleneck, and load imbalance in WMNs.

1.6. Research Goal and Objectives

The goal and the objectives of this study are presented in subsection 1.6.1 and subsection 1.6.2, respectively.

1.6.1. Goal

The goal of the study is to compare existing transport layer protocols and make recommendations or design features for an ideal TLP applicable to WMN.

1.6.2. Objectives

i. To develop the framework for the analysis of the related work.

ii. To use the framework developed in objective one, to choose a sample of transport layer protocols.

iii. To simulate and evaluate the selected transport layer samples.

iv. To recommend the design criteria/features an ideal transport layer protocol should have.

1.7. Research Methodology

This section presents the research method used. This research method consists of the following three steps:

i. Literature Survey using a special categorization framework

This section answered the first research question and we used it as a vehicle for achieving the first and second objective. We surveyed relevant literature to discover what other scholars have done to solve the problem (what are existing solutions) and how they went about it. The literature review enabled us to come up with a framework to classify the TLPs. The framework and classification allowed us to identify transport layer protocols with common features, protocols utilizing common approach and determining the trends of the TLPs development, so as to choose the samples for evaluations.

ii. Case-Study: by evaluating the performance of the simulated representative protocol samples

The case-study is a means of fulfilling objectives number three and four. Two existing TLPs were selected from each category and simulated using a network simulation tool, called NS2. The following QoS performance metrics were considered: throughput, delay, packet delivery ratio, round trip time and packet retransmission. The network size, number of flows, distance between nodes and simulation time would be varied.

iii. Recommendation of the design features for ideal transport TLP for WMNs

This answered objective number four. The knowledge gathered from the literature and the simulation (objective three) results was analyzed and utilized in recommending design features/criteria for an ideal TLP (perfect protocol) for WMNs. The advantage of our methodology is that, there are many TLPs that were simulated and implemented in NS2 already. Therefore, we have added some modification on the existing TLPs.

1.8. Organization of the Dissertation

The rest of the dissertation is organized as follows: Chapter Two presents the analysis of TLPs for WMNs. The framework used for literature review is also developed in this chapter. We utilized this framework to select the evaluation samples (TLPs evaluated for comparisons). In Chapter three we give a detailed description of the TLPs selected for performance evaluation and comparison. Chapter Four details the implementation results and evaluation of the selected TLPs of WMNs. Analysis of the results is also given in the same chapter. Finally, Chapter Five concludes the study and highlights future work.

CHAPTER TWO

LITERATURE REVIEW

2.1. Introduction

In the previous chapter we introduced our study area and presented the background for TLPs in WMNs. In this chapter we review and analyze the existing TLPs applicable to wireless multi-hop networks (WMNs). A number of researchers have decided to develop TLPs from the scratch to improve the performance of WMNs. This action by researchers has led to an immediate increase in the number of TLPs. While there are multiple TLPs that exist in the literature, these TLPs are not compared in a consistent manner when applied in WMNs.

Therefore, the identification of the optimally performing TLP in the WMN domain remains a challenge. In Chapter One (Section 1.2), we identified the need for using the existing scholarship to determine an optimal performing TLP applicable to WMNs and come up with the recommendations for features of an ideal TLP. The framework provided in the chapter describes our approach to selecting appropriate TLPs for this study.

Section 2.2 details a TLP classification framework which is subsequently used to critically analyze a number of TLPs that have been proposed in the literature for WMNs. Research trends in TLPs applicable to WMNs is presented in section 2.3. Section 2.4 gives the framework for analyzing the related work. In Section 2.5 we give the analysis of the TLPs applicable to WMNs. The selection of TLPs for evaluation is presented in

Section 2.6.

2.2. Classification of Transport Layer Protocols (TLPs).

A number of TLPs have been proposed for wireless multi-hop networks (WMNs) (Navatatnam *et al*, 2007), (Sundaresan *et al*, 2003). Some TLPs are the extensions of traditional TLPs (Akylidiz *et al*, 2005) and some are designed specifically for Ad hoc and Wireless Mesh networks (Su and Zurich, 2005). Most of the TLPs were proposed due to the fact that TCP cannot perform well in wireless environment for reasons such as high bit error ratio, route failure, mobility, link failure and medium contention. WMN TLPs are classified into four categories (Akylidiz *et al*, 2005). These categories are clearly shown in Table 2-27. The categories are as follows:

1. TCP variants
2. UDP variants
3. Hybrid TLPs
4. Entirely New TLPs.

TCP variant is an improved version of TCP. TCP degrades when used in wireless networks because it does not distinguish between congestion and non-congestion losses of packets (Fu *et al*, 2003). While congestion is the main source of packet losses in wired networks (Holland, and Vaidya, 1999), (Rath and Sahoo, 2005), this is not the only case in wireless networks, because the losses in the wireless networks environment are sometimes due to other causes such as medium related errors (e.g. attenuation) (Eckhardt and Steenkiste, 1996), mobility related routing (link breakage) failures (Holland and Vaidya, 1999), MAC contention and erroneous wireless channel (Franklin *et al*, 2008).

UDP variants are enhanced versions of UDP. In real-time delivered applications, UDP is employed as the TLP, although UDP cannot guarantee real-time delivery (Postel, 1980), (Casner and Jacobson, 1999). Therefore, protocols such as Real-Time Transport Protocol (RTP) (Schulzrinne *et al*, 1996), (Laron , 1999) and RTP Control Protocol (RTCP) (Schulzrinne *et al*, 1996) were proposed to work over UDP. Ratio Control Protocol (RCP) (Fu *et al*, 2003) is one of the protocols designed to alleviate the congestion control challenge within the UDP.

Hybrid protocols are a combination of TCP and UDP based protocols. This is because in wireless multi-hop networks, a number of clients can generate TCP and UDP traffic which goes in the same multi-hop path from one edge node to another edge node or edge node to gateway node (Franklin and Murthy, 2008). Applications such as audio and video streaming coexist with TCP traffic, but improving only the performance of TCP protocol, not the overall performance of the WMNs is not enough. Therefore, Hybrid transport layer protocols have been proposed to cater for both UDP and TCP traffic.

Most of the Internet based applications use TCP as a transport protocol, since it provides end-to-end reliable data transmission. Many applications such as audio and video streaming utilize UDP as a transport protocol, since they require faster delivery of data rather than reliable transmission. As a result, networks such as WMNs may be used to carry traffic from numerous applications which use different transport protocols, thus requiring TLPs that can cater for both real time and non-real time applications.

Entirely New TLPs (ENTs) are protocols designed to avoid the fundamental problems in both TCP and UDP. Such protocols are developed to tailor to the characteristics of WMNs. By using ENTs with a new set of mechanisms for reliable data transport, WMNs achieve an improved performance when compared to TCP and UDP variants (Su and Zurich, 2005), (Akyildiz and Wang, 2005).

Despite the fact that there are various categories of TLPs, it is not clear which is an optimal TLPs applicable to WMNs and from which category, since they are not compared in a consistent manner. Therefore, it is absolutely necessary to consistently evaluate and compare the existing TLPs in order to suggest or recommend features of an ideal TLP applicable to WMNs. Such recommendations would be taken into account when developing a TLP applicable to WMNs in future.

In this chapter we present a critical analysis of the existing TLPs and their applicability to WMNs. The analysis is based on the specially designed framework in Table 2-1.

2.3. Research Trends in TLPs for WMNs

The research trends for this study are given in a pictorial form in Figure 2-1. The trends indicate the focus of the researchers concerning transport layer protocols from 1980 up to 2010. This research trend clearly indicates what problem was being solved by TLP in the particular year. Our trends start from the wired technology until the wireless technology come into play. The first TLP (TCP) came into play as the network congestion caused the poor performance of the network as a whole.

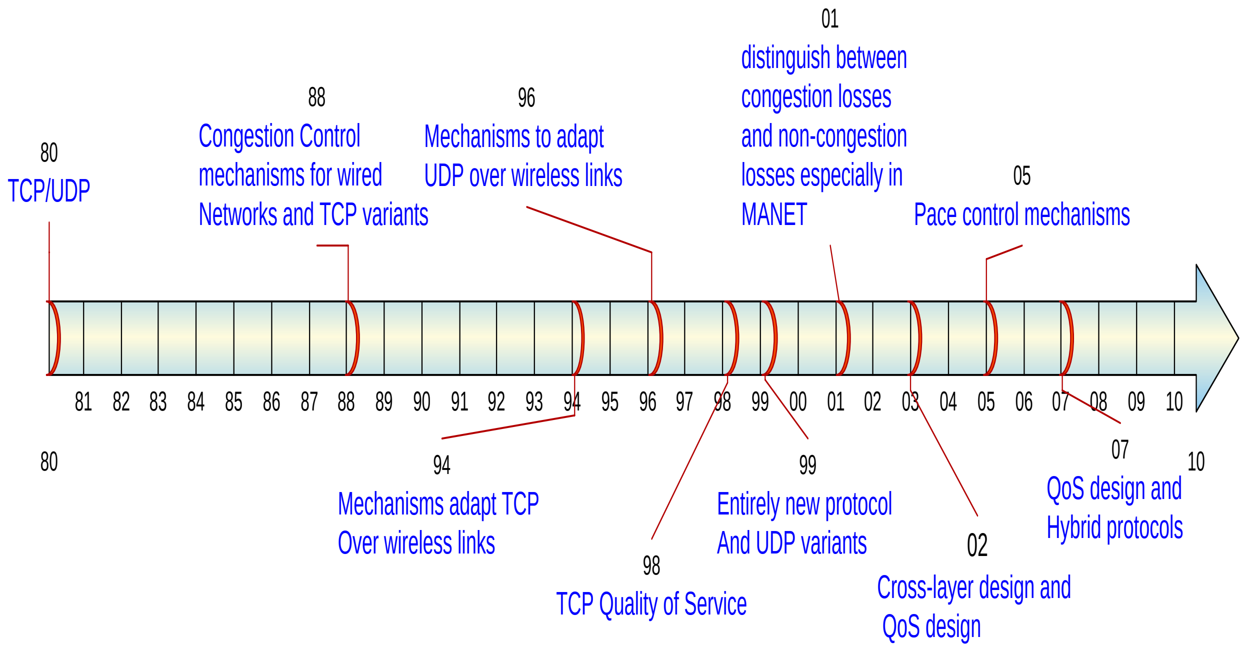

Figure 2-1: Trends for TLPs of Wireless Multi-hop networks

Such situation occurred on the early Internet (wired network) and lead to the development of the TCP with congestion control mechanism (Jacobson and Karels, 1988). Initially TCP was for wired network where congestion is a regular cause for packet loss (Jacobson and Karels, 1988), (Holland and Vaidya, 1999). As much as TCP provides the congestion control mechanisms for wired network but it was not suitable for real time applications. Therefore, UDP came into consideration to set TCP free from real-time applications problems associated with audio and video streaming (Postel, 1981), (Casner and Jacobson, 1999).

As from 1988, TCP was being upgraded, different version of TCP using various approaches such as Slow Start and Congestion Avoidance (Jacobson, 1988), Fast Retransmit and Fast Recovery (Braden *et al*, 1998), (Mathis *et al*, 1996) were proposed to minimize the network congestion. Many TLPs were designed to extend TCP (TCP related) in order to increase the level of congestion control. UDP related transport layer protocols were developed to work out the UDP-specific problems on wired networks. In 1990s, the wireless technology was introduced as far as the TLPs are concerned. When the wireless technology comes into play, the traditional TLPs (TCP and UDP) were adopted in the wireless network environment. The traditional TLPs performed poorly in wireless environments. TCP assumes that all packet losses taking place in any network are due to network congestion (Gerla *et al*, 1999), (Fu *et al*, 2002), (Fu *et al*, 2003). In the late 1990s and early 2000s, the focus of researchers shifted from network congestion to Quality of Service (QoS).

As the wireless technology emerges, different types of networks come into play and more new networks problems discovered. As the number of network type increases, TLPs also increase, with some extending the existing TLPs and some designed from scratch for specific type of network. The framework used to analyze the TLPs for wireless multi-hop networks is presented in the following section, tabulated in Table 2-1.

The review of related work was based on the framework shown in Table 2-1. The properties of the framework were identified from the literature (Wang *et al*, 2002), (Navaratnam *et al*, 2007), (Franklin and Murthy, 2008). The major aim of the framework was to categorize TLPs and be used to thoroughly analyze the existing work in the field to enable easy selection of transport layer protocols for evaluations and comparisons.

2.4. Framework for Analyzing Related Work of the TLPs for WMNs

Table 2-1: TLPs Literature Analysis Framework

Framework Attribute	Description
Year	In which year was the transport layer protocol developed? This helps us to find out, what were the main problems of the TLPs in the beginning and what are the current problems. It gives us the trends of the TLPs.
Problem	What is the main problem being solved by the protocol. This permits us to identify protocols solving the same problem utilizing different approaches and enables us to identify various types of problems being solved by TLPs.
Evaluation Approach	How did they go about providing the proof of concept? We should be able to identify protocols using common approach, so that when choosing the representative samples we take one from those utilizing the common approach.
Type	TLPs classified into four groups as mentioned in the beginning of this chapter. Since there are various types of networks running various types of applications, so the classification of TLPs us to identify the optimal type of TLPs, and which application is suitable for it.
Does it works in conjunction with other protocols	Some protocols can employ certain features of other protocols to solve different problems or same problem and some protocols cannot work hand in hand with other protocols.
Control of network Congestion	Some TLPs are able to control the situation when the excessive amount of data is being sent into the network. This is the major problem that almost all TLPs are trying to solve in the network as whole regardless of whether is a wired or wireless network.
Real-time or non real-time	There are protocols designed for real-time, non-real-time and some for both real and non-time applications. This should tell which protocol you should use when you are dealing with real-time application and non-real time applications.
QoS aware	What is the performance level of a service offered by the transport layer protocol to the user? We all looking for the protocol that will yield the services of a highest quality, so this will tell which protocol you can choose for your service.
Cross-layer aware	Does it allow the knowledge sharing between all layers to obtain the highest possible adaptively to take place.

	The protocol might perform very well but due to some problems from other source (i.e. other layers), then the performance degrades
Implemented in real-world scenario	At what degree the TLPs have been implemented in real-world scenario.

2.5. Analysis of TLP Applicable to WMNs

The review of TLPs applicable to WMNs is presented in the following order, TCP variants, UDP variants, Entirely New TLPs (ENTs) and Hybrid TLPs to depict the evolution of the TLPs proposed in the literature to date.

2.5.1. TCP Variants

TCP variants enhanced TCP in order alleviate the poor performance offered by the TCP in wireless multi-hop networks (Brown and Singh, 1997), (Chandran *et al*, 1998), (Wang and Zhang, 2002), (Chen *et al*, 2004), (ElRakabawy *et al*, 2005). There is a huge number of TCP variants proposed in the literature. The TCP related TLPs reviewed in the study are as follows:

2.5.1.1. Indirect-TCP (Bakre and Badrinath, 1995)

Indirect-TCP (I-TCP) is the protocol designed due to the fact that the traditional TLPs suffer from poor performance in wireless mobile ad hoc networks. The poor performance is due to the repeated disruption in wireless network connectivity due to mobility and unreliable nature of wireless link. I-TCP uses the resources of Mobility Support Routers (MSRs) to provide transport layer communication between mobile nodes.

Table 2-2: Characteristics of I-TCP (Bakre and Badrinath, 1995)

Characteristics	Result
Year	1995
Problem	Link/connection failure in the wireless multi-hop networks (network congestion)
Approach	splits the TCP connection into two separatio connections
Work in conjunction with other protocol	Not specified
Congestion control	Yes
Real-time/non-real-time	Non-real-time
QoS aware	No
Cross-layer aware	No
Real-world scenario	No

In I-TCP, TCP connection between the mobile nodes is established by MSR. If the mobile node moves to another position during life time of TCP connection, the new MSR takes over the connection from the previous MSR. For I-TCP the most applications that utilize TCP for bulk data transmission have some support built-in for application layer acknowledgments and error recovery. These acknowledgments are frequently required since TCP does not provide any notification to the source application when data is received by the peer application.

I-TCP is not QoS and cross-layer aware. Evaluation of the I-TCP was based on real world scenario i.e. on testbed. I-TCP has shown the improvement compared to the traditional TLP (TCP) on the mobile wireless environment only. Therefore, it is not clear whether it can outperform the TLP from other categories such UDP variants, ENTs and Hybrid TLPs. Additional characteristics of this TLP are listed in Table 2-2. M-TCP was developed to provide a solution to the problem of improving TCP's efficiency in mobile wireless network environment.

2.5.1.2. M-TCP (Brown and Singh, 1997)

Table 2-3: Characteristics of M-TCP (Brown and Singh, 1997)

Characteristics	Result
Year	1997
Problem	High bit error ratio(BER), frequent disconnection and low variable bandwidth on TCP in wireless links
Approach	Split connection approach
Work in conjunction with other protocol	Not specified
Congestion control	Yes
Real-time/non-real-time	Non-real-time
QoS aware	yes
Cross-layer aware	No
Real-world scenario	No

Like I-TCP, M-TCP improves TCP's performance in the following aspects: high bit error ratio (BER) and frequent disconnections of mobile user. M-TCP is different from I-TCP in case of maintaining end-to-end TCP semantics. M-TCP is also designed in such way that it is able to deal with problems caused by lengthy disconnection. The M-TCP protocol is able to adapt to dynamically changing bandwidth over already starved wireless links.

M-TCP ensures that handoffs (as mobile nodes roams) are efficient. To implement this protocol, split connection (like in I-TCP) approach was used because it fits well with the general design philosophy and because it allows the mobile network to respond better to disconnections and low wireless bandwidth. Similar to I-TCP, M-TCP performance was tested against TCP only while there are several TLPs from other categories which are not considered. A summary of this work can be found in Table 2-3.

2.5.1.3. WTCP (Ratnam and Matta, 1998)

WTCP is the new scheme to enhancing the performance of the TCP in network with wireless links. Like M-TCP, it also maintains end-to-end TCP semantics. WTCP effectively shields wireless link errors and attempts to hide the time spent by the node to recover locally so that the TCPs round trip time (RTT) estimation at the source is not affected. The shielding process is critical or else the ability of the source to effectively detect congestion in the intermediate node will be hindered. The Three above-mentioned TLPs such as I-TCP, WTCP and M-TCP were designed to deal with the problem such as link and route failure and a few errors experienced by TCP when it is applied to WMNs.

These three protocols deal with mechanisms to reduce link and route failure, but they do not consider the same features when comparing their performance to TCP, i.e. M-TCP considers transfer time against disconnection length performance measure parameters whereas WTCP considers throughput against mean bad duration. M-TCP and WTCP provide different mechanisms solving the same problem but not are compared in a consistent manner. The performance of WTCP was compared against three TCP variants namely Snoop, TCP-Tahoe and I-TCP. WTCP is compared with TCP related TLPs only while the other three other categories of the TLPs which are not measured. Table 2-4 gives additional characteristics of WTCP.

Table 2-4: Characteristics of WTCP (Ratnam & Matta, 1998)

Characteristics	Result
Year	1998
Problem	Data losses because of wireless channel errors or host mobility when using TCP.
Approach	Splits connection approach (split the TCP connection into two separatio connections)
Work in conjunction with other protocol	Not specified
Congestion control	Yes
Real-time/non-real-time	Non-real-time
QoS aware	No
Cross-layer aware	No
Real-world scenario	No

2.5.1.4. TCP-Feedback (Chandran *et al*, 1998)

A mobile ad hoc network topology is often changed due to the movements of the nodes and sudden movements of the node cause the sudden packet losses and delays. TCP misinterprets such impairments as congestion and invokes congestion control mechanisms which result in unnecessary retransmission and loss of throughput. Chandran *et al* (1998) proposed a feedback scheme, whereby the traffic source can distinguish between route failure and network congestion losses. TCP-Feedback provides the source with explicit notification routing failure using a Route Failure Notification (RFN), and when a new route has been discovered using Route Re-establishment Notification (RRN) message.

When a route to the destination is currently unavailable, i.e. after receiving an RFN message, the TCP-Feedback source enters a “snooze” state. State values such as timers and window sizes are frozen. Intermediate nodes generate RFN message when link failures on routes are detected. Once a node was previously generating or forwarding, an RFN learns about a new route to the destination node, a RRN message is generated and sends it to the source node. When the source

Table 2-5: Characteristics of TCP-Feedback (Chandran *et al*, 1998)

Characteristics	Result
Year	2001
Problem	Packet losses and delays due to suddenly movement of the host.
Approach	Feedback mechanism
Work in conjunction with other protocol	Not specified
Congestion control	Yes
Real-time/non-real-time	Non-real-time
QoS aware	Not specified
Cross-layer aware	No
Real-world scenario	No

node receives a RRN, it resumes the TCP session is resumed with the previous state value. In order to prevent TCP-Feedback session from remaining in a snooze state indefinitely in the event of last RRN message, an additional timeout is used as a feedback. TCP-Feedback protocol compares itself only with TCP in ad hoc network when sending with and without feedback. Some additional characteristics of this work are presented in Table 2-5.

2.5.1.5. Explicit Link Failure Notification (Holland and Vaidya, 1999)

Explicit Link Failure Notification (ELFN) was proposed to solve the TCP problem of not performing well when it is applied in wireless multi-hop networks. ELFN means feedback from lower layers to notify TCP explicitly about link or routing failures. In case of such a failure, the source enters standby mode, which is the equivalent to TCP-F's snooze state. In contrast to the TCP-F proposal no explicit notification in case of a re-established route is used.

Table 2-6: Characteristics of ELFN (Holland and Vaidya, 1999)

Characteristics	Result
Year	1999
Problem	Link and route failure in mobile ad hoc connection failure in the wireless multi-hop networks (network congestion)
Approach	Explicit Link Failure notification
Work in conjunction with other protocol	Not specified
Congestion control	Yes
Real-time/non-real-time	Non-real-time
QoS aware	No specified
Cross-layer aware	No
Real-world scenario	No

Instead, a TCP-ELFN source sends probe packets in regular intervals when in a standby mode. A standby mode is left as soon as a probe packet is acknowledged by the destination node. Likewise, for route failure notifications no special control packet is introduced in ELFN. The authors propose to either piggyback the notification message onto a route failure message sent by the routing protocol (as used, for example, in Dynamic Source Routing), or to use an ICMP (Internet Control Message Protocol) host unreachable message for that purpose. ELFN has become very well-known and has served as a basis for many later approaches.

Although ELFN can indeed improve the performance of TCP; there are situations where severe performance degradation is possible (Monks e*t al*, 2000). This is mainly true for conditions where you find a number of working links where an ELFN-like causes a TCP performance to be more aggressive. As a result, the MAC layer conflict is normally higher. The MAC conflict in turn causes more interference, higher packet loss rate on the MAC layer and finally the fake link breakdown detections. Therefore, incorrect route failure notifications are sent and unnecessary route discoveries are performed. This observation is quite fundamental and should generally be taken into account in wireless multi-hop networks. It does not only apply to ELFN or TCP-Feedback, but also to other protocols such as TCP Door and EXACT. The problem with ELFN is that still a number of data packets and ACKs may get lost before the state is frozen (Yu, 2004). MAC contention has the negative effects after the state is restored, because missing packets or missing ACKs will then cause timeouts or duplicate ACKs.

The comparisons of the ELFN with other TCP variants such as base TCP-Reno and TCP-Reno was based on simulation using NS network simulator from Lawrence Berkeley National Laboratory (LBNL) (Fall and Varadham, 1998). Only TCP variant TLP was considered for comparisons. See Table 2-6 for some additional characteristics of ELFN.

2.5.1.6. TCP Door (Wang and Zhang, 2002)

TCP Detection of Out-of-Order (TCP Door) focuses on the idea that out-of-order (OOO) packets can happen frequently in WMNs environment as a result of node mobility, and it might be enough to indicate link failure inside the network. TCP Door detects OOO events and responds accordingly. Since not only data packet but also acknowledgement (ACK) packets can experience OOO deliveries, TCP Door implements detection mechanisms in both source and destination.

This is achieved by adding a one-byte option for ACKs and a two-byte option for data packets in the TCP options. For every data packet the source increments its own stream sequence number inside the two-byte option regardless of whether it is a retransmission

Table 2-7: Characteristics of TCP Door (Wang and Zhang, 2002)

Characteristics	Result
Year	2002
Problem	Link failure inside the network
Approach	TCP door implements detection mechanisms at both source and destination
Work in conjunction with other protocol	Not specified
Congestion control	Yes
Real-time/non-real-time	Non-real-time
QoS aware	No
Cross-layer aware	No
Real-world scenario	No

or not, as opposed to standard TCP which does not increment the sequence number of retransmitted packets. This enables the destination to detect OOO data packets and notify the source via an ACK. The destination subsequently increments its own ACK stream sequence number inside the one-byte option for every retransmitted ACK so that the source can distinguish the exact order of packets sent even if it is a retransmission. These mechanisms provide the source with reliable information about the order of the packet stream in both directions, allowing the TCP source to act accordingly.

A TCP Door source can respond to OOO events by temporarily disabling congestion control and instant recovery during congestion avoidance. When congestion control is disabled, the TCP source keeps its state variables constant for a while (T1) after the OOO detection in order to avoid unnecessary congestion control invocation. In the instant recovery mechanism, when an OOO condition is detected TCP source checks if the congestion control mechanism has been invoked in the recent past (T2). If so, the connection state prior to the congestion control invocation is restored, since such an invocation may have been caused by temporary disruption instead of by congestion

Table 2-8: Characteristics of Snoop (Ch Ng *et al*, 2002)

Characteristics	Result
Year	1995
Problem	Link/connection failure in the wireless multi-hop networks
Approach	Implement the Snoop agents in the OPNET WLAN device.
Work in conjunction with other protocol	No
Congestion control	Yes
Real-time/non-real-time	Non- Real-time
QoS aware	No
Cross-layer aware	No
Real-world scenario	No

itself. The TCP Door is another like TCP-Feedback (Chandran *et al*, 2001) and ELFN (Holland and Vaidya, 1999), which provide TCP source with information about the link and rout failures so that it can avoid respond to failures as if congestion occurred, but it was compared to traditional TLP (TCP) only. TCP Door, TCP-Feedback and ELFN provide solution to the similar problem through different approaches, but they are not compared in a consistent manner. A summary of this work can be found in Table 2-7.

2.5.1.7. Snoop (Ch Ng *et al*, 2002)

Snoop is first TLP selected from TCP variants category for evaluations and comparisons. Snoop is fully reviewed in Chapter Three. Table 2-8 presents the summary of the characteristics of this work.

2.5.1.8. ST-PD (Xu *et al*, 2002)

ST-PD employs a self-tuning proportional and derivative (ST-PD) control based TCP congestion control protocol to decouple the congestion control and error control

Table 2-9: Characteristics of ST-PD (Xu *et al*, 2002)

Characteristics	Result
Year	2002
Problem	Network congestion
Approach	uses a PD controller to change the TCP congestion window size to keep the buffer occupancy of the bottleneck node on the connection path at a desired operating level
Work in conjunction with other protocol	Yes
Congestion control	Yes
Real-time/non-real-time	Non-real-time
QoS aware	No
Cross-layer aware	No
Real-world scenario	No

functionalities. It is control-theory based and uses a proportional and derivative (PD) controller to change the congestion window size to keep the buffer occupancy of the bottleneck node on the connection path at a desired operating level. The input to the PD controller is the control error, which is the difference between the designed buffers occupancy and the observed buffer occupancy. The output of the PD controller is the TCP congestion window adjustment. Owing to the fact that the TCP operates over a variety of networking environments, the bandwidth-delay product of different TCP connections can vary significantly.

To handle bandwidth-delay, control gains of the PD controller are tuned online by a fuzzy logic controller to estimate bandwidth-delay product of the TCP connection.ST-PD is an end-to-end design that uses acknowledgement from destination to probe available bandwidth without causing buffer overflows. ST-PD protocol does not support QoS and cross-layer aware phenomenon. During ST-PD performance evaluation through NS2, ST-PD performance was compared with the performance of TCP without considering other

Table 2-10: Characteristics of (TCP-AP) (ElRakabawy *et al*, 2005)

Characteristics	Result
Year	2005
Problem	Packet bursts result in increased contention on the wireless channel
Approach	TCP end-to-end together with coefficient of variance (i.e. measures the degree of contention on the network path).
Work in conjunction with other protocol	No
Congestion control	Yes
Real-time/non-real-time	Non- Real-time
QoS aware	Not specified
Cross-layer aware	No
Real-world scenario	No

categories of TLPs. A summary of the characteristics of this work is presented in Table 2-9.

2.5.1.9. TCP-AP (ElRakabawy *et al*, 2005)

TCP Adaptive Pacing (TCP-AP) is the second TCP variants selected for evaluations and comparisons. The critical analysis for TCP-AP is presented in Chapter Three. Additional characteristics of this work are shown in Table 2-10.

2.5.1.10. Explicit rate-based flow ConTrol (Chen *et al*, 2004)

Explicit rate-based flow ConTrol (EXACT) is ratio based and is supported by the network itself, i. e., by the intermediate nodes. These nodes have dedicated state variables for all flows passing through them. All nodes determine their current bandwidth to their neighbors and calculate local fair bandwidth shares for all flows. Explicit ratio

Table 2-11: Characteristics of EXACT (Chen *et al*, 2004)

Characteristics	Result
Year	2004
Problem	Link and route failure in mobile ad hoc because of TCP's implicit AIMD flow control
Approach	Explicit rate-based flow control mechanism
Work in conjunction with other protocol	Not specified
Congestion control	Yes
Real-time/non-real-time	Non-real-time
QoS aware	Not specified
Cross-layer aware	No
Real-world scenario	No

information is inserted into all passing packets by the intermediate nodes to transmit the minimum bandwidth at the bottleneck to the destination of the flow. Each node checks whether the ratio it can supply for the flow of a packet it processes is lower than the ratio currently specified in the packet header. In this case, the lower ratio is written into the header before the packet is forwarded. Thus the bottleneck ratio is reported in the destination nodes. The lowering ratio mechanism is used twice, i. e. on two different header fields. One field contains the current ratio of the source and another one the ratio requested by the sending application.

On the one hand, with this procedure it is possible for the intermediate routers not to give a flow more bandwidth than it needs, and, on the other hand, the source is notified when it is allowed to increase its ratio above the current level. A safety window prevents the source from overloading the network in case of a route failure. A source is not allowed to have more unacknowledged packets underway than the size of the safety window. Some limitations on EXACT's practical usage and scalability might be imposed by the fact that it requires explicit state information for each flow in each intermediate node. By NS2 simulations, it was shown that EXACT outperforms TCP in terms of fairness and efficiency, in highly dynamic mobile ad hoc network environment where flow control is huge problem. The other TLPs from various categories such as UDP variants, Entirely New and Hybrid TLPs are not included for comparisons. A summary of the characteristics of this work can be found in Table 2-11.

In all TLPs we review under TCP variant category, there is no TLP that has compared itself with TLP from all four categories. TLPs have been compared either with TCP or TCP variants. Therefore, TCP variants are not compared in a consistent manner. In the following Section (2.5.2) we reviewed the UDP related transport layer protocols

2.5.2. UDP Variant

A huge number of time-sensitive applications such as Audio and Video streaming (real-time applications) utilize UDP. UDP is connectionless and time-sensitive (Xylomenos and Polyzoz, 1999). It requires faster delivery of data rather than reliable transmission. UDP has an advantage of not introducing additional delays to the carried data due to retransmission as in TCP (Mujica *et al*, 2004).

To support end-to-end delivery of real-time traffic, UDP instead of TCP is applied as a transport layer protocol (Akyildiz *et al*, 2005). Several TLPs developed to extend UDP (UDP variants) for wireless multi-hop networks. The following are UDP related TLPs reviewed and critically analyzed for wireless multi-hop networks:

2.5.2.1. M-UDP (Brown and Singh, 1996)

Table 2-10: Characteristics of M-UDP (Brown and Singh, 1996)

Characteristics	Result
Year	1996
Problem	In wireless mobile environment, high bit-error ratio (resulting in lost data)
Approach	Split connection approach
Work in conjunction with other protocol	Not specified
Congestion control	Yes
Real-time/non-real-time	Real-time
QoS aware	Yes
Cross-layer aware	Not specified
Real-world scenario	No

If UDP is used unmodified over a wireless channel a large percentage of packets could be lost due one of two conditions. Wireless links tend to be susceptible to bit errors. This problem may be alleviated to some degree by using some form of Forward Error Correction FEC (Clark, 1982) encoding. The protocol is based on an idea similar to the one used in I-TCP（Bakre *et al*, 1995) and M-TCP (Brown and Singh, 1996). The UDP connection is broken into two at some nodes near the non-static user.

In order to keep the number of lost packets small, this node tries to use any free bandwidth to retransmit lost packets. The M-UDP was implemented in (NetBSD) and compared its performance against that of UDP (traditional TLP) in an experimental mobile network that was developed at the University of South Carolina. The results of M-UDP depicted better performance than the traditional UDP. Although ENT and Hybrid TLPs were not yet proposed but even TCP variant which was already there was not considered for comparisons. Some additional characteristics of this work are given in Table 2-12.

2.5.2.2. RAP (Rejaie *et al*, 1999)

Table 2-11: Characteristics of RAP (Rejaie *et al*, 1999)

Characteristics	Result
Year	1999
Problem	non-congestion-controlled real-time applications and congestion control and loss detection
Approach	Additive-increase, multiplicative-decrease (AIMD) algorithm, rate-based congestion control.
Work in conjunction with other protocol	Not specified
Congestion control	Yes
Real-time/non-real-time	Real-time
QoS aware	Not specified
Cross-layer aware	Not specified
Real-world scenario	Yes

The Ratio Adaptive Protocol (RAP) protocol machinery is mainly implemented at the source. A RAP source sends data packets with sequence numbers, and a RAP sink acknowledges each packet, providing end-to-end feedback. Each acknowledgment (ACK) packet contains the sequence number of the corresponding delivered data packet. Using the feedback, the RAP source can detect losses and sample the round trip time (RTT). To design a ratio-adaptation mechanism, three issues must be addressed (Jain, 1989).

These are the decision function, the increase or decrease algorithm, and the decision frequency. RAP was evaluated through extensive ns2 simulation (McCanne and Floyd, 1995), and compared it to TCP Tahoe, TCP Reno, New Reno (Fall and Floyd, 1996) and Sack (Mathis *et al*, 1996) and also run real world experiment. Although RAP was compared to four TLPs but all of these TLPs are TCP variants. Thus UDP variant category was not included. Table 2-13 presents additional characteristics of RAP.

2.5.2.3. TFRC (Floyd, 2005)

Table 2-12: Characteristics of TFRC (Floyd, 2005)

Characteristics	Result
Year	2005
Problem	TCP link fail when sending audio and video.
Approach	An equation based ratio control mechanism for unicast UDP flows.
Work in conjunction with other protocol	Yes
Congestion control	Yes
Real-time/non-real-time	Real-time
QoS aware	Not specified
Cross-layer aware	Not specified
Real-world scenario	Yes

TCP Friendly Ratio Control (TFRC) attempts to match the long-term throughput of TCP and is smooth, fair and TCP friendly in wired networks. However, with the increasing popularity of wireless and mobile devices, it is highly desirable to have video transport schemes also work properly across wireless networks. Unfortunately, wireless links are usually error-prone. Since TFRC attempts to faithfully match the throughput of TCP, it suffers the same efficiency problem in the presence of moderate to high random errors (Su *et al*, 2001). Recently, TFRC has been extended for better efficiency in wireless networks.

Among the solutions are TFRC Wireless (Cen *et al*, 2003) and MULTFRC by Chen and Zakhor (2004). In particular, Video Transport Protocol (VTP) in (Su *et al*, 2004) have proposed and presented preliminary comparison between VTP and TFRC extensions in (Su *et al*, 2005). The newly designed ratio control mechanism in VTP keeps monitoring the end-to-end Achieved Ratio (AR) to achieve smoother ratio adaptation while maintaining TCP friendliness.

Table 2-13: Characteristics of HERC (Xu, 2007)

Characteristics	Result
Year	2007
Problem	TCP and TFRC poor performance in high-speed and long-distance networks, when streaming audio and video.
Approach	An equation based ratio control mechanism for unicast UDP flows
Work in conjunction with other protocol	Yes
Congestion control	Yes
Real-time/non-real-time	Real-time
QoS aware	Not specified
Cross-layer aware	Not specified
Real-world scenario	No

Moreover, a Loss Discrimination Algorithm (LDA) is used to distinguish between congestion and non-congestion loss and minimize the impact of random errors. TFRC performance was tested comprehensively across the public Internet, in the Dummynet emulator (Rizzo *et al*, 1999), and in (NS2) network simulator. The results reported that TFRC outperform TCP across various network conditions. TFRC was not compared across other categories of TLPs besides TCP (traditional). The additional characteristics of the TFRC transport layer protocol are presented in Table 2-12.

2.5.2.4. HERC (Xu, 2007)

An equation-based congestion control protocol (HERC) is the first UDP variants selected for evaluations and comparisons. The critical analysis for HERC is presented in Chapter Three. Additional characteristics of this work are shown in Table 2-13.

2.5.2.5. LATP (Navaratnam *et al*, 2007)

Table 2-14: Characteristics of LATP (Navaratnam *et al*, 2007)

Characteristics	Result
Year	2007
Problem	Medium contention on transport layer
Approach	Rate-based, controlling the ratio at which data packets are send into network
Work in conjunction with other protocol	Yes
Congestion control	Yes
Real-time/non-real-time	Real-time
QoS aware	Yes
Cross-layer aware	Yes
Real-world scenario	No

LATP is second TLP selected from UDP variants category for evaluations and comparisons. LATP is fully reviewed in Chapter Three. Table 2-14 presents the summary of the characteristics of this work.

2.5.3. Hybrid TLPs

As wireless multi-hop networks (WMNs) are used as backbone network for accessing the Internet as well as for community networking, the traffic in WMNs is from multiple applications which use different transport layer protocols (both reliable and unreliable) (Franklin and Murthy, 2008). Applications such as audio and video streaming coexist with TCP traffic; therefore, improving only TCP performance cannot improve overall performance of the WMNs better (Kliazovich *et al*, 2007). Thus, hybrid transport layer protocols proposed. The following are the hybrid transport layer protocols:

2.5.3.1. Implicit Hop-by-hop Congestion Control (Scheuermann *et al*, 2007)

Table 2-15: Characteristics of Implicit Hop-by-hop Congestion Control (Scheuermann *et al*, 2007)

Characteristics	Result
Year	2007
Problem	Congestion due to shared medium nature of WMNs.
Approach	Hop-by-hop in nature and implicit feedback
Work in conjunction with other protocol	Yes
Congestion control	Yes
Real-time/non-real-time	Both
QoS aware	Yes
Cross-layer aware	Not specified
Real-world scenario	Yes

IHCC is the TLP designed to tailor the specific properties of the shared medium over wireless multi-hop networks. Recently, it has been shown that traditional TLPs (TCP and UDP) generally not work well in WMNs as it do in wired networks, due to the ratios of multiple flows do not necessarily converge to a fair sharing of bandwidth owing to shared medium (Raghunathan *et al*, 2005). In the implicit hop-by-hop congestion control, exploits wireless broadcast medium in order to access the necessary information for a backpressure mechanism that reliably limits the number of packets to per flow and hop, and thereby implicitly avoids network congestion.

Implicit hop-by-hop congestion control has a lightweight error detection and correction mechanism, which guarantees a fast reaction to changing medium conditions and low overhead. The implicit comparison to traditional TLPs (TCP and UDP) and other TLP for ad hoc networks ADTCP (Fu *et al*, 2002), good fairness properties and competitive throughput can be observed. In order to examine and compare the performance of implicitly hop-by-hop congestion control, extensive simulation using NS2 network simulator, version 2.29 was done. However, as simulation for wireless multi-hop networks are not able to model all factors that might influence a protocol in real world. Therefore, to compliment simulations, the implicitly hop-by-hop congestion control was also implemented with Request For Acknowledgment (RFA) in a real hardware testbed, and conducted measurements with this implementation.

The implicitly hop-by-hop congestion control exhibits some remarkable advantages over common TLP end-to-end mechanism. In particular these are the ability to deal with UDP as well as with TCP related traffic, very fast reaction times, a low packet delay and a simple protocol designed which greatly eases the adaptation of the protocol new usage scenarios and environments. Additional characteristics of ATCP are shown in Table 3-17.

2.5.3.2. LLE-TCP (Kliazovich *et al*, 2007)

There are two Hybrid TLPs that have been identified in this study with as the LLE-TCP first one. The review of this transport layer protocol is presented in Chapter Three. Table 2-18 shows the summary of the characteristics for LLE-TCP.

2.5.3.3. Link Layer Adaptive Pacing (LLAP) (Franklin and Murthy, 2008)

LLAP is the second Hybrid transport layer protocol that has been identified out of this study and evaluated in order to compare it with other TLPs applicable to WMNs. The review of this TLP can be found in Chapter Three. Table 2-19 shows the summary of the characteristics for LLAP.

Table 2-16: Characteristics of LLE-TCP (Kliazovich *et al*, 2007)

Characteristics	Result
Year	2007
Problem	Medium busy time reduction, link error, network congestion.
Approach	LLE-TCP avoids TCP ACK packet transmission over the wireless channel.
Work in conjunction with other protocol	Yes
Congestion control	Yes
Real-time/non-real-time	Both
QoS aware	Yes
Cross-layer aware	Yes
Real-world scenario	No

Table 2-17: Characteristics of LLAP (Franklin and Murthy, 2008)

Characteristics	Result
Year	2008
Problem	if each edge node pushes data into the network, congestion and MAC contention increase and the overall utilization of the network reduces significantly
Approach	Estimates the four hop transmission delay in the network path without incurring any additional overhead (Control packets) and accordingly paces the packet transmission to reduce MAC contentions in the network.
Work in conjunction with other protocol	Yes
Congestion control	Yes
Real-time/non-real-time	Both
QoS aware	Yes
Cross-layer aware	Yes
Real-world scenario	No

2.5.4. Entirely New TLPs (ENTs)

These TLPs were designed from a fresh start, to avoid the fundamental problems experienced by traditional TLPs in particular type of networks. Entirely New is to tailor the characteristics of WMNs. Existing works have approached the problem of reliable transport in ad-hoc networks by proposing mechanisms to improve TCPs performance over such networks (Akyildiz *et al*, 2005). The following are the ENTs:

2.5.4.1. ATCP (Lui and Singh, 2001)

Ad-hoc TCP (ATCP) proposed because wireless ad-hoc networks are plagued by problems such as high bit errors and frequent route changes. If TCP run over such connection, the throughput of the connection is found to be very poor because TCP treats loss or delay acknowledgements as congestions. ATCP does not impose changes to the standard TCP itself. It implements an intermediate layer between the network and transport layers. In particular, this approach relies on the ICMP (Internet Control Message Protocol) and ECN (Explicit Congestion Notification) scheme to detect network partition and congestion, respectively.

In this manner, the intermediate layer keeps track of the packets to and from the transport layer so that TCPs congestion control is not invoked when not required. When three duplicate ACKs are detected, indicating a loss channel, ATCP sets TCP into persistent mode and quickly retransmits the lost packet from the TCP buffer. After receiving the next ACK the normal state is resumed. In the event than an ICMP destination unreachable message arrives, pointing out a network partition, ATCP also sets the TCP in persistent

Table 2-18: Characteristics of ATCP (Lui and Singh, 2001)

Characteristics	Result
Year	2001
Problem	Network partition and congestion
Approach	ICMP (Internet Control Message Protocol) and ECN (Explicit Congestion Notification)
Work in conjunction with other protocol	Yes
Congestion control	Yes
Real-time/non-real-time	Not specified
QoS aware	Not specified
Cross-layer aware	Yes
Real-world scenario	No

mode which only ends when the connection is re-established. Finally, when the receipt of an ECN message detects network congestion, the ATCP does not do anything, but forwards the packets to TCP so that it can invoke its congestion control. As the ATCP presented the solution to the problem of running TCP in ad hoc wireless networks, therefore, ATCP is an almost ideal for ad-hoc networks solution. The ATCP was implemented in FreeBSD for performance evaluation.

The results from extensive experimentation done in ad hoc networks indicated that ATCP performs better than TCP over wireless ad hoc network. Several TLPs such as TCP-EXACT (TCP variant) and UDP-EXACT (UDP variants) from different categories designed to provides solution to the similar problem but, they are not considered for comparisons. Additional characteristics of ATCP are shown in Table 3-20.

2.5.4.2. ATP (Sundaresan *et al*, 2003)

Table 2-19: Characteristics of ATP (Sundaresan *et al*, 2003)

Characteristics	Result
Year	2003
Problem	TCP's performance degradation in wireless ad hoc network
Approach	Transmissions in ATP are rate-based, and quick start is used for initial ratio estimation. The congestion detection is a delay-based approach.
Work in conjunction with other protocol	Not specified
Congestion control	Yes
Real-time/non-real-time	Non-real time
QoS aware	Not specified
Cross-layer aware	Not specified
Real-world scenario	No

ATP is the first TLP select for evaluations and comparisons from the ENT category. ATP is reviewed in Chapter Three. The characteristics of this TLP are listed in Table 2-21.

2.5.4.3. TPA (Anastasi *et al*, 2005)

The Transport Protocol for Ad-hoc network (TPA) is the TLP developed from scratch specifically for ad-hoc networks. TPAs congestion control mechanism is inspired by TCP, but designed to minimize the number of required packet retransmissions. Packets are transmitted in blocks using a window-based scheme. A fixed number of packets is grouped into a block and transmitted reliably to the destination before any packet of the next block is transmitted. Packet retransmissions are not performed before every packet of a block has been transmitted once, thus a block is transmitted in several rounds.

Table 2-20: Characteristics of TPA (Anastasi *et al*, 2005)

Characteristics	Result
Year	2005
Problem	Link and route failure leading to more packet retransmission.
Approach	Block packet transmission, using window-based approach
Work in conjunction with other protocol	Yes
Congestion control	Yes
Real-time/non-real-time	Real-time
QoS aware	Yes
Cross-layer aware	Not specified
Real-world scenario	No

Every packet must be first transmitted once, then not yet acknowledged packets of this block are retransmitted until every packet of the block has been delivered and acknowledged If an ELFN mechanism is available, TPA can make use of it and enters a freeze state upon route failures, decreasing the window size to one. If ELFN is not available, TPA detects route failures by a number of consecutive timeouts. Similar to TCP, TPA uses an estimation of the RTT to set the retransmission timeout.

In the event of route changes, new RTT values are given a greater weight in the sliding average in order to speed up the RTT estimation method to achieve quickly a reliable estimate for new RTT. For congestion control TPA uses a window mechanism with a tightly limited maximum window size. Actually, only two different cwnd values are used: a “large” window of 2 or 3 segments during normal operation and the minimum value of 1 when congestion is detected. TPA shows that even a quite simple end-to-end protocol without additional intelligence in the intermediate nodes has the potential to increase throughput in comparison to TCP. TPA works interchangeably with other TLPs, however, it is not yet clear if these benefits can be maintained in more complex, dynamic scenarios. The performance of the TPA and TCP was compared in terms of both throughput achieved by destination node at the application layer and the percentage of retransmission were done through NS2 network simulator tool. Among existing TLP categories only TCP (traditional) considered for comparisons. Some additional characteristics of this work are shown in Table 2-22.

2.5.4.4. WXCP (Su and Zurich, 2005)

Wireless eXplicit Congestion control (WXCP) is the second selected TLP from the Entirely New category. The review of the WXCP can be found in Chapter four. The Table 2-23 presents the characteristics of WXCP TLP.

2.5.4.5. AR-TP (Pace *et al*, 2007)

Adaptive and Responsive Transport Protocol (AR-TP) proposed for wireless mesh networks in order to fairly allocate the network resources among multiple flows, while minimizing the performance overhead. AR-TP is an adaptive transport layer protocol based on hop-by-hop congestion control and coarse-grained end-to-end reliability mechanisms, which are designed to achieve high throughput performance and reliable data transmission in Wireless Mesh Networks.

Table 2-23: Characteristics of WXCP (Su and Zurich, 2005)

Characteristics	Result
Year	2005
Problem	TCP variants problem of being unfair and inefficient in WMNs
Approach	It uses an eXplicit Congestion control Protocol (XCP) to control network congestion.
Work in conjunction with other protocol	Yes
Congestion control	Yes
Real-time/non-real-time	Real-time
QoS aware	Yes
Cross-layer aware	Not specified
Real-world scenario	No

Table 2-24: Characteristics of AR-TP (Pace *et al*, 2007)

Characteristics	Result
Year	2007
Problem	Limited link capacity for Wireless Mesh Networks
Approach	AR-TP is based on hop-by-hop congestion control and coarse grained end-to-end reliability mechanisms
Work in conjunction with other protocol	Yes
Congestion control	Yes
Real-time/non-real-time	Real-time
QoS aware	Not specified
Cross-layer aware	Not specified
Real-world scenario	No

Furthermore, compared to end-to-end ratio control schemes, the hop-by-hop ratio adaptation strategy of AR-TP enables each router to keep track of dynamic wireless channel conditions. Using a hop-by-hop strategy, each mesh router can adapt its data transmission ratio opportunistically in the case of multi-channel WMNs. Performance evaluation via extensive simulation (NS2 network simulator version 2.29) experiments

fairness when compared to several other TLPs, namely TCP New Reno, TCP-Sack representing the most widely used TLP family over Internet, TCP-FEW and TCP-ELFN which are TLPs specially proposed for wireless ad hoc networks. The characteristics summary of the AR-TP is found in Table 2-24.show that the AR-TP protocol achieves high performance in terms of throughput and TCP as the most used protocol for Internet has been researched in various networks.

2.5.4.6. CLM-TCP (Nascimento *et al*, 2008)

Table 2-25: Characteristics of CLM-TCP (Nascimento *et al*, 2008)

Characteristics	Result
Year	2008
Problem	Packet loss due to channel error, medium contention and route failures.
Approach	Cross-layer aware congestion control mechanisms
Work in conjunction with other protocol	Yes
Congestion control	Yes
Real-time/non-real-time	Non-real-time
QoS aware	Not specified
Cross-layer aware	Yes
Real-world scenario	No

This TLP has a problem in WMNs due to its origin and implementation for wired networks. Therefore, CLM-TCP proposed as an adaptation TLP on congestion control mechanisms using information from the network layer to improve TCP through the vertical calibration across layer technique. CLM-TCP adjusts congestion window (cwnd) and slow-start threshold (ssthresh) values in the occurrence of a timeout in less aggressive manner. This protocol uses information received from the network layer in order to improve network throughput.

The collected information by CLM-TCP used to check the link quality and delay, to verify if there are necessary modification in the variables cwnd and ssthresh. The verifications are done by CLM-TCP only takes place after timeout event that need cwnd and ssthresh to be changed. To evaluate the performance of the CLM-TCP, NS2 was used. The CLM-TCP was compared to several TCP variants in Wireless Mesh Networks based on congestion overhead, throughput and congestion window input behaviour. Additional characteristics of CLM-TCP are found in Table 2-25.

2.5.4.7. WCCP (Mahendra and Sethnil, 2010)

Wireless Congestion Control Protocol (WCCP) was designed because traditional TLP (TCP) encounters several problems and lead to poor performance if the IEEE 802.11 MAC protocol is used in wireless multi-hop networks. The huge number of problems experienced by TCP in multi-hop Ad hoc is due to medium contention at MAC layer. WCCP is based on channel busyness ratio. In this TLP, each forwarding node along a traffic flow exercises determines the inter-node and intra-node for fair channel resource allocation and allocating the resource to the passing flows by monitoring and possibly overwriting the feedback field of the data packets according to its measured channel busy ratio.

The feedback is then carried back to the source by the destination, while copies it from data packet to its corresponding acknowledgement. The source finalizes by adjusting the sending ratio accordingly. The WCCP performance was compared with TCP through extensive simulations using NS2 the comparisons were done based on the following

Table 2-26: Characteristics of WCCP (Mahendra and Sethnil, 2010)

Characteristics	Result
Year	2010
Problem	Packet loss due to channel error, medium contention and route failures.
Approach	Ratio based congestion control
Work in conjunction with other protocol	Yes
Congestion control	Yes
Real-time/non-real-time	Non-real-time
QoS aware	Not specified
Cross-layer aware	Not specified
Real-world scenario	No

performance metrics: channel utilization, delay, and fairness. The advantage of WCCP, it reduces starvation problem suffered by TCP. A summary of WCCP is found in Table 2-26. In this section we afforded to review the transport layer protocols applicable to wireless multi-hop networks (WMNs) from each of the four categories and tables are provided for additional characteristics of the TLPs. In the following section, the list of the selected TLPs for evaluation id provided.

2.6. Selection of the TLPs for Performance Evaluations in WMNs

The analyses of the existing TLPs applicable WMNs using framework described in Section 2.5 eases the selection of the TLP for performance evaluation. TLPs are categorized into four types and two TLPs from each type were selected. The traditional transport protocols from the baseline of our comparisons. In Table 2-27, the selected TLPs for evaluations are bolded and highlighted. The list of the selected TLPs has been provided also in Table 2-28.

During the selection of these transports layer protocol numbers of features were considered especially the ones in the framework Table 2-1 and Table 2-27. Where TLPs are found to be dealing with common problem, the most recent protocols were considered for evaluation. All TCP variants in Table 2-27 deal mostly with network congestions but

Table 2-27: Transport Layer Protocols

Group	Protocol	Real-Time Transfer	Congestion Control	Cross-layer Aware	Testbed	QoS Aware
TCP Variant	I-TCP (Bakre *et al*, 1995)	×	√	×	√	×
	M-TCP(Brown and Singh, 1997)	×	√	×	×	×
	WTCP (Ratnam & Matta, 1998)	×	√	×	×	√
	TCP-Feedback(Chandran,2001**)**	×	√	×	×	×
	SNOOP (Ch Ng *et al*, 2002)	×	√	×	×	×
	ST-PD TCP (Xu *et al*, 2002)	×	√	×	×	×
	TCP Door (Wang *et al*, 2002)	×	√	×	×	×
	EXACT(Chen *et al*, 2004)	×	√	×	×	×
	TCP-AP(ERakabawy *et al*, 2005)	×	√	×	×	√
UDP Variant	RAP (Rejaie *et al*, 1999)	√	√	×	×	×
	TFRC (Handley *et al*, 2003)	√	√	×	×	√
	ARC (Kan and Akyildiz, 2004)	√	√	×	×	×
	DCCP (Kohler, 05 ,06)	√	√	×	×	√
	LATP(Navaratnam *et al*, 2007)	√	√	√	×	√
	HERC(Xu, 2007)	√	√	×	×	×
Hybrid	LLE-TCP(Kliazovich *et al*, 2007)	√	√	√	√	√
	LLAP(Franklin & Murthy, 2008)	√	√	√	×	√
Entirely New	WXCP (Su *et al*, 2005)	×	√	×	×	√
	ATP (Sundaresan *et al*, 2003)	×	√	×	×	√
	AR-TP (Pace, 2007)	×	√	√	×	√
	ATP(Lui and Singh, 1999)	×	√	×	×	×

using different approaches because some TLPs are the extension of other TLPs, therefore, the recent TLP (has improved performance) was considered. TLPs already implemented in NS2 were considered as the ones with the highest priority in our selection. On the other hand, if the TLPs of the same category have provided a common feature e.g. solving the common problem, using similar approach, we select one of them for performance evaluations. If the TLP is an extension of another TLP which is already implemented on NS2, therefore, our task was to modify the previous TLP to suit the new

TLP's behavior. Modifying the TLP existing already in NS2 was also given the highest

Table 2-28: Selected TLPs

GROUP	Representative TLPs
TCP Variants	Snoop [Ch Ng *et al*, 2002]
	TCP-AP [ERakabawy *et al*, 2005]
UDP Variants	HERC [Xu, 2007]
	LATP [Navaratnam *et al*, 2007]
Hybrid Representative	LLE-TCP [Kliazovich *et al*, 2007]
	LLAP [Franklin and Murthy, 2008]
Entirely New	WXCP [Su and Zurich, 2005]
	ATP [Sundaresan *et al*, 2003]
Traditional	TCP
	UDP

priority because using something that has been tried and tested is better that starting something from scratch. The time needed for modifying the TLP already existing in NS2 is much less than the time needed for implementing the protocol from the scratch. For TLP which is not implemented in NS2 and not extending other TLP that is already implemented, the flowcharts and pseudo code developed in Chapter Three were followed to implement them in NS2.

2.7. Summary

The analysis of the four categories of the TLPs, the research trends and the framework followed to review and analyse the TLPs are given in this chapter. This chapter has also presented the literature of the TLPs applicable to WMNs reviewed using formulated literature analysis framework. The framework in this chapter assisted in speeding-up the process of identifying the eight TLPs for evaluation and comparison. Thus, the list of the selected TLPs for evaluation is provided in the same chapter. The following chapter (Chapter Three) gives the descriptions, flowcharts and pseudo codes for the selected

TLPs applicable to WMNs.

CHAPTER THREE

REVIEW OF THE SELECTED PROTOCOLS FOR THIS STUDY

3.1. Introduction

After the review of the literature in Chapter Two using the framework given in Table 2-1, four classes or categories of TLPs applicable WMNs were identified. Two TLPs were selected from each category for simulation i.e. eight TLPs chosen. We wanted to simulate many TLPs per category, but due to time constraint we opted to consider only two. The selected TLPs were simulated in NS2 in order to evaluate and identify an optimal performing TLP in various WMNs conditions in Chapter Four. In this Chapter we presented the pseudo code and flowcharts for the selected TLPs.

In order to verify the correctness of pseudo code and flowcharts as well as requesting their source code, we emailed them to original authors. However, out of eight authors, only one author responded i.e. Kliazovich *et al* (2007) for LLE-TCP. He verified and approved our pseudo code and flowchart, but he did not offer the source code. The two traditional transport layer protocols (i.e. TCP and UDP) are taken as default protocol for our performance evaluations, since they have been tried and tested.

This chapter gives the descriptions, pseudo codes and flowcharts for the selected TLPs for performance evaluations. Pseudo-code and flowcharts were to implement TLPs in NS2. The pseudo-code and the flowchart for the traditional transport layer protocols are not provided since traditional TLPs are most common TLPs.

3.2. Selected Transport Layer Protocols applicable to WMNs

In this section, we describe the eight selected TLPs for evaluations and presented the pseudocode and flowcharts for these TLPs.

3.2.1. TCP variants

The descriptions for two TCP variants are given in this section. Figure 3-1 presents the flowchart for Snoop (TCP variant) and Figure 3-2 gives the pseudocode for Snoop TLP. The flowchart and pseudocode for TCP-AP (TCP variant) are presented in Figure 3-3 and Figure3-4, respectively.

3.2.1.1. Snoop

The major goal of Snoop is to improve the TCP performance of communication over wireless links without triggering the retransmission and window reduction polices at the transport layer. The protocol is implemented using an *agent* that *monitors packet*s that pass through *TCP connections* in both *directions* and *caches* the packets sent across the link that has not yet been "*acknowledged"* by the destination. After *caching* the agent *forwards packets* to their destination and monitors the corresponding *acknowledgements (ACKs).*

The data *packets loss* is indicated by the reception of *duplicate ACKs* from the destination or by a local timeout. Snoop *agent retransmits* the lost packets when it has *cached* the *duplicate* and *suppresses* the *duplicate acknowledgments*. It retransmits them locally *without forwarding* the ACKs to the source. Hence, since the TCP layer is not aware of the *packet loss*, and the *congestion control algorithm* is not triggered. In addition, the Snoop *agent starts* a retransmission timer for each TCP connection.

When the *retransmission timer expires*, the agent retransmits the packets that have not been *acknowledged yet*. This timer is called persist timer because, unlike TCP retransmission timer, it has a fixed value. The Snoop protocol *intercepts TCP packets*, analyzes them, and retransmits them if necessary. As a result, no additional packet formats are introduced into the protocol, and all packets sent and received still conform to the TCP protocol.

The major advantage of the Snoop protocol is that it suppresses duplicate ACKs for TCP packet lost and retransmitted locally, therefore, avoids needless fast retransmission and congestion control invocation by the source. Snoop is unable to completely protect the source from wireless losses. To study the effect of the Snoop on the performance of TCP over wireless links, Snoop has been implemented and simulated in OPNET wireless devices.

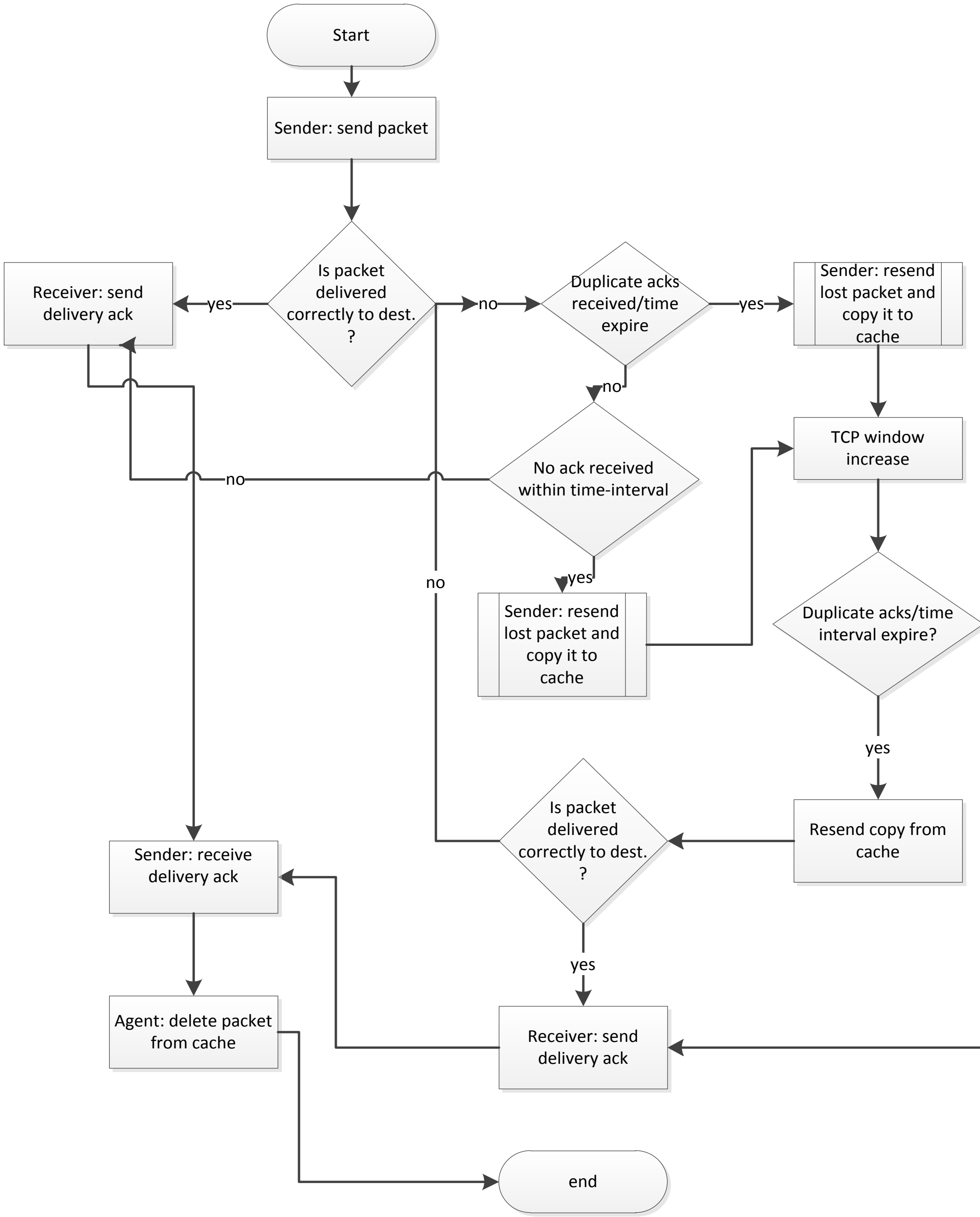

Figure 3-1: Snoop flowchart

The performance improvement measured by comparing the performance of Snoop and TCP. However, the Snoop mechanisms used to detect differentiated between congestion and non-congestion losses and control congestions are not compared to other mechanism constant manner. These mechanisms are only compared with TCP over wireless, but

```
Start {
    sender: start sending () //after send request has been accepted
         agents (monitor packet to dist.) //manages packet to destination
            receiver: receive packet (send acks) //delivery acknowledgement
               if (packet reach correct dest.)
                 {
                 delivery acks received;
                 }
               else {
                 duplicate acks received (lost packet) //duplicate indicate lost
       retransmit lost packet () // sender
      increase window () // tcp mechanism
      }
       if (sender receives no acks) //within the expected/estimated period
            {
               reduce window size by half () //reduced by TCP
             copy packet to cache () //agent
           retransmission timer start;
     }
            if (duplicate ack received or timer expires)
                 {
      retransmit packet () // packet retransmitted by agent
      delivery acks;
    delete cache; //agent, after delivery acks
 }
 End
}
```

Figure 3-2: Pseudocode for Snoop

using different values and parameters. The step-by-step processing of the Snoop is given in Figure 3-2 and the flow of the events of the protocol is reported in Figure 3-1. The flowchart and pseudo code simplify the implementation of Snoop in NS2 for our performance evaluations. Table 2-8 presents the summary of the characteristics of Snoop TLP.

3.2.1.2. TCP with Adaptive Pacing (TCP-AP)

TCP-AP is a hybrid of *window-based* and a *rate-based* approach, adding rate-based mechanisms to TCP in order to avoid the large bursts of packet problem. TCP-AP *spreads* the transmission of successive data packets according to the *computed transmission ratio*, this accounts for the spatial reuse constraint in IEEE 802.11 multi-hop wireless networks. Furthermore, by *proactively* identifying *incipient* c*ongestion*, i.e. before congestion-related losses actually occur, TCP-AP *adjusts transmission* ratio and, hence, reduce contention on the MAC layer.

TCP-AP allows *routers* to provide *explicit congestion information* in the IP header of each data packet, and allows *intermediate routers* to *modify* the data *sending ratio*. The ratio information returns from the destination to the source as feedback on the network path of the TCP connection and proactively throttles the transmission ratio before losses occur. In order to retain the end-to-end semantics of TCP, TCP-AP uses a measure obtainable at the TCP entities, which quantifies the degree of contention on the network path.

The Coefficient of Variation of recently measured round trip time is key measure for the degree of the contention on the network path. The TCP-AP designed to improve TCP performance over WMNs and it uses measure obtainable at the TCP entities to detects or quantify the degree of contention on the network. These mechanisms yield better improvement in the TCP performance over wireless network but we cannot tell which one performs better since they are not compared in a consistent manner.

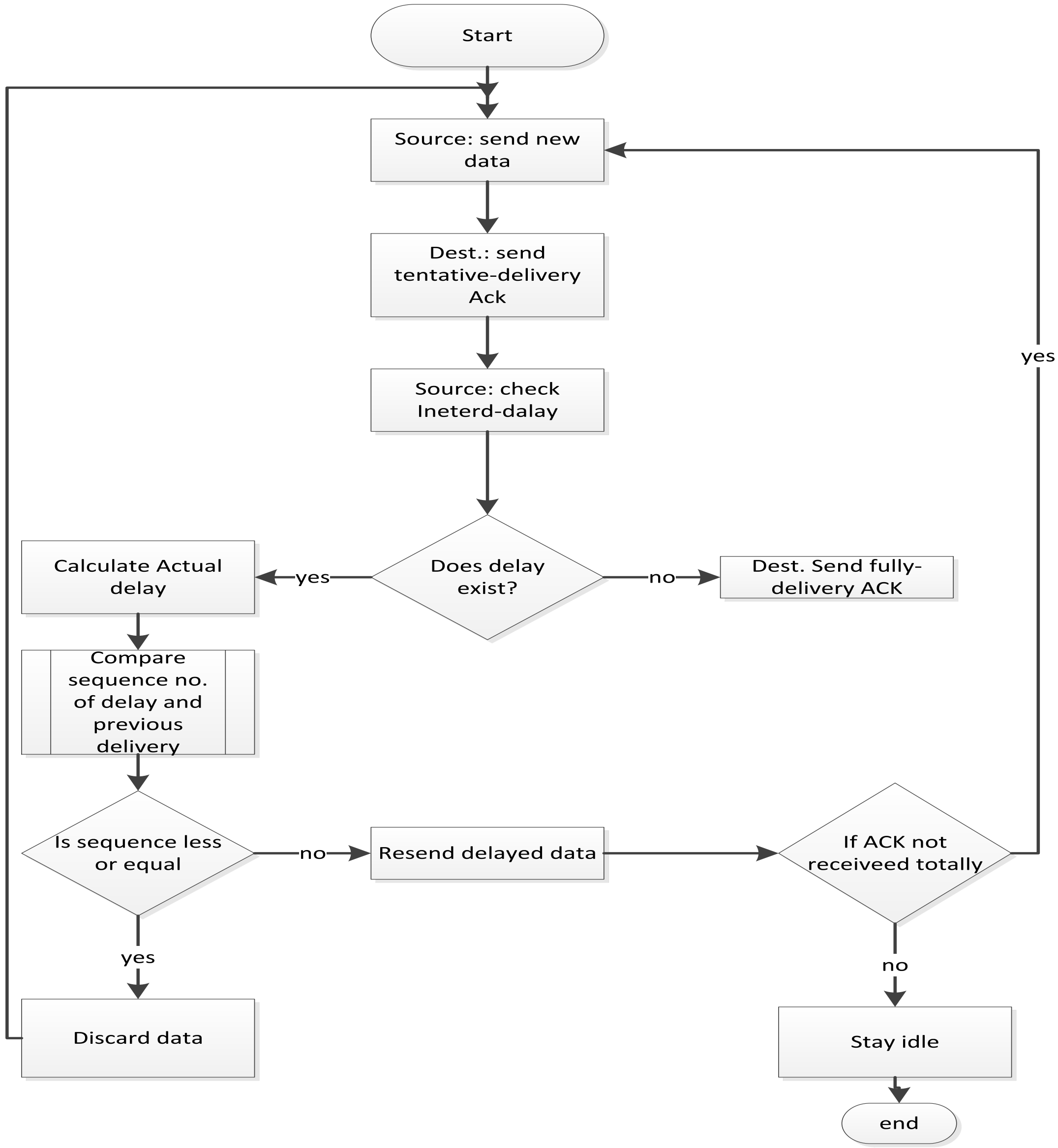

Figure 3-3: TCP-AP Flowchart

The complete simulation study utilizing NS2 depicts that TCP-AP attains more throughput than TCP New Reno, provides excellent fairness in different conditions and is greatly responsive to varying traffic scenarios. Additional characteristics of this work are shown in Table 2-12. The flow of the events (flowchart) in the TCP-AP is presented in Figure 3-3 whereas Figure 3-4 gives the pseudocode for TCP-AP.

```
Start {
     sender: start sending data () // after request to send approved
      receiver: send tentative-delivery Ack () // while checking IP's
           render: check InterpacketDelay() // after receiving Ack
                  if (delay doesn't exist) //delivered in estimated time
                       {
                  receiver: send fully-delivery Ack ()
                 }
              else{
                   compare (actual delay and expected delay) // check if there was delay
                    }
            if (delay doesn't exceed expected)
                 {
                    receiver: send fully-delivery Ack ()
                  }
          else
                 {
               calculate actual delay()
       compare sequence (delayed-data  & previous-delivered data)
      }
       if (sequence is less or equal)
     {
     discard data and send new data () //destroy data from intruders
  }
else
    {
      resend delayed data ()
 }
if(delivery Ack not received) // lost due to unknown reason
     {
    resend data ()
 }
else
 {
    Stay idle () //no data need to be send
}
End
}
```

Figure 3-4: Pseudocode for TCP-AP

3.2.2. UDP Variants

Two variants of UDP namely LATP and HERC are described in the subsections below. The flowchart and the pseudocode for LATP are shown in Figure3-5 and Figure 3-6, respectively. In the subsection below, Figure 3-7 and Figure 3-8 present the flowchart and algorithm.

3.2.2.1. Link Adaptive Transport Protocol (LATP)

Link Adaptive Transport Protocol (LATP), was proposed to fix the challenges such as congestion, which is mainly due to medium contention in multi-hop wireless networks, challenges the performance of the traditional TLP in such networks. LATP support quality of service requirements in multi-hop wireless networks. It provides symmetric way of controlling end-to-end ratio for multimedia streaming applications, based on the *degree* of *medium contention* information received from the network.

LATP sources transmit a stream of data packets to the destination and *control the sending ratio* based on the *feedback information* received from the *destination*. Three components responsible for monitoring the data packets transfer from source to target destination namely: *Intermediate node* which calculates the *permissible throughput* “P” for each and every outgoing data packet based on channel *busyness ratio* and *throughput estimation*, to efficiently utilize channel, while avoiding *severe medium contention*.

It then updates the ratio-feedback, “*R*” in the header to the value of *P*, if *P* is smaller than *R*, LATP Destination sends feedback packets at *regular reporting periods* in order to

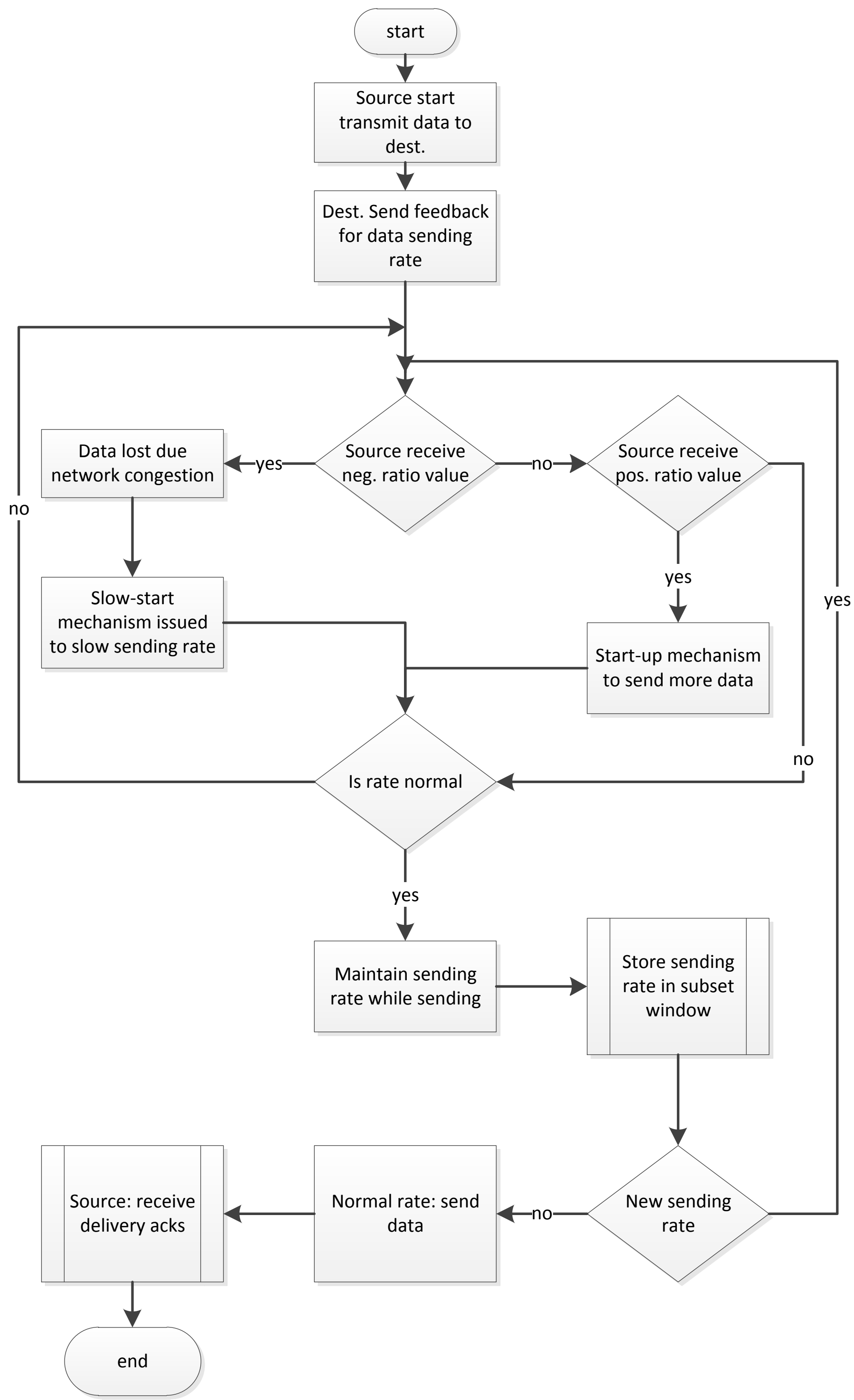

Figure 3-5: LATP Flowchart

```
Start () {
  source-LATP: transmit a number of data; // after request to send approved
       dest-LATP: return feedback;
           control sending rate based on feedback; // Source-LATP
                update every rrt; // Source-LATP
                   if (source receives negative ratio value)
                     {
                    network congested; // indicating possible data lost
              then: slow-start mechanism is triggered; // to reduce sending rate
              reduce transmission rate to normal; //Slow-start mechanism
                }
           else-if (source receives positive ratio value)
           {
                    network is not congested; //indicating possible channel under used
           then: start-up mechanism is triggered; //to increase sending ratio
                 increase transmission rate to normal; // start-up mechanism
           LATP: maintain normal transmission rate; //once the normal state reached
  store normal value in subset window;
}
    if (new rate experienced)
    {
   compare new rate and rate in subset window;
    }
   if (new rate suggest data lost) // due to congestion
{
      calculate new sending rate;
}
else
{
       send data; // use new rate
}
End ()
}
```

Figure 3-6: Pseudocode for LATP

assist the source to determine the sending ratio according to network conditions. The destination copies *R* and measures the average *ratio-feedback*, and LATP Source, on connection initiation the source transmits data utilizing a small opening *transmitting rate* until it experiences the first ACK from the destination. When the first feedback packet is *achieved*, it takes a *slow start mechanism* to probe the *network capacity*. Although LATP is QoS aware and cross-layer aware but, it not yet implemented in real life.

The performance evaluation of LATP over a variety of network scenarios was done using NS2 simulations. LATP is compared with TFRC (UDP variants) and TCP for end-to-end delay, jitter, loss ratio, and throughput and fairness performance over chain, grid and random topologies. The Flowchart indicating the flowing of the events in LATP is shown in Figure 3-5 and the step-by-step functioning of LATP is presented in Figure 3-6. A summary of the characteristics of this work is presented in Table 2-18.

3.2.2.2. High-speed Equation-based Ratio Control (HERC)

An equation-based congestion scheme such as TFRC has been a promising substitute to TCP for real time traffics. However, it also utilizes a similar TCP response technique and it also has disgraced the functioning as TCP applicable to high-speed and long-distance wireless networks (Floyd and Kohler, 2005). Thus, Xu (2006) proposed High-speed Equation-based Ratio Control (HERC), extending the TFRC by replacing the TCP by substituting the TCP response technique buy a high-speed response technique. HERC supports applications, such as high-definition video streaming, and distant collaboration involving high-resolution visualization, which prefers a *high-speed* and relatively smooth

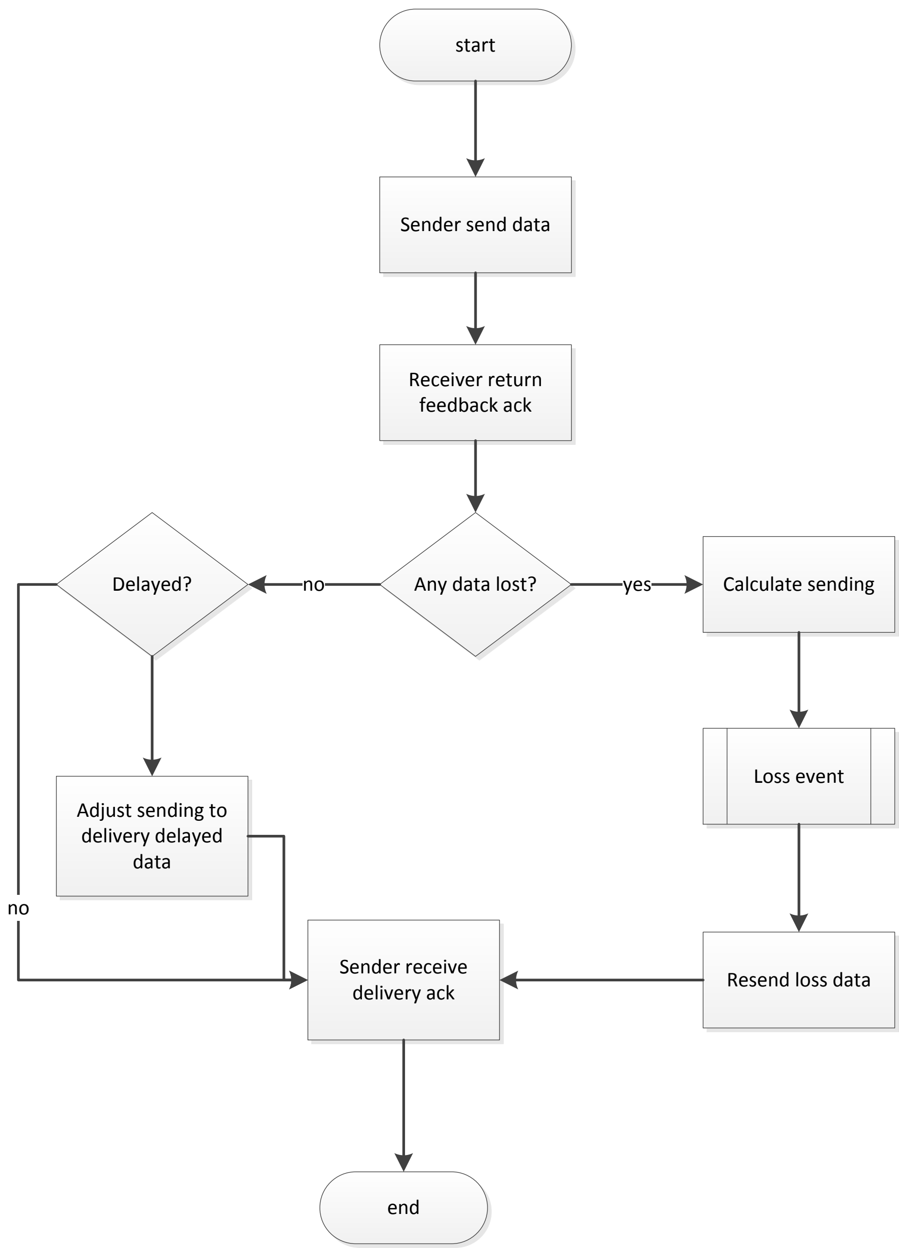

Figure 3-7: HERC Flowchart

sending ratio. The response function of a *high-speed* TCP variant determines *several important properties* of the protocol, such as bandwidth scalability. The bandwidth scalability of a protocol, defined as the ability of the protocol to achieve high throughput in a high-speed network, is typically determined by its sending ratios under low loss.

```
Start ()
        {
    sender: start sending data; // after request to send approved
       return ack feedback; //receiver
          verify packet loss event from ack feedback; //Sender:
if (packet lost is less or equal to 0.54)  // lost=0.54 packet lost occurred
                {
              packet loss event occurred ();
             sending rate: calculated; // to get normal sending rate
               sending rate: gives number of loss event;
                     Loss event: gives number of packet to resend;
                 sender: resend lost packet;
                         sending rate adjusted; // to normal sending rate
                       }
              else-if (packet lost is greater or equal to 0.54)
                  {
              packets not reach dest. during expected time; // due to delay
          adjust sending rate;   //Sender reduce equation
         data delivery;   // delivery ack send by receiver to sender
  }
End
}
```

Figure 3-8: Pseudocode for HERC

event rates. The TCP friendliness defines whether a protocol is being fair to TCP, and it is critical to the safety of deploying a protocol in the Internet. The impact of a general high-speed response function on throughput and smoothness of HERC was verified by using NS2. The result indicated that using a high-speed TCP variant and fine-tuning. Figure 3-7 presents flowchart for HERC and Figure 3-8 depicts the pseudo code for step-by-step processing of HERC. An additional properties summary for HERC is listed in Table 2-15.

3.2.3. Hybrid TLPs

The two Hybrid TLPs selected for evaluations are LLE-TCP and LLAP and their analysis are given in this section. Subsection 3.2.3.1 gives the studies of the LLE-TCP. The studies of the LLAP are presented in subsection 3.2.3.2.

3.2.3.1. Link Layer Exploitation TCP (LLE-TCP)

Link layer ARQ (Automatic Repeat Request) Exploitation TCP (LLE-TCP), is cross-layering approach, where the main performance advantages are achieved through the optimization of the interlayer ARQ scheme functionality. LLE-TCP does not change or override any TCP flow control mechanisms. However, *suppression* of TCP ACK transmission over the *wireless channel* and corresponding impact on the *delay component reduces* the *round trip time* (RTT) of the connection. The logical association of the ARQ to the link layer brings scalability to LLE-TCP.

The main functionality of AQR is to suppress TCP ACKs. LLE-TCP provides dynamic means to reduce error rate available on wireless present on wireless connected via delivery delay, using a "*stop & wait*". ARQ mechanism where the transmitter is unallowed to transmit the following packet in the queue until the destination confidently indicates the positive receiver of the packet that was sent in first place. Thus, LLE-TCP ensures that the *spoiled packets* are retransmitted, and introducing a degree of *overhead*

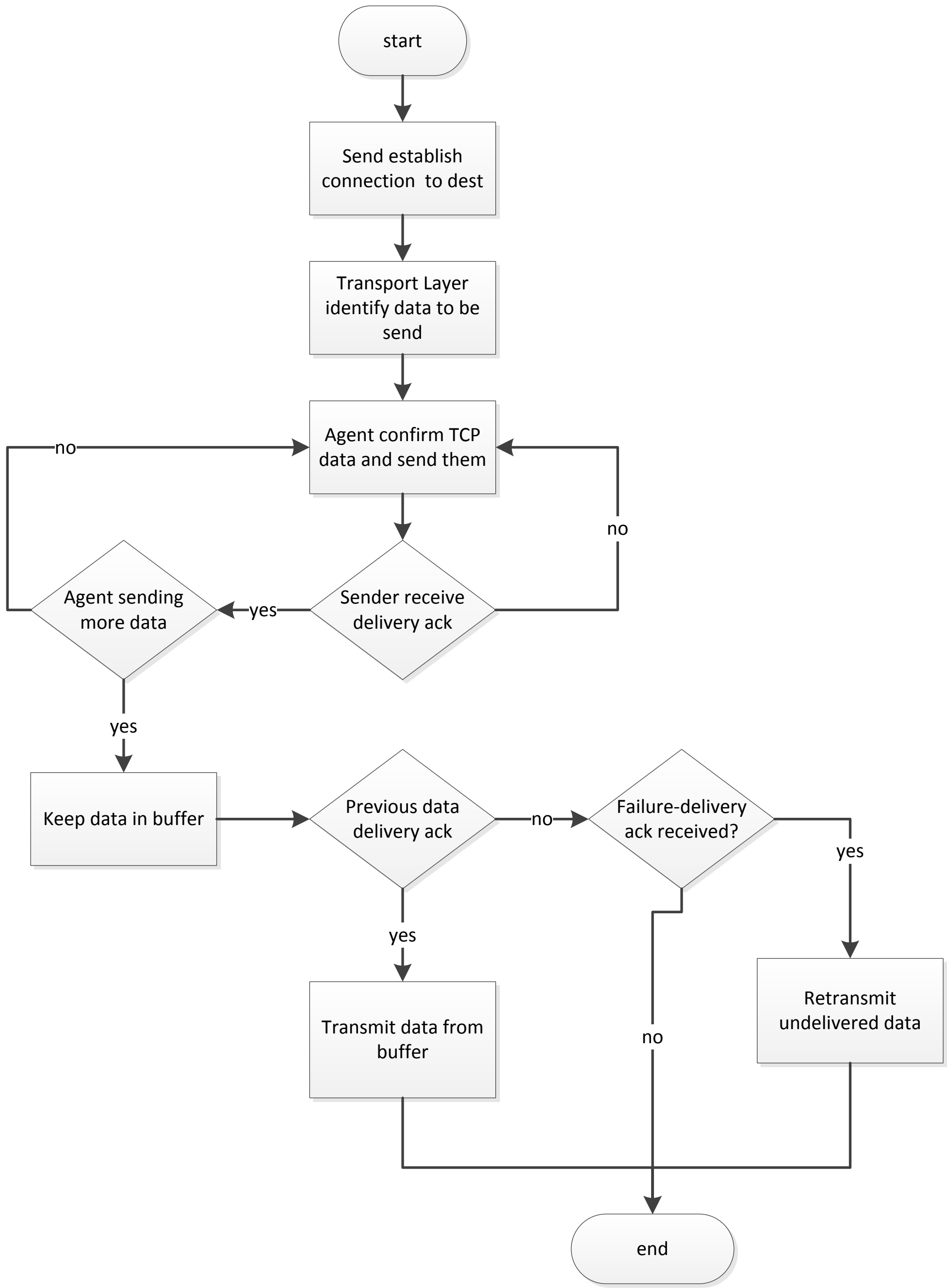

Figure 3-9: LLE-TCP Flowchart

adjusted to conform to the state of the connection. LLE-TCP solve the congestion problem by its possibility of accessing the *receiver advertise window* (rwnd), i.e. to *misuse* the *TCP header field* in all ACK for identifying how much *unused buffer space* on the network.Among the factors contributing to LLE-TCP performance enhancement is:

```
Start ()
{
 sender: send packet to dest.; // after request to send approved
   detect new packet to be sent;  //by Transport Layer
ARQAgent: confirm if ( TCP packet)  //to make sure that its TCP related packet
                 {
                transmit TCP packet to dest.; //using agent:
  recieve delivery ack;  //Sender:
              }
     else-If (agent busy sending other TCP packet)
        {
         agent: store new packet buffer; //waiting for resources to send
         }
 agent; wait for successful-delivery acks (previous data);
       }
agent: retransmit packet from buffer;

if (failure-delivery acks received)
     {
       retransmit undelivered data;
     }
agent: only transmit data from buffer (when success-delivery ack received)
          {
  End
}
```

Figure 3-10: Pseudocode for LLE-TCP

medium busy time reduction, minimized sensibility to *connection errors*, *RTT*, and *advanced congestion mechanism*. The performance enhancement of this TLP brought by cross-layer optimization of ARQ mechanisms utilized in the various layers of the OSI model. Performance evaluation of LLE-TCP is performed via NS2 network simulator and verified using experiments on the IEEE 802.11b testbed. Apart from the traditional TLPs (UDP and TCP), the TCP Reno was selected for comparisons as the common reference implemented of the TCP TLP, presently driving systems. All traditional TLPs and one

TLP from TCP variants considered for comparisons. A step-by-step functioning of the LLE-TCP is presented in Figure 3-9 whereas the LLE-TCP flowchart is depicted in Figure 3-10. Additional characteristics of this transport layer protocol are presented in Table 3-19.

3.2.3.2. Link Layer Adaptive Pacing (LLAP)

LLAP is the scheme that *adaptively* controls the offered *load* into the network. LLAP *reduces* the *contention* in the network by properly *scheduling* the packets at edge nodes thereby *increasing* the channel *spatial reuse* in the network, a cross-layer approach used for scheduling of packets and estimation of Four-Hop transmission Delay (FHD) in a path. LLAP *estimates* the FHD in a path by measuring the *queuing* and *transmission* delay incurred at the bottleneck node in a distributed manner. The LLAP improves the *performance* of higher layer protocols without any modifications to them.

LLAP scheme estimates the FHD in the network path without *incurring* any additional overhead (Control packets) and accordingly *paces* the packet transmissions to *reduce MAC contentions* in the network. As the performance of TCP and UDP is greatly affected by the packet losses in the network, thus LLAP provides great improvement by reducing the losses to improve WMNs performance considerably. The nodes in the network channel *measure* the degree *of congestion at the bottleneck* node in a distributed manner. The main contributions of LLAP paper are as follows:

i. In order to reduce the contention in the network for achieving better spatial channel reuse, a scheme for pacing of packets (based on their destination) at the Link layer

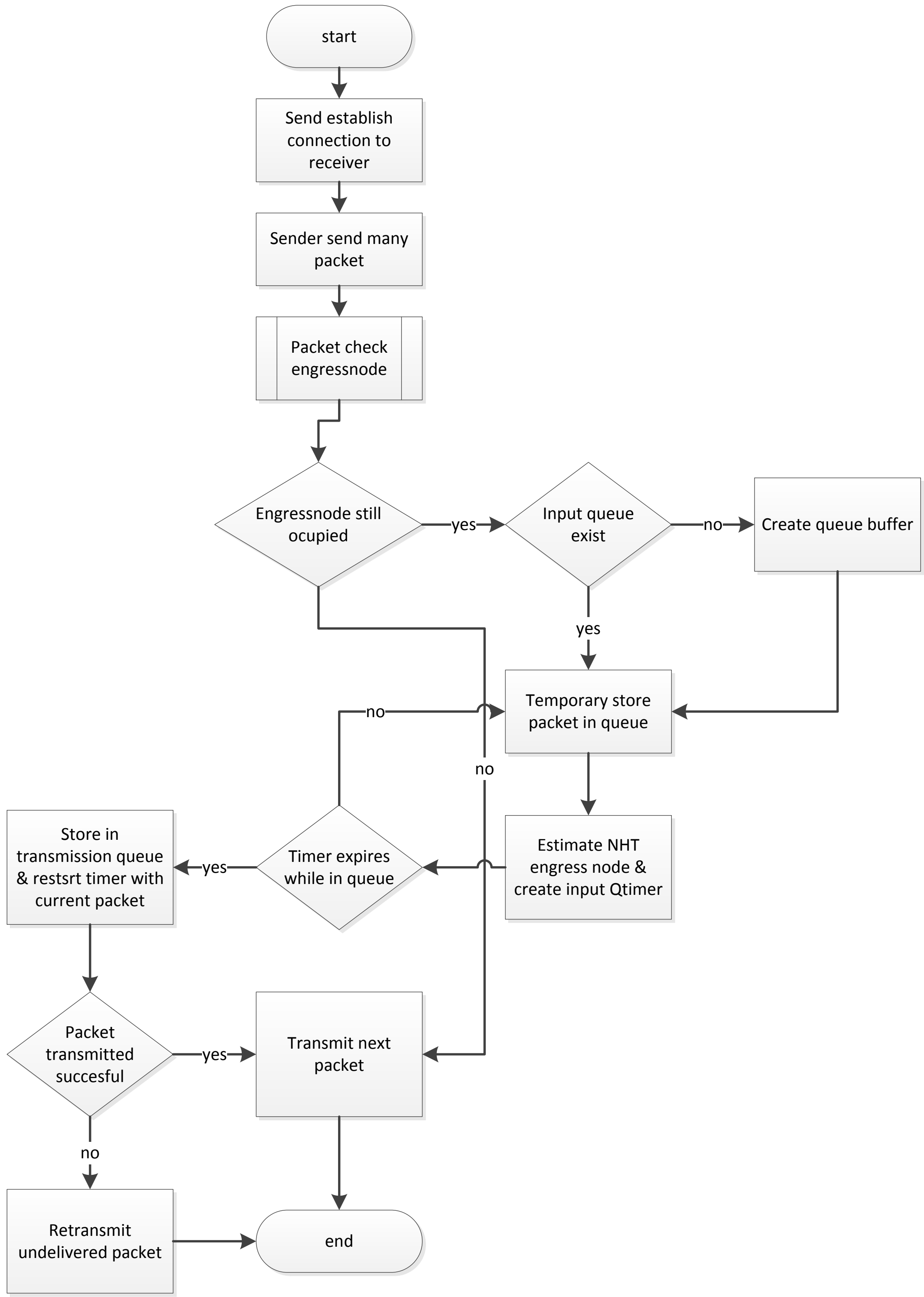

Figure 3-11: LLAP Flowchart

```
Start ()
        {
  sender: send many packet concurrent; // after request to send approved
         packet: check engress node; // responsible for routing data
    if (engress node still occupied)
      {
then: check if (input queue exist) // to keep packet until d engross is available
                          {
                        store packet temporary (input queue);
             }
        estimate NHT engress node; //help in creating IQT
  create input queue timer (IQT); //to tell how long to keep packet in queue
         if (packet timer expires while in queue)
                   {
              move packet to transmission queue (TQ);
         then: restart timer with current packet;
     if (packet transmitted successful)
        {
          transmit next packet;
     }
  else {
        retransmit undelivered packet;
}
End
}
```

Figure 3-12: Pseudocode for LLAP

was proposed. ii. For the estimation of pacing delay, no any additional control packet exchange between the nodes was needed. The LLAP mechanism is implemented in NS-2.29 and comprehensively investigated its ability to for both UDP and TCP related applications in different network scenarios. The LLAP performance indicated better improvement when compared to both traditional TLPs (UDP and TCP) as well as TCP New Reno for throughput and fairness. Table 2-19 in Chapter Two shows the summary of the characteristics for LLAP. Figure 3-12 presents the pseudo code for LLAP while flow of events (flowchart) of LLAP is shown in Figure 3-11.

3.2.4. Entirely New TLPs (ENTs)

The full analyses and the descriptions of the ENTs (WXCP and ATP) are given in this section. The analysis of WXCP and ATP are presented in subsection 3.2.4.1 and Subsection 3.2.4.2, respectively.

3.2.4.1. Wireless eXplicit Congestion Control Protocol (WXCP)

The Wireless eXplicit Congestion control protocol (WXCP) was designed to eliminate the problems experienced with TCP variants such as being unfair and inefficient in wireless multi-hop networks environment (Sundaresan *et al*, 2003). WXCP uses *explicit ratio feedback* instead of *probing the available bandwidth*. It contains *Congestion Metrics* to measure *resource usage* and *level of conges*tion in the network. The following metrics are used, i.e. i. Available bandwidth (ABW), ii. Interface Queue (IFQ), iii. Length and Average link layer retransmission (LAR).

ABW is used to indicate the present network ability in order to determine and if the *incoming data rate* is *more than the outgoing data rate*, data starts to be overloaded and input traffic is stopped from *entering the network*, whether there is high probability of *congestion occurring* or *not*. *IFQ* controls the *input traffic*. The third congestion metric detects the *degree of self-interference*. WXCP is a QoS aware transport layer protocol

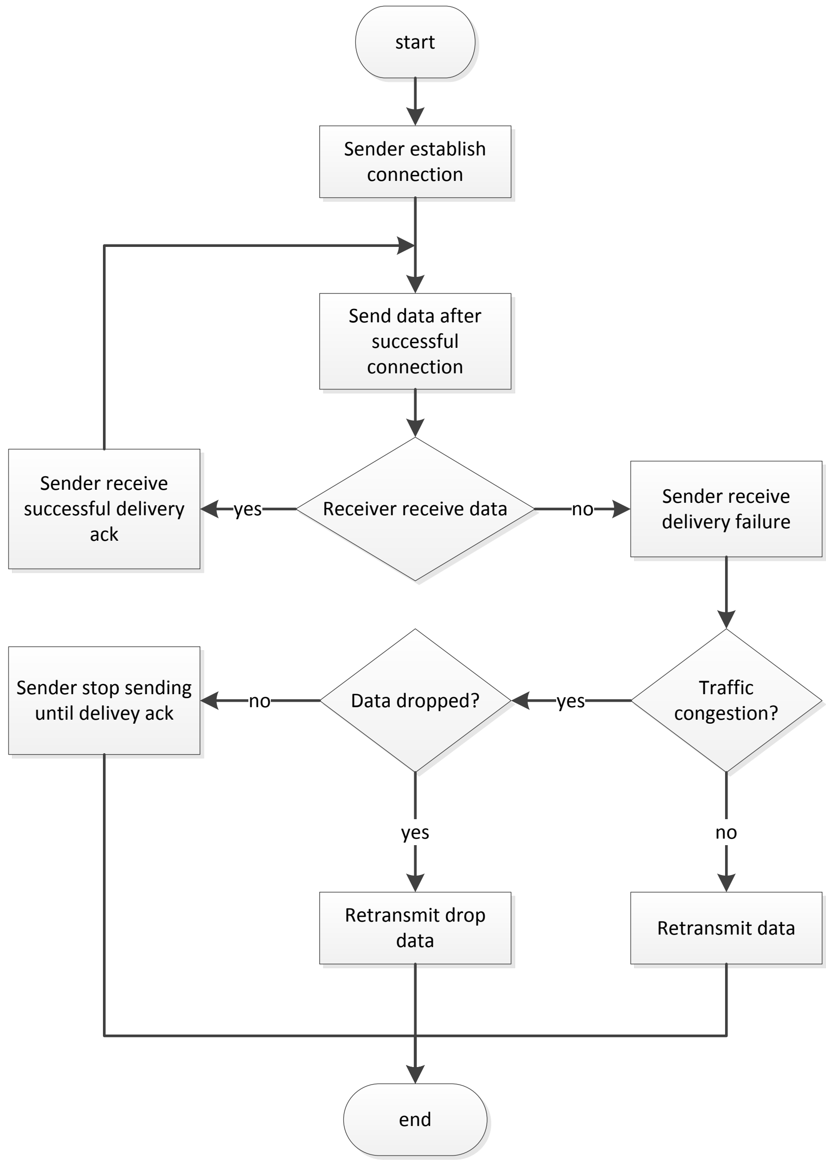

Figure 3-13: WXCP Flowchart

although it not cross-layer aware. A *window based* strategy with some rate-based constituent in WXCP can *enable* the *source* to change from *window-based default* to a *slow rate-based control mechanism*. Via the *discovery mode*, WXCP enables *the source* to keep on checking the *recent loss outline*. The performance evaluation for WXCP was performed using NS2 (version 2.29) network simulator. TCP New Reno was utilized as the basis for performance comparisons.

```
Start ()
{
  sender: establish connection to receiver;   //handshake
    if (connection established succcessful)  //receiver allow sending to start
        {
          send data to receiver; //sender
           if (receiver receives data)
            {
          receiver ; send delivery ack;  // indicating successful data delivery
        }
     else {
          undelivered data ack;    // indicating data didn't reach destination
      }
  if (delay or congestion ack)    // indicating data delay
   {
   check if (data drop)    //check it was drop due to congestion or delay
    {
      retransmit lost data;   //lost due to either delay or congestion
   increase transmission pace ; //fasten data delivery
  }
 End
}
```

Figure 3-14: Pseudocode for WXCP

The results indicated that WXCP outperforms TCP New Reno when tested using performance metrics namely throughput, congestion, window and packet lost. UDP variants, other Entirely New and Hybrid TLP were not included for performance comparisons. Before we actually coded the WXCP, we developed the flowchart presented in Figure 3-13 which we followed to implement WXCP in NS2. Figure 3-14 gives pseudo code as entire logic followed to implement WXCP. The characteristics of WXCP are listed in Table 2-24 in the previous chapter.

3.2.4.2. Ad Hoc Transport Protocol (ATP)

The Ad hoc Transport Protocol (ATP) was designed to improve TCP's performance degradation in wireless ad hoc network. ATP primarily consists of *mechanisms* at the *sender* to achieve effective congestion *control* and *reliability*. However, unlike in TCP, ATP relies on *feedbac*k not just from the *receiver*, but also from the *intermediate nodes* in the connection path. In terms of specific functionality, the intermediate nodes provide *congestion feedback* to the sender, while the *receiver provides* feedback for both *flow control* and *reliability*.

The receiver also acts as a *collator* of the congestion information offered by the *intermediate nodes* in the network before the information is sent back to the *sender*. The receiver provides the *reliability*, *flow control*, and collated *congestion control* information via *periodic messages*. The *sender*, on the other hand, is responsible for *connection control*, *start-up ratio estimation*, *congestion control*, and *reliability provision*. Given that ad hoc networks are typically stand-alone approach to the problem of reliable TLP from the perspective that it justifiable to develop an ENT (ATP) TLP that is not TCP variant.

The performance evaluation results through NS2 based simulation show that ATP outperforms both default TCP and TCP-ELFN. The performance metrics considered to measure the performance of the ATP are instantaneous throughput, congestion window, and packet lost. UDP variant, ENT as well as Hybrid TLP categories are not included for performance comparisons.

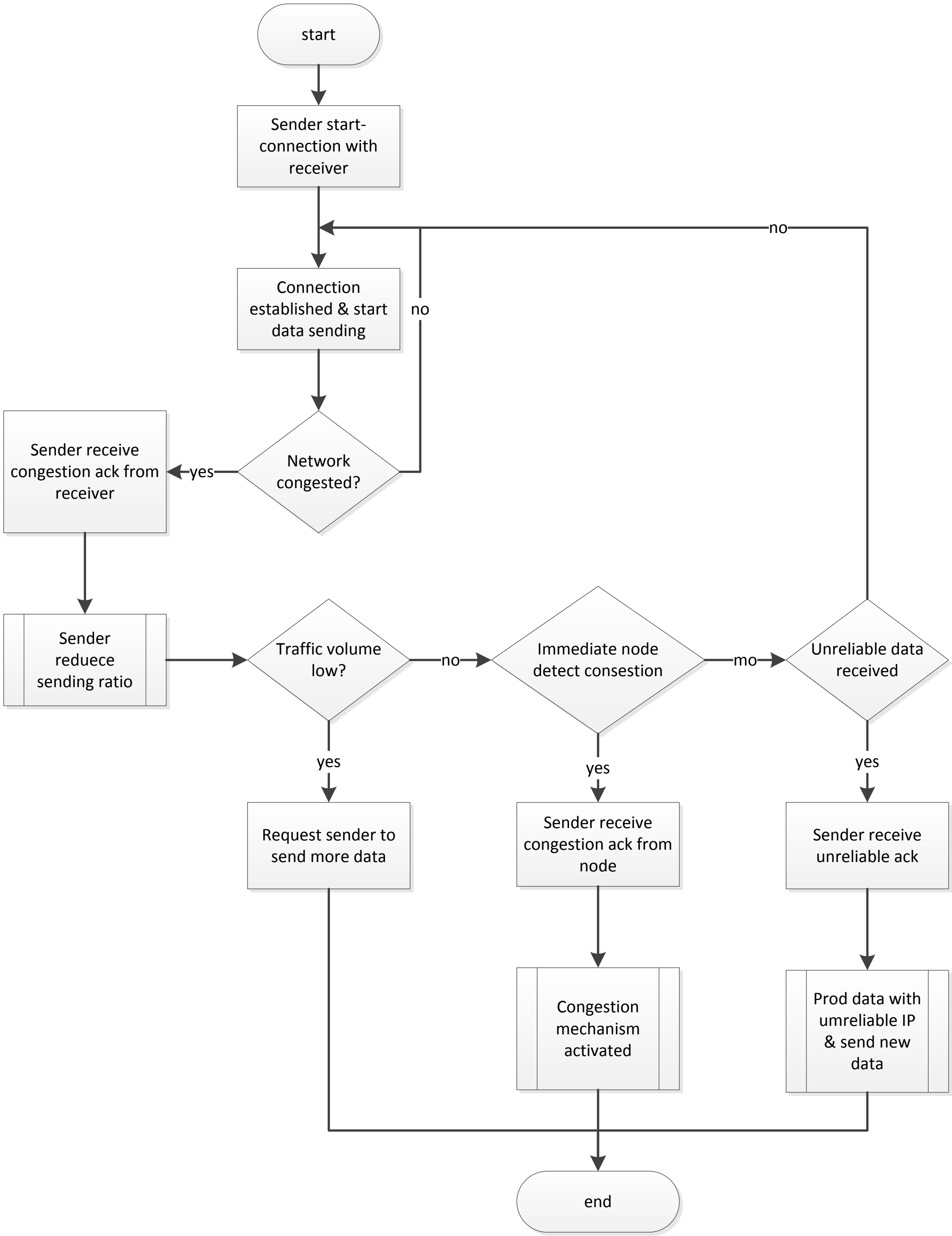

Figure 3-15: ATP Flowchart

```
Start ()
{
  sender: start-up connection;
      accept connection;  //receiver
       send packets;  // Sender
     feedback to sender;  //receiver, delivery ack
   if (traffic is congested)
         {
           receiver: inform the sender and control the packets flow;
        sender reduce sending ratio;
     when: traffic volume is low;
 receiver acknowledge sender to send more packets; //receiver keeps sending
else if (nodes detect congestion) // before even reaching the receiver
 immediate Node: send congestion ack to sender;
     sender: activate congestion control mechanism;
       if (unreliable data received)    // receiver get data out of order
          {
             receiver: send unreliable ack;
        sender: confirm the sequence and address;
 if (address does not correspond)
      {
       discard data;   // discard data from intruders
        }
       else
       {
      resend data; // if the address is correct
     }
  End
}
```

Figure 3-16: Pseudocode for ATP

The flowchart in Figure 3-15 gives the steps we followed to implement ATP in ns2. The entire logic followed to implement ATP is shown in Figure 3-16. The characteristics of this TLP are listed in Table 2-22 in the previous chapter.

3.3. Summary

This chapter has discussed the analysis, pseudocode and flowcharts of the eight TLP selected for evaluations. The analysis of the TCP variants is presented in section 3.2.1. Section 3.2.2 has given analysis of the variants of UDP. Hybrid TLPs have been analyzed in section 3.2.3. Finally, Section 3.2.4 has explained the analysis of the Entirely New TLPs. Chapter Four gives the experiment results and their analysis.

CHAPTER FOUR

PERFORMANCE ANALYSIS OF THE SELECTED TLPs

4.1. Introduction

Currently, there is a large number of TLPs applicable to WMNs, but we are not sure which TLP is most suitable for WMNs, since these TLPs are not compared in a consistent way. Therefore, instead of developing the new TLP from the sketch, while there are TLPs existing already, we critically analyzed the existing protocols with the aid of a literature review framework developed in Chapter Two in order to select the TLP to be compared.

In this chapter we evaluate (through simulation using NS2) eight selected TLPs applicable to WMNs in order to compare them to determine the most suitable TLP for WMNs amongst the four identified categories. Section 4.2 gives the setup for the whole experiments. Experimental results are presented in section 4.3. The summary of the results and simulator and experiment limitations are given in section 4.4 and section 4.5, respectively. Finally, section 4.6 gives the recommendations we made for an ideal TLP applicable to WMNs.

4.2. Experimental Setup

This section is made up of two subsections. The first subsection presents the simulations metrics that have been used to evaluate our selected TLPs, whereas the second subsection gives the details about the simulations environment.

4.2.1. Recorded Performance Metrics

The performance metrics to evaluate the relative performance of the selected TLPs applicable to WMNs were identified from the literature. The chosen metrics are as follows:

i. *Packet Delivery Ratio (PDR)*: the percentage of application layer packets containing unique packet identifiers (Ids) received at the intended destination. As TLPs attempt to ensure that all data packets being send rich their destination safety, therefore, we considered PDR to examine the degree of data packet delivery.

ii. *End-to-End Delay*: an expression of how much of time it takes for a data packet to get from designated point (source) to another point (destination) when a certain TLP defined in Transport Layer applied in WMNs. Delay is a function of protocol and traffic characteristics. Therefore, when comparing protocols, it is necessary to compare them based on the same traffic parameters.

iii. *Throughput*: how much data packets can be transferred and delivered successful at given amount of time. The objective of TLPs is to maximize the throughput (allowing more data packet to be delivered) while minimizing the network delay.

iv. *Packet Retransmission*: refers to the number of retransmitted packets at a given time during the transmission of data packets from source to destination over a network. Some TLPs have retransmission mechanism to retransmit the lost data packets in order to guarantee high reliability.

v. *Round Trip Time (RTT)*: refers to the amount of time it takes for a data to travel in both directions (from source to designated destination and vice versa) over a network.

4.2.2. Simulation Environment

The representative TLPs were simulated using Network Simulator (NS2) tool version 2.33 [http://www.isi.edu/nanam/ns] running on Ubuntu Linux 9.10 operating system which support IEEE 802.11 standards with many subsequent patches/updates published by the NS2 user community to improve the IEEE 802.11 simulation model (Marco Fiore, 2004). All data was collected using Aho Weinberger Kernighan (AWK) scripts (See Appendix A), which is the programing language that is designed for processing text-based data, either in files or data streams, and was created at Bell Labs in 1970s. Table 4-1 represents the parameters used in our simulations. In the following subsections we describe the models of the various layers of the IEEE 802.11 protocol stack that were used in this simulation.

4.2.2.1. Physical and Data Link Layer Model

In this study nodes were assumed to be making use of omni-directional antennas. This format is valuable for broadcasting a signal to all directions, or when listening for signals from all directions. This model provides data transfer for reliable or unreliable applications, in the case of transmission errors the higher-level protocols (TLPs) provide flow control, error checking, acknowledgements and retransmission.

4.2.2.2. Medium Access Control

The Medium Access Control (MAC) is a sub layer of Data Link Layer and it provides addressing and channel control mechanisms that make it possible for network nodes to

Table 4-1: Simulation parameters Considered

Parameters	Environment
Simulation Time	1000s
Number of Nodes	20, 40, 60….200
Map Size	2500x1500
Mobility	None
Packet size	512 byte
Connection Ratio	4.0 for all nodes
TLPs	TCP, UDP, Snoop, TCP-AP, HERC, LATP, WXCP, ATP, LLEP-TCP, LATP

communicate within a multi-hop network. The link layer of our simulator is based on the IEEE 802.11standard MAC protocol distributed Coordination Function (DCF), in order to accurately model the node contention for wireless medium. DFC designed to minimize the possibility interference owing to hidden terminals utilizing Physical Carrier Sense and Virtual Carrier Sense mechanisms.

The transmission of the packets is preceded by a Request-To-Send (RTS), Clear-To-Send (CTS) exchange that reserves the wireless channel for transmission of data packets. When each packet received by a destination node, an ACK is send to the source. If the ACK is not received for particular period of time, the source retransmits a packet a limited number of times until these ACKs are delivered.

4.2.2.3. Packet Buffering Model

All wireless multi-hop network nodes in the simulation used a buffer for both data and packets that are awaiting transmission. The buffer was able to accommodate fifty packets and implements the drop-tail queue management algorithm which requires minor management. In addition, in this buffer, packets are transmitted in the first come first served basis. If the buffer is full, any new packets are dropped.

4.2.2.4. Data Traffic Model

To simulate the Transport Layer Protocol communication between nodes in the network we used constant bit ratio (CBR) and TCP. Despite the lack of realism, it was deemed that the use of CBR and TCP traffic would not have impacted on the relative abilities of the network TLPs being investigated to facilitate the delivery of the packet to their intended destinations.

4.2.2.5. Routing Protocol

The Hybrid Wireless Mesh Protocol (HWMP) (Bahr *et al*, 2006) routing protocol was selected to assist in the performance measurement of the TLPs applicable to WMNs. The HWMP is a newly produced standard routing protocol for Wireless Mesh Networks. Thus, we decided use the protocol of the latest standard. On the other hand, wireless mesh network is more static. Therefore, since our evaluations were based on a static wireless networks we considered HWMP as the most reliable routing protocol for our experiments.

4.2.2.6. Traffic Generation

In the study, we used connection pattern file generator in order to generate traffic. The following is used directory: *~ns/indep-utils/cmu-scen-gen*. Two different connection pattern files: cbrgen.tcl- for generating CBR connections and TCP connections is used. We use built-in scenario and traffic file generators. Method to create CBR/TCP connection to create CBR/TCP connections, run the script: *ns cbrgen.tcl [-type cbr/tcp] [-nn nodes] [-seed seed] [-mc connections] [-ratio ratio*], for example, *ns cbrgen.tcl -type cbr -nn 25 -seed 1 -mc 8 -ratio 4* OR *ns cbrgen.tcl -type tcp -nn 25 -seed 1 -mc 8*.

4.3. Experimental Results

This section presents the results obtained from our investigation to identify optimal TLPs applicable to WMNs among the five existing TLP categories, identified and described in Chapter Three. The TLPs being evaluated can be found in Table 2-28. The data is collected from four perspectives namely: network size, number of flows, distance between nodes and simulation time, respectively.

4.3.1. Performance Metrics versus Network Size

The purpose of the network size is to measure the scalability of the TLPs over WMNs, since the study is based on a small rural village community with the potential to grow more in the future, WMNs may consist of large number of nodes when the village has grown up, hence TLP scalability should be taken into considerations. Therefore, TLPs applicable to WMNs address the challenges of scalability through network size in this study. In this section, we describe and analyse the results of Experiment 1, Experiment 2, Experiment 3, Experiment 4, and Experiment 5 against the network size. In these experiments we recorded the following performance parameters: throughput, delay, packet delivery ratio, packet retransmission and round trip time against network size. The results analysis of these experiments can be found in *subsection a*, b, c, d and e, respectively.

a. Experiment 1: Effects of network size on the Throughput

The purpose of this experiment was to determine the degree of successful data packets transmitted over the WMNs communication channel using the ten selected TLPs. Throughput is one of the quality of service (QoS) parameter (Naeem *et al*, 2010), (Linn *et al*, 2005), thus, it was considered in the study to indicate the QoS level yield by TLPs applicable to WMNs. The good throughput signifies the good QoS offered by TLPs over WMNs. Table 4-2 shows the data results for the network per network size per TLP. The data is graphically depicted in Figure 4-1 to highlights the relative performance of TLPs applicable to WMNs, when throughput against the different network sizes considered.

Traditional (TCP and UDP) TLPs had the least throughput than all other TLP categories. TCP performance degrades over WMNs since TCP is greatly affected by packet loss due to link layer contentions rather than loss due to congestion whereas UDP performance over WMN is poor due to the fact that it does not contain a congestion control mechanisms. The UDP variants (HERC and LATP) outperform traditional TLPs, as UDP variants have control mechanisms for media streaming applications in contention-based

Table 4-2: Results for the network throughput per network size

NN	TCP	UDP	SNOOP	TCP-AP	HERC	LATP	WXCP	ATP	LLE-TCP	LLAP
20	119.36	138.46	252.24	263.66	205.73	186.43	342.93	310.71	344.74	331.99
40	103.21	127.73	232.89	242.60	190.99	170.38	308.23	272.50	319.89	305.39
60	96.22	106.79	195.09	201.95	156.13	157.85	247.73	275.23	341.59	330.84
80	81.92	92.72	222.35	214.60	184.17	121.42	270.43	246.18	300.60	290.60
100	93.31	97.34	179.70	175.12	106.62	141.62	255.70	247.20	324.70	299.95
120	79.48	87.74	180.24	189.96	134.85	131.36	231.74	210.99	299.74	259.24
140	89.34	96.13	152.98	168.10	95.95	128.45	215.48	193.98	285.48	264.48
160	72.14	89.55	141.72	154.10	107.39	122.64	230.22	187.72	273.72	260.47
180	53.97	76.08	137.67	153.78	81.66	118.91	213.93	160.66	284.92	258.17
200	52.44	66.33	114.53	138.53	75.62	96.62	181.35	154.35	252.03	236.53

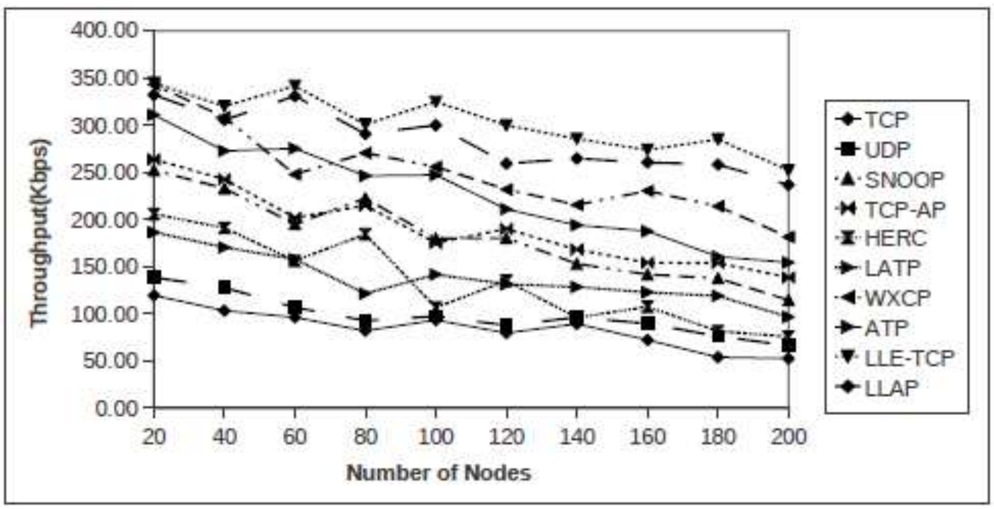

Figure 4-1: Throughput versus network size

wireless multi-hop networks. During the transmission of data packets, UDP variants control the sending ratio based on the feedback received from the destination. According to this result, HERC with speed control technique allows more packets to reach destination quickly when network size is small and allows more packets to get lost when the network size is big. UDP variants achieve lesser throughputs than TCP variants, ENT as well as Hybrid. The poor performance of UDP variants results from an unreliable service provided by UDP variants with the assumption that error checking and correction is either not necessary or performed in the application, avoiding the overhead of such processing at the network interface level. The performances of UDP variants diminish as the network sizes enlarge. TCP variants (Snoop and TCP-AP), with adaptive pacing mechanism produces better throughput than both traditional and UDP variants based on the results presented on above. This adaptive pacing mechanism controls the level of data inserted into the network. When there is more data consuming the network bandwidth, it reduces ratio of inserting the data into the network and vice versa. This mechanism reduces the probability of network congestion as well as the number of data lost.

Snoop produces lesser throughput than TCP-AP despite the fact that Snoop can control the ratio of data packets when loaded into network. The outstanding performance by TCP-AP comes from the fact that it has an adaptive pacing based on hybrid ratio and congestion control mechanisms. Figure 4-1 shows that throughput for ENTs is better than all categories but not Hybrid TLPs. The higher throughput produced by the ENT is due to two facts. Firstly, they cater for the exact need for that particular type of network .i.e. WXCP was designed to cater for the characteristics of WMNs as results of poor performance by TCP.

Secondly, ENTs congestion control and reliability mechanism are decoupled. Therefore, congestion control is performed only if network is congested. Similarly, the packet retransmission takes place only when there is packet lost. Congestion control is performed using feedback from network, while reliability is ensured through coarse grained destination feedback and selective ACKs. ENTs have better throughput than other TLP categories, but not Hybrid TLPs. ENTs have better throughput because of the ability to make more precise estimation of the congestion conditions and compute the ratio feedback based on multiple congestion metrics. The ENTs attempt to ensure that a congestion control mechanism is being applied only if the network is congested through decoupling congestion control reliability. Using explicit ratio feedback instead of probing the available bandwidth, ENT flows are able to converge to a transmission state where better throughput is achieved.

Although the ENTs achieve better throughput, but when network size grow up their performance declined. As shown in Figure 4-1, Hybrid TLPs outperform all other TLP categories. Using cross-layer aware and QoS mechanisms the Hybrid TLPs produce better throughput. This category enhanced TCP over large variety of wireless networks working with both real time and non-real time applications. The adaptive controls mechanisms to control the traffic load offered into the networks allows the Hybrid TLPs to achieve better throughput while it also improves the performance of high layer protocols without any modification to them.

LLE-TCP outperforms LLAP by reduces the medium busy time for the transmission of data packets over wireless link. Hybrid TLPs are considered as an optimal performing category based on the results presented in Figure 4-1. Despite the observation that Hybrid TLPs are found to be the better performing category, but their throughput level decreases as the network size is escalating.

b. Experiment 2: Effects of the network size on Delay

The purpose of this experiment (delay) was to determine the average time taken to deliver the Application Layer data packets from the source to the intended destination. Packets delay may result in the packet losses and false indication of the network congestion (Reaz and Atiquzzaman, 2005). Therefore, delay was considered in the study to find out the impact it has on the performance of TLPs applicable to WMNs. This experiment enabled to identify the TLP that mitigate packet loss and delay in WMNs.

On the other hand, delay signifies the QoS offered by TLP when applied to WMNs, thus it was considered from the fact that is one of the QoS parameters (Naeem *et al*, 2010), (Navaratnam *et al*, 2008), (Linn *et al*, 2008). The results for this experiment are presented in Table 4-3, while Figure 4-2 shows the effects of delay against the network size. This process cost the transmission of packet by TLPs, including the time taken to establish a route as well as the actual time taken to deliver the Application Layer data packets, via intermediate nodes to their intended destination. Traditional TLPs such as TCP and UDP, both experienced more end-to-end delay than all TLPs categories.

The traditional TLPs were designed without considering wireless network characteristics whereas the other TLP categories were designed with wireless in mind. Among the two Traditional TLPs, TCP fares more end-to-end delay than UDP. TCP has more delay since it assumes that all data lost are due to network congestions but packet losses in WMNs are usually due to medium contention, link failure, and route failure (Fu *et al*, 2003), (Navaratnam *et al*, 2007).

Table 4-3: Results for Delay per network size

NN	TCP	UDP	SNOOP	TCP-AP	HERC	LATP	WXCP	ATP	LLE-TCP	LLAP
20	512.10	459.85	324.80	317.66	452.88	402.31	252.87	225.87	149.30	117.05
40	587.42	500.69	384.01	356.22	455.30	436.76	277.63	234.15	163.76	138.01
60	635.71	554.66	407.93	368.66	500.05	435.55	302.72	272.47	191.68	165.93
80	586.52	533.56	428.03	394.89	471.79	505.79	314.01	299.89	172.03	143.28
100	616.35	601.25	455.53	423.94	525.47	512.22	385.03	326.78	207.03	196.03
120	653.16	637.20	471.67	401.18	546.10	540.85	341.67	360.67	240.42	228.92
140	645.31	610.32	458.92	437.52	587.99	523.49	389.42	336.67	242.67	224.92
160	668.39	653.98	499.34	478.55	603.43	563.93	409.09	351.09	289.09	252.59
180	700.88	663.68	527.54	458.40	629.00	592.50	418.76	395.76	323.79	275.04
200	712.74	683.54	552.56	539.31	649.43	626.93	449.26	421.76	347.81	286.81

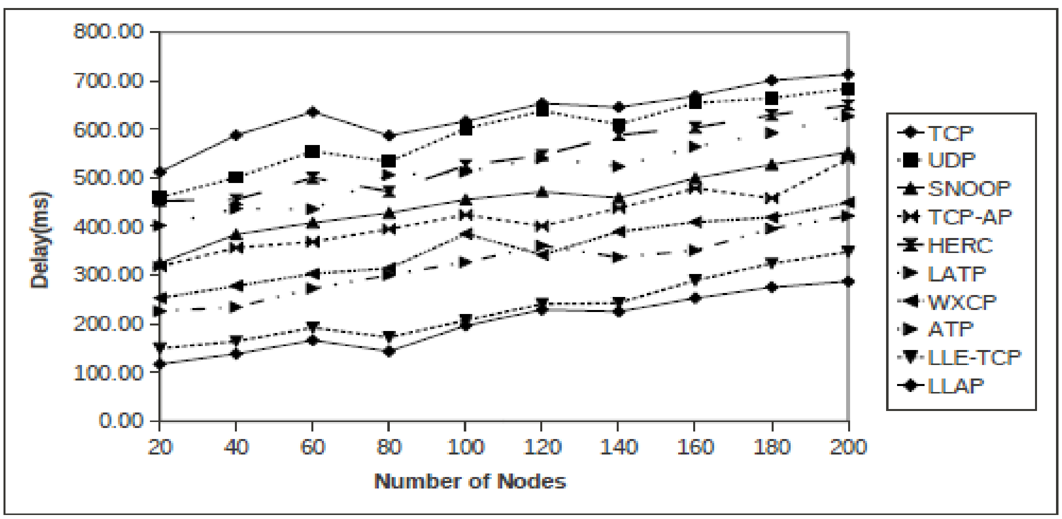

Figure 4-2: Delay versus network size

However, unlike TCP, UDP uses a simple transmission model without implicit hand-shaking dialogues for guaranteeing reliability, ordering, or data integrity. Therefore, UDP provides an unreliable service, data packet may arrive out of order, appear duplicated, or go missing without notice but UDP is time-sensitive, it attempts to ensure that delay is reduced. Both the UDP variants experienced high levels of latency than all TLPs but not than traditional TLPs. As UDP variants have no congestion control mechanisms, therefore, ratio control protocol (RCP) is added on top of them. The RCP was designed for wired network as a result; it does not differentiate between losses caused by buy congestion or wireless channels. Therefore, UDP variants produce more delay as well packet loss when applicable to WMNs. Within the UDP variants, HERC experiences more delay than LATP. The performance of HERC depends on the fact that it replaces the TCP response function with high-speed response function to provide speed. However, the problem with HERC is that it does not consider the order in which data is delivered. Although it allows quick data delivery, more data gets lost and delivered out of order.

LATP experienced less delay than HERC in WMNs because its mechanisms to control medium contention enable the data packets destination quickly (Navaratnam *et al*, 2007). Both UDP variants encounter more delays as the network size grows. The delay experienced by ENT TLPs in WMNs is far lesser than the delay experienced by traditional TLPs and UDP variants but not Hybrid. TCP variants perform better than traditional TLPs as they are designed with the considerations of wireless networks characteristics.

Snoop and TCP-AP have been designed to tailor traditional TLPs problems in wireless environments such invoking congestion control, while the networks is experiencing bit errors not congestions (Akyildiz and Wang, 2005), (Sundaresan *et al*, 2003), (Liu *et al*, 2001). TCP-AP produces less delay than Snoop by using adaptive pacing mechanism to adjust the pace of data loaded into the network for transmission. TCP variants, at somepoints experience same delay i.e. at 140 and 160 node; it is because both TCP variants have similar mechanisms to differentiate between the congestion and non-congestion loss. All TCP variants produce more delay as the network size enlarges. ENTs are designed to solve fundamental problems such as route changes, link failure, and medium contention in traditional TLPs for specific types of networks. These TLPs utilize a special set of mechanisms for reliable data transport and achieve much better performance in delay as well throughput than TCP variants as well as UDP variants in WMNs. ATP experienced more end-to-end network delay than WXCP due to the fact that WXCP consists of an explicit flow control mechanism designed specifically for WMNs.

This explicit flow control mechanism reduces network congestion. The congested network increases the chances of the data packet being delayed. Both ENTs experience more delay than hybrid TLPs and delay produced by ENTs increase as the network size increases. The Hybrid TLPs were found to be the best TLPs as far as delay is concerned according to the results presented in Figure 4-2. Less delay produced by Hybrid TLPs when applied to WMNs as they work efficiently with both real time and non-real time applications.

The outstanding performance of the Hybrid TLPs also comes from the fact that it is cross-layer aware; being cross-layer aware allows these TLPs to work efficiently at the different layers of the protocol stack. However, Figure 4-2 shows that LLAP performs better than LLE-TCP, as WMNs are greatly affected by packet loss due to link layer contentions rather than loss due to congestion (buffer overflow at the intermediate nodes)

(Franklin *et al*, 2006). Hence, LLAP with its mechanism to distinguish congestion losses from non-congestion losses and link layer adaptive pacing mechanism, has least delay. Similar to all other TLP categories, Hybrid TLPs encounter more network delay as the network size increases. These results show that Hybrid TLPs outperform all TLP categories with the LLAP as the best performing TLP.

c. Experiment 3: Effects of the network size on the Packet Delivery Ratio (PDR)

The purpose of this experiment was to determine the network's ability to successfully deliver the data packets being sent, using particular type of TLPs applicable to WMNs. A PDR 0% represents the total failure of the network to deliver data packets whilst the PDR 100% shows that the all sent data packets in the network were successful delivered. PDR signifies quality of service (QoS) of TLP over WMNs as PDR is the QoS parameter (Naeem *et al*, 2010), (Navaratnam *et al*, 2008), (Linn *et al*, 2008), therefore, PDR was considered in our study to indicate the QoS as the network size increasing.

Table 4-4 shows data for the effect of network size on packet delivery ratio (PDR) using TLPs applicable to WMNs. This data is plotted in Figure 4-3, which is used to graphically depict the behaviour of TLPs on the packet delivery ratio against the network size. As depicted in Figure 4-3, one of traditional TLP (UDP) has a very low packet delivery ratio, while Hybrid TLPs have a very high PDR. This indicates that TLPs with good throughput are the ones with high packet delivery ratio also. UDP has less data packets delivery than all TLP, with maximum and minimum of 53% and 30%. The poor performance of UDP is due to the fact that UPD is an unreliable TLP.

Table 4-4: Results for the packet delivery ratio per network size

NN	TCP	UDP	SNOOP	TCP-AP	HERC	LATP	WXCP	ATP	LLE-TCP	LLAP
20	63.74	51.97	66.48	70.60	55.76	60.94	74.93	80.08	86.01	92.35
40	63.37	45.98	64.91	68.66	54.57	58.67	71.86	76.68	85.93	92.51
60	58.80	46.80	65.55	67.52	50.31	55.28	72.55	76.60	83.33	87.06
80	57.24	43.97	62.93	64.04	46.65	52.60	70.83	74.64	81.38	84.97
100	53.95	41.85	58.84	64.04	48.65	48.63	68.88	73.32	80.41	81.37
120	48.09	36.75	55.27	59.21	44.37	49.09	64.33	70.29	76.04	84.77
140	50.25	37.60	52.54	56.21	43.54	46.42	66.68	67.92	75.76	78.35
160	46.96	33.27	49.93	52.26	38.12	40.67	62.59	67.53	73.88	77.02
180	43.83	33.57	46.71	50.42	35.87	38.76	58.03	66.75	72.85	74.03
200	40.25	29.84	44.70	46.04	34.85	37.95	54.62	61.22	68.05	72.60

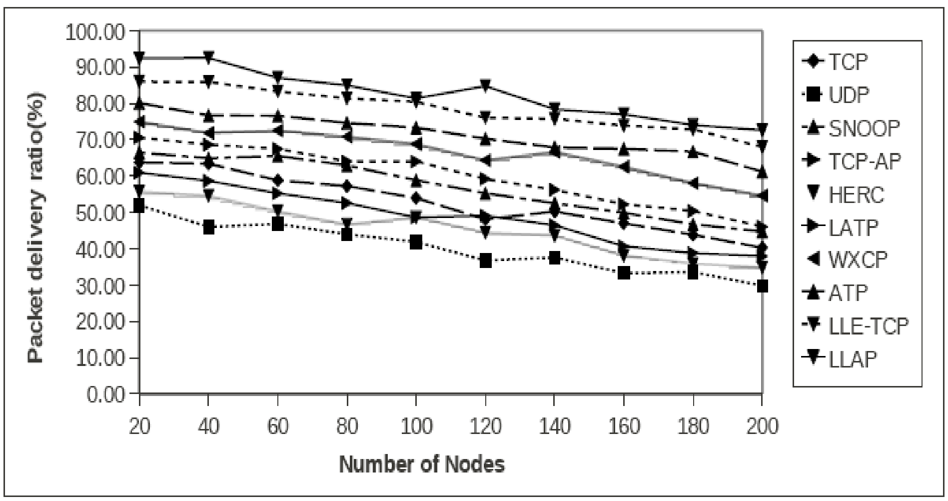

Figure 4-3: Packet delivery ratio versus network size

TCP with the ability to retransmit the packet that did not reach an intended destination until it reaches the destination has more delivery ratio than UDP and UDP variants. UDP variants experience better PDR when network size is small, but as the network size expands the number of successfully delivered packets is reduced. The PDR for UDP variants applied to WMNs lower than all TLP categories and one of the traditional (TCP) but not UDP. The multi-metric joint detection based rate control mechanism enables the

UDP variants to drop fewer packets than UDP TLP. As UDP variants use unreliable equation based rate control to reduce packet losses, which attempts to control network congestion but is not aware of the medium contention and channel errors in WMNs. Therefore, UDP variants deliver less number of data packets in WMNs. LATP with end-to-end rate control scheme based on medium access control (MAC) layer feedback of the bottleneck node's permissible packet delivery information, has more packet delivery ratio compared to HERC.

The permissible packet delivery information indicates the degree of medium contention on the path from source to destination. LATP control the ratio of data packets into the network accordingly in order reduce network overload. Figure 4-3 depicts that TCP variants outperform traditional TLP and UDP variants. The feedback mechanism to detect congestion and the mechanism differentiating different packet losses allow the TCP variants to successfully deliver more data packets than traditional and UDP variants in WMNs.

Snoop communicates over wireless links without triggering retransmission and window reduction policies at the transport layer makes Snoop performance better than traditional TLPs and UDP variants. While Snoop reducing retransmission, TCP-AP, on the other hand, incorporates rate-based transmission algorithm into TCP's window-based congestion control to minimise network congestion. As shown in Figure 4-3, ENTs successfully deliver more data packets all TLP categories but not Hybrid TLPs, with the maximum and minimum of 75% and 65%, and 80% and 70% for both WXCP and ATP, respectively. The ENTs, with loss discovery and pacing mechanisms introduced at the source to deal with tiny window and burst problem accordingly; thus, gives the ENT ability to reduce the number of lost data packets. A rate based transmission technique which quick-start during connection initiation and route switching minimize the number of packets dropped by ENTs applicable to WMNs. ATP which totally incompatible with TCP decouples reliability and congestion control mechanisms in order to use them in an interchangeable manner.

Hybrid category, to the best of our knowledge is the only category with the ability to support the transmission of both real time and non-real time applications. Figure 4-3 highlights that the Hybrid category outperforms all TLP categories with higher delivery ratio percentage ranges between 85% to 75% and 95% and 78% both LLE-TCP and LLAP, respectively. These TLPs are QoS and Cross-layer aware. The cross-layer mechanisms break the ISO/OSI principle of layers as it permits the layers of the stack protocol to work with more than one layer, therefore optimizing the protocol stack (Nascimento *et al*, 2008).

The Hybrid TLPs yield far better performance in terms of PDR as these TLPs contain mechanisms to adaptively control the load offered into the networks. Hybrid TLPs use a positive acknowledgement schemes which specify TCP destination to acknowledge data successfully received from source.

d. Experiment 4: Effects of the network size on the Packet Retransmissions (PR)

The main purpose for this experiment was to determine how many packets are being retransmitted during the transmission of data packets from source to destinations when particular TLPs applicable over a WMNs. PR signifies the level of network congestions, since congestion control mechanisms assist in reducing the number of packet retransmissions (Iyer *et al*, 2005), thus, the number of packet retransmissions is considered to indicate the degree of network congestion experienced by WMNs when the selected TLPs applied to it.

On the other hand, this experiment assisted to investigate whether the TLP is reliable over WMNs as the PR is one of the mechanisms used to ensure reliability and occur when it is certain that a packet to be retransmitted was actually lost (Chen *et al*, 2004) or not. Table 4-5 shows the data results which indicate the number of retransmitted packets per various network sizes. Figure 4-4 depicts how much throughput can be produced against various number of packet retransmission by the TLPs applicable to WMNs. UDP and UDP variants are not considered in this experiment since they do not have packet retransmission mechanisms.

The results highlighted in Figure 4-4 depict that for all TLP categories the number of retransmitted packets increases as the network size increases. The traditional TCP has more number of retransmitted packets, TCP encounters more packet errors ratio in WMNs, which force more packets to be retransmitted and attempt to ensure reliability. A huge number of packet retransmission is experienced by traditional TLP when the

Table 4-5: Number for retransmitted packets

NN	TCP	SNOOP	TCP-AP	WXCP	ATP	LLE-TCP	LLAP
20	131.00	91.00	77.00	67.00	68.00	51.00	35.00
40	155.00	133.00	130.00	111.00	100.00	77.00	55.00
60	179.00	169.00	143.00	124.00	138.00	134.00	101.00
80	358.00	228.00	179.00	119.00	128.00	139.00	140.00
100	439.00	398.00	298.00	236.00	196.00	188.00	165.00
120	421.00	367.00	334.00	270.00	240.00	212.00	190.00
140	506.00	396.00	289.00	250.00	265.00	178.00	204.00
160	472.00	382.00	353.00	336.00	312.00	276.00	244.00
180	519.00	476.00	349.00	342.00	294.00	282.00	268.00
200	563.00	488.00	378.00	327.00	353.00	290.00	288.00

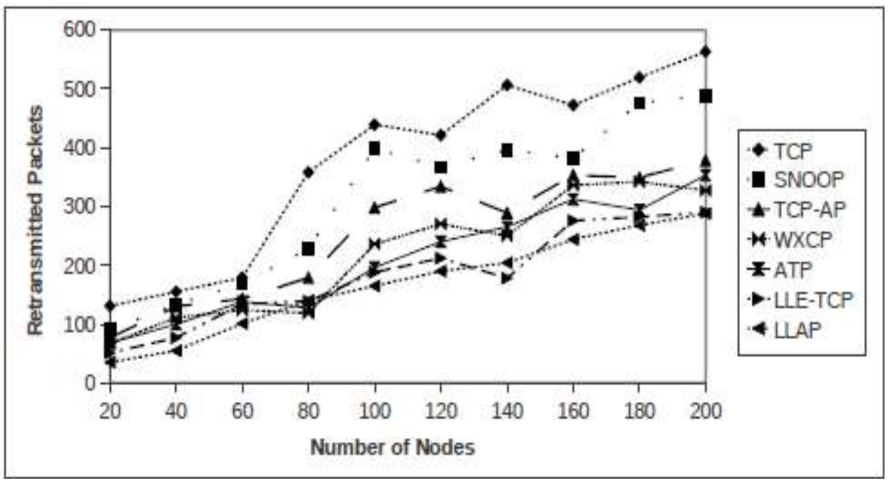

Figure 4-4: Number of retransmitted packet versus network size

network size is vast. In wireless multi-hop networks, the most repeated cause of packet lost is bit error in packets. Therefore, TCP experienced more packet losses in WMNs as it treats all packet losses as congestion losses. TCP variants fare much better the number of packet retransmissions than traditional TLPs (TCP) as they have feedback for detecting congestion and differentiating congestion and non-congestion packet losses. Among the TCP variants, Snoop experiences more packet retransmissions than TCP-AP, since TCP-AP sends packet at a pre-determined ratio instead of sending new packets into the network only when old packets have been acknowledged. Therefore, TCP-AP allows more packets to reach their destination without retransmission. The adaptive pacing based on hybrid ratio and congestion control by TCP-AP reduces network congestion and packet drop ratio. As result the number packet retransmissions is minimized. Figure 4-4 shows that Entirely New Transport Layer Protocols (ENTs) have less packet retransmissions than traditional TLP and TCP variants, but not Hybrid. TCP and TCP variants experience more packet retransmissions than ENTs, since they lack accuracy of contention estimation.

On the other hand, ENTs make more precise estimation of congestion conditions and computes the ratio feedback based on multiple congestion metrics. ENTs also utilize an explicit ratio feedback instead of probing the available bandwidth. Therefore, ENTs are able to converge to transmission state where better throughput and less packet retransmission is achieved which not the case in TCP and TCP variants. In Figure 4-4, Hybrid (LLE-TCP and LLAP) TLPs outperformed all the other TLPs concerning packets retransmissions (PR). Hybrid reduces the PR by suppressing the TCP ACK which reduces round trip time (RTT) by the time required for TCP ACK transmission over the wireless link.

The LLAP produce far less number of retransmitted packets than LLE-TCP. LLE-TCP only achieved less number of retransmitted packets than LLAP when the network is made up of 130 to 150 nodes, since LLAP adaptively estimates the Four-Hop transmission Delay (FHD) on the path and transmits the packets with estimated FHD interval. The

Hybrid TLPs control network congestion by propagating the congestion information to the source without any additional control overhead and do the pacing at the link layer. Thus, this TLP category improves the performance of the WMNs irrespective of higher layer protocols.

e. Experiment 5: Effects of network size on the Round Trip Time (RTT)

The purpose of RTT is to determine the amount of time taken by network communications (acknowledgements) starting from when specific source sends packets to a specific destination and back again to the source. This experiment investigates the TLP with less round trip delay when applied to WMNs with various network sizes to indicate the degree of the network congestion. The Table 4-6 shows the data results for round trip time against network size of TLPs applicable to WMNs. The data presented in Table 4-6 was used plotted Figure 4-5 to depict the round trip time versus network size for TLPs applicable to WMNs.

Similar to Experiment 4, only seven TLPs considered in this experiment as well. The traditional TLP produces more RTT than all TLPs categories studied in this experiment. TCP encounters several problems in WMNs because of it tendency to assume that all packets losses are due to network congestions whereas the network congestion is not the main problem in wireless networks. TCP variants have lesser RTT than TCP as the performance of TCP is greatly affected by the packet loss in the wireless networks owing to problems such as route and link failure, and channel bit error ratio.

Table 4-6: Results for Round Trip Time per network size

NN	TCP	SNOOP	TCP-AP	WXCP	ATP	LLE-TCP	LLAP
20	1078.00	789.00	621.00	534.00	398.00	274.00	186.00
40	1098.00	997.00	789.00	596.00	462.00	356.00	213.00
60	5321.00	3547.00	3123.00	2896.00	2134.00	1979.00	993.00
80	8185.00	5345.00	6296.00	5987.00	3487.00	2675.00	1678.00
100	9077.00	8563.00	7890.00	5687.00	4567.00	3923.00	2345.00
120	9114.00	8231.00	7436.00	6976.00	6023.00	5043.00	3946.00
140	11471.00	10567.00	9840.00	8887.00	8234.00	4689.00	2987.00
160	17308.00	15456.00	12458.00	10786.00	8976.00	6976.00	5231.00
180	19000.00	16889.00	14239.00	11985.00	10342.00	9843.00	7869.00
200	21107.00	18987.00	15768.00	13875.00	12567.00	10264.00	9864.00

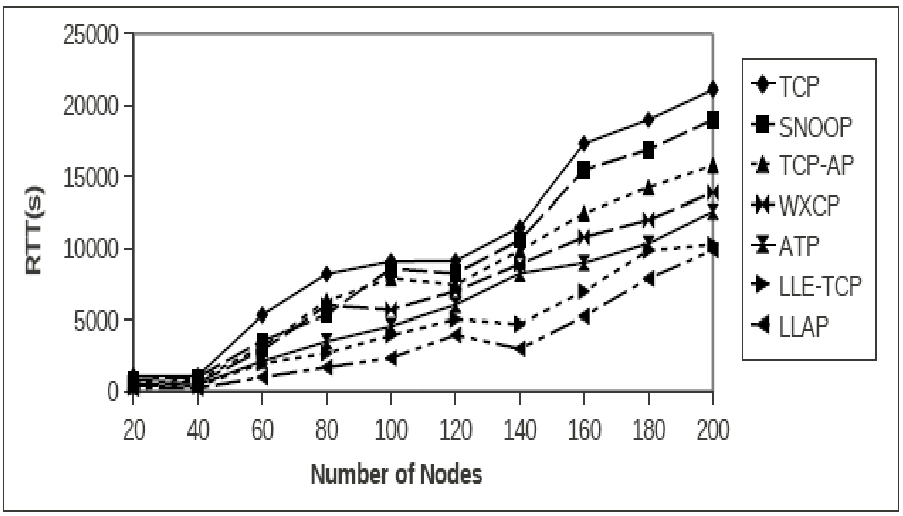

Figure 4-5: Round Trip Time versus network size

The bursty traffic by TCP increases the medium contention and the packet loss in the wireless network, thereby affecting the performance of TCP.TCP variants differentiate between the congestion and non-congestion lost, thus, do not waste time invoking congestion mechanisms where there is no congestion. A TCP variant, therefore, attempts to apply a solution specific to a particular problem. ENTs experienced less RTT and more throughputs compared to TCP and TCP variants. The fact that ATP (antithesis of TCP) has no retransmission timeouts and it does decouple congestion control and reliability makes the performance of the ENTs to be better than TCP and TCP variants. The TCP variants with feedback mechanism to detect congestion and differentiating data losses have far less throughput than ENTs because feedback notifications may be lost and mechanisms of differentiations may not be accurate for all types of networks. Both (ATP and WXCP) ENTs experienced almost the same RTT as they are interchanging in all angles.

They work interchanged due to the fact that they TLPs both utilize the ratio based transmission system. Similarly to experiment 4 (number of retransmitted packets against the network size), Hybrid TLPs outperform all TLP categories studied in this experiment. Hybrid suppresses TCP ACK to reduce RTT by the time required for TCP ACK transmission over wireless link. Another reason for Hybrid category to produce far lesser RTT as depicted in Figure 4-5 is the ability of LLAP and LLE-TCP to allow faster reaction to packet losses.

4.3.2. Performance Metrics versus Number of Flows

The purpose of recording the different number of flows in this study is to find out the TLP that can monitor network congestion when the number of flows increases. Reducing the number of flows can significantly increase the levels of service that the WMN can provide to applications. Therefore, consideration of the number of flows enabled us to determine the TLP that can increase the level of service provisioning, even if the number of flows increased. Continuous and event-driven flows should be supported in TLPs applicable to WMNs (Iyer *et al*, 2005). Therefore, we also considered the heterogeneous number of flows to find out the TLPs that are applicable to WMNs and supporting continuation and event-driven flows. This section describes and analyses the results of Experiment 6, 7 and 8. In these experiments we considered throughput, delay and packet delivery ratio against the number of flows. The data, results and results analysis of these experiments can be found in subsection a, b and c, respectively.

a. Experiment 6: Effects of the number of flows on the Throughput

The purpose of this experiment was to determine the degree of successful data packets transmitted over the WMNs communication channel using the ten selected TLPs against number of flows. As the TLP is one of the mechanisms to provide QoS in the multi-hop wireless network (WMNs), therefore, we considered throughput when we evaluate the performance of TLPs applicable to WMNs because throughput is the QoS parameter (Naeem *et al*, 2010), (Navaratnam *et al*, 2008), (Linn *et al*, 2008).

Experiment 2 presents throughput behaviour of TLPs applicable to WMNs against the network size, Table 4-7 depicts the data results for throughput against the number of flows for all selected TLPs applicable to WMNs. Figure 4-6 shows the graphical representation of the data results in Table 4-7. Similarly to Experiment 2 Experiment 6 considers throughput against number of flows. Traditional TLPs produced poor throughput than all TLP categories since Traditional TLPs are designed for wired networks not for WMNs.

Table 4-7: Results for the throughput per number of flows

NF	TCP	UDP	SNOOP	TCP-AP	HERC	LATP	WXCP	ATP	LLE-TCP	LLAP
20	77.33	64.36	171.39	184.00	116.83	154.17	163.40	181.12	199.34	212.23
40	84.18	59.56	166.41	178.51	101.25	164.38	166.29	187.23	192.23	209.96
60	69.52	57.34	162.29	166.35	105.62	141.44	156.98	197.34	232.30	242.24
80	79.66	48.00	166.79	164.13	99.09	159.94	180.87	203.12	243.56	246.67
100	54.06	53.56	196.43	188.49	100.58	153.29	186.33	212.23	227.68	234.34
120	59.43	55.56	168.60	176.98	95.06	160.03	174.93	186.35	241.33	221.12
140	61.69	47.56	163.10	136.52	96.64	142.65	159.01	178.23	229.22	196.66
160	60.19	50.46	183.70	177.05	96.89	151.92	166.78	179.99	217.56	198.55
180	58.19	53.44	165.78	180.17	96.64	137.23	156.72	178.56	213.44	189.57

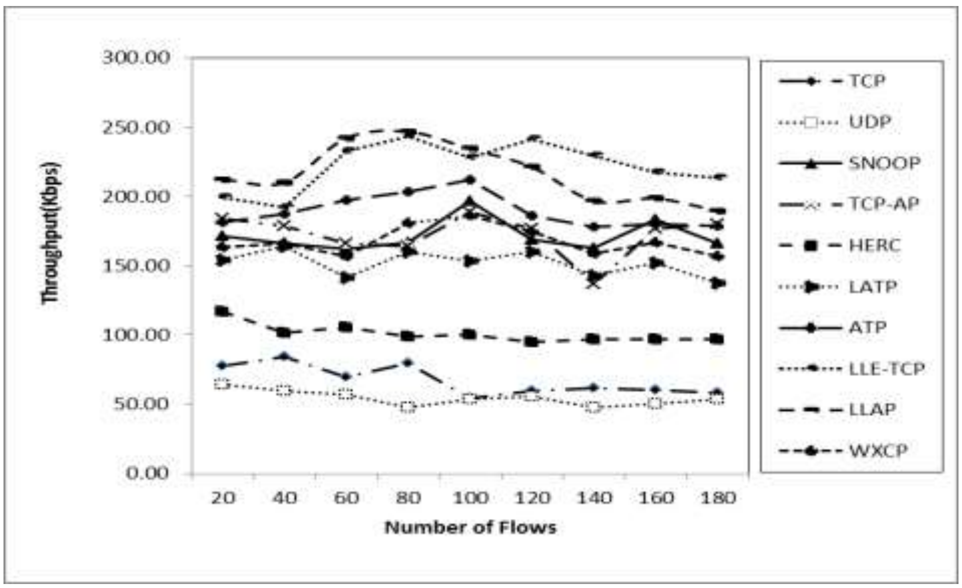

Figure 4-6: Throughput versus number of flows

The difference between Experiment 6 and Experiment 2 is that in Experiment 2 throughput for all TLPs decrease as the network size increases whereas in Experiment 6 the throughput increases slightly when the number of flows increases. Thus, the increase in number of flows does not have big impact on the level of throughput achieved by the TLPs. The throughput does not increase as the number of flows increase, since the increase in the number of flows also increase in the number of packet drops. The UDP variants produced more throughput than traditional TLPs due to the similar reasons mentioned in Experiment 2. As shown in Figure 4-6, UDP variants produce fewer throughputs than all TLP categories. The UDP variants end-to-end multi-metric joint detection approach designed to support real-time delivery for multimedia traffic in WMNs. The accuracy of the detection scheme is not sufficient for WMNs. Therefore, the performance of UDP variants degrades in WMNs. The increase in the number of flows causes the network to overload.

The network overload results in the increase of the number of the drop data packets. Thus, UDP variants drop more packets due to overloaded network as they do not control congestion. As shown in Figure 4-6 TCP variants produce a better throughput compared to Traditional TLPs, since traditional TLPs were mainly designed for wired networks and do not function properly in WMNs. TCP variant's reliability contributes to their better performance compared to UDP variants which are not reliable.

TCP-AP fares better throughout than Snoop as it adjusts the transmission ratio via considering many metrics such as medium contention, congestion window and spatial-reuse constraint. TCP variants produce fewer throughputs than ENT and Hybrid TLPs, since their performance limited by accuracy of contention estimation as well as the hardness of deriving the optimal window. The ENTs with the ability to decouple the congestion and reliability mechanisms, they have better throughput than traditional TLPs as well as UDP variants as shown in Figure 4-6.

TCP-AP (TCP variant) with pacing mechanism controlling load into the network is the only TCP variants that outperform ATP (ENT). ENTs produce better throughput than TCP variants since ENTs precisely estimate network congestion conditions for the WMNs. The estimations by ENT reduce the degree of network congestion. Congestion is the factor that contributes into poor network performance. Therefore, the reduction of network congestion increases the level of the throughput. As a result, when the number of flows increases, the performances of the ENTs decrease slightly. Similar to Experiment 5, Hybrid TLPs outperform all other TLPs categories.

The outstanding Hybrid TLPs perform congestion control by propagating the information to source without any additional control overheads. Optimal Hybrid TLPs reduces the round trip time (RTT) as well as medium busy time enabling the increase in the throughput. The LLAP performs better than LLE-TCP when the number of flow is 20 until 100, whereas LLE-TCP outperforms from 100 to 180 numbers of flows. The performance keeps changing since they both provide congestion control, but with different techniques. The common observation, for all TLP decrease very slightly when the number of flows increase.

b. Experiment 7: Effects of the flow size on the Delay

The purpose of this experiment was to determine the average time taken to successfully deliver the data packets from the source to an intended destination using our selected TLPs in WMNs. Packets delay may result in the packet losses and false indication of the network congestion (Reaz and Atiquzzaman, 2005). Therefore, delay was considered in this study to find which TLP experienced more packet losses over WMNs as the size of the traffic increases. Thus, we used delay to evaluate the performance of the TLPs applicable to WMNs find out which TLP can reduce packet loss and congestion in WMNs. On the other hand, delay is one of the QoS parameters (Naeem *et al*, 2010), (Navaratnam *et al*, 2008), (Linn *et al*, 2008), thus, it was considered in this study to indicate the TLP that yield better QoS when applicable to WMN with different traffic sizes.

We ran each number of flows with the same network size (200 nodes) four times and considered the average, starting from 20 to 180 numbers of flows. Table 4-8 presents the data results for delay against number of flows for all selected TLPs applicable to WMNs and Figure 4-7 shows the graphically representation of the network delay against the number of flows. Traditional TLPs produced more delay than all other TLPs considered in the study. Comparing the Traditional TLPs, UDP has less delay than TCP as UDP does not require any acknowledgements of data delivery.

TCP, therefore, experienced more delay especially when the data packets are retransmitted as results of packet loss before successfully delivered to their destination. TCP variants experienced more delay than all TLP categories but not traditional TLPs. Unlike UDP variants, TCP variants contain congestion mechanisms and attempt to ensure that data packet successfully reach the intended destination. Therefore, when the packet lost is detected, TCP variant retransmits the lost packet until the delivery note is received.

Table 4-8: Results for the delay per number of flows

NF	TCP	UDP	SNOOP	TCP-AP	HERC	LATP	WXCP	ATP	LLE-TCP	LLAP
20	686.30	624.02	446.96	432.42	120.40	294.54	380.59	318.66	253.12	173.77
40	700.86	670.72	438.77	491.16	122.07	297.45	392.95	309.90	261.90	175.41
60	848.09	735.45	533.10	571.03	111.81	297.44	409.92	319.87	242.45	197.55
80	806.45	756.45	526.97	570.57	118.22	299.57	410.61	329.57	259.34	198.23
100	837.67	781.21	580.55	558.33	156.08	354.36	437.45	312.35	269.45	208.45
120	823.57	785.74	510.66	625.22	149.85	389.79	433.12	354.87	298.12	196.99
140	961.77	839.04	584.66	662.43	151.99	377.56	463.92	360.88	316.65	229.98
160	904.99	879.99	589.66	630.94	154.49	381.46	479.80	387.56	342.68	295.43
180	942.67	893.41	621.07	692.67	156.99	389.34	474.58	392.57	352.12	298.26

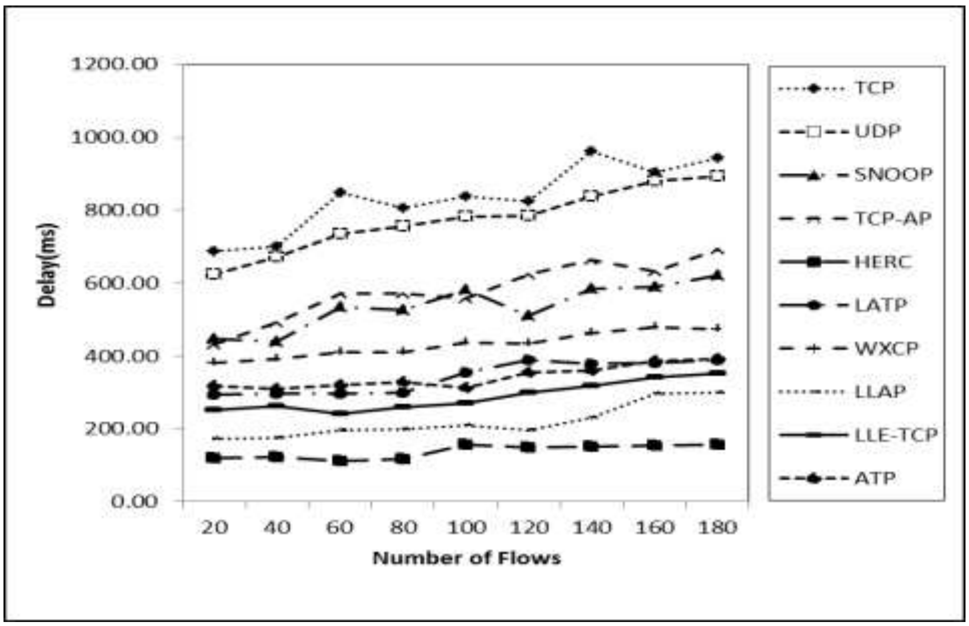

Figure 4-7: Delay versus number of flows

TCP variants require acknowledgements, therefore, experience more delay than UDP variants. UDP variants are more appropriate for sending limited amount of data per packet and suitable for low-latency applications. Therefore, UDP variants experienced by far lesser delay than TCP variants. The QoS aware LATP (UDP variant) found to be a suitable candidate to reduce delay in WMN environment compare to HERC, as this protocol provides an effort to perform end-to-end ratio symmetric control for multimedia streaming applications based on the degree of medium contention information received from the network. As depicted in Figure 4-7, ENTs received less delay than all TLP categories but not Hybrid TLP. The outstanding performance of the ENTs results from fact that they enable the sources to adjust the transmission ratio based on the received feedback and make precise estimation of congestion conditions. Thus, these TLPs also reduce the chances of the WMNs to be congested. Although ENTs reduce the network delay, but as the number of traffic connections grow up the level of network delay also increases.

Hybrid TLPs as a category with the ability to support both real time and non-real time applications produced less delay than all other categories. These TLPs reduce delay effectively through sending packets into the network with pacing delay (interval between adjacent packet transmissions) of FHD. Hybrid TLPs provide mechanisms to monitor the transmission of data packet from source to intended destination. In all categories studied delay against the number of flows, the experienced delay increase very slightly as the number of connected traffic increases.

c. Experiment 8: Effects of the flow size on the Packet Delivery Ratio

The purpose of this experiment was to determine the average ratio of successful data packets delivery (throughput) over different flow sizes. A 0% PDR represents the total failure of the network to deliver data packets whilst 100% PDR shows that the entire sent data packet was delivered successfully. The packet delivery ratio (PDR) was considered since it is the quality of service (QoS) parameter (Naeem *et al*, 2010), (Linn *et al,* 2008). TLPs are the means of QoS provisioning, therefore, to indicate the degree of QoS offered by TLP applied to with the increasing number of flows. Figure 4-8 shows the effect of flow size on the packet delivery ratio (PDR) of the selected TLPs applied to WMNs. The results for the PDR against the number of flows are shown in Table 4-9. Similar to Experiment 6 and Experiment7, we ran this experiment four times with the different number of flows with the same network size made up of 200 nodes, and then the average was considered.

Traditional TLP category drops more packets compare to all other categories, whereas Hybrid drops fewer packets compare to all other TLP categories. In the traditional TLP, UDP drops more data packets than TCP, since TCP provides more reliability and used acknowledgements to indicate successful data packets delivery. If the packet is lost TCP attempts to ensure that the packet reaches the destination through packet retransmissions. TCP variants successfully deliver more data than Traditional and UDP variants since they are able to distinguish between congestion and non-congestion losses meaning these TLPs only apply the congestion control mechanisms when the network is congested.

TCP variant such as TCP-AP sends packet at a pre-determine rate; therefore, reducing the potential of the network congestion which drives to the large number of packet lost. The UDP variants drop more packet TCP variants as they do not control network congestion and no packet retransmission takes place in these TLPs. The results graphically highlighted in Figure 4-8 determine that ENTs outperform all TLP categories but not the

Table 4-9: Results for the Packet delivery ratio per number of flows

NF	TCP	UDP	SNOOP	TCP-AP	HERC	LATP	WXCP	ATP	LLE-TCP	LLAP
20	54.48	46.32	71.04	74.26	61.79	67.92	91.17	89.23	94.01	92.70
40	46.06	42.99	75.81	71.58	62.99	67.03	91.20	89.18	94.19	93.08
60	52.62	43.95	69.99	79.01	60.38	70.03	90.05	88.04	93.79	91.90
80	51.60	45.91	70.96	74.26	57.90	65.73	89.00	86.57	93.66	90.92
100	43.99	41.26	70.31	78.56	54.68	64.72	87.77	85.46	92.57	90.82
120	45.41	39.05	68.43	76.03	55.46	59.49	88.32	86.06	92.12	90.38
140	46.85	41.54	63.23	71.48	53.35	59.02	87.46	85.23	92.54	90.51
160	49.51	40.80	64.76	70.87	52.34	56.72	87.18	86.48	92.68	90.19
180	46.61	36.66	64.54	69.48	51.68	54.69	87.80	86.21	92.95	90.74

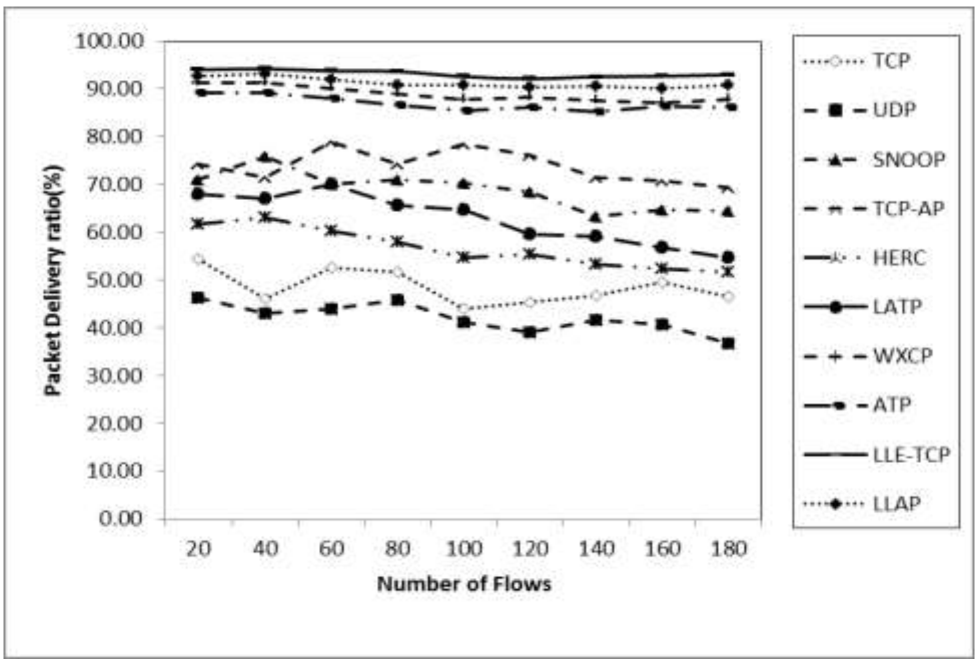

Figure 4-8: Packet delivery ratio versus number of flows

Hybrid. TLP. ENTs deliver more data packet successful than other categories, since the ENTs classical decouples congestion control and reliability mechanisms. This phenomenon enables the ENT TLPs to increase the number of data packets delivered successfully. The Hybrid TLPs supporting cross-layer as well as QoS aware approaches transmits more data packets successful compared to all TLP categories. The Hybrid reduces PDR by utilizing the link layer adaptive pacing mechanisms to control amount of data loaded into the network while support any type applications i.e. real-time and non-real time applications. Figure 4-8 depicts that for all TLPs, the level of PDR diminishes as the number of flows increase. The number of flows has less impact in the performance of TLPs applicable to WMNs compared to number of flows, as the performance of TLPs slightly affected by the increase in the number of flows unlike in number of nodes.

4.3.3. Performance Metrics versus Distance between nodes

In these experiments, we were interested in finding out the impact of distance between nodes over the performance of TLPs applicable to WMNs. We adopted the distance between the nodes from (Johnson and Lysko, 2008), (Johnson and Hancke, 2008) where they compare two routing metric in OLSR and where they compare MANET routing protocols, respectively. From the best of our knowledge, the distance between the nodes has not been considered when evaluating the performance of TLPs over the networks.

Therefore, we wanted to investigate if we can find the new trends for the TLPs over WMNs. This section presents Experiment 9, 10 and 11, where the performance evaluations of the ten TLPs based on the throughput, packet retransmission and packet drop ratio all versus various distance between the nodes. The full analysis of these experiments can be found in subsection *a, b, and c, respectively.*

a. Experiment 9: Effects of the distance between nodes on the throughput

The purpose of this experiment was to determine the average of successful data packets delivery over different distances between the nodes with our selected TLPs applied to WMNs. It was stated in Chapter One that for performance evaluation we considered the performance metrics for QoS. As throughput is one of the quality of service (QoS) parameters (Naeem *et al*, 2010), (Navaratnam *et al*, 2008), (Linn *et al*, 2008), therefore, it was considered in this study to indicate the TLP that offers more QoS when applied to WMNs with increasing distance between nodes.

The previous experiment (such Experiment 1 and Experiment 6) pertains throughput have been considered throughput against network size, number of flows and simulation time, respectively, but not distance between nodes. Table 4-10 depicts the data results for TLPs throughput behaviour against different distances between the nodes in WMNs. Figure 4-9 shows the effect of node distance on throughput of the TLPs applied to WMNs. Figure 4-9 shows that traditional TLPs have fewer throughputs than all TLP categories.

Traditional TLPs were originally designed for wired networks, as a result their performance degrades in WMNs. TCP is slightly better than UDP. However, TCP designed to provide reliability (it retransmits the data packet if the packet has failed to rich the destination) whereas UDP considers timeliness and is unreliable. As the distance between the nodes increases the performance of traditional TLPs degrades. When the distance between the nodes increases the chances of link failures also increases.

Therefore, more packet drops experienced by traditional TLPs as they assume that all packet losses are due to congestion (Akyildiz and Wang, 2002), (Allman *et al*, 1999). As a result, if the non-congestion losses occur, the network throughput quickly drops. The UDP variants (HERC and LATP) perform worse than all TLP categories but not traditional TLPs. Even though UDP variants performance is slightly better than traditional TLPs in the previous experiment such as Experiment 1, but in this experiment, the TCP outperformed LATP.

The poor throughputs produced by UDP variants are due to fact that their end-to-end multi-metric joint detection approaches still lack accuracy of detection (Fu *et al*, 2003). HERC outperforms TCP as it replaces the TCP response function which is yielding poor performance in wireless environment with high-speed equation-based ratio control. Figure 4-9 shows that TCP variants (Snoop and TCP-AP) achieve better throughput than traditional and UDP variants TLP categories but not ETN and Hybrid TLP categories.

The ability of TCP variants to differentiate between the congestion and non-congestion losses allows the better throughput produced by TCP variants. This differentiation avoids the TCP variants from invoking the congestion control while the pocket losses are due to route or link failure not congestion (Akyildiz *et al*, 1999). TCP-AP achieve a much better throughput with various distance between nodes as it this TLP adjusts the transmission ratio by considering multiple performance metrics such as congestion window, contention on the end-to-end path, and spatial-reuse constraint. Hybrid TLPs achieve far better throughput than all TLP categories but not Entirely New TLPs (ENT).

Table 4-10: Throughput per distance between nodes

DBN	TCP	UDP	SNOOP	TCP-AP	HERC	LATP	ATP	WXCP	LLE-TCP	LLAP
100	104.76	65.48	156.85	178.04	97.26	149.35	240.06	272.56	191.85	212.56
200	116.78	51.73	161.50	182.14	101.78	135.50	225.66	268.16	173.78	213.16
300	128.64	42.45	157.22	183.70	90.89	147.22	239.02	246.02	180.22	220.77
400	103.27	45.47	172.02	194.94	78.27	152.02	224.94	254.94	183.35	202.44
500	102.94	42.00	128.49	167.08	77.19	135.99	216.54	249.04	145.39	195.79
600	83.44	41.32	123.28	149.03	65.94	118.28	176.92	201.92	158.08	169.42
700	68.77	38.99	145.44	152.50	53.77	135.44	188.89	226.39	145.44	163.89
800	73.49	38.70	140.08	170.52	73.49	110.08	205.22	215.22	169.17	184.72
900	70.49	33.64	146.41	185.32	65.49	113.34	215.66	220.66	162.02	199.41
100	67.26	36.66	120.64	154.45	59.76	104.39	184.91	192.41	143.85	194.91

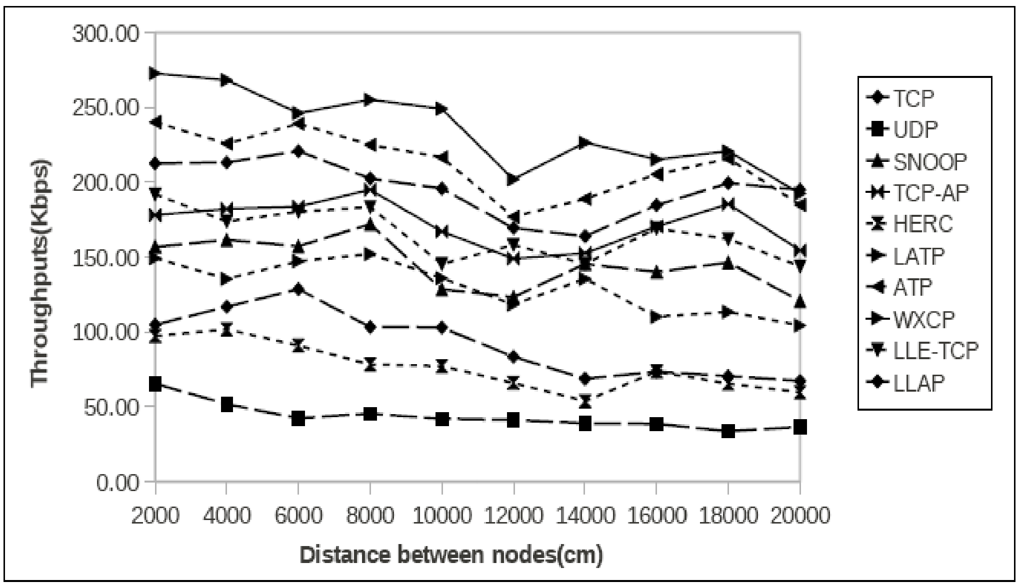

Figure 4-9: Throughput versus distance between nodes

On the other hand, TCP-AP (TCP variant) outperforms LLE-TCP. As shown in Figure 4-9, TCP-AP has a hybrid adaptive-pacing mechanism of sender ratio control and congestion control. TCP-AP uses the TCP end-to-end semantics which are not impacted and lower layer protocols such as routing and MAC require no change. A good throughput produced by Hybrids result from the fact that they can work efficiently with both UDP and TCP traffic. Hybrid TLPs produce better throughput, but the disadvantage of these TLPs, they deal with MAC contention and congestion whereas most of data packets are lost due link failures or route failures in a wireless network with long distance between nodes (Xylomenos *et al*, 2001). As show in Figure 4-9, ENTs have more throughput than other TLP categories. In the ENTs, congestion detection is a delay-based approach, and, therefore, there is no ambiguity between the losses which are due to congestion and non-congestion.

ENTs separate reliability and congestion control mechanisms, i.e. ENTs invoke the mechanism related to the current problem experienced by TLPs WMNs.In addition, loss discovery and pacing mechanisms are introduced at the both sender and receiver to get rid of the tiny window and burst problem. The WXCP achieve better throughput than ATP (total incompatible with TCP) due to the fact that WXCP utilizes explicit ratio feedback instead probing an available bandwidth, WXCP flows are able to converge to transmission state where better throughput is achieved.

b. Experiment 10: Effects of the distance between nodes on Packet retransmissions

The main purpose for this experiment is to determine how the number of packet retransmission (PR) affects the performance of the TLPs applicable to WMNs. Does the number of PR increase or decrease if the distance between nodes increases? This experiment assists to investigate whether we need to advance the congestion control mechanisms that are currently used to reduce PR or not. Since TCP retransmissions should only occur when it is certain that a packet to be retransmitted was actually lost

(Chen *et al*, 2001). The congestion control mechanisms assist in reducing the number of packet retransmissions (Iyer *et al*, 2005), thus, the number of PR is considered to indicate the degree of network congestion experienced by WMNs when the selected TLPs applied to it. On the other hand, PR was considered in the study since it indicates how reliable the TLP applicable to WMNs is. Table 4-11 shows the data results which indicate the number of retransmitted packets on different distances between nodes. Figure 4-10 depicts how many packets can be retransmitted by the TLPs applicable to WMNs when the distance between the nodes changes.

Similar to Experiment 4, UDP and UDP variants are not considered in this experiment since they do not contain packet retransmission mechanisms. Therefore, out of ten TLPs only seven TLPs were considered in this experiment. Results highlighted in Figure 4-10 show that traditional TLP (TCP) has a larger number of retransmitted packets than all other TLP categories, i.e. TCP experienced more data packet losses compare to other TLPs when applicable to WMNs because the number of packet retransmissions indicates the loss level of the transmitted packets.

There is a large number of packet drops in TCP due to the fact that TCP always invoke congestion control mechanisms even if the packets loss is not due to congestion more especially in wireless environments. Therefore, packet retransmission is usually used to insist network reliability. The number of packet retransmissions increase as the distance between the nodes increases. TCP variants have less number of retransmitted packets compare to classical traditional TCP, but TCP variants also experienced the same problem

Table 4-11: Packet retransmission per distance between nodes

DBN	TCP	SNOOP	TCP-AP	ATP	WXCP	LLE-TCP	LLAP
100	785.98	806.22	457.46	302.98	187.29	502.29	572.29
200	795.55	811.35	427.80	385.55	219.60	559.60	634.60
300	1009.43	883.60	478.57	432.18	220.14	607.64	682.64
400	1018.33	868.60	524.92	448.33	223.27	680.77	658.27
500	1017.95	919.71	690.25	430.45	201.27	646.27	743.77
600	1180.08	1031.17	835.99	450.08	178.68	625.43	680.68
700	1046.87	916.33	809.74	401.87	205.14	680.14	730.14
800	1033.35	890.95	715.17	428.35	229.38	546.88	621.88
900	1036.30	857.27	707.41	406.73	169.79	606.29	656.29
1000	1041.52	871.15	776.99	527.27	222.50	640.00	702.50

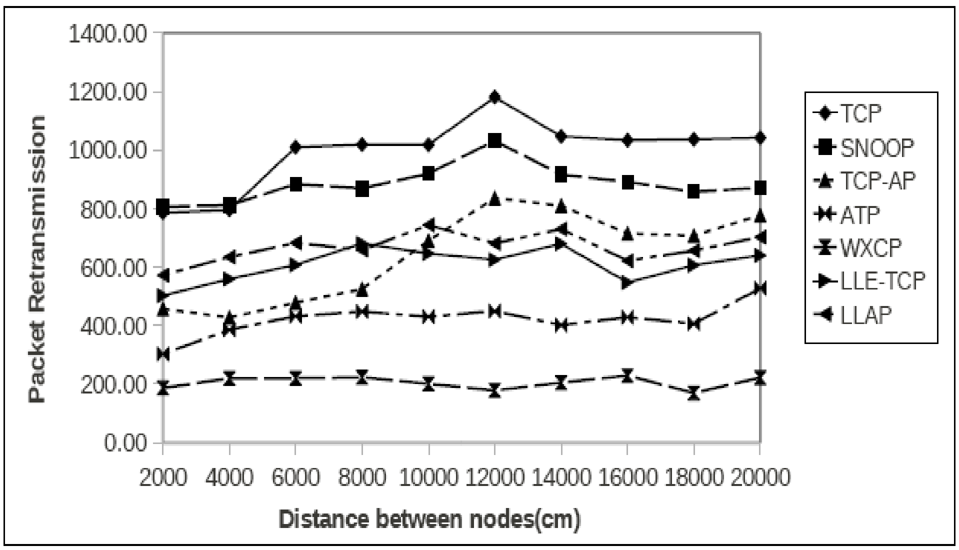

Figure 4-10: Packet retransmission versus distance between

of producing more packet retransmission as the distance between nodes increases. A much better performance yield by TCP variants, results from the fact that TCP variants differentiate between congestion and non-congestion losses. Therefore, TCP variants do not issue congestion control mechanisms in wireless environment even if packets are lost due to a few errors and link or route failures but not network congestion. Figure 4-10 depicts that TCP-AP outperforms Snoop protocol as well as one of Hybrid TLPs, but only outperforms Hybrid TLPs when the distance between nodes is small (i.e. from 100 to 480 mm). TCP-AP experiences lesser number of retransmitted packets as they use an adaptive ratio based scheme to monitor the amount of data packets offered into network and calculates optimal TCP window for wireless networks. The performance of the Hybrid TLPs is better than TCP-AP, when the distance between nodes is bigger because the TCP variant's performance can be limited by the lack of accuracy. For example, notification may be lost, differentiations mechanism may not be accurate and hardness of deriving an optimal window (Fu *et al*, 2003), (de Oliveira and Braun, 2005), (ElRakabawy *et al*, 2005).

Hybrid TLP produced less number of retransmitted packets than other TLP categories but not ENTs. The reduction of packet retransmission when Hybrid TLPs are applied to WMNs is provided by link layer adaptive pacing scheme. This pacing scheme adaptively controls the amount of data offered into network, thus, the number of packets loss reduced as a results, the number of packets retransmission reduced as well. On the other hand, Hybrid TLPs use its advantage of being cross-layer aware to improve the performance by interacting with higher layer protocols without modifying them.

The LLE-TCP retransmits less data packets than LLAP but, as LLE-TCP enhances protocol stacks with cross-layer ARQ agents that support ACK suppression, therefore, LLE-TCP has less number of retransmitted packets than TCP and its variants. The drawback of LLE-TCP is that its performance improvement depends on the transmitted datagram size. Similar to TCP variants, Hybrid TLPs retransmit more and more packets as the distance between the nodes increases. As depicted in Figure 4-10, ENTs have better mechanisms to minimize the number of packet retransmission than other TLP categories when applicable to WMNs. Multiple mechanisms are integrated in ENTs applicable to WMNs in order to make accurate estimation of network congestion circumstances. Therefore, better throughput and minimum number of packet retransmissions are achieved in the wireless multi-hop networks. On the other hand, the congestion control and reliability mechanisms are separated, therefore, congestion control mechanisms are not applied when packet losses detected in wireless environment are not related to congestion. In ENTs, the number of packet retransmissions increase slightly as the distance increases between the nodes.

c. Experiment 11: Effects of the distance between nodes on the Packet delivery ratio

The purpose of this Experiment was to determine the WMN's level of delivering data packets being sent when our ten selected TLPs are utilized during the data packets transmission. A 100% PDR represents all data packet sent in the network delivered successful and whilst 0% PDR indicates that the entire sent data packet was dropped. The packet delivery ratio (PDR) is the quality of service (QoS) parameter (Naeem *et al*, 2010), (Premalatha and Balasubramanie, 2010) (Navaratnam *et al*, 2008), (Linn *et al*, 2008), therefore, PDR was considered in our study and it indicates the TLP that alleviates the QoS challenges over WMNs made up of nodes with different distances between them.

Table 4-12 shows the data results which indicate the PDR of TLPs when applicable to WMNs made up of nodes with different distances between them. Figure 4-11 highlights the ratio at which the data packets are delivered when applied to WMNs. Figure 4-11 shows that traditional (UDP) TLPs deliver fewer packets than TCP (traditional) and all TLP categories. UDP TLP experienced more packet losses when applied to WMNs since it was not meant to be used in wireless environment. On the other hand, UDP does not have mechanisms to retransmit the packets that have failed to reach destination. UDP has lower-level of PDR as it is usually a tolerant of packet losses but delay-sensitive. TCP deliver more packets than UDP and UDP variants, since TCP is delay resilient, and imposing reliability by retransmitting the lost packets until they reach their intended destinations.

More packet losses take place as the distance between nodes increase. The packet loss tolerant UDP variants drop more packet than other TLP categories but not traditional (UDP). The better performance achieved by UDP variants compared to UDP, results from the rate control and link adaptive mechanisms provided by UDP variants. HERC with high-speed equation-based rate control for multimedia streaming applications increases the packet delivery ratio. LATP with a link adaptive mechanism outperforms HERC when there are applied to WMNs.

Figure 4-11 depicts that TCP variants deliver more packets than traditional and UDP variants. The reduction of packet loss by TCP variants results from the fact that in TCP variants congestion and packet loss due to network blackout (such as route or link failures) are controlled separately (Chandran *et al*, 2001). Several mechanisms such as explicit link failure, feedback-based scheme, out-of-order detection and etc have been designed for TCP variants to improve TCP performance in wireless environments.

Table 4-12: Packet drop ratio per distance between nodes

DBN	TCP	UDP	SNOOP	TCP-AP	LATP	HERC	WXCP	ATP	LLE-TCP	LLAP
2000	25.72	13.33	37.17	53.22	26.71	19.02	88.25	70.75	45.82	58.12
4000	28.25	14.97	36.86	53.95	20.18	15.27	87.75	67.00	42.22	57.89
6000	30.10	12.16	35.38	56.76	23.71	17.68	86.50	68.50	43.27	66.40
8000	27.89	13.46	34.79	56.34	20.22	16.42	89.75	72.11	44.24	59.04
10000	29.97	10.44	41.45	50.10	19.34	14.01	88.00	66.25	43.57	56.13
12000	26.12	15.61	35.87	48.99	15.16	17.12	85.50	65.45	43.01	50.02
14000	31.79	12.25	35.35	47.89	19.64	16.00	82.25	64.25	42.15	55.29
16000	24.67	10.23	34.92	47.09	28.50	15.55	83.25	59.25	43.21	52.60
18000	23.77	14.00	33.11	48.12	15.00	14.00	70.23	63.50	40.12	52.21
20000	20.35	9.54	39.52	48.00	14.00	10.78	62.56	61.00	39.79	51.00

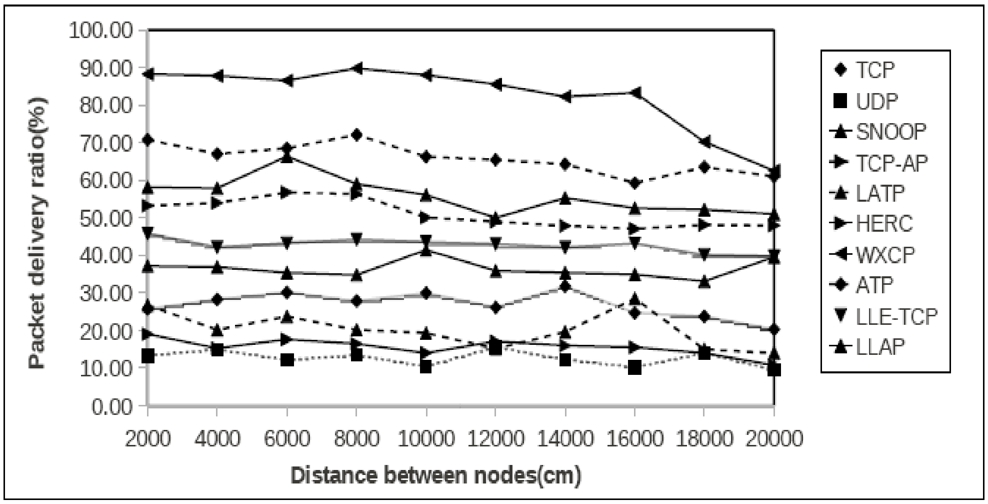

Figure 4-11: Packet delivery ratio versus distance between nodes

These mechanisms make the TCP variants to reduce the number of packet drops compared to traditional and UDP variants. TCP-AP outperforms Snoop and the LLE-TCP (Hybrid TLP). The reason for TCP-AP to experience maximize packet delivery more than Snoop and LLE-TCP is, the mechanisms of rate control is effective and the estimation of contention and spatial-reuse constraint is accurate. Hybrid TLPs experienced a maximum number of packet delivery compare to traditional TLPs and UDP variants. One of the Hybrid (LLAP) outperforms all TLP categories but not ENTs.

LLAP reduces the number of packet losses using the link layer adaptive pacing mechanism which adaptively manages the amount of traffic offered into the network. On the other hand, LLAP mechanisms estimates the four hop transmission delay in the network path without incurring any additional overhead and accordingly paces the packet transmissions to reduce MAC contention in the wireless network. As a cross-layer aware and QoS aware TLPs, Hybrid TLPs reduce the number of the packet drops in the network with different distances between nodes.

There is few packets delivery experienced by WMNs when Hybrid TLPs are applied to the network as the distance between nodes increases. Figure 4-11 shows that ENTs experienced more number of packets delivered than all TLP categories when applied to WMNs. In ENTs transmissions are rate-based, and quick-start mechanism is used for initial rate estimation. The number of packet losses is reduced in ENTs because their congestion detection mechanism is delay-based; as a result, the confusion between the congestion losses and non-congestion is avoided. In addition, there is no retransmission timeout, and congestion control and reliability mechanisms are decoupled.

4.3.4. Performance Metric versus simulation time

This section presents the Experiment 12 in which the performance evaluations of the ten TLPs based on the throughput versus various simulation times. The full analysis of this experiment can be found in *subsection a* below. Different simulation time is considered to investigate if time has an impact on the performance of TLPs applicable to WMNs.

a. Experiment 12: Effects of simulation time on the throughout

The purpose of this experiment was to determine the average rate of successful data packets delivery over different simulation times with the selected TLPs applicable to WMNs. Throughput is considered as it signifies the degree of QoS offered by TLP(QoS mechanism) over WMNs. This phenomenon is based on the fact that throughput is the QoS parameter (Naeem *et al*, 2010), (Navaratnam *et al*, 2008), (Linn *et al*, 2008). The previous experiment (such Experiment 1 and Experiment 5) pertains throughput have been considered throughput against network size and number of flows, respectively, but not simulation time.

Table 4-13 depicts the data results for TLPs throughput behavior against different simulation times in WMNs. Figure 4-12 shows the effect of time on throughput of the TLPs applied to WMNs. In Traditional TLPs, TCP achieved far better throughput than UDP and UDP variants. The TCP is resilient to delay, but demands reliability and has congestion control mechanisms as results, the level of TCP throughput is better than that of UDP and UDP variants as they are usually tolerant to packet drops but they are delay-sensitive and not have congestion mechanisms.

Actually, Figure 4-12 shows that UDP and its variants produce less throughput compare to all TLPs as they are not reliable (i.e. they do not retransmitted any lost packets). UDP TLPs only assume that they have performed their duty correctly if the data packets reach destination in real time, without considering the order in which the data packets delivered.

Table 4-13: Throughput per network simulation time

Time	TCP	UDP	SNOOP	TCP-AP	HERC	LATP	WXCP	ATP	LLE-TCP	LLAP
100	199.88	93.04	197.61	180.69	111.34	118.15	197.61	201.54	210.56	212.15
200	206.67	124.84	199.06	194.63	126.34	132.67	197.61	212.79	226.76	224.98
300	208.15	137.96	206.39	194.29	136.32	148.21	211.89	218.65	218.91	209.56
400	205.44	144.34	217.55	194.79	140.89	162.92	217.19	228.78	239.63	234.64
500	208.15	148.32	217.08	196.87	149.78	155.68	219.93	223.31	243.82	235.76
600	207.17	150.89	220.70	198.18	157.58	163.66	224.05	231.81	242.88	242.35
700	209.23	152.85	226.23	195.95	152.98	158.53	224.10	236.14	248.98	236.88
800	210.09	154.17	222.32	196.87	173.83	178.12	221.16	233.97	254.45	249.99
900	210.73	155.42	224.43	198.61	163.46	176.89	221.86	236.99	259.86	253.82
1000	210.43	156.27	228.02	198.72	182.95	189.88	221.15	232.62	261.90	243.86

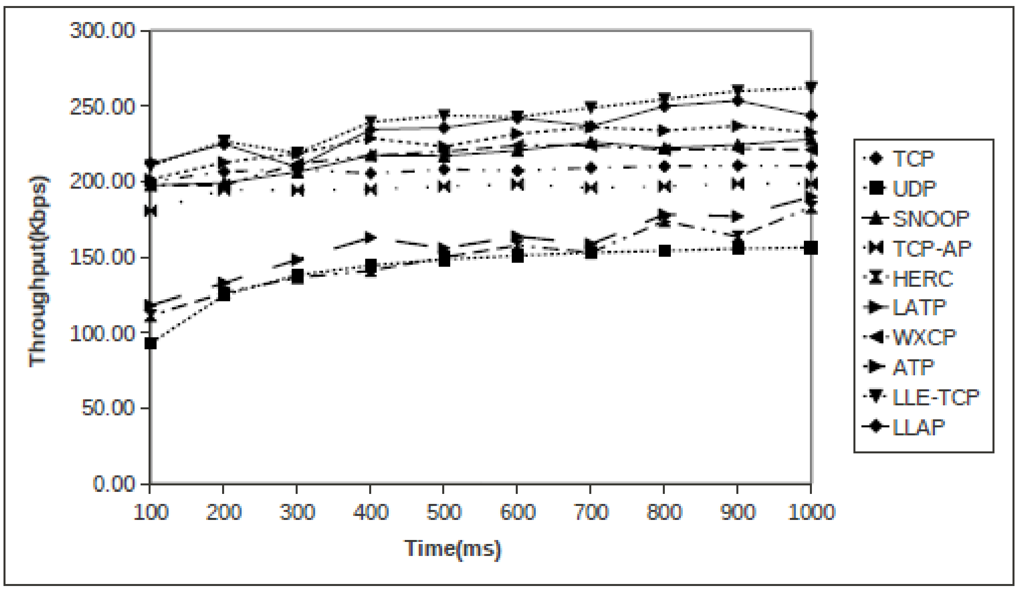

Figure 4-12: Throughput versus simulation time

LATP with link adaptive mechanisms performs better than both UDP and HERC. TCP variants (Snoop and TCP-AP) have more throughput than UDP and UDP variants while they produce fewer throughputs than ENT and Hybrid TLPs. TCP-AP performance is almost constant throughout. Snoop with network input rate control mechanisms achieve better throughput than TCP and TCP-AP. The performances of Snoop slightly improve as the simulation time increases. ENT (ATP) outperforms traditional, UDP variants and TCP variants but not Hybrid TLPs. The better throughput achieved by ENTs is due to their rate-based approach, and quick-start they used for initial ratio estimation. On the other hand, this ENT TLPs have entirely new set of mechanisms for reliable data transfer. WXCP increases throughput slowly as the simulation time increases. The ATP yield the good throughput as it does not consist of retransmission timeout, and congestion control and reliability are decoupled. Hybrid TLPs produce by far better network throughput than all other TLP categories. These TLPs produce the same throughput from 100 to 200s, but LLE-TCP outperforms LLAP at 200s to 600s.

Their performance increases as the time increases. The results indicate that Hybrid TLPs performance is optimal compared to other TLPs categories when applicable to WMNs. The throughput achieved with various network sizes decreases as the network size increase. But the throughput achieved in various simulation times increase as the simulation times increase. This is because an increase in time gives the packet chances to reaching the destination even if the original one was lost or delayed, but through recovery mechanisms such as retransmission, slow start, and fast recovery the packet reaches the destination.

4.4. Results Summary

Table 4-14 summarizes the performance of network TLPs with regard to the various performance metrics that we have utilized in our experiments. Numbers are used in rating the performance of TLPs over wireless multi-hop networks ranging from '**1**' to '**10**', with "**1**" representing the best performance transport layer protocol and '**10**' representing worst performance transport layer protocol. The full meanings of performance metrics used to evaluate the performance of the TLPs in Table 4-14 are as follow: ***T1*** = throughput against network size, ***T2*** = throughput against number of flows, ***T3*** = throughput versus distance between nodes, ***T4*** =throughput against simulation time, ***D1*** = delay against the network size, ***D2*** = delay against number of flow, ***PD1*** = packet delivery ratio against network size, ***PD2*** = packet delivery ratio against number of flow, ***PD3*** = packet delivery ratio versus distance between nodes, ***RP*** = retransmitted packets versus network size, ***RP2*** = retransmitted packets versus distance between nodes , and ***RTT*** = round trip time against network size.

Table 4-14 assists to easily determine an optimal performing TLP among the category. The experiments reported in subsection 4.4.1 until 4.4.9, as well as the concise summary provided in Table 4-14 enabled us to identify an optimal performing TLP. The results show that an optimal performing is LLAP followed by another hybrid TLP LLE-TCP. These TLPs show the optimal performance for both UDP and TCP traffic in different network scenario scenarios. In our study we wanted to identify an optimal transport layer protocol among the four TLP categories.

In the process of identifying the optimal TLP we also managed to identify an optimal performing TLP category among the five considered TLP categories for our experiments. An optimal category is the Hybrid category since the first and the second best TLP are both from this category. The second best performing category is the ENT category, although it does not outperform the other categories in all performance metrics.

Table 4-14: Performance Summary

GROUP	TLPs	T1	T2	T3	T4	D1	D2	PR1	PR2	PD1	PD2	PD3	RTT
Hybrid TLPs	LLAP	1	2	3	2	1	1	1	5	1	2	3	1
	LLE-TCP	2	1	5	1	2	2	2	4	2	1	5	2
Entirely New TLPs	WXCP	3	4	1	5	4	4	4	1	4	3	1	4
	ATP	4	3	2	3	3	3	3	2	3	4	2	3
TCP Variants	SNOOP	5	5	6	4	6	7	6	6	6	6	6	6
	TCP-AP	6	6	4	7	5	8	5	3	5	5	4	5
UDP Variants	LATP	7	7	7	8	7	6	-	-	8	7	8	-
	HERC	8	8	9	9	8	5	-	-	9	8	9	-
Traditional TLPs	UDP	9	10	10	10	9	9	-	-	10	10	10	-
	TCP	10	9	8	6	10	10	7	7	7	9	7	7

An Entirely New category of TLPs is followed by the TCP variant category. This tells us that the worse performing category according to our findings is traditional TLP category. This outcomes indicated clear that the application independent TLP (hybrid TLPs) should be considered first when someone intends to develop a new TLP.

4.5. Simulator and Experiment Limitations

Simulation experiments are not at the best approximation of the real network scenario. Therefore, there are bound to be assumptions made in an effort to model the environment being considered. This section highlights the assumptions made; any limitations on the experiments conducted as well any inherent limitations of the simulation tool that was utilized. It is possible that one or more of the assumptions made and the limitations of the experiments and simulation tool could have affected the results presented. The assumptions and limitations are:

i. The nodes in the network were static.

ii. The IEEE 802.11 RTS/CTS mechanism was disabled.

iii. We managed to evaluate the performance using few performance metrics and varied them with few parameters, while there many parameters that could be considered for performance evaluation.

iv. Lack of realistic Application Layer modeling. A constant bit ratio model was used whereas realistic Application Layer traffic resembles a variables bit ratio traffic stream,

v. The terrain was assumed to be flat without obstacles, realistic terrain models consider the elevation of the nodes as well objects such as walls, poles and buildings.

4.6. Main Conclusions and Recommendations

i. T*he use of congestion control algorithms*: From our results, it is clear that TLPs with an advanced congestion control mechanism (i.e. the one that differentiate between congestion and non-congestion losses) yield excellent throughput, less delay and less packet drop ratio. Congestion control mechanisms reduce number of packet lost and packet retransmissions. The results show that TCP had more retransmissions because it cannot distinguish between the congestion loss and non-congestion loss. We, conclude that sophisticated congestion mechanisms are crucial requirements for optimal TLP performance. There are several congestion control and avoidance mechanisms or schemes that have been designed already for different types of networks and from different perspective such as cross-layer, QoS aware of recent, but still to the best of our knowledge, there is no congestion control algorithm designed to cater for all this congestion schemes combined into one which can be applied to any network scenario to control and avoid network congestion. Therefore, the study to combine several congestion into one (One fits all congestion control scheme) scheme is important.

ii. *TLPs should be cross-layer aware*: From the results given above all the cross-layer aware TLPs outperform all the TLPs, which are not cross-layer aware. This is because transmission of data does not involve TLPs only. The process includes many other protocols e.g. routing protocol which are found in other layers. The cross-layer aware TLPs (such as LLE-TCP and LLAP) outperform the non cross-layer aware TLPs in most of the aspects that were considered in this research. Although cross-layer aware TLPs perform better based on the study, but cooperative Load and Traffic Managing process is needed. Thus cross-layer optimization (CLO) can be used to provide cooperative management, where Load and Traffic managing processes can be made possible using interface between cross-layer optimization entity and the management in the application, transport and network layer stacks.

iii. *Protocol should be application independent*: Hybrid TLPs outperformed all other TLP categories, at in most cases. These transport layer protocols work well with both real time and non-real time applications. This gives an impression that a good TLP is the one which is applicable to both real-time and non-real-time applications.

iv. *Quality of Service (QoS) aware*: taking from the results we highlighted above, QoS is still an issue. There are TLPs which are QoS aware, but their performance still needs to be improved especially TCP and UDP variants. This is due to problems such as medium contention, temporary link failure, frequently route changes, and, etc. The Hybrid TLPs have better degree of user satisfaction.

The problem with TLPs as far as QoS is concern is that, they do not synchronize QoS loading at each layer i.e. mating cross-layer and QoS provisioning processes. As load shifts or reallocates, increase or decrease, and when problems occurs, some adjustments are important due to the fact that the process such network delay statistics, existing source-receiver (transmission) relationship and the statistics regarding allocating source to receiver may require some QoS aware adjustments.

v. *Scalability*: The performance of all TLPs degrades as the network size and number of flows increase. Even the Hybrid TLPs which were found to be the best also experienced the same problem. All these conditions confirm that the issue of network scalability needs attention. The fact we have identified that network scalability requires attention does not mean scalable network does not have any problem. In a large geographical area, thus the long distance can incur some network delays while trying to provide network scalability. Therefore, delay intolerant applications can be affected even if the network is scalable. As results we recommend anyone who would be interested to inflict a scalable TLP should make it a point that it is an *anti-delay scalable TLP* applicable to WMNs.

vi. *TLPs should be distance tolerant*: The performance of all TLPs mortifies as the distance between the nodes increases. The Hybrid TLPs which performed better in many cases but they are even outperformed by ENTs and one of the TCP variants (i.e. TCP-AP) when throughput, packet retransmission and packet delivery ratio versus distance between nodes were concerned.

4.7. Summary

In this chapter we have presented the simulation results of the selected TLPs, comparing their performance when they are applied to WMNs. Experiments on the *throughput* against network size, number of flows and simulation times, *delay* against network size and number of flows, *packet delivery ratio* against network size and number of flows, *packet retransmission* against network size and *RTT* against network were done. Table 4-14 depicts the ratings of the TLPs performance which enabled us to simply identify an optimal performing TLP. Section 4.5, presents the simulation and experimental limitations. The main conclusions have been drawn and recommendations have been made from the experiments are given in Section 4.7. Chapters Five makes the overall conclusion of this study and the future work.

CHAPTER FIVE

CONCLUSION AND FUTURE WORK

5.1. Conclusion

This study presents the analysis of the work done to determine the optimal TLPs within the five TLP categories applicable to wireless WMNs. This analysis was carried out, through comparing the performance of the TLPs selected from literature. TLPs were categorized into four categories namely: 1) TCP variants, 2) UDP variants, 3) Hybrid and 4) Entirely New. Traditional (TCP and UDP) TLP category was included as the basis of our study, since most of the existing TLPs were derived from them and they are the main and the first TLPs that are widely used in real life scenario.

The main goal of the study was to compare existing TLPs and make recommendations for ideal TLP applicable to WMNs. To achieve the goal we came up with four objectives. The first one was to develop the framework for the analysis of the related work. The literature review analysis framework was developed and presented in Chapter Two. The second objective was to use the framework developed in objective one to select the TLPs for evaluations.

Table 2-26 gives the selected TLPs for evaluations and the fully analysis of these TLP is presented in Chapter Three. On the other hand, the trends for TLPs applicable to WMNs developed based on the analysis were also used to make things easier in the selection of the TLP for evaluation. These trends are graphically presented in Figure 2.1. The third objective was to simulate and evaluate the selected TLPs. We simulated and evaluated the performance of TLP over WMNs using NS2 in Chapter Four. The psuedocodes and flowcharts followed to implement the TLPs in NS2 were developed and presented in Chapter Three. Performance evaluations enabled us to compare the performance of the selected TLPs to identify the optimal performing TLP over WMNs, since it was not clear which TLP is most suitable for WMNs.

We managed to determine a TLP with an optimal performance when applied to WMNs based on the simulations and evaluations of the selected TLPs. According to our findings reported in Chapter Four, LLAP is the one that performs better than the other nine TLPs evaluated in this study. On the other hand, an optimal performing TLP category was identified while investigating our main focus was on the optimal TLP. We found the Hybrid category to be the optimal performing category with the first two TLPs regarded as most optimal coming from this category.

The fourth objective was to recommend the design criteria or features an ideal TLP should have. Based on the results reported in Chapter Four we were able to make recommendations about what should be the design criteria for ideal TLP applicable to WMNs in section 4.7. From this work we have drawn the following conclusions and these conclusions are the basis of our design criteria:

1. Sophisticated congestion control mechanisms are needed for an optimal performing TLP applicable to WMNS.
2. Cross-layer optimization can enable the cross-layer aware TLP to be more optimal in

WMNs.

3. TLP should be application-independent

4. QoS aware TLPs should synchronise read QoS load at each layer, mating cross-layer and QoS provisioning process.

5. The TLP applicable to WMNs should be an anti-delay scalable TLP and,

6. Finally, the TLP applicable to WMNs should be distance tolerant.

5.2. Future Work

Actually, there are several performance metrics (such as jitter, throughput smoothness, fairness) that we did not afford to evaluate because of our work schedule. As multi-hop wireless network is made up of various types of networks, these different types of networks work well with different network topology and nodes such as stationary and mobile and many more parameters. We have only to focus on the stationary WMNs. Mobility was not considered in this study; therefore, in future we should consider Mobile Ad hoc Networks (MANETs) and compare the results.

Wireless Sensor Network was also not considered in our work, and it is not clear whether the results we have achieved could be same in Sensor Network. As a result, Wireless Sensor Networks should be taken into consideration using similar parameters in every aspect in future. When we reviewed the literature we noted that, most of the TLPs for wireless multi-hop networks are not implemented in real life scenario. This gives an impression that even though more work has been done concerning TLPs in this area, but translating it to real world scenario is still a challenge. We, therefore, wish to focus our attention on testing these protocols in real-life scenarios. We intend to compare the NS2 and Test-bed results and come up with solid conclusions and recommendations in future.

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APPENDIX A - AWK Scripts for analysing the trace file

```
Begin{
recvdSize = 0
startTime = 1e6
stopTime = 0
}
{
# Trace line format: normal
if ($2 != "-t") {
event = $1
time = $2
if (event == "+" || event == "-") node_id = $3
if (event == "r" || event == "d") node_id = $4
flow_id = $8
pkt_id = $12
pkt_size = $6
flow_t = $5
level = "AGT"
}
# Trace line format: new
if ($2 == "-t") {
event = $1
time = $3
node_id = $5
flow_id = $39
pkt_id = $41
pkt_size = $37
flow_t = $45
level = $19
}
# Store packets send time
if (level == "AGT" && flow_id == flow && node_id == src &&
sendTime[pkt_id] == 0 && (event == "+" || event == "s") && pkt_size >= pkt)
{
if (time < startTime) {
startTime = time
}
sendTime[pkt_id] = time
this_flow = flow_t
}
# Update total received packets' size and store packets arrival time
```

```
if (level == "AGT" && flow_id == flow && node_id == dst &&
event == "r" && pkt_size >= pkt) {
if (time > stopTime) {
stopTime = time
}
# Rip off the header
hdr_size = pkt_size % pkt
pkt_size -= hdr_size
# Store received packet's size
recvdSize += pkt_size
# Store packet's reception time
recvTime[pkt_id] = time
}
}
END {
# Compute average delay
delay = avg_delay = recvdNum = 0
for (i in recvTime) {
if (sendTime[i] == 0) {
printf("\nError in delay.awk: receiving a packet that wasn't sent %g\n",i)
}
delay += recvTime[i] - sendTime[i]
recvdNum ++
}
if (recvdNum != 0) {
avg_delay = delay / recvdNum
} else {
avg_delay = 0
}
# Compute average jitters
jitter1 = jitter2 = jitter3 = jitter4 = jitter5 = 0
prev_time = delay = prev_delay = processed = deviation = 0
prev_delay = -1
for (i=0; processed<recvdNum; i++) {
if(recvTime[i] != 0) {
if(prev_time != 0) {
delay = recvTime[i] - prev_time
e2eDelay = recvTime[i] - sendTime[i]
if(delay < 0) delay = 0
if(prev_delay != -1) {
jitter1 += abs(e2eDelay - prev_e2eDelay)
jitter2 += abs(delay-prev_delay)
jitter3 += (abs(e2eDelay-prev_e2eDelay) - jitter3) / 16
jitter4 += (abs(delay-prev_delay) - jitter4) / 16
}
# deviation += (e2eDelay-avg_delay)*(e2eDelay-avg_delay)
```

```
prev_delay = delay
prev_e2eDelay = e2eDelay
}
prev_time = recvTime[i]
processed++
}
}
if (recvdNum != 0) {

jitter1 = jitter1*1000/recvdNum
jitter2 = jitter2*1000/recvdNum
}
# if (recvdNum > 1) {
# jitter5 = sqrt(deviation/(recvdNum-1))
# }
# Output
if (recvdNum == 0) {
printf(
"##############################################################
###########\n" \
"# Warning: no packets were received, simulation may be too short #\n" \
"##############################################################
###########\n\n")
}
printf("\n")
printf(" %15s: %g\n", "flowID", flow)
printf(" %15s: %s\n", "flowType", this_flow)
printf(" %15s: %d\n", "srcNode", src)
printf(" %15s: %d\n", "destNode", dst)
printf(" %15s: %d\n", "startTime", startTime)
printf(" %15s: %d\n", "stopTime", stopTime)
printf(" %15s: %g\n", "receivedPkts", recvdNum)
printf(" %15s: %g\n", "avgTput[kbps]", (recvdSize/(stopTime-startTime))*(8/1000
))
printf(" %15s: %g\n", "avgDelay[ms]", avg_delay*1000)
printf(" %15s: %g\n", "avgJitter1[ms]", jitter1)
printf(" %15s: %g\n", "avgJitter2[ms]", jitter2)
printf(" %15s: %g\n", "avgJitter3[ms]", jitter3*1000)
printf(" %15s: %g\n", "avgJitter4[ms]", jitter4*1000)
# printf(" %15s: %g\n", "avgJitter5[ms]", jitter5*1000)
# %9s %4s %4s %6s %5s %13s %14s %13s %15s %15s %15s %15s %15s\n\n", \
# "flow","flowType","src","dst","start","stop","receivedPkts", \
# "avgTput[kbps]","avgDelay[ms]","avgJitter1[ms]","avgJitter2[ms]", \
# "avgJitter3[ms]","avgJitter4[ms]","avgJitter5[ms]")
# printf(" %6g %9s %4d %4d %6d %5d %13g %14s %13s %15s %15s %15s %15s\n\n",
\
```

```
# flow,this_flow,src,dst,startTime, stopTime, recvdNum, \
# (recvdSize/(stopTime-startTime))*(8/1000),avg_delay*1000, \
# jitter1,jitter2,jitter3*1000,jitter4*1000,jitter5*1000)
}
function abs(value) {
if (value < 0) value = 0-value
return value
}
```

APPENDIX B- NS-2 Simulation Script for TLPs

```
#
=================================================================
# Define options
#
=================================================================
set val(chan) Channel/WirelessChannel ;# channel type
set val(prop) Propagation/TwoRayGround ;# radio-propagation model
set val(netif) Phy/WirelessPhy ;# network interface type
set val(mac) Mac/802_11 ;# MAC type
set val(ifq) Queue/DropTail/PriQueue ;# interface queue type
set val(ll) LL ;# link layer type
set opt(ll) LL/LLSnoop
set val(ant) Antenna/OmniAntenna ;# antenna model
set opt(tcp) Snoop
set opt(Sink) TCPSink
set opt(app) FTP
set opt(seed) 0
set opt(bw) 10mb
set val(ifqlen) 50 ;# max packet in ifq
set val(nn) 200 ;# number of mobilenodes
set val(rp) HWMP ;# routing protocol
set opt(scen) "scen10c"
set opt(tfc) "die1"
#
=================================================================
=======
# Main Program
#
=================================================================
=======
#
# Initialize Global Variables
#
set ns_ [new Simulator]
set tracefd [open scene10c.tr w]
$ns_ use-newtrace
$ns_ trace-all $tracefd
set f0 [open MyTrace.xls w]
set j 0
# set up topography object
set topo [new Topography]
$topo load_flatgrid 2500 1500
#
```

```
# Create God
#
#create-god $val(nn)
set god_ [create-god $val(nn)]
#
# Create the specified number of mobilenodes [$val(nn)] and "attach" them
# to the channel.
# Here two nodes are created : node(0) and node(1)
$ns_ node-config -adhocRouting $val(rp) \
-llType $val(ll) \
-macType $val(mac) \
-ifqType $val(ifq) \
-ifqLen $val(ifqlen) \
-antType $val(ant) \
-propType $val(prop) \
-phyType $val(netif) \
-channelType $val(chan) \
-topoInstance $topo \
-agentTrace ON \
-routerTrace ON \
-macTrace OFF \
-movementTrace OFF \
for {set i 0} {$i < $val(nn) } {incr i} {
set node_($i) [$ns_ node]
$node_($i) random-motion 0 ;# disable random motion
}
#
# Provide initial (X,Y, for now Z=0) co-ordinates for mobilenodes
#
puts "Loading connection pattern..."
source $opt(scen)
puts "Loading traffic file..."
source $opt(tfc)
# Setup traffic flow between nodes
# TCP connections between node_(0) and node_(1)
#
# Tell nodes when the simulation ends
#
for {set i 0} {$i < $val(nn) } {incr i} {
$ns_ at 100 "$node_($i) reset";
}
$ns_ at 100.1 "stop"
$ns_ at 100.1 "puts \"NS EXITING...\" ; $ns_ halt"
proc stop {} {
global ns_ tracefd f0
$ns_ flush-trace
```

```
close $tracefd
close $f0
#exec grep -e "d -t" testing50.tr > testing50r.tr &
proc record {TCP} {
global f0 j
#set RET [$TCP set nrexmit_]
set RET1 [$TCP set nrexmitpack_]
puts $f0 "$RET1"
set j [expr $j + $RET1]
#puts "we are here now ... "
puts "$j"
}
puts "Starting Simulation..."
$ns_ run
```

E: