An Adaptive Packet Aggregation Algorithm (AAM) for Wireless Networks

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2014-02-12

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by

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A thesis submitted to the Dublin Institute of Technology

for the degree of *Doctor of Philosophy*

Supervisor: Dr. Mark Davis

School of Electronic and Communications Engineering

November 2013

Acknowledgements

This thesis would not exist without the support from my supervisor, families and friends etc. I am very happy to take this opportunity to give my appreciation to those people who have helped and guided me during my graduate career.

First and foremost I would like to acknowledge my supervisor Dr. Mark Davis who gave me the opportunity to study at the Communications Network Research Institute (CNRI) in Dublin Institute of Technology (DIT) for several years and guided me on how to do PhD research. Dr. Mark Davis has been tremendous supervisor proving me invaluable guidance and advices about research, academic skills and the Ireland culture. He is open-minded, supportive and caring. He gave me the confidence and knowledge to help me to turn my dreams into reality. I own him a great many heartfelt thanks.

I also would like to express my gratitude to Prof. Gerald Farrell, Prof. Zhiguang Qin, and Prof. Bing Wu, who helped me to gain the scholarship from China Scholarship Council (CSC) and the opportunity of studying in DIT. Prof. Gerald Farrell gave me a warm reception on the first day I arrived in DIT and I appreciate that he always is ready to help me. Prof. Zhiguang Qin always gives me good suggestions and encourages me to make plan for the future. Prof. Bing Wu leads me to know Ireland and introduced me to Prof. Gerald Farrell. Prof. Bing Wu is very kind heart and friendly. I appreciate what they have done for me.

During the life time in Ireland, the friendship always supports me. I apology I am not listing you all by name and there is a few people mentioned: Dr. Mirosław Narbutt, Dr. Tanmoy Debnath, Dr. Mustafa Ramadhan, Dr. Brian Keegan, Chengzhe Zhang, Yin Chen, Dr. Rong Hu, Dr. Erqiang Zhou, Dr. Yi(熠) Ding, Jianfeng Wu, Fuhu Deng, Yi (一) Ding, Dr. Yupeng Liu, Dr. Qiaohuan Chen, Bilu, Jenny, Panpan Lin, Albuto Dotto and Claude Dyer. I have sweet memories from these friends of Dr. Rong Hu, Dr. Erqiang Zhou, Dr. Yi (熠) Ding, Jianfeng Wu, Fuhu Deng, Yi (一) Ding who cooked for me, discussed with me and made the house feel like home.

I would like to give high respect to my fantastic wife Rong Yang who always loves, encourages and supports me since we met in seven years ago. Without her loves, supports and sacrifices, this work would have never come to fruition. I would like to thank Rong Yang for sharing these years and for brightening my life. In the middle of the night of 14th March 2011, the most original creation I was involved finally arrived, Kaihao Deng, my dearest son. He makes my life different on every day from his birth. He always uses his smiles and voices including laugh and cry to tell me that he loves me which makes me powerful to face all of the difficulties.

Declaration

I certify that this thesis which I now submit for examination for the award of Doctor of Philosophy, is entirely my own work and has not been taken from the work of others, save and to the extent that such work has been cited and acknowledged within the text of my work.

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Abstract

Packet aggregation algorithms are used to improve the throughput performance by combining a number of packets into a single transmission unit in order to reduce the overhead associated with each transmission within a packet-based communications network. However, the throughput improvement is also accompanied by a delay increase. The biggest drawback of a significant number of the proposed packet aggregation algorithms is that they tend to only optimize a single metric, i.e. either to maximize throughput or to minimize delay. They do not permit an optimal trade-off between maximizing throughput and minimizing delay. Therefore, these algorithms cannot achieve the optimal network performance for mixed traffic loads containing a number of different types of applications which may have very different network performance requirements. In this thesis an adaptive packet aggregation algorithm called the Adaptive Aggregation Mechanism (AAM) is proposed which achieves an aggregation trade-off in terms of realizing the largest average throughput with the smallest average delay compared to a number of other popular aggregation algorithms under saturation conditions in wireless networks. The AAM algorithm is the first packet aggregation algorithm that employs an adaptive selection window mechanism where the selection window size is adaptively adjusted in order to respond to the varying nature of both the packet size and packet rate. This algorithm is essentially a feedback control system incorporating a hybrid selection strategy for selecting the packets. Simulation results demonstrate that the proposed algorithm can (a) achieve a large number of sub-packets per aggregate packet for a given delay and (b) significantly improve the performance in terms of the aggregation trade-off for different traffic loads. Furthermore, the AAM algorithm is a robust algorithm as it can significantly improve the performance in terms of the average throughput in error-prone wireless networks.

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Abbreviations and Acronyms

A^3	Adjustable Aggregation Algorithm
AAM	Adaptive Aggregation Mechanism
AARF	Adaptive auto rate fallback
ACK	Acknowledgement
AD	Additive Decrease
AF	Adaptive with FIFO Packet Aggregation Algorithms
AI	Additive Increase
AIMD	Additive Increase Multiplicative Decrease
A-MPDU	Aggregate MAC Protocol Data Unit
A-MSDU	Aggregate MAC Service Data Unit
ANF	Adaptive with Non-FIFO Packet Aggregation Algorithms
AP	Access Point
APA	Aggregate Packet Analyzer
API	Application Program Interface
ARF	Auto Rate Fallback
ATA	Aggregate Tuning Algorithm
BA	Block ACK
BC	Back-off Counter
BER	Bit Error Rate
BSS	Basic Service Set
CCDF	Complementary Cumulative Distribution Function
CDF	Cumulative Distribution Function
CRC	Cyclic Redundancy Check
CSMA/CA	Carrier Sense Multiple Access with Collision Avoidance
CTS	Clear-to-Send
CW	Contention Window
DCF	Distribution Coordination Function
DIFS	DCF Inter Frame Space

DLL	Delay Lower Limit
DDSS	Direct Sequence Spread Spectrum
DTMC	Discrete Time Markov Chain
EIFS	Extended Inter-Frame Space
FCS	Frame Check Sequence
FER	Frame Error Rate
FF	Fixed with FIFO Packet Aggregation Algorithms
FHSS	Frequency Hopping Spread Spectrum
FIFO	First-In First-Out
FNF	Fixed with Non-FIFO Packet Aggregation Algorithms
FR	Frame Relay
FSE	Frame Size Estimation
Gbps	Gigabits Per Second
HCF	Hybrid Coordination Function
IBSS	Independent Basic Service Set
IEEE	Institute of Electrical and Electronics Engineers
IR	Infrared
ISM	Industrial, Scientific and Medical
LLC	Logical Link Control
MAC	Medium Access Control
Mbps	Megabits Per Second
MD	Multiplicative Decrease
MI	Multiplicative Increase
MIMO	Multiple-Input Multiple-Output
MU-MIMO	Multi-user MIMO
NAV	Network Allocation Vector
NEST	Network Simulator Test-bed
OFDM	Orthogonal Frequency Division Multiplexing
PDF	Point Coordination Function

PHY	Physical Layer
PIFS	PCF Inter-Frame Space
PLCP	Physical Layer Convergence Protocol
PPDU	Physical Layer Protocol Unit
PS-Poll	Power-Saver Poll
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
RTS	Request-to-Send
SIFS	Short Inter-Frame Space
SSFS	Smallest-Size First-Served
TUL	Throughput Upper Limit
VoIP	Voice over IP
WDS	Wireless Distribution System
WLANs	Wireless local area networks
WMANs	Wireless Metropolitan Area Networks
WMNs	Wireless Mesh Networks
WNICs	Wireless Network Interface Controllers
WPANs	Wireless Personal Area Networks
WWANs	Wireless Wide Area Networks

Chapter 1
Introduction

In 1997 the IEEE LAN/MAN Standards Committee approved the first version of the IEEE 802.11 standard [IEE97]. Since then, there have been numerous amendments to the standard to achieve the goal of realizing ever higher throughputs. Increasing the transmission rate and the use of ever more complex modulation schemes have allowed for a further improvement in the throughput performance in wireless local area networks (WLANs). However, as a consequence of the protocol headers, there exists an upper limit on the achievable throughput which has been demonstrated by the authors in [XiR02] where a lower limit on the delay has also been demonstrated. The existence of such limits indicate that simply increasing the data rate without reducing the PHY (Physical Layer) and MAC (Medium Access Control) overheads is bounded even if the data rate is increased indefinitely. This has lead to the use of packet aggregation where the throughput is increased as the protocol headers are reduced by combining a number of small size packets into a single large size (or aggregate) packet.

Packet aggregation is the process of combining multiple packets together into a single transmission unit in order to reduce the overhead associated with each transmission within a packet-based communications network. In 2009 the IEEE 802.11n standard defined two packet aggregation algorithms that are also employed in the IEEE 802.11ac standard draft: Aggregate MAC Service Data Unit (A-MSDU) and Aggregate MAC Protocol Data Unit (A-MPDU). However, the throughput improvement is also associated with a delay increase as the packet aggregation algorithm may have to wait for packets to arrive in order to be assembled into an aggregate packet.

1.1 Problem Statement

As most of the proposed packet aggregation algorithms don't take account of the varying nature of the traffic loads particularly the random nature of the packet size and packet rate, these algorithms tend to optimize a single metric, i.e. either to maximize throughput or to minimize delay. In general, they do not permit an optimal trade-off between the two metrics which would allow for greater flexibility in operating under a wide range of mixed traffic loads.

Generally, in modern networks the traffic load is a mix of different types of application (e.g. VoIP and E-mail) which often have very different network performance requirements. Consequently, optimal network performance cannot be achieved simultaneously for mixed traffic loads by employing a packet aggregation algorithm that only optimizes a single metric.

So there is a need for an adaptive packet aggregation algorithm that is better suited to the mixed traffic loads found in modern data networks. This adaptive algorithm not only achieves an optimal trade-off between maximizing throughput and minimizing delay in a data network but also provides a good performance over a wide range of mixed traffic loads.

1.2 Objectives and Contributions

In this thesis an adaptive packet aggregation algorithm called the Adaptive Aggregation Mechanism (AAM) is proposed which can operate over a wide range of different traffic loads in order to achieve the best aggregation trade-off in terms of realizing the largest average throughput with the smallest average delay compared to a number of other popular aggregation algorithms under saturation conditions in wireless networks. The AAM algorithm is a robust adaptive packet aggregation algorithm where a feedback control scheme incorporating a hybrid selection strategy and a tunable selection window mechanism is employed in order to respond to the varying nature of the packet size and packet rate. The operation of the AAM algorithm is based upon the use of a selection window whose size is adaptively adjusted. In general, increasing the selection window size will increase the probability of achieving the target aggregate packet size (accompanied by a larger delay), while reducing the selection window size will reduce the delay but will also reduce the probability of attaining the target aggregate packet size. There are three elements configured in a feedback control system in order to achieve the robustness for the AAM algorithm: Adjustable Aggregation Algorithm (A^3), Aggregate Packet Analyzer (APA) and Aggregate Tuning Algorithm (ATA). The AAM algorithm generates an aggregate packet whose size approaches the target aggregate packet size as closely as possible within a given delay.

In this thesis, the results will demonstrate that:

- The AAM algorithm is an adaptive algorithm that can aggregate the largest number of sub-packets per aggregate packet with a given average packet delay compared to the FIFO (First-In First-Out) and SSFS (Smallest-Size First-Served) algorithms.

- The AAM algorithm has the best performance in terms of the aggregation trade-off in achieving the largest average throughput with the smallest average delay for all three algorithms considered (i.e. AAM, FIFO, and SSFS) under saturation conditions in wireless networks.

- The AAM algorithm is a robust algorithm as it can significantly improve the throughput by up to 28% in error-prone wireless networks.

- The AAM algorithm can operate over a wide range of different traffic loads in wireless networks with and without transmission errors present.

1.3 Organization

This thesis is organized as follows.

Chapter 2 describes the main technologies that are used throughout the course of this research by introducing the general technical background regarding wireless networks before concentrating on the operation of packet aggregation. Chapter 2 overviews parts of the IEEE 802.11 standards, the architecture of the WLANs, the MAC mechanism of the IEEE 802.11 standards and the structure of the IEEE 802.11 frames which are relevant to the thesis. The transmission errors in WLANs, the PHY rate adaption mechanism, network simulator and packet sniffer are also discussed in the final sections of this chapter.

Chapter 3 provides a literature review of packet aggregation algorithms in WLANs that have been proposed by other researchers. This chapter also highlights the recent advances in the area of packet aggregation research.

Chapter 4 describes the design and the development of the AAM algorithm. A fundamental analysis of the AAM algorithm is presented after a detailed description of each stage of the proposed algorithm. A description of the simulation process for the AAM algorithm implemented in two different test scenarios is given that includes all the modeling assumptions adopted in the simulation.

Chapter 5 presents the results for the two performance validation test scenarios. The first section analyses the performance of the AAM algorithm aggregation process only. The next section presents the results of the AAM algorithm when it is implemented in wireless networks with and without transmission errors present. A comparison between the performances is provided in order to further highlight the advantages of the AAM algorithm compared to two other aggregation algorithms (i.e. FIFO and SSFS) based on 16 captured traffic trace files.

Chapter 6 provides a summary of the main findings and conclusions from this research carried out. This chapter also gives some suggestions for the future research in this area.

Chapter 2
Technical Background

In this chapter, relevant background knowledge about IEEE 802.11 wireless local area networks (WLANs), the IEEE 802.11 MAC mechanism, transmission errors and PHY rate adaption mechanism in WLANs, network simulators and packet sniffers will be introduced. In the first section, an introduction to the main standards of IEEE 802.11 WLANs and the architecture of wireless networks are presented. The second section focuses on the MAC mechanism of the IEEE 802.11 WLAN standards and then the formats of some of the IEEE 802.11 frames are presented. The detrimental impact of transmission errors in WLANs are described in the fourth section and some PHY rate adaption mechanisms are introduced in the following section. A discussion of the network simulator ns-3 is given in the sixth section and the packet sniffer application *Wireshark* is described in the last section.

2.1 IEEE 802.11 Wireless Local Area Networks

In the last decade, Wireless Local Area Networks (WLANs) based on the IEEE 802.11 standards have been widely employed in the home and enterprise networks across the world. The IEEE 802.11 standard was approved by the IEEE LAN/MAN Standards Committee in 1997 [IEE97]. The original version of the IEEE 802.11 standard defined a single Medium Access Control (MAC) accessed by the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) mechanism and a Physical Layer (PHY) which defined PHY rates of 1 Mbps and 2 Mbps. The PHY defined three types of modulation technique: Infrared (IR), Frequency Hopping Spread Spectrum (FHSS) and Direct Sequence Spread Spectrum (DSSS).

Further enhancements to the original standard, namely the IEEE 802.11b [IEEb99] and IEEE 802.11a [IEa99] standards were both published in 1999. The IEEE 802.11b standard supports 1, 2, 5.5 and 11 Mbps PHY rates in the license-free 2.4 GHz ISM (Industrial, Scientific and Medical) band, while the IEEE 802.11a standard by using the Orthogonal Frequency Division Multiplexing (OFDM) provides 8 PHY rates (i.e. 6, 9, 12, 18, 24, 36, 48 Mbps and 54 Mbps) in the license-free 5 GHz ISM band. In June of 2003, the IEEE 802.11g [IEE03] standard was approved which provides a maximum 54 Mbps PHY rate in the 2.4 GHz ISM band. The IEEE 802.11n standard [IEn09] was published in September of 2009 which allows for a maximum of 100 Mbps PHY rate in both the 2.4 GHz and 5 GHz ISM bands by using channel bonding with up to 72 Mbps without channel bonding. The new multiple antenna technology MIMO (Multiple-Input Multiple-Output) and the packet aggregation are employed in the IEEE 802.11n standard. The standard for the next generation of wireless networks is the IEEE 802.11ac which is still under development. The draft 5.0 was published at the beginning of 2013 [IEE13]. It provides higher throughput for WLANs on the 5 GHz ISM bands [R&S11]. Theoretically, this specification will enable multi-station WLAN throughput of at least 1 Gbps and a maximum single link throughput of at least 500 Mbps by using some new technologies, such as extended channel bonding, Multi-user MIMO (MU-MIMO) and packet aggregation [Any12]. The IEEE 802.11ac will provide backwards compatibility with the IEEE 802.11a and IEEE 802.11n devices operating in the 5 GHz ISM band [War12]. The IEEE 802.11ac standard is expected to be ratified in the early 2014 and the maximum PHY rate will be in excess of 5 Gbps.

Some members of the IEEE 802.11 family of standards are shown in Table 2-1 where there are 5 main versions: IEEE 802.11a, IEEE 802.11b, IEEE 802.11g, IEEE 802.11n which are now widely used to provide wireless connectivity in homes and businesses, and the latest standard IEEE 802.11 ac is still under development.

Table 2-1: Some members of the IEEE 802.11 family of WLAN standard

Standard	**Comments**
802.11a	Extends the PHY rate to up to 54 Mbps in the 5 GHz ISM band
802.11b	Extends the PHY rate to 11 Mbps in the 2.4 GHz ISM band
802.11c	Incorporates bridging in Wireless Bridges or AP (Access Point)
802.11d	Supports operation in additional regulatory domains
802.11e	Defines the QoS (Quality of Service) enhancement mechanisms
802.11f	Provides AP communications among multivendor systems
802.11g	Extends the PHY rate to up to 54 Mbps in 2.4 GHz ISM band
802.11h	Supports the power control mechanisms in 5 GHz ISM band
802.11i	Specifies the security mechanisms
802.11n	Extends the PHY rate to up to 600 Mbps and supports Frame Aggregation
802.11p	Supports access in vehicular environment
802.11s	Supports the creation of mesh networks
802.11ac	Extends the PHY rate to up to 5 Gbps and is still under development
802.11ad	Extends the PHY rate to up to 7 Gbps and is still under development

2.1.1 IEEE 802.11a standard

The IEEE 802.11a standard was ratified in 1999 and uses Orthogonal Frequency Division Multiplexing (OFDM) in the unlicensed 5 GHz ISM band to extend the PHY rate maximum of 54 Mbps but it also supports lower PHY rates at 6, 9, 12, 18, 24, 36 and 48 Mbps. The OFDM is a mechanism for encoding digital data on multiple orthogonal subcarriers [IEa99]. Actually, the OFDM is a digital modulation method in which a signal is split into several narrowband channels at different frequencies. This technology is also used in the IEEE802.11g and IEEE 802.11n standards. In this thesis, all the PHY rates in the IEEE 802.11a standard are used to demonstrate the performance of the proposed AAM algorithm.

2.1.2 IEEE 802.11n Standard

The IEEE 802.11n standard was introduced to increase the PHY rate from 54 Mbps to 600 Mbps by adding the Multiple-Input Multiple-Output (MIMO) mechanism and 40 MHz channels to the Physical Layer (PHY) and also by employing a packet aggregation algorithm at the MAC layer.

MIMO is a technology that allows multiple antennas to send and receive multiple spatial streams at the same time in order to coherently resolve more information than that of using a single antenna. Using multiple antennas the data can be sent and received through multiple signals and more antennas usually equates to higher speeds [IEE09]. The IEEE 802.11n standard specified that the devices can use up to 4 antennas to transmit data at the same time.

Packet aggregation is a method used to improve throughput by sending a large aggregate packet which contains more than one smaller size data packet. Two packet aggregation algorithms are defined in the IEEE 802.11n standard: Aggregate MAC Service Data Unit (A-MSDU) and Aggregate MAC Protocol Data Unit (A-MPDU). Both algorithms combine several data packets into a single large packet to improve the throughput. More accurately, packet aggregation is used to reduce the impact of header overhead on throughput. The ratio of the payload to the transmitted frame size is higher as the frame header information needs to be specified only once per aggregate packet [IEE09]. In this thesis, the basic algorithm A-MSDU is employed as the typical benchmark packet aggregation algorithm to study the performance of the AAM algorithm.

2.1.3 IEEE 802.11ac standard

The goal of the IEEE 802.11ac standard is to provide new PHY rates from 500 Mbps to 5 Gbps by employing some new technologies [IEE13]. It extends the air interface concepts embraced by the IEEE 802.11n standard to accomplish even higher throughputs. It extends the channel band from the 40 MHz in the IEEE 802.11n standard to 80 MHz or even to 160 MHz and increases the number of MIMO spatial streams to twice that of the IEEE 802.11n standard. The IEEE 802.11ac standard uses the MU-MIMO technology which exploits the availability of multiple independent radio terminals in order to enhance the communication capabilities of each individual terminal and improves the modulation to 256-QAM (Quadrature Amplitude Modulation) [War12]. It also uses the packet aggregation algorithms specified in the IEEE 802.11n standard, i.e. A-MSDU and A-MPDU. The standard was finalized in early 2012 with final IEEE 802.11 Working Group approval expected in early 2014 [Wik13]. According to a study, devices with the IEEE 802.11ac specification are expected to be widely used by 2015 with an estimated one billion devices globally [Tim13]. In the future work, the proposed AAM algorithm will be implemented based on the IEEE 802.11 ac standard.

2.1.4 Architecture of WLANs

A WLAN implements a flexible data communication system frequently augmenting rather than replacing a wired LAN within a building or campus. WLANs use radio frequency communication to transmit and receive data over the air, minimizing the need for wired connections [CIS13]. WLANs have become popular in the home due to easy installation and in commercial complexes offering wireless access to their customers. A WLAN is one type of wireless network and other types defined by their coverage range include the following: Wireless Personal Area Network (WPAN), Wireless Mesh Network (WMN), Wireless Metropolitan Area Network (WMAN), Wireless Wide Area Network (WWAN) and the Mobile Network.

A WLAN links two or more devices using some wireless distribution method, Spread-Spectrum, Orthogonal Frequency Division Multiplexing (OFDM), or MIMO radio, and usually provides a connection through an access point (AP) to the wired network. This gives user the mobility to move around within a local coverage area and still remain connected to the network and most of the modern WLANs are based on the IEEE 802.11 standards. All components that can connect into a wireless medium in a network are referred to as station. All the stations are equipped with wireless network interface controllers (WNICs). Wireless stations fall into one of two categories: access points (APs) and client stations [Fra03]. Access points (APs), or routers, essentially act as base stations for wireless networks that connect wireless enabled client devices to a backbone network. Wireless client stations can be mobile devices such as laptops, personal digital assistants, IP phones and other smart phones, or fixed devices such as desktops and workstations that are equipped with a wireless network interface. In this thesis, the simulation is based on a single hop WLAN in which a single AP and a single client are implemented to investigate the performance of the AAM algorithm.

2.2 IEEE 802.11 MAC Mechanism

There are three ways to access the wireless medium that are defined in MAC specification of the IEEE 802.11 standard: Point Coordination Function (PCF) and Hybrid Coordination Function (HCF) and Distributed Coordination Function (DCF).

The PCF provides contention-free services in infrastructure networks but it has not been widely implemented. The HCF supports the high Quality of Service (QOS) through the hybrid DCF and PCF and also allows stations to utilize multiple service queues when accessing the medium. Although specified in the IEEE 802.11e standard, the HCF has not been widely implemented. The DCF is the basic mechanism to access the wireless medium and is based upon a random back-off scheme.

There are four types of inter-frame spaces defined in the MAC specification: DCF Inter-Frame Space (DIFS), Short Inter-Frame Space (SIFS), PCF Inter-Frame Space (PIFS) and Extended Inter-Frame Space (EIFS) as shown in Figure 2-1. The first three of them are employed to control access the medium while the EIFS is used when there is a transmission error present in packet transmission and it does not have a fixed duration.

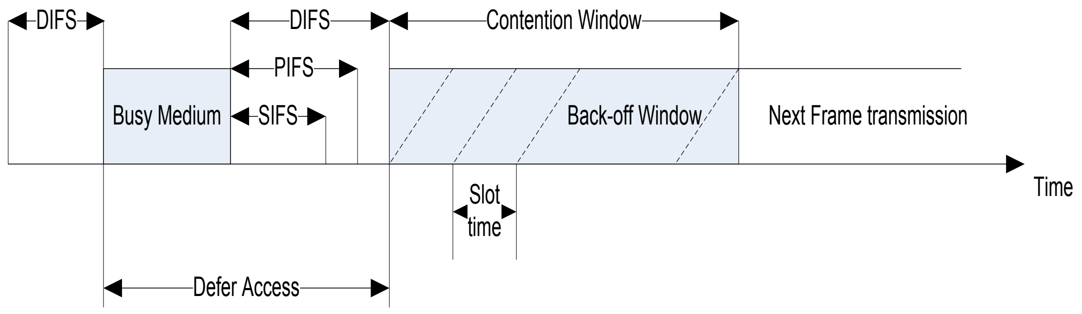


Figure 2-1: The use of Inter-Frame Spaces in accessing the medium.

The DIFS is the minimum medium idle time for contention based services in general. The PIFS is shorter than DIFS and employed by PCF in contention-free operation. The SIFS is shorter than PIFS but is only used for the highest priority transmission of control frames (e.g. ACK). In the IEEE 802.11a, IEEE 802.11b, IEEE 802.11g, IEEE 802.11n and IEEE 802.11ac standards, the durations of SIFS, DIFS and the Slot Time are shown in Table 2-2.

When packets are awaiting transmission in a buffer, the client station has to determine whether the channel is busy or not by using a carrier-sensing function. There are two types of carrier-sensing mechanism supported in the IEEE 802.11 standard: Physical carrier sensing supported by the physical layer and the virtual carrier sensing provided by the network allocation vector (NAV). The NAV is a timer used to indicate the amount of the time that the medium will be reserved [IEa99].

Table 2-2: The values of slot time, SIFS, DIFS and *CW* for the different IEEE 802.11 standards

Standard	**Slot Time (µs)**	**SIFS (µs)**	**DIFS (µs)**	**Min. *CW***	**Max. *CW***
IEEE 802.11a	9	16	34	15	1023
IEEE 802.11b	20	10	50	31	1023
IEEE 802.11g	9 or 20	10	28 or 50	31 or 15	1023
IEEE 802.11n	9	16	34	15	1023
IEEE 802.11ac	9	16	34	15	1023

If the channel is busy, all the stations have to wait for a duration of DIFS until the channel is idle and then employ the random back-off scheme to initialize a Back-off Counter (*BC*) which starts to decrease at every slot time in which the medium remains idle. The *BC* is frozen whenever the channel becomes busy. The *BC* is initialized by randomly picking an integer from a Contention Window (*CW*) which is divided into slots whose duration depends on the modulation format and frequency band used. The values of the slot time for the different IEEE 802.11 standards are shown in Table 2-2. When a *BC* has decremented to zero, the station gains the authorization to use the channel and transmit its packet. If there is more than one station trying to access medium, the station whose *BC* first reaches zero gains the authorization to transmit its packet. A collision occurs when two or more *BCs* reach zero at the same time [IEa99]. In this case, they continue to transmit their frames; however the collision causes the frames to be received incorrectly by the receiver which does not respond with an ACK frame. This in turn triggers a re-transmission of the frames by the stations involved. Therefore, they have to restart the random access process again to reset the BC but the size of the CW has been doubled. The size of CW is calculated by the Binary Exponential Back-off Algorithm which is 1 less than an integer power of 2 (i.e. 1, 3, 7…. 511 and 1023). The CW moves to the next greater power of two [IEa09] every time when the BC is reset as a failed transmission. The CW is reset to the minimum size when a packet is transmitted successfully, or the associated re-try counter limit is reached and the packet is discarded. The maximum and minimum sizes allowed for CW are presented in Table 2-2. This scheme ensures a low delay when only a few station nodes collide but also ensures that the collision is resolved within an acceptable time interval when large numbers of station nodes collide.

Figure 2-2 illustrates an example of the operation of the DCF in accessing wireless medium. There are two station nodes, A and B. After the station node B receives an ACK and waits a time of DIFS, the channel is idle. Both nodes try to transmit their packets, so they have to set their back-off counter (BC) values: A is set to 4 and B is set to 9. The BC of A decreases to zero after 4 time slots have elapsed and can transmit its packet while B has to freeze its BC at 5 and waits until A completes its transmission. After a successful transmission A waits for a DIFS time and resets the BC (this time it has chosen 8) and B just restarts the BC (which is 5). The station node B can transmit its packet when its BC reaches zero after 5 time slots.

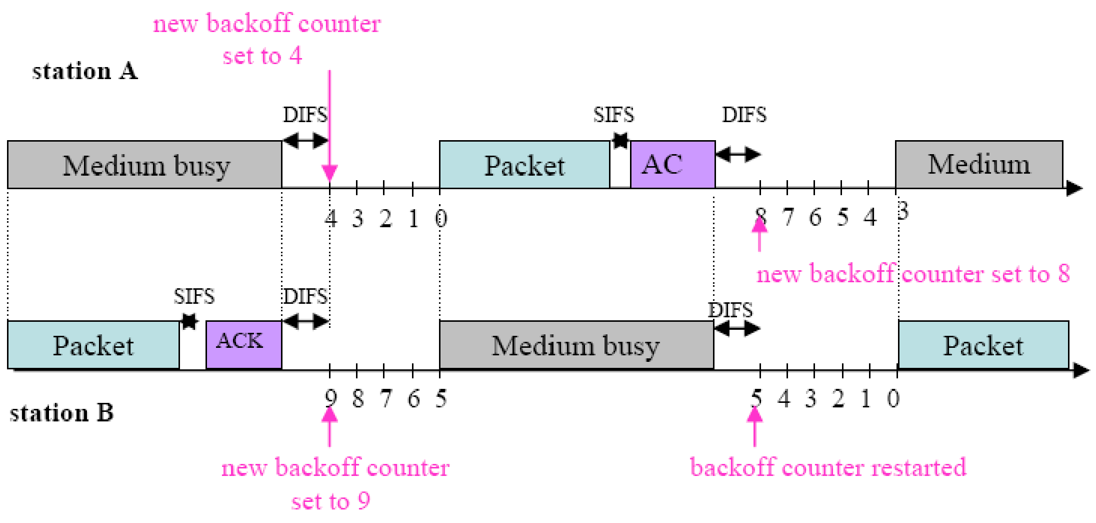


Figure 2-2: An example of the DCF operation used to access the medium.

If the channel is idle, the station node has to wait for a time of DIFS and when its back-off counter (*BC*) has reached zero before it may transmit its packet. When a packet is received by the destination node, the destination node has to wait a time of SIFS and then sends an Acknowledgement (ACK) packet back to the source node to indicate a successful reception of the data packet. In this thesis, there is a single client station used in the wireless network of the simulation for the AAM algorithm and the station can always gain the authorization to use the medium as collisions do not occur as there is no contention for access. The AAM algorithm is intended for use on a single hop link. Therefore, it is sufficient to investigate the performance in a single station.

2.3 IEEE 802.11 Frames

In the IEEE 802.11 standards, there are three types of frame defined: Data frame, Management frame and Control frame.

2.3.1 IEEE 802.11 Data Frame Format

In the IEEE 802.11 standard there are a number of data frame types defined. One way to classify these data frames are as contention-based service data frames and contention-free service data frames. The data frame of the contention-free service can only be used in the contention-free period and cannot be used in IBSS (Independent Basic Service Set). The generic IEEE 802.11 MAC data frame is shown in Figure 2-3. The standard MAC frame of the IEEE 802.11 standards includes two fields: the header information and frame body data. Both of them are defined in the IEEE 802.11 standards but the data frame doesn't include the type/length files and the preamble.

bytes	2	2	6	6	6	2	6	0 -- 2312	4
	Frame Control	Duration /ID	Address 1	Address 2	Address 3	Seq-Ctl	Address 4	Frame Body	FCS

Figure 2-3: The generic IEEE 802.11 MAC data frame format.

As shown in Figure 2-3, the header information in the data frame format includes 6 fields: Frame Control, Duration/ID, Address, Sequence Control and Frame Check Sequence (FCS) fields. The length of header is defined in the standard as 34 bytes but in practice only 28 bytes are used. The reason for this is that for most of the applications, only the first 3 address fields are used and the fourth address file is just employed by bridging services (i.e. the Wireless Distribution System (WDS)). The frame control field is 2 bytes and contains most of the frame information which includes the protocol version, subtype file, re-try bit and protected frame bit and so on. The Duration/ID field follows the frame control field. There are 4 address fields in the IEEE 802.11 frame to set the receiver's address, transmitter's address and filtering address of receiver. The 16-bit sequence control field is employed for both defragmentation and discarding duplicate frames.

In the IEEE 802.11 standard, the maximum payload is 2312 bytes which includes the 8 bytes of the Logical Link Control (LLC) header. In the IEEE 802.11 frame format, there is no padding to ensure a minimum frame length.

The FCS field uses the Cyclic Redundancy Check (CRC) as in the Ethernet frame to check whether a transmission error has occurred or not in the reception of the frame. If a transmission error occurs at the receiver, the receiver will not return an ACK frame to the sender. The frame will then be re-transmitted by the sender.

2.3.2 IEEE 802.11 Control Frame Format

The format of the control frame is shown in Figure 2-4 and it supports the transmission of data frames by helping the station nodes to manage the MAC access. One type of the control frame is the ACK frame which is employed in the positive acknowledgement of received data. Other frames are used to provide for more reliable communication by helping to avoid collisions, such as Request-to-Send (RTS), Clear-to-Send (CTS) and Power-Saver Poll (PS-Poll). In this thesis, only the ACK frame is employed to determine whether the transmission is successful or not.

bits	2	2	4	1	1	1	1	1	1	1	1
	Protocol	Type	Sub-type	ToDs	FromDs	More Frag	Retry	Pwr Mgmt	More data	Protected Frame	Order

Figure 2-4: The frame control field in the IEEE 802.11 control frame.

ACK Frame

The ACK frame as shown in Figure 2-5 is 14 bytes in length and is used to indicate a positive acknowledgment of the frame transmission as required by the MAC and with data frame transmissions frames preceded by the RTS/CTS handshake and fragmented frames. In the IEEE 802.11n and IEEE 802.11ac standards, the Block ACK (*BA*) scheme is employed to improve the MAC efficiency. The *BA* is a special ACK frame which can be used to acknowledge multiple MPDUs. The *BA* is helpful in improving the MAC efficiency when all the frames in a burst are successfully transmitted. However, the whole *BA* must be re-transmitted if any frame in the burst is missing or the acknowledgment itself is corrupted.

bytes	2	2	6	4
	Frame Control	Duration	Destination Address	FCS

Figure 2-5: The format of the IEEE 802.11 ACK frame.

2.3.3 IEEE 802.11 Management Frame Format

The management frames are used to determine the timing, authentication and synchronization of stations in IEEE 802.11 WLANs [IEEb99]. The format of the management frame is shown in Figure 2-6. There are two types of management frames based on the frame body size: fixed-length where the body size is fixed and variable-length where the body size can be varied. A station node uses the Beacon frames to determine which BSS (Basic Service Set) and AP are available and uses the authentication frame to gain the authorization to access the network, then it sends an association frame to join the AP's BSS.

bytes	2	2	6	6	6	6	0 -- 2312	4
	Frame Control	Duration	Destination Address	Source Address	BSS ID	Seq-Ctl	Frame Body	FCS

Figure 2-6: The format of the IEEE 802.11 management frame.

2.4 Transmission Errors in WLANs

In WLANs, path loss, thermal noise, fading, and interference can cause significant packet errors which will have a detrimental impact on the system performance [SHW10]. These transmission errors are often characterized by the bit error rate (BER). The transmission errors can also have a detrimental impact on the performance of a packet aggregation algorithm as they increase the probability of a frame re-transmission [HLL08]. Although packet aggregation can increase the throughput under ideal channel conditions, a larger size aggregate packet may cause each station to wait longer before its next transmission opportunity. However, in error-prone channels, corrupting a large size aggregate packet may waste a long period of channel time and leads to a lower MAC efficiency. So some packet aggregation algorithms [Lin06] [KSP12] have been proposed to improve the performance in error-prone wireless networks. We will discuss this in more detail in the next chapter.

2.5 PHY Rate Adaption Mechanisms in WLANs

In the IEEE 802.11 standards, the PHY allows for a set of different transmission modes to adapt to the channel variations. Each PHY mode uses a specific modulation and channel coding scheme to offer different performance in terms of throughput. Table 2-3 shows the IEEE 802.11b/g/n PHY rates giving the modulation/coding/MIMO details where only the IEEE 802.11n mandatory PHY rates are shown and the other IEEE 802.11n PHY rates can be calculated based on the diagram shown in Figure 2-7.

Table 2-3: The details of the PHY rate for IEEE standards 802.11b/g/n.

Standard	PHY Rate (Mbps)	Modulation	Coding Rate	MIMO
802.11b	1	DBPSK	----	No
	2	DQPSK	----	No
	5.5	CCK	----	No
	11	CCK	----	No
802.11g	6	BPSK	1/2	No
	9	BPSK	3/4	No
	12	QPSK	1/2	No
	18	QPSK	3/4	No
	24	16-QAM	1/2	No
	36	16-QAM	3/4	No
	48	64-QAM	2/3	No
	54	64-QAM	3/4	No
802.11n (mandatory PHY rate)	6.5	BPSK	1/2	1
	13	QPSK	1/2	1
	19.5	QPSK	3/4	1
	26	16-QAM	1/2	1
	39	16-QAM	3/4	1
	52	64-QAM	2/3	1
	58.5	64-QAM	3/4	1
	65	64-QAM	5/6	1

Where, DBPSK: Differential Binary phase-shift keying; DQPSK: Differential Quadrature phase-shift keying; CCK: Complementary Code Keying; QAM: Quadrature amplitude modulation; 16-QAM: 16-state quadrature amplitude modulation; 64-QAM: 64-state quadrature amplitude modulation.

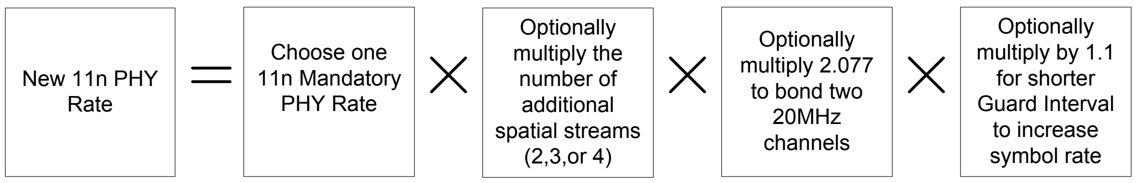

New 11n PHY Rate
=
Choose one 11n Mandatory PHY Rate
×
Optionally multiply the number of additional spatial streams (2,3,or 4)
×
Optionally multiply 2.077 to bond two 20MHz channels
×
Optionally multiply by 1.1 for shorter Guard Interval to increase symbol rate

Figure 2-7: Calculating the new IEEE 802.11n PHY rate.

The PHY rate adaption is the process of dynamically switching the PHY mode to match the channel conditions. The goal is to select the most effective rate that will achieve the maximum throughput for a given channel condition [HVB01]. The effect of transmission errors also impacts on the selection of PHY rate for transmission. There are many PHY rate adaption mechanisms that have been proposed, such as [HVB01] [KaM97] [WYL06] [MLT08], to achieve the goal of realizing a maximum throughput in error-prone wireless network channels.

The PHY rate adaption mechanism can reduce the number of re-transmissions caused by the transmission errors. The ARF (Auto Rate Fallback) [KaM97] mechanism is a simple and widely adopted scheme which is based on the number of consecutive successful or unsuccessful transmission attempts to determine whether to increase or decrease the transmission rate. The disadvantage of the ARF mechanism is that it tries a higher transmission rate every time after it successfully transmits a fixed number of packets even if the current rate is the most effective rate to achieve the maximum throughput. The AARF (Adaptive ARF) mechanism is proposed to alleviate this problem [WYL06]. The AARF mechanism behaves like the ARF mechanism except that the number of consecutive successfully transmission attempts is exponentially incremented when the higher transmission rate has failed. In the Receiver Based Auto

Rate (RBAR) mechanism the RTS/CTS handshake is mandatory and the RTS, CTS frames structure has been modified [HVB01]. The Robust Rate Adaption Algorithm [MLT08] mechanism is composed of the rate selector mechanism and the adaptive RTS mechanism which does not always make the best choice for the rate [YWA04] as the rate selected depended on the used rate. In this thesis, we use the popular AARF mechanism to select the PHY rate in an error-prone wireless network.

2.6 Network Simulators

A network simulator is an important research tool in which a computer program simulates the behavior of a network either by calculating the interaction between the different network entities using mathematical formulas or by actually capturing and playing back observations from a live network. It models the behaviors of the network and the various applications and services which can be observed in a test laboratory. Various attributes of the environment can also be modified in a controlled manner to assess how the network would behave under different conditions.

There are a number of network simulators available such as OPNET, GloMoSim, ns-2 and ns-3 etc. OPNET is a commercial software package for analyzing the performance of computer networks and applications. GloMoSim is another popular network simulator tool that is employed for network research and laboratory experimentation and covers many technologies. The network simulator (ns) has a long history and is derived from REAL (Real and Large). It (i.e. ns) is a name for a series of discrete event network simulators particularly ns-1, ns-2 and ns-3. They are free open source discrete-event network simulators primarily used in research and teaching [TNS12]. The ns-1, the first version of ns, was developed at Lawrence Berkeley National Laboratory (LBNL) between 1995 and 1997. The second version of ns, called ns-2, was based on a refactoring by Steve McCanne in 1996-1997 [BBE99].

The network simulator ns-3 [BEF00] is the third generation of this family of network simulators. The project started in 2006 and is still being actively developed today. It is not an extension of ns-2 and does not support the APIs (Application Program Interface) from ns-2, but some modules of ns-2 have been ported to ns-3 [NS312]. The ns-3 is a new simulator and built on C++ and Python. It is essentially a C++ library where many network simulation modules are implemented as C++ objects and are wrapped by Python. Normally, the C++ or Python applications can instantiate a set of simulation modules to set up the simulation scenario of interest, enter the simulation main loop, and exit when the simulation is completed. It provides support for TCP/UDP, routing, and most of the IEEE 802 standards for wired and wireless networks.

The advantages of ns-3 over other discrete-event network simulators are as follows [NS313]: (i) It uses the object oriented language C++ and Python which allow the user to take advantages of the full support from each language. (ii) The callback-driven events scheme is used to make it easy to turn any function into an event and be scheduled. In fact, the simulation events in ns-3 are simply function calls that are scheduled to execute at a prescribed simulation time. (iii) Different levels of user flexibility. It allows the expert user to configure the core from the low-level APIs which are powerful and the normal users to configure it from invoking the high-level easier-to-use APIs. (iv) An emphasis on simulation that allows the simulator to interact with the real world. Several different simulation-in-the-loop and virtual machine integration frameworks have been developed. (v) Alignment with real-world interfaces where objects (e.g. sockets, net devices) are aligned with those in a Linux computer which facilitates code reuse and improves the realism of the models and makes the simulator control flow easier compared to real system. (vi) Configuration management is developed which uses an integrated attribute-based system to manage default and perinstance values for simulation parameters. In this thesis the ns-3 network simulator is employed to simulate the proposed AAM algorithm.

One of the disadvantages of the ns-3 simulator is that it does not maintain an Integrated Development Environment (IDE) to configure, debug, execute, and visualize simulations in a single application window, such as found in other simulators.

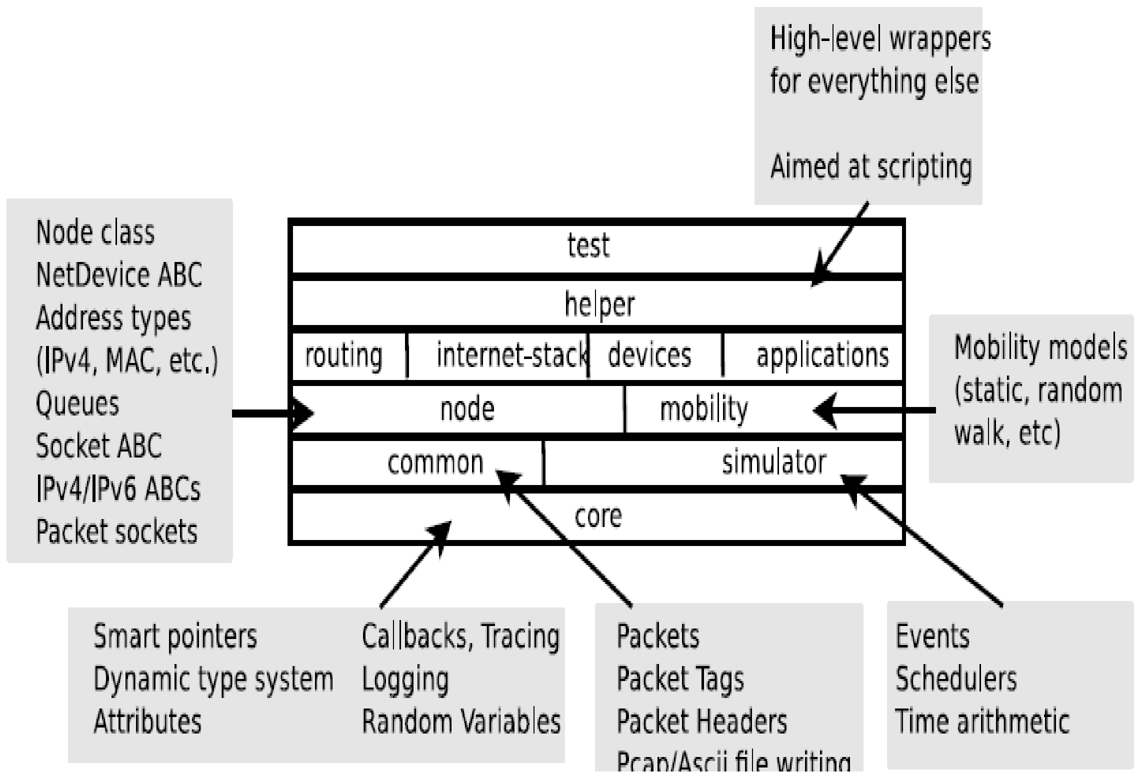
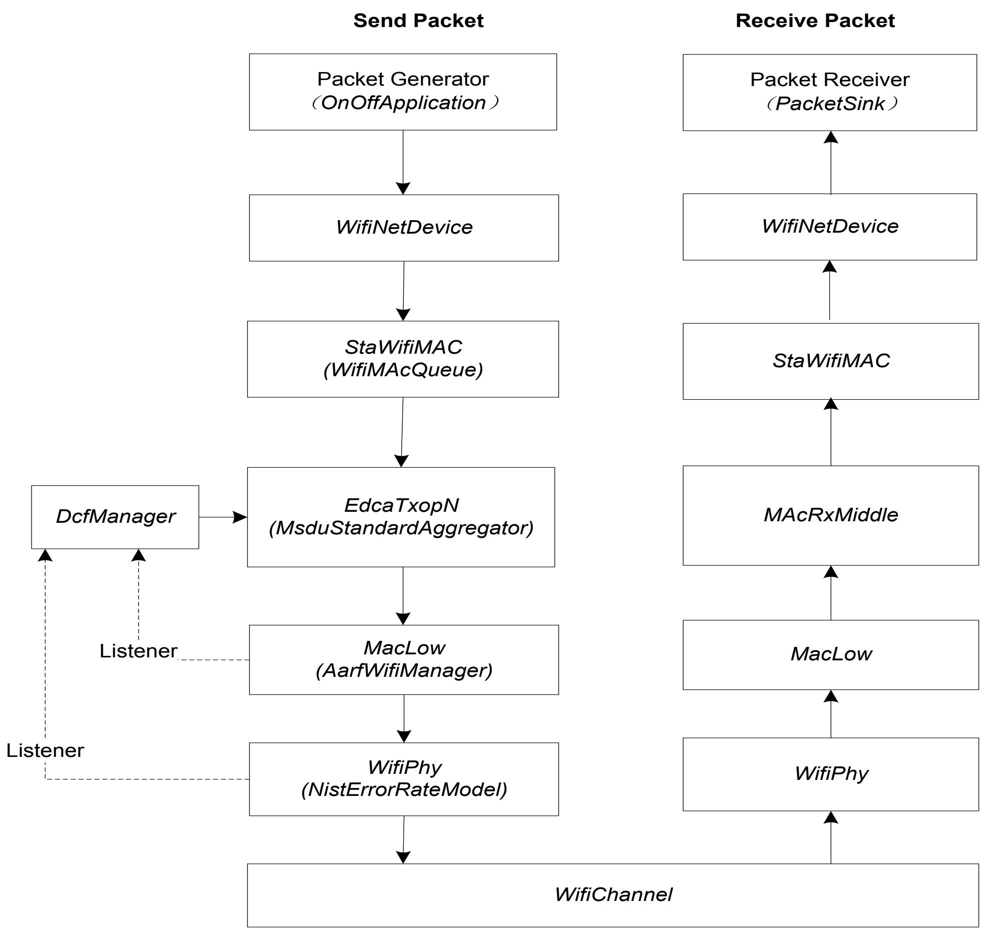


Figure 2-8: The software organization of the ns-3 simulator [NsP12].

The software of ns-3 is currently at version 3.18 and the organizational structure is shown in Figure 2-8 [NS312]. It can be organized into six layers. The bottom layer is the core layer which defines the fundamental modules which include all protocols, hardware and environmental modules, such as *tracing system* and *logging system*. The second layer includes two elements: the *common* module which defines the traffic object packet including how to generate and trace and the *simulator* module is concerned with the events, schedulers and the time arithmetic. The upper layer also has two elements: the *node* module in which a lot of classes are defined to abstract the basic computing device and the *mobility* module that provides the library to support node mobility. The fourth layer is concerned with *routing, device* and *application* modules. The fifth is the *helper* module which provides a set of APIs to help to interact with all other modules. The top layer is the *test* module which contains test cases to allow the user to test the system or modules.

In this thesis, the *wifi* module in the core layer has been modified to include the proposed AAM algorithm and the basic simulation setup for packet aggregation in ns-3 is shown in Figure 2-9. It shows the basic processes required in sending and receiving an aggregate packet to and from another node in a wireless node. For example, to send an aggregate packet from the source node to another node requires the following steps. At first, the packet generator is used to generate the packets. In our simulation, the module *OnOffApplication* is used to generate the required packets which have the same key characteristics (e.g. length, destination IP address) as real world packets. Then the *WifiNetDevice* pushes these packets into the *WifiMacqueue* in the MAC. After gaining the authorization to transmit a packet, the *EdcaTxopN* module invokes the *MsduStandardAggregator/MsduAggregator* to combine packets from the queue. If required, the PHY rate is selected by the PHY rate adaption modules (e.g. *AarfWifiManager*) to send the aggregate packet. Transmission errors can also be included by employing some modules (e.g. *NistErrorRateModel*). Then the aggregate packet is sent through the *WifiChannel* module which is set by the *WifiPhy* module. Having finished the transmission, it waits for the ACK frame. If it does not receive the ACK frame, the frame is re-transmitted by the *MacRxMiddle* module if required. If it receives the ACK, the transmission is successful. This is the basic transmission aggregate packet process. We just selected the most suitable modules (e.g. *AarfWifiManager, NistErrorRateModel*) to implement the algorithm in our simulations.

We also modified some of these modules to implement some new functions which will be described further in chapter 4.



Send Packet
Receive Packet
Packet Generator (OnOffApplication)
Packet Receiver (PacketSink)
WifiNetDevice
WifiNetDevice
StaWifiMAC (WifiMAcQueue)
StaWifiMAC
DcfManager
EdcaTxopN (MsduStandardAggregator)
MAcRxMiddle
Listener
MacLow (AarfWifiManager)
MacLow
Listener
WifiPhy (NistErrorRateModel)
WifiPhy
WifiChannel

Figure 2-9: Basic simulation of packet aggregation in ns-3.

2.7 Packet Sniffers

A packet sniffer is a program running on a network attached device that passively receives frames passing through the device's network adapter [ARC02]. A packet sniffer is also known as a Network Analyzer or Protocol Analyzer or Wireless Sniffer. The packet sniffer can monitor all data transmitted on the network and save it for analysis later. The packet sniffer can be used as an administrative tool to monitor and troubleshoot network traffic [AGS03]. Figure 2-10 shows a typical packet sniffer program running in a wireless network.

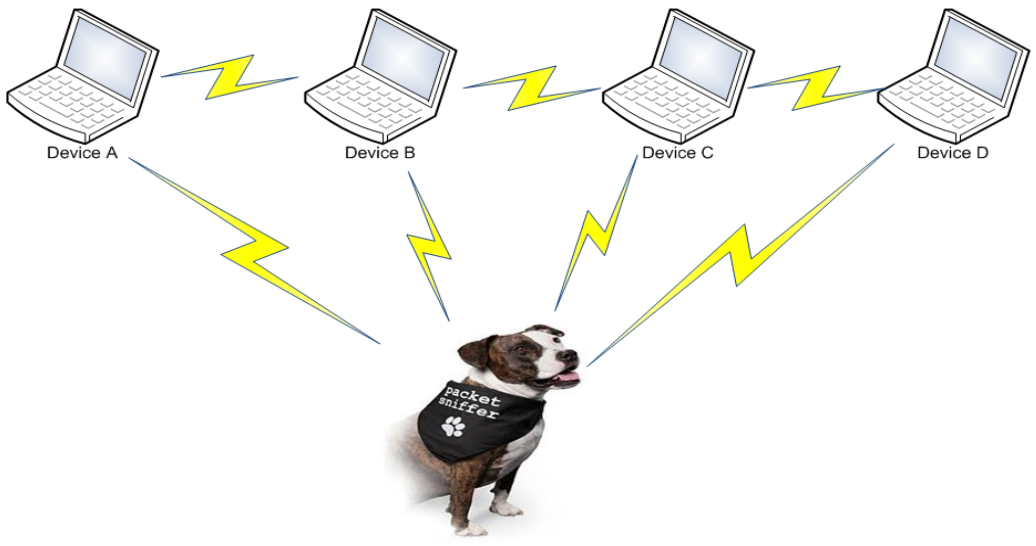
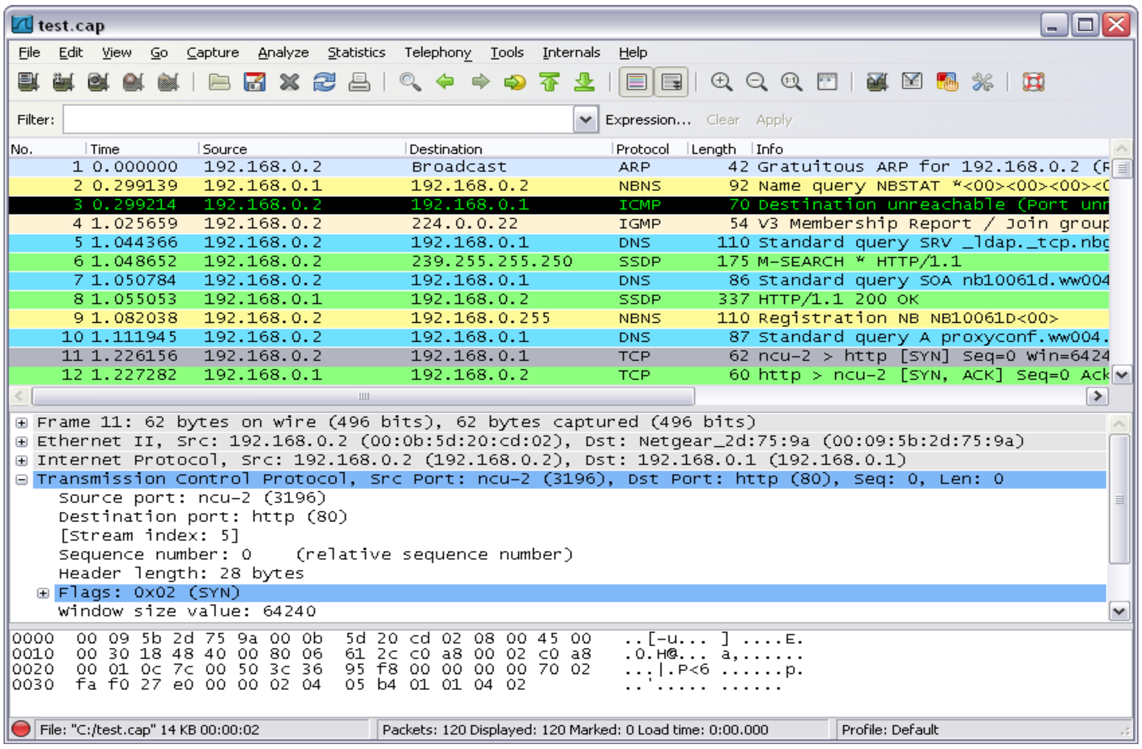


Figure 2-10: The operation of a packet sniffer application in a wireless network environment.

There are four devices (A, B, C, and D) in the wireless network and the device running a packet sniffer programmer listens to the data which arrives at the Network Interface Card (NIC). Usually, the NIC works in two modes: Non promiscuous mode (normal mode) and promiscuous (or monitor) mode. In a normal device, when a packet is received by a NIC, it first compares the MAC address of the packet to its own. If it matches, the NIC accepts the packet otherwise it ignores the packet. So in order to capture packets, the NIC has to be set in the promiscuous mode to receive all packets even those are not intended for it. There are a number of packet sniffer software applications available such as *Wireshark*, *tcpdump*, *snoop* etc.

The *Wireshark* packet sniffer application is one of the most widely used. *Wireshark* is a free and open-source packet analyzer and the latest stable version is *Wireshark* 1.8.6 [Wir13]. It is used for network troubleshooting, analysis, software and communications protocol development and education. *Wireshark* is a cross-platform application using a file format (i.e. pcap) to save the captured packets and it runs on various Unix-like operating systems including Linux, MacOS, BSD, Solaris, and Microsoft

Windows[ThW12]. The *Wireshark* not only captures network packets but also displays the captured packet data in a detailed format where an example is shown in Figure 2-11. In this thesis, the simulation is based on 16 traffic trace files which were captured by the *wireshark* from a number of different live Wi-Fi hotspot networks at different times and locations and whose details will be described in chapter 4.


test.cap
File Edit View Go Capture Analyze Statistics Telephony Tools Internals Help
Filter:
Expression... Clear Apply

```
No.   Time      Source        Destination      Protocol Length Info
    1 0.000000  192.168.0.2   Broadcast        ARP         42 Gratuitous ARP for 192.168.0.2 (R
    2 0.299139  192.168.0.1   192.168.0.2      NBNS        92 Name query NBSTAT *<00><00><00><0
    3 0.299214  192.168.0.2   192.168.0.1      ICMP        70 Destination unreachable (Port unr
    4 1.025659  192.168.0.2   224.0.0.22       IGMP        54 V3 Membership Report / Join group
    5 1.044366  192.168.0.2   192.168.0.1      DNS        110 Standard query SRV _ldap._tcp.nbc
    6 1.048652  192.168.0.2   239.255.255.250  SSDP       175 M-SEARCH * HTTP/1.1
    7 1.050784  192.168.0.2   192.168.0.1      DNS         86 Standard query SOA nb10061d.ww004
    8 1.055053  192.168.0.1   192.168.0.2      SSDP       337 HTTP/1.1 200 OK
    9 1.082038  192.168.0.2   192.168.0.255    NBNS       110 Registration NB NB10061D<00>
   10 1.111945  192.168.0.2   192.168.0.1      DNS         87 Standard query A proxyconf.ww004.
   11 1.226156  192.168.0.2   192.168.0.1      TCP         62 ncu-2 > http [SYN] Seq=0 Win=6424
   12 1.227282  192.168.0.1   192.168.0.2      TCP         60 http > ncu-2 [SYN, ACK] Seq=0 Ack

⊞ Frame 11: 62 bytes on wire (496 bits), 62 bytes captured (496 bits)
⊞ Ethernet II, Src: 192.168.0.2 (00:0b:5d:20:cd:02), Dst: Netgear_2d:75:9a (00:09:5b:2d:75:9a)
⊞ Internet Protocol, Src: 192.168.0.2 (192.168.0.2), Dst: 192.168.0.1 (192.168.0.1)
⊟ Transmission Control Protocol, Src Port: ncu-2 (3196), Dst Port: http (80), Seq: 0, Len: 0
    Source port: ncu-2 (3196)
    Destination port: http (80)
    [Stream index: 5]
    Sequence number: 0    (relative sequence number)
    Header length: 28 bytes
  ⊞ Flags: 0x02 (SYN)
    Window size value: 64240

0000  00 09 5b 2d 75 9a 00 0b  5d 20 cd 02 08 00 45 00   ..[-u... ] ....E.
0010  00 30 18 48 40 00 80 06  61 2c c0 a8 00 02 c0 a8   .0.H@... a,......
0020  00 01 0c 7c 00 50 3c 36  95 f8 00 00 00 00 70 02   ...|.P<6 ......p.
0030  fa f0 27 e0 00 00 02 04  05 b4 01 01 04 02         ..'..... ......

File: "C:/test.cap" 14 KB 00:00:02   Packets: 120 Displayed: 120 Marked: 0 Load time: 0:00.000   Profile: Default
```

Figure 2-11: An example of how *Wireshark* captures packets and parses their contents.

2.8 Chapter Summary

In this chapter, we have introduced the development and general concepts of the IEEE 802.11 WLANs including the structure of MAC frames, transmission errors and PHY rate adaption mechanisms in WLANs, network simulators and packet sniffer applications. The architecture of the wireless network was presented which is the main network topology used to implement the proposed AAM algorithm. A number of the IEEE 802.11 standards and the main components of WLANs were discussed and the typical network topology was shown. In particular, the IEEE 802.11n and IEEE 802.11ac standards were introduced which support the packet aggregation algorithm and proposed two packet aggregation algorithms (i.e. A-MSDU and A-MPDU) whose details will be described in the next chapter. The MAC mechanism of the IEEE 802.11 standard was presented and the details of how to access the medium, inter-frame times and the back-off contention scheme were also introduced. The impact of transmission errors and various PHY rate adaption mechanisms were discussed.

Network simulation tools are important tools to research the performance of network algorithms, protocols and environments. There are currently a number of free open source simulation tools available for researchers. One of the network simulation tools is the ns-3 which is still under developing and has a number of advantages. The ns-3 is employed in this thesis to implement and simulate the operation of the proposed AAM algorithm. The packet sniffer was presented in this chapter and the *Wireshark* software application was also introduced.

Chapter 3
Review of Packet Aggregation Algorithms

In this chapter a number of packet aggregation algorithms proposed for wireless networks will be presented. In particular we will attempt to describe what research is being carried out in the development of packet aggregation algorithms targeted at wireless networks. As will be shown, current approaches failed to achieve the goal of realizing an optimal trade-off between maximizing throughput and minimizing delay which is concerned with improving maximum throughput at the cost of the least delay increase. The majority of current researchers are only concerned with optimizing a single metric algorithm which attempts to achieve the goal of either maximizing throughput or minimizing delay. For these algorithms, they don't take account of the varying nature of mixed traffic loads. Some algorithms can achieve large throughputs at the expense of large delays while others achieve the goal of minimizing delay associated with small throughputs. Throughput and delay are the two most important performance metrics used to analyze a packet aggregation algorithm.

3.1 Throughput and Delay

3.1.1 Throughput

In modern communication networks such as WLANs or Ethernet networks, the throughput or network throughput is the average rate at which data is successfully transmitted through a communications channel [Rap02]. In general, throughput is measured in bits per second (bps) or it can sometimes be measured in packets per second (pps). In this thesis the throughput is defined as *the average payload in bits successfully transmitted in unit time from the source node to the destination node in a wireless network.*

The maximum throughput of a network is important for both user and system designer as it is essentially synonymous with the capacity of the network. The maximum throughput can be defined in a number of different ways such as the maximum achievable throughput, the peak measured throughput or the maximum sustained throughput. In this thesis, we define the *maximum average throughput as the average throughput under saturation conditions in wireless networks.*

For the IEEE 802.11 wireless networks, Xiao and Rosdahl [XiR02] [XiR03] showed that a throughput upper limit (TUL) exists. The TUL of wireless network is defined in [XiR02] and the authors assumed a wireless network where one sender and one receiver operate in the DCF mode. The sender always has packets to be transmitted and each packet has the same size. This throughput is determined at the Link Layer Control. It is assumed that there are no transmission errors present in order to emphasize the impact of the overheads which includes the PLCP (Physical Layer Convergence Protocol) preamble and the PLCP header. The TUL is given by the following equation:

$$TUL = \frac{8 * L_{Data}}{2 * T_p + 2 * T_{PHY} + 2 * \tau + T_{DIFS} + T_{SIFS} + \frac{CW_{min} * T_{slot}}{2}} \cdots\cdots (3.1)$$

Where L_{Data} denotes the payload of the packet in bytes, T_p denotes the transmission time of the PLCP preamble, T_{PHY}denotes the transmission time of the PLCP header, T_{DIFS} is the time of DIFS, T_{SIFS} is the time of SIFS, CW_{min} is the minimum size of contention window in unit of a slot time, T_{slot} is the time duration of a slot time and τ is the propagation delay that is the propagation time between the nodes by the radio signal which usually can be ignored as its value is negligible compared to that of other times. The distance between wireless nodes is less than 50 meters, so the time of propagation is less than 0.2 μs. The values of the parameters are shown in Table 3-1 for the IEEE 802.11a, IEEE 802.11b, IEEE 802.11n and IEEE 802.11ac standards.

Table 3-1: The PHY parameter values for some of the IEEE 802.11 standards

Parameter	**IEEE 802.11 a**	**IEEE 802.11 b**	**IEEE 802.11 n**	**IEEE 802.11 ac**	**Comment**
$Tslot$	9 µs	20 µs	9 µs	9 µs	slot time
T_{DIFS}	34 µs	50 µs	34 µs	34 µs	DIFS duration time
CW_{min}	15	31	15	15	Minimum contention window size in unit of slot time
Tp	16 µs	144 µs	16 µs	16 µs	PLCP preamble duration
T_{PHY}	4 µs	48 µs	4 µs	4 µs	PLCP header duration
T_{SIFS}	16 µs	10 µs	16 µs	16 µs	SIFS duration time
τ	1 µs	1 µs	1 µs	1 µs	Propagation delay

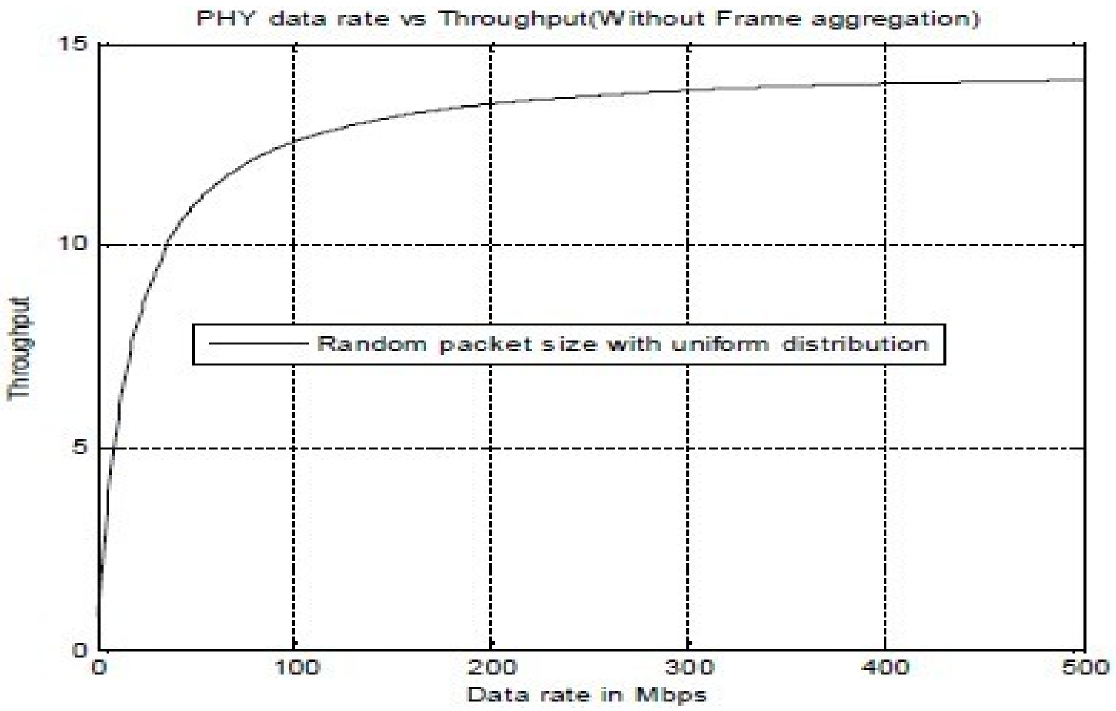


Figure 3-1: The throughput against data rate without packet aggregation [Hud09].

The TUL is defined as the maximum throughput when the PHY rate increases indefinitely. As shown in Figure 3-1, for a certain PHY rate, the throughput is almost independent of data rate when packets have a fixed size [Hud09]. This is due to the large amount of overhead added to every packet. This suggests that there are two methods to improve the throughput by increasing the average size of payload or specifically by reducing the ratio of header size to payload size in a frame. Both methods can be achieved by using a packet aggregation algorithm which is why packet aggregation algorithms have become so popular. For example, the latest IEEE 802.11n standard and the IEEE 802.11ac standard (which is still under development) support the use of packet aggregation. There are two packet aggregation algorithms proposed in the IEEE 802.11n standard namely Aggregate MAC Service Data Unit (A-MSDU) and Aggregate MAC Protocol Data Unit (A-MPDU) which will be described in section 3.3.

3.1.2 Delay

Network delay in wireless networks specifies how long it takes for a data packet to travel across the wireless network from one node to another. It is an important performance characteristic of an IEEE 802.11 wireless network. The network delay is usually divided into several parts depending on the location of the specific pair of communicating nodes: processing delay, queuing delay, transmission delay and propagation delay. In this thesis, the delay is defined as *the average time to successfully transmit a packet from the MAC layer of the source node to the MAC layer of the destination node in wireless networks* and *the minimum average delay is defined as the average delay under saturation conditions in wireless networks.*

There is a certain minimum level of delay which will be experienced due to the time it takes to transmit a packet serially through a link [ZNN10]. The delay lower limit (DLL) of the DCF model in the IEEE 802.11 wireless network is derived in [XiR02] [QCS02]. To derive the DLL, the system needs to be operated under a best-case scenario: (i) The channel is an ideal channel without transmission errors present; (ii) At any transmission cycle, there is one and only one active station which always has packets to send and other stations can only receive packets and provide acknowledgements (ACKs). The DLL is given by equation (3.2) [XiR02].

$$\mathrm{DLL} = T_P + T_{PHY} + \tau + T_{DIFS} + \frac{CW_{min} * T_{slot}}{2} \text{ .... (3.2)}$$

Where all of the parameters have been defined as in equation (3.1) and their values are given in Table 3-1. This DLL does not consider the queuing time and waiting time at the MAC.

3.1.3 Discussion of Throughput and Delay

So far, we have introduced the concepts of the throughput and the delay, and also defined how they are used in the analysis of the proposed AAM algorithm. In WLANs, an upper limit on the throughput exists and is given by [XiR02] [XiR03] [QCS02]. Due to the protocol overhead associated with the transmission packet, the throughput cannot be further increased without reducing the protocol overheads even though the data rate increases indefinitely. A packet aggregation algorithm is used to reduce the average protocol overhead and can significantly improve the throughput. Also a lower limit on the delay in wireless networks exists and is defined by [XiR02] [QCS02] where the delay of both the queuing time and waiting time in the MAC are not considered. However, the time spent waiting for more packets to arrive is the main delay for the packet aggregation algorithm [TYH10]. Therefore, there is a trade-off in terms of an increased throughput and an increased delay when employing a packet aggregation algorithm. This trade-off will be discussed in the next section.

3.2 Trade-off between Throughput and Delay

As discussed in the last section, throughput is the key metric that packet aggregation algorithms try to improve. Reducing the protocol overheads is an important approach to improving the throughput as an upper limit on throughput exists. Studies have shown that the packet aggregation can significantly improve the throughput. For example, adopting a packet aggregation algorithm in [KGL06] increases by a factor of 7 the number of calls that can be supported for VoIP applications; [JDB10] improves the throughput by a factor of 2.5 times; and [MaA12] achieves a 200% improvement in the throughput.

However, the delay can also be increased as the throughput increases when packet aggregation algorithms are employed. The existence of a delay lower limit has been defined and demonstrated in wireless networks by [XiR02]. But the delay lower limit does not consider the queuing delay and waiting delay both of which are the dominant delay components that are increased when using a packet aggregation algorithm. We will describe the delay associated with a packet aggregation algorithm.

3.2.1 Delay Associated with a Packet Aggregation

Delay is the key cost associated with the use of a packet aggregation algorithm to improve the throughput. There have been many studies conducting regarding the delay associated with packet aggregation in wireless networks.

In [Lin06], a model was proposed to calculate the packet aggregation delay. This model studies error-prone channels using A-MPDU and A-MSDU packet aggregation algorithms both of which are popular packet aggregation algorithms defined in the IEEE 802.11n standard and will be described in the next sections. In this model, the network saturation throughput is defined based on a wireless network where *M* station nodes use the RTS/CTS scheme to access the same channel. The network saturation throughput (*S*) is given by equation (3.3).

$$S = \frac{E_p}{E_t} \qquad (3.3)$$

Where, E_p is the successfully transmitted payload size (in bits) in unit time, E_t is the expected length of the time slot, which is defined as the time when a station starts to check the channel state (i.e. idle or busy) for transmitting a packet until it receives an ACK of the packet. As there are M station nodes competing for transmission, the average access delay is given by equation (3.4)

$$D = M * \frac{L_p}{S} \qquad (3.4)$$

Where L_p is the aggregate packet size (in bits). The simulation results show that the delay increases as the BER increases for the packet aggregation algorithm. This delay only considers the transmission delay and does not include other components such as queuing time.

In [TYH10], a delay calculation is proposed which calculates the delay of a packet's life time and is the first work focusing on packet delay of packet aggregation algorithm. The average delay (D_a) of a packet includes two parts: the queuing delay (D_Q) and the transmission delay (D_T) which equals the value of delay (D) in the equation (3.4).

$$D_a = D_Q + D_T \qquad (3.5)$$

$$D_T = D \qquad (3.6)$$

If the length of MAC queue is L_Q in bytes where it assumes that the queue is always full, and the aggregate packet size is L_p bytes which are transmitted at each period, the queuing delay is shown in (3.7)

$$D_Q = D_T * \left(\frac{L_Q}{L_P}\right) \qquad (3.7)$$

From the equations (3.4), (3.5), (3.6) and (3.7), the average delay (D_a) is given by (3.8)

$$D_a = M * \left(1 + \frac{L_Q}{L_p}\right) * \frac{L_P}{S} \quad \text{………………..…………..} \quad (3.8)$$

This results show that the average delay is affected by the BER, the aggregate packet size and the number of competing nodes. As the BER and the number of competing nodes cannot be controlled, one way that the delay can be changed is to adjust the aggregate packet size.

The packet aggregation algorithm proposed in [ZIF08] is based on the Packet Reservation Multiple Access (PRMA) protocol. In this model, the PRMA data transmission is based on the Markov chain model and it assumes that the aggregation process starts after the arrival of a certain number of packets into the buffer. In [ZIF08], the packet aggregation delay is the delay experienced by the first arrival packet (i.e. the time from when it arrives at a terminal's buffer until the time it can start to be transmitted). The delay consists of two parts: packet aggregation delay and channel access delay. The packet aggregation delay is the queuing time of the first arrival packet waiting in the buffer, so the mean aggregation delay is given based on the Markov chain:

$$D_{agg} = \begin{cases} 0 & n = 1 \\ \alpha \left\lceil \frac{n-1}{\lambda \alpha} \right\rceil & n \geq 2 \end{cases} \quad \text{……………………..…} \quad (3.9)$$

Where α denotes the average packet transmission period, λ denotes the packet arrival rate. As discussed in [ZIF08], a number of conclusions have been drawn: (i) The aggregate packet has a long packet aggregation delay when the packet arrival rate is low, this means that the first arrival packet could wait a longer time in the buffer for the size of packets in the buffer to exceed the specified target aggregate packet size; (ii) With a fixed arrival rate, the aggregate packet has a higher aggregation delay with a bigger target aggregate packet size; (iii) The proposed algorithm cannot achieve throughput gains by packet aggregation algorithm at the expense of high delay when the packet arrival rate is low; (iv) There is a trade-off between throughput and delay when using packet aggregation. As the packet arrival rate is unpredictable, the packet aggregation delay is controlled by the target packet aggregation size.

3.2.2 Trade-Off between Throughput and Delay

It is clear that packet aggregation algorithms can significantly improve the throughput but any improvement will also have an associated cost in terms of a delay increase. As a delay lower limit and a throughput upper limit exist in the IEEE 802.11 wireless networks, most of the proposed packet aggregation algorithms try to asymptotically approach these limits.

Most of the proposed packet aggregation algorithms attempt to optimize a single metric, i.e. either to maximize throughput or to minimize delay. For example, for time sensitive applications (e.g. VoIP, video streams), the packet aggregation algorithms focus on the delay, such as in [TYH10] [KuD06]. For some other applications, such as E-mail, they can tolerate a relatively longer delay than time sensitive applications. They focus on the improvement in throughput by using packet aggregation algorithms. However, for a significant number of wireless networks, the traffic load is mixed containing many different types of applications that can have very different network performance requirements. It requires that the packet aggregation algorithm should be adaptive to achieve an optimal trade-off between throughput and delay. The adaptive algorithm should be capable in improving the maximum throughput with the least cost in terms of a delay increase.

An aggregation algorithm has a superior performance over other algorithms in terms of the trade-off between maximizing throughput and minimizing delay if it can deliver a throughput greater than that of other algorithms for a given delay increase. This is illustrated in Figure 3-2 where there are two packet aggregation algorithms A and B operating under the same conditions. The algorithm A has the better performance in terms of the trade-off between maximizing throughput and minimizing delay than that of the algorithm B as the algorithm A has a throughput greater than that of the algorithm B for a given delay increase.

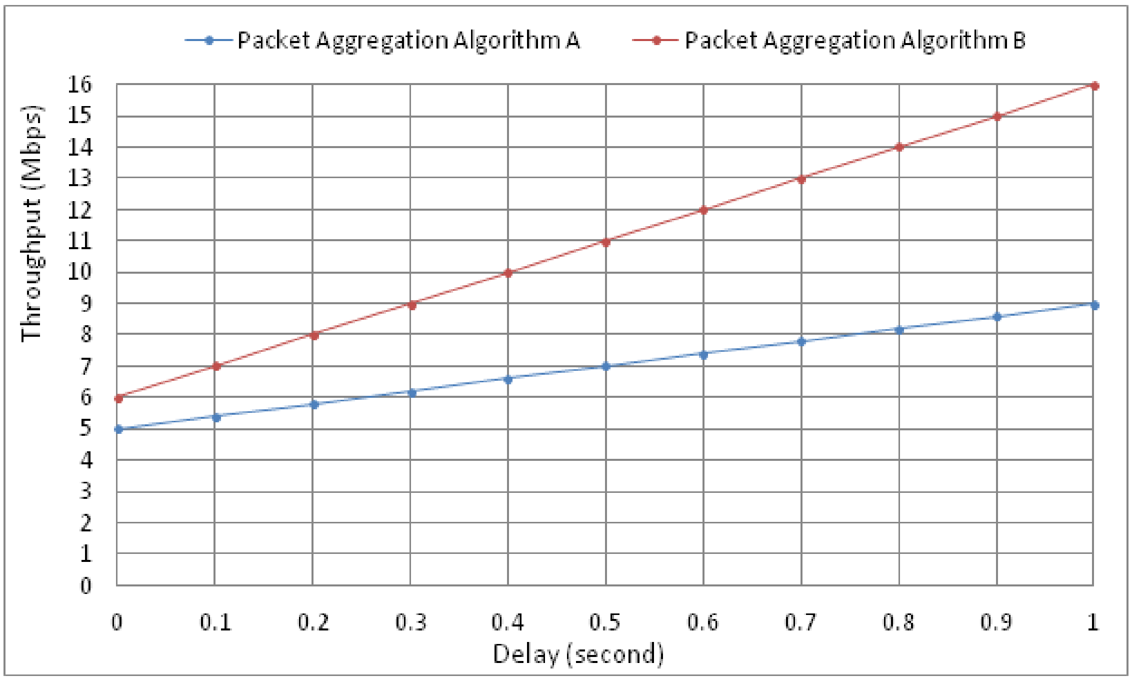


Figure 3-2: An example of trade-off between throughput and delay for different aggregation algorithms.

In [GuL12], a study of the effect of packet aggregation on video streaming performance on an experimental IEEE 802.11n test-bed was performed. They found that the video application naturally takes advantage of packet aggregation in both single- and multi-stream environments. The packet aggregation algorithm can severely impact on the average delay and quality of a video stream through limiting the aggregation packet size in the IEEE 802.11n wireless networks. This algorithm tries to minimize the delay by reducing the throughput increase caused by limiting the target aggregate packet size but it does not consider how to increase the throughput.

In [Wla12], the authors proposed a new model to obtain the optimal packet aggregation size with regard to the delay constraints of nodes. This model specifies a parameter for the time limit of a node to access the channel which means that the node cannot wait a longer time than this to access the channel. The simulation results show that the delay increases with an increase in the number of nodes attempting to access the channel for the normal IEEE 802.11n scheme and the delay is less than the value of the specified parameter. The idea of this scheme is different to that of the previous schemes. This scheme attempts to fix the upper limit of delay to control the average packet delay, while the previous scheme (i.e. [GuL12]) attempts to limit the packet aggregation size to achieve the goal. In other words, the proposed algorithm in [Wla12] bounds the delay increase at the cost of limiting the throughput increase. In particular, if there are a large number of nodes attempting to transmit in a wireless network, the throughput is not increased and may even be decreased under the proposed scheme compared to that of non-aggregation.

In [ZIF08], the authors found that the packet arrival rate is an important factor that affects the delay: the higher the packet arrival rate the shorter the waiting time in the buffer and the smaller the delay. However, in practice the packet arrival rate is essentially random and uncontrollable. So there is a need to develop adaptive packet aggregation algorithms that are better suited to the variations in the packet arrival rate.

In this thesis, we investigate the performance of the proposed AAM algorithm in terms of the aggregation trade-off where the *aggregation trade-off is defined as the maximum average throughput with the minimum average delay in wireless networks.*

3.2.3 Discussion of Trade-off between Throughput and Delay

In this section, we discuss the trade-off between the throughput and the delay when employing a packet aggregation algorithm in wireless networks. Generally, the throughput is improved at the expense of a delay increase. In modern networks the traffic load is a mix of different types of applications which can have very different requirements in terms of network performances. Therefore there is a requirement to realize some form of an optimal trade-off between maximizing throughput and minimizing delay.

There are a number of factors that impact the throughput and the delay, such as target aggregate packet size, channel noise, contention for access etc. But there is one factor that is the target aggregate packet size which can be controlled to manage both throughout and delay. The others are essentially unpredictable and uncontrollable, such as the packet arrival rate and transmission errors. Generally, the larger the size of the aggregate packet, the higher the throughput but also the larger the delay as it may wait longer for packets to arrive. Therefore many researchers have attempted to achieve the goal of finding the optimal packet aggregation size that minimizes the delay and maximizes the throughput.

As it is difficult to achieve the optimal trade-off for mixed traffic loads by using packet aggregation algorithm, some studies are only focused on some special applications, such as VoIP or video streaming. In this thesis, the proposed AAM algorithm is used to achieve the best aggregation trade-off in terms of achieving the maximum average throughput with the minimum average delay for different traffic loads in wireless networks.

3.3 Packet Aggregation Algorithms

It is clear that increasing the data rate or changing the modulation scheme can improve the performance in terms of throughput in IEEE 802.11 WLANs. However, due to the IEEE 802.11 protocol overheads (e.g. MAC header), a throughput upper limit (TUL) exists which was shown by Xiao and Rosdahl [XiR02] [XiR03]. The MAC is inefficient due to the MAC protocol headers, back-off time, inter-frame spacing and ACKs, and this inefficiency is the most pronounced when the data rate is high or the payload is small. To achieve higher throughputs it is necessary to reduce protocol overheads particularly for small size packets. The protocol overhead is the key factor for small sized packet to lower the MAC efficiency which is clearly demonstrated by Dionysius et al. [SNC08]. The idea of packet aggregation algorithm was proposed by Shaffer [SWC99] in 1999 and Gopalakrishna [Gop03] proposed their packet aggregation method in 2003. A lot of research has been conducted to show how packet aggregation can improve the throughput in wireless networks. D. Skordoulis [SNC08] has demonstrated the influence of aggregation, block acknowledgement on the throughput of the IEEE 802.11 WLANs. In [BMS09], the study shows that the packet aggregation algorithm has significantly improved the throughput for the various application data traffic in an IEEE 802.11 experiment test-bed.

Packet aggregation is shown in Figure 3-3 where there are three packets aggregated into a single packet at the sender. This reduces the number of MAC headers required from three to just one header. The two MAC headers do not need to be transmitted which represents a saving of two MAC overheads.

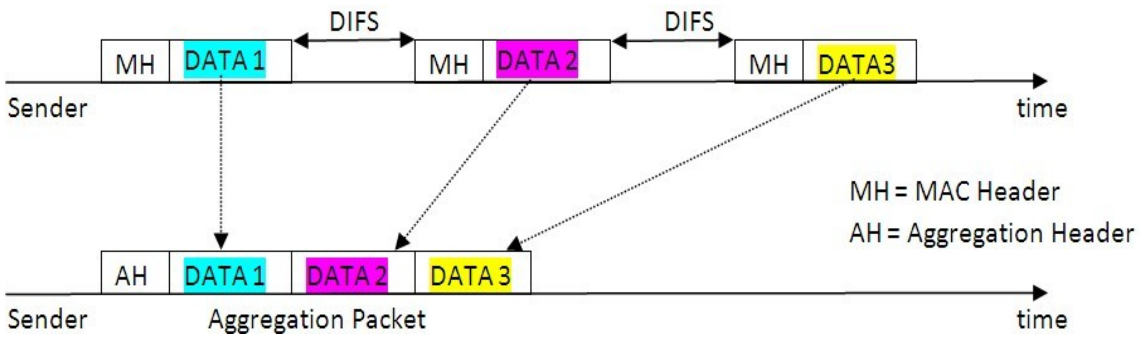


Figure 3-3: The aggregation process for packet aggregation algorithm in wireless networks.

Packet aggregation algorithms can be divided into different classes based on the different strategies employed. For example, in [HeC04], the packet aggregation may be performed at different granularities: aggregating all the packets (full aggregation), aggregating packets from the same traffic class (per-class aggregation) and aggregating packets from the same flow (per-flow aggregation). While, there are eight ways to classify them as proposed in [Xia05]: (i) distributed vs. centrally controlled; (ii) ad-hoc vs. infrastructure; (iii) uplink vs. downlink; (iv) single-destination vs. multi-destination; (v) PHY-level vs. MAC-level; (vi) single-rate vs. multi-rate; (vii) immediate ACK vs. delay ACK; (viii) no spacing vs. SIFS spacing.

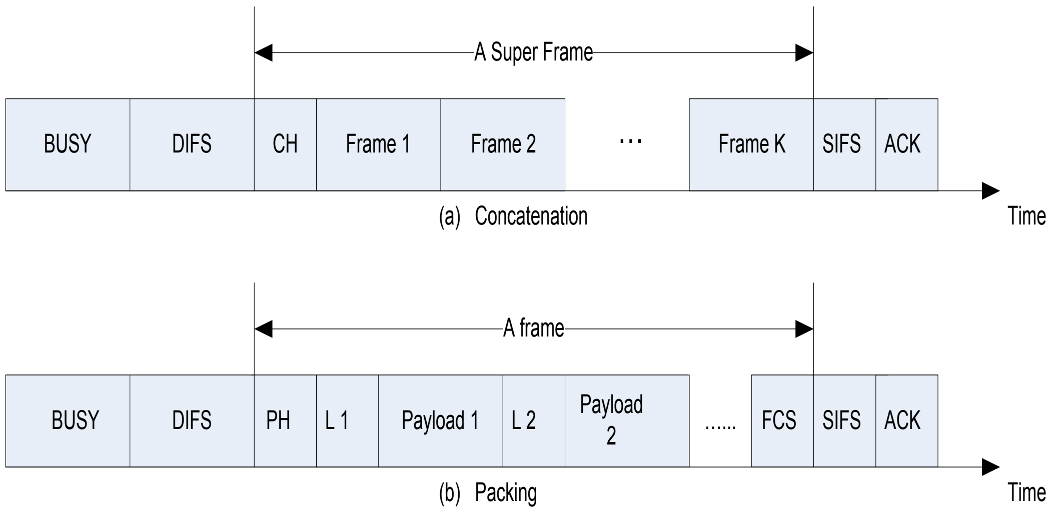
In this thesis, the different aggregation algorithms will be divided into 4 categories based upon the aggregation discipline and selection strategy employed. The aggregation packet selection strategy describes the way in which the packets are selected for assembling. The two categories here are first-in first-out (FIFO) and Non-FIFO. The FIFO discipline selects the packets based on the time of arrival into the buffer which is also the benchmark algorithm used here to compare with the proposed AAM algorithm. Non-FIFO uses other methods to select the packets.

The aggregation discipline can either be fixed or adaptive depending on whether the algorithm parameters are fixed or are dynamically adjusted in response to variations in the network conditions. Combining these two approaches results in 4 categories for aggregation algorithms: (1) Fixed with FIFO selection strategy (FF); (2) Fixed with Non-FIFO selection strategy (FNF); (3) Adaptive with FIFO selection strategy (AF); (4) Adaptive with Non-FIFO selection strategy (ANF).

3.3.1 Fixed with FIFO Packet Aggregation Algorithms (FF)

The FF packet aggregation algorithm does not automatically adjust the parameters of the algorithm to adapt to the variations in network conditions and uses the FIFO selection strategy to select packets for aggregation. These algorithms were proposed at the early stages of the packet aggregation development.

In [KoG03], an algorithm is presented that has been developed for multi VoIP streams and it shows the relationship between the number of VoIP calls and output link rate, and the network performance is measured in terms of teletraffic parameters.



(CH: Concatenation Header; PH: Packing Header; L1~L2: Length of the following payload.)

Figure 3-4: The format of concatenation and packing.

Concatenation and packing [Xia04] shown in Figure 3-4, as well as aggregation schemes [YCJ04] are early aggregation attempts. Concatenation shown in Figure 3-4 (a) is the process of concatenating multiple frames into a large frame. Packing shown in Figure 3-4 (b) is the process of combining multiple data units from a higher layer into a single MAC protocol data unit. For the packing scheme, it involves combining all concatenated frames into a larger frame with one header instead of many concatenated frames and it is more efficient than the concatenation scheme at the expense of complexity and delay of combining and decomposing frames. However, the disadvantage of the algorithm is that the aggregation headers are considered too large for small payloads and the behavior in noisy channels has not been addressed [SOS11].

In [YCJ04], the performance of frame aggregation is evaluated by both numerical analysis and experimental measurements obtained from a test-bed. According to the measurement results, the frame aggregation can improve the throughput performance of the IEEE 802.11b WLAN by 2 to 3 Mbps, when multiple frames are aggregated. However, it does not consider the delay.

There has been renewed interested in their applications for wireless networks [KGL06] [RLI06], where fairness and inefficiency issues in the IEEE 802.11-based wireless systems are examined. In [KGL06], a distributed packet aggregation algorithm is proposed for Voice over IP (VoIP) used in multi-hop wireless networks. The experimental results demonstrate that the number of calls supported increases to 8 for the proposed aggregation algorithm compared to 1 in the case of no aggregation. The disadvantage of the proposed algorithm is that it only operates for a single application, i.e. VoIP. VoIP is one of the most important applications to be researched in the development of packet aggregation algorithms as the average size of a VoIP packet is small. However it imposes a constraint on the maximum delay allowed, typically less than 150 ms [ITU03]. Normally, the packet aggregation algorithms for the time sensitive applications (e.g. VoIP or video streaming) are focused on the increased delay resulting from an improvement the throughput.

In [Hud09], a scheme called the frame aggregation and block acknowledge (FABA) is proposed which is shown to be capable of providing a throughput that is sufficient for multimedia applications, even at rates of over 100 Mbps. FABA combines the packet aggregation mechanism and the block acknowledgement mechanism to improve the throughput. In this scheme, a number of aggregate packets will be transmitted from the sender in a single back-off period and a special ACK frame will be sent to the sender from the receiver. However, it is based on a new MAC scheme designed by the author and not easily implemented.

In [LNM09] [MaE07], the authors proposed a scheme called Aggregation and Fragment Retransmission (AFR) scheme to aggregate as many packets as possible into a large frame. This large frame is, in turn, fragmented into smaller fragments before being transmitted. If transmission errors occur during the transmission, only the corrupted fragments of the large frame are re-transmitted. The simulation results show that AFR achieves between 2.5 and 3 times the throughput of DCF over a range of network conditions for TCP traffic. However, a new data format is developed and an ACK frame which has a new format is proposed by the author and it should be received for every fragment transmitted by the source node which significantly affects the throughput even if some packets are divided into fragments. And the new format ACK is larger than the typical ACK frame. This algorithm has some disadvantages as it can only improve the throughput in the high packet rate wireless networks and cannot be easily implemented as a new format data and ACK are required.

In [JDB10], the authors have proposed three packet aggregation algorithms to improve the throughput for VoIP applications. Here, the simulation results show that these algorithms have better performances than those without packet aggregation. For the proposed scheme, the supported VoIP calls increased to 80 from 33. However, it just considers improving the system efficiency of uplink VoIP packet transmission and assumes the AP is equipped with a special smart antenna with a beam width of 90^0 with gain. Furthermore, this scheme produces the aggregation packets at a fixed rate.

3.3.2 Fixed with Non-FIFO Packet Aggregation Algorithms (FNF)

The FNF packet aggregation algorithm is the algorithm that does not automatically adjust the parameters of the algorithm to adapt to variations in the network conditions but employs a Non-FIFO selection strategy to select packets for aggregation.

In [EEV06], the authors describe a packet aggregation algorithm that can increase the throughput of WLAN for voice communication by decreasing the overhead of the back-off process at the beginning of each packet transmission. It also implements the packet aggregation on the IEEE 802.11 WLANs and an analysis of the results shows that it can considerably improve the performance of VoIP. The algorithm picks the VoIP packet from the buffer (based on the packet size) assuming that the VoIP packet size is smaller than others. So it is easy to miss the VoIP packet and also easy to pick the wrong packet which is not a VoIP packet but has a similar size.

3.3.3 Adaptive with FIFO Packet Aggregation Algorithms (AF)

The AF aggregation algorithm has parameters that can be automatically adjusted to follow the variations in the network conditions and employs the FIFO selection strategy. These algorithms have been proposed to satisfy the variations in the application's demands and under changing network conditions.

An adaptive algorithm proposed by Yuxia and Vincent [Lin06], defines an optimal packet size based on the A-MSDU aggregation scheme. The algorithm operates in three steps as follows: (i) the source station evaluates the channel BER before transmitting an aggregate packet; (ii) it calculates the optimal packet size for unidirectional and bi-directional transmissions respectively; (iii) it assembles the aggregate packet with a size as close as possible to the optimal packet size. The research demonstrated that the adaptive packet aggregation has a better performance in terms of throughput than that for both the fixed packet aggregation and randomized packet aggregation where the aggregate packet sizes are randomly distributed in a range.

In [ZIF08], the authors proposed a packet aggregation algorithm called PRMA (Packet Reservation Multiple Access) to improve throughput for data traffic. According to the proposed scheme, a generic Markov chain model is developed. It shows that the throughput increases as the packet arrival rate increases. When the arrival rate is low, the achievable throughput has no difference with or without packet aggregation. In [HLF09], a scheme based upon automatic repeat request (ARQ) is employed called aggregated selective repeat ARQ (ASR-ARQ) to improve the throughput based on the Markov chain. It confirms that the aggregate packet successful transmission probability increases as the BER decreases. The authors in [WeL11] proposed an adaptive scheme also based on the Markov chain to constrain the delay by adjusting the times of re-transmission due to collisions and transmission errors respectively. The proposed algorithm achieves a better performance in terms of average delay by limiting the re-transmission time than the normal fixed packet aggregation. However, the packet loss rate is high if a large number of stations try to transmit packets at the same time using the proposed algorithm.

The algorithm described in [KCK11] is a joint rate and fragment size adaption packet aggregation algorithm which is implemented within the context of the proposed algorithm AFR (Aggregation and Fragment Retransmission) in [LNM09] to improve the throughput. This scheme is based on the current estimated fragment error probability which can characterize channel quality without using explicit feedback information. But this estimation sometimes may not accurately characterize the channel quality as the channel quality can change rapidly. It also has the same disadvantage with [MaE07] as it needs a large buffer and an extra ACK. Generally, the feedback control scheme is a good approach to determine the channel quality. In this thesis, a feedback control scheme is also employed by the proposed AAM algorithm.

3.3.4 Adaptive with Non-FIFO Packet Aggregation Algorithms (ANF)

The ANF algorithm has parameters that are automatically adjusted to adapt to variations in the network conditions and uses a Non-FIFO selection strategy. These algorithms are adaptive and employ different selection strategies based on different characteristics to select the packets for aggregation, such as based on packet size, packet life time, priority etc.

In [LeP07], a scheme is proposed to manage the delay budget and control the packet aggregation which is based upon a rotating priority queue (RPQ) scheme [LWM96] at the cost of large number of queues as it requires queues for every traffic types. The proposed scheme uses a priority strategy to select the packets for assembling that is similar to [RMP08]. In [WaH08], the authors also find that the scheduling of packet aggregation is a knapsack problem which is a NP-hard problem and an algorithm is proposed called Largest Unit Urgency First (LUUF) to approximate the optimal solution. The largest unit urgency packet is selected at first to be aggregated. The analysis result shows that the total LUUF complexity can be reduced to *O(n)* from *O(nlog n)* in each cycle. But it requires that all users have the same QoS (Quality of Service) requirements.

An adaptive aggregation and differentiation scheme, in which a priority mechanism and scheduling is implemented at the top of the MAC, was proposed by Riggio [RMP08]. The priority selection strategy is employed in the proposed algorithm where the packets are pushed into 4 different aggregating buffer based on the different priorities. The test results on a Wi-Fi test-bed show that the proposed scheme can attain a large gain in the voice call capacity. The proposed algorithm can effectively differentiate services and improve the network scalability. The disadvantage of the scheme is that it needs a large memory for the pool of queues and only works for the packets which use tagging.

Another aggregation scheduler presented by Selvam and Srikanth [SeS10] adaptively estimates the deadline of packet transmission and selects the aggregation type based on the size of the aggregation buffer which has the smallest size packet. The results show that the different packet aggregation algorithms (e.g. A-MSDU, A-MPDU) have different advantages in different network environments to improve the throughput. For example, the A-MSDU algorithm is very effective under ideal channel conditions due to the reduced protocol overhead. However, in error-prone wireless networks it yields poor performance due to the lack of an individual FCS (Frame Check Sequence) for each sub-packet. On the other hand, A-MPDU is robust against transmission errors as the presence of individual CRC (Cycle Redundancy Check) per MPDU and the aggregated packet size can be up to 64 KB. But the receiver nodes have a large delay to reorder the large size packet. The proposed aggregation scheduler employs the selection strategy based on the packet size where the frames are saved in ascending order to wait for aggregation which is used as the comparison algorithm with the AAM algorithm.

In [MaA12], an aggregation scheme is proposed to improve the throughput. It first formulates the problem of optimal aggregation as being NP-Hard and then proposes two heuristics to solve the aggregation problem for multi-rate WLANs. The first heuristic is called Data Rate based Aggregation protocol (DRA) that divides packets in the MAC queue into different groups based on the data rate with which they are to be transmitted. DRA also achieves up to a 200% increase in the number of VoIP calls supported by a single IEEE 802.11g AP compared to using the Destination based Aggregation (DA), a state-of-the-art aggregation protocol. DA combines these packets that have the same destination address and then sends them in a single aggregate packet to the destination node [CDK07]. The second heuristic is called Data Rate based Aggregation with Selective Demotion (DRASD) which enables cross data rate aggregation and allows limited cross data rate aggregation, and it shows that selective packet demotion could be used to reduce WLAN delays in certain cases. In [MaA12], the algorithm selects the packets based on the life time and the priority. The first packet selected is the packet which has the smallest life time, or has the highest priority if two of more packets have the same life time. It is shown that selectively demoting packets can further improve performance. However, this algorithm may not accurately characterize the data rate as the channel conditions can change rapidly.

3.3.5 Transmission Errors and Packet Aggregation Algorithms

As discussed in chapter 2, transmission errors can have a detrimental impact on the performance of packet aggregation in wireless networks as corrupting a large size aggregated packet may waste a long period of channel time and leads to a lower MAC efficiency. Therefore some packet aggregation algorithms were proposed to reduce the detrimental impact on the performance of wireless networks.

[YWA04] experimentally studied the effect of packet size in an error-prone channel for the IEEE 802.11 DCF and concluded that there is an optimal packet size under a certain BER to achieve the maximum throughput with the saturated traffic. A model based on an optimal frame size adaptation algorithm was proposed to study the saturation throughput and delay performance in [Lin06] which was introduced previously. This performance for the proposed model was investigated under error-prone channels by using the A-MPDU and A-MSDU packet aggregation algorithms. However, it cannot always accurately characterize the channel BERs as the channel conditions can change rapidly. In [KSP12], the authors proposed the adaptive frame size estimation (FSE) depending on the channel condition which can improve the throughput for A-MSDU in the error-prone WLAN environments. In this thesis, the proposed AAM algorithm will be implemented in an error-prone WLAN.

3.3.6 Discussion of Packet Aggregation Algorithms

This survey provides a good insight into the different packet aggregation algorithms that have been developed for wireless networks. We have divided these proposed packet aggregation algorithm into 4 categories, FF, FNF, AF and ANF based on the aggregation discipline and the selection strategy. Some proposed packet aggregation algorithms whose goal is to ameliorate the detrimental impact on the performance of wireless networks are introduced.

It is shown that the adaptive packet aggregation algorithm has a better performance than the fixed packet aggregation, [Lin06] [WeL11], as the adaptive packet aggregation algorithm can adaptively adjust the parameters to suit different types of traffic loads. FIFO selection strategy is the most popular selection strategy [RLI06] [MaE07] [SOS11] [KCK11] [KSP12]. Some other Non-FIFO selection strategies, such as based on packet size, life time, priority, are used in some proposed packet aggregation algorithms. However, if the packet sizes are all similar, the selection strategy cannot significantly impact on the performance of packet aggregation [LYY09]. The different selection strategies have different advantages and disadvantages. For example, the biggest advantage of the FIFO selection strategy is a short waiting time in the queue and the disadvantage of the FIFO selection strategy is that it cannot always achieve the maximum throughput. The performance of the selection strategy of Smallest-Size First-Served (SSFS) is the opposite to that of the FIFO selection strategy. This means that the SSFS can aggregate a larger number of sub-packets in an aggregate packet but it needs longer waiting time in the queue. Table 3-2 highlights the contributions and disadvantages of some of the reported packet aggregation algorithms sorted by the aggregation discipline and the selection strategy. In this thesis, the A-MSDU [IEn09] is a typical FIFO algorithm used to compare with the AAM algorithm which will be described in the next section and the other comparison algorithm in [SeS10] used the typical SSFS algorithm will also be introduced in the next section as the Non-FIFO selection strategy algorithm.

Table 3-2: A comparison between some packet aggregation algorithms

Reference	Approach	Main Contribution	Main Disadvantage
[IEn09]	FF	Proposes two aggregation algorithms, A-MSDU and A-MPDU, which are widely used and employed by the IEEE 802.11ac standard. It defines the new format of the two packet aggregation algorithms.	Whole A-MSDU aggregation packets need to be re-transmitted when a bit is corrupted [KML12].
[LNM09]	FF	Aggregates as many packets as possible into a large packet and only the corrupted sub-packet needs to be re-transmitted. It is can significantly improve the throughput.	It is not easily implemented as a new format ACK and a new data format are used [KCK11].
[Hud09]	FF	Supports the multimedia applications and improves the throughput even up to 100Mbps	Based on a new MAC mechanism proposed by author and is not easily implemented [ZKH13].
[SOS10]	FF	Adjusts the header size so that it has a more significant impact for the small size packet than that of larger size packet and proposes algorithm on the small size packets.	Only works well for the small size application packets. [ArS12]
[EEV06]	FNF	Can considerably improve the performance of VoIP operating on IEEE 802.11 WLANs.	Easy to miss VoIP packets as assumption the VoIP packets sizes are smaller than other type packets sizes [LCB10].
[KCK11]	AF	Proposes a joint rate and fragment size adaption packet aggregation algorithm based on the current estimated fragment error probability to improve the throughput.	The estimation sometimes may not accurately characterize the channel quality and it needs a large buffer and an extra ACK [SaA12].
[Lin06]	AF	Defines the saturation throughput and delay on the A-MSDU aggregation scheme.	Cannot always accurately characterize the channel BERs as the channel conditions can change rapidly [LFH13].
[ZIF08]	AF	(i) long aggregation delay if the packet arrival rate is low or if a large target aggregate packet size with fixed arrival rate; (ii) cannot achieve throughput gain by packet aggregation at the expense of high delay when the packet arrival rate is low; (iii) there is a trade-off between throughput and delay by using packet aggregation.	Operation of the proposed model is based on the PRMA protocol. [MaS11]
[HLL08]	AF	Proposes an adaptive target aggregate packet size algorithm for A-MPDU in the IEEE 802.11n networks to maximize the throughput by selecting the optimal frame length under different channel conditions.	Cannot accurately determine the optimal target packet size by selecting the optimal packet size that is calculated off-line under typical BERs. [ZaL13]
[WeL11]	AF	Has better performance in delay by limiting the packet re-transmission times.	Has a high loss rate if large numbers of stations try to transmit.
[KSP12]	AF	Determines the optimal packet size for the next transmission by using the current channel conditions.	May not accurately characterize the channel quality as the channel rapidly changes.
[MBR12]	AF	Supports time sensitive applications and satisfies the QoS requirement.	May produce high loss as the packet waits too long.
[LeP07]	ANF	Supports the management of the delay budget and controls the packet aggregate by using the priority strategy.	Needs a large numbers of queues. [MBR12]
[WaH08]	ANF	The scheduling of packet aggregation is a knapsack problem and the proposed algorithm can reduce the complexity to *O*(n) from *O*(n log n).	Assumes that the users have the same QoS requirements.
[MaA12]	ANF	Formulates the optimal aggregation is the NP-hard problem and proposes 2 algorithms to resolve and the selection strategy is based on the data rate and priority.	May not accurately characterize the data rate as the channel rapid change. [KJL13]
[RMP08]	ANF	Attains a large gain in the voice call capacity by the priority selection strategy.	Needs large memory for the pool of queue and only works for the tagging packets [SOS10].
[SeS10]	ANF	Proves that the A-MSDU and A-MPDU have different advantages based on the packet size in which the smallest size packet is first service.	Needs the packet to be ordered in the queue and the determinable optimal waiting time is not accurate [MAH12].

3.4 A-MSDU and A-MPDU Schemes

In the IEEE 802.11n [IEn09] standard there are two aggregation algorithms defined, namely Aggregate MAC Service Data Unit (A-MSDU) and Aggregate MAC Protocol Data Unit (A-MPDU). The IEEE 802.11n MAC sub-layers can be divided into two entities, upper and lower MACs, based on its time sensitivity [KML11]. The A-MSDU operates at the upper MAC while the A-MPDU is performed at the lower MAC.

For the A-MSDU algorithm, multiple MSDUs are aggregated into a single A-MSDU with a single MAC header and then it is transmitted within a single MPDU [KML11] [LYY09]. The A-MSDU increases the maximum frame transmission size from 2304 bytes to 7935 bytes. The frame format of A-MSDU is shown in Figure 3-5. All the sub-frames in a single A-MSDU should have the same transmitter address and receiver address, which means that all the sub-frames are intended to be received by a single receiver and necessarily they are all transmitted by the same transmitter. However, the sub-frames (i.e. MSDUs) are allowed to have different source and destination addresses which are indicated in the sub-frame header. There is a distinction between the source address and the transmitter address and a parallel distinction between the destination address and the receiver address. The transmitter address is the address of the transmitter which sends a frame onto the wireless medium but does not necessarily to create the frame, while source creates a frame and sends it. A similar distinction holds for destination address and receiver address. A receiver may be an intermediate destination, but frames are processed by higher protocol levels only when they reach the destination.

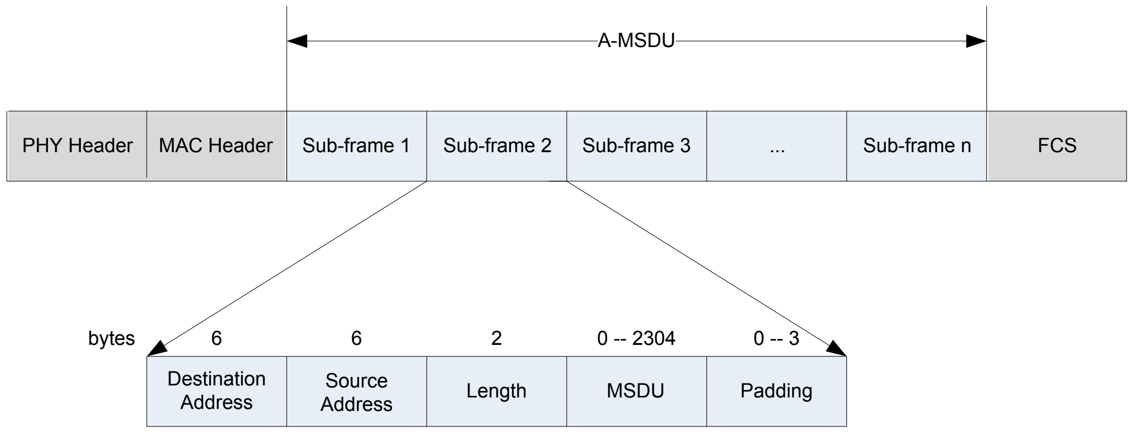


Figure 3-5: The format of A-MSDU frame.

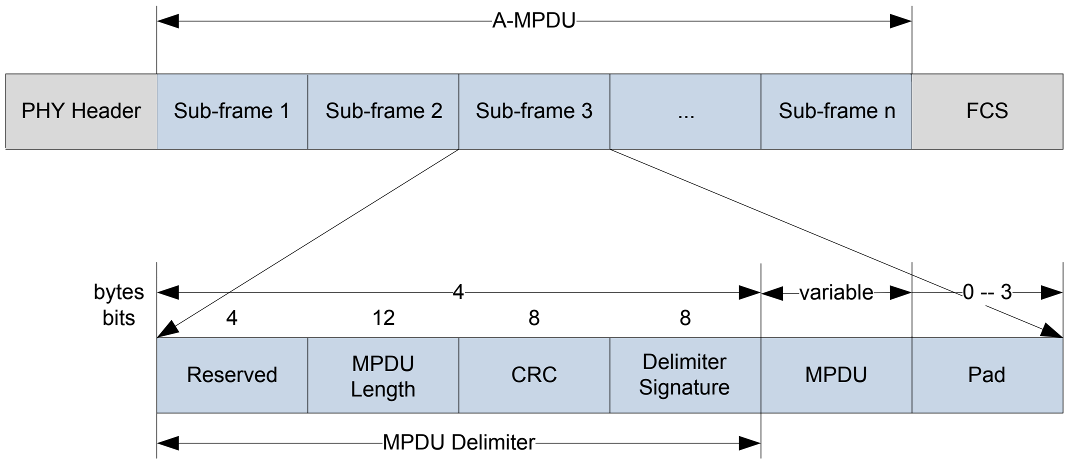


Figure 3-6: The format of A-MPDU frame.

As the unit for an ACK is an MPDU, if any bit within an A-MSDU is corrupted at the receiver, the whole A-MSDU has to be re-transmitted. In the A-MPDU algorithm, multiple MPDUs are aggregated into a single A-MPDU which is delivered to the PHY layer as a single Physical Layer Service Data Unit (PSDU). It is then processed as a single Physical Layer Protocol Unit (PPDU) to be sent to the channel. Figure 3-6 shows the frame format of an A-MPDU. Like the A-MSDU, all the sub-frames have the same sender and receiver addresses in a single A-MPDU. If one or more frames are received with errors, the structure of the A-MPDU can usually be recovered [KML11] as each sub-frame is preceded by an MPDU delimiter signature as shown in Table 3-6. As already mentioned in the IEEE 802.11n MAC, the ACK unit is an MPDU, each sub-frame in a single A-MPDU should be individually ACKed. As multiple MPDUs are transmitted within a single PPDU, the Block ACK (*BA*) is used for the A-MPDU scheme. The relationship between the A-MSDU and A-MPDU is shown in Figure 3-7.

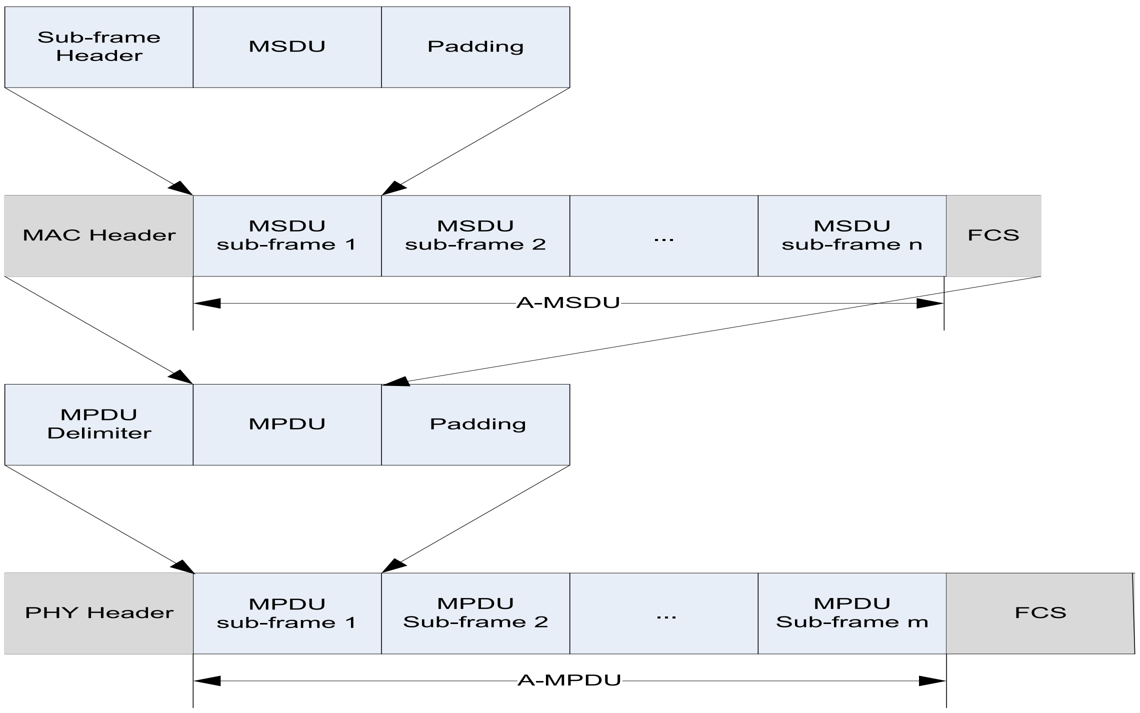


Figure 3-7: The relationship between A-MSDU and A-MPDU frames.

Ginzburg and Kesselman [GiK07] were the first to study the A-MPDU and A-MSDU algorithms to estimate the maximum throughput of the IEEE802.11n standard and they concluded that the performance of A-MSDU aggregation significantly degrades for high packet error rates and high PHY rates.

In [SNC08], the authors present a simulation based performance comparison of the maximum throughput for the aggregation algorithms. In [Lin06], an analytical study of the performance for the A-MSDU and A-MPDU algorithms is performed under uni-directional and bi-directional data transfers. In [KuD06], a transmission queue model of an IEEE 802.11n station is proposed to estimate the impact of packet aggregation size on the delay and channel utilization. They studied the impact of the packet aggregation size over a wide range of operating conditions and the results showed that the aggregate packet size is impacted by the packet arrival rate and data frame size. The authors in [KHS08] propose an analytical model to evaluate the throughput performance based on an enhanced discrete time Markov chain (DTMC). The results have shown that the target aggregate packet size has little impact on the throughput in an unsaturated network, while the throughput varies according to the target packet size in a saturated network and the larger target aggregate packet size do not always yield better throughput performance.

In [HLL08], an adaptive target aggregate packet size algorithm for A-MPDU in the IEEE 802.11n networks is proposed to maximize the throughput by selecting the optimal frame length under different channel conditions. The network throughput can be further improved if it is employed together with the PHY rate adaption mechanism. The performance of the algorithm with adaptive target aggregate packet size is better than that of the algorithm with fixed target aggregate packet size. The idea of combining fragmentation with A-MPDU aggregation is also presented in [SYC06] while the authors in [SeS10] proposed a simple scheduling algorithm to determine which aggregation option is used based on the packet size where the smallest size packet is served first and this is referred to SSFS. However, in [SYC06], the authors ignore the delay and just consider the reliability and throughput. It has the disadvantage that it needs to change the format of MSDU which is not easy to implement.

In [SWS10], the authors compare the throughput performance of A-MSDU, A-MPDU and PHY super-frames on different aggregation types and fragmentation types under delay limits in an ultrahigh-speed WLAN. The authors in [SOS10] show that the header size has a larger significant impact for the small sized MSDUs than that of large sized MSDUs by using the packet aggregation algorithm. They present a packet aggregation algorithm (mA-MSDU) which is described in [SOS11] to reduce the protocol overheads and implement a re-transmission control over the individual sub-packets at the MSDU level. The simulations results and analysis show a significant performance improvement in terms of throughput for the proposed scheme particularly for applications that have a small packet size such as VoIP.

[AbA11] shows the impact of the multi-rate operation on the A-MSDU and A-MPDU based on the experiments using certified IEEE 802.11n equipment. Within A-MSDU operation, an A-MSDU enabled station that operates at low data rate affects the performance of A-MSDU disabled stations transmitting at higher rates. However, within A-MPDU operation, the effect of not enabling A-MPDU for all stations is even worse than the effect of multi-rate operation.

In [KML11], an adaptive aggregation scheme is proposed in order to resolve the potential problem in A-MPDU where the sender transmits A-MSDUs within A-MPDUs in an adaptive manner. A new analytical model to evaluate the performance of A-MSDU and A-MPDU aggregation schemes is defined in [DAM11] where the model is defined for a reliable multicast transport and allows the estimation of the MAC layer efficiency. The proposed algorithm can improve the throughput over A-MSDU by up to 19% in the single hop network topology but it has poor performance in a multiple hop network.

There are also some researchers who have studied some special applications (e.g. video) for the A-MSDU and A-MPDU algorithms. The authors in [BaA12] study the performance of packet aggregation to improve the efficiency and quality of the video transmissions over the IEEE 802.11n wireless networks. In [ZCY10], they study the impact of the video transmission for the packet aggregation, especially for the A-MPDU algorithm in IEEE 802.11n wireless networks. It was found that when the optimal packet aggregation size was changed following the channel conditions the throughput was improved but this had little effect on the video quality.

For the technique described in [SeS10] may drop the packets in the queue as the packet waits too long. This disadvantage is similar to [MBR12] where a real time scheduler scheme is proposed which relies on traffic priority in order to support time sensitive applications and satisfy the QoS requirements by employing A-MSDU. A scheme is proposed in [KSP12] to determine the optimal frame size for the next transmission using current channel information. In [KSP12], the authors employ frame size estimation (FSE) with extended Kalman Filter (EKF) which uses a tight frame error rate (FER) bound for OFDM system to obtain the instantaneous link quality. The researchers found that the number of video streams that can be supported on the IEEE 802.11n networks depends on the implementation of the packet aggregation in [LYK08].

3.4.1 Discussion of A-MSDU and A-MPDU Schemes

A-MSDU and A-MPDU are the most popular packet aggregation algorithms which are defined in the IEEE 802.11n standard and employed by the IEEE 802.11ac standard draft to achieve the goal of high-throughput. The algorithm of A-MSDU combines several MSDU packets into a large packet with a single MAC header and the A-MPDU algorithm aggregates multiple MPDU packets into a large frame with a single PHY header. Research studies on these two methods can be divided into two categories: comparison of the performance for the two algorithms and determining the optimal packet aggregation size and selection schemes to aggregate the packets.

Generally speaking, under a high BER environment, the A-MPDU algorithm is more efficient in terms of throughput than that of the A-MSDU algorithm as only the corrupted sub-packet needs to be re-transmitted in the A-MPDU algorithm while the whole aggregate packet has to be re-transmitted in the A-MSDU algorithm if an error occurs. Until now, a significant number of the proposed packet aggregation algorithms based on the A-MSDU and A-MPDU algorithms try to determine the optimal target aggregate packet size in different environments. Some proposed algorithms have good performance in terms of throughput [KML11] [MBR12], however, they are not based on the real live traffic loads. In this thesis, the A-MSDU algorithm is the typical FIFO algorithm that is employed as the benchmark algorithm to be compared with the proposed AAM algorithm. The other comparison algorithm used is the SSFS algorithm [SeS10].

3.5 Chapter Summary

In this chapter, we have reviewed a number of different packet aggregation algorithms which have been proposed by other researchers. As an upper limit on throughput and a lower limit on delay exist in wireless networks, packet aggregation needs to be used to improve the throughput. There exists a trade-off between the maximizing throughout and minimizing delay. However, most of these researches were focused either on the improvement in throughput or the reduction in delay. A number of different categories of packet aggregation algorithms have been presented and the effect of the packet aggregation algorithm on throughput and delay performances was studied in IEEE 802.11 WLANs. Some algorithms were presented that attempt to reduce the delay increase to asymptotically approach the lower limit delay. The proposed packet aggregation algorithms were divided into 4 categories: FF, FNF, AF and ANF. Two most important packet aggregation algorithms, A-MSDU and A-MPDU, were introduced which are defined in the IEEE 802.11n and IEEE 802.11ac standards. Many studies of the two algorithms have been carried out.

Chapter 4
Proposed Packet Aggregation Algorithm

In chapter 2 and chapter 3, we discussed the background and the performance of the packet aggregation algorithms proposed by other researchers. The most important performance metrics for the packet aggregation algorithms are the throughput and the delay. As described in the chapter 3, most of the proposed packet aggregation algorithms attempt to optimize a single metric, i.e. either to maximize throughput or minimize delay. These packet aggregation algorithms don't take account of the varying nature of the traffic load particularly the random nature of the packet size and packet rate. For example, in [ZhN08] [SWC99], [BMS09], the authors focus on the optimal aggregate packet size to achieve the maximum throughput but they don't consider the delay, while in [WeL11] and [LeP07] they just consider how to achieve the minimum delay.

In this chapter we will outline the proposed packet aggregation algorithm called Adaptive Aggregation Mechanism (AAM) which has been designed to achieve the goal of the best aggregation trade-off in terms of realizing the maximum average throughput with the minimum average delay compared to a number of popular aggregation algorithms for different traffic loads in wireless network environments. The AAM algorithm is an adaptive algorithm in that it responds to the varying nature of the packet size and packet rate and attempts to assemble the target size aggregate packet with the minimum delay.

In Figure 4-1, the structure of the AAM algorithm is shown. As can be seen, the AAM algorithm is essentially a feedback control system which comprises three elements. The first of these is the Adjustable Aggregation Algorithm (A^3) which aggregates the packets that are selected from a selection window in the input buffer. The selection window whose size *N* is adjustable contains the front *N* packets in the input buffer. The second element is the Aggregate Packet Analyzer (APA) which analyzes the number of sub-packets in the aggregate packet and the aggregate packet delay of the assembled aggregate packet. The sub-packet is the MAC service data unit (MSDU) that is received from the logical link control (LLC) sub-layer. The aggregate packet delay includes two elements: the waiting delay which is defined as the duration from the first sub-packet from its arrival in the input buffer to being aggregated in the output buffer, and the transmission delay which is defined as the time from when the selected packets are aggregated to when the ACK of the aggregate packet is received. The waiting time starts from the arrival of the first sub-packet of an aggregate packet into the input buffer.

The last element is the Aggregate Tuning Algorithm (ATA) which uses the analysis results from the APA to dynamically adjust the size of the selection window in A^3. The ATA also has two user input parameters which are specified per-queue: the target aggregate packet size and the maximum acceptable delay.

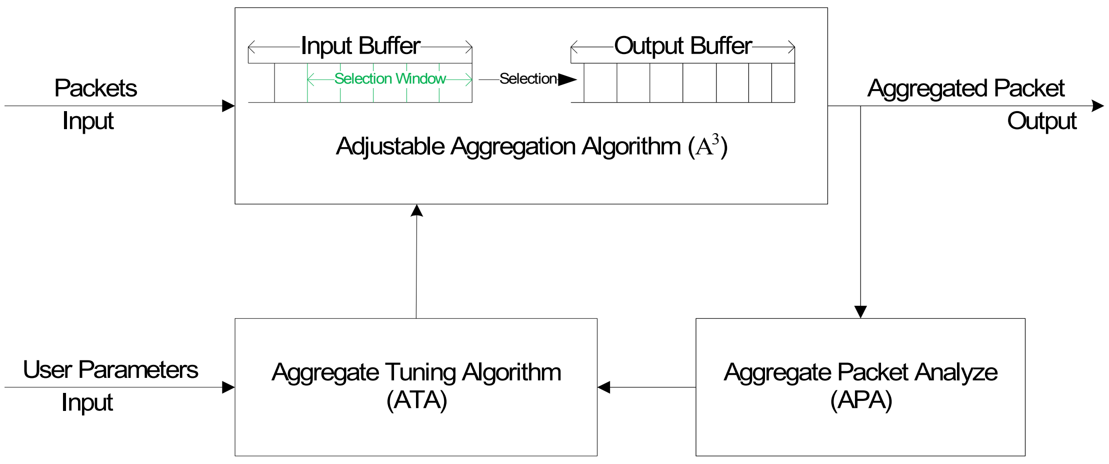


Figure 4-1: The structure of the AAM algorithm.

4.1 Adjustable Aggregation Algorithm (A^3)

The Adjustable Aggregation Algorithm (A^3) is employed to select packets from the selection window in the input buffer used for assembling the aggregate packet. The sizes of the packet and the inter-arrival times between packets in the input buffer are considered to be random. For example, the size of a data packet can be up to 7935 bytes (i.e. A-MPDU) and the inter-arrival times between packets arriving into the input buffer can be short in busy wireless networks (e.g. microseconds) or long in idle wireless networks (e.g. milliseconds).

There are two buffers used in the A^3 algorithm: one is the input buffer which is a buffer for receiving the incoming packets from the network or upper layers and all sub-packets are selected from it; the other one is the output buffer which is the buffer used for assembling the selected packets into an aggregate frame. Figure 4-2 shows how packets are selected from the input buffer and moved into the output buffer. There are more than 7 packets in the input buffer and 4 packets are selected by the A^3 and moved into the output buffer. After completing the selection process, all the selected packets in the output buffer are aggregated together and transmitted as a single frame.

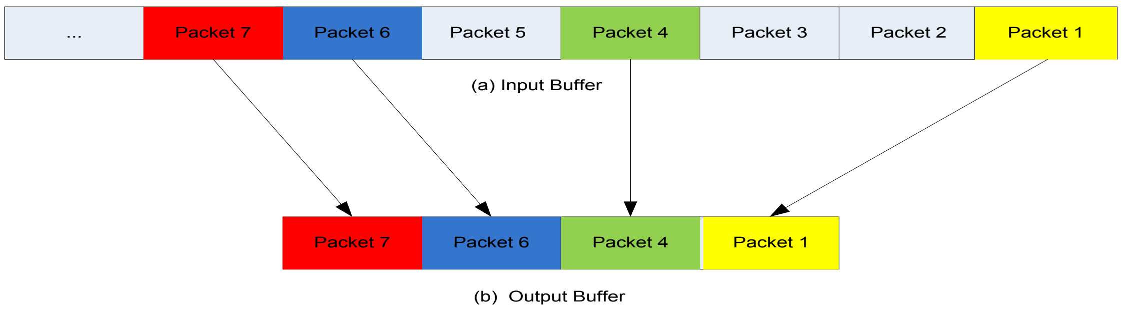
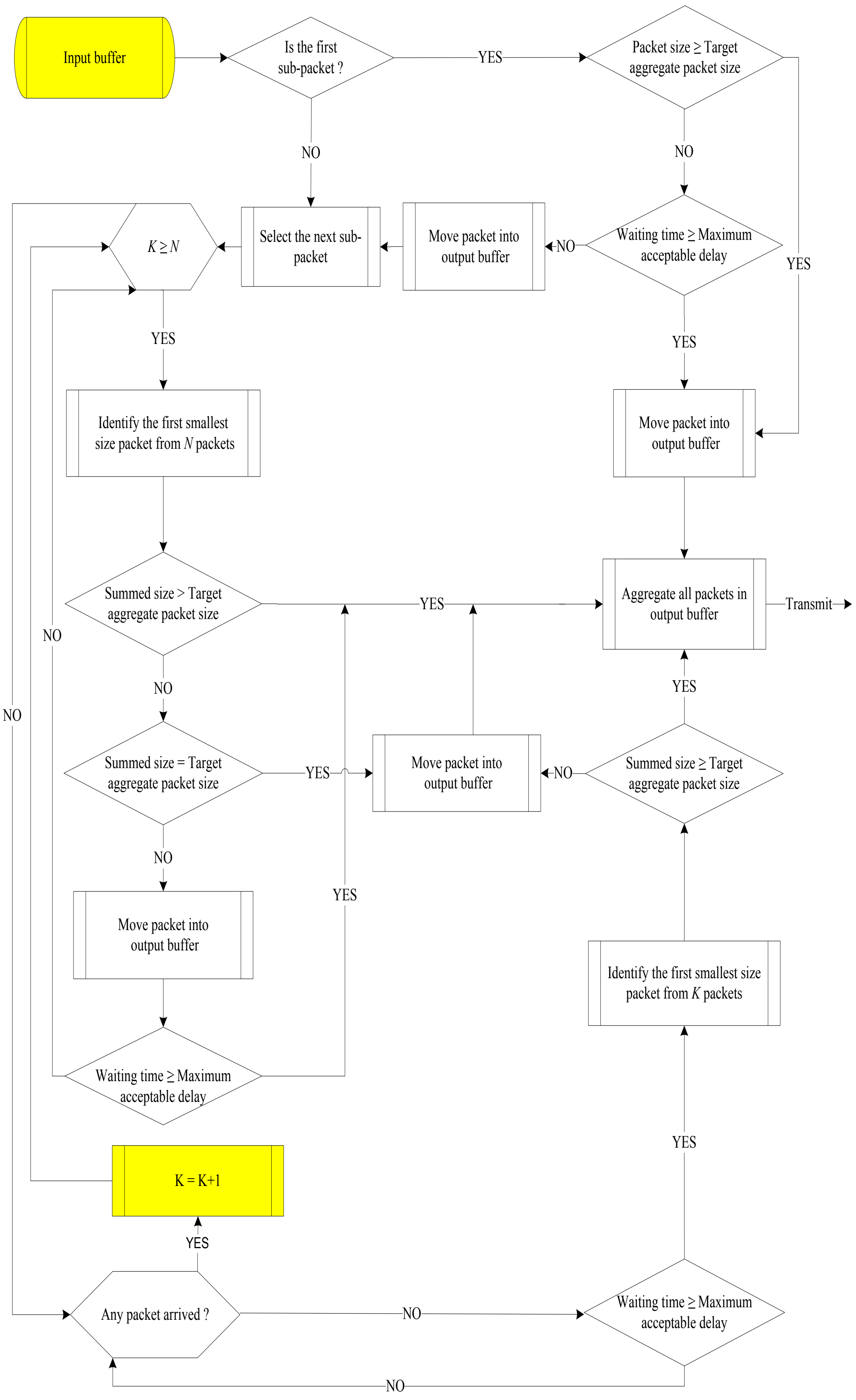


Figure 4-2: How packets are selected from the input buffer and moved into the output buffer.



Input buffer
Is the first sub-packet ?
YES
Packet size ≥ Target aggregate packet size
NO
NO
YES
K ≥ N
Select the next sub-packet
Move packet into output buffer
NO
Waiting time ≥ Maximum acceptable delay
YES
YES
Identify the first smallest size packet from N packets
Move packet into output buffer
Summed size > Target aggregate packet size
YES
Aggregate all packets in output buffer
Transmit
NO
NO
NO
YES
Summed size = Target aggregate packet size
YES
Move packet into output buffer
NO
Summed size ≥ Target aggregate packet size
NO
YES
Move packet into output buffer
Identify the first smallest size packet from K packets
Waiting time ≥ Maximum acceptable delay
K = K+1
YES
YES
Any packet arrived ?
NO
Waiting time ≥ Maximum acceptable delay
NO

Figure 4-3: The flowchart of the A^3.

The flowchart of A^3 is shown in Figure 4-3. The size, *N,* is first initialised. Next, the front packet in the input buffer is selected as the first sub-packet and its size is compared with the target aggregate packet size.

(1) The packet is moved into the output buffer and transmitted if its size is greater than or equal to the target aggregate packet size.

(2) If the packet size is smaller than the target aggregate packet size, the packet waiting time in the input buffer is compared with the maximum acceptable delay.

(3) If the waiting time is greater than or equal to the maximum acceptable delay, the packet is moved into the output buffer and transmitted as soon as possible without waiting for other packets to arrive.

(4) If the waiting time is less than the maximum acceptable delay, the packet is moved into the output buffer and the algorithm selects the next sub-packet.

Assuming that the number of packets in the input buffer is *K* and the selection window size is *N*, there are two outcomes that result from a comparison of *K* and *N*.

When $K \geqslant N$, there are sufficient packets available for selection in the selection window. At first, the first smallest size packet in the selection window is identified where the first smallest size packet is the minimum length packet or the first one to have arrived if more than one packet has the same minimum length in the selection window. A^3 compares the sum of this packet size and the selected packets sizes (i.e. the summed size) with the target aggregate packet size.

(i) If the summed size is greater than the target aggregate packet size, the first smallest size packet is not selected and all the selected packets in the output buffer are aggregated together and transmitted as soon as possible.

(ii) If the summed size equals the target aggregate packet size, the first smallest size packet is selected and moved into the output buffer to be aggregated with other selected packets. Then the aggregate packet is transmitted as soon as possible.

(iii) If the summed size is less than the target aggregate packet size, the first smallest size packet is selected and moved into the output buffer. Then the algorithm checks the waiting time of the first sub-packet in the output buffer.

(iv) If it exceeds the maximum acceptable delay, all the selected packets in the output buffer are aggregated into a single packet to be transmitted.

(v) Otherwise, the A^3 resumes the process of selecting another sub-packet.

When $K < N$, there are insufficient packets to be selected from the selection window in the input buffer and the algorithm must wait for packets to arrive. The maximum time to wait is determined by the waiting time of the first sub-packet and the specified maximum acceptable delay. During this time, if there are further packet arrivals into the input buffer, the algorithm checks whether the inequality $K < N$ applies or not.

(a) The case where $K \geqslant N$ has been described above.

(b) If $K < N$ applies, then the A^3 waits until the waiting time of first sub-packet exceeds the maximum acceptable delay or $K \geqslant N$ applies.

(c) If the waiting time of first sub-packet exceeds the maximum acceptable delay and $K < N$ still applies, the first smallest size packet is selected from the K packets. Then the algorithm processes the packet according to steps (i), (ii) and (iii) described above until the summed size reaches the target aggregate packet size or all K packets are selected. However, in step (iii) the algorithm does not wait for more sub-packets and all the selected packets are aggregated and transmitted as soon as possible.

The first sub-packet in every aggregate packet is always the front packet in the input buffer. This rule ensures that the waiting time of the first sub-packet does not increase indefinitely which can happen if the front packet were to be the biggest size packet in the selection window.

In this scheme, the sub-packets may need to be reordered in the receiver. Based on the IEEE 802.11n standard, the receiver contains a reordering buffer which is responsible for reordering packets, so that the packets are eventually passed up to the next MAC process in the order of received sequence number [IEn09]. The reordering process may increase the delay as a packet may need to wait for other packets to arrive.

4.2 Aggregate Packet Analyzer (APA)

The aggregate packet analyzer (APA) is used to analyze the number of sub-packets and the aggregate packet delay of the aggregate packet in order to determine the selection window size for the next aggregate packet.

The APA analyses the current aggregate packet and the previous aggregate packet in order to determine the value of *N* for the next aggregate packet where two registers are used. Each register has two members, one is a counter used to record the number of sub-packets and the other one is a timer used to record the aggregate packet delay. The value of the counter of the number of sub-packets is incremented by 1 when a packet is moved into the output buffer. When the first sub-packet is moved into the output buffer, the waiting time of this packet in the input buffer is set as the value of the aggregate packet delay timer and then it is incremented until the ACK for the aggregate packet is received or the life time of the aggregate packet has been exceeded. The selection process is stopped when the waiting time of the first sub-packet exceeds the maximum acceptable delay or the summed size is greater than or equal to the target aggregate packet size. The timer of the aggregate packet delay is frozen when an ACK for the aggregate packet is received which also means the aggregate packet delay can be determined. Each register counter and timer will be reset when the ACK for the aggregate packet is received.

After an ACK frame is received, the APA algorithm calculates the differences in the number of sub-packets and the aggregate packet delay between the current aggregate packet and the previous aggregate packet. The outcomes are used to determine the selection window size *N* in the ATA algorithm.

4.3 Aggregate Tuning Algorithm (ATA)

The third element of the AAM algorithm is the aggregate tuning algorithm (ATA) which uses the analysis results from the APA to dynamically adjust the selection window size *N*. The basic idea in developing the tuning rules is that the selection window size is increased in order to improve the throughput when the network performance is improved, while the selection window size is decreased in order to reduce the delay when the network performance deteriorates. The analysis results from the APA are used in ATA to determine the performance of network. The analysis results are the change in the number of sub-packets and the change in the aggregate packet delay, both of which are calculated between the current and the previous aggregate packets. There are three outcomes for the change in the number of sub-packets: increase, decrease and unchanged, and two outcomes for the change in the aggregate packet delay: decrease and no decrease. Generally, a decrease in the aggregate packet delay means that the performance of network is improving while an increase in the aggregate packet delay means that the performance of network is getting worse.

Therefore, the ATA makes appropriate adjustments to *N* as shown in Table 4-1 where *N*-- means that the value of *N* is decremented by 1 and *N*++ means that the value of *N* is incremented by 1. When *N* is greater than 1, the change is based on the rule in column (A) and when *N* equals 1 it is based on the rule in column (B) where the value *N* cannot be reduced any more. Generally, the range of the value of *N* is between 1 and the input buffer size.

Table 4-1: The rules for tuning the size *N* of the selection window

		Change in the Aggregate Packet Delay			
		No Decrease	Decrease	No Decrease	Decrease
Change in the Number of Sub-packets	Increase	*N*	*N*++	*N*	*N*++
	Decrease	*N*--	*N*++	*N*	*N*++
	Unchanged	*N*--	*N*	*N*	*N*
		$N > 1$ *(A)*		$N = 1$ *(B)*	

The rules in Table 4-1 are explained as follows:

A. If the number of sub-packets has increased and the aggregate packet delay has not decreased, the value of *N* is maintained.

B. If the number of sub-packets has increased and the aggregate packet delay has decreased, the value of *N* is incremented by 1.

C. When the number of sub-packets has decreased and the aggregate packet delay has not decreased: (a) if *N* is greater than 1, *N* is decremented by 1; (b) if *N* equals 1, *N* is maintained at 1.

D. If both the number of sub-packets and the aggregate packet delay have decreased, *N* is incremented by 1.

E. When the number of sub-packets is maintained and the aggregate packet delay has not decreased: (a) if *N* is greater than 1, *N* is decremented by 1; (b) if *N* equals 1, *N* is maintained at 1.

F. If the number of sub-packets is maintained and the aggregate packet delay has decreased, *N* is maintained.

Increasing *N* increases the probability of achieving the target size of the aggregate packet, but at the expense of a delay increase. Conversely, decreasing *N* reduces the delay, but also may reduce the probability of achieving the target size of the aggregate packet.

4.4 User Specified Input Parameters

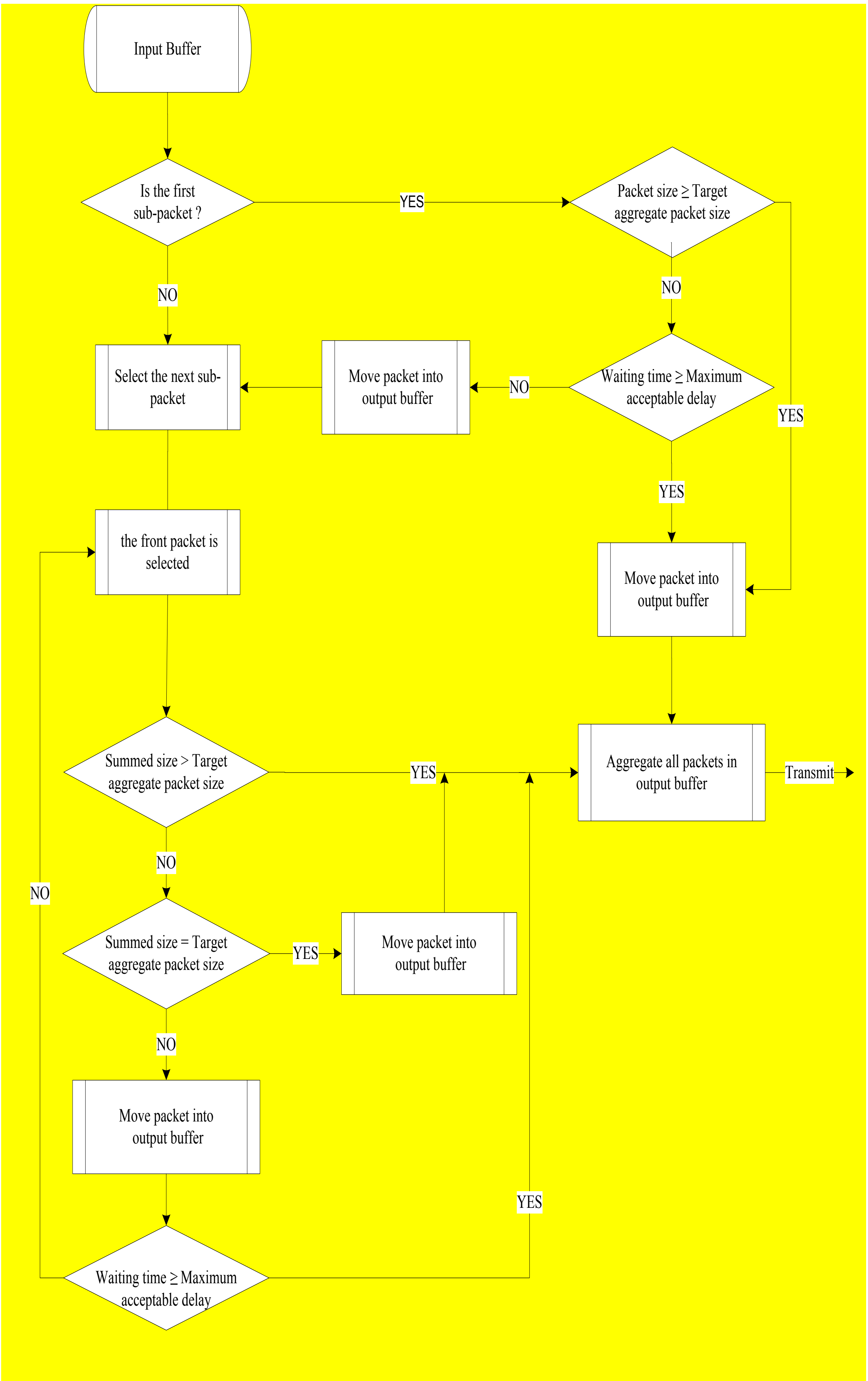
In the AAM algorithm, the target aggregate packet size and the maximum acceptable delay are specified by the user. The target aggregate packet size is the maximum size of the aggregate packet and the maximum acceptable delay is the maximum time that the A^3 is allowed to wait in order to achieve the target aggregate packet size. The values of these parameters are determined by the application being used. For example, if the user wants to use a VoIP application (e.g. *skype)*, the value of maximum acceptable delay is set to 150 ms or less and the maximum acceptable delay could be set to 1 second when the user wishes to use an email application. If the user wants to use both of them at the same time, the maximum acceptable delay could be set to some appropriate value by the user. Both of these user specified parameters, the target aggregate packet size and the maximum acceptable delay, are the threshold values used to control the aggregation process. The two parameters are set in the ATA and the values are sent to the A^3 with the selection window size *N*.

4.5 Analysis of All Three Aggregation Algorithms

In this thesis, the AAM algorithm is compared to two other packet aggregation algorithms, FIFO and SSFS. The FIFO algorithm is used as the benchmark algorithm which employs the most basic and popular packet selection strategy where all packets are aggregated based on the packet arrival time and a selection window scheme is not employed. The flow chart of the FIFO is shown in Figure 4-4. The A-MSDU algorithm [IEn09] is employed as the typical FIFO algorithm to compare with the AAM algorithm. The other algorithm is the SSFS algorithm (Smallest-Size First-Served) where all packets are aggregated based on their size and a selection window scheme is also not employed. The goal of the SSFS algorithm is to achieve the maximum number of sub-packets in an aggregate packet [SeS10] associated with a large delay as it always tries to wait for the smallest size packet to arrive. The flow chart of the SSFS is shown in Figure 4-5. The AAM algorithm employs a selection window scheme and a hybrid selection strategy which combines the FIFO and SSFS selection strategies where the first sub-packet uses the FIFO selection strategy and the other sub-packets use the SSFS selection strategy. The FIFO selection strategy ensures that the delay does not increase indefinitely, while the SSFS selection strategy ensures that the maximum number of sub-packets in an aggregate packet is achieved. The net result of this hybrid approach is that the AAM algorithm can achieve the largest average aggregate packet size for all three algorithms considered.

The throughput improvement is dependent upon the number of packets combined into an aggregate packet in unit time compared to the non-aggregation case. In general, the larger the number of packets that are assembled in unit time, the greater the throughput improvement. As discussed in section 4.2, the aggregate packet delay includes two elements, the waiting delay and the transmission delay. When the packet rate is high (i.e.

the inter-arrival time is small), the packet aggregation does not require extra time for packets to arrive compared to non-aggregation [ZIF08]. Therefore, the aggregate packet delay is dependent upon the transmission delay which is dependent on the PHY rate and the aggregate packet size based on the equation (3.2) [XiR02]. With a fixed PHY rate, the aggregate packet delay is dependent on the aggregate packet size for small values of the inter-arrival time. The reason is that generally a larger average size of the aggregate packet corresponds to a larger average number of sub-packets per aggregate packet which can reduce some of the delays associated with each transmission such as the access medium time (e.g. DIFS) for the sub-packet and the transmission time of the MAC header and ACK.



Input Buffer
Is the first sub-packet ?
YES
Packet size ≥ Target aggregate packet size
NO
NO
Select the next sub-packet
Move packet into output buffer
NO
Waiting time ≥ Maximum acceptable delay
YES
YES
the front packet is selected
Move packet into output buffer
Summed size > Target aggregate packet size
YES
Aggregate all packets in output buffer
Transmit
NO
NO
Summed size = Target aggregate packet size
YES
Move packet into output buffer
NO
Move packet into output buffer
YES
Waiting time ≥ Maximum acceptable delay

Figure 4-4: The flow chart of FIFO.

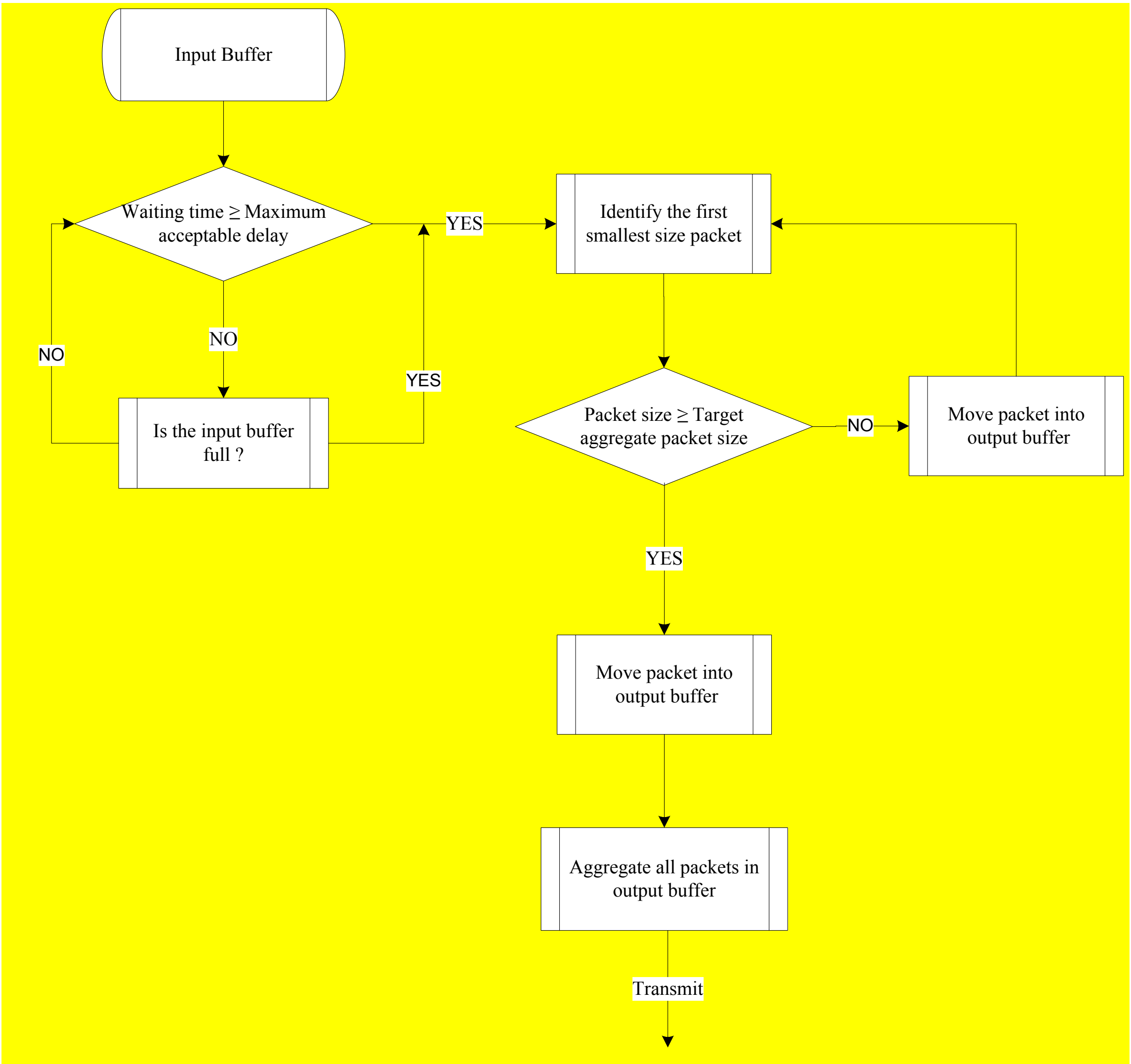

Input Buffer
Waiting time ≥ Maximum acceptable delay
YES
NO
NO
YES
Is the input buffer full ?
Identify the first smallest size packet
Packet size ≥ Target aggregate packet size
NO
Move packet into output buffer
YES
Move packet into output buffer
Aggregate all packets in output buffer
Transmit

Figure 4-5: The flow chart of SSFS.

In this section, we will discuss the interactions between the different parameters for the three packet aggregation algorithms. Figure 4-6 shows the interaction between the different parameters of the FIFO algorithm which is an example of an open control system. The positive sign is used to indicate that if the inter-arrival time increases this causes an increase in waiting delay because the FIFO algorithm aggregates the packets based on the packet arrival time. The negative sign is used to indicate that if the arrival packet size decreases this causes an increase in the number of sub-packets per aggregate packet because it requires a combining of more packets into an aggregate packet in order to reach the target aggregate packet size. This can be explained as follows: if the arrival packet size is small, an aggregate packet needs to combine more sub-packets in order to achieve the target aggregate packet size that is specified (e.g. 1500 bytes). The positive sign is used to indicate that if the number of sub-packets per aggregate packet increases this causes an increase in the aggregate packet size as the packets are selected based on the arrival time. The positive sign is used to indicate that if the aggregate packet size increases this causes an increase in the transmission delay.

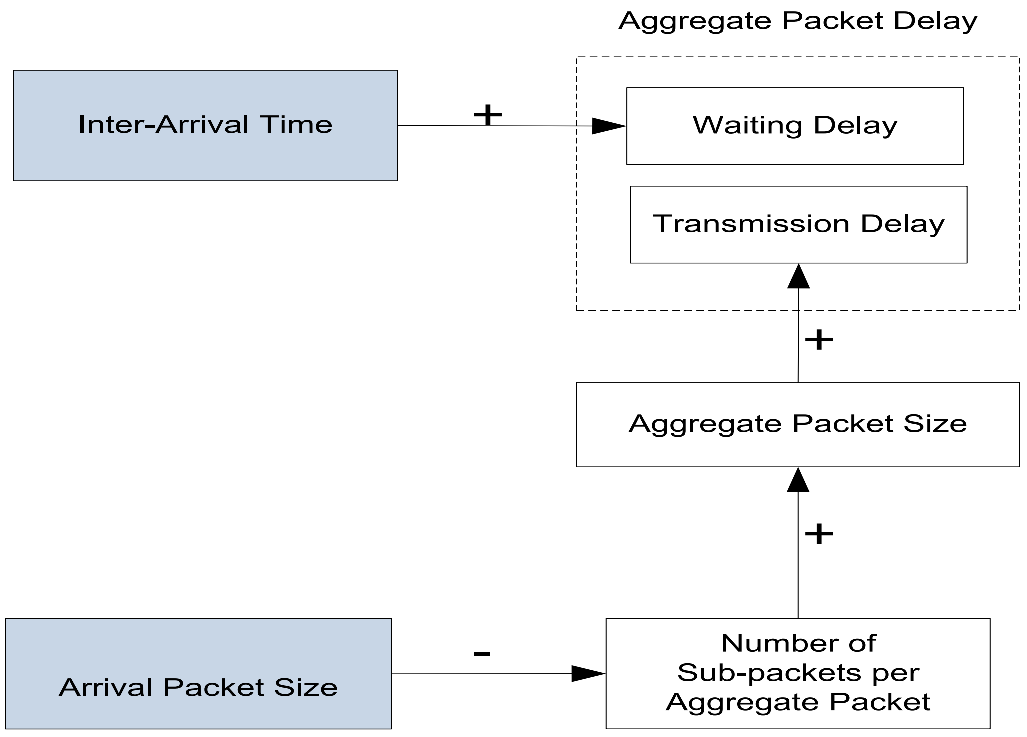


Figure 4-6: The interaction between the different parameters of the FIFO algorithm.

Figure 4-7 shows the interaction between the different parameters of the SSFS algorithm which is also an example of an open control system in its operation. The negative sign is used to indicate that if the inter-arrival time decreases this causes an increase in the number of sub-packets per aggregate packet because there are more packets available for selection in order to assemble a larger number of small sized packets as this is the basis of the SSFS algorithm. The negative sign is used to indicate that if the arrival packet size decreases this causes an increase in the number of sub-packets per aggregate packet as it requires assembling more packets to reach the target aggregate packet size.

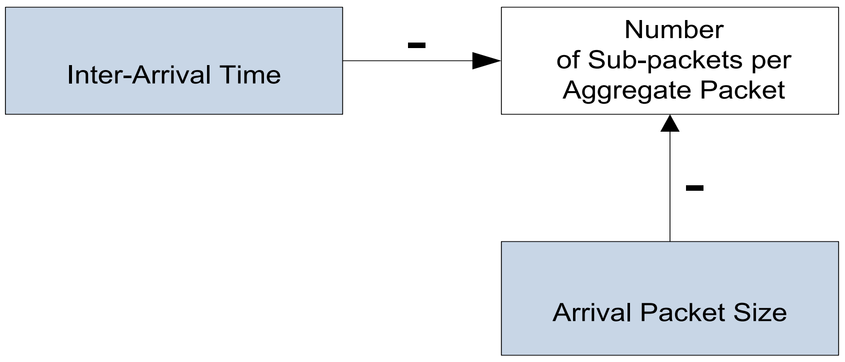


Figure 4-7: The interaction between the different parameters of the SSFS algorithm.

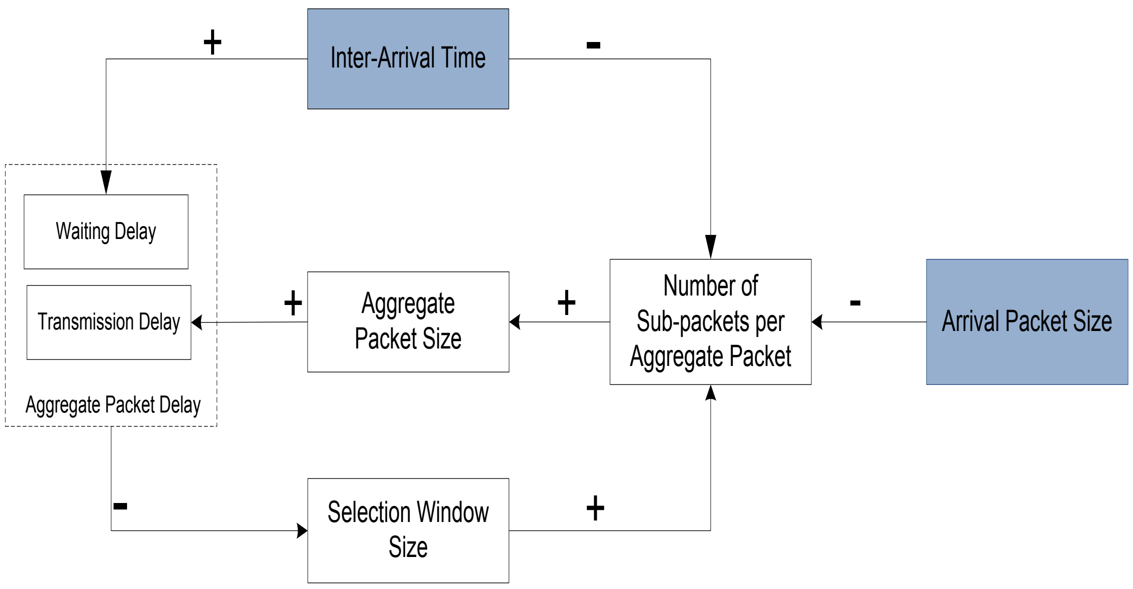


Figure 4-8: The interaction between the different parameters of the AAM algorithm.

Figure 4-8 presents the interaction between the different parameters of the AAM algorithm which is an example of a feedback control system. The negative sign is used to indicate that if the inter-arrival time decreases this causes an increase in the number of sub-packets per aggregate packet where the explanation is the same as that for the SSFS algorithm. The positive sign is used to indicate that if the inter-arrival time increases this causes an increase in waiting delay because the first sub-packet is selected based on the arrival time. The negative sign is used to indicate that if the arrival packet size decreases this causes an increase in the number of sub-packets per aggregate packet where the explanation is the same as that for the SSFS algorithm. The positive sign is used to indicate that if the number of sub-packets per aggregate packet increases this causes an increase in the aggregate packet size where the explanation is the same as that for the FIFO algorithm. The positive sign is used to indicate that if the aggregate packet size increases this causes an increase in the transmission delay. The negative sign is used to indicate that if the aggregate packet delay decreases this causes an increase in the selection window size according to the tuning rules shown in Table 4-1. The positive sign is used to indicate that if the selection window size increases this causes an increase in the probability of achieving a larger number of sub-packets per aggregate packet.

In Figure 4-8, there is a negative feedback loop formed between the number of sub-packets per aggregate packet, aggregate packet size, aggregate packet delay and selection window size when raw packets can be aggregated (i.e. when packet aggregation can occur). Increasing the number of sub-packets per aggregate packet leads to an increase in the aggregate packet size and an increase in the aggregate packet size increases the aggregate packet delay as the transmission delay increases. An increase in the aggregate packet delay decreases the selection window size according to the tuning rules shown in Table 4-1 and a decrease in the selection window size decreases the probability of achieving a larger number of sub-packets per aggregate packet.

Based on the previous discussion, the AAM algorithm is a feedback control system and can operate with random packet size and packet rate. The AAM algorithm has a better performance compared to the other two algorithms (i.e. FIFO and SSFS) as will be demonstrated by the results presented in chapter 5.

4.6 Simulation

In this thesis, the AAM algorithm is implemented in two scenarios. In the first scenario, the aggregation process of the AAM algorithm is implemented as a standalone C++ computer program. In this test scenario, there are two objectives: (i) To demonstrate that the AAM algorithm is an adaptive algorithm that can operate over a wide range of different traffic loads; (ii) To demonstrate that the AAM algorithm has a superior performance compared to that of the FIFO and SSFS algorithms in terms of the number of sub-packets per aggregate packet for a given delay (i.e. waiting delay) by employing a selection window scheme associated with the hybrid selection strategy. In the second test scenario, the AAM algorithm is implemented in a wireless network containing an AP and a client station. This test scenario has been implemented in the ns-3 simulator. The objectives of the second scenario is to demonstrate that the AAM algorithm (i) has a superior performance compared to the other two algorithms in terms of the aggregation trade-off in achieving the maximum average throughput with the minimum average delay in wireless networks; and (ii) can significantly improve the performance in terms of the average throughput in error-prone wireless networks.

In the two test scenarios, 16 captured traffic trace files are used as the input. All of these traffic trace files were captured from live Wi-Fi hotspot networks by using the network sniffer tool *wireshark*. These traffic trace files were captured at different locations, such as in a library, university campus, coffee shop, train station and airport, and were also captured at different times from 29th May 2012 to 17th July 2012. The details of the 16 captured traffic trace files are shown in Table 4-2.

There are two parameters selected as input to the simulation namely the packet size and the packet arrival time. The advantages of using this approach to generate the network test traffic profiles are the following: Firstly, these captured traffic trace files better represent the typical traffic patterns found on data networks. Secondly, these captured traffic trace files represent the characteristics of a traffic load containing different types of application. These captured traffic trace files differ from each other in terms of the packet size and the inter-arrival time between packets.

Table 4-2: The details of the 16 captured traffic trace files

ID	**Capture Date**	**Capture Time**	**Number of captured Packets**	**Average Packet Size (bytes)**	**Average Packet Rate (pps)**	**Capture Location**
1	29th May 2012	10:30—11:30	331649	552	92.1	*JAVA City*, DIT, Dublin
2	29th May 2012	12:00—13:00	410444	576	114.0	*JAVA City*, DIT, Dublin
3	29th May 2012	14:00—15:00	272349	550	75.7	*JAVA City*, DIT, Dublin
4	29th May 2012	16:00—17:00	400514	588	111.3	*JAVA City*, DIT, Dublin
5	29th May 2012	17:30—19:30	393944	590	109.4	*JAVA City*, DIT, Dublin
6	19th June 2012	09:30—10:30	33958	442	9.4	*Costa Coffee*, Dawson St, Dublin
7	19th June 2012	11:00—12:00	20255	235	5.6	*Parliament Square,* TCD, Dublin
8	19th June 2012	12:30—13:30	33156	440	9.2	*Costa Coffee*, Dawson St, Dublin
9	19th June 2012	16:00—17:00	12998	223	3.6	*Parliament Square,* TCD, Dublin
10	19th June 2012	17:00—18:00	23933	571	6.6	*Costa Coffee*, Dawson St, Dublin
11	24th June 2012	12:00-13:00	24747	317	6.9	*Hueston* train station, Dublin
12	24th June 2012	13:30-14:30	15242	94	4.2	*Hueston* train station, Dublin
13	24th June 2012	15:00—16:00	22358	137	6.2	*Hueston* train station, Dublin
14	26th June 2012	10:30—11:30	69299	399	19.2	Library, Kevin Str., DIT, Dublin
15	26th June 2012	12:00—13:00	13785	155	3.8	Library, Kevin Str., DIT, Dublin
16	17th July 2012	19:00—19:50	24962	694	8.3	*Shuangliu* Airport, Chengdu, China

Where *JAVA City* is a name of a popular student coffee shop on the campus of Dublin Institute of Technology and TCD is Trinity College Dublin. As a lot of RTS/CTS packets were captured, the average packet size of the captured traffic trace file 12 is small.

4.6.1 The Aggregation Process Only Scenario

In this test scenario, the packet size and the packet arrival time for these captured traffic trace files are used as the input to a standalone C++ computer program employed to implement the aggregation process of the AAM algorithm.

Table 4-3: Explanation of the key parameters used in the C++ simulation

Key Parameters	**Mathematical Symbol**
Packet size	*S_pkt*
Packet arrival time	*T_arriv*
Selection window size	*N*
Maximum acceptable delay	*Max_accedelay*
Target aggregate packet size	*S_target*
Waiting time of the first sub-packet	*T_waitfirstpkt*
Number of packets in the input buffer	*K*
Number of sub-packets of current aggregate packet	*N_curre_pkt*
Number of sub-packets of previous aggregate packet	*N_previ_pkt*
Aggregate packet delay of current aggregate packet	*Max_curre_aggdelay*
Aggregate packet delay of previous aggregate packet	*Max_previ_aggdelay*
Summed size with the selected packets sizes in output buffer and the selecting packet size in input buffer	*S_summed*

In this implementation, it is assumed that the packets arrive into the input buffer and start to be processed by the AAM algorithm. The packet size *S_pkt* is used as the input to generate the traffic load and the packet arrival time *T_arriv* is set as the time that the packet arrived into the input buffer. The description of the key parameters of this scenario is given in Table 4-3.

Table 4-4: An example of the calculation of the aggregate packet delay

Parameters	**Value**
Number of sub-packets in aggregate packet (*N_spkt*)	11
Arrival time of the first sub-packet (*T_arriv*)	11.2374 s
Selection window size (*N*)	5
Arrival time of the last packet in the selection window for selection the 11th sub-packet (*T_arriv*)	11.6874 s
Aggregate packet delay (*Max_aggdelay*)	0.4500 s (i.e. = 11.9874 – 11.2374)

In this test scenario, the waiting delay is based on the inter-arrival time between the captured packets. The inter-arrival time corresponds the waiting time which equals the interval between the arrival times of the captured packets. For example, the first packet's arrival time is 16.3394 s and the second packet's arrival time is 16.6678 s, so the waiting time of the first sub-packet *T_waitfirstpkt* is 0.3278 s (i.e. = 16.6678 - 16.3394). In this test scenario, we use an assumption that the transmission delay is zero, so the aggregate packet delay *Max_aggdelay* equals the waiting time of the first sub-packet which equals the inter-arrival time between the first sub-packet and the last arriving packet in the selection window. For example, as shown in Table 4-4, an aggregate packet contains 11 sub-packets, the arrival time of the first sub-packet is 11.2374 s and the selection window size *N* is 5. When selecting the 11th sub-packet, the arrival time of the packet which is the last one arriving into the selection window is 11.6874 s, so the *Max_aggdelay* is 0.4500 s (i.e. = 11.9874 – 11.2374).

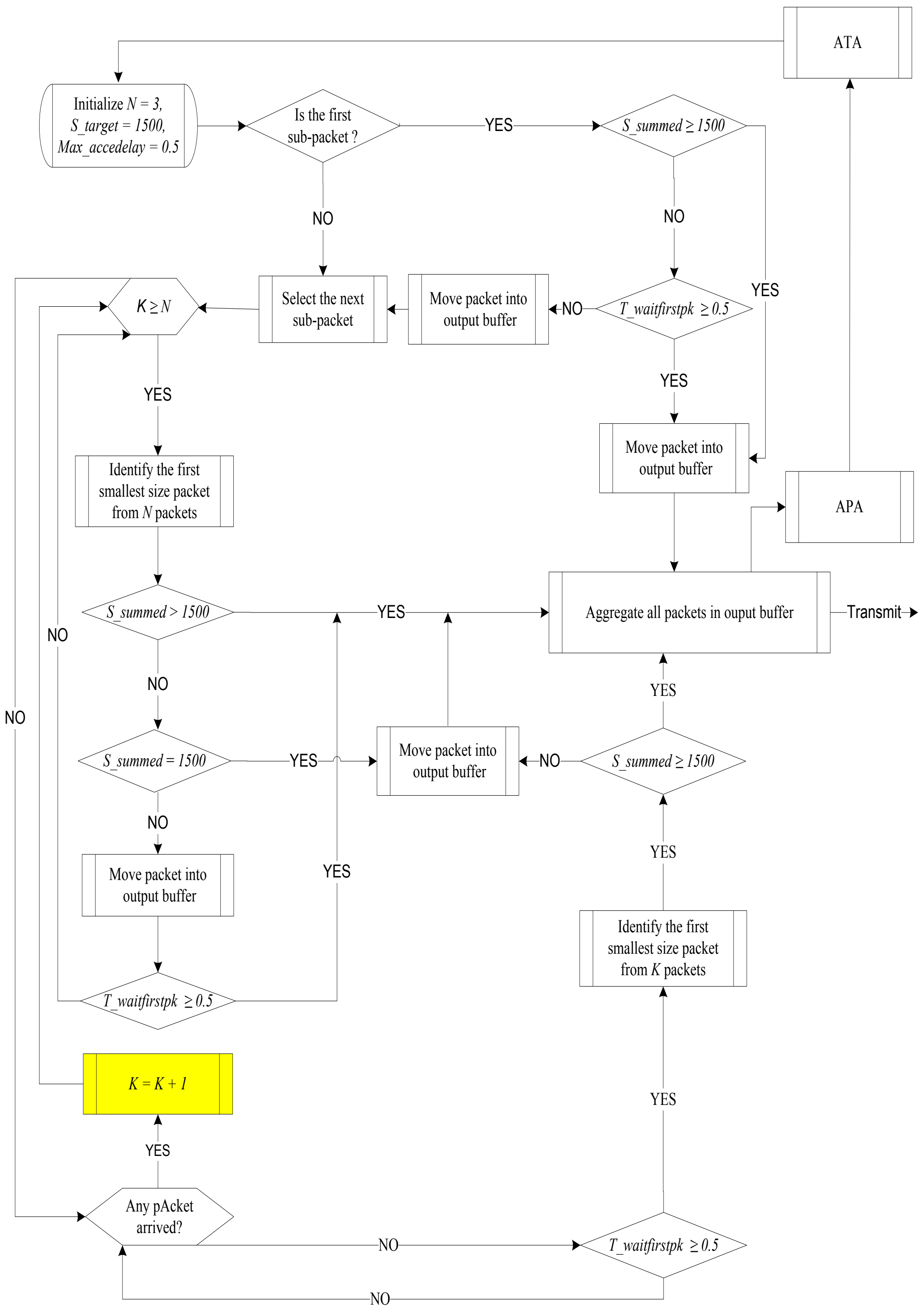

ATA
Initialize N = 3, S_target = 1500, Max_accedelay = 0.5
Is the first sub-packet ?
YES
S_summed ≥ 1500
NO
NO
YES
Select the next sub-packet
Move packet into output buffer
NO
T_waitfirstpk ≥ 0.5
K ≥ N
YES
YES
Move packet into output buffer
Identify the first smallest size packet from N packets
APA
S_summed > 1500
YES
Aggregate all packets in ouput buffer
Transmit
NO
NO
NO
YES
S_summed = 1500
YES
Move packet into output buffer
NO
S_summed ≥ 1500
NO
YES
Move packet into output buffer
YES
Identify the first smallest size packet from K packets
T_waitfirstpk ≥ 0.5
K = K + 1
YES
YES
Any pAcket arrived?
NO
T_waitfirstpk ≥ 0.5
NO

Figure 4-9: Flowchart showing the operation of the AAM algorithm.

The operation of the AAM algorithm in this test scenario is shown in Figure 4-9. First the parameters are initialized; the selection window size *N* is set to 3, the target aggregate packet size *S_target* is fixed at 1500 bytes, the maximum acceptable delay *Max_accedelay* is 0.5 s where the sizes of the input buffer and the output buffer are both 100 packets and the other parameters are set to zero (i.e. the two groups of registers in the APA).

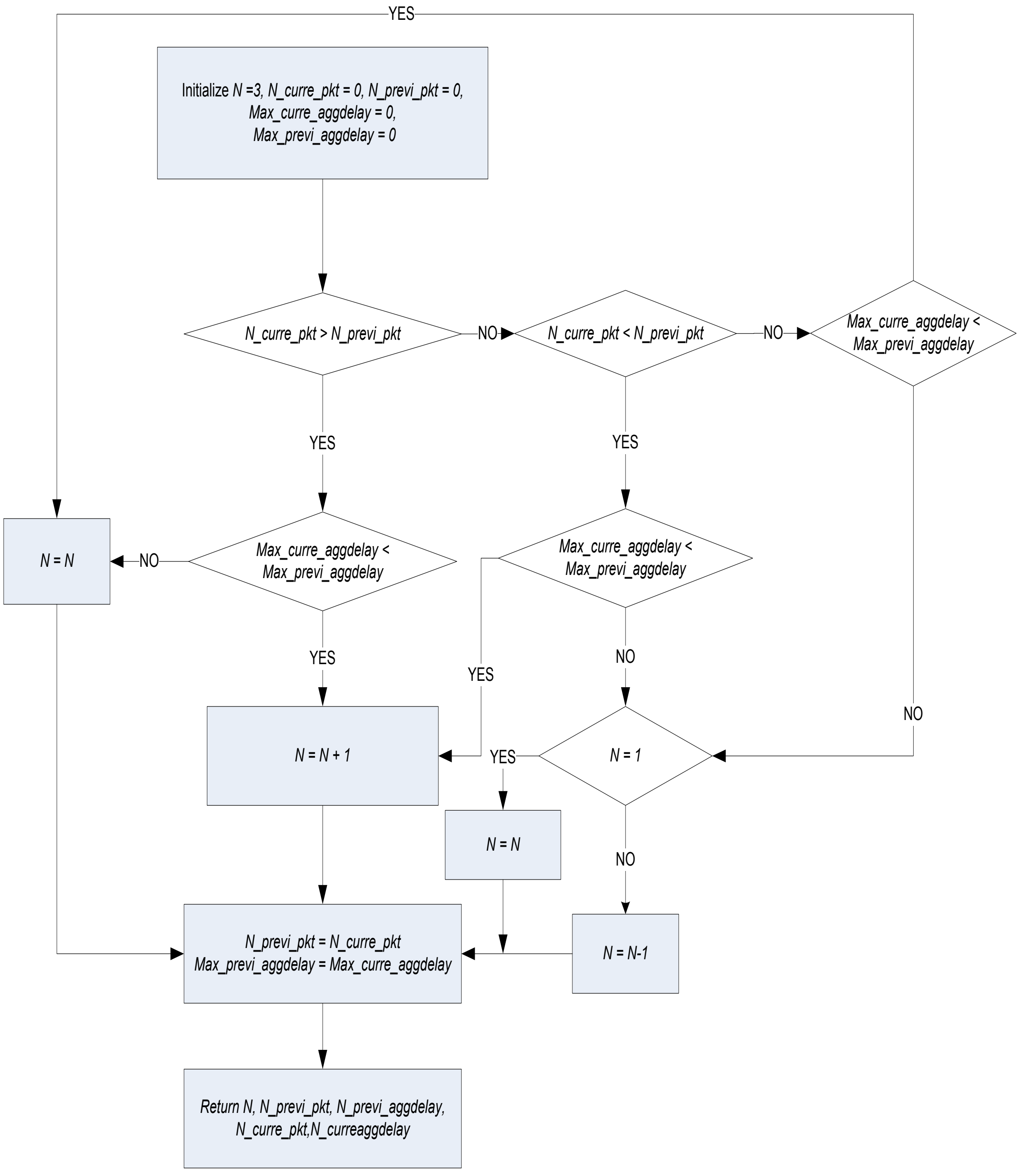

YES
Initialize N =3, N_curre_pkt = 0, N_previ_pkt = 0, Max_curre_aggdelay = 0, Max_previ_aggdelay = 0
N_curre_pkt > N_previ_pkt
NO
N_curre_pkt < N_previ_pkt
NO
Max_curre_aggdelay < Max_previ_aggdelay
YES
YES
N = N
NO
Max_curre_aggdelay < Max_previ_aggdelay
Max_curre_aggdelay < Max_previ_aggdelay
YES
YES
NO
NO
N = N + 1
YES
N = 1
N = N
NO
N = N-1
N_previ_pkt = N_curre_pkt
Max_previ_aggdelay = Max_curre_aggdelay
Return N, N_previ_pkt, N_previ_aggdelay, N_curre_pkt,N_curreaggdelay

Figure 4-10: Flowchart showing the operation of the APA and ATA algorithms.

The input traffic loads *S_pkt* and *T_arriv* are generated based upon the captured traffic trace files. The sub-packets are selected from the selection window in the input buffer based on the A^3 until the selection process is stopped. If the condition where ((*Max_aggdelay* ≥ *Max_accedelay*) OR (*S_summed* ⩾ *S_target*)) is satisfied, the selection process is stopped and then the selected packets in the output buffer are aggregated. The operations of the APA and ATA are shown in Figure 4-10. The APA starts to analyze the *N_curre_pkt* and the *Max_curre_aggdelay* for the current aggregate packet which are compared to that for the previous aggregate packet *N_previ_pkt* and *Max_previ_aggdelay*, and the outcomes are sent to the ATA. After receiving the analysis results from the APA, the ATA adjusts the value of *N* for the next aggregate packet based on the rules shown in Table 4-1.

Table 4-5: The definitions of the performance metrics for the AAM algorithm in the scenario of aggregation process only

Performance Metrics	Comment
Number of sub-packets	The number of packets contained in an aggregate packet which provides a measure of the number of packets combined per aggregate packet.
Selection window size	The size of the selection window.
Sub-packet delay	The waiting time of every sub-packet in an aggregate packet which provides a measure of the average delay of each sub-packet in an aggregate packet.
Aggregate packet delay	The waiting delay which equals the waiting time of the first sub-packet. It provides a measure of the delay for an aggregate packet in the buffer.
Average packet delay	Calculated from the average aggregate packet delay based on per sub-packet count which provides a measure of the average waiting delay of the aggregate packets that contains the same number of sub-packets.

In this scenario, the performances for all 16 captured traffic trace files in terms of the selection window size, the CCDF (Complementary Cumulative Distribution Function) of the number of sub-packets and the CDF (Cumulative Distribution Function) of the sub-packet delay, the number of sub-packets against the average aggregate packet delay for the AAM algorithm will be presented in chapter 5. The average packet delay is the mean aggregate packet delay based on the per sub-packet count. This means that all aggregate packets which have the same number of sub-packets are used to calculate the average packet delay. For example, an average packet delay is calculated for all aggregate packets containing 7 sub-packets aggregate packets and similarly for all aggregate packets containing 8 sub-packets and so on. The parameters used to analyze the performance of the AAM algorithm in this scenario are shown in Table 4-5.

4.6.2 Deployment Scenario in Wireless Networks

The second test scenario is implemented on the ns-3.14 simulation tool where the AAM algorithm has been deployed in a wireless network. The AAM algorithm is implemented in an IEEE 802.11 WLAN with and without transmission errors present, and the payloads of the packets are based on the packet sizes found in the captured traffic trace files.

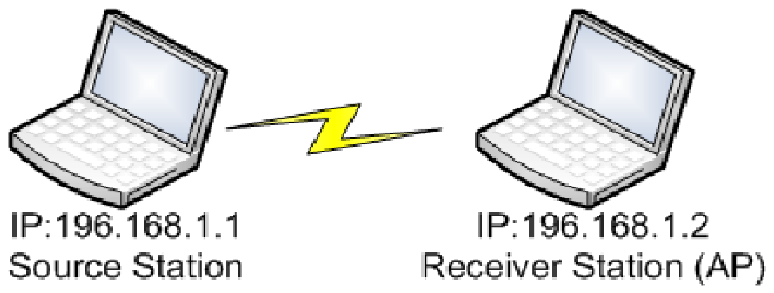


Figure 4-11: The topology of the wireless network in the ns-3 simulation.

The topology used is shown in Figure 4-11. The wireless network contains two stations, one station is the receiver station (i.e. with IP address 196.168.1.2) which is also the access point (AP), and the other one is operated as the source station (i.e. with IP address 196.168.1.1) which is also the client station. The distance between the stations is 50 meters.

In this test scenario, the source station employs the AAM algorithm to aggregate the packets at the MAC layer. The basic simulation details of packet aggregation in ns-3 are presented in chapter 2 and the operation of the AAM algorithm in ns-3 is shown in Figure 4-12.

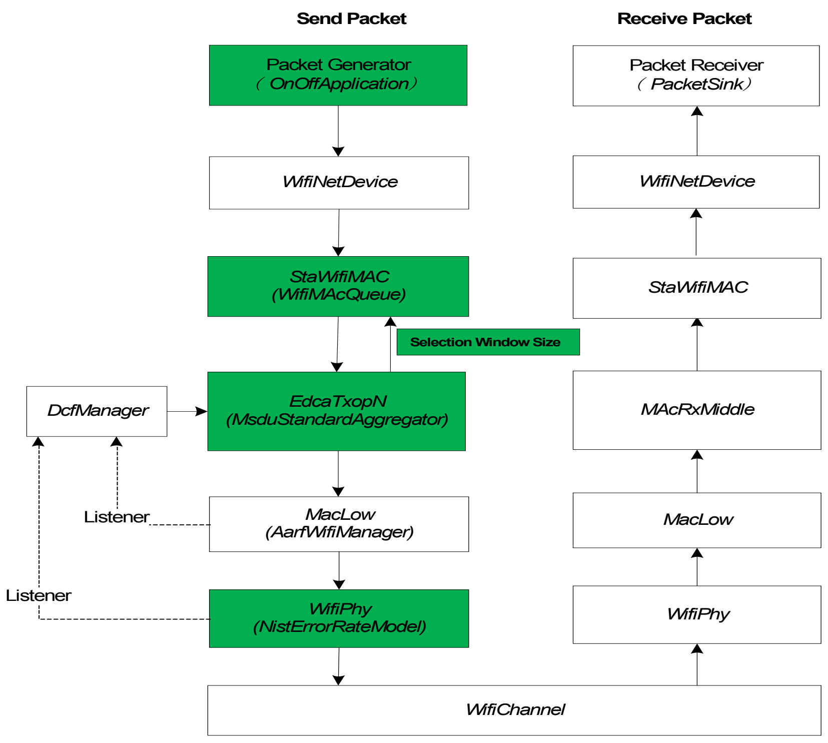


Figure 4-12: The operation of the AAM algorithm in the ns-3 simulation.

There are several modules that have been modified in ns-3 in order to implement the AAM algorithm. The modified module *OnOffApplication* is used to generate the packets whose sizes are based on the captured traffic trace files as the original *OnOffApplication* generates packets at a fixed size. The original *WifiMacQueue* module selects the packets based on the FIFO selection strategy, so it has been modified to select the packets based on the A^3. The module of *EdcaTxopN* is used to aggregate the packets and has been modified to implement the APA and ATA to adjust the size of the selection window which is sent to the *WifiMacQueue* module. With transmission errors present, the module *NistErrorRateModule* is employed which has also been modified in order to change the value of BER. The parameter of *SetRemoteStationManager* is used to determine whether the PHY rate adaption mechanism is employed or not. These modified modes are list in Table 4-6.

Table 4-6: The list of the modified ns-3 module files

Module Name	**Modification**
OnOffApplication	Modified to accept an input from the captured traffic trace files.
WifiMacQueue	Modified to achieve the operation of the packet selection based on the A^3 algorithm from the selection window whose size *N* is passed from the modified *EdcaTxopN* module.
EdcaTxopN	-Modified to implement the APA algorithm to analyze the aggregate packets. - Modified to implement the ATA algorithm to calculate the selection window size *N* of the next aggregate packet
DataRate of *OnOffApplication*	- modified in order to change the data rate based on the varying interval between the raw packets of the traffic trace files.
NistErrorRateModule	Modified to allow the value of BER to be changed.

The simulation parameters used for the implementation of the AAM algorithm in ns-3 are shown in Table 4-7. After these parameters are initialized, the packets are generated by the modified module *OnOffApplication* and sent to the queue in MAC layer (i.e. *WifiMacQueue*). The parameter *DataRate* in *OnOffApplication* is the generated packet rate which will be called the *data rate* in this thesis. At the start when a packet arrives into the input buffer, the arrival time is recorded by using the time stamp (*tstamp*) and the size is also recorded by the class of *GetSize*. Then *EdcaTxopN* invokes the module of *WifiMacQueue* to select the packets and packets are aggregated by the module of *MsduStandardAggregator* after the selection process is completed and then the aggregate packet is transmitted. After an ACK is received for the aggregate packet, the results of the number of sub-packets and the aggregate packet delay between the current and previous aggregate packets are calculated in *EdcaTxopN*. Next the module of *EdcaTxopN* adjusts the selection window size for the next aggregate packet based on the rules shown in Table 4-1 and then it sends the updated value of *N* to the module of *WifiMacQueue.*

Table 4-7: The simulation parameters used to implement the AAM algorithm in ns-3

Parameter	**Value**
Number of stations	2
Initial size of the selection window	3
Distance between stations (m)	50
Maxpacketnumber in *WifiMacQueu*	400
Max. acceptable delay in *EdcaTxopN* (second)	0.05
MaxSlrc in *AarfWifiManager*	10
MaxSuccessThreshold in *AarfWifiManager*	100
MaxAmsduSize in *MsduStandardAggregator*	1500
PHY rate adaption module	*AarfWifiManager*
Transmission errors module	*NistErrorRateModule*
PHY rate (Mbps)	6, 9, 12, 18, 24, 36, 48, 54
DataRate in *OnOffApplication*	2, 4, 6, 8, 10, 12, 14, 16,18, 20, 22, 24, 26, 28,30
BER in *NistErrorRateModule*	10^{-5}, 10^{-4}, 5×10^{-4}, 10^{-3}, 1.5×10^{-3}, 1.6×10^{-3}, 1.7×10^{-3}, 1.8×10^{-3}, 1.9×10^{-3}, 2×10^{-3}, 2.1×10^{-3}, 2.2×10^{-3}, 2.3×10^{-3}, 2.5×10^{-3}, 3×10^{-3}, 4×10^{-3}, 6×10^{-3}, 8×10^{-3}, 10^{-2}, 2×10^{-2}

When the AAM algorithm is implemented in a wireless network without transmission errors present and also without employing the PHY rate adaption mechanism, the PHY rate can be set to 6 Mbps, 9 Mbps, 12 Mbps, 18 Mbps, 24 Mbps, 36 Mbps, 48 Mbps and 54 Mbps, which is implemented by the parameter *SetRemoteStationManager* set to *ConstantRateWifiManager*, and the data rate is changed by controlling the parameter *DataRate* in the *OnOffApplication* module. The transmission errors module

NistErrorRateModule and PHY rate adaption mechanism module *AarfWifiManager* are not employed.

Table 4-8: The performance metrics for analysis the AAM algorithm in the deployment scenario in wireless networks

Performance Metric	**Definition**
Throughput	The average payload in bits per second successfully transmitted from the source node to the destination node.
Maximum average throughput	The average throughput when the wireless network is saturated.
Average delay	The average time required to successfully transmit packets from the MAC layer of the source node to the MAC layer of the destination node.
Minimum average delay	The average delay when the wireless network is saturated.
Aggregation trade-off	The maximum average throughput with the minimum average delay in wireless networks.
Deviation	Defined as the difference between the target aggregate packet size and the aggregate packet size.
Mean square deviation	The average value of the square of the deviation.
Data rate	The data rate (in bits per second) arriving into the buffer which equals the generated data rate of the generator (i.e. *DataRate* in *OnOffApplication*)
BER	The Bit Error Rate used to characterize the transmission errors

In this test scenario, different values of BER are used (as shown in Table 4-7) in order to demonstrate that the AAM algorithm is a robust algorithm. When the AAM algorithm is implemented in the error-prone wireless networks the *NistErrorRateModule* is modified to set the different values of the BER. The PHY rate adaption module *AarfWifiManager* which is based on the adaptive auto rate fallback (AARF) [LMT04] is employed to select the PHY rate, where the module *AarfWifiManager* is invoked by the parameter *SetRemoteStationManager.*

The performance metrics used to analyze these performances for the AAM algorithm in this scenario are shown in Table 4-8 and the results will be presented in chapter 5.

4.7 Chapter Summary

This chapter has presented the proposed packet aggregation algorithm AAM that is an adaptive algorithm and is essentially a feedback control system which tries to achieve the best aggregation trade-off in terms of realizing the maximum average throughput with the minimum average delay for the different traffic loads typically found in real wireless networks. To the best of our knowledge the AAM algorithm is the first packet aggregation algorithm to employ a tunable selection window scheme for the selection of sub-packets.

The AAM algorithm comprises three elements: Adjustable Aggregation Algorithm (A^3), Aggregate Packet Analyzer (APA) and Aggregate Tuning Algorithm (ATA). The adjustable aggregation algorithm (A^3) assembles the aggregate packet by selecting packets from a selection window. The size of this selection window is adaptive. Increasing the size of the selection window increases the probability of achieving the target aggregate packet size of the aggregate packet at the expense of a delay increase. Conversely, decreasing the size of the selection window reduces the delay, but also reduces the probability of achieving the target aggregate packet size. The aggregate packet analyzer (APA) analyzes the number of sub-packets in the aggregate packet and the aggregate packet delay associated with assembling the aggregate packet. The aggregate tuning algorithm (ATA) uses the analysis results from the APA to adaptively adjust the size of the selection window. The ATA has two input parameters specified by the user: the target aggregate packet size and the maximum acceptable delay which are the threshold values used to control the aggregation process of the AAM algorithm. The interaction between the different parameters for all three algorithms considered (i.e. FIFO, SSFS and AAM) is discussed.

After an introduction and an analysis of the proposed AAM algorithm, the two test scenarios which are employed to implement the AAM algorithm were described. The first test scenario was used to demonstrate that the AAM algorithm is an adaptive algorithm that has a superior performance in terms of the number of sub-packets per aggregate packet for a given average packet delay. In this test scenario, a standalone computer program was developed using C++ and 16 captured traffic trace files which were captured from live Wi-Fi hotspot networks and were used to provide an input traffic profile. In the second test scenario, the AAM algorithm was deployed in a wireless network by using the ns-3 simulator. The AAM algorithm was implemented in the source node to analyze the performances in terms of throughput, delay and aggregation trade-off in achieving the maximum average throughput with the minimum average delay in wireless networks with and without transmission errors present.

Chapter 5
Results and Analysis

In this chapter, we will present the performance results for the AAM algorithm described in chapter 4 and provide an analysis of them. There were 16 traffic trace files captured from live Wi-Fi hotspot networks whose details were described in chapter 4. The performance analysis is based on an analysis of all 16 captured traffic trace files, however for convenience the results for the captured traffic trace files 2 and 14 only will be discussed here. The details are presented in Table 5-1.

Table 5-1: Some details of the captured traffic trace files 2 and 14

ID	Capture Time	Capture Date	Number of captured Packets	Average Packet Rate (pps)	Capture Location
2	12:00 – 13:00	29th May 2012	410444	114	*JAVA City*, DIT, Dublin
14	10:30 –11:30	26th June 2012	69299	19.2	Library, Kevin street, DIT, Dublin

5.1 Performance in the Scenario of Aggregation Process Only

The objective of this scenario is to demonstrate that the AAM algorithm is an adaptive algorithm which can operate over a wide range of input traffic loads. Also, it serves to demonstrate that the AAM algorithm has a superior performance over the FIFO and SSFS algorithms in terms of the number of sub-packets that can be aggregated within a given average packet delay.

The results of the performance in terms of the selection window size for these captured traffic trace files are presented in appendix A. The results of the CCDF (Complementary Cumulative Distribution Function) of the number of sub-packets and the CDF (Cumulative Distribution Function) of the sub-packet delay are presented in appendices B and C, and the results of the performance in terms of the number of sub-packets against the average packet delay are presented in appendix D. The CCDF of the number of sub-packets and the CDF of the sub-packet delay are used to analyze the performance of the AAM algorithm. The CCDF and CDF allow us to make a meaningful comparison of the performances for the different algorithms. The CCDF of the number of sub-packets is the probability that the number of sub-packets takes on a value greater than or equal to a certain value which allows us to compare the performance in terms of the average number of sub-packets per aggregate packet for the different algorithms. The CDF of the sub-packet delay is the probability that the sub-packet delay has a value less than or equal to a certain value which allows us to compare the performance in terms of the average sub-packet delay for the different algorithms.

In this scenario, the target aggregate packet size is set at 1500 bytes with a maximum acceptable delay of 0.5 seconds and the sizes of input buffer and output buffer are both 100 packets and the selection window size is initialized to 3.

5.1.1 Impact of the Selection Window Size on Performance

As the captured traffic trace file 2 contains over 410,000 raw packets in a 3600 second period, we present the average packet rate based on a 10-second interval. The average packet rate for the captured traffic trace file 2 is shown in Figure 5-1. The corresponding performance in terms of the selection window size is presented in Figure 5-2 where the selection window sizes are sampled every ten aggregated packets.

Figure 5-3 and Figure 5-4 show the average packet rate and the selection window size for the captured traffic trace file 14.

From Figure 5-1 and Figure 5-2, it can be seen that the selection window size follows the variation in the average packet rate. When the average packet rate is high the selection window size is large and when the average packet rate is low the selection window size is small. The same conclusion also can be drawn by observing Figure 5-3 and Figure 5-4. A similar result is shown for all captured traffic trace files in appendix A. These results clearly demonstrate that the selection window size can successfully track the changes in the traffic load as the AAM algorithm is an adaptive feedback control system.

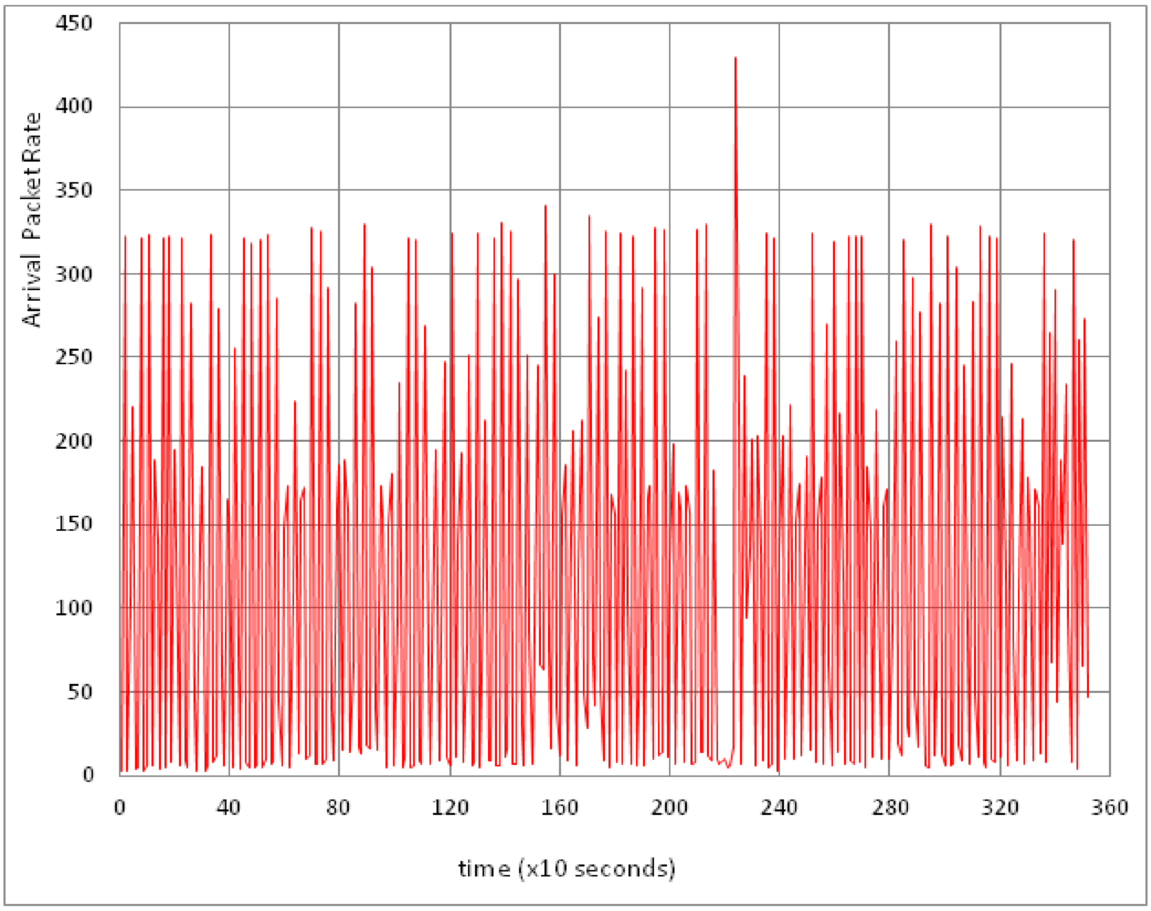


Figure 5-1: The average packet rate for the captured traffic trace file 2.

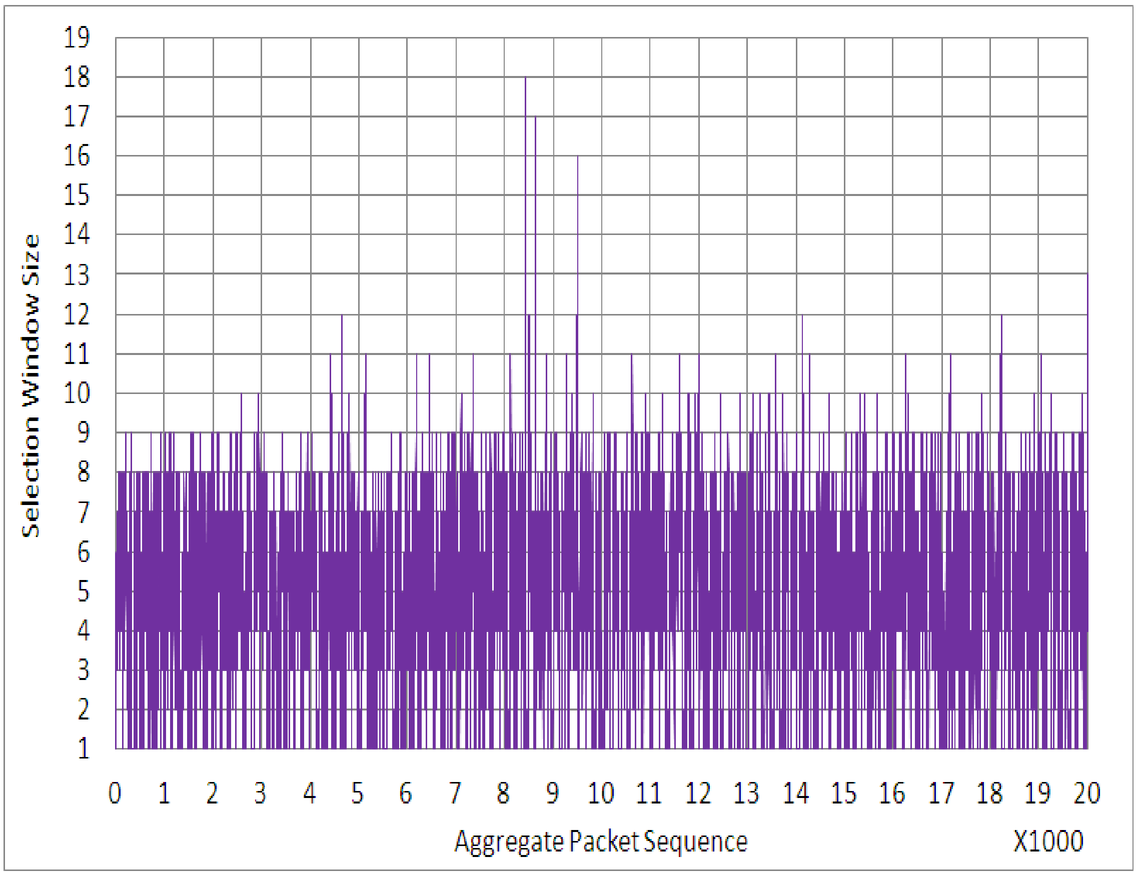


Figure 5-2: The selection window size of the one in ten aggregate packets generated for the captured traffic trace file 2.

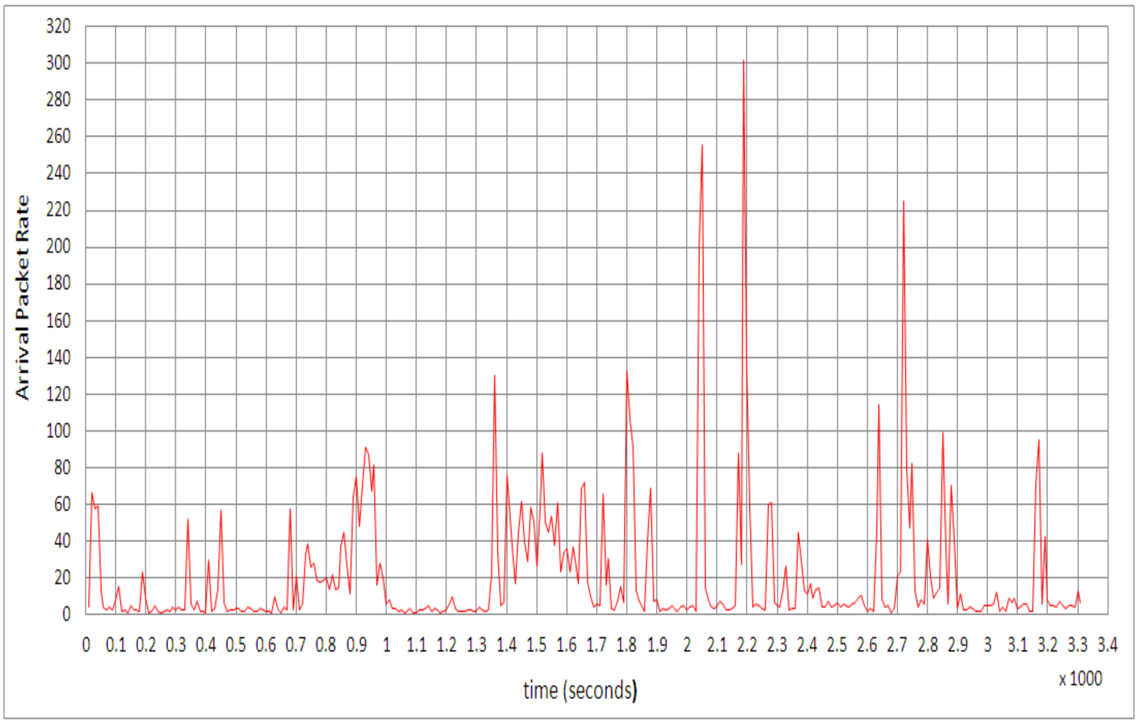


Figure 5-3: The average packet rate for the captured traffic trace file 14.

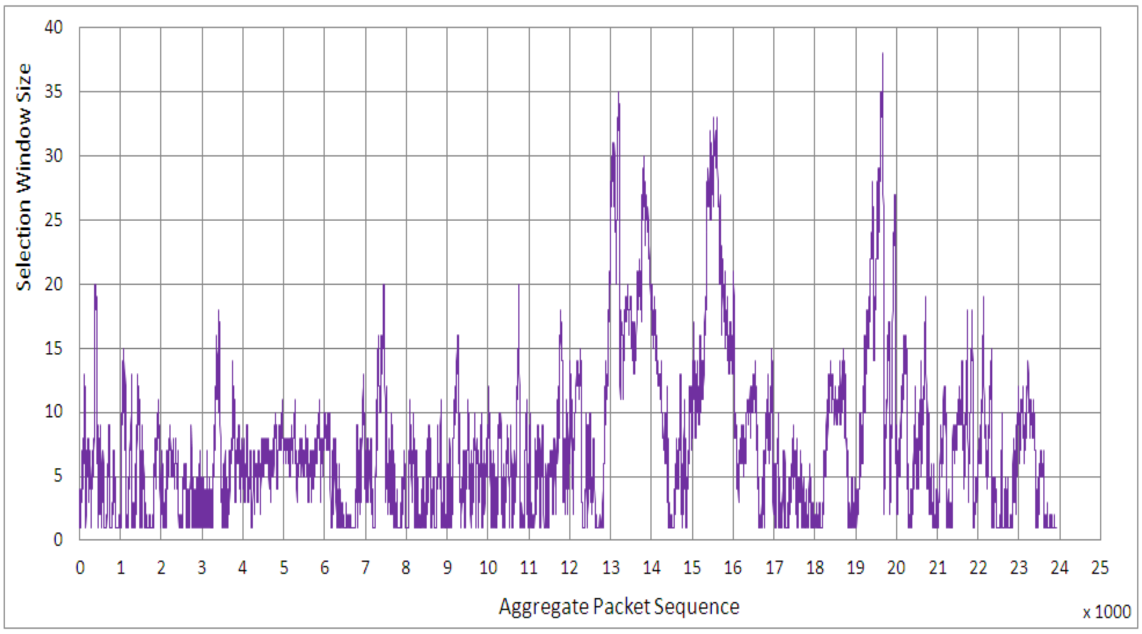


Figure 5-4: The selection window size for the aggregate packets generated for all raw packets input for the captured traffic trace file 14.

5.1.2 CCDF of the Number of Sub-packets

The CCDF of the number of sub-packets is the probability that the number of sub-packets has a value greater than or equal to a certain value. The CCDF allows us to compare the performances in terms of the average number of sub-packets per aggregate packet for the different algorithms. The CCDF of the number of sub-packets for the captured traffic trace file 2 is shown in Figure 5-5 and that for the captured traffic trace file 14 is shown in Figure 5-6.

As shown in Figure 5-5, Figure 5-6 and in appendix B, the captured traffic trace files are tested in a number of different cases:

(1) The FIFO algorithm;

(2) The SSFS algorithm;

(3) The AAM algorithm;

(4) The AAM algorithm with a fixed selection window size at 3, 8 and 10.

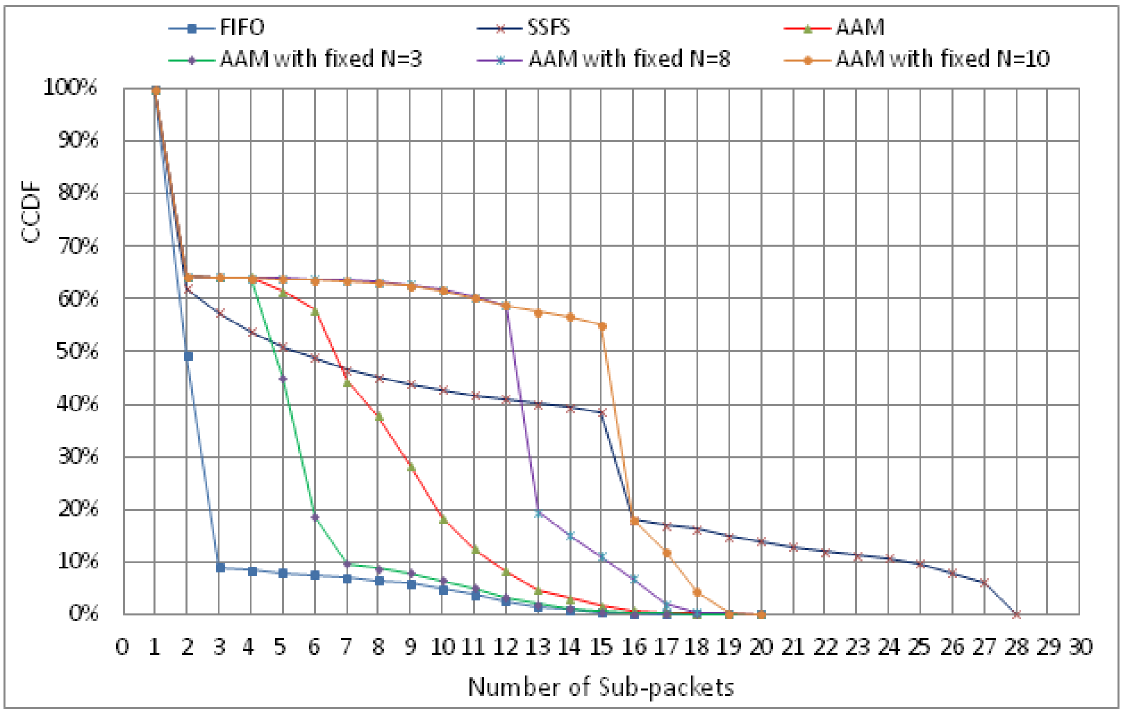


Figure 5-5: The CCDF of the number of sub-packets for the captured traffic trace file 2.

From the result for the FIFO algorithm in Figure 5-5, over 50% of the aggregate packets contain just 1 sub-packet and the probability is less than 0.1 that the number of sub-packets is greater than 2. While for the AAM algorithm, the probability is over 0.6 that the number of sub-packets is greater than 2. This indicates that the average number of sub-packets per aggregate packet produced by the AAM algorithm is greater than that generated by the FIFO algorithm. The performance in terms of the CCDF of the number of sub-packets for the SSFS algorithm is the best for all three algorithms considered. This means that the SSFS algorithm produces the largest average number of sub-packets per aggregate packet and the FIFO algorithm produces the smallest average number of sub-packets per aggregate packet.

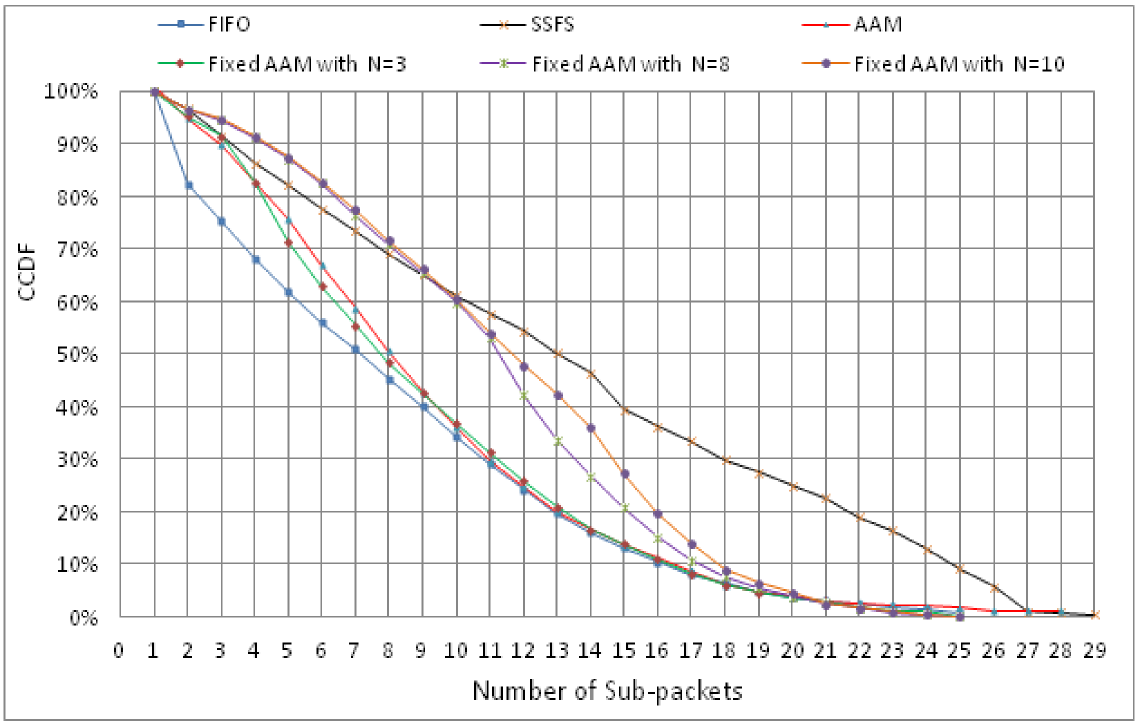


Figure 5-6: The CCDF of the number of sub-packets for the captured traffic trace file 14.

The AAM algorithm with a fixed selection window size (i.e. 3, 8 and 10) is employed in order to demonstrate that the different sizes of the selection window result in different performances in terms of the CCDF of the number of sub-packets. It can be seen that these performances are better than those for the FIFO algorithm. In particular, the larger the selection window size, the better the performance in terms of the CCDF of the number of sub-packets. This demonstrates that the selection window size has an impact on the average number of sub-packets in an aggregate packet.

From the graphs in appendix B, we can infer that the performance in terms of the CCDF of the number of sub-packets for the AAM algorithm is similar to that for the benchmark FIFO algorithm when the packet rate is low. However, it has a better performance than that for the FIFO algorithm when the packet rate is high. The explanation for this is that for low values of packet rate, the average number of sub-packets per aggregate packet for the AAM algorithm is similar to that of the FIFO algorithm as there are insufficient packets available for selection in the selection window. While for large values of packet rate, the average number of sub-packets per aggregate packet for the AAM algorithm is larger than that for the FIFO algorithm as there are sufficient packets available for selection in the selection window.

The performance in terms of the CCDF of the number of sub-packets for the AAM algorithm is always poorer than that for the SSFS algorithm as the SSFS algorithm tries to aggregate as many sub-packets as possible associated with a large delay.

5.1.3 CDF of the Sub-packet Delay

The CDF of the sub-packet delay represents the probability that the sub-packet delay takes on a value less than or equal to a certain value which allows us to compare the performance in terms of the average sub-packet delay for the different algorithms. The CDF of the sub-packet delay for all three algorithms considered SSFS, FIFO and AAM are shown in Figure 5-7 and Figure 5-8 for the captured traffic trace files 2 and 14 respectively.

From the two graphs, it can be seen that the benchmark FIFO algorithm has the best performance in terms of the CDF of the sub-packet delay and the performance for the AAM algorithm is poorer than that of the FIFO algorithm, but it is better than that of the SSFS algorithm. This means that the average sub-packet delay for the FIFO algorithm is the lowest and for the SSFS algorithm is the largest. As shown in Figure 5-7, the sub-packet delays are less than 0.05 seconds for over 90% of the sub-packets for the FIFO algorithm and 80% of the sub-packets for the AAM algorithm, while only 20% of the sub-packets for the SSFS algorithm. However, in Figure 5-8, the number of sub-packets whose delay is less than 0.05 seconds is reduced to 60% for the FIFO algorithm, 50% for the AAM algorithm and 10% for the SSFS algorithm.

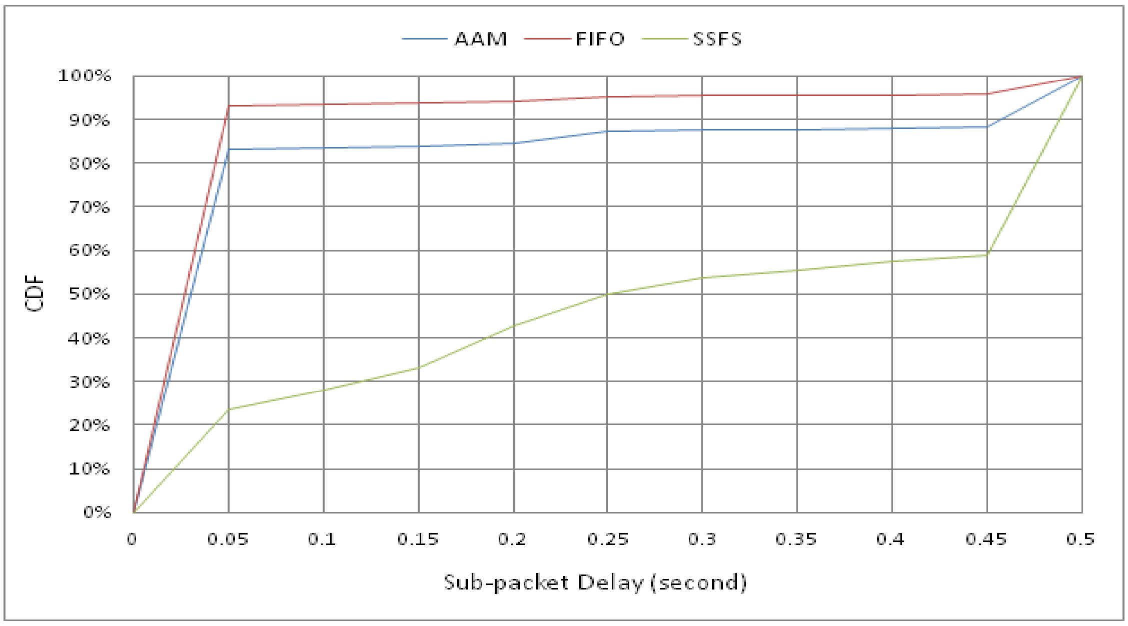


Figure 5-7: The CDF of the sub-packet delay for the captured traffic trace file 2.

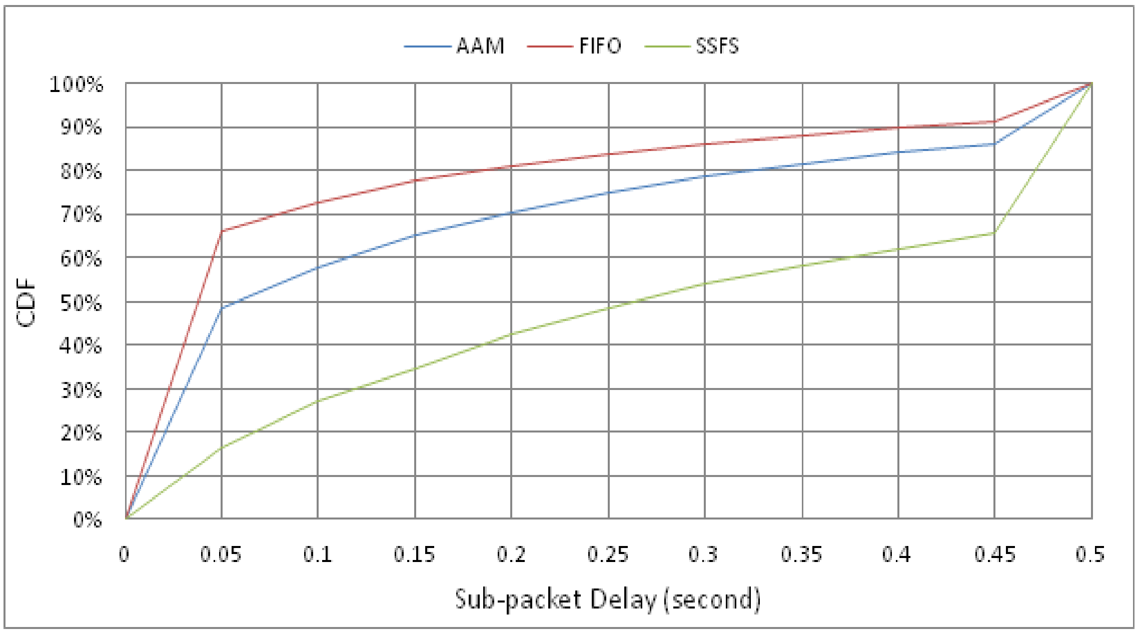


Figure 5-8: The CDF of the sub-packet delay for the captured traffic trace file 14.

By observing the results in appendix C, it can be seen that when the packet rate is low, the performances in terms of the CDF of the sub-packet delay for the AAM and FIFO algorithms are similar. However, they still have a superior performance over the SSFS algorithm in terms of the CDF of the sub-packet delay. This means that the average sub-packet delay for the AAM algorithm is similar to that for the FIFO algorithms but it is smaller than that for the SSFS algorithm when the packet rate is low as the AAM algorithm may need a longer time to wait for the packets to arrive. It also shows that the average sub-packet delay increases as the packet rate decreases for each algorithm as all algorithms may have to wait longer for packets to arrive when the average packet rate is low. Also, it can be seen that the sub-packet delays does not exceed the specified maximum acceptable delay of 0.5 seconds.

5.1.4 Number of Sub-packets against Average Packet Delay

Figure 5-9 and Figure 5-10 show the performance in terms of the number of sub-packets against the average packet delay for the captured traffic trace files 2 and 14 respectively where it can be seen that the performance for the AAM algorithm is superior to that for the FIFO and SSFS algorithms. The reason for this is that the AAM algorithm combines a larger number of sub-packets for a given average packet delay than that of the FIFO and SSFS algorithms. For low values of the number of sub-packets, the AAM algorithm results presented in Figure 5-9 and Figure 5-10 show that the average packet delay decreases as the number of sub-packets increases. However, for large values of the number of sub-packets, the average packet delay does not significantly increase as the number of sub-packets increases. This can be explained as follows: The aggregation process of the AAM algorithm is controlled by the maximum acceptable delay and the target aggregate packet size thresholds. When the number of sub-packets is small, this indicates that the aggregation process is dominated by the maximum acceptable delay threshold which means that a significant number of aggregate packet delays achieved the maximum acceptable delay. When the number of sub-packets is large, this indicates that the AAM algorithm aggregation process is dominated by the target aggregate packet size requirement which means that a significant number of aggregate packet sizes reached the target aggregate packet size. The main reason why the average packet delay is so small when the number of sub-packets is 1 is that the size of the first sub-packet achieves the target aggregated packet size and therefore it is immediately transmitted.

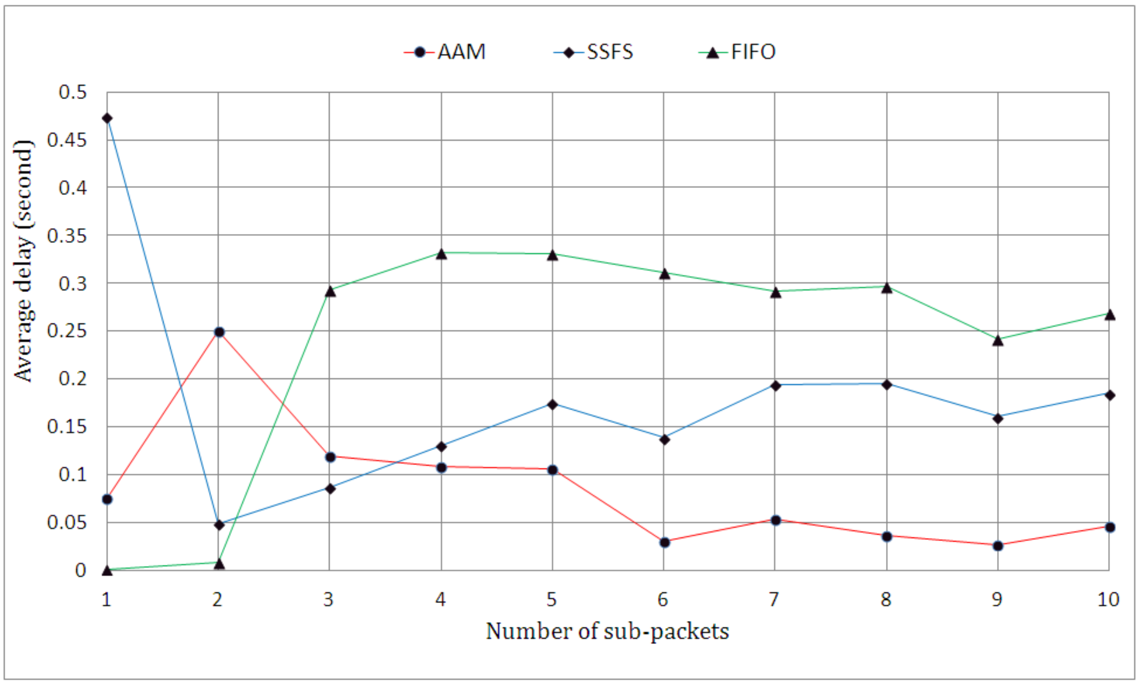


Figure 5-9: The number of sub-packets against the average packet delay for the captured traffic trace file 2.

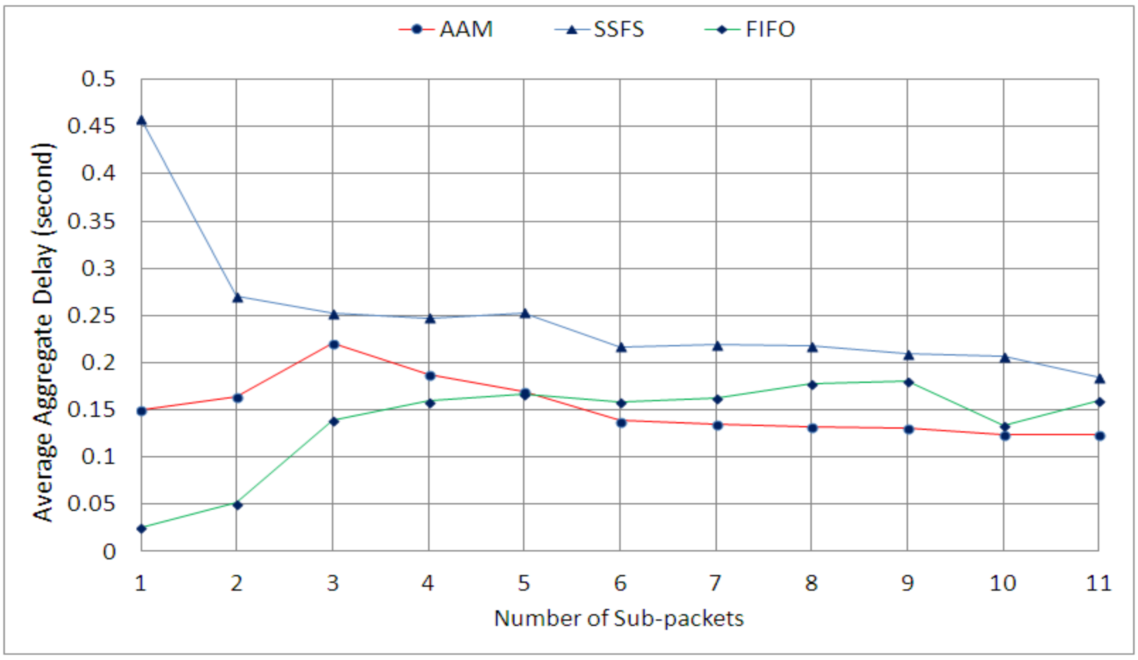


Figure 5-10: The number of sub-packets against the average packet delay for the captured traffic trace file 14.

The two figures have quite different shapes as the two captured traffic trace files have different packet rates which are shown in Table 4-2. The packet rate of the captured traffic trace file 2 is 114 pps while it is 19.2 pps for the captured traffic trace file 14. From all the graphs for the 16 captured traffic trace files presented in appendix D, when the packet rate is large, the performance in terms of the number of sub-packets against the average packet delay for the AAM algorithm is the best for all three algorithms considered. For low values of the packet rate, the performance for the AAM algorithm is similar to that for the FIFO algorithm as the number of sub-packets that can be combined into an aggregate packet by the AAM algorithm is similar to that for the FIFO algorithm because there are insufficient packets available for selection from the selection window.

5.1.5 Conclusion

The performances in terms of the selection window size, the CCDF of the number of sub-packets, the CDF of the sub-packet delay and the number of sub-packets against the average packet delay for all three algorithms considered have been presented. The following conclusions can be drawn:

- The AAM algorithm is an adaptive algorithm which can operate over a wide range of different traffic loads. The selection window size is adaptively adjusted to follow the variations in the packet rate of the traffic load. The size of the selection window has an impact on the performance in terms of the CCDF of the number of sub-packets. In particular, the larger the size of the selection window, the better the performance in terms of the CCDF of the number of sub-packets.
- The AAM algorithm has a better performance in terms of the CCDF of the number of sub-packets than that for the benchmark FIFO algorithm but is poorer than that for the SSFS algorithm. This means that the average number of sub-packets per aggregate packet generated by the AAM algorithm is larger than that for the FIFO algorithm but is smaller than that for the SSFS algorithm.

- The AAM algorithm has a better performance in terms of the CDF of the sub-packet delay than that for the SSFS algorithm but is poorer than that for the FIFO algorithm. This means that the AAM algorithm has a shorter average sub-packet delay than that for the SSFS algorithm but is longer than that for the FIFO algorithm.
- The performance in terms of the number of sub-packet against the average packet delay for the AAM algorithm is the best for all three algorithms considered. This means that the AAM algorithm can aggregate a larger number of sub-packets for a given average packet delay than the other two algorithms considered due to its adaptive nature. However, when the packet rate is low, the performance for the AAM algorithm is similar to that for the FIFO algorithm as there are insufficient packets available for selection in the selection window.

5.2 Performance in Wireless Networks

In this section, we will present the results for the AAM algorithm implemented in a wireless network test scenario which was described in chapter 4. The performance of the FIFO and SSFS algorithms will also be shown. In this test scenario, the AAM algorithm is implemented in a wireless network under two different operating environments. The first one is an ideal environment where transmission errors are absent and the PHY rate adaption mechanism is also disabled and the other one is the same wireless network but with transmission errors present where the PHY rate adaption mechanism is employed. The objective of this test scenario is to demonstrate that the AAM algorithm has a superior performance over the other two algorithms in terms of the aggregation trade-off. It is also to demonstrate that the AAM algorithm is a robust algorithm which has a superior performance over the other two algorithms in terms of the throughput in error-prone wireless networks.

5.2.1 Performance in an Ideal Wireless Network

The AAM, FIFO and SSFS algorithms were implemented in an ns-3 simulation wireless network where transmission errors are absent and the PHY rate adaption mechanism is disabled. In this scenario, as described in section 4.6.2, the *DataRate* parameter of *OnOffApplication* is modified in order to change the data rate by changing the interval between the raw packets of the captured traffic trace files. The interval can be changed by using the *OffTime* parameter of *OnOffApplication* module. Figure 5-11 and Figure 5-12 show the performances in terms of the throughput against the data rate for the different IEEE 802.11 PHY rates for the captured traffic trace files 2 and 14 respectively. It can be seen that the AAM algorithm has the best performance in terms of the throughput for the 8 different PHY rates for all three algorithms considered. The FIFO algorithm has the poorest performance in terms of the throughput. The improvement in the throughput increases as the PHY rate increases. When the PHY rate is 6 Mbps, the throughput improvement is 6% compared with 30% when the PHY rate is 54 Mbps for the AAM compared to the FIFO algorithm. For example, as shown in Figure 5-11, when the PHY rate is fixed at 54 Mbps, the maximum throughput for the FIFO algorithm is 20 Mbps and for the SSFS algorithm it is 23 Mbps, while the maximum throughput for the AAM algorithm is as large as 26 Mbps. This is because the AAM algorithm can combine a larger number of packets per aggregate packet for a given delay in wireless networks where the delay is defined as the maximum acceptable delay.

From Figure 5-11 and Figure 5-12, it can be seen that when the PHY rate is fixed and the data rate is low, the AAM, FIFO and SSFS algorithms have similar performances in terms of their throughput. However, if the data rate is large, the AAM algorithm has a superior performance compared to the other two algorithms in terms of the throughput. For example, when the data rate is below 19 Mbps with a PHY rate fixed at 54 Mbps, the performances in terms of the throughput are similar for all three algorithms considered; while if the data rate is greater than 23 Mbps, the throughput for the AAM algorithm is greater than that of both the FIFO and SSFS algorithms. The reason is that when the data rate is low, the performance of the AAM algorithm degrades to that for the FIFO algorithm as there are insufficient packets to be selected from the selection window. When the data rate is large, there are sufficient packets for selection in the selection window to generate a larger size aggregate packet for a given delay by the AAM algorithm.

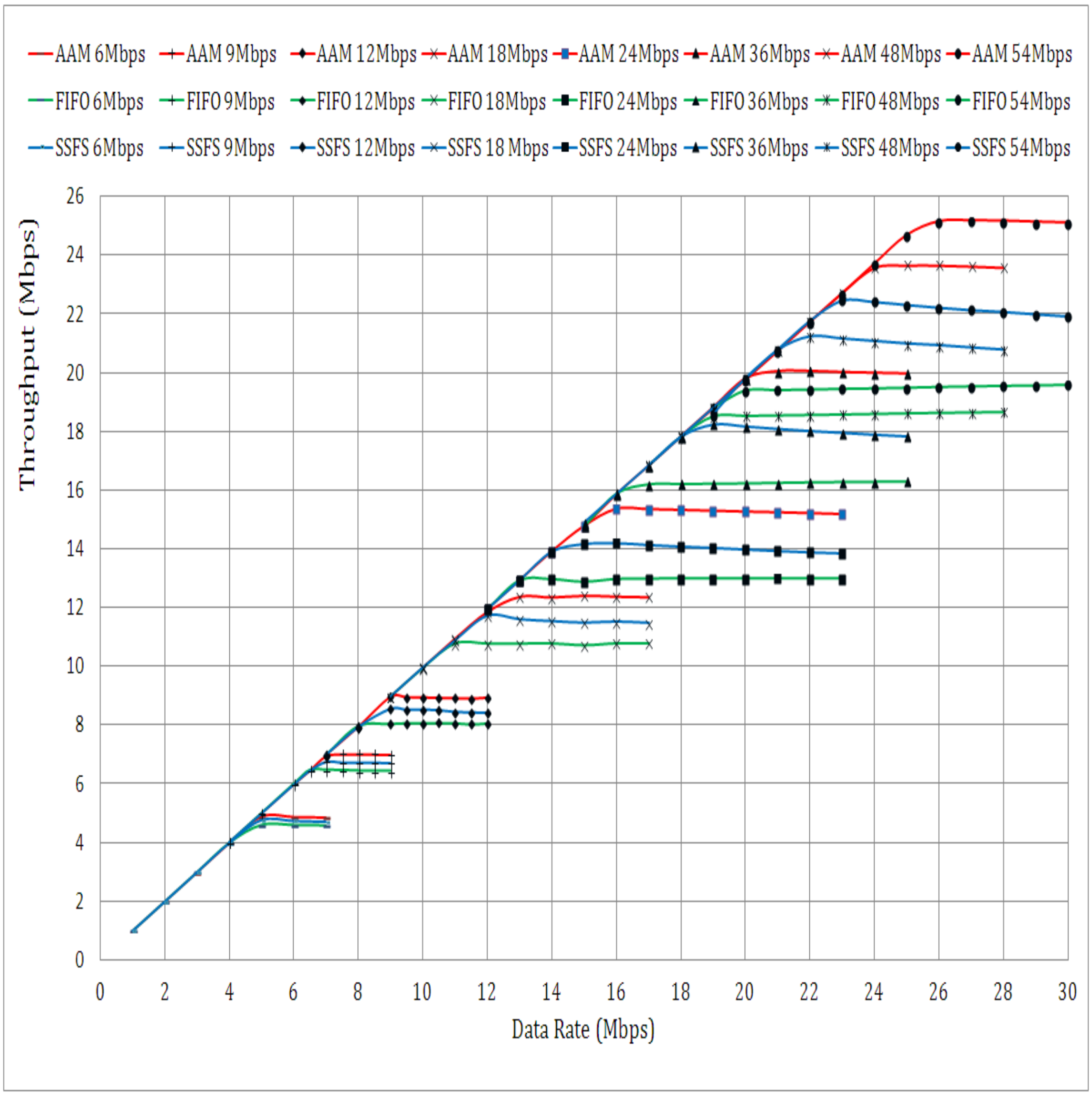


Figure 5-11: The throughput against data rate for the different PHY rates for the captured traffic trace file 2.

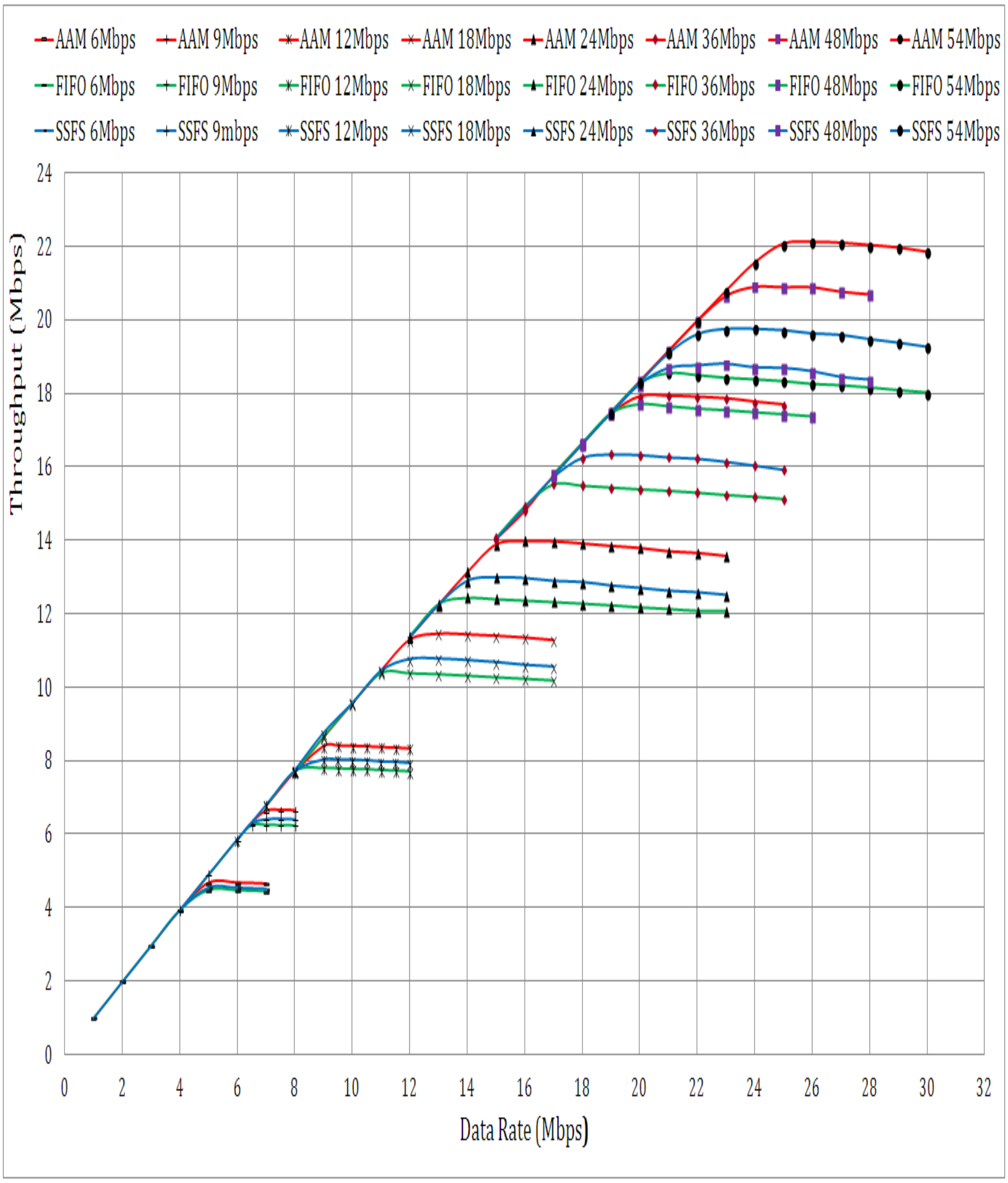


Figure 5-12: The throughput against data rate for the different PHY rates for the captured traffic trace file 14.

Figure 5-13 and Figure 5-14 present the average delay against the data rate for the different PHY rates for the captured traffic trace files 2 and 14 respectively. It can be seen here that the performance in terms of the average delay for the AAM algorithm is the best for all three algorithms considered and the performance for the FIFO algorithm is the poorest. When the PHY rate is fixed at 6 Mbps, the reduction in the average delay is 5% compared with 15% when the PHY rate is fixed at 54 Mbps for the AAM algorithm compared to the FIFO algorithms. This is because the AAM algorithm takes the least time to transmit a packet from the source to the destination. Moreover, the average delay decreases as the PHY rate increases. For example, the average delay is 1.15 milliseconds for the AAM algorithm when the PHY rate is fixed at 6 Mbps and the data rate is 4 Mbps. When the PHY rate is fixed at 54 Mbps and the data rate is 26 Mbps, the average delay is 0.153 milliseconds. The AAM algorithm has the best performance in terms of the average delay compared to the FIFO and SSFS algorithms. The reason is that when the data rate is large with a fixed PHY rate, there are sufficient packets available for selection from the selection window, so the AAM algorithm can on average produce larger size aggregate packets to reduce the average delay which will be explained in the next section. However, there is not a significant decrease in the average delay as the data rate increases for a fixed PHY rate for all three algorithms considered. This is because the size of the aggregate packet cannot increase indefinitely as a target aggregate packet size exists. So the average transmission delay of each aggregate packet has a lower limit. This is why the average delay level-offs at high data rates.

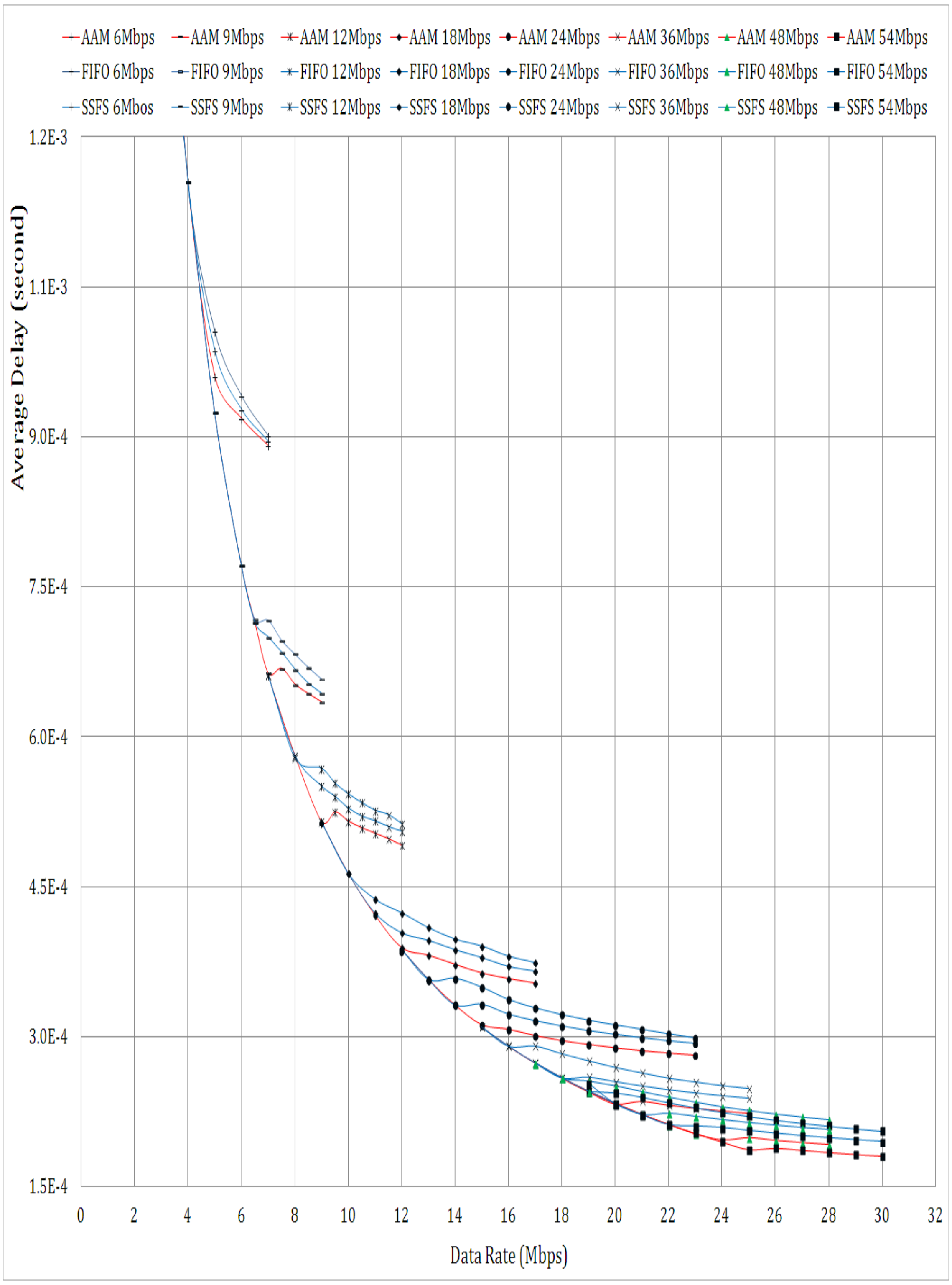


Figure 5-13: The average delay against the data rate for the different PHY rates for the captured traffic trace file 2.

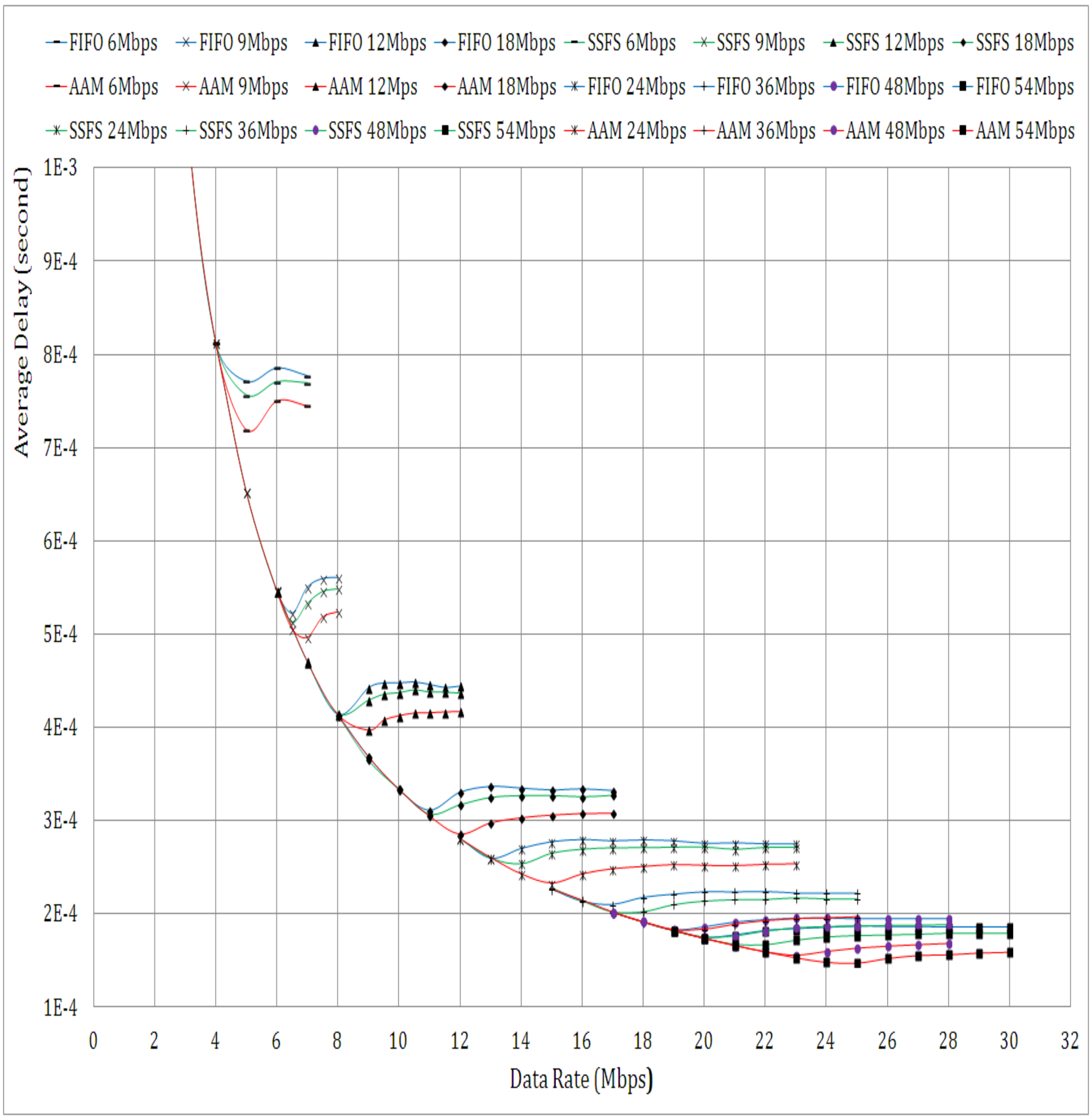


Figure 5-14: The average delay against the data rate for the different PHY rates for the captured traffic trace file 14.

Figure 5-15 and Figure 5-16 present the performance in terms of the average aggregate packet size for the different PHY rates under saturation for all three algorithms considered. It can be seen that the AAM algorithm has the largest average aggregate packet size for the different PHY rates in saturated wireless networks, while the FIFO algorithm has the smallest average aggregate packet size. It can be seen that the AAM algorithm can improve the average aggregate packet size by as much as 50% compared to the FIFO algorithm for the captured trace file 2 as the average size of an AAM aggregate packet is over 1200 bytes while that of a FIFO aggregate packet is 800 bytes. As discussed in chapter 4, the minimum average delay is determined by the average size of the aggregate packet with a fixed PHY rate under saturation. The minimum average delay for the AAM algorithm is the smallest for the different PHY rates for all three algorithms considered which was shown in Figure 5-13 and Figure 5-14. In general, the larger the aggregate packet size, the larger the number of sub-packets per aggregate packet.

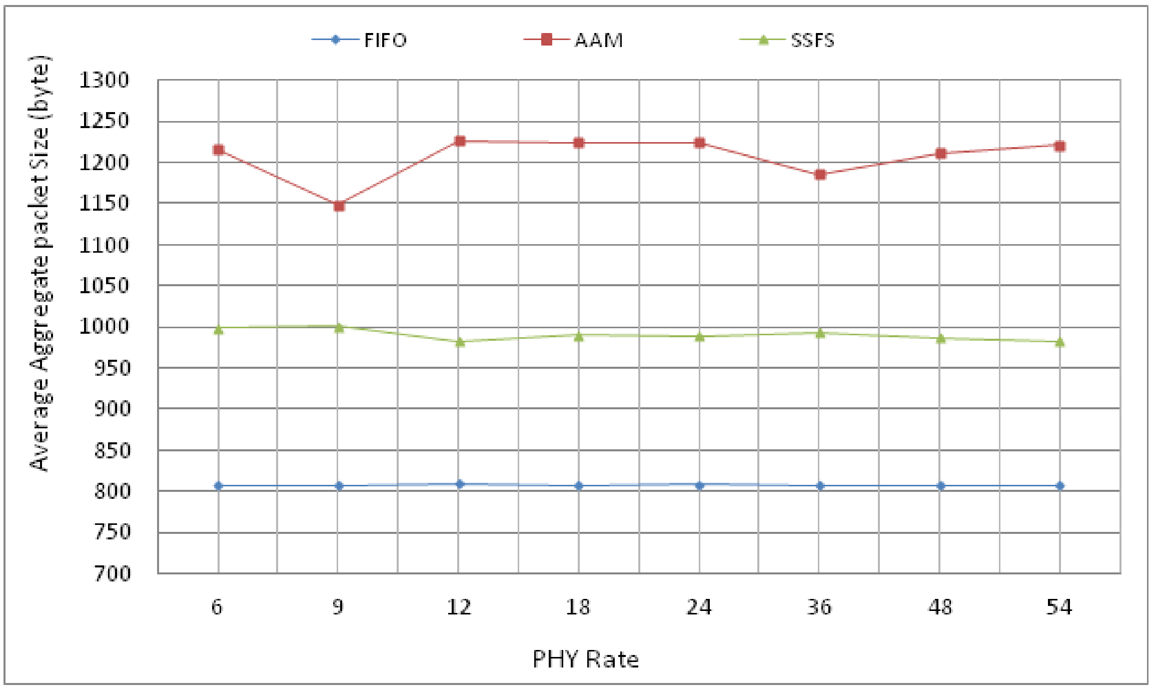


Figure 5-15: The average aggregate packet size for the different PHY rates under saturation for the captured traffic trace file 2.

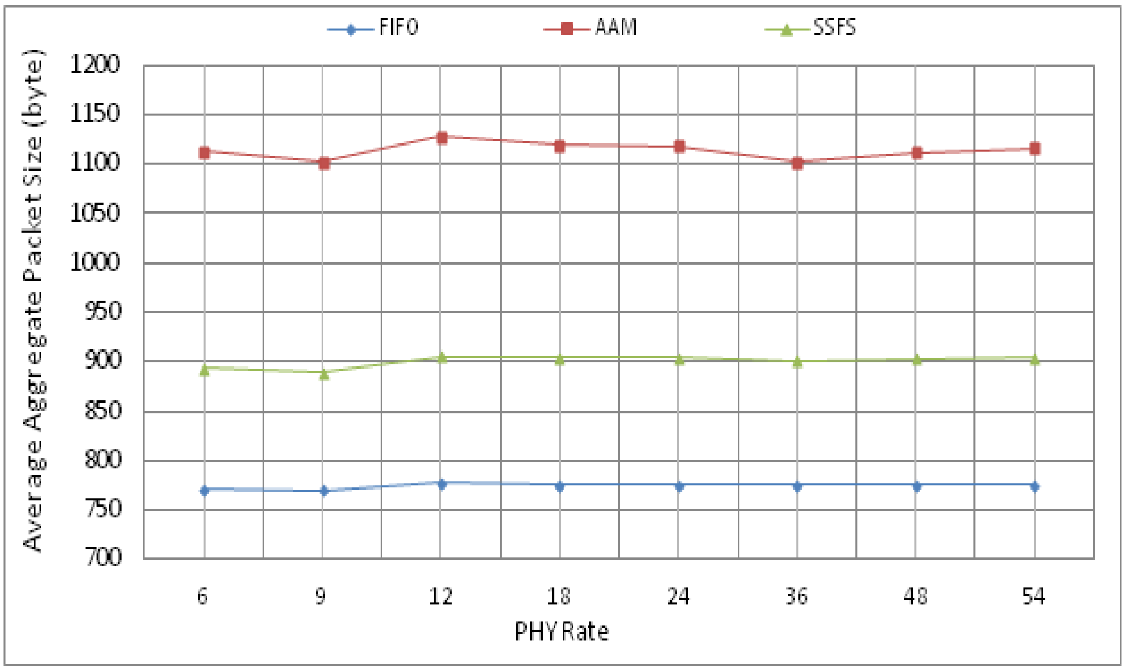


Figure 5-16: The average aggregate packet size for the different PHY rates under saturation for the captured traffic trace file 14.

Here, the mean square deviation is employed to analyze the extent to which the aggregate packet size is spread out from the target aggregate packet size. *The mean square deviation is defined as the mean value of the square of the difference between the target aggregate packet size and the aggregate packet size.*

The performances in terms of the mean square deviation for the different PHY rates under saturation for the captured traffic trace files 2 and 14 are shown in Figure 5-17 and Figure 5-18 respectively. The two figures show that the AAM algorithm has the smallest value of mean square deviation which means that the aggregate packet sizes tend to be the closest to the target aggregate packet size and the average aggregate packet size is the largest. However, the mean square deviation for the FIFO algorithm is the largest which means that the aggregate packet sizes is the farthest from the target aggregate packet size on average and the average aggregate packet size is the smallest. The SSFS algorithm has a superior performance over the FIFO algorithm in terms of the mean square deviation for the different PHY rates under saturation.

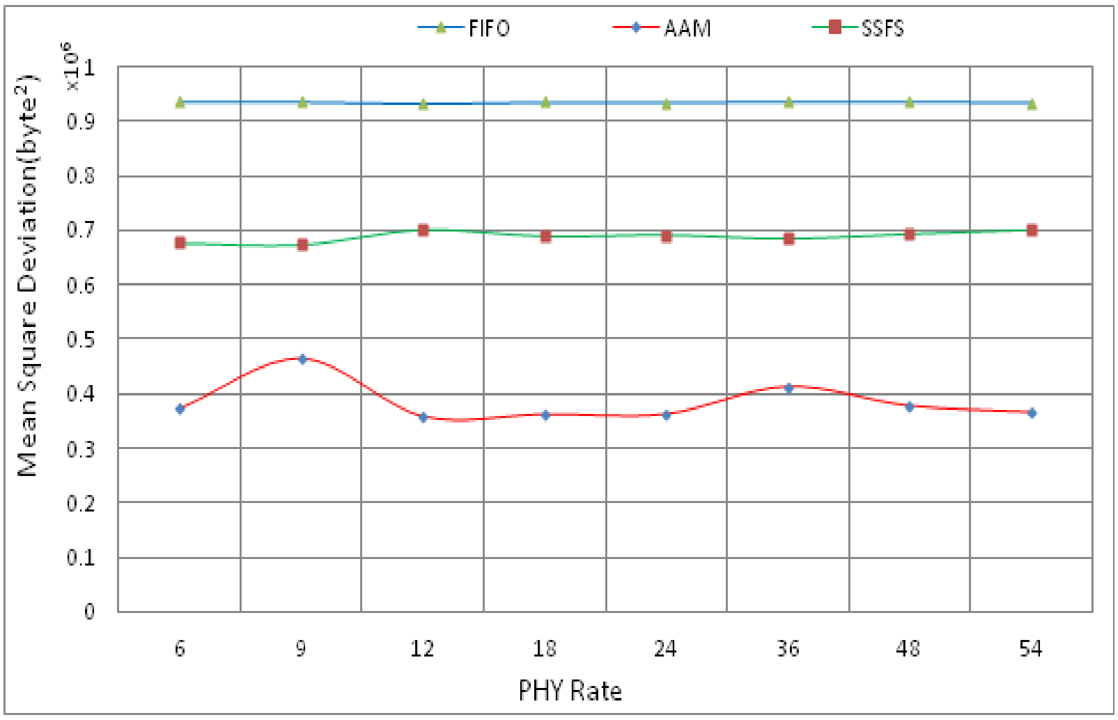


Figure 5-17: The mean square deviation for the different PHY rates under saturation for the captured traffic file 2.

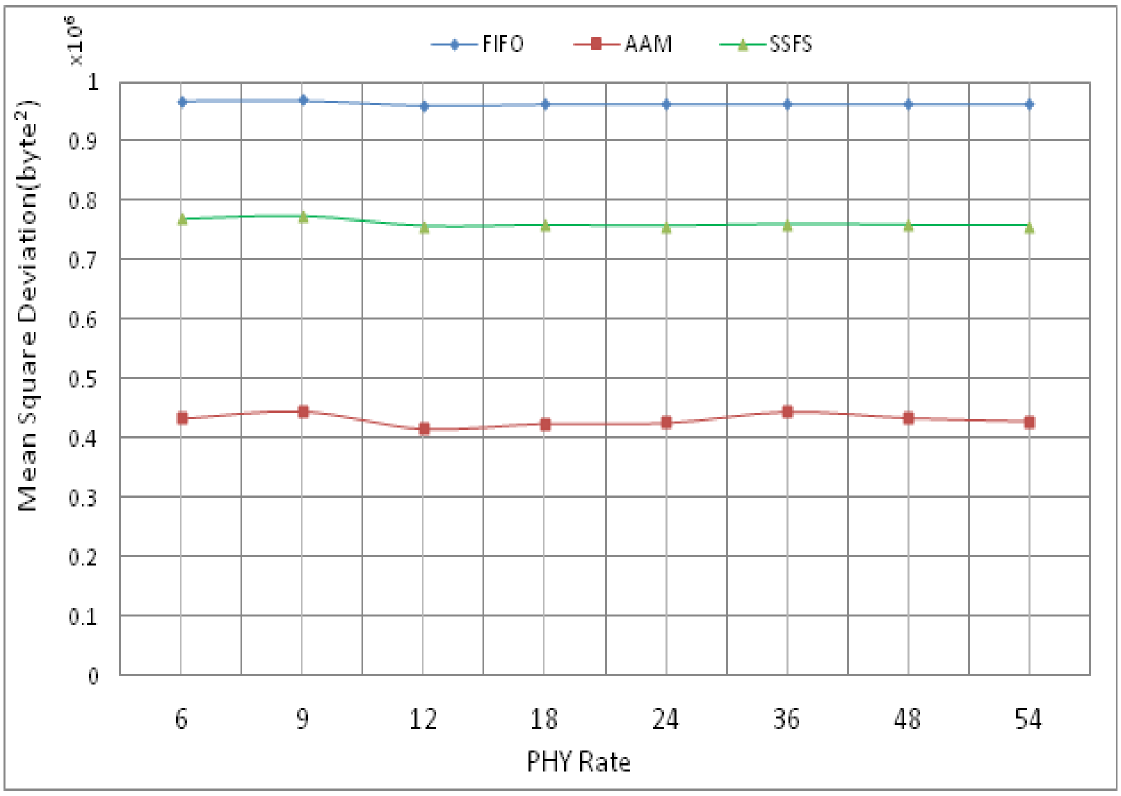


Figure 5-18: The mean square deviation for the different PHY rates under saturation for the captured traffic file 14.

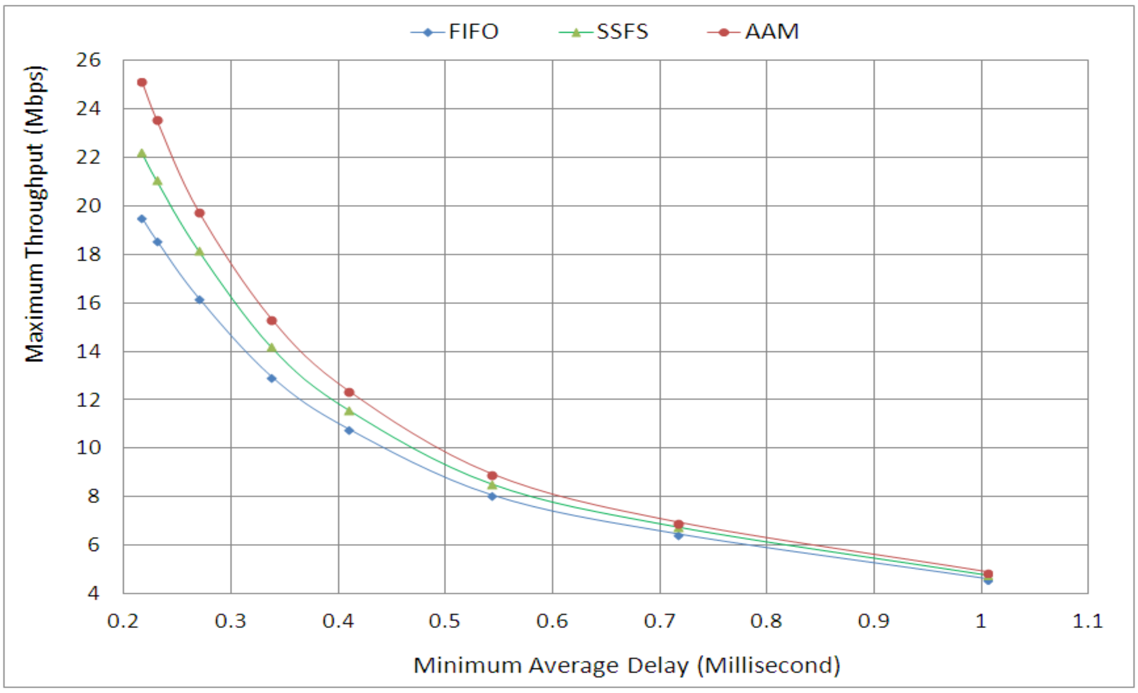


Figure 5-19: The aggregation trade-off in terms of achieving the maximum average throughput with the minimum average delay for the capture traffic trace file 2.

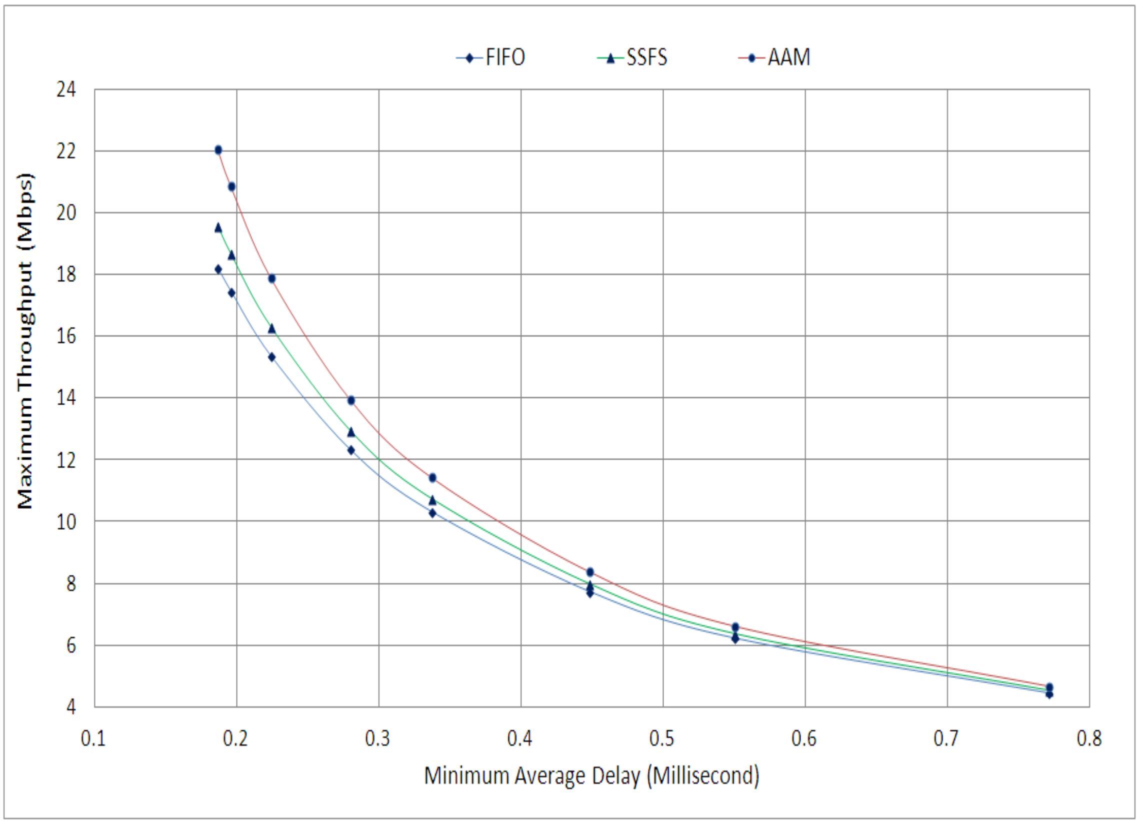


Figure 5-20: The aggregation trade-off in terms of achieving the maximum average throughput with the minimum average delay for the capture traffic trace file 14.

Figure 5-19 and Figure 5-20 show the performance in terms of the aggregation trade-off in achieving the maximum average throughput with the minimum average delay for the captured traffic trace files 2 and 14 respectively. It can be seen here that the AAM algorithm has the best performance in terms of the aggregation trade-off due to the adaptive nature of the AAM algorithm. As the minimum average delay is dependent upon the PHY rate, a low value of the minimum average delay corresponds to a large PHY rate. From these two figures, it can be seen that the improvement in the aggregation trade-off increases as the PHY rate increases. When the PHY rate is large (i.e. for a low value of minimum average delay), the aggregation trade-off improvement is the best for the AAM algorithm compared to the other two algorithms as the AAM algorithm can successfully transmit more data for a given minimum average delay. When the PHY rate is low (i.e. for a large value of minimum average delay), the performance in terms of aggregation trade-off for the AAM algorithm is similar to that for the other two algorithms. This shows that when the PHY rate is low, the AAM algorithm still can improve the performance in terms of aggregation trade-off compared to the FIFO algorithm but cannot significantly improve it.

Conclusion

Based on the previous discussions for the AAM algorithm implemented in a wireless network without transmission errors present and with the PHY rate adaption mechanism disabled, a number of conclusions can be draw:

- The AAM algorithm has the best performance in terms of the throughput for all three algorithms considered and the improvement in the throughput increases as the PHY rate increases. It improves the throughout from 6% with a PHY rate fixed at 6 Mbps up to 30% with a PHY rate fixed at 54 Mbps compared to the FIFO algorithm.

- The AAM algorithm has a similar performance to the other algorithms in terms of the throughput when the data rate is low with a fixed PHY rate. The reason is that the wireless network is unsaturated which can be determined by checking the AAM algorithm to see if it needs to wait for further packet arrival or not. If it needs to do it, the network is unsaturated. When the data rate is large, the AAM algorithm has a superior performance over the other algorithms in terms of the throughput as there are sufficient packets for selection from the selection window to generate a larger size aggregate packet for a given delay.
- The AAM algorithm has the best performance in terms of the average delay. The AAM algorithm has the best performance in terms of the average delay for all three algorithms considered and it reduces the average delay by up to 15% compared to the FIFO algorithm with a PHY rate at 54 Mbps. The average delay decreases as the PHY rate increases for all three algorithms considered.
- The AAM algorithm has the best performances in terms of the average aggregate packet size and the mean square deviation under saturation for all three algorithms considered. The average size of an AAM aggregate packet is 1.5 times that for a FIFO aggregate packet. This demonstrates that the AAM algorithm produces the largest average size of the aggregate packet. The AAM algorithm has the smallest value of the mean square deviation which demonstrates that it has the largest number of aggregate packets whose sizes are closest to the target aggregate packet size.
- The AAM algorithm has the best performance in terms of the aggregation trade-off in achieving the maximum average throughput with the minimum average delay for all three algorithms considered as it achieves the largest throughput at the cost of the least average delay.

5.2.2 Performance in an Error-Prone Wireless Network

The AAM, SSFS and FIFO algorithms are implemented in the same wireless network simulation but with transmission errors present and the PHY rate adaption mechanism enabled. The transmission errors are characterized by the BER and the PHY rate adaption mechanism is implemented by the *AarfWifiManager* module in the ns-3 simulator. As described in chapter 4, the objective of the PHY rate adaption mechanism is to improve the throughput by selecting the most effective PHY rate to transmit the frames. However, in this case it will be shown that the AARF mechanism is not the main factor contributing to the throughput improvement when using the AAM algorithm.

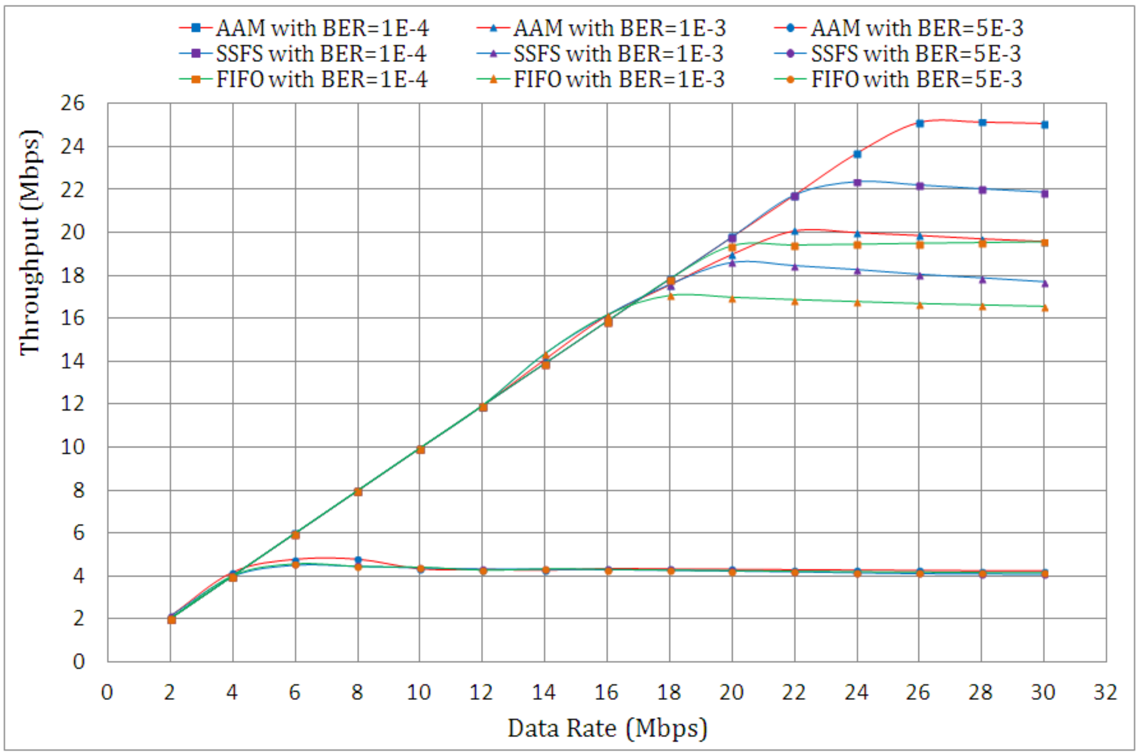


Figure 5-21: The throughput against data rate for the different BERs for the captured trace file 2.

Figure 5-21 and Figure 5-22 show the performance in terms of the throughput against data rate for the different BERs (i.e. 10^{-4}, 10^{-3} and 5×10^{-3}) for the captured traffic trace files 2 and 14 respectively. The graphs show that the throughput improvement increases as the BER decreases. As can be seen, the AAM algorithm has the best performance in terms of the throughput for all three algorithms considered when the BER is 10^{-4} and 10^{-3}. For example, for the captured traffic trace file 2 the throughput for the AAM algorithm is 25 Mbps while it is 19.5 Mbps for the FIFO algorithm with a fixed BER of 10^{-4} which corresponds to a throughput improvement of 28%. This value of the throughput improvement is 18% for a BER of 10^{-3}. However, when the BER is large (i.e. 5×10^{-3}) the AAM algorithm has a similar performance to that for the FIFO and SSFS algorithms in terms of the throughput. The reason is that when the BER is small, the AAM algorithm has more packets than the other algorithms to be successfully transmitted; When the BER is large, a significant number of aggregate packets are corrupted and therefore need to be re-transmitted.

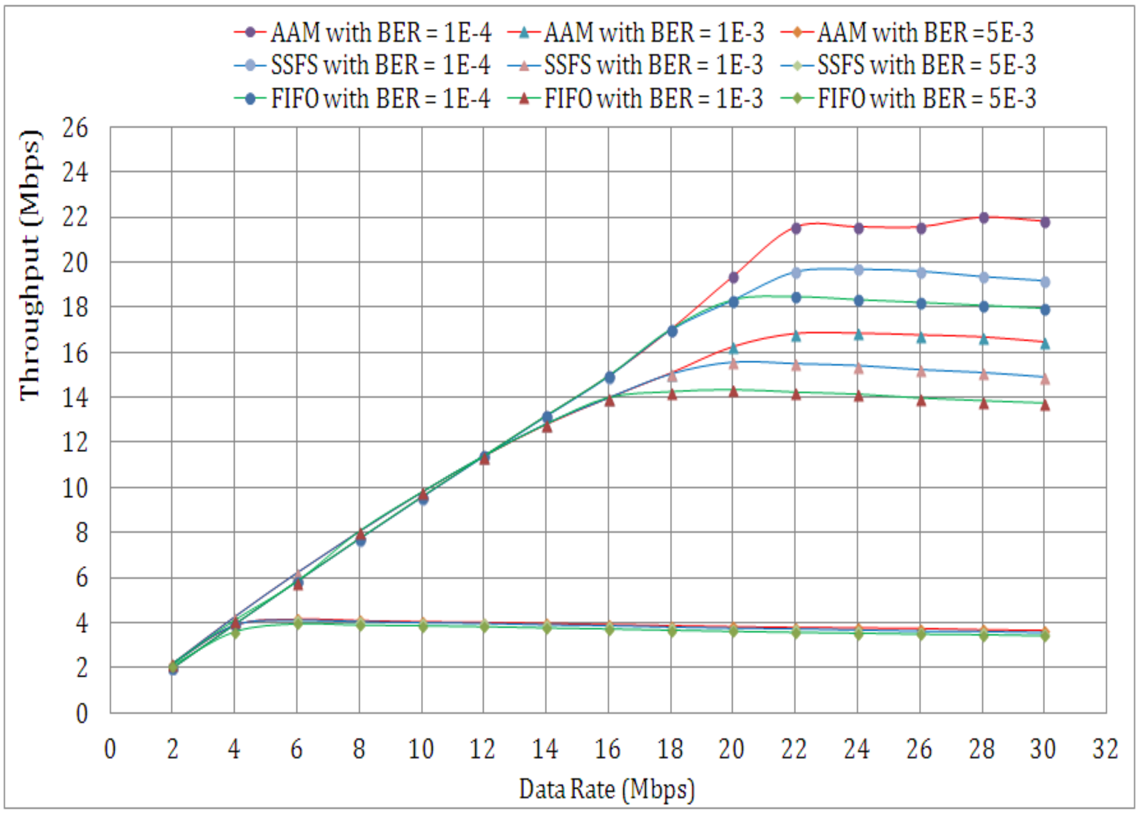


Figure 5-22: The throughput against data rate for the different BERs for the captured trace file 14.

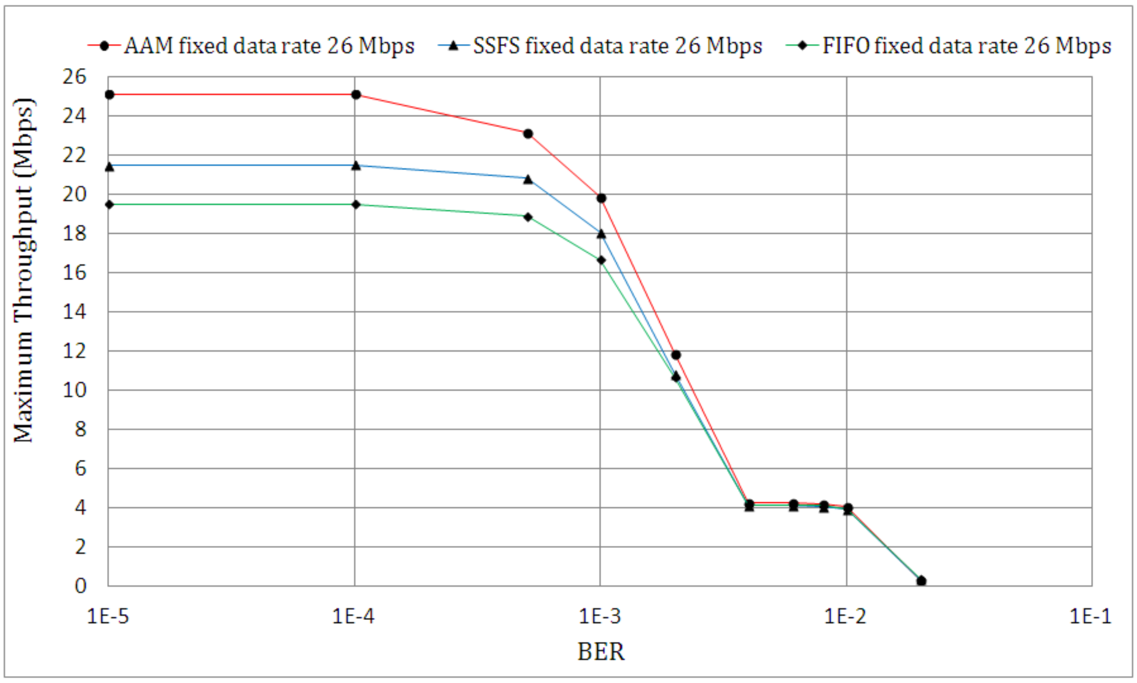


Figure 5-23: The maximum throughput against BER with a fixed data rate of 26 Mbps for the captured traffic trace file 2.

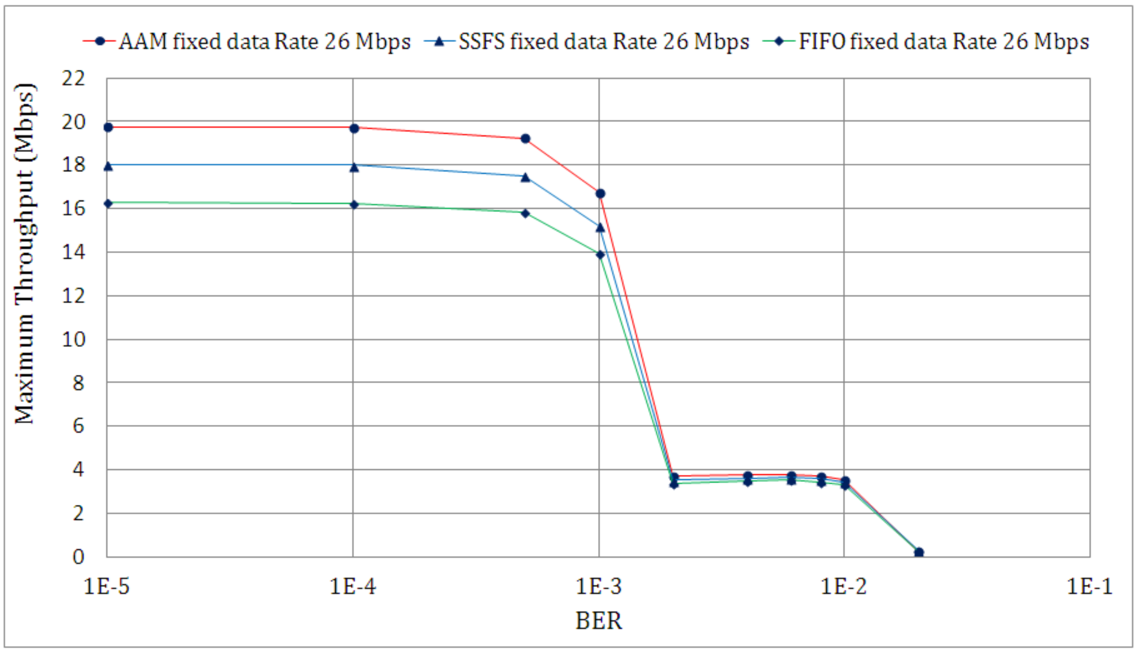


Figure 5-24: The maximum throughput against BER with a fixed data rate of 26 Mbps for the captured traffic trace file 14.

The performances in terms of the maximum throughout against BER with a fixed data rate of 26 Mbps for the captured traffic trace files 2 and 14 are presented in Figure 5-23 and Figure 5-24 respectively. The maximum throughput improvement decreases as the BER increases. When the BER is 10^{-3}, the AAM algorithm can improve the maximum throughput by 20% compared to the FIFO algorithm. For low values of BER, the AAM algorithm can significantly improve the maximum throughput; while for large values of BER, the AAM algorithm has a similar performance to that for the other algorithms in terms of the maximum throughput. The maximum throughputs for all three algorithms considered decrease as the BER increases. However, the maximum throughput levels-off when the values of BER are between 4×10^{-3} and 10^{-2}. The reason for this is that a significant number of packets are transmitted at the lowest PHY rate (i.e. 6 Mbps) which has been determined by the AARF mechanism. The maximum throughput is almost zero when the BER is greater than 2×10^{-2} as only a few packets are successfully transmitted. This result demonstrates that when the BER is large (e.g. 2×10^{-2}), the performances in terms of the maximum throughput for all three algorithms considered are similar as a significant number of packets are corrupted.

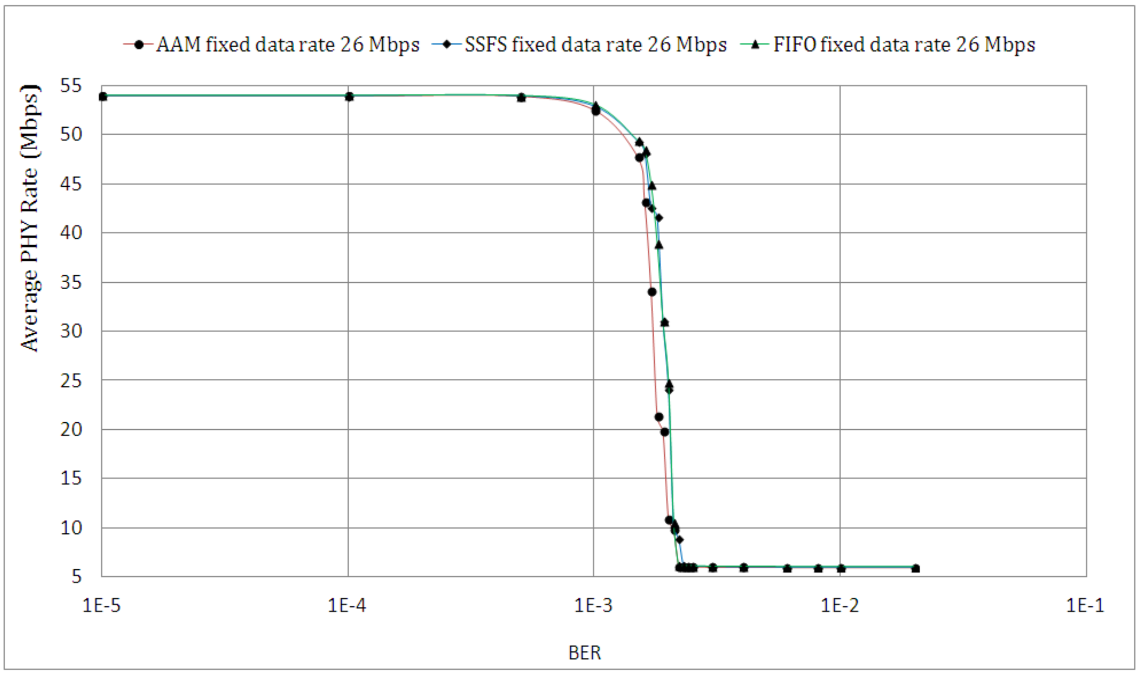


Figure 5-25: The average PHY rate against BER with a fixed data rate of 26 Mbps for the captured traffic trace file 2.

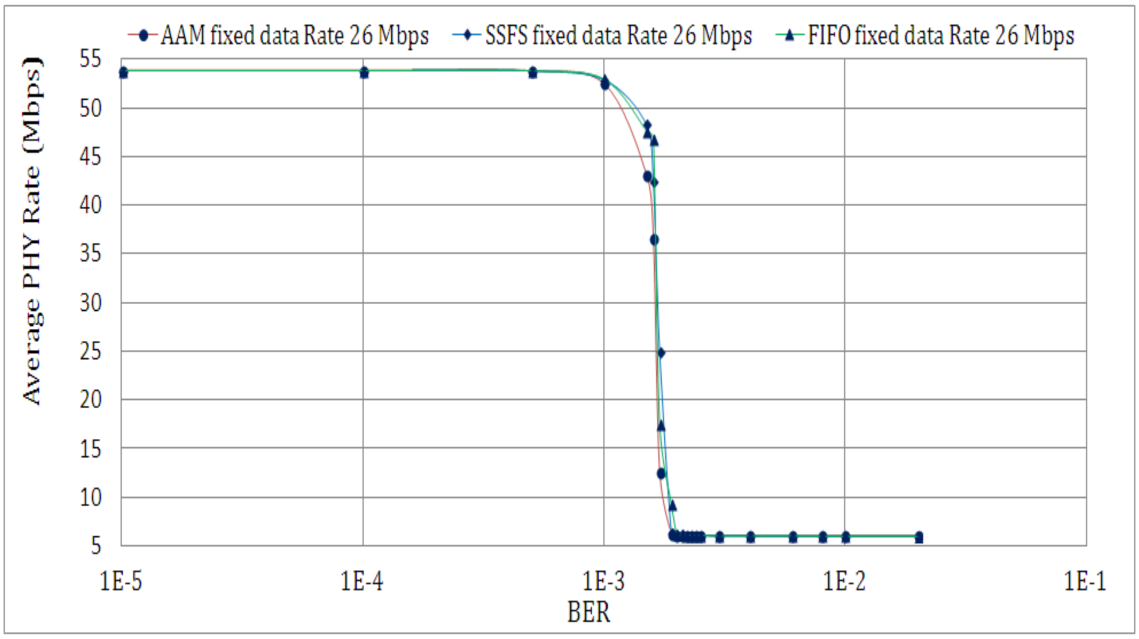


Figure 5-26: The average PHY rate against BER with a fixed data rate of 26 Mbps for the captured traffic trace file 14.

The performances in terms of the average PHY rate against BER with a fixed data rate of 26 Mbps for the captured traffic trace files 2 and 14 are presented in Figure 5-25 and Figure 5-26 respectively. It can be clearly seen that the performances in terms of the average PHY rates are similar for all three algorithms considered. When the BER is less than 10^{-3} the PHY rate is close to 54 Mbps and the PHY rate is almost 6 Mbps if the BER is greater than 4×10^{-3}. The PHY rate sharply decreases from 54 Mbps to 6 Mbps when the BER increases from 10^{-3} to 4×10^{-3}. The result shows that the PHY rate adaption mechanism AARF is not the main factor contributing to the improvement in the throughput by using the AAM algorithm but rather the adaptive nature of the AAM algorithm.

Conclusion

Based on the previous discussions for the AAM algorithm implemented in a wireless network with transmission errors present and with the PHY rate adaption mechanism enabled, the following conclusions can be draw:

- The AAM algorithm has a superior performance compared to the other two algorithms in terms of the throughput. Moreover, the improvement in the throughput decreases as the BER increases. When the BER is 10^{-4}, the AAM algorithm improves the throughput by 28%, while it cannot significantly improve the throughput when the BER is large (i.e. 5×10^{-3}).
- The PHY rate adaption mechanism AARF is not the main factor responsible for the improvement in the throughput as the performances are similar for all three algorithms considered in terms of the average PHY rate against BER with a fixed data rate of 26 Mbps. The main factor that contributes to the improvement in the throughput is the adaptive nature of the AAM algorithm.

5.3 Chapter Summary

In this chapter the main findings of the analysis of the AAM algorithm implemented in two test scenarios have been presented. The performance of the AAM algorithm implemented in the first scenario involving the aggregation process only demonstrated that the AAM algorithm is an adaptive packet aggregation algorithm which can operate over a wide range of different traffic loads by employing a tunable selection window scheme and a hybrid selection strategy. An analysis of the CCDF of the number of sub-packets and the CDF of the sub-packet delay demonstrated that the AAM algorithm has a superior performance compared to the FIFO and SSFO algorithms.

The AAM algorithm also has the best performance in terms of the number of sub-packets against the average packet delay as it can combine a larger number of sub-packets per aggregate packet for a given average packet delay. It was also shown that when the packet rate is low, the performances are similar in terms of the number of sub-packets against the average packet delay as the performance for the AAM algorithm degrades towards that for the FIFO algorithm.

The results of the second scenario where the AAM algorithm has been deployed in a simulated wireless network with and without transmission errors present were presented. In the ideal wireless network (i.e. without transmission errors present) where the PHY rate adaption mechanism is not employed, the AAM algorithm has the best performance in terms of the throughput and the average delay for the different IEEE 802.11 PHY rates. The AAM algorithm can improve the throughput by 30% and reduce the average delay by 15% compared to the FIFO algorithm with a fixed PHY rate of 54 Mbps. However, when the data rate is low with a fixed PHY rate, the performances are similar in terms of the throughput and the average delay for all three algorithms considered. The performances in terms of the average aggregate packet size and the mean square deviation for the AAM algorithm are the best for all three algorithms considered. The AAM algorithm also has the best performance in terms of the aggregation trade-off in achieving the maximum average throughput with the minimum average delay for all three algorithms considered. The improvement in this performance increases as the PHY rate increases. When the PHY rate is 6 Mbps, the performances are similar in terms of the aggregation trade-off in achieving the maximum average throughput with the minimum average delay for all three algorithms considered.

In the case of a wireless network with transmission errors present where the PHY rate adaption mechanism is employed, the AAM algorithm has the best performance in terms of the throughput when the BER is low. The AAM algorithm can improve the throughput by 28% compared to the FIFO algorithm when the BER is 10^{-4}. The improvement in the throughput decreases as the BER increases. When the BER is 10^{-3}, the AAM algorithm can improve the throughout by 20% compared to the FIFO algorithm. However, the throughputs are almost zero for all three algorithms considered when the BER is 2×10^{-2}. Therefore the AAM algorithm cannot provide any significant benefits in a simulated wireless networks with large values of BER as a significant number of packets are corrupted and need to be re-transmitted. Furthermore, the results demonstrate that the PHY rate adaption mechanism is not the main factor contributing to the improvement in the throughput but rather the adaptive nature of the AAM algorithm.

Chapter 6
Conclusions and Future Work

Packet aggregation is a technique which combines a number of data packets into a single large data packet in order to achieve higher throughputs by reducing the overhead associated with protocol headers in packet-based communications networks. In the IEEE 802.11n standard, two packet aggregation algorithms have been proposed to improve the throughput: A-MSDU and A-MPDU. However, a packet aggregation algorithm can also increase the delay as it may need to wait for more packets to arrive in order to be aggregated into an aggregate packet. Consequently, there exists a trade-off between the throughput and the delay for all packet aggregation algorithms. However, most of the packet aggregation algorithms proposed so far just tend to optimize a single metric, i.e. either to maximize throughput or to minimize delay. In other words, they do not take account of the varying nature of the mixed traffic load particularly the random nature of the packet size and inter-arrival time.

In this thesis, an adaptive packet aggregation algorithm called the Adaptive Aggregation Mechanism (AAM) has been proposed to achieve the best aggregation trade-off in realizing the maximum average throughput with the minimum average delay compared to two other packet aggregation algorithms (i.e. FIFO and SSFS) for different traffic loads. The AAM algorithm is essentially a feedback control system that can operate over a wide range of different traffic loads by employing an adaptive selection window mechanism and a hybrid selection strategy. There are three elements to the AAM algorithm: Adjustable Aggregation Algorithm (A^3), Aggregate Packer Analyzer (APA) and Aggregate Tuning Algorithm (ATA). The A^3 selects packets from the selection window in the input buffer based on a hybrid selection strategy and then aggregates them together in the output buffer. The hybrid selection strategy results in the first sub-packet being selected based on the packet arrival time and the other sub-packets being selected based on the packet size. The selection window size can be adaptively adjusted. The APA analyses the number of sub-packets and the aggregate packet delay between the current and previous aggregate packet. The ATA determines the size of the selection window for the next aggregate packet based on the analysis results from the APA. The aggregation process is determined by two user specified threshold values: the target aggregate packet size and the maximum acceptable delay.

In order to demonstrate the performance for the AAM algorithm over a wide range of different traffic loads, there were 16 traffic trace files used which were captured from a number of live Wi-Fi hotspot networks at different times and locations. These captured traffic trace files were used as the input traffic source in the test simulations. There were two test scenarios used to analyze the performance of the AAM algorithm:

- In the first test scenario, the AAM algorithm was implemented as a standalone aggregation process only. This scenario was used to demonstrate that the AAM algorithm is an adaptive packet aggregation algorithm which can combine the largest number of sub-packets per aggregate packet for a given average packet delay compared to other two aggregation algorithms, namely the FIFO and SSFS algorithms.

- In the second test scenario, the AAM algorithm was implemented in the ns-3 simulator and deployed in a test wireless network with and without transmission errors present. This scenario was used to demonstrate that the AAM algorithm has the best performance in terms of the aggregation trade-off in achieving the maximum average throughput with the minimum average delay compared to other two aggregation algorithms for different traffic loads. In the same wireless network with transmission errors present where the PHY rate adaption mechanism AARF was employed, it was demonstrated that the AAM algorithm is also a robust algorithm. The AAM algorithm can significantly improve the performance in terms of the throughput for low values of BER (e.g. not greater than 10^{-3}), while for large values of BER (e.g. 10^{-2}) the AAM algorithm can still improve the throughput but cannot significantly improve it.

We briefly summarize the key contributions of this work and discuss several future research directions. The central claim of this thesis is that the adaptive packet aggregation algorithm AAM can significantly improve the aggregation trade-off in terms of achieving the maximum average throughput with the minimum average delay over a wide range of different traffic loads in wireless networks. Moreover, it is a robust algorithm as it has the best performance in terms of the throughput compared to the FIFO and SSFS algorithms in error-prone wireless networks. The results of the performance investigation for the AAM algorithm presented in chapter 5 support this claim.

6.1 Summary of Contributions and Achievements

Throughout this thesis, we have presented arguments as to why this work represents an important contribution to the development of packet aggregation. The objective of this section is to collect these arguments together in order to create a more coherent and concise picture of how this work contributes to the packet aggregation research. Specific contributions include:

- *An adaptive adjustable selection window mechanism for aggregating packets.* This adaptive selection window mechanism for the AAM algorithm helps to achieve the best aggregation trade-off in terms of realizing the maximum average throughput with the minimum average delay for all three algorithms considered (i.e. AAM, FIFO and SSFS) under different traffic loads. To the best knowledge of the author, the AAM algorithm is the first packet aggregation algorithm that employs an adaptive selection window mechanism. Chapter 5 presented the performances in terms of the size of the selection window and the average packet rate to show that the selection window size follows the variations in the average packet rate. The performance in terms of the number of sub-packets against the average packet delay, and the aggregation trade-off in achieving the maximum average throughput with the minimum average delay in wireless networks are presented which demonstrate that the adaptive adjustable selection window mechanism can help to achieve the goal of the best performance in terms of the aggregation trade-off for the AAM algorithm in wireless networks.

- *A hybrid selection strategy for selecting packets.* In the AAM algorithm, a hybrid selection strategy is employed to take account of the random packet rate and the packet size where the front packet is always selected as the first sub-packet in order to avoid the possibility of the delay for the first sub-packet increasing indefinitely and other sub-packets are selected based on their size in order to maximize the number of sub-packets in an aggregate packet. Chapter 5 showed that the AAM algorithm has the best performance in terms of the throughput for different PHY rates; even in error-prone wireless networks, the AAM algorithm still can improve the throughput by up to 28% compared to the FIFO algorithm. The AAM algorithm also achieves the goal of the best aggregation trade-off in terms of realizing the maximum average throughput with the minimum average delay for all three algorithms considered by employing this hybrid selection strategy.

- *An adaptive feedback mechanism that ensures robustness in error-prone wireless networks.* The AAM algorithm is based upon an adaptive feedback mechanism that allows it to operate over a wide range of different traffic loads in error-prone wireless networks. Chapter 5 presented the performances for the AAM algorithm in a wireless network where transmission errors were present and it showed that the throughput can be improved by up to 28% compared to the FIFO algorithm. Even if the BER is increased up to 10^{-3}, the AAM algorithms still can improve the throughput by 18%.

In other words, the three elements (i.e. A^3, ATA and APA) that comprise the AAM algorithm allow it to achieve the best aggregation trade-off in terms of realizing the maximum average throughput with the minimum average delay compared to the FIFO and SSFS algorithms for different traffic loads in wireless networks.

6.2 Open Problems and Future Work

The research in this thesis represents important progress in using packet aggregation to improve the throughput in WLANs. However, every new solution naturally generates more questions. Therefore, this section introduces some of the research directions which are closely related to the work in this thesis and appear promising for future research:

- *Employing a more appropriate channel model for the analysis of the performance of the AAM algorithm*. The AAM algorithm has been shown to have good performances under different levels of static BER conditions in wireless networks. However, the simulation scenario used here for the performance analysis is poor in the sense that it uses a simplistic and unrealistic loss model for the wireless channels. This would suggest that it needs to use a more appropriate channel model that includes the time-varying and the busty nature of real wireless channels where transmission errors tend to occur in bursts due to the effects of fading and interference. A number of models may be employed to demonstrate the performance of the AAM algorithm, such as the log-distance path loss model [Rap96] which assumes an exponential path loss based on the distance from the sender to the receiver, or the Jakes model [ZeX03] which calculates the propagation loss by modeling a set of rays transmitted from the sender to the receiver via different paths.

- *Carrying out an experimental validation of the AAM algorithm to better gauge its performance under realistic wireless network conditions*. In this thesis, the AAM algorithm is simulated using the ns-3 simulator. The AAM algorithm can be implemented in the MAC layer of the station nodes in an experimental wireless network by modifying the open source ath9k [ath9] or ath10k [ath10] driver. The ath9k driver can support all the Atheros IEEE 802.11n WLAN based chipsets and the ath10k driver can support the IEEE 802.11ac chipsets. There is a need to develop three modules to implement the three algorithms considered (i.e. AAM, FIFO and SSFS) and these modules then need to be incorporated into the ath9k or ath10k driver. The modified driver is deployed in the MAC layer of all the stations in a multiple-hop wireless network which contains an AP station and multiple client stations. The 16 captured traffic trace files can still be used as the input in the sender stations. After that, other applications (e.g. VoIP) can be used to test the performance of the AAM algorithm in the same test-bed scenario.
- Employing an adaptive step size for tuning the size of the selection window. The current tuning rules used in the ATA increase/decrease the size of the selection window in steps of 1 which has been shown to produce a good performance in terms of the throughput and the delay. An adaptive step size for increasing/decreasing the step size may achieve an even better performance. For

example, the AIMD (Additive Increase Multiplicative Decrease) [ChJ89] strategy which combines the linear growth of the selection window size with an exponential reduction could be used by the AAM algorithm. The AIMD strategy may further improve the throughput and reduce the delay. Another adaptive strategy can be used to adaptively tune the step size of the selection window which is still based on the analysis results from the APA. The rules for tuning the step size of the selection window are shown in Table 6-1 where MI means multiplicative increase, MD means multiplicative decrease, AI means additive increase and AD means additive decrease. These rules can be explained as follows: (1) If the delay has increased and the number of sub-packets has not decreased, this means that the network performance is slowly deteriorating and the selection window size needs to be slowly decreased. Therefore, the step size is additively decremented. (2) If the delay has increased and the number of sub-packets has decreased, this means that the network is rapidly deteriorating. Therefore, the selection window size needs to be rapidly reduced by using a multiplicative decrement. (3) If the delay has decreased and the number of sub-packets has increased, this means that the network performance is slowly improving and the selection window size needs to be slowly increased. Therefore, the step size is additively incremented. (4) If the delay has decreased and the number of sub-packets has not increased, the step size is additively incremented. (5) If the delay has maintained, the step size is additively incremented. The biggest challenge of this adaptive strategy is that one needs to determine what values for the multiplicative factors and additive steps to use in order to achieve the best network performance.

Table 6-1: The rules for tuning the step size of the selection window size

		Aggregate packet delay		
		> 0	< 0	= 0
Number of sub-packets	> 0	AD	AI	AI
	< 0	MD	AI	AI
	= 0	AD	AI	AI

- *Investigating the performance of the AAM algorithm when combined with a routing protocol*. It is well known that routing is critical to the network throughput in WMNs. Packet size is one of the most important metrics that impacts on the routing decision of routing protocols in WMNs [YWK05A], such as the WCETT (Weighted Cumulative ETT) [DPZ04] and MIC (Metric of Interference and Channel-switching) [YWK05]. There is an optimal aggregate packet size associated with some routing protocols to achieve the maximum network throughput in WMNs [GoY13]. In particular for the ETT (Expected Transmission Time) [DPZ04] routing protocol which is a popular routing protocol and chooses the routing based on the packet size, the AAM algorithm may significantly impact on the routing decision as the size of the AAM aggregate packet can be controlled. Therefore, the interaction between the routing protocol ETT and the AAM algorithm should be investigated [GoY13] in order to achieve an optimal network performance. Furthermore, there is a need for an investigation to determine which routing protocol works best with the AAM algorithm.

- *Optimizing the selection strategy in the A^3 algorithm in wireless networks*. The A^3 selection strategy is a hybrid selection strategy which has successfully helped to achieve the goal of the best aggregation trade-off in terms of realizing the maximum average throughput with the minimum average delay. There may be some other selection strategies that could be used to further improve the throughput. For example, a priority strategy based on the least life time could be used to reduce the delay by selecting the packets which have the smallest life time [LYY09]. In particular, for the multi-hop wireless networks, the value density selection strategy [Mod82] can be used where the packets are selected based on the value density. The value density of a packet is defined as the number of transmitted hop-counts per payload in byte. The largest value density packet in the input buffer is serviced first in order to reduce the packet dropped.

- *Specifying the values of the target aggregate packet size and the maximum acceptable delay parameters for different application types.* Currently, the values of the both parameters are fixed and specified by the user at the outset. There may be further benefits in terms of network performance if these parameters can be specified according to the application types. There is one method that can be implemented in several steps. At first, all the applications within a mixed traffic load are divided into two classes based on tagging and packet size [EEV06], the time sensitive application (e.g. VoIP, video streaming) and the time insensitive application (e.g. E-mail) which are pushed into separate buffers. Here, the biggest drawback is that it may fail to identify the time sensitive application packets as they tend not to use tagging. Then the values of the two parameters for each class of application can be specified for the different applications in order to improve the network performance. For example, for the time sensitive applications the maximum acceptable delay and the target aggregate packet size can be specified in order to minimize the delay. For the time insensitive applications, the values of the two parameters can be specified in order to maximize the throughput. After that, the two classes of applications each employ separate AAM algorithms to aggregate packets based on the different specified input parameters. Finally, the aggregate packets are moved into the output buffer and are transmitted based on the arrival time. The aggregate packet of the time sensitive application is moved into the output buffer first if there are two aggregate packets arrivals into the output buffer at the same time. Another method that can be used is to develop a smart algorithm which can adaptively adjust the values of the two parameters based on the different traffic loads. The values of the two parameters can be adaptively adjusted based on the ratio of the small size packets to the overall packets which arrive within a certain duration of time (e.g. 20 ms). The values of the two parameters are inversely proportional to the value of this ratio. The reason for this is that the VoIP packet is a small size packet and is time sensitive packet. Here, the biggest challenge is that how to determine what constitutes a small size packet and what time duration to use and how to adjust the values of the two parameters based on the ratio in order to achieve the best network performance.

6.3 Publications

The publications arising from this thesis are as followings:

[1] Jianhua Deng, Mark Davis, “An Adaptive Packet Aggregation Algorithm for Wireless Networks,” *International Conference on Wireless Communications and Signal Processing 2013 (WCSP 2013),* pp. 449-504, Hangzhou, China, 24 -26 October 2013.

[2] Jianhua Deng, Mark Davis, “The Performance of the Adaptive Aggregation Mechanism (AAM) in Lossy Wireless Networks,” acceptable for publication at *20th IEEE Symposium on Communications and Vehicular Technology in the Benelux 2013 (IEEE SCVT 2013)*, Namur, Belgium, 21 November 2013.

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Appendix A

The captured traffic trace file 1

Table A-1: The details for the captured traffic trace file 1.

Time	Date	Location	PPS
10:30 – 11:30	29th ,May, 2012	*JAVA City*, DIT, Dublin	92.1

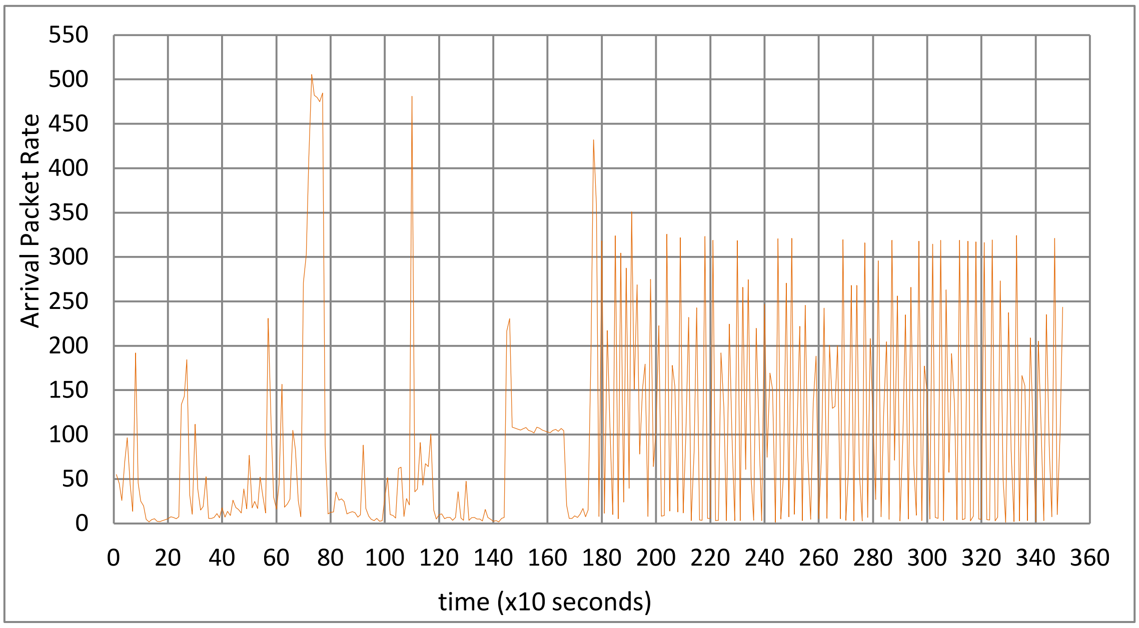

Figure A-1: The average packet rate for the captured traffic trace file 1.

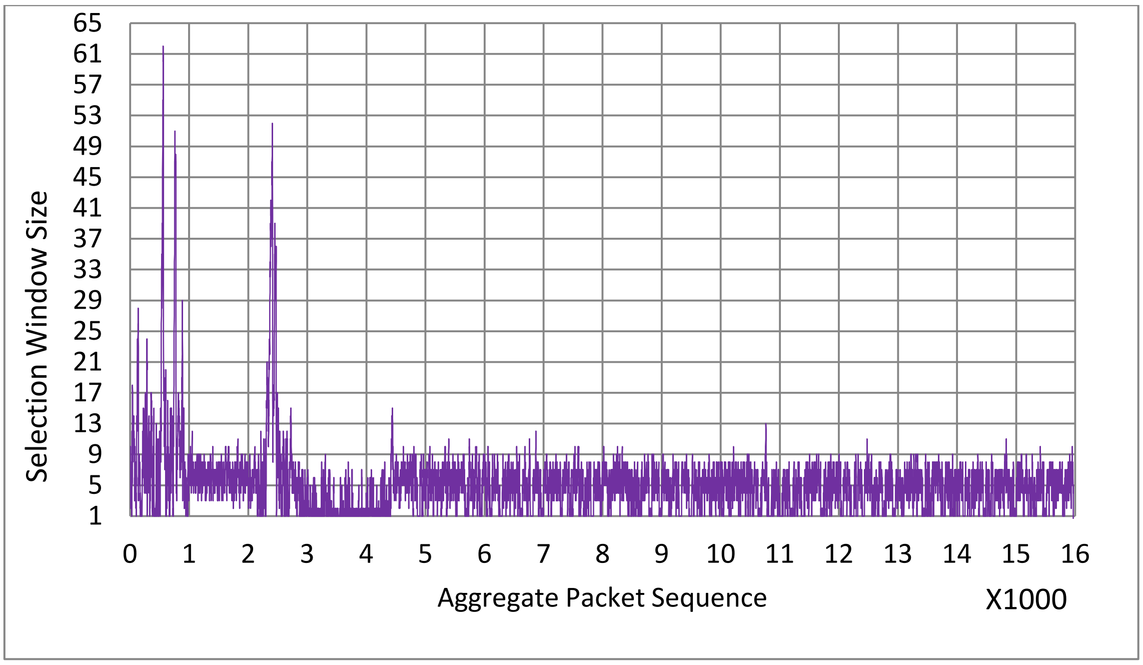

Figure A-2: The selection window size sampled every ten aggregated packets generated for the captured traffic trace file 1.

The captured traffic trace file 2

Table A-2: The details for the captured traffic trace file 2.

Time	Date	Location	PPS
12:00 – 13:00	29th ,May, 2012	*JAVA City*, DIT, Dublin	114

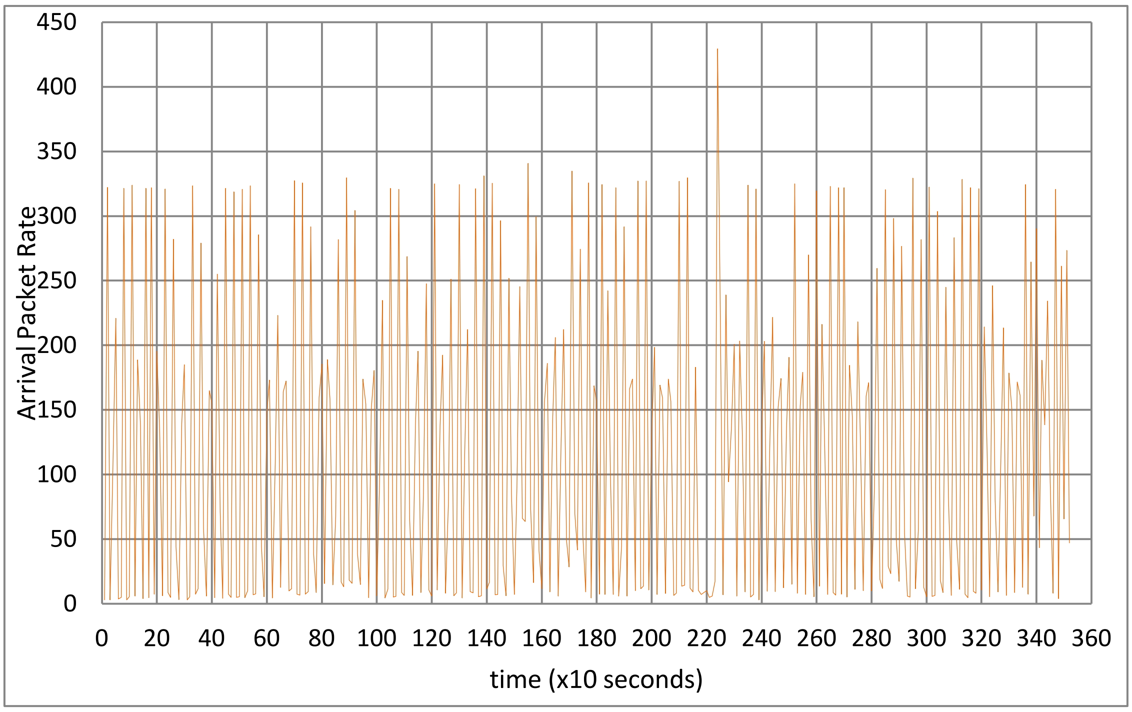

Figure A-3: The average packet rate for the captured traffic trace file 2.

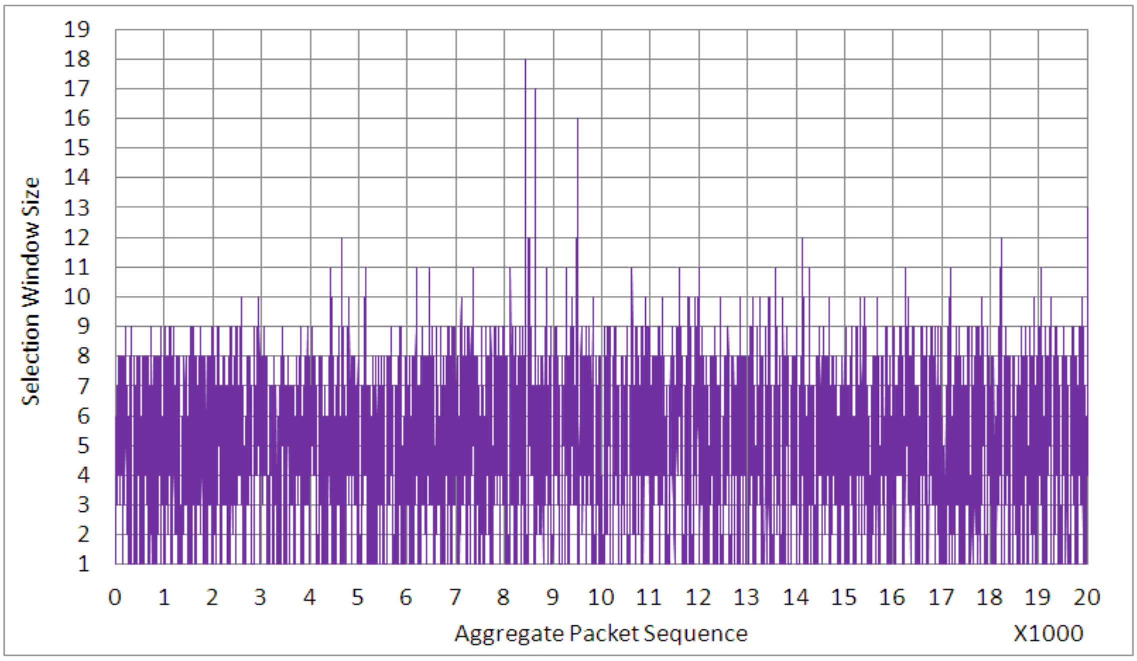

Figure A-4: The selection window size sampled every ten aggregated packets generated for the captured traffic trace file 2.

The captured traffic trace file 3

Table A-3: The details for the captured traffic trace file 3.

Time	Date	Location	PPS
14:00 – 15:00	29th ,May, 2012	*JAVA City,* DIT, Dublin	75.7

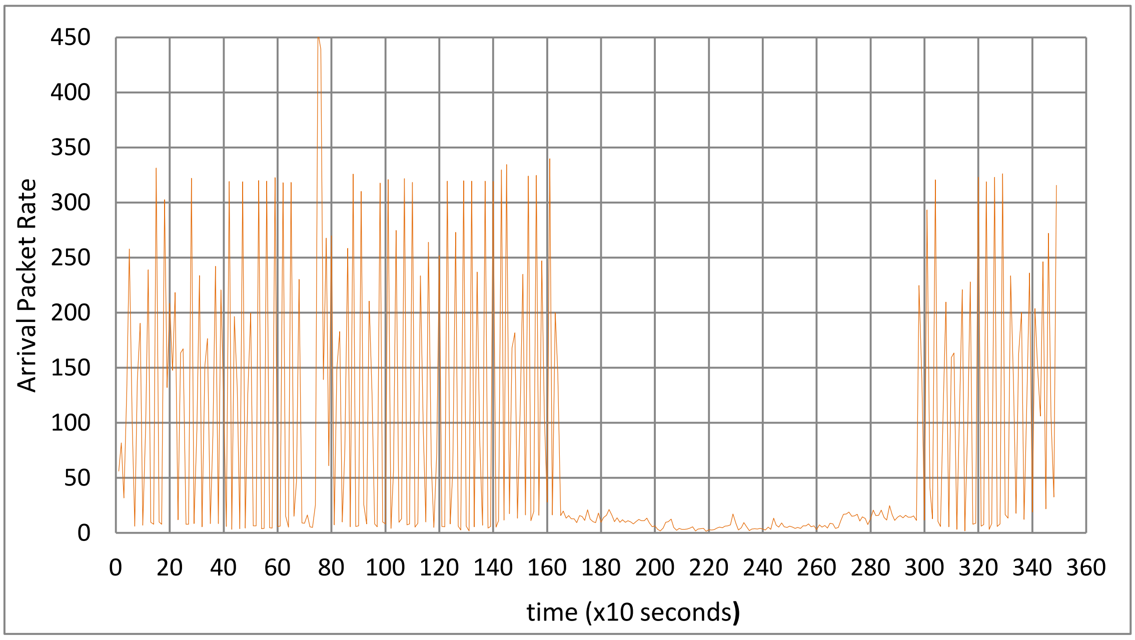

Figure A-5: The average packet rate for the captured traffic trace file 3.

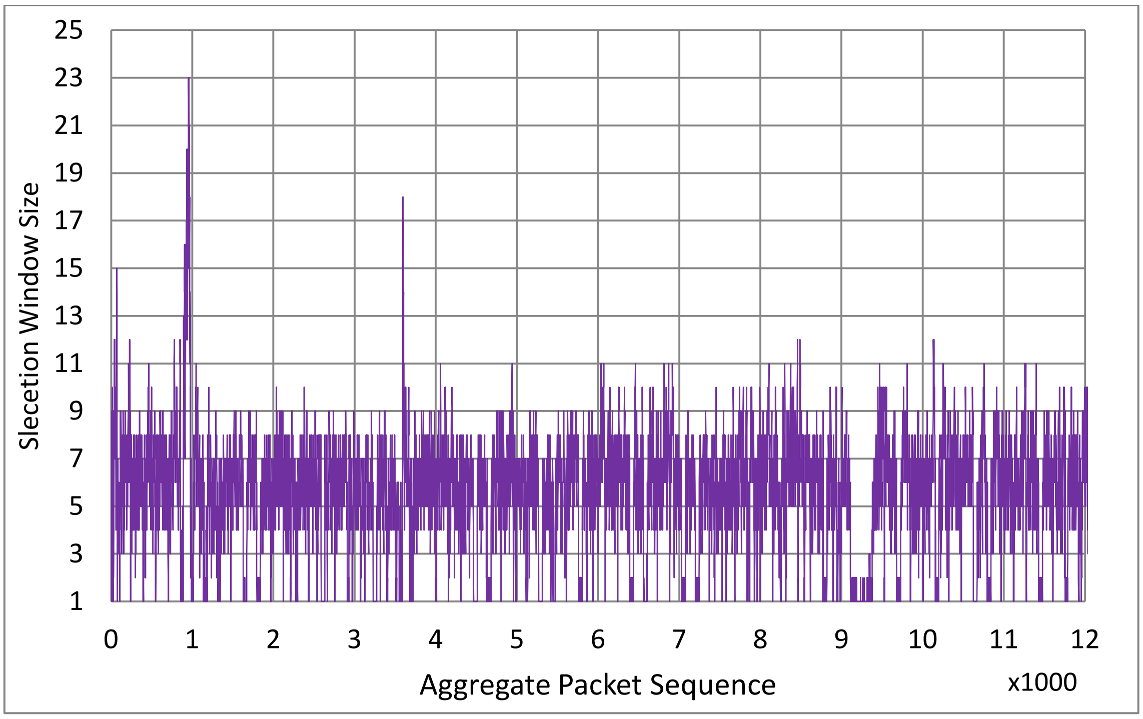

Figure A-6: The selection window size sampled every ten aggregated packets generated for the captured traffic trace file 3.

The captured traffic trace file 4

Table A-4: The details for the captured traffic trace file 4.

Time	Date	Location	PPS
16:00 – 17:00	29th ,May, 2012	*JAVA City,* DIT, Dublin	111.3

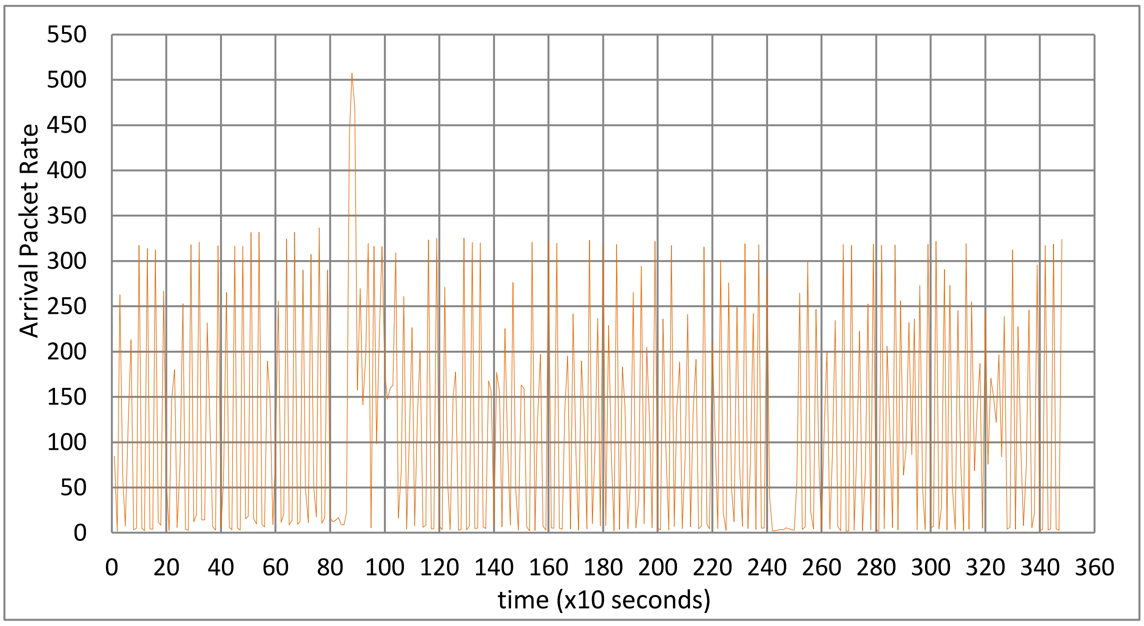

Figure A-7: The average packet rate for the captured traffic trace file 4.

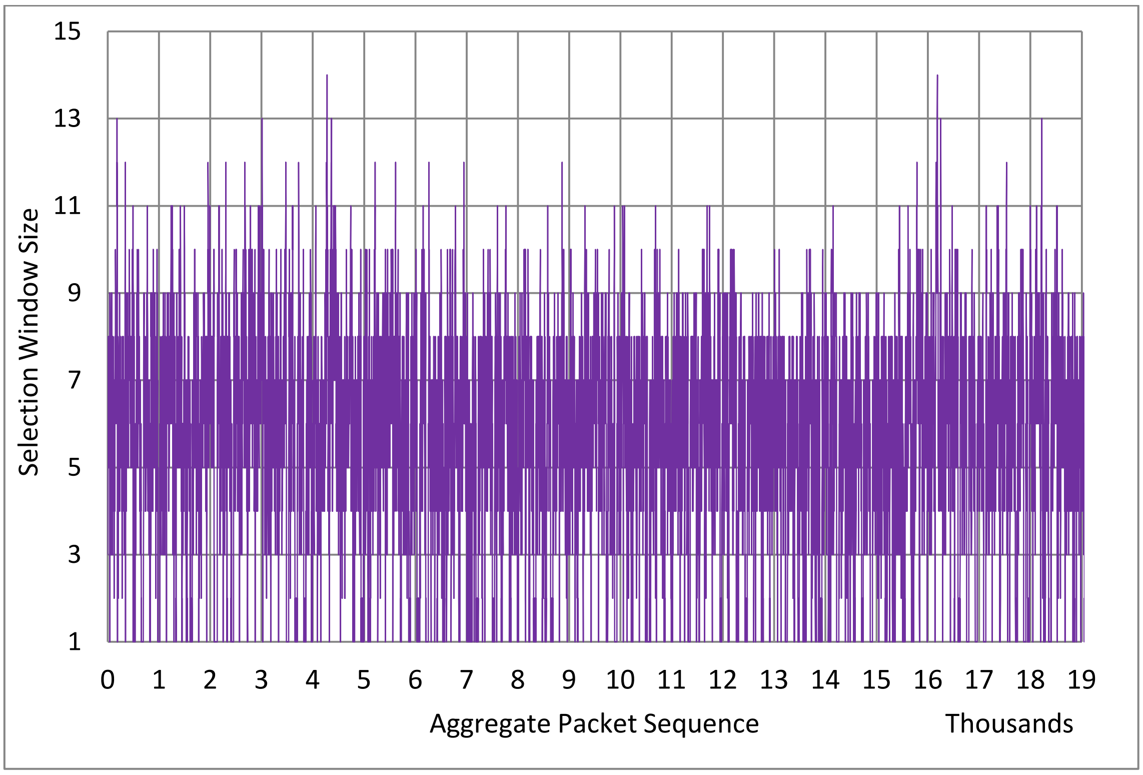

Figure A-8: The selection window size sampled every ten aggregated packets generated for the captured traffic trace file 4.

The captured traffic trace file 5

Table A-5: The details for the captured traffic trace file 5.

Time	Date	Location	PPS
17:30 – 18:30	29th ,May, 2012	*JAVA City,* DIT, Dublin	109.4

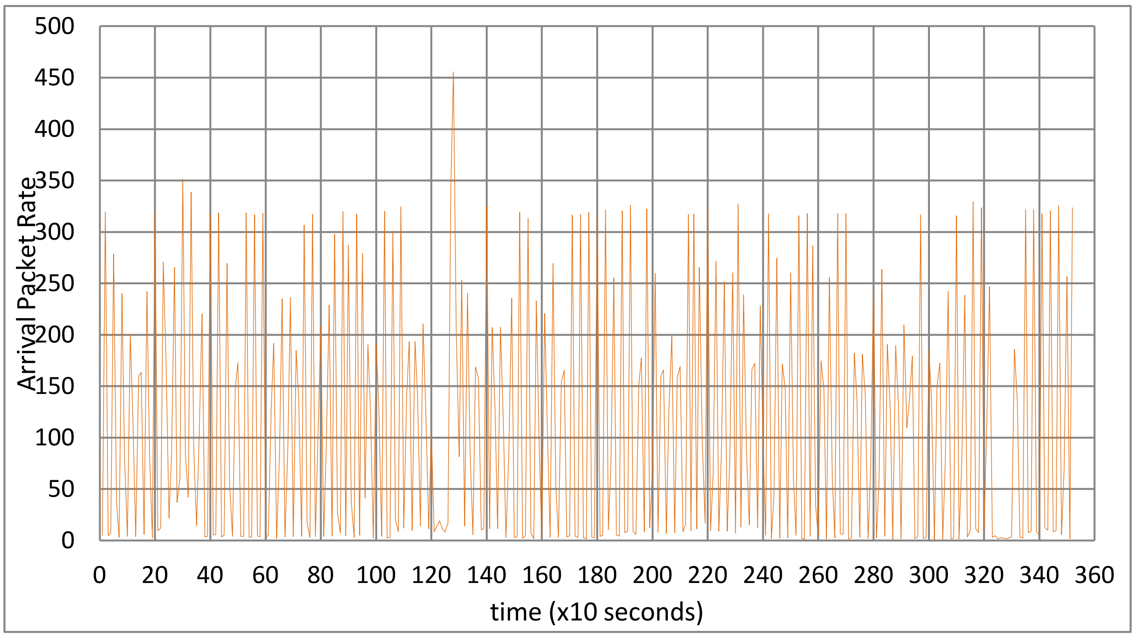

Figure A-9: The average packet rate for the captured traffic trace file 5.

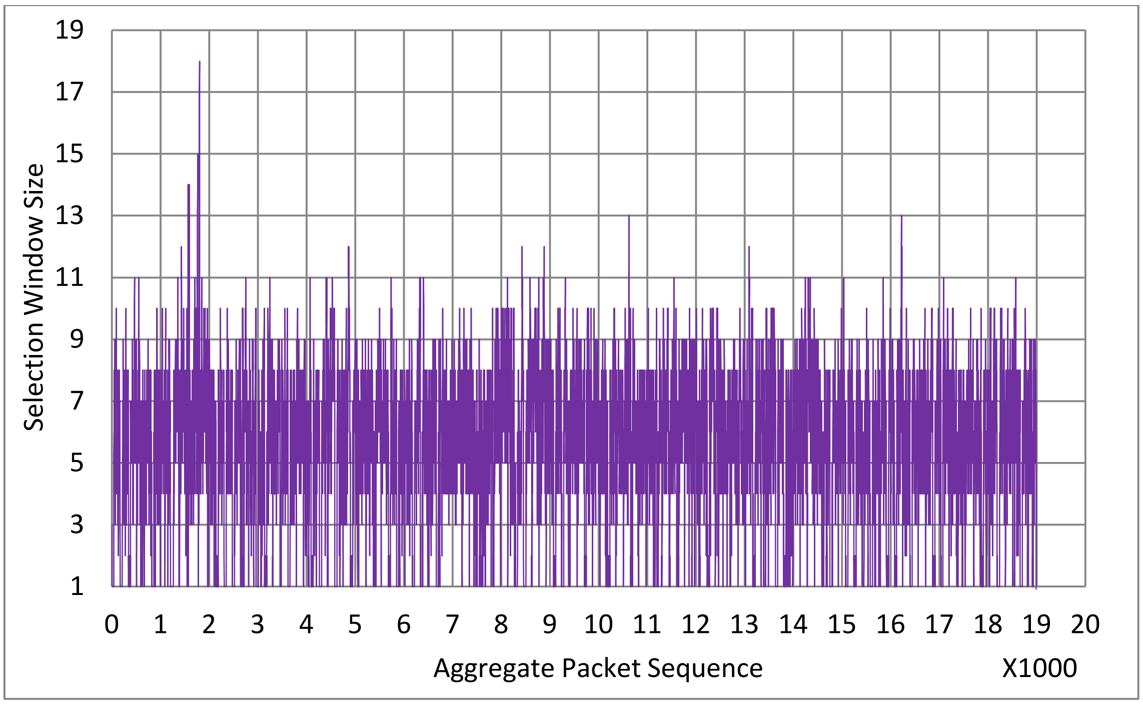

Figure A-10: The selection window size sampled every ten aggregated packets generated for the captured traffic trace file 5.

The captured traffic trace file 6

Table A-6: The details for the captured traffic trace file 6

Time	Date	Location	PPS
09:30 – 10:30	19th, June, 2012	*Costa coffee shop,* Dublin	9.4

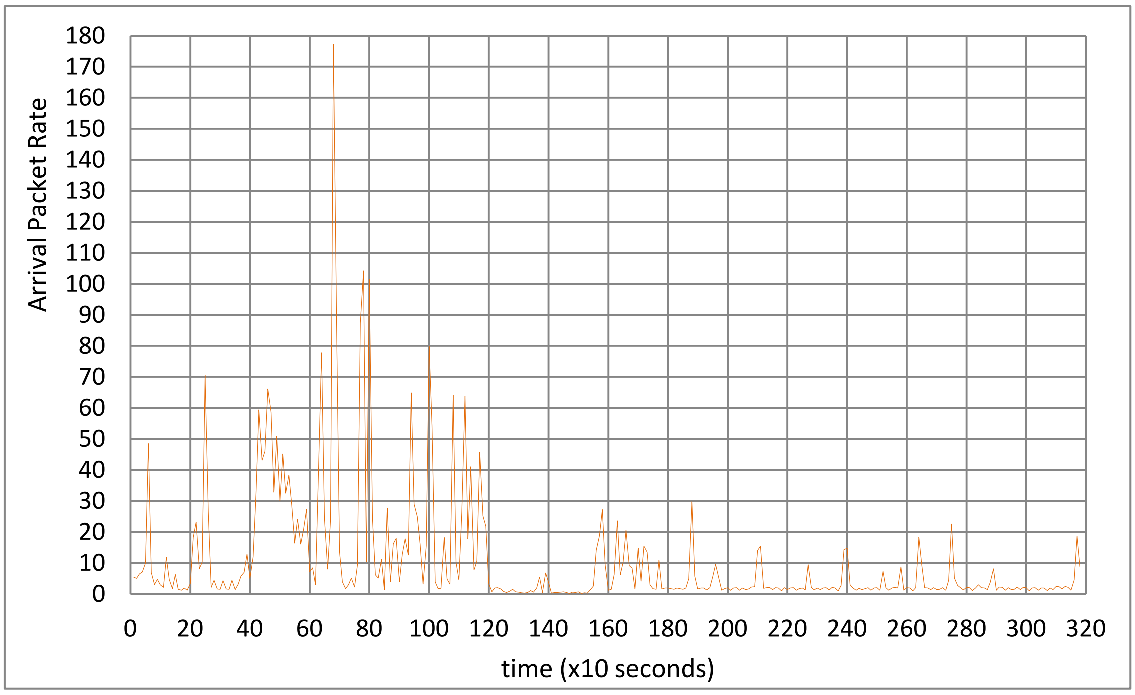

Figure A-11: The average packet rate for the captured traffic trace file 6.

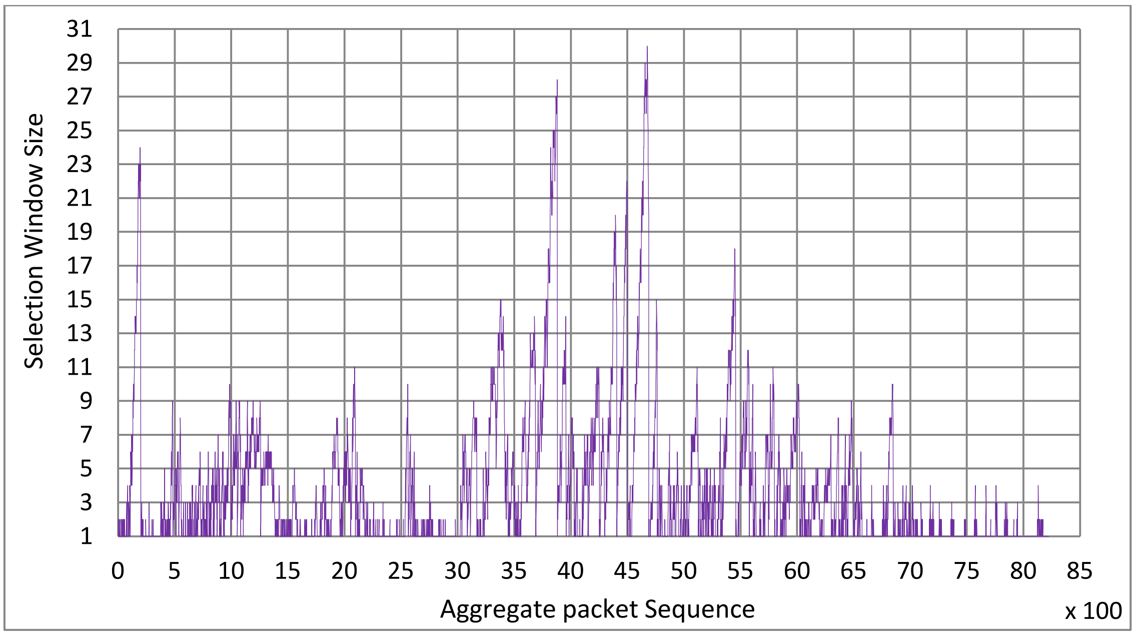

Figure A-12: The selection window size generated by the AAM algorithm for the captured traffic trace file 6.

The captured traffic trace file 7

Table A-7: The details for the captured traffic trace file 7

Time	Date	Location	PPS
11:00 –12:00	19th, June, 2012	*Parliament Square*, TCD, Dublin	5.6

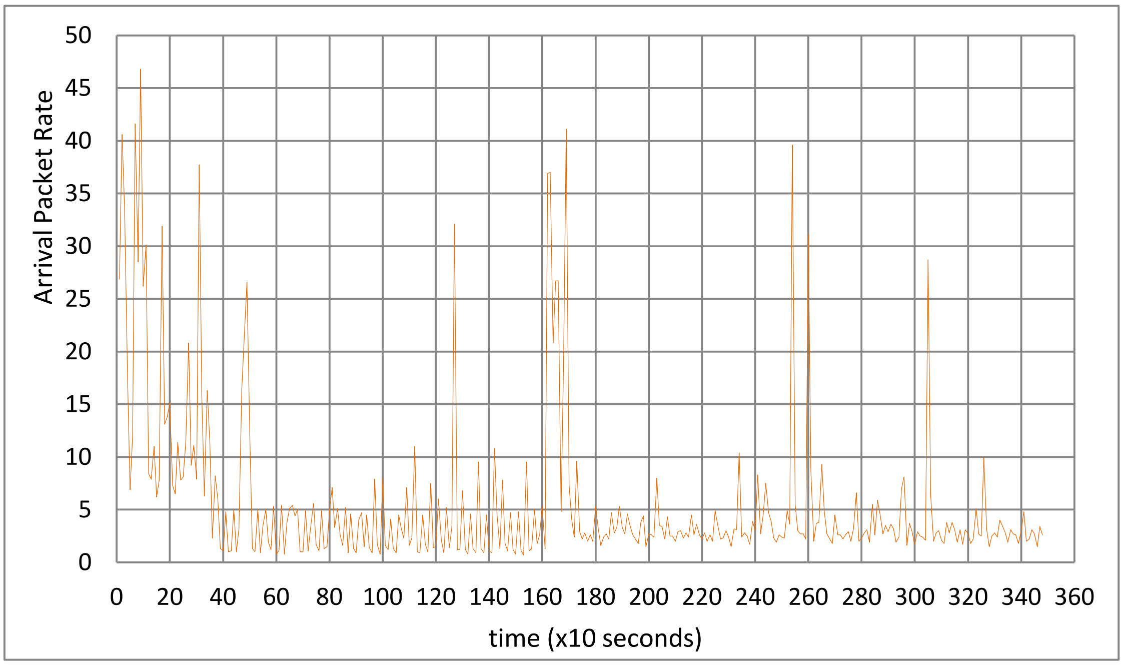

Figure A-13: The average packet rate for the captured traffic trace file 7.

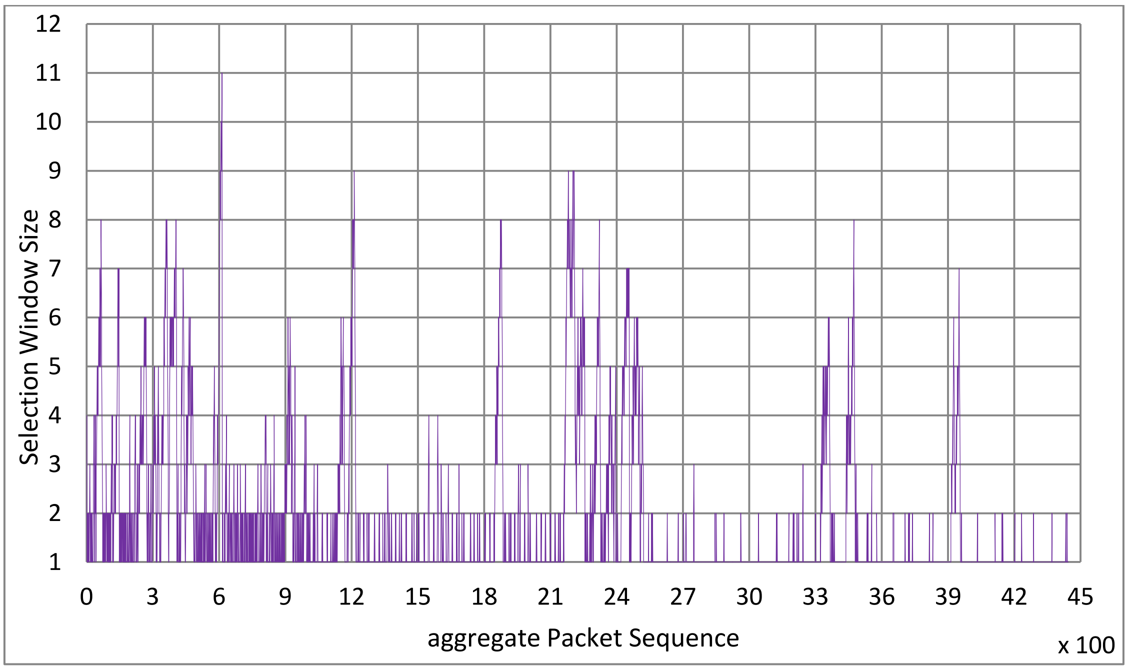

Figure A-14: The selection window size generated by the AAM algorithm for the captured traffic trace file 7.

The captured traffic trace file 8

Table A-8: The details for the captured traffic trace file 8

Time	Date	Location	PPS
12:30 – 13:30	19th, June, 2012	*Costa coffee shop,* Dublin	9.2

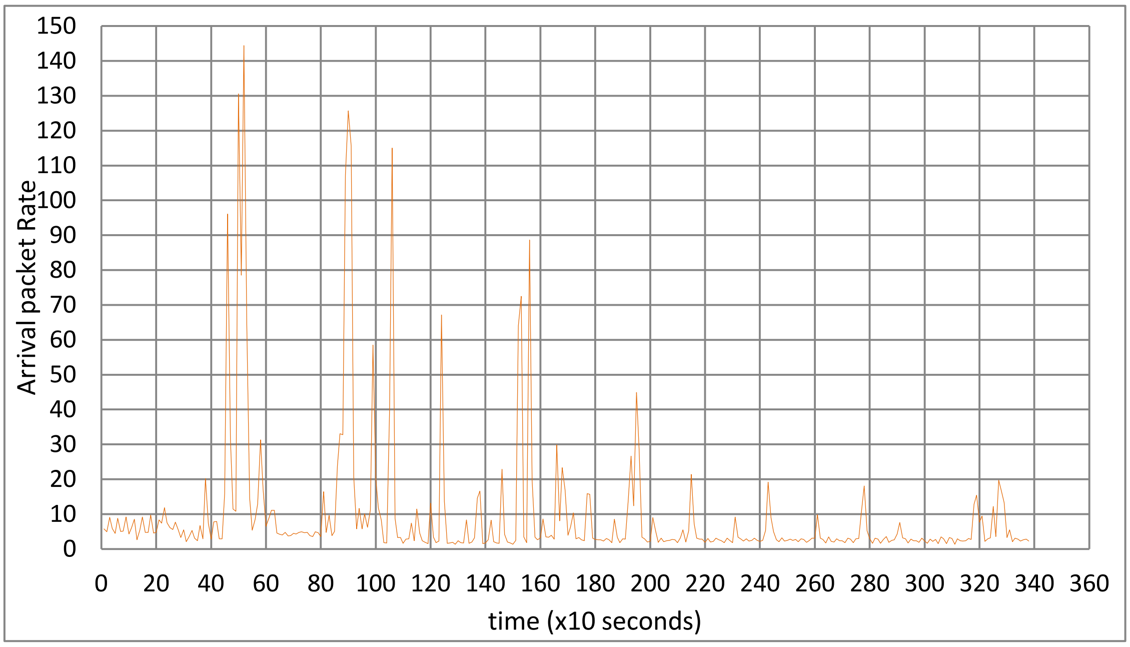

Figure A-15: The average packet rate for the captured traffic trace file 8.

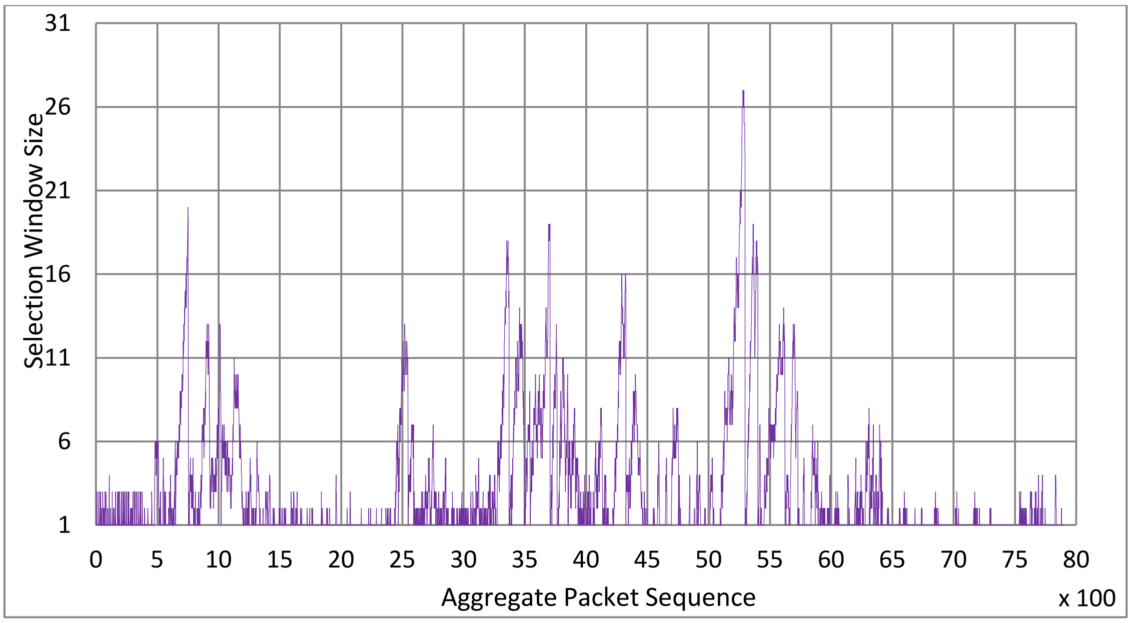

Figure A-16: The selection window size generated by the AAM algorithm for the captured traffic trace file 8.

The captured traffic trace file 9

Table A-9: The details for the captured traffic trace file 9

Time	Date	Location	PPS
16:00 –17:00	19th, June, 2012	*Parliament Square*, TCD, Dublin	3.6

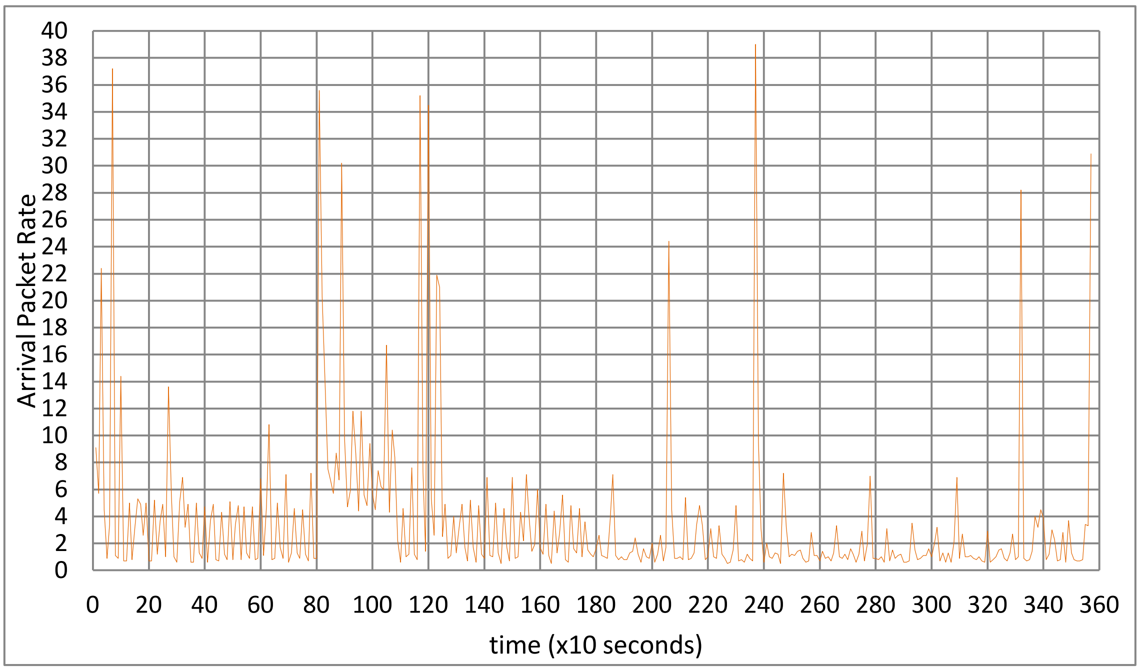

Figure A-17: The average packet rate for the captured traffic trace file 9.

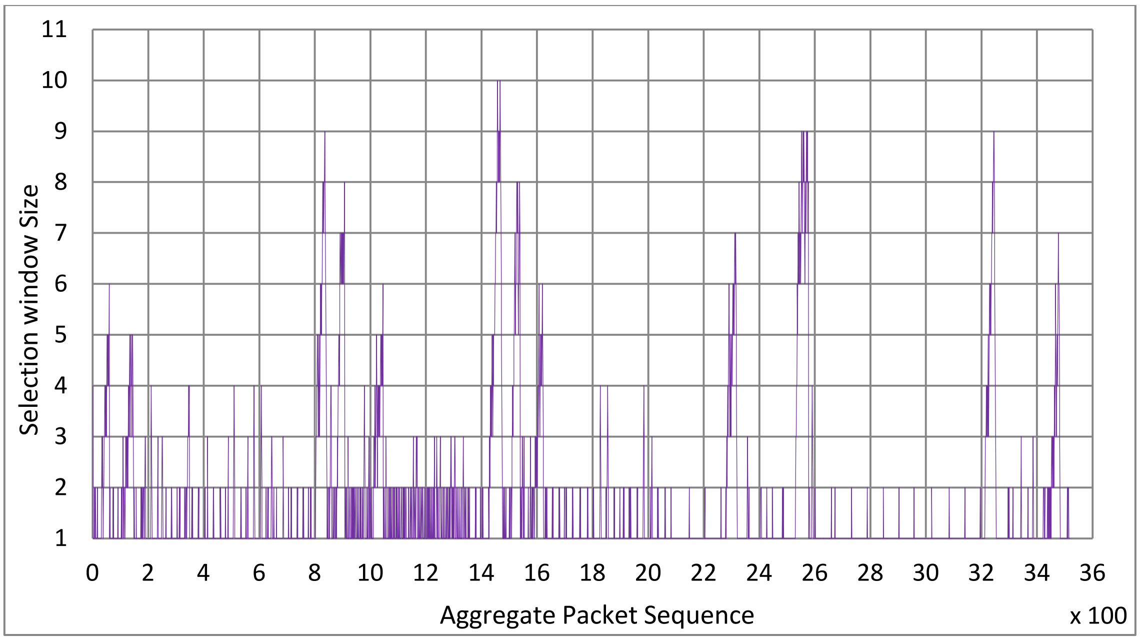

Figure A-18: The selection window size generated by the AAM algorithm for the captured traffic trace file 9.

The captured traffic trace file 10

Table A-10: The details for the captured traffic trace file 10

Time	Date	Location	PPS
17:00 – 18:00	19th, June, 2012	*Costa coffee shop,* Dublin	6.6

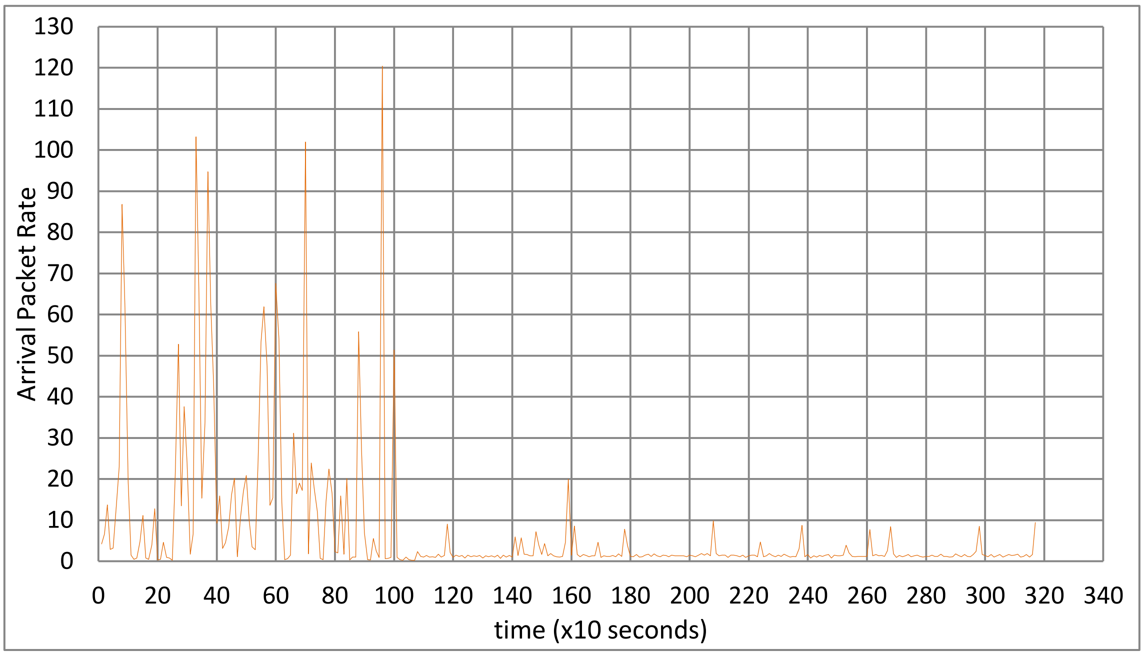

Figure A-19: The average packet rate for the captured traffic trace file 10.

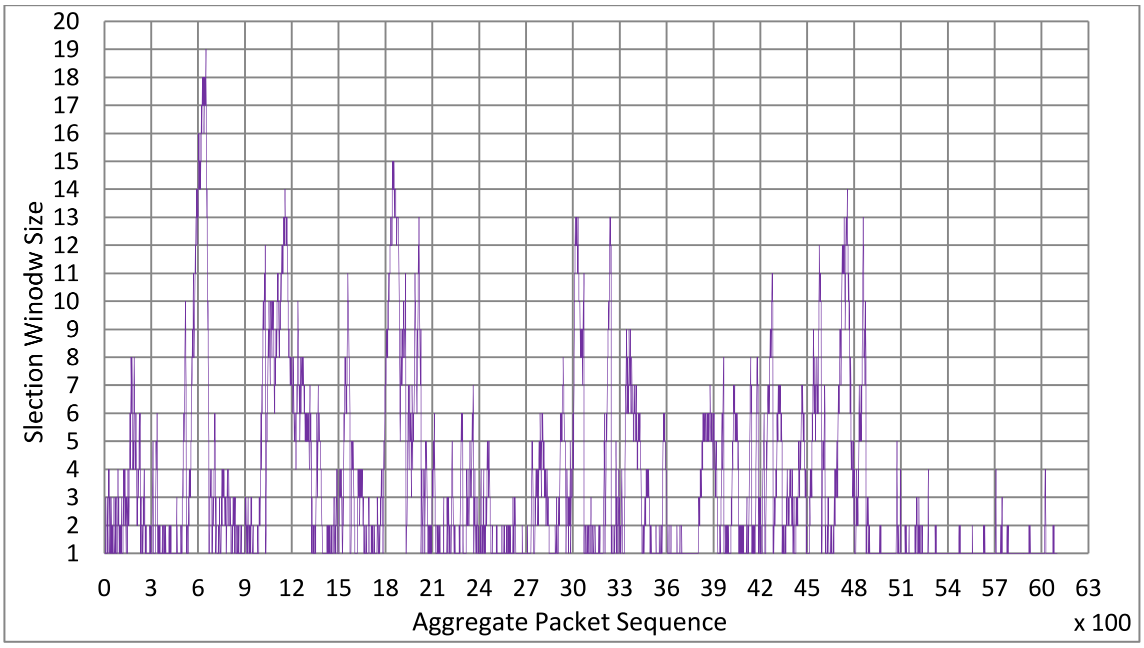

Figure A-20: The selection window size generated by the AAM algorithm for the captured traffic trace file 10.

The captured traffic trace file 11

Table A-11: The details for the captured traffic trace file 11

Time	Date	Location	PPS
12:00 –13:00	24th, June, 2012	*Hueston* train station, Dublin	6.87

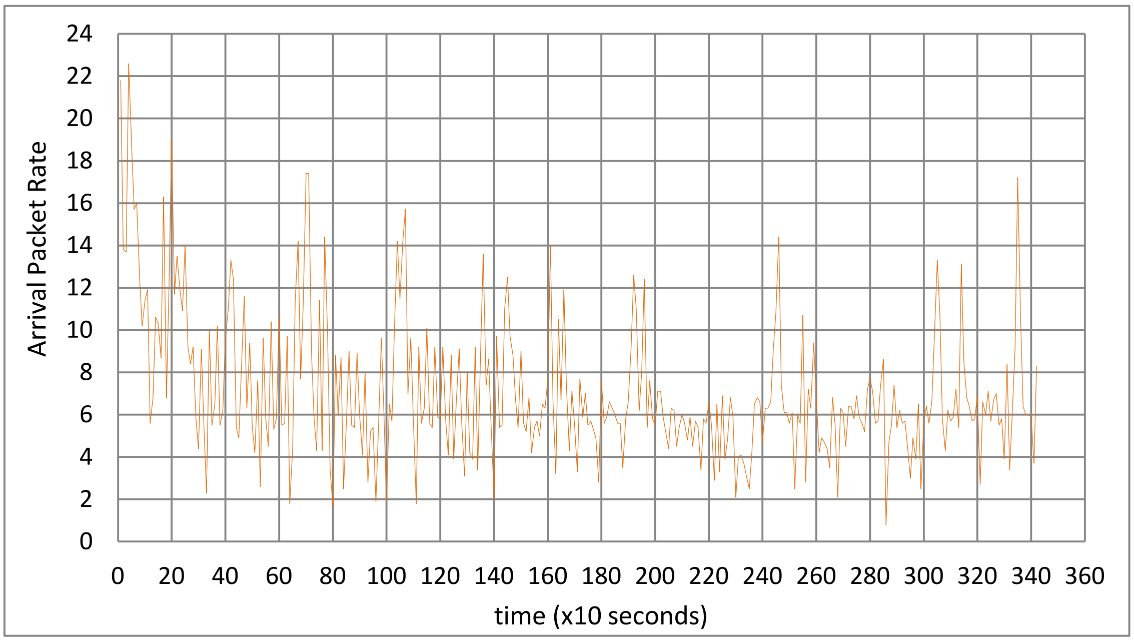

Figure A-21: The average packet rate for the captured traffic trace file 11.

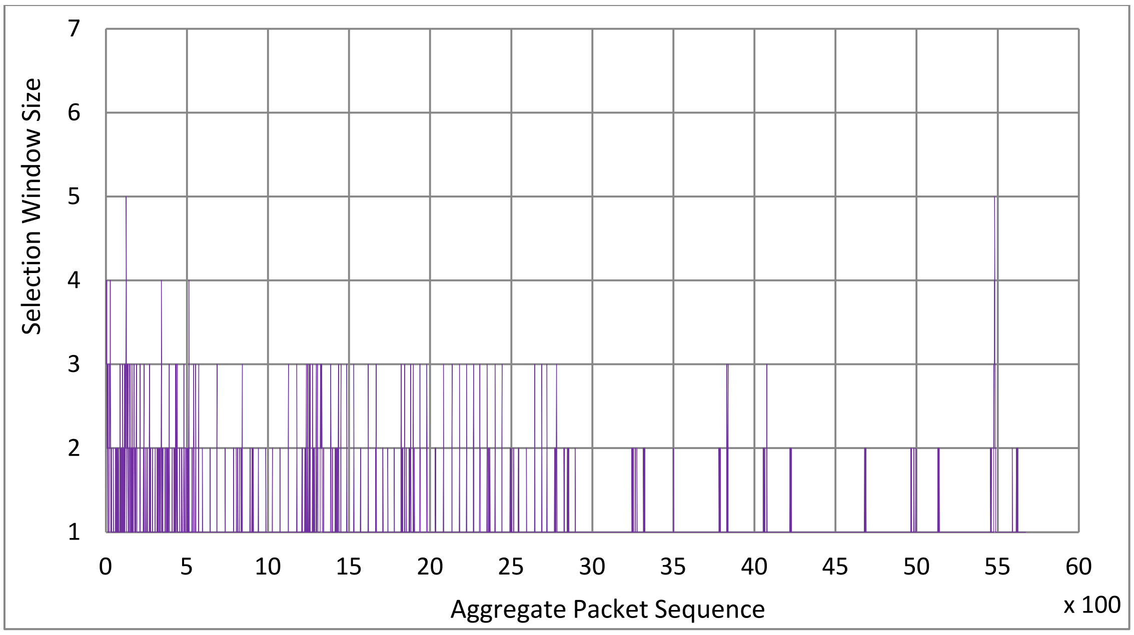

Figure A-22: The selection window size generated by the AAM algorithm for the captured traffic trace file 11.

The captured traffic trace file 12

Table A-12: The details for the captured traffic trace file 12

Time	Date	Location	PPS
13:30 – 14:30	24th, June, 2012	*Hueston* train station, Dublin	4.2

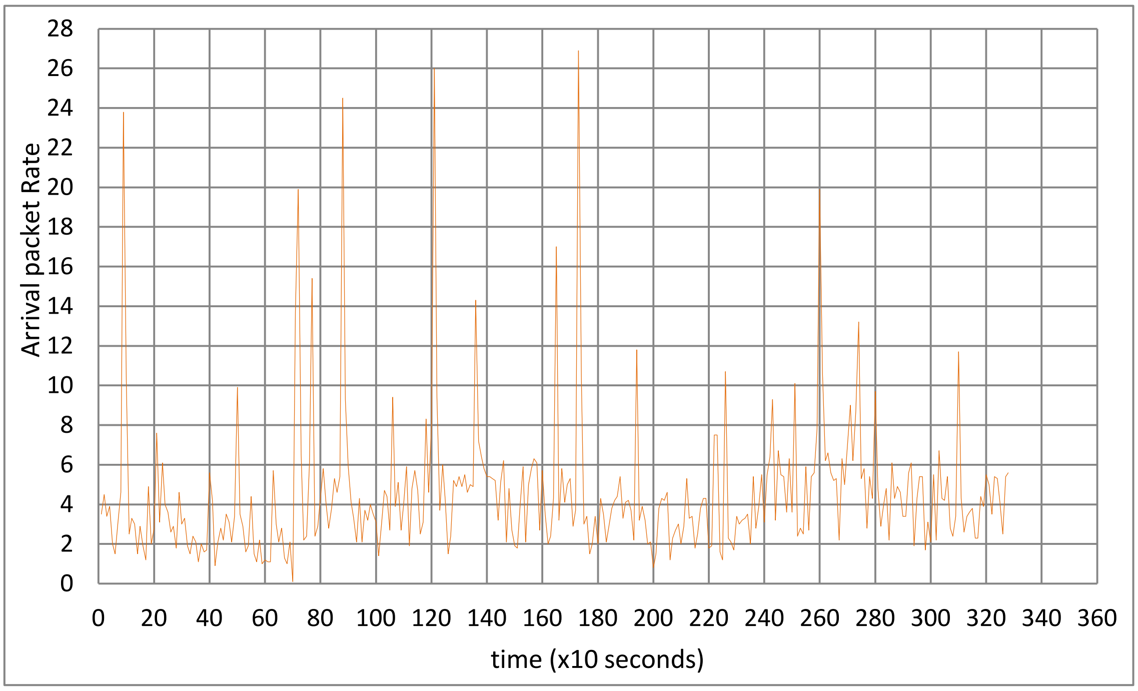

Figure A-23: The average packet rate for the captured traffic trace file 12.

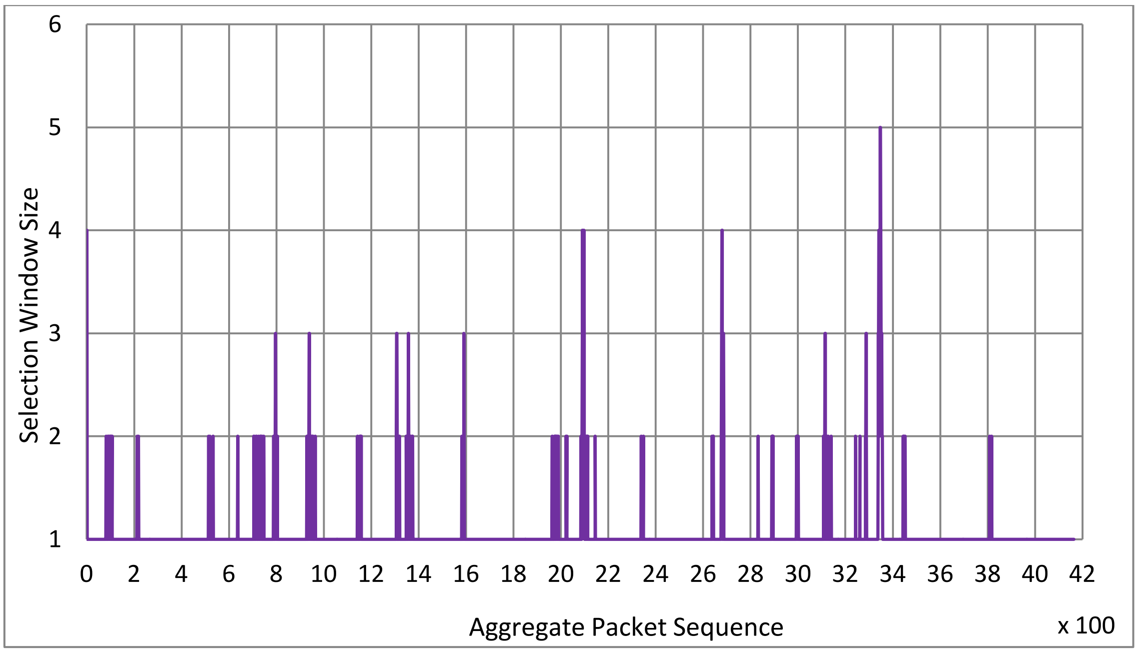

Figure A-24: The selection window size generated by the AAM algorithm for the captured traffic trace file 12.

The captured traffic trace file 13

Table A-13: The details for the captured traffic trace file 13

Time	Date	Location	PPS
15:00 –16:00	24th, June, 2012	*Hueston* train station, Dublin	6.2

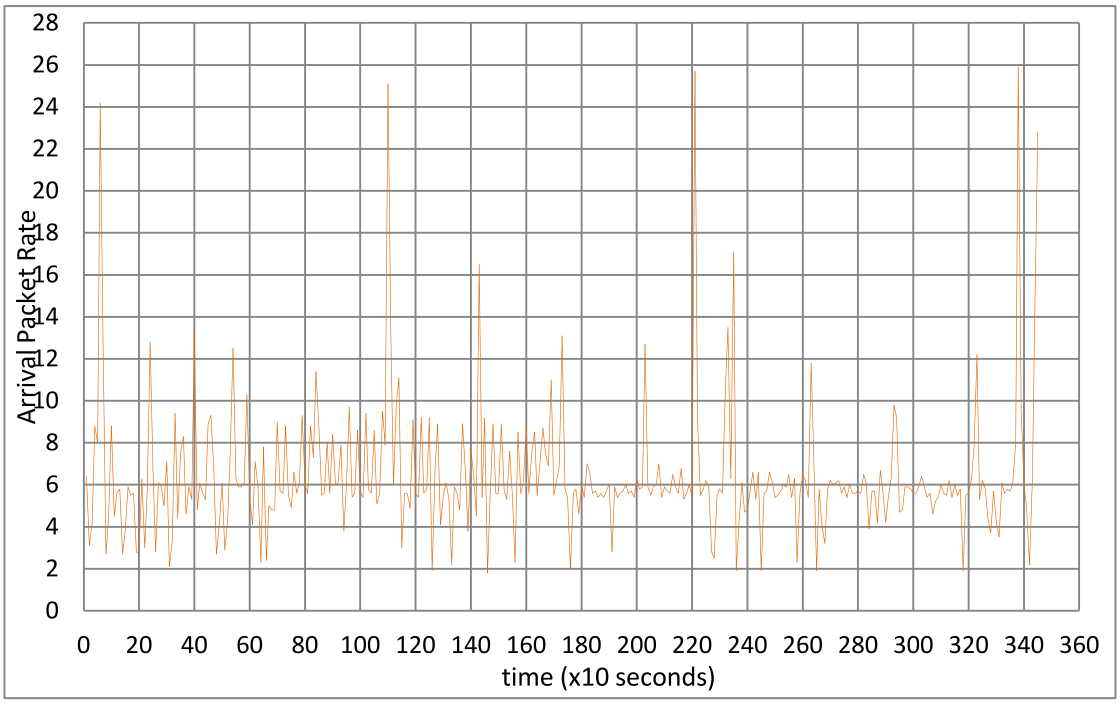

Figure A-25: The average packet rate for the captured traffic trace file 13.

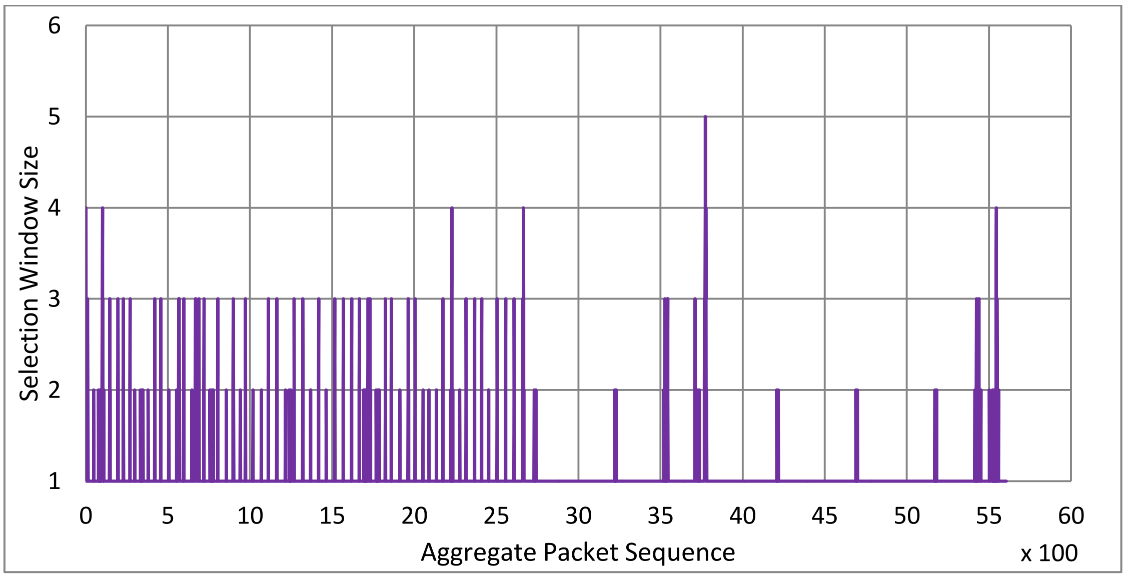

Figure A-26: The selection window size generated by the AAM algorithm for the captured traffic trace file 13.

The captured traffic trace file 14

Table A-14: The details for the captured traffic trace file 14

Time	Date	Location	PPS
10:30 –11:30	26th, June, 2012	Library, Kevin street, DIT, Dublin	19.2

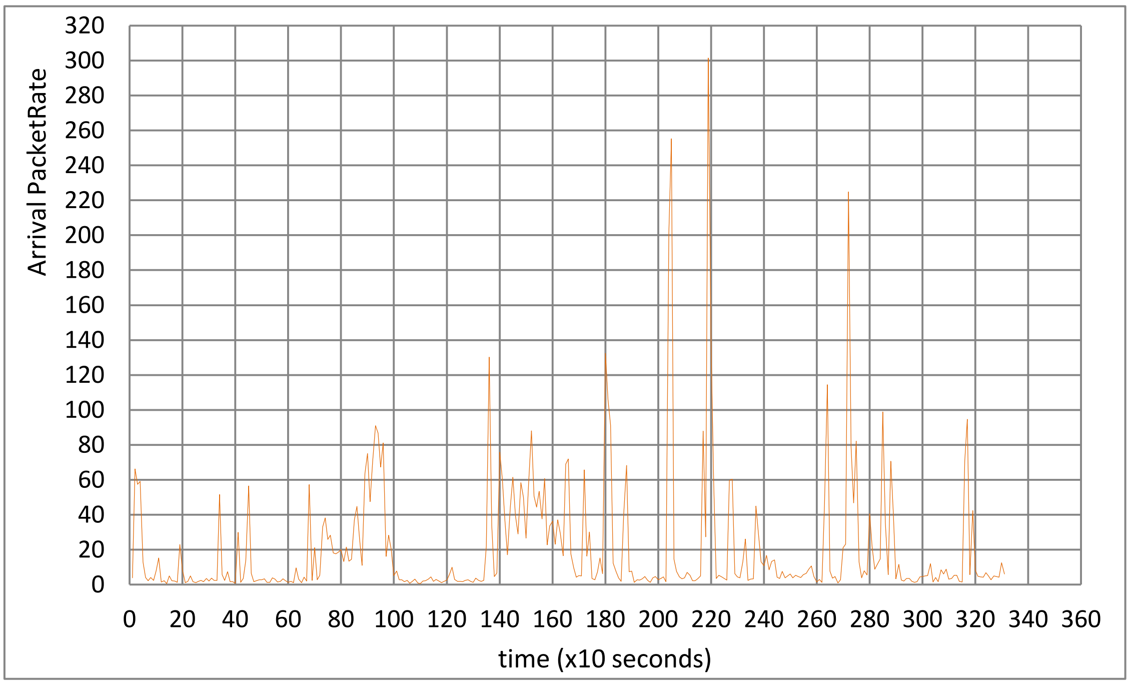

Figure A-27: The average packet rate for the captured traffic trace file 14.

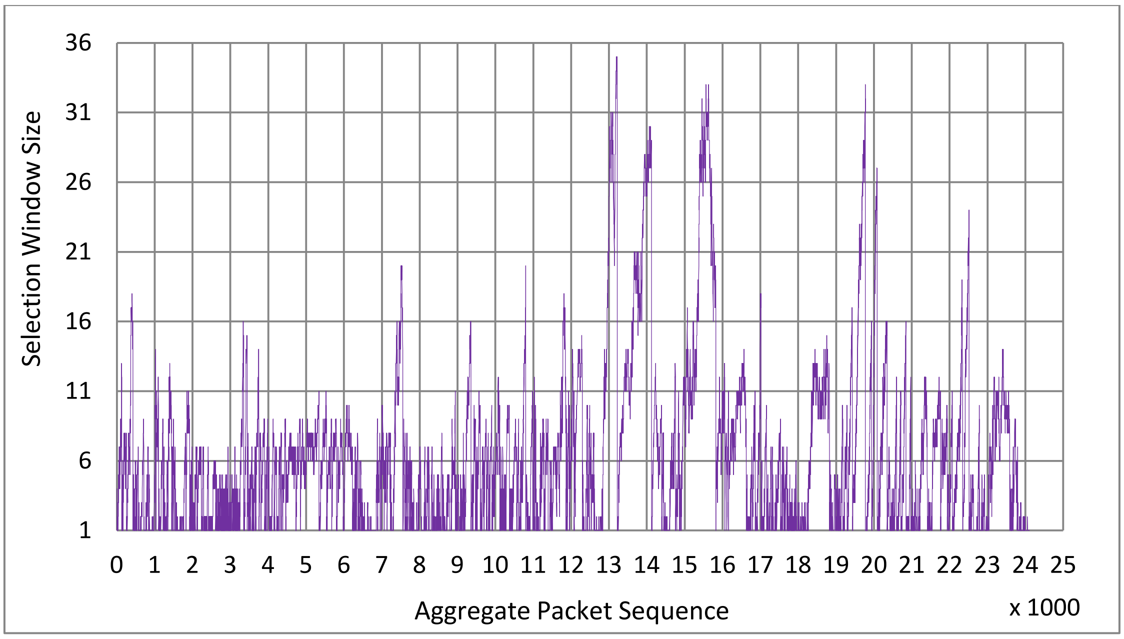

Figure A-28: The selection window size generated by the AAM algorithm for the captured traffic trace file 14.

The captured traffic trace file 15

Table A-15: The details for the captured traffic trace file 15

Time	Date	Location	PPS
12:00 –13:00	26th, June, 2012	Library, Kevin street, DIT, Dublin	3.8

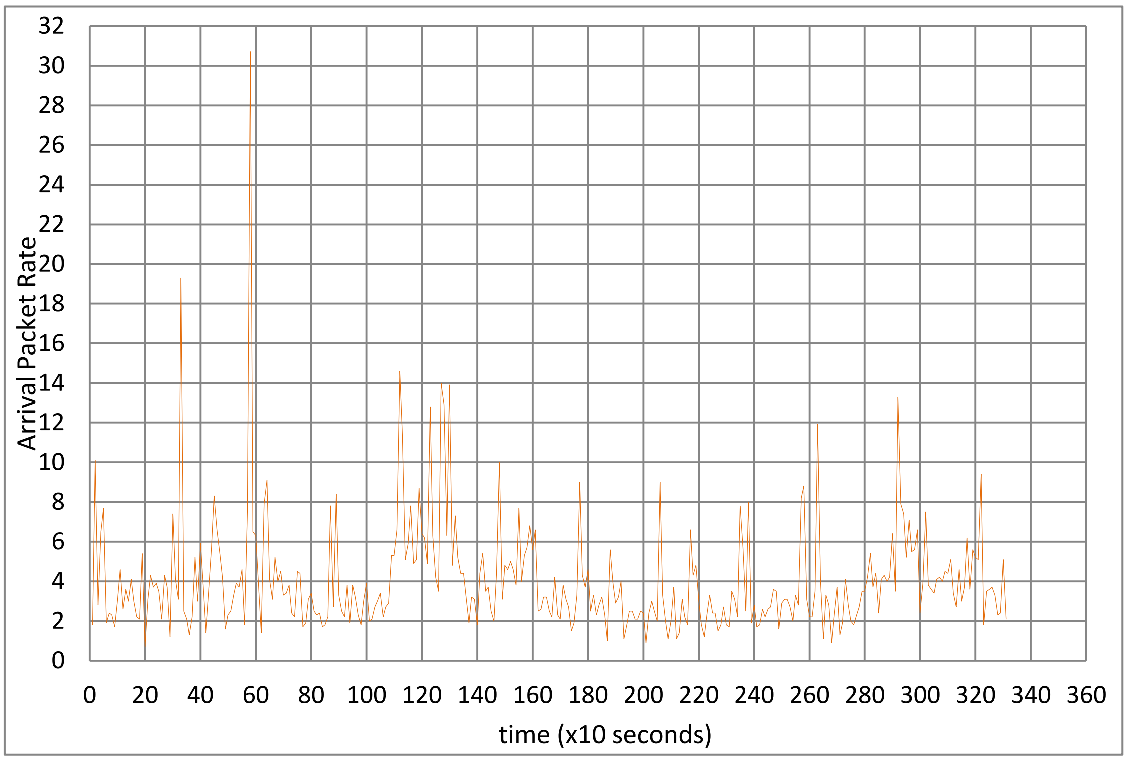

Figure A-29: The average packet rate for the captured traffic trace file 15.

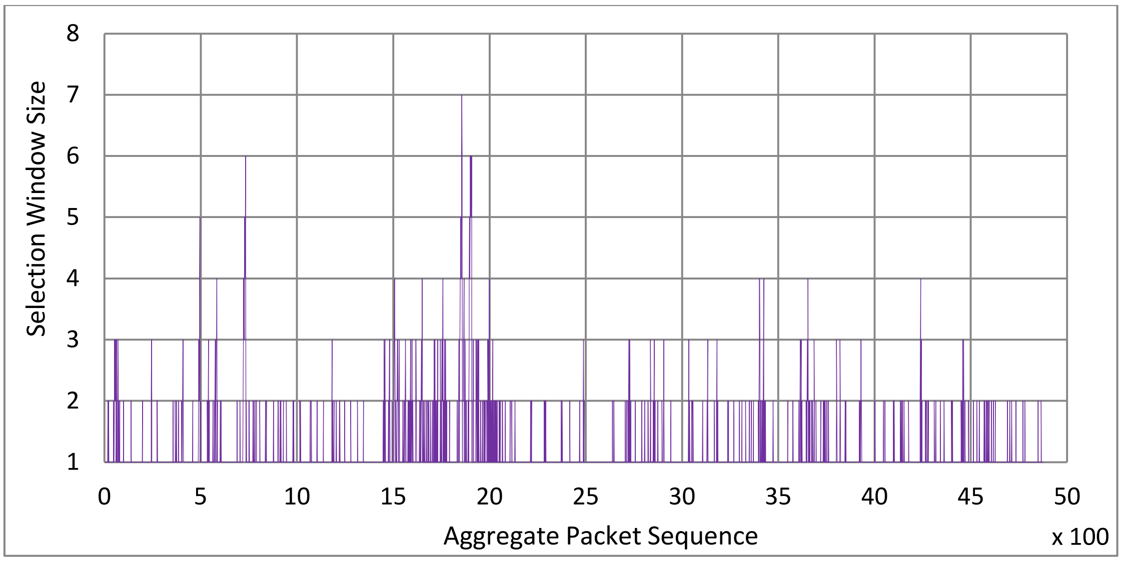

Figure A-30: The selection window size generated by the AAM algorithm for the captured traffic trace file 15.

The captured traffic trace file 16

Table A-16: The details for the captured traffic trace file 16

Time	Date	Location	PPS
19:00 –19:50	17th ,July, 2012	*Shuangliu* airport, Sichuan, China	6.9

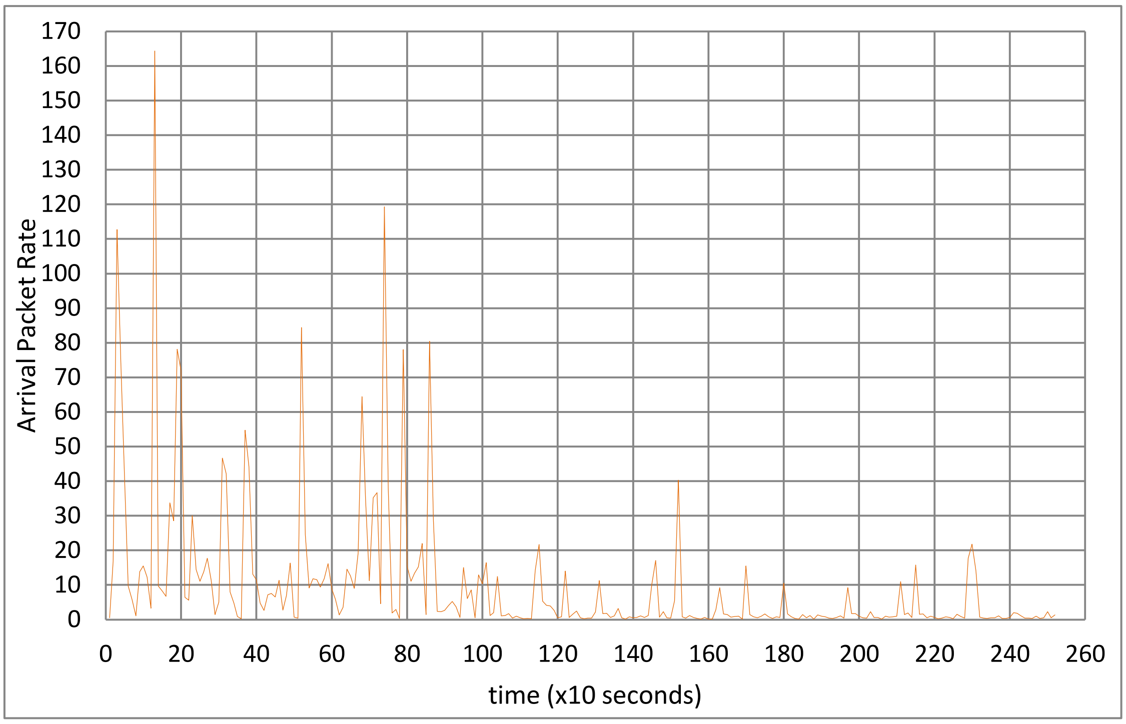

Figure A-31: The average packet rate for the captured traffic trace file 16.

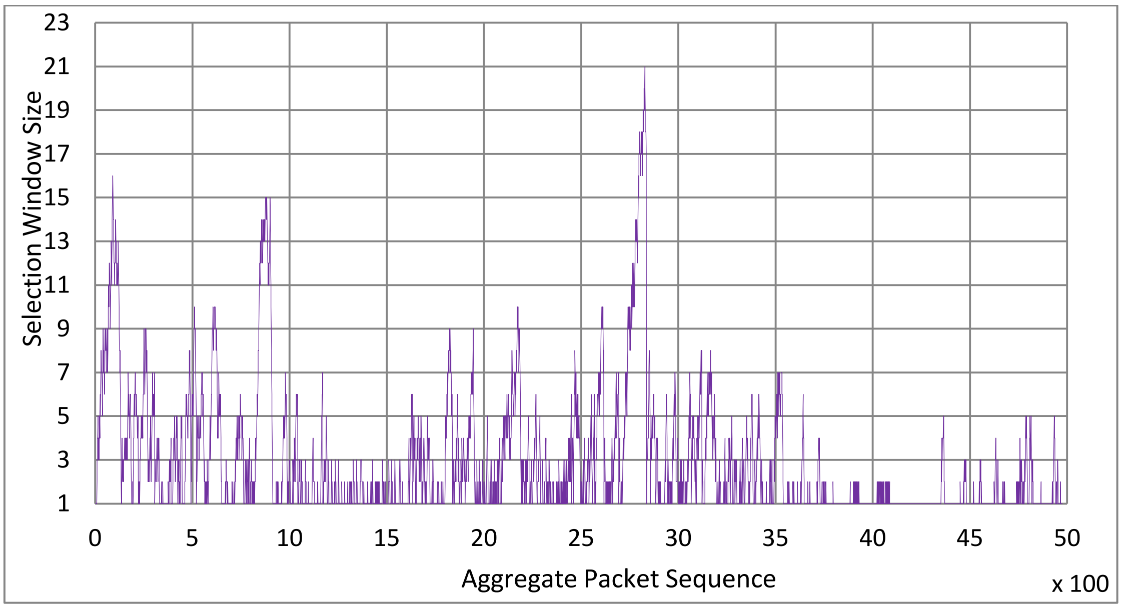

Figure A-32: The selection window size generated by the AAM algorithm for the captured traffic trace file 16.

Appendix B

The captured traffic trace file 1

Table B-1: The details for the captured traffic trace file 1

Time	Date	Location	PPS
10:30 – 11:30	29th ,May, 2012	*JAVA City,* DIT, Dublin	92.1

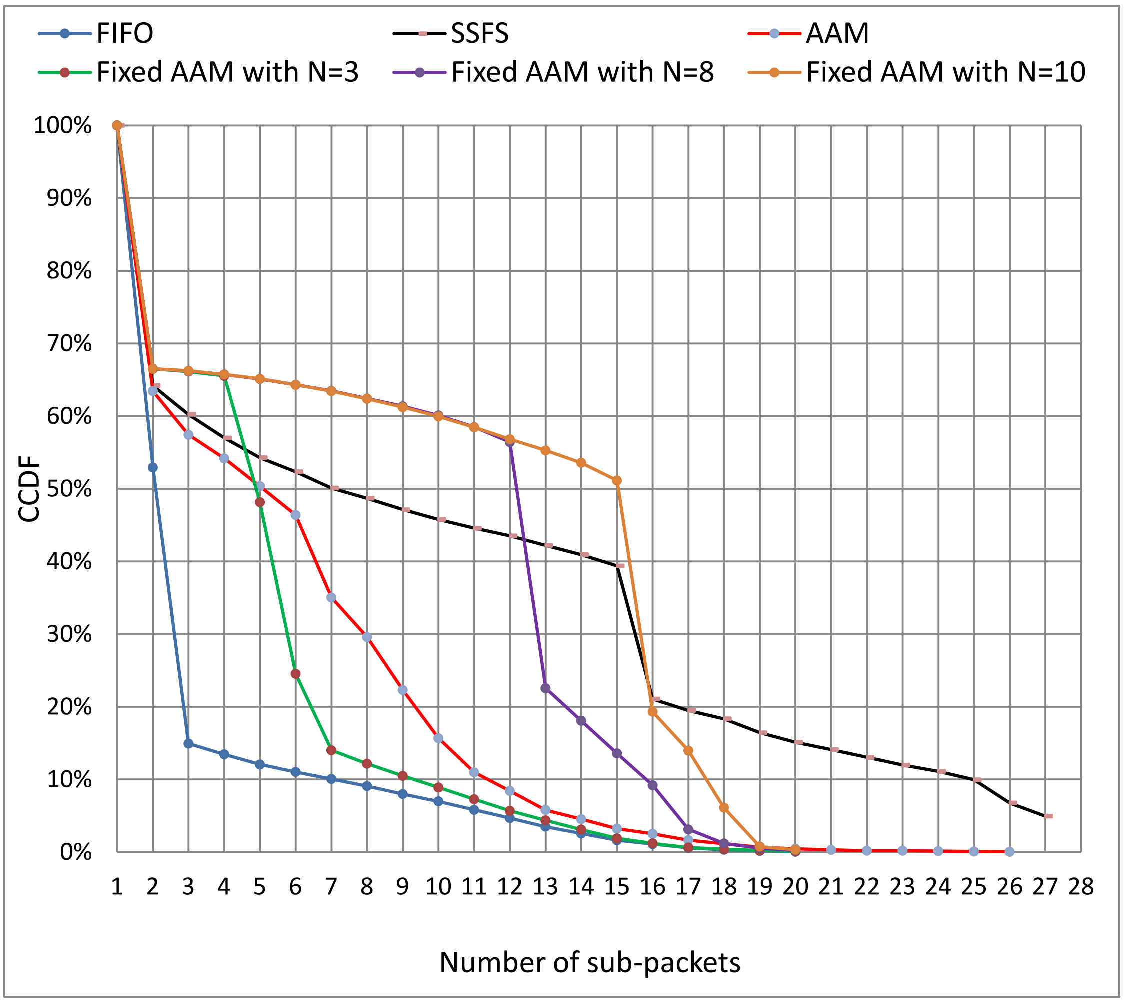

Figure B-1: The CCDF of the number of sub-packets for the captured traffic trace file 1.

The captured traffic trace file 2

Table B-2: The details for the captured traffic trace file 2

Time	Date	Location	PPS
12:00 – 13:00	29th ,May, 2012	*JAVA City,* DIT, Dublin	114

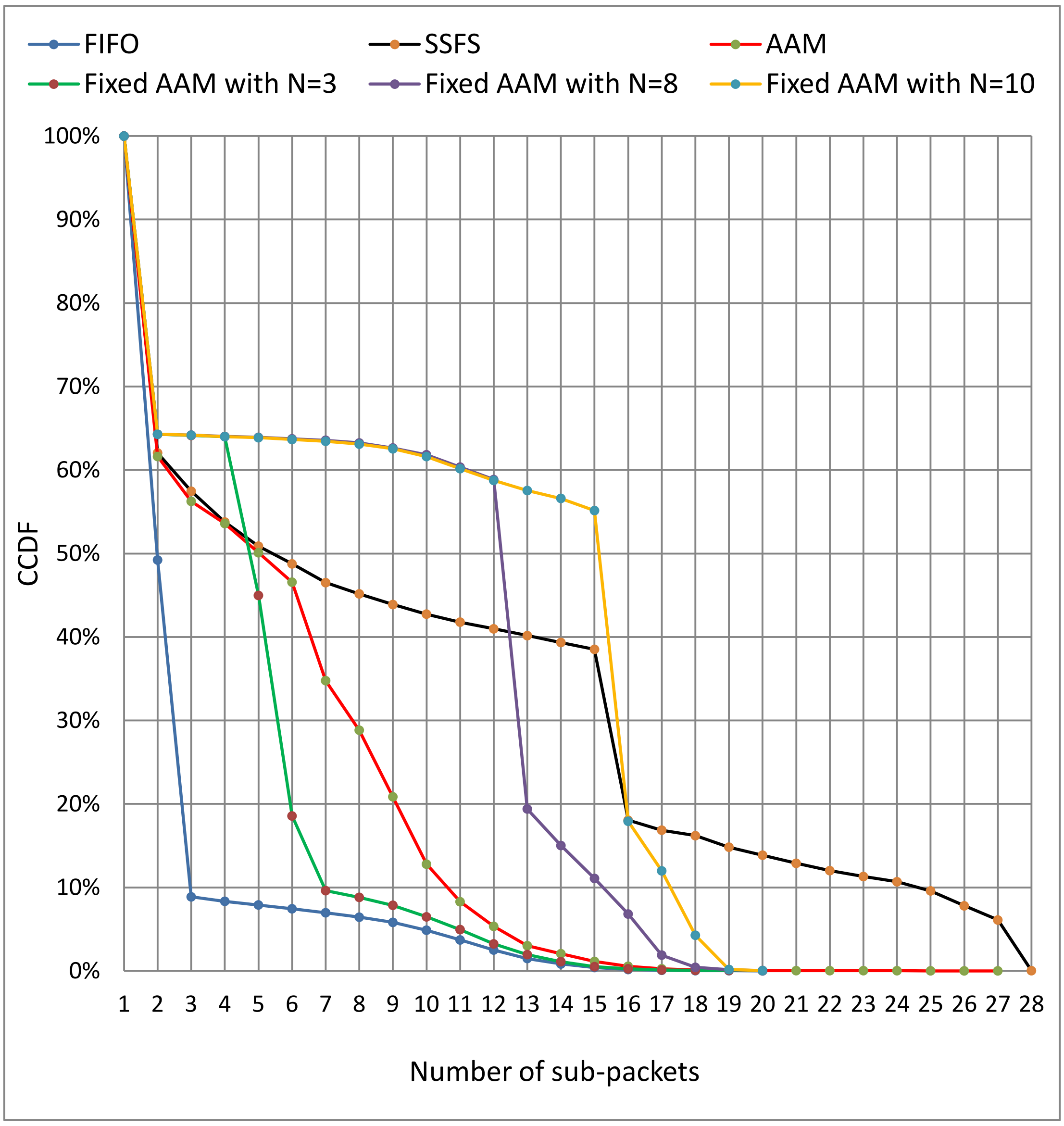

Figure B-2: The CCDF of the number of sub-packets for the captured traffic trace file 2.

The captured traffic trace file 3

Table B-3: The details for the captured traffic trace file 3

Time	Date	Location	PPS
14:00 – 15:00	29th ,May, 2012	*JAVA City,* DIT, Dublin	75.7

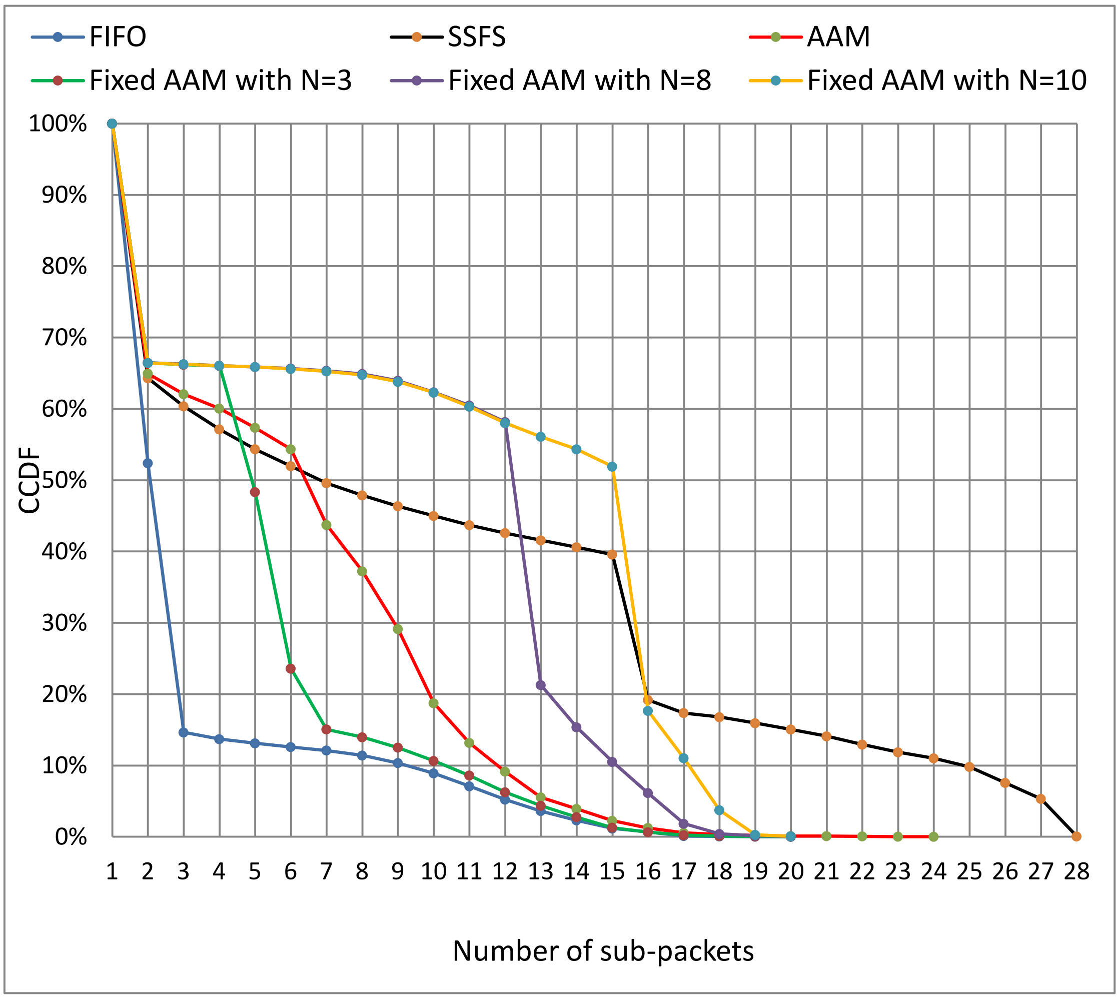

Figure B-3: The CCDF of the number of sub-packets for the captured traffic trace file 3.

The captured traffic trace file 4

Table B-4: The details for the captured traffic trace file 4

Time	Date	Location	PPS
16:00 – 17:00	29th ,May, 2012	*JAVA City,* DIT, Dublin	111.3

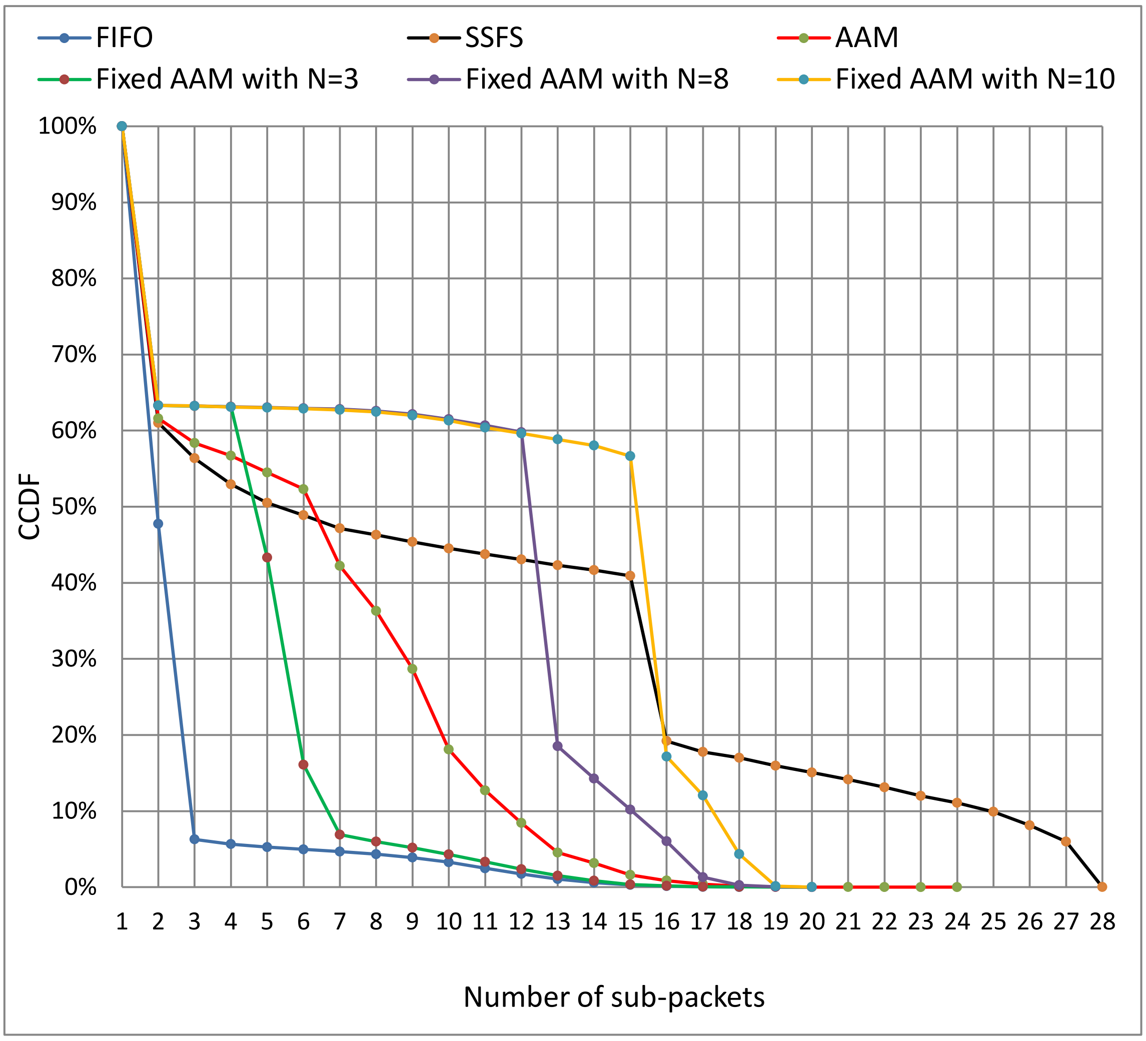

Figure B-4: The CCDF of the number of sub-packets for the captured traffic trace file 4.

The captured traffic trace file 5

Table B-5: The details for the captured traffic trace file 5

Time	Date	Location	PPS
17:30 – 18:30	29th ,May, 2012	*JAVA City,* DIT, Dublin	109.4

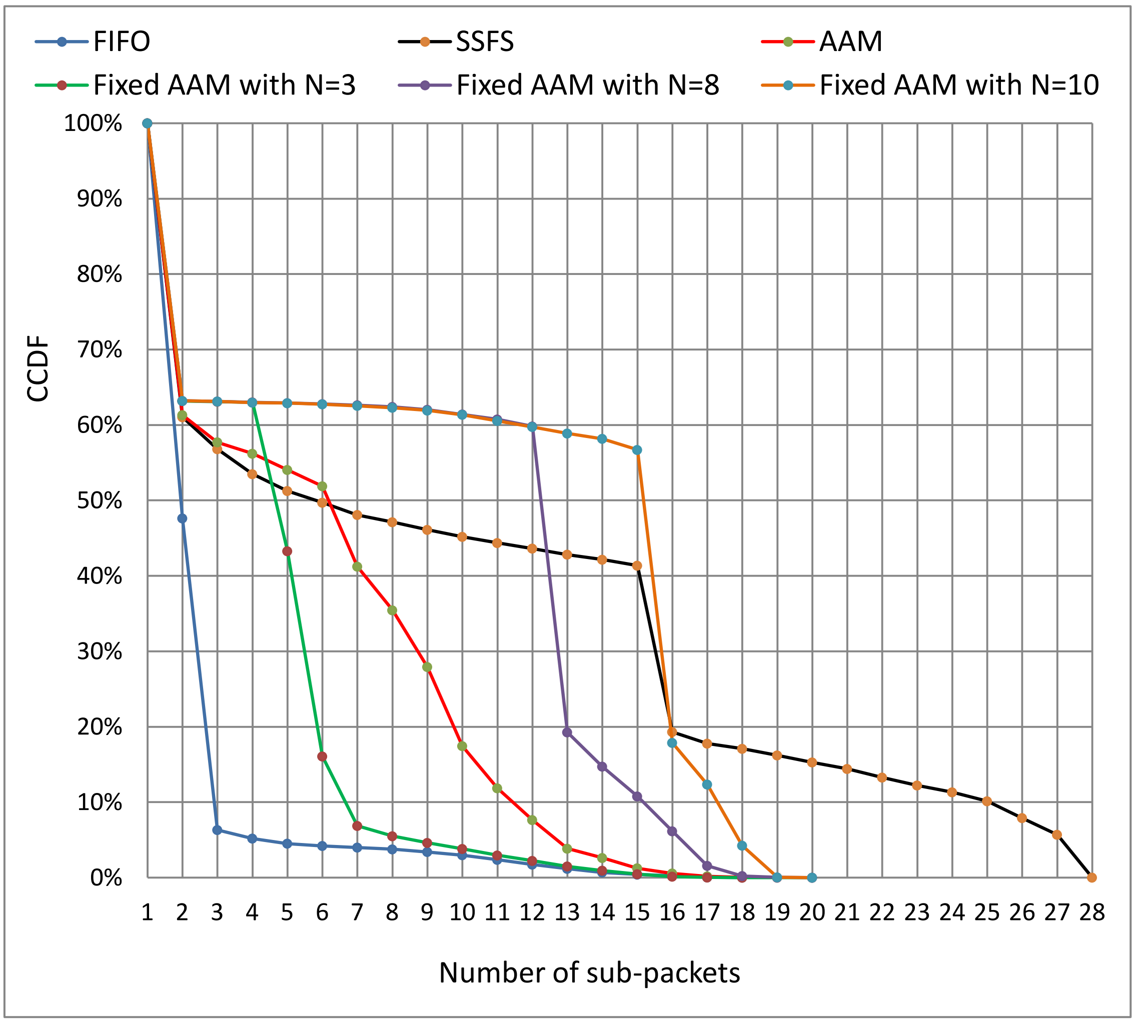

Figure B-5: The CCDF of the number of sub-packets for the captured traffic trace file 5.

The captured traffic trace file 6

Table B-6: The details for the captured traffic trace file 6

Time	Date	Location	PPS
09:30 – 10:30	19th, June, 2012	*Costa coffee shop,* Dublin	9.4

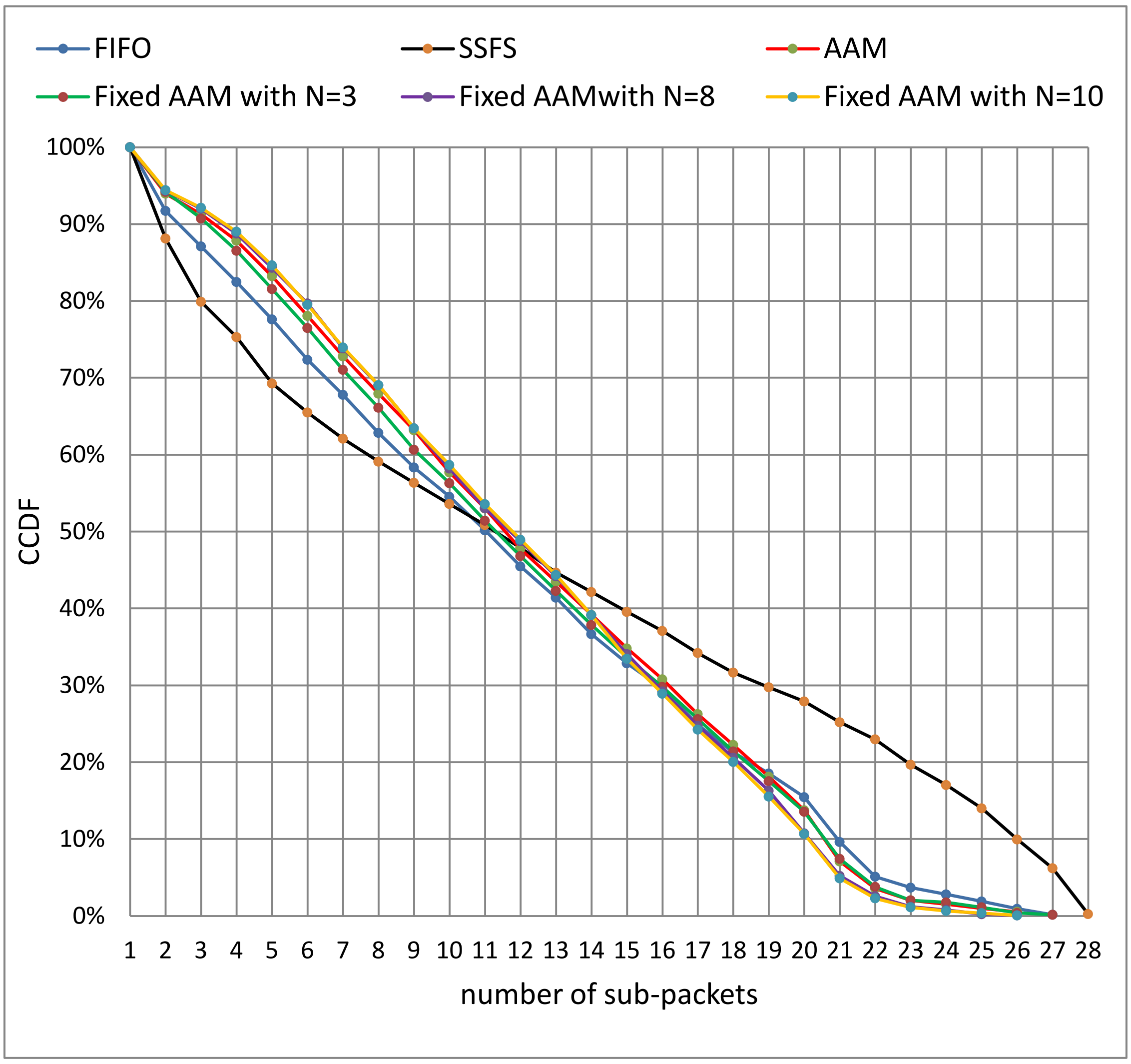

Figure B-6: The CCDF of the number of sub-packets for the captured traffic trace file 6.

The captured traffic trace file 7

Table B-7: The details for the captured traffic trace file 7

Time	Date	Location	PPS
11:00 –12:00	19th, June, 2012	*Parliament Square*, TCD, Dublin	5.6

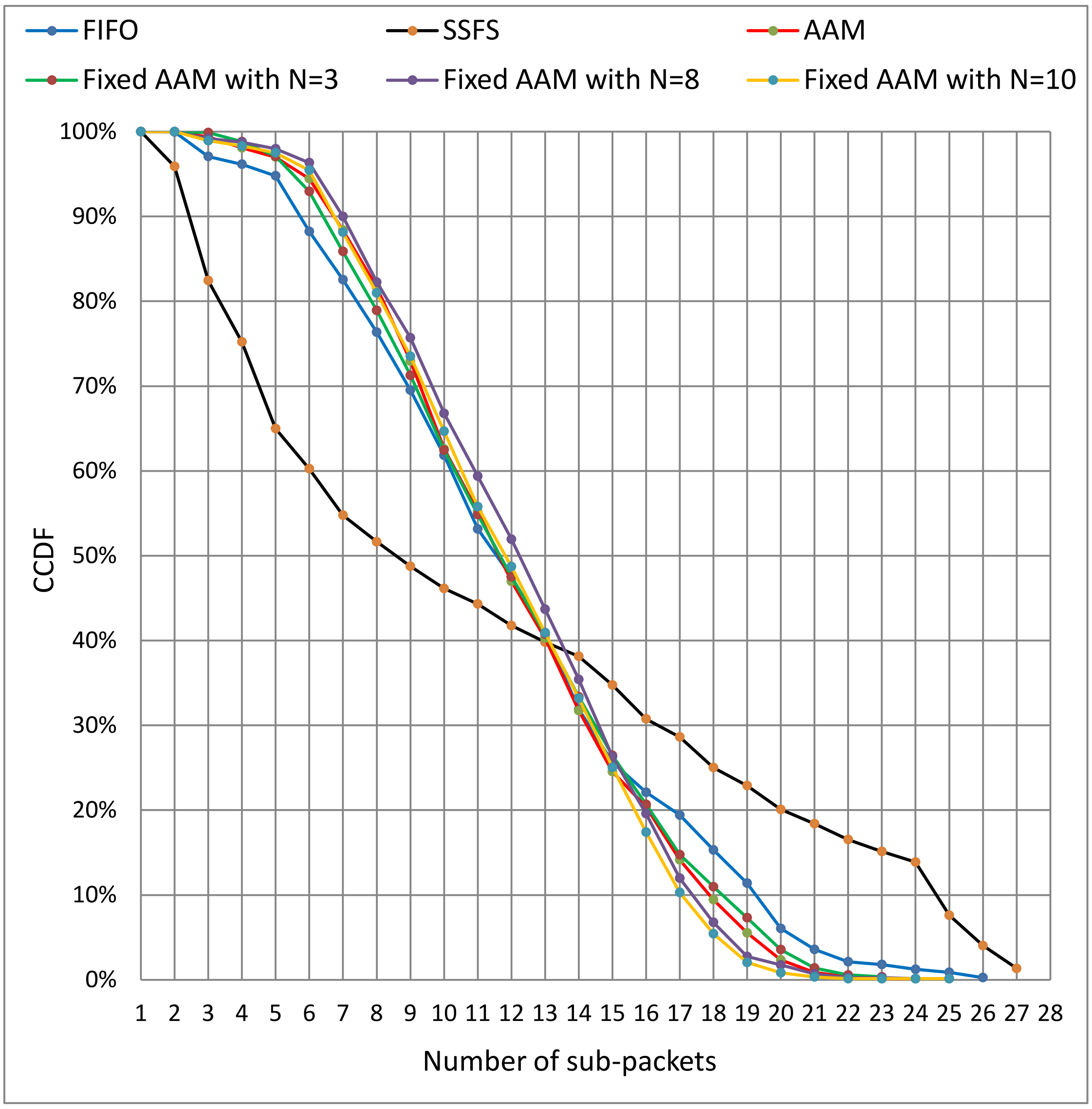

Figure B-7: The CCDF of the number of sub-packets for the captured traffic trace file 7.

The captured traffic trace file 8

Table B-8: The details for the captured traffic trace file 8

Time	Date	Location	PPS
12:30 – 13:30	19th, June, 2012	*Costa coffee shop,* Dublin	9.2

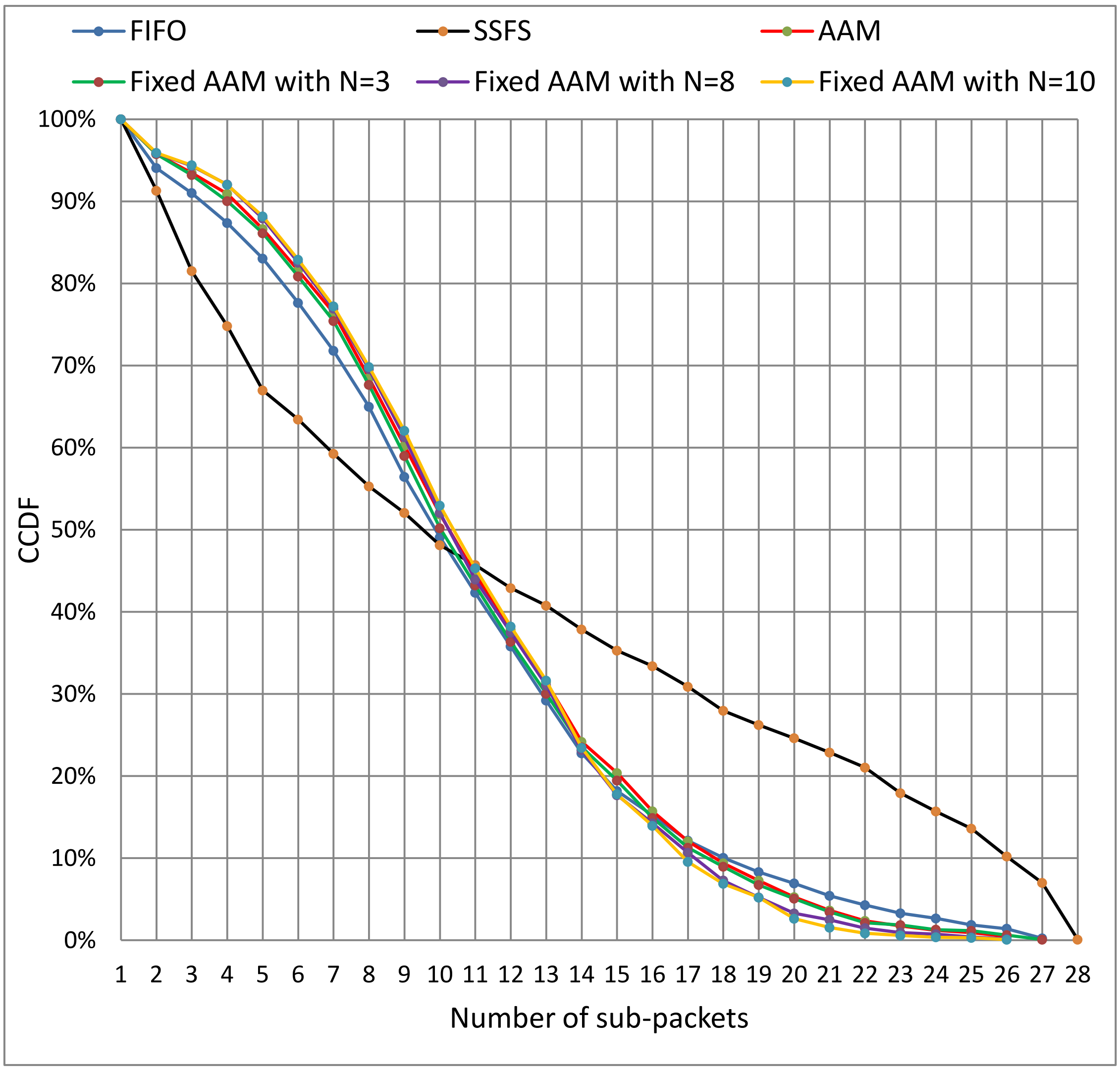

Figure B-8: The CCDF of the number of sub-packets for the captured traffic trace file 8.

The captured traffic trace file 9

Table B-9: The details for the captured traffic trace file 9

Time	Date	Location	PPS
16:00 –17:00	19th, June, 2012	*Parliament Square*, TCD, Dublin	3.6

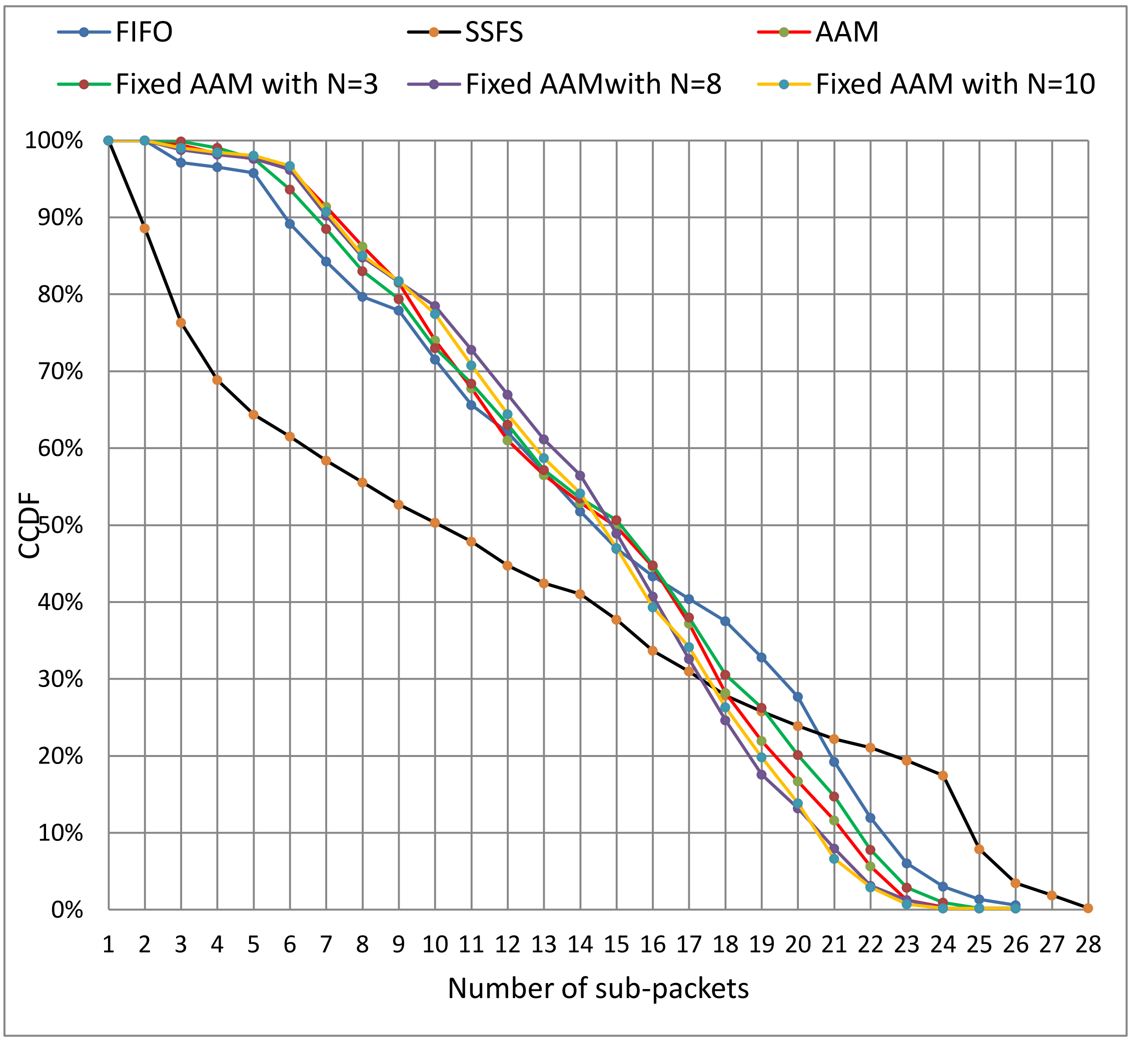

Figure B-9: The CCDF of the number of sub-packets for the captured traffic trace file 9.

The captured traffic trace file 10

Table B-10: The details for the captured traffic trace file 10

Time	Date	Location	PPS
17:00 – 18:00	19th, June, 2012	*Costa coffee shop,* Dublin	6.6

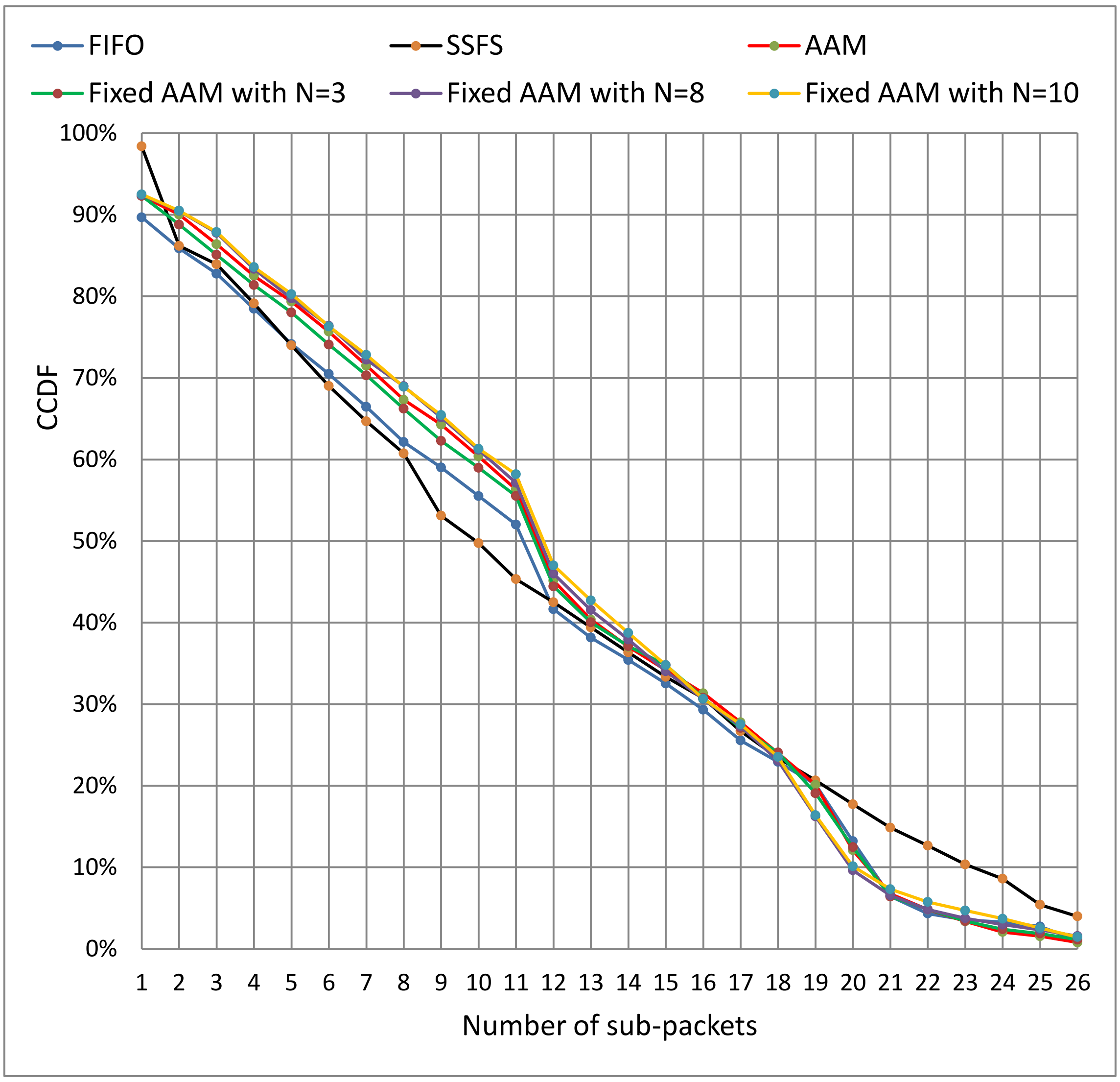

Figure B-10: The CCDF of the number of sub-packets for the captured traffic trace file 10.

The captured traffic trace file 11

Table B-11: The details for the captured traffic trace file 11

Time	Date	Location	PPS
12:00 –13:00	24th, June, 2012	*Hueston* train station, Dublin	6.87

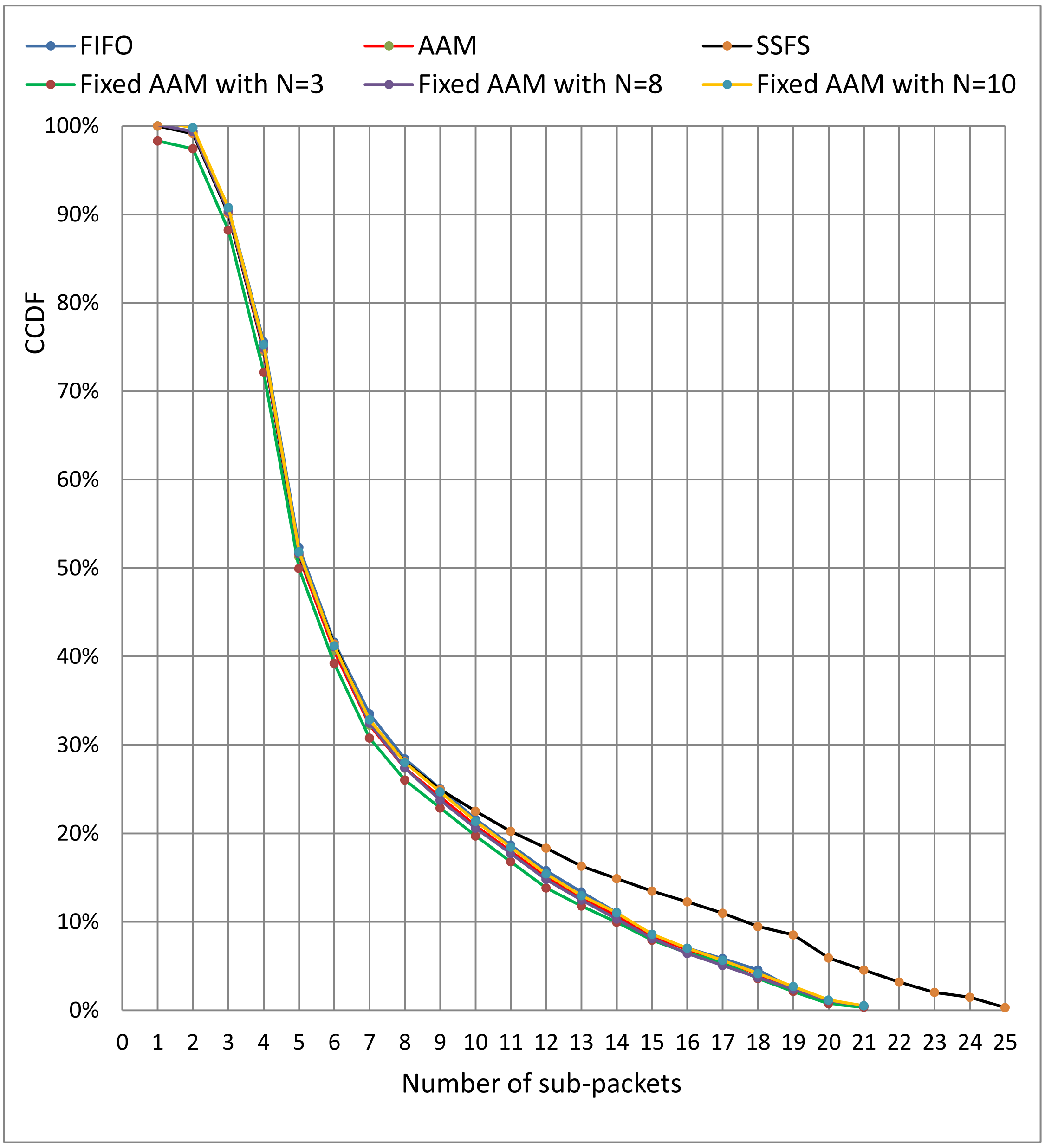

Figure B-11: The CCDF of the number of sub-packets for the captured traffic trace file 11.

The captured traffic trace file 12

Table B-12: The details for the captured traffic trace file 12

Time	Date	Location	PPS
13:30 – 14:30	24th, June, 2012	*Hueston* train station, Dublin	4.2

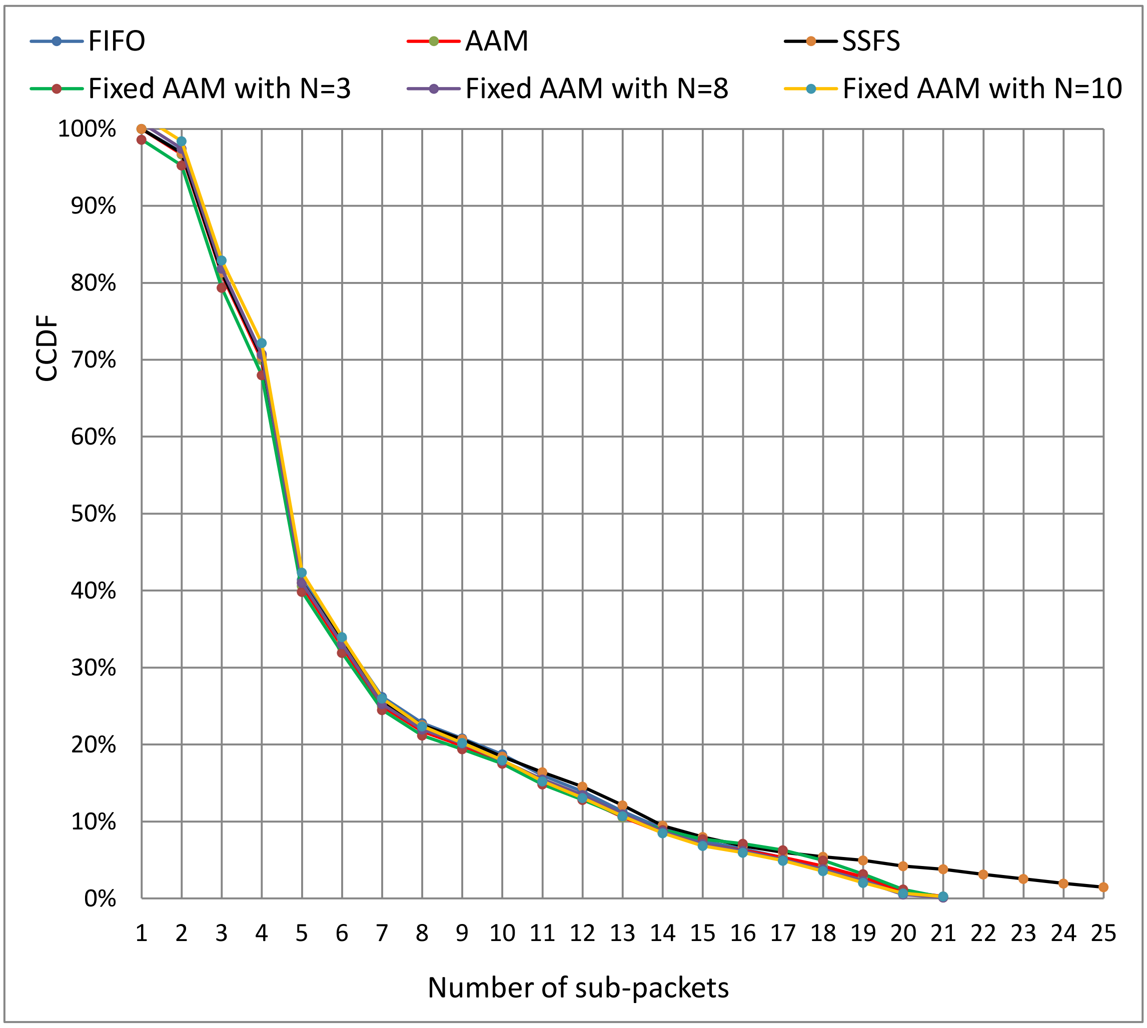

Figure B-12: The CCDF of the number of sub-packets for the captured traffic trace file 12.

The captured traffic trace file 13

Table B-13: The details for the captured traffic trace file 13

Time	Date	Location	PPS
15:00 –16:00	24th, June, 2012	*Hueston* train station, Dublin	6.2

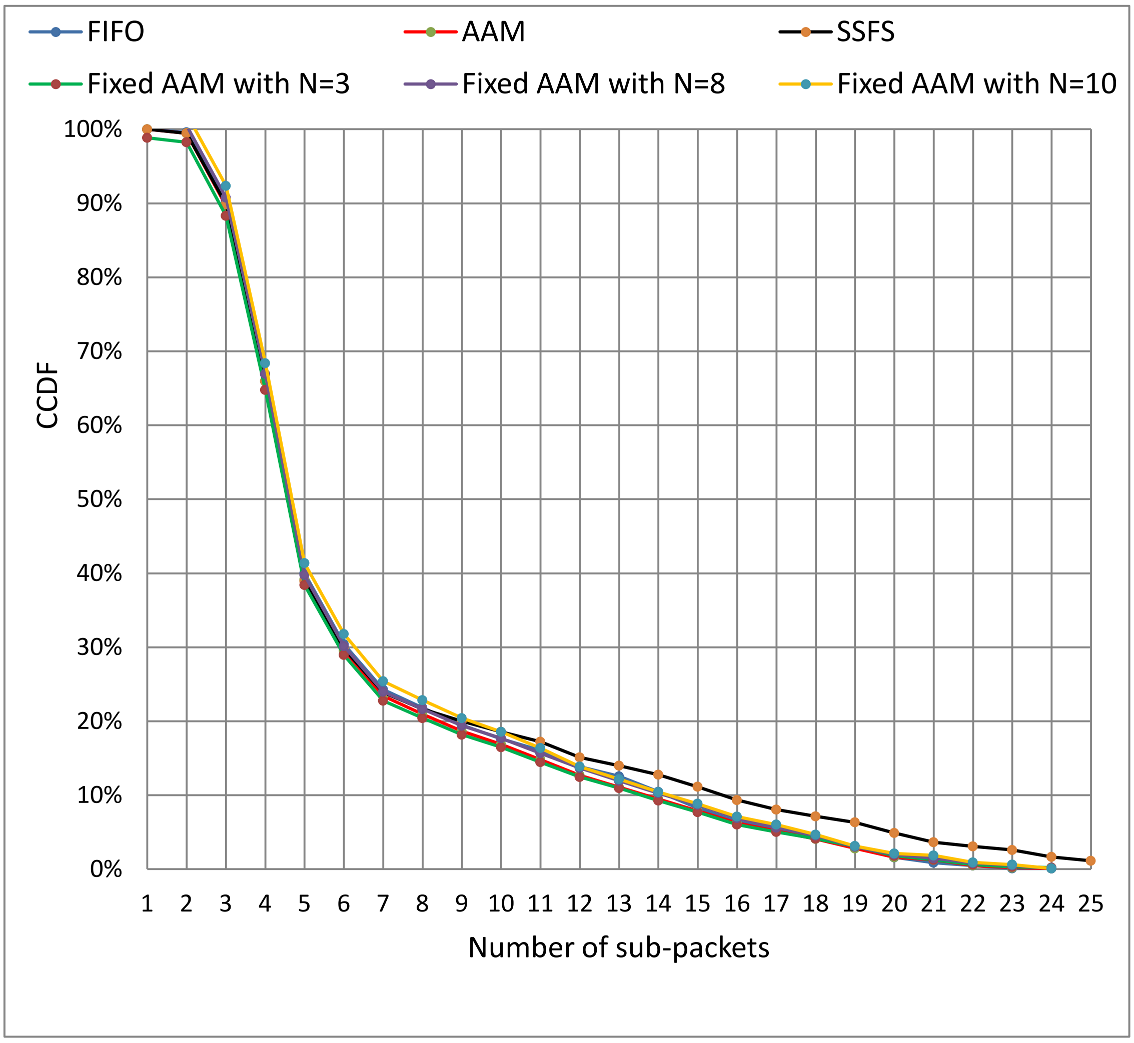

Figure B-13: The CCDF of the number of sub-packets for the captured traffic trace file 13.

The captured traffic trace file 14

Table B-14: The details for the captured traffic trace file 14

Time	Date	Location	PPS
10:30 –11:30	26th, June, 2012	Library, Kevin street, DIT, Dublin	19.2

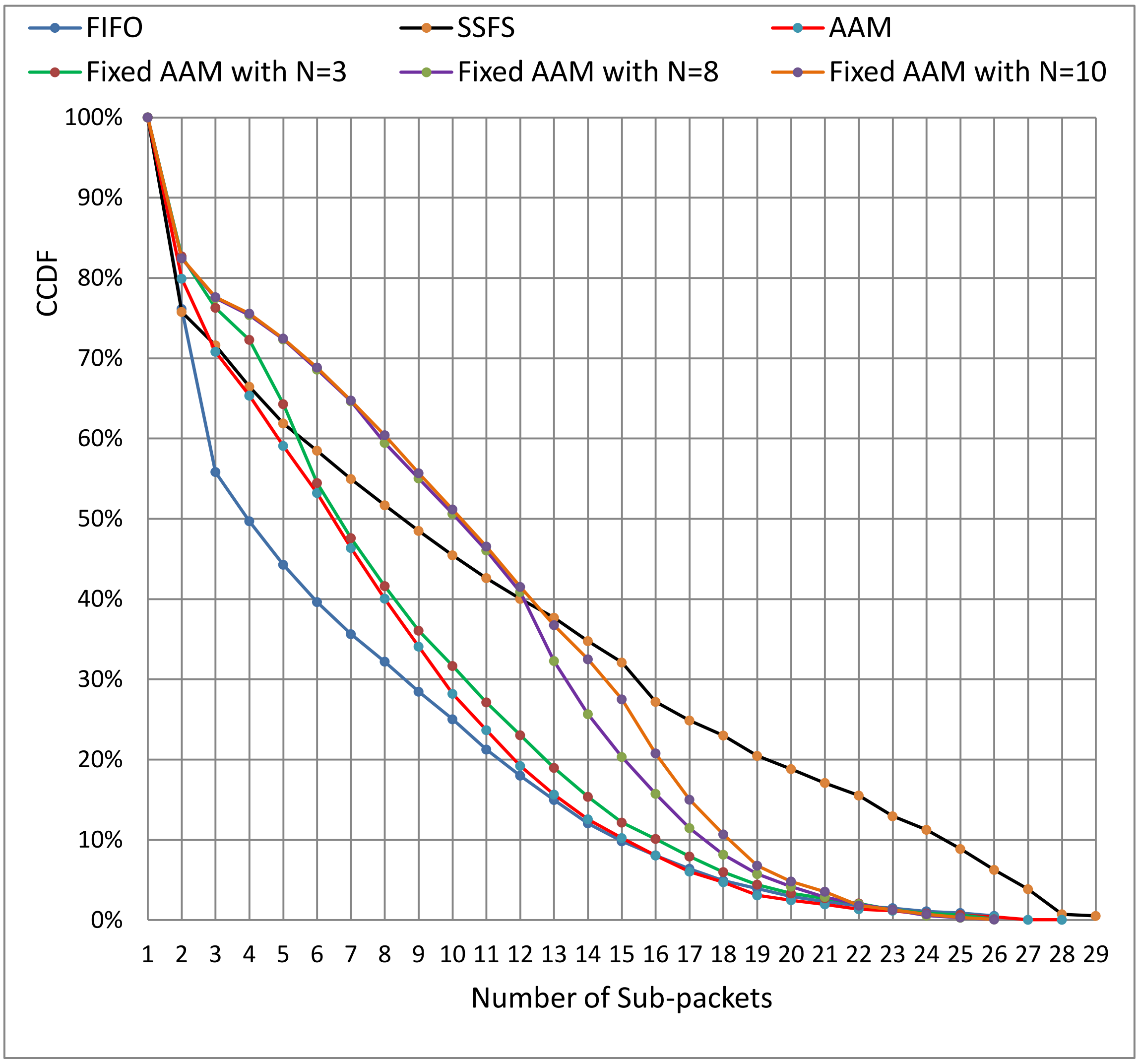

Figure B-14: The CCDF of the number of sub-packets for the captured traffic trace file 14.

The captured traffic trace file 15

Table B-15: The details for the captured traffic trace file 15

Time	Date	Location	PPS
12:00 –13:00	26th, June, 2012	Library, Kevin street, DIT, Dublin	3.8

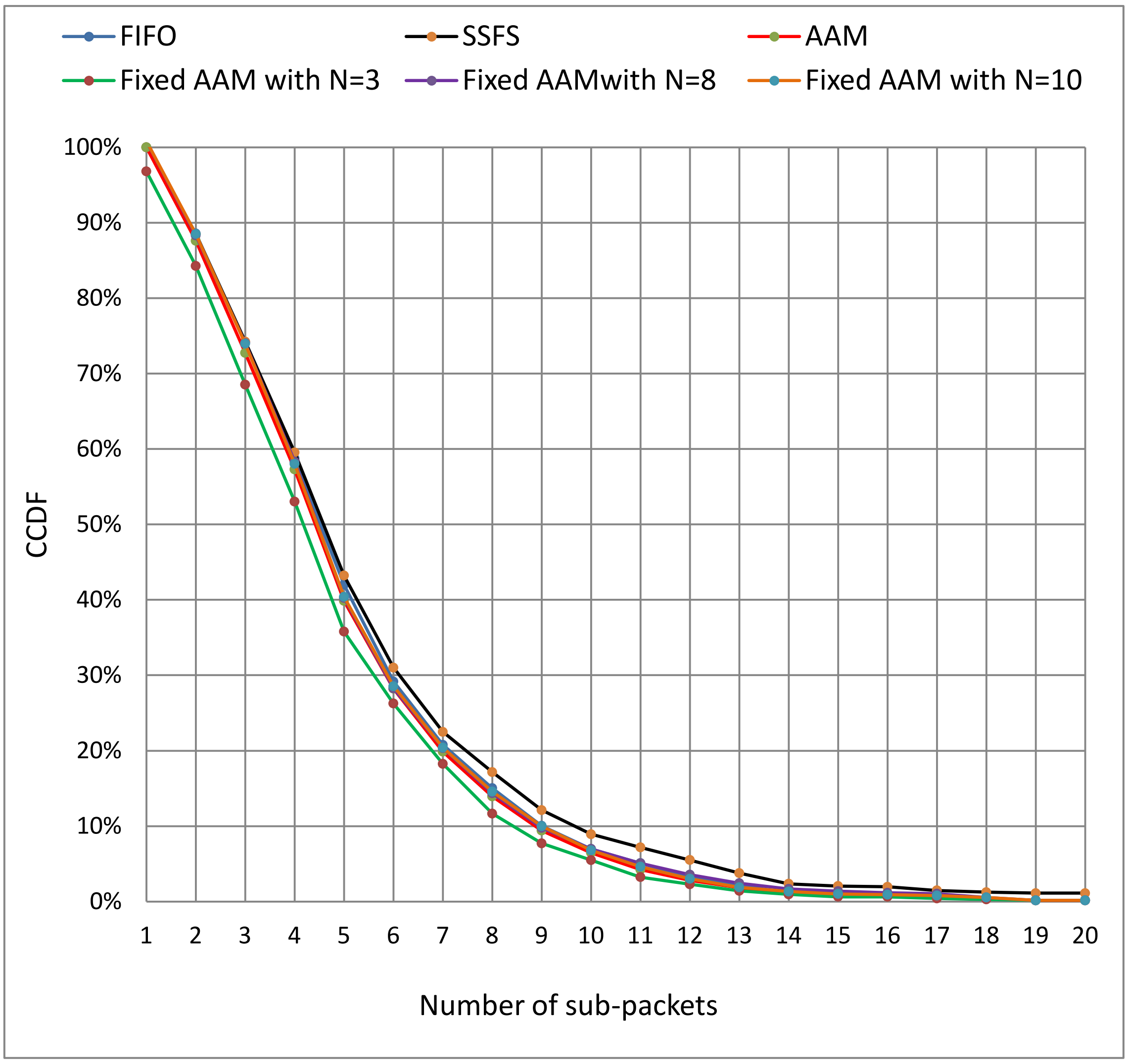

Figure B-15: The CCDF of the number of sub-packets for the captured traffic trace file 15.

The captured traffic trace file 16

Table B-16: The details for the captured traffic trace file 16

Time	Date	Location	PPS
19:00 –19:50	17th ,July, 2012	*Shuangliu* airport, Sichuan, China	6.9

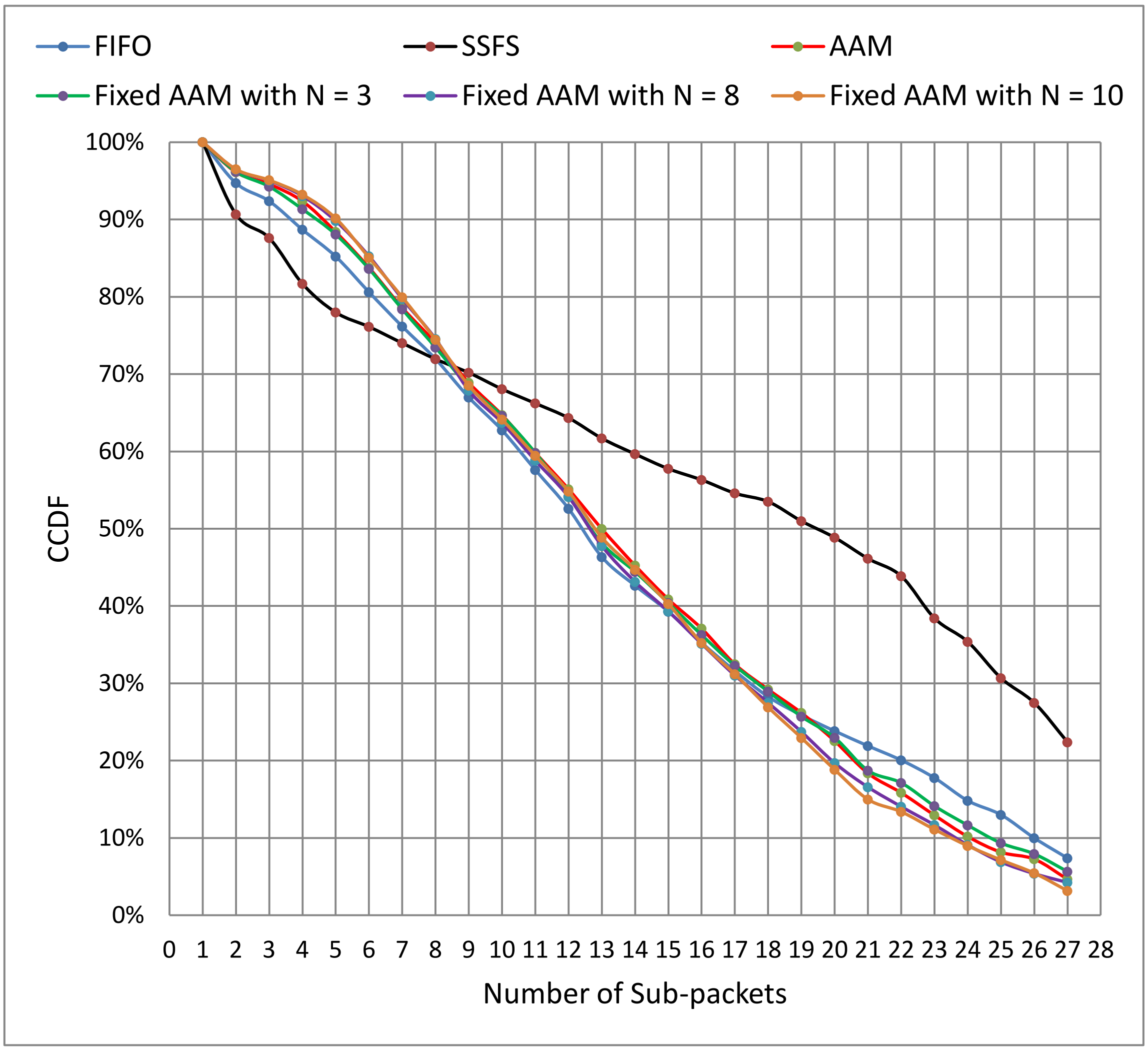

Figure B-16: The CCDF of the number of sub-packets for the captured traffic trace file 16.

Appendix C

The captured traffic trace file 1

Table C-1: The details for the captured traffic trace file 1

Time	Date	Location	PPS
10:30 – 11:30	29th ,May, 2012	*JAVA City*, DIT, Dublin	92.1

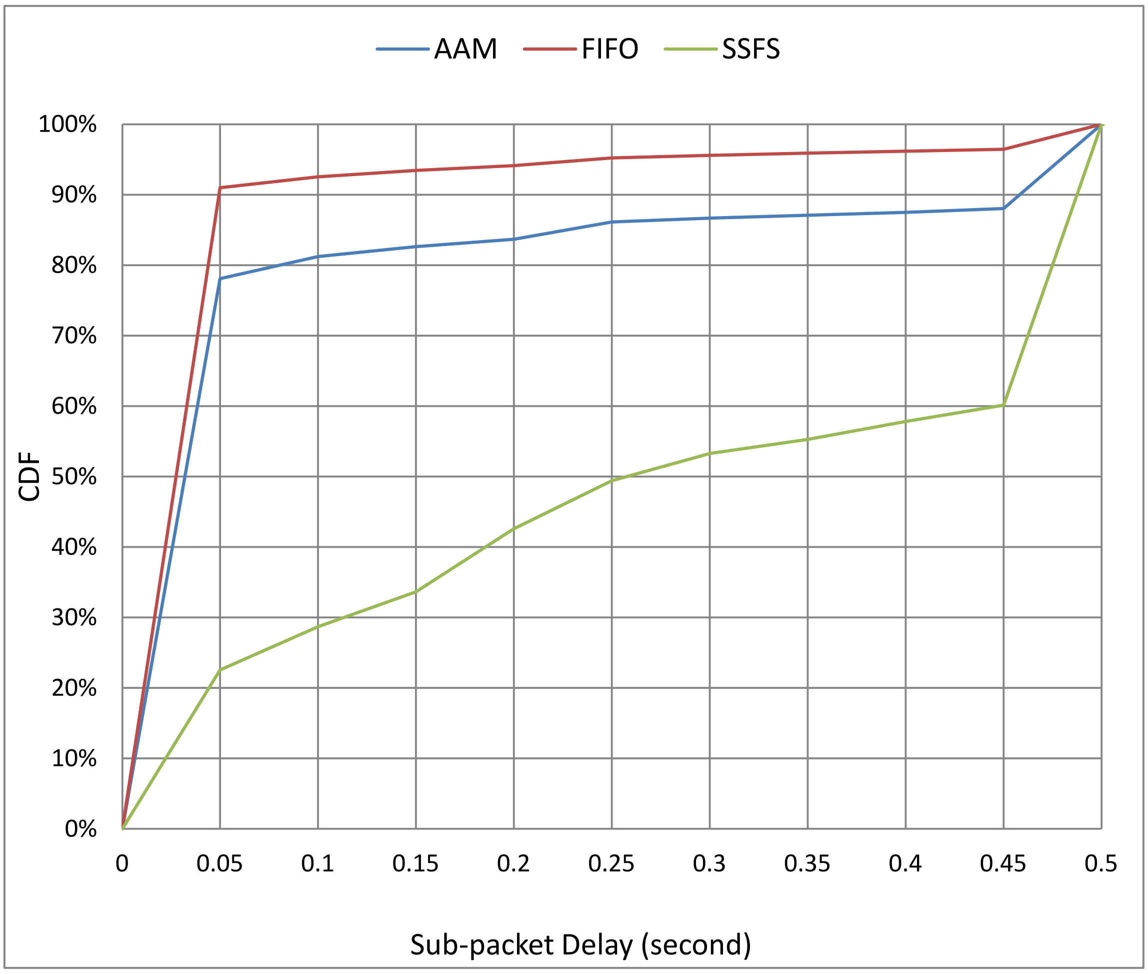

Figure C-1: The CDF of sub-packet delay for the captured traffic trace file 1.

The captured traffic trace file 2

Table C-2: The details for the captured traffic trace file 2

Time	Date	Location	PPS
12:00 – 13:00	29th ,May, 2012	*JAVA City*, DIT, Dublin	114

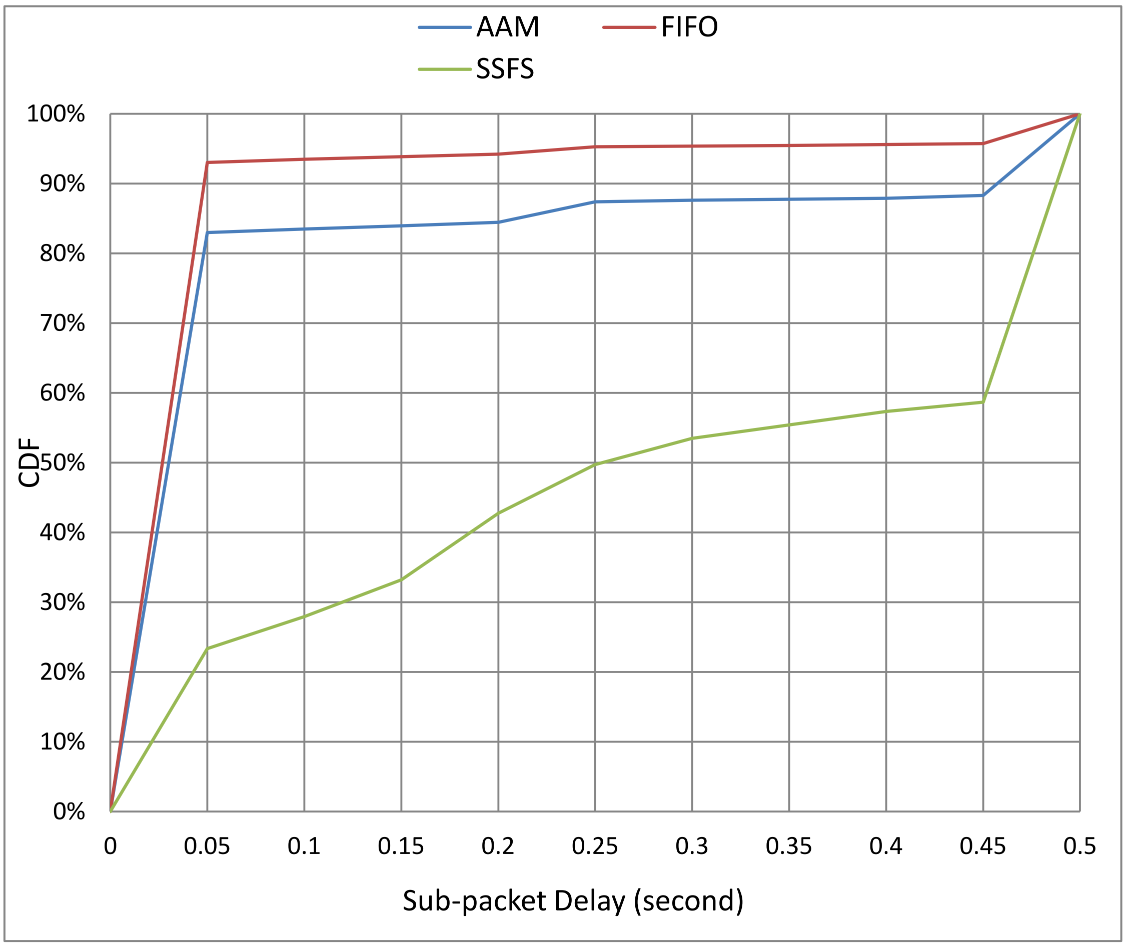

Figure C-2: The CDF of sub-packet delay for the captured traffic trace file 2.

The captured traffic trace file 3

Table C-3: The details for the captured traffic trace file 3

Time	Date	Location	PPS
14:00 – 15:00	29th ,May, 2012	*JAVA City*, DIT, Dublin	75.7

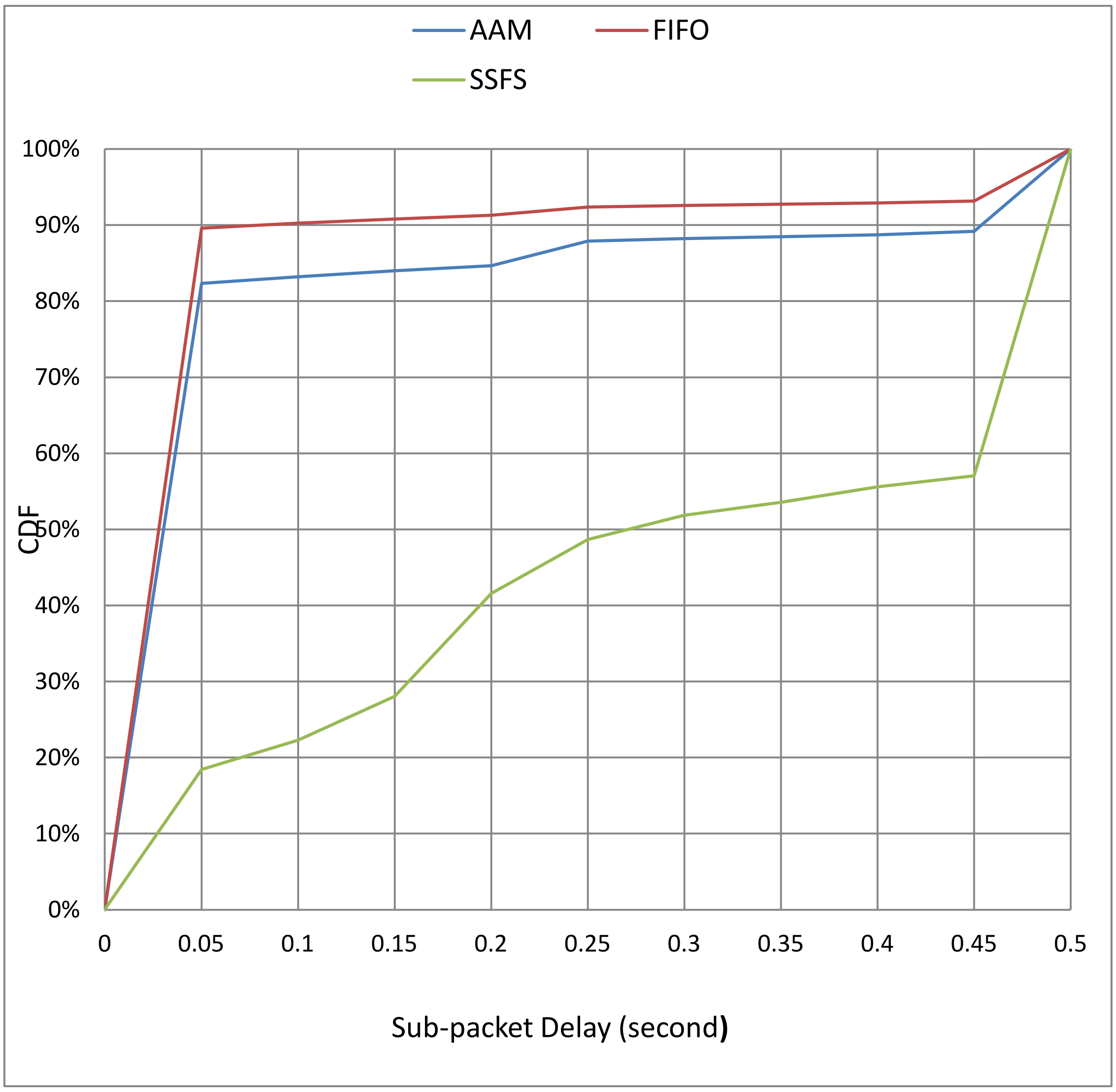

Figure C-3: The CDF of sub-packet delay for the captured traffic trace file 3.

The captured traffic trace file 4

Table C-4: The details for the captured traffic trace file 4

Time	Date	Location	PPS
16:00 – 17:00	29th ,May, 2012	*JAVA City*, DIT, Dublin	111.3

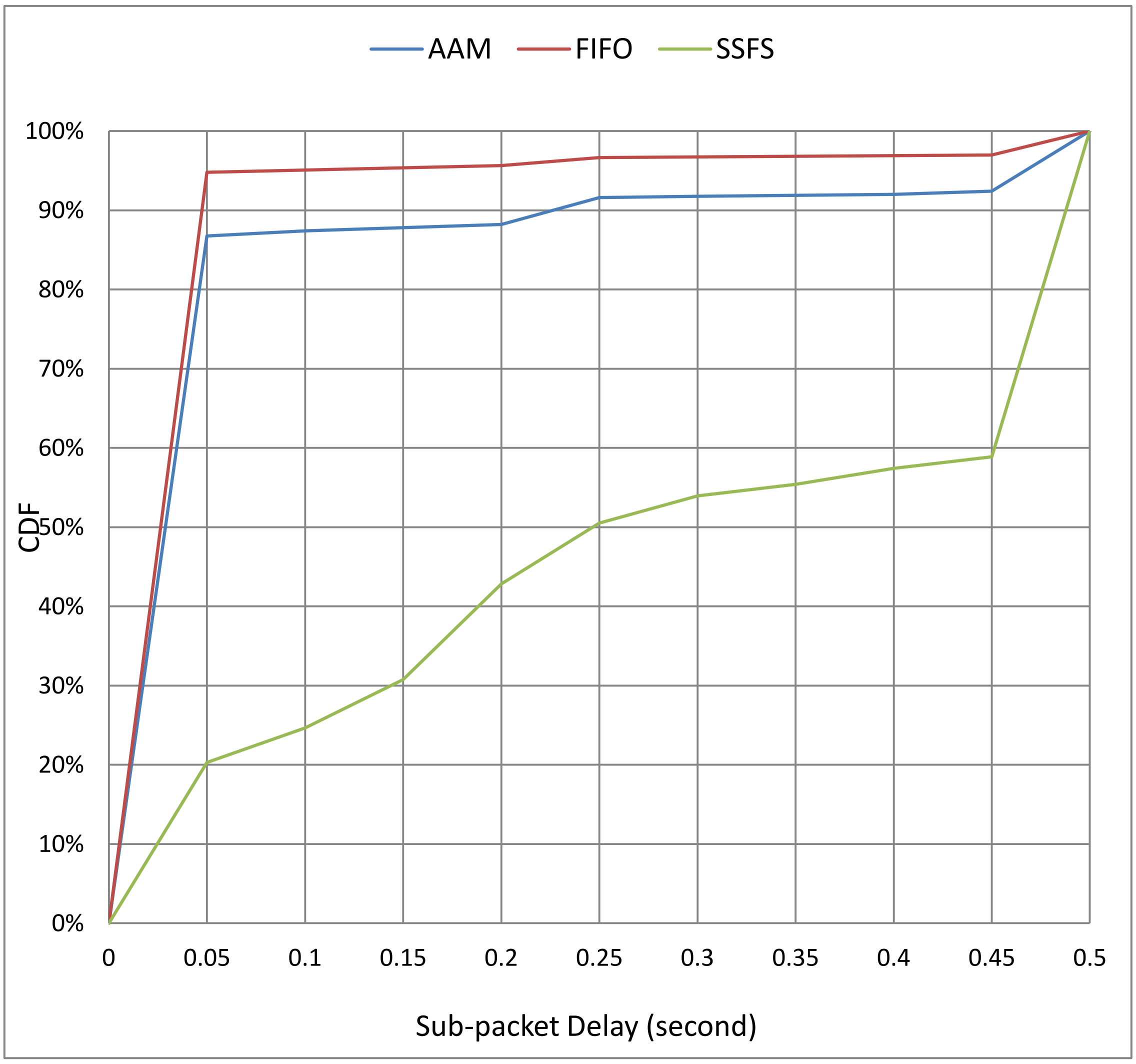

Figure C-4: The CDF of sub-packet delay for the captured traffic trace file 4.

The captured traffic trace file 5

Table C-5: The details for the captured traffic trace file 5

Time	Date	Location	PPS
17:30 – 18:30	29th ,May, 2012	*JAVA City*, DIT, Dublin	109.4

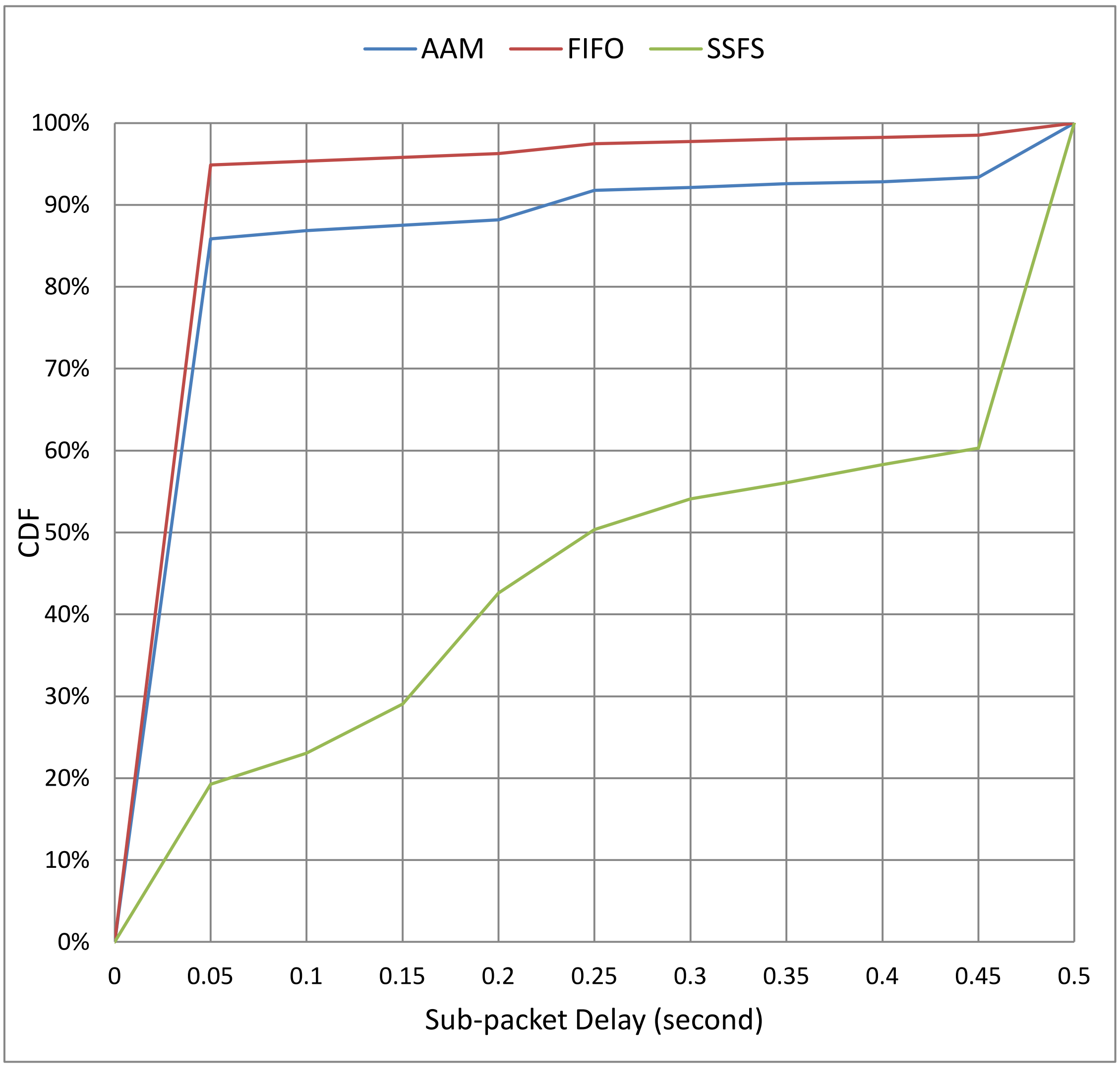

Figure C-5: The CDF of sub-packet delay for the captured traffic trace file 5.

The captured traffic trace file 6

Table C-6: The details for the captured traffic trace file 6

Time	Date	Location	PPS
09:30 – 10:30	19th, June, 2012	*Costa coffee shop*, Dublin	9.4

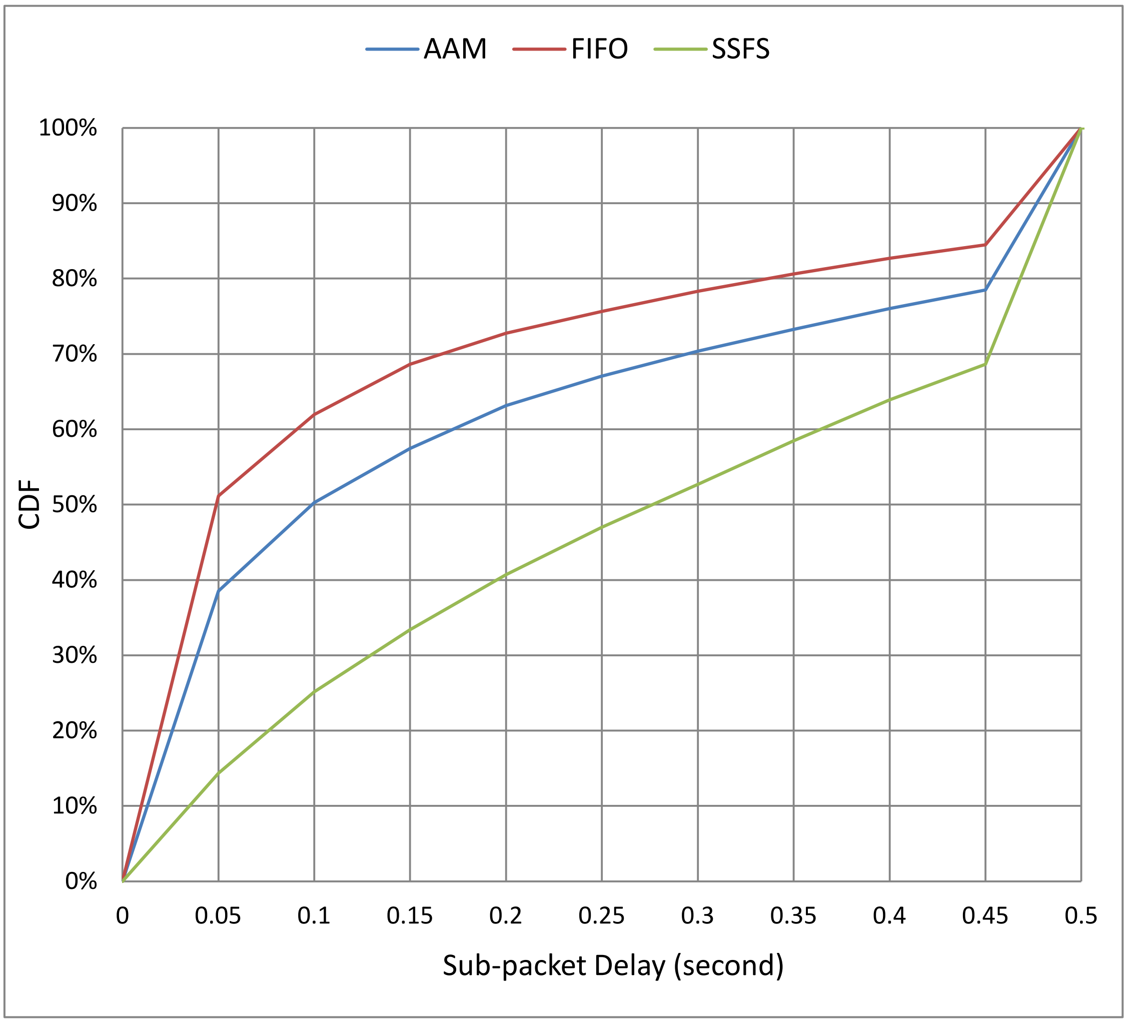

Figure C-6: The CDF of sub-packet delay for the captured traffic trace file 6.

The captured traffic trace file 7

Table C-7: The details for the captured traffic trace file 7

Time	Date	Location	PPS
11:00 –12:00	19th, June, 2012	*Parliament Square*, TCD, Dublin	5.6

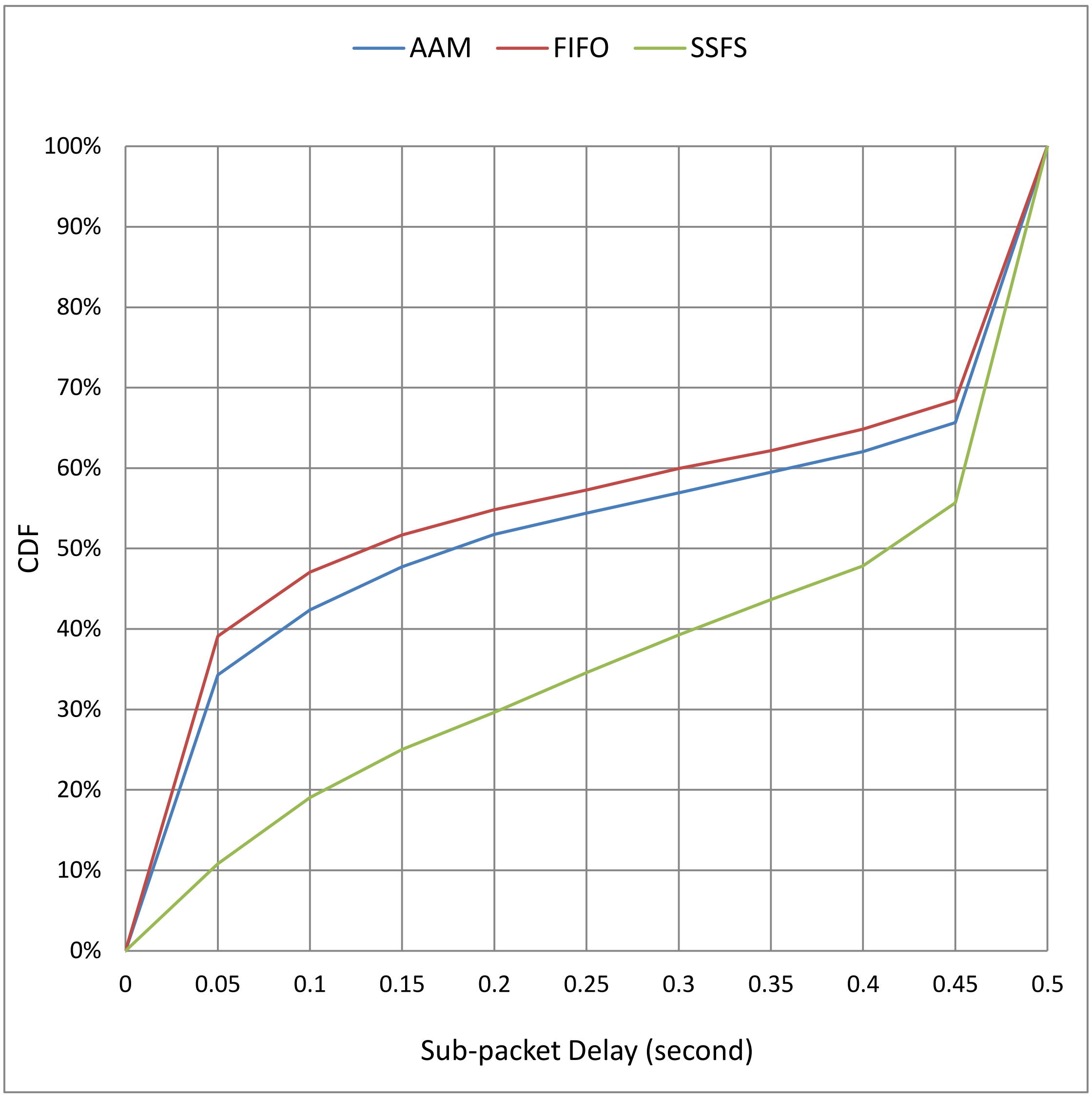

Figure C-7: The CDF of sub-packet delay for the captured traffic trace file 7.

The captured traffic trace file 8

Table C-8: The details for the captured traffic trace file 8

Time	Date	Location	PPS
12:30 – 13:30	19th, June, 2012	*Costa coffee shop*, Dublin	9.2

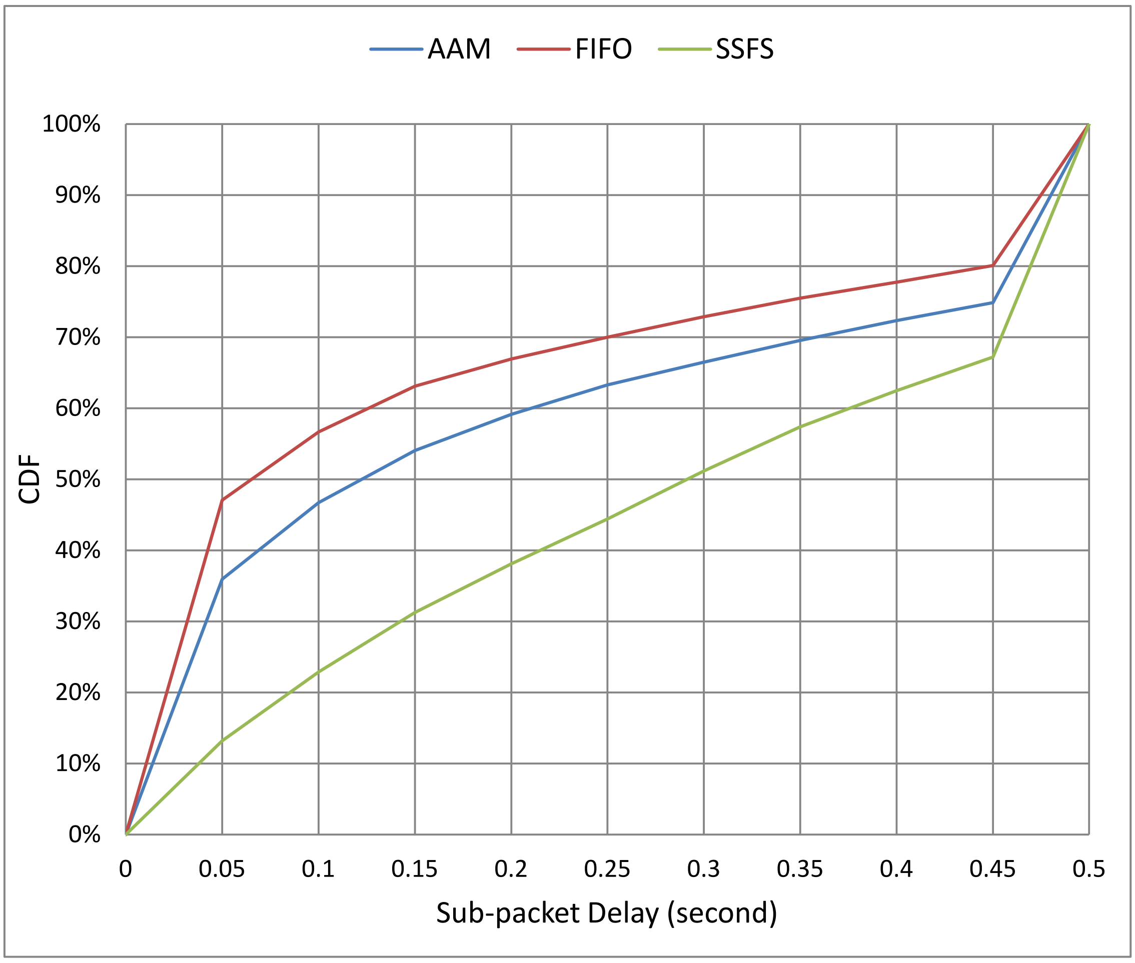

Figure C-8: The CDF of sub-packet delay for the captured traffic trace file 8.

The captured traffic trace file 9

Table C-9: The details for the captured traffic trace file 9

Time	Date	Location	PPS
16:00 –17:00	19th, June, 2012	*Parliament Square*, TCD, Dublin	3.6

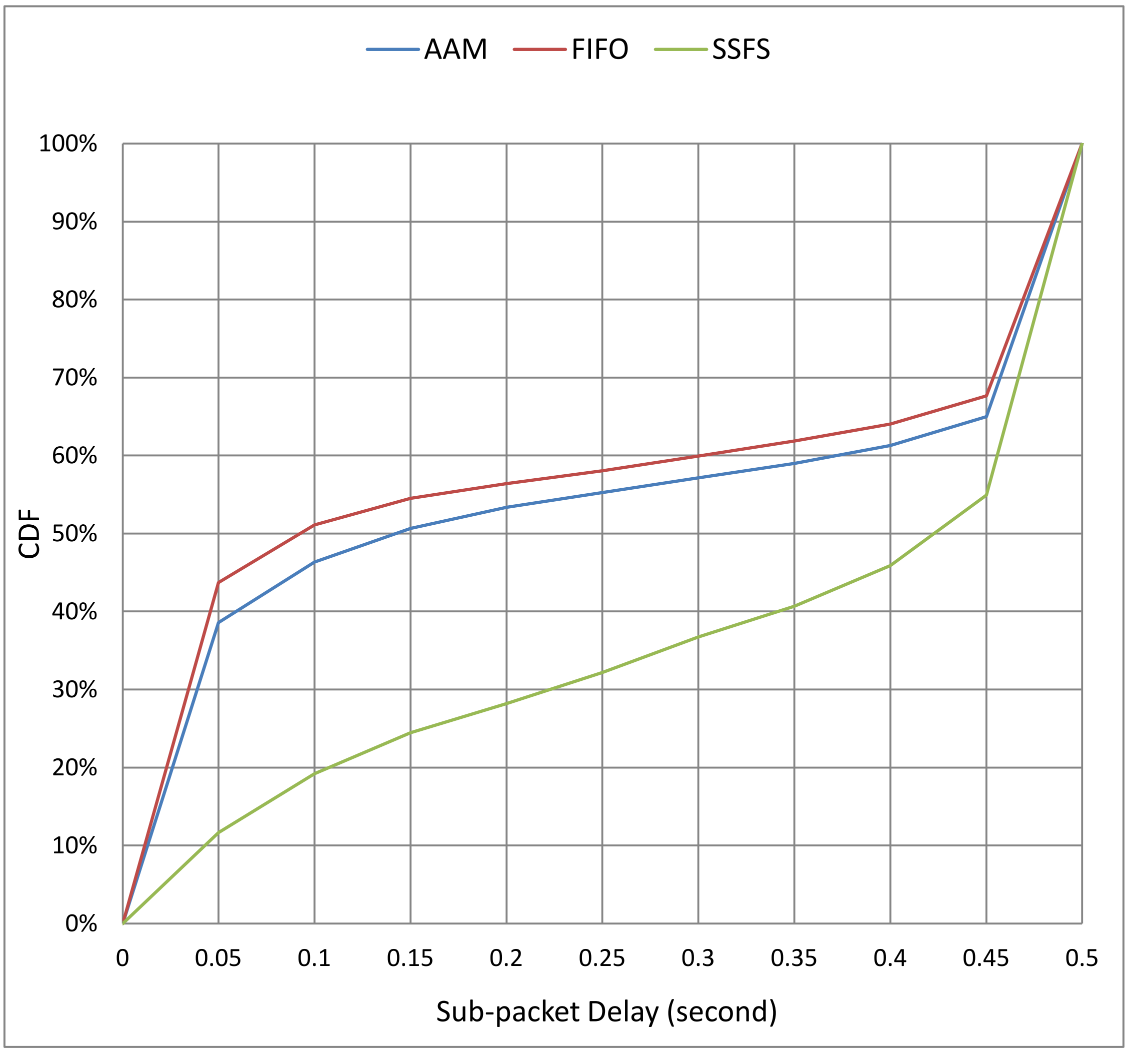

Figure C-9: The CDF of sub-packet delay for the captured traffic trace file 9.

The captured traffic trace file 10

Table C-10: The details for the captured traffic trace file 10

Time	Date	Location	PPS
17:00 – 18:00	19th, June, 2012	*Costa coffee shop*, Dublin	6.6

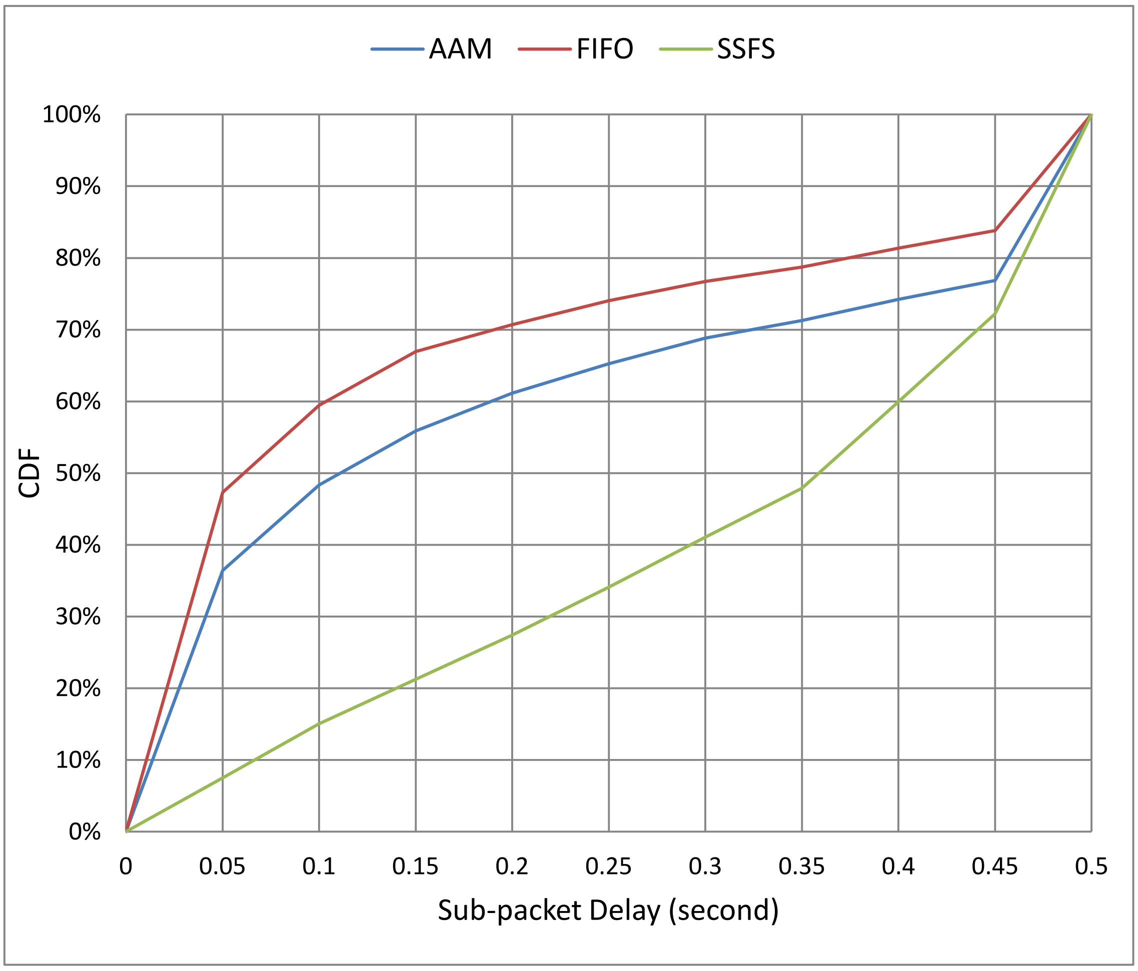

Figure C-10: The CDF of sub-packet delay for the captured traffic trace file 10.

The captured traffic trace file 11

Table C-11: The details for the captured traffic trace file 11

Time	Date	Location	PPS
12:00 –13:00	24th, June, 2012	*Hueston* train station, Dublin	6.87

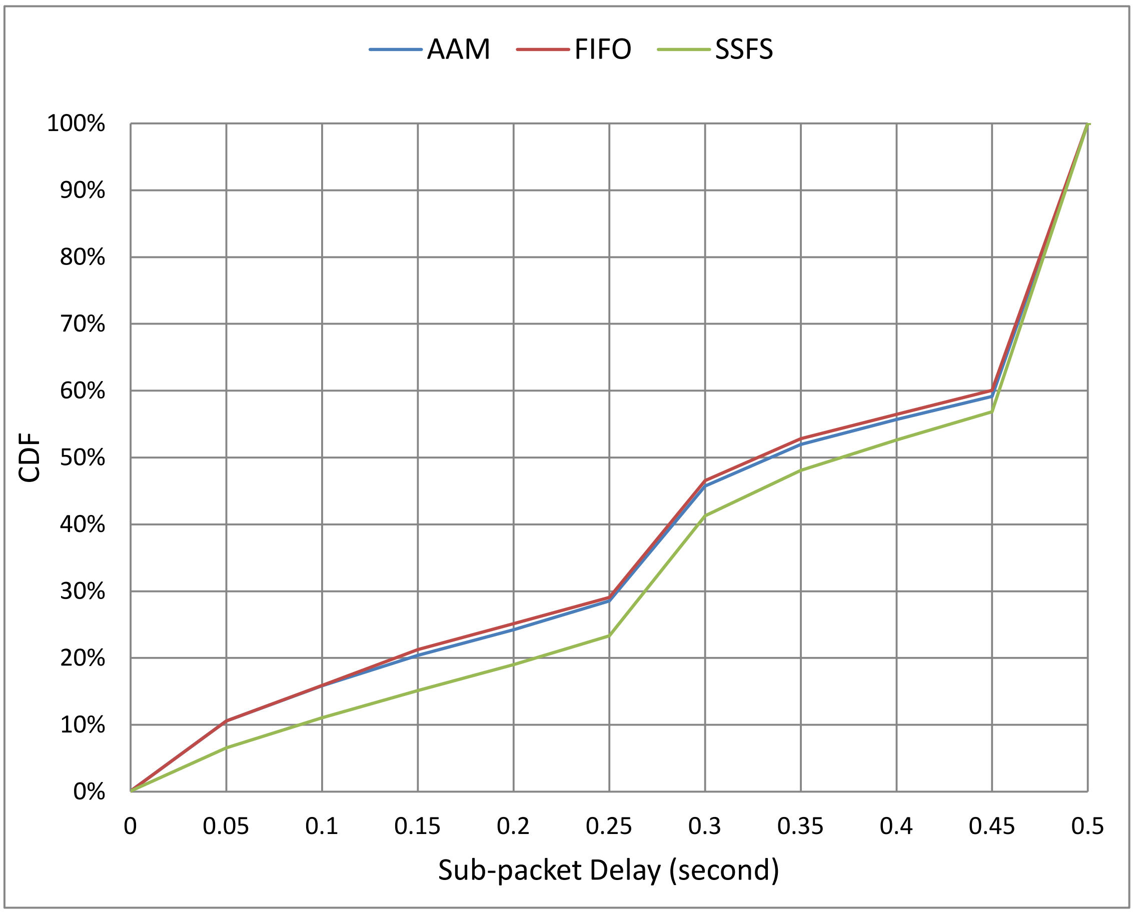

Figure C-11: The CDF of sub-packet delay for the captured traffic trace file 11.

The captured traffic trace file 12

Table C-12: The details for the captured traffic trace file 12

Time	Date	Location	PPS
13:30 – 14:30	24th, June, 2012	*Hueston* train station, Dublin	4.2

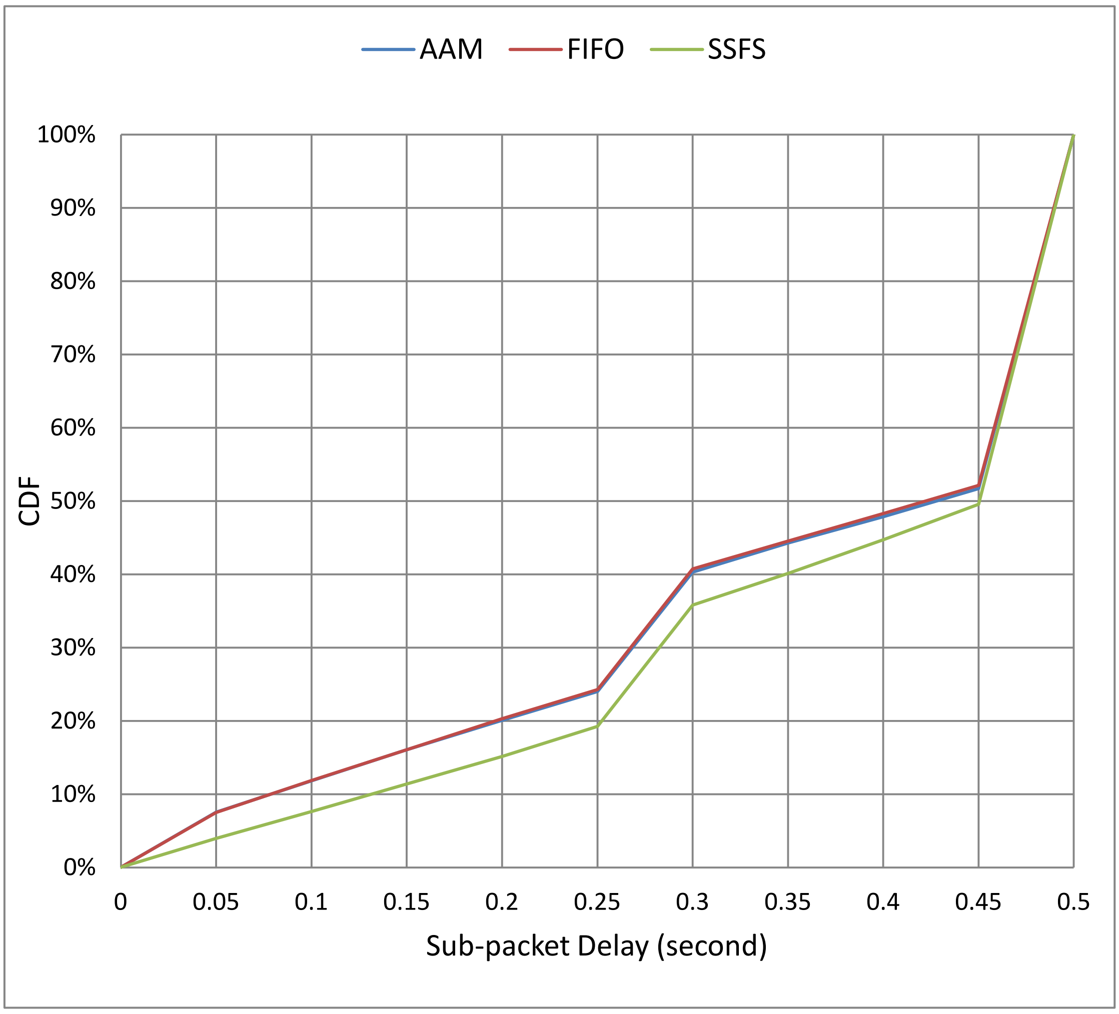

Figure C-12: The CDF of sub-packet delay for the captured traffic trace file 12.

The captured traffic trace file 13

Table C-13: The details for the captured traffic trace file 13

Time	Date	Location	PPS
15:00 –16:00	24th, June, 2012	*Hueston* train station, Dublin	6.2

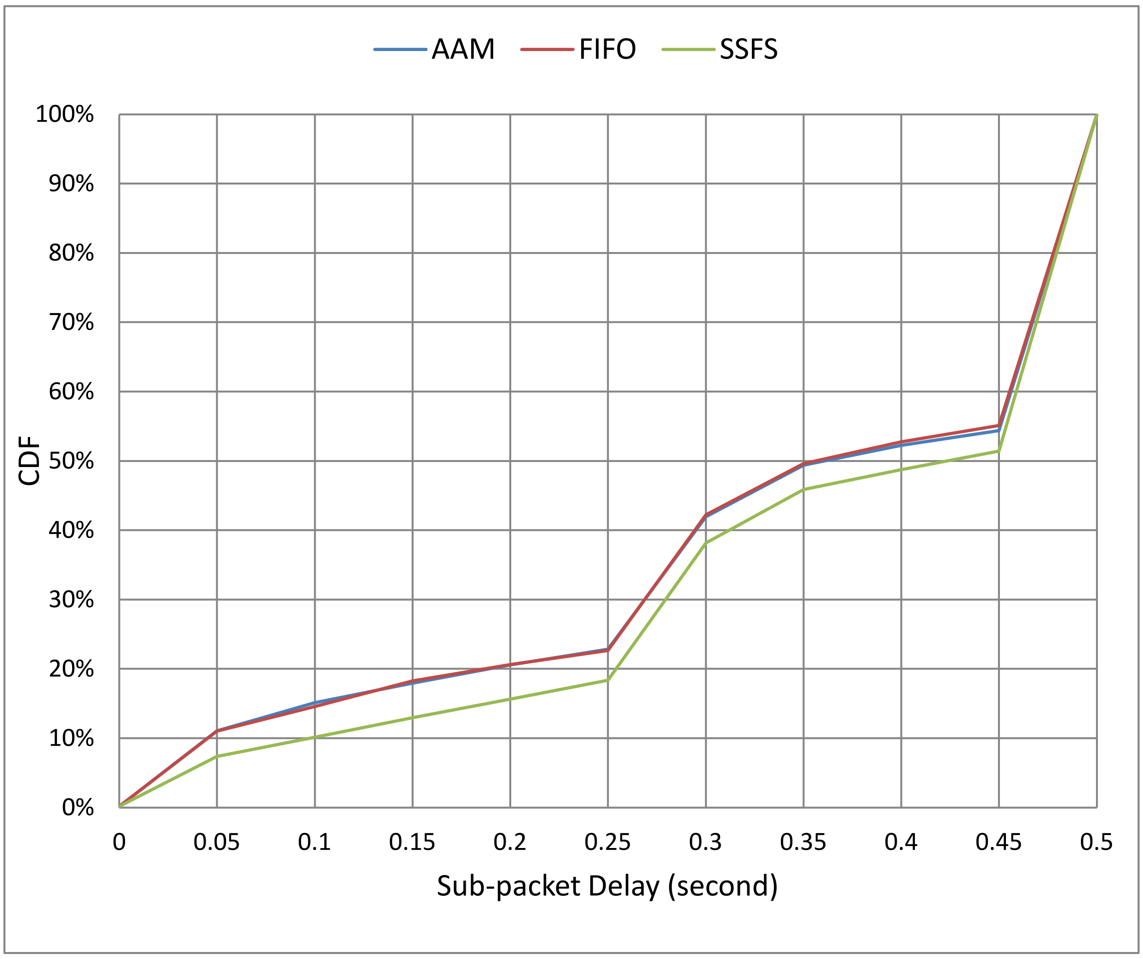

Figure C-13: The CDF of sub-packet delay for the captured traffic trace file 13.

The captured traffic trace file 14

Table C-14: The details for the captured traffic trace file 14

Time	Date	Location	PPS
10:30 –11:30	26th, June, 2012	Library, Kevin street, DIT, Dublin	19.2

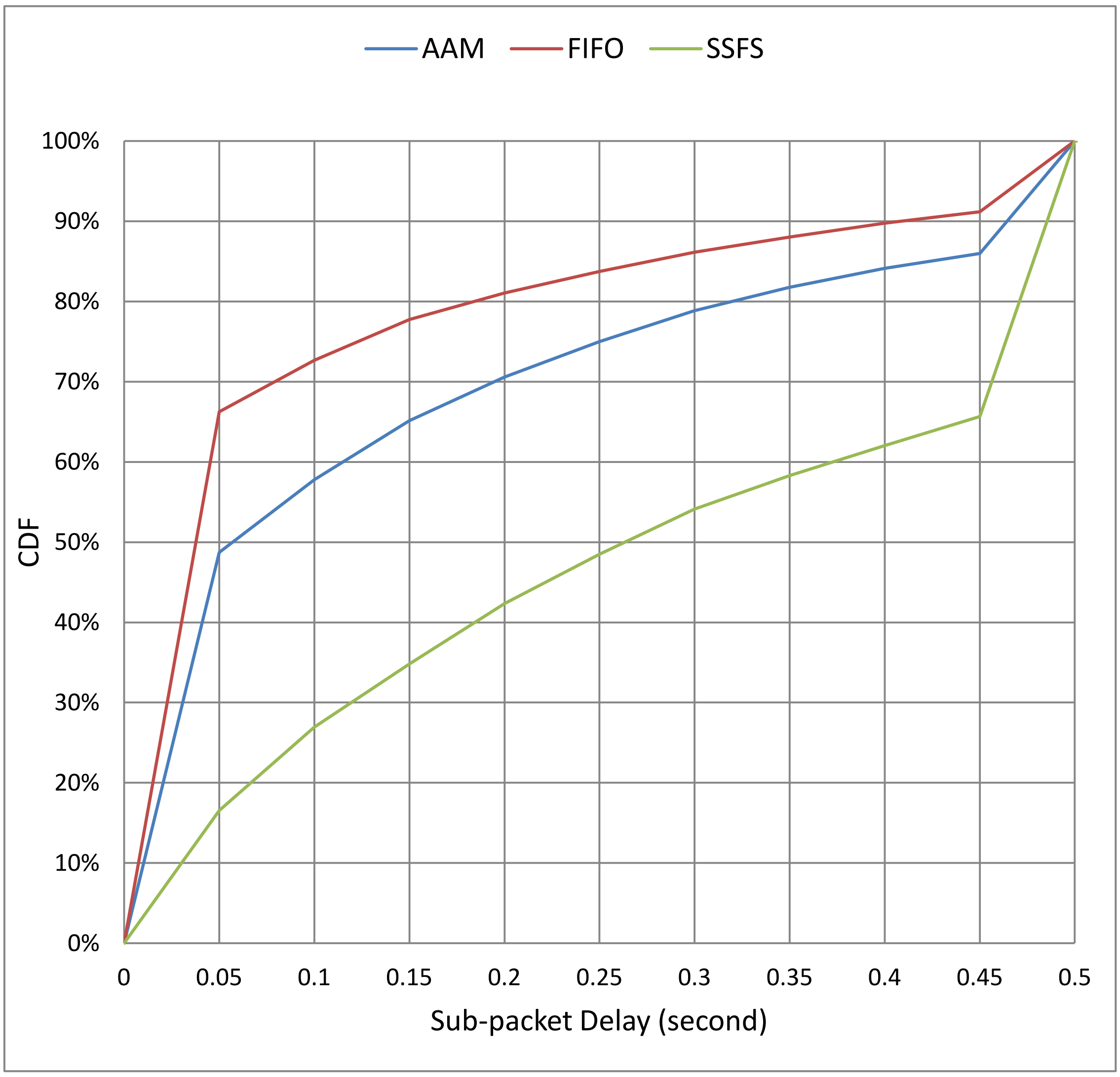

Figure C-14: The CDF of sub-packet delay for the captured traffic trace file 14.

The captured traffic trace file 15

Table C-15: The details for the captured traffic trace file 15

Time	Date	Location	PPS
12:00 –13:00	26th, June, 2012	Library, Kevin street, DIT, Dublin	3.8

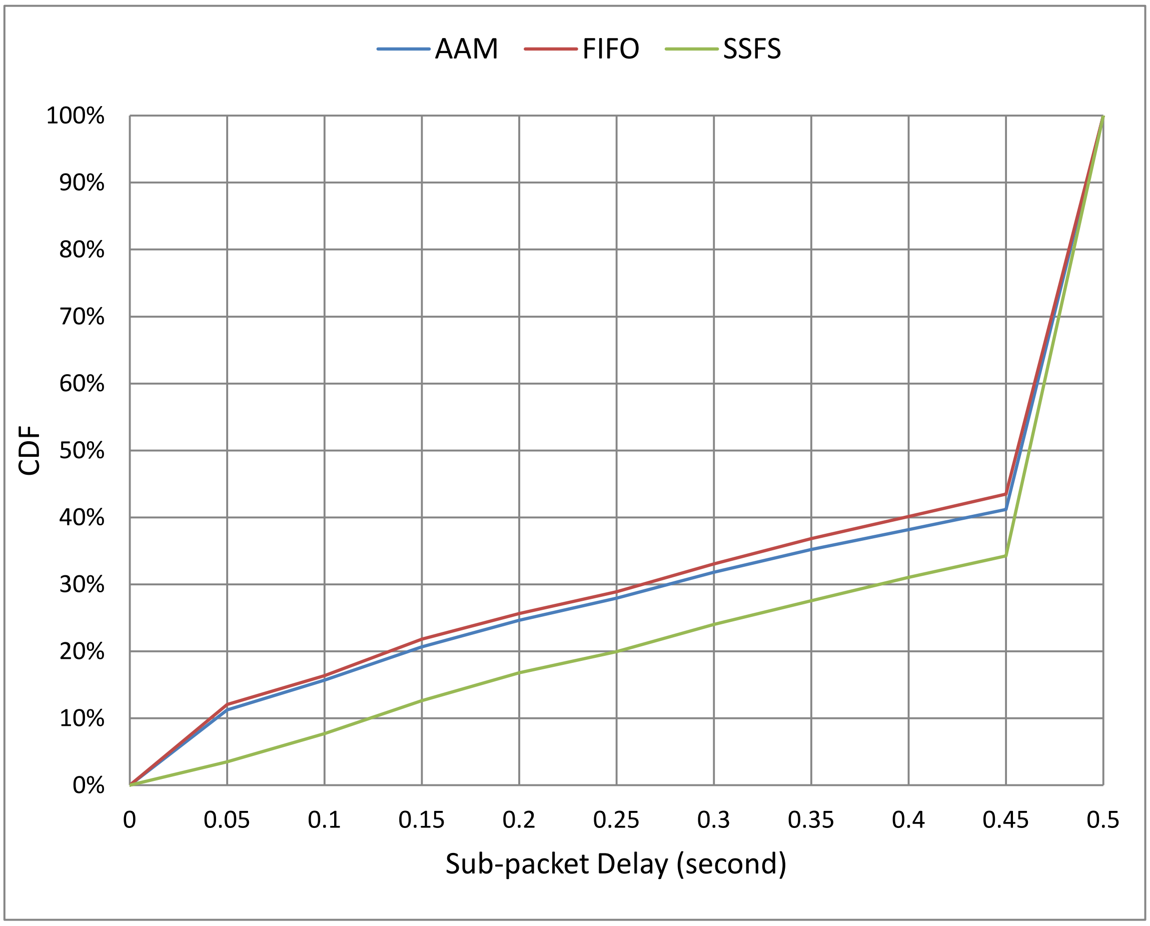

Figure C-15: The CDF of sub-packet delay for the captured traffic trace file 15.

The captured traffic trace file 16

Table C-16: The details for the captured traffic trace file 16

Time	Date	Location	PPS
19:00 –19:50	17th ,July, 2012	*Shuangliu* airport, Sichuan, China	6.9

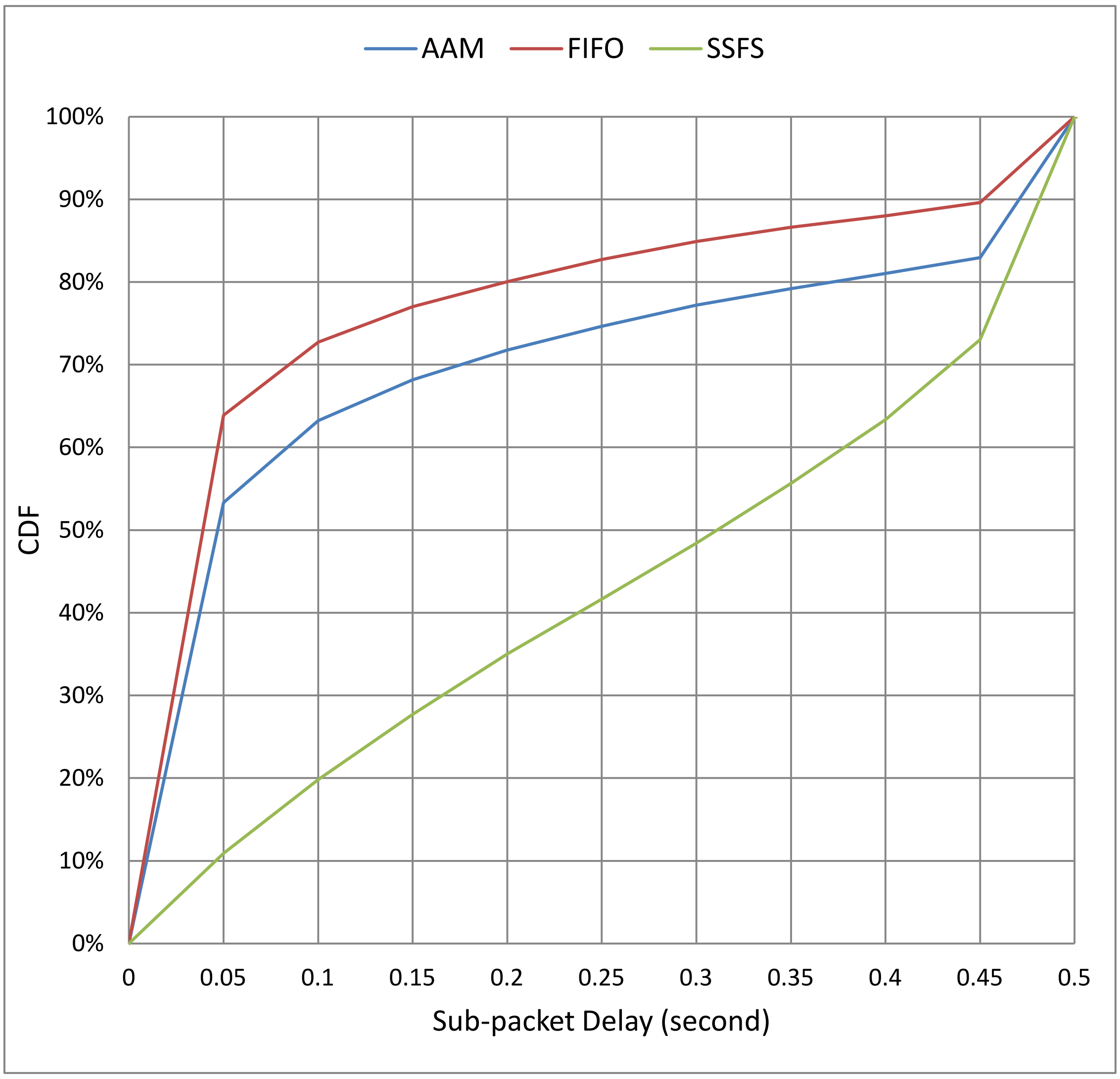

Figure C-16: The CDF of sub-packet delay for the captured traffic trace file 16.

Appendix D

The captured traffic trace file 1

Table D-1: The details for the captured traffic trace file 1

Time	Date	Location	PPS
10:30 – 11:30	29th ,May, 2012	*JAVA City,* DIT, Dublin	92.1

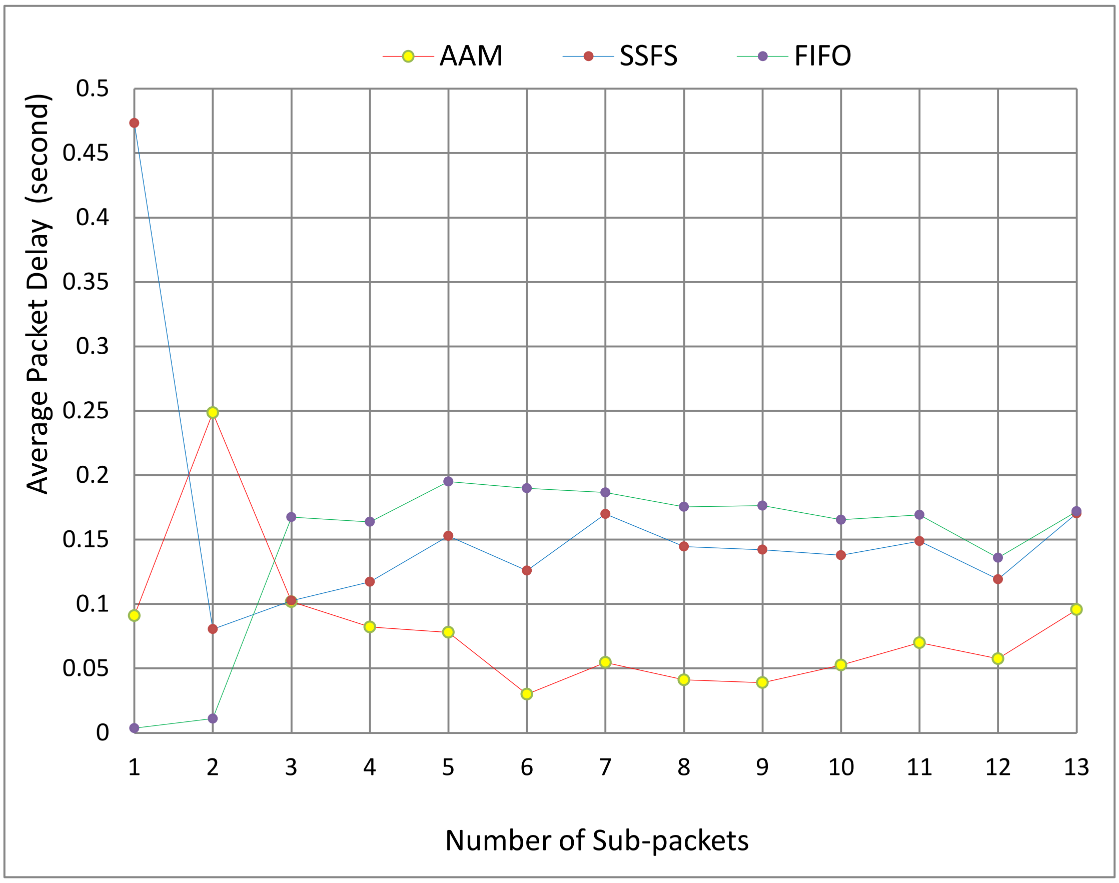

Figure D-1: The number of sub-packets against the average packet delay for the captured traffic trace file1.

The captured traffic trace file 2

Table D-2: The details for the captured traffic trace file 2

Time	Date	Location	PPS
12:00 – 13:00	29th ,May, 2012	*JAVA City,* DIT, Dublin	114

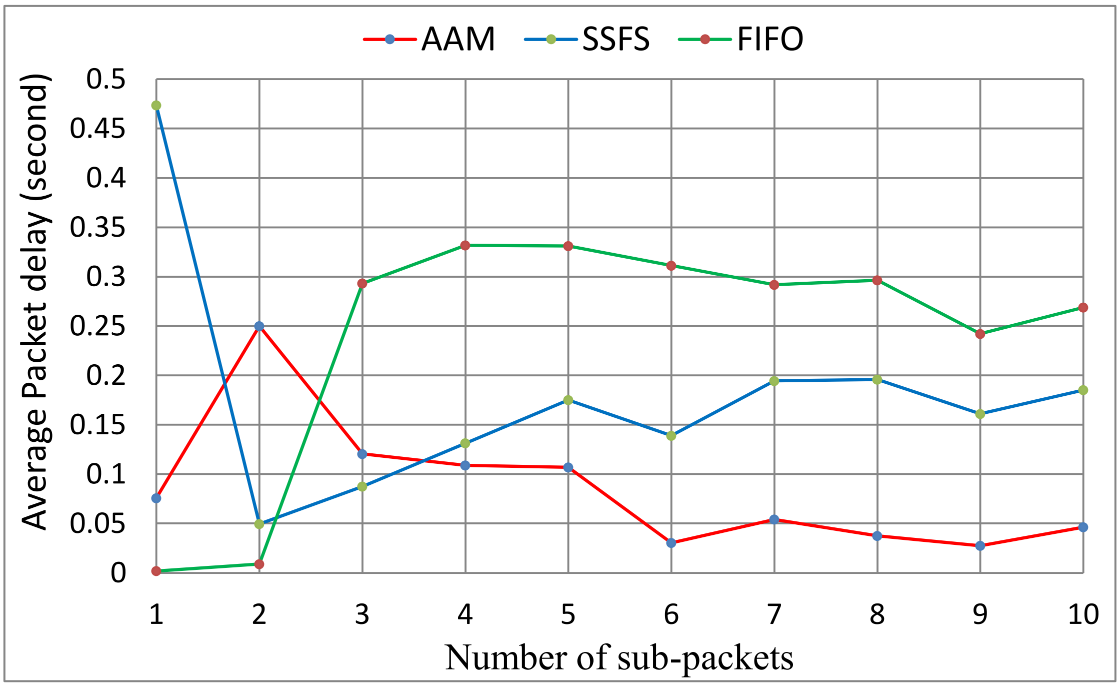

Figure D-2: The number of sub-packets against the average packet delay for the captured traffic trace file 2.

The captured traffic trace file 3

Table D-3: The details for the captured traffic trace file 3

Time	Date	Location	PPS
14:00 – 15:00	29th ,May, 2012	*JAVA City,* DIT, Dublin	75.7

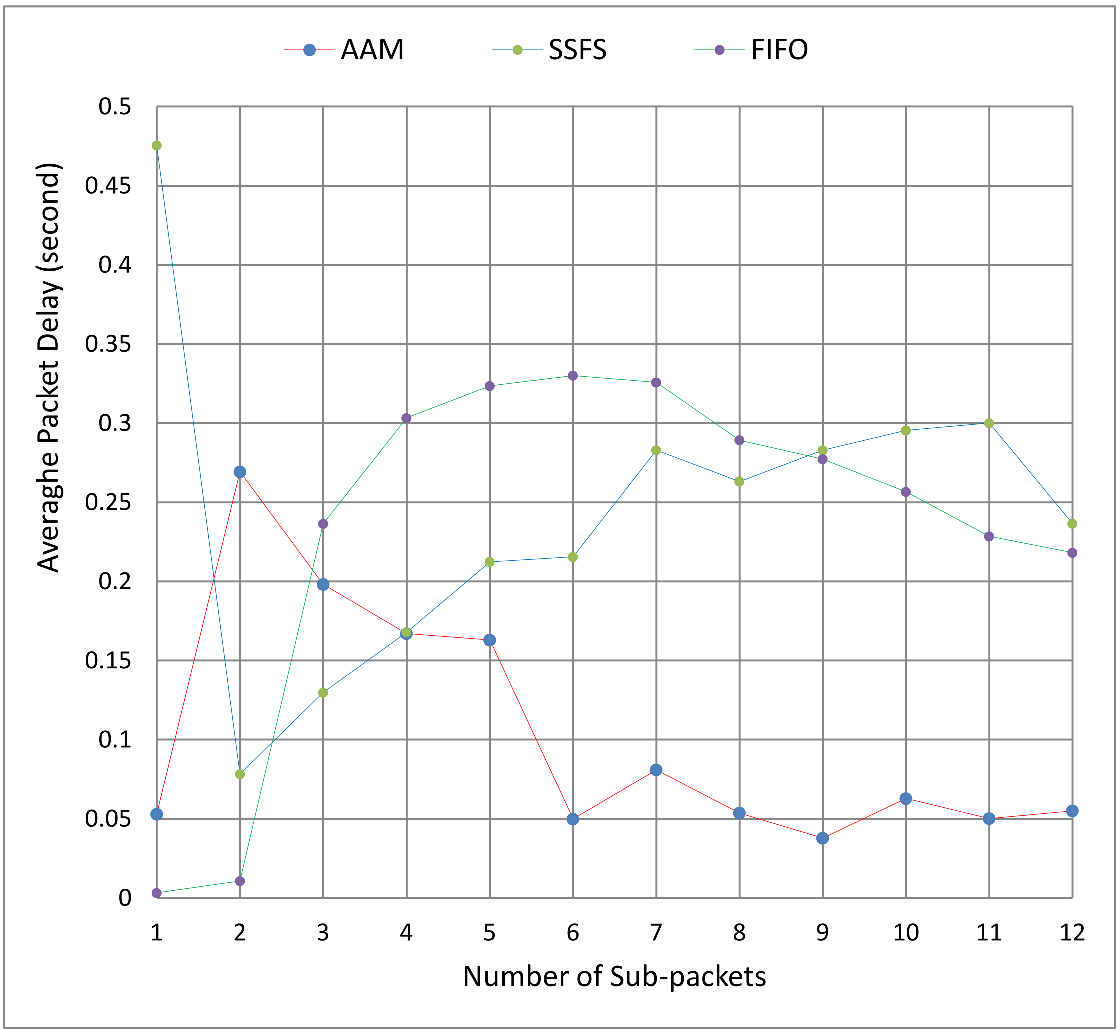

Figure D-3: The number of sub-packets against the average packet delay for the captured traffic trace file 3.

The captured traffic trace file 4

Table D-4: The details for the captured traffic trace file 4

Time	Date	Location	PPS
16:00 – 17:00	29th ,May, 2012	*JAVA City*, DIT, Dublin	111.3

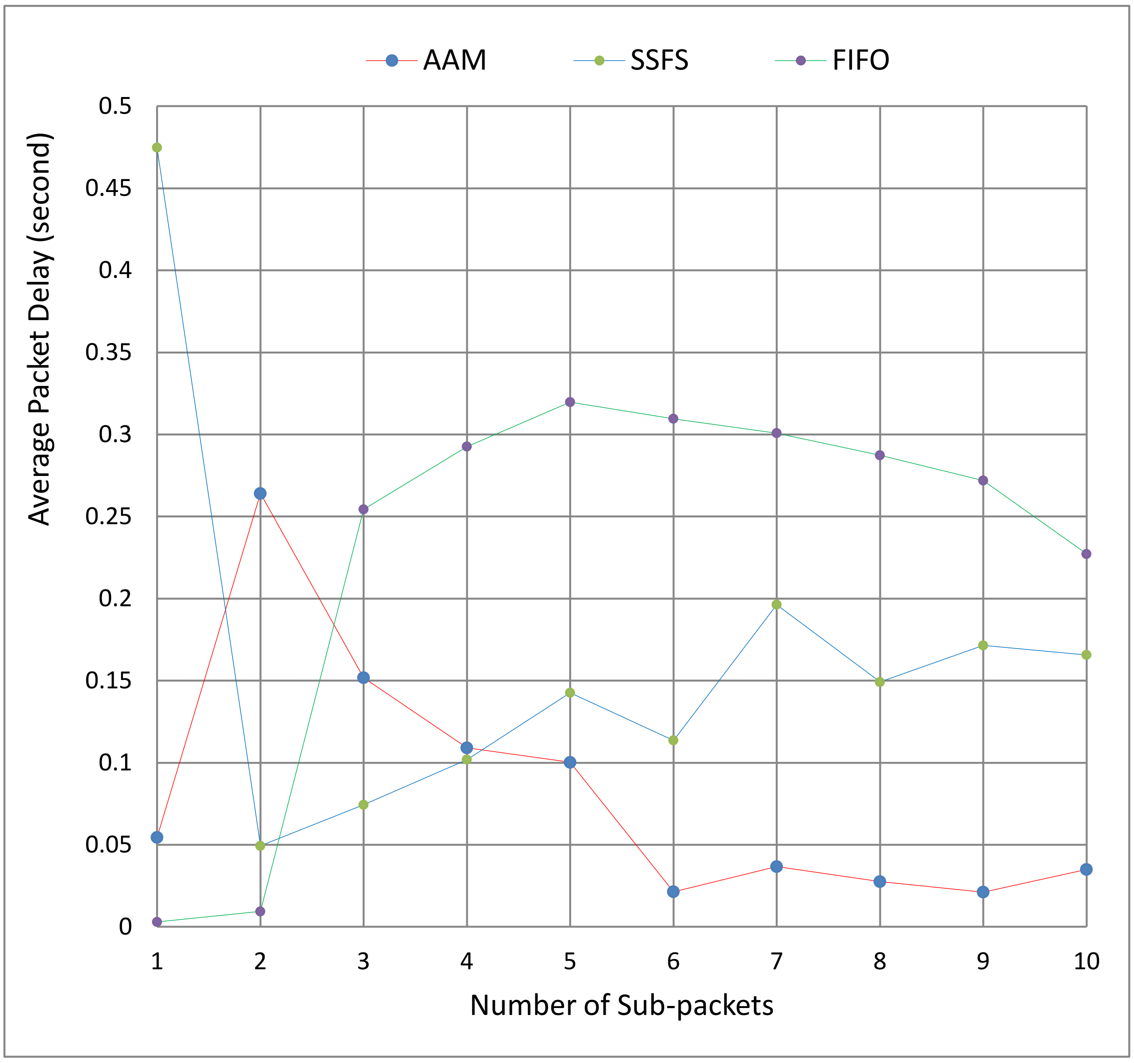

Figure D-4: The number of sub-packets against the average packet delay for the captured traffic trace file 4.

The captured traffic trace file 5

Table D-5: The details for the captured traffic trace file 5

Time	Date	Location	PPS
17:30 – 18:30	29th ,May, 2012	*JAVA City,* DIT, Dublin	109.4

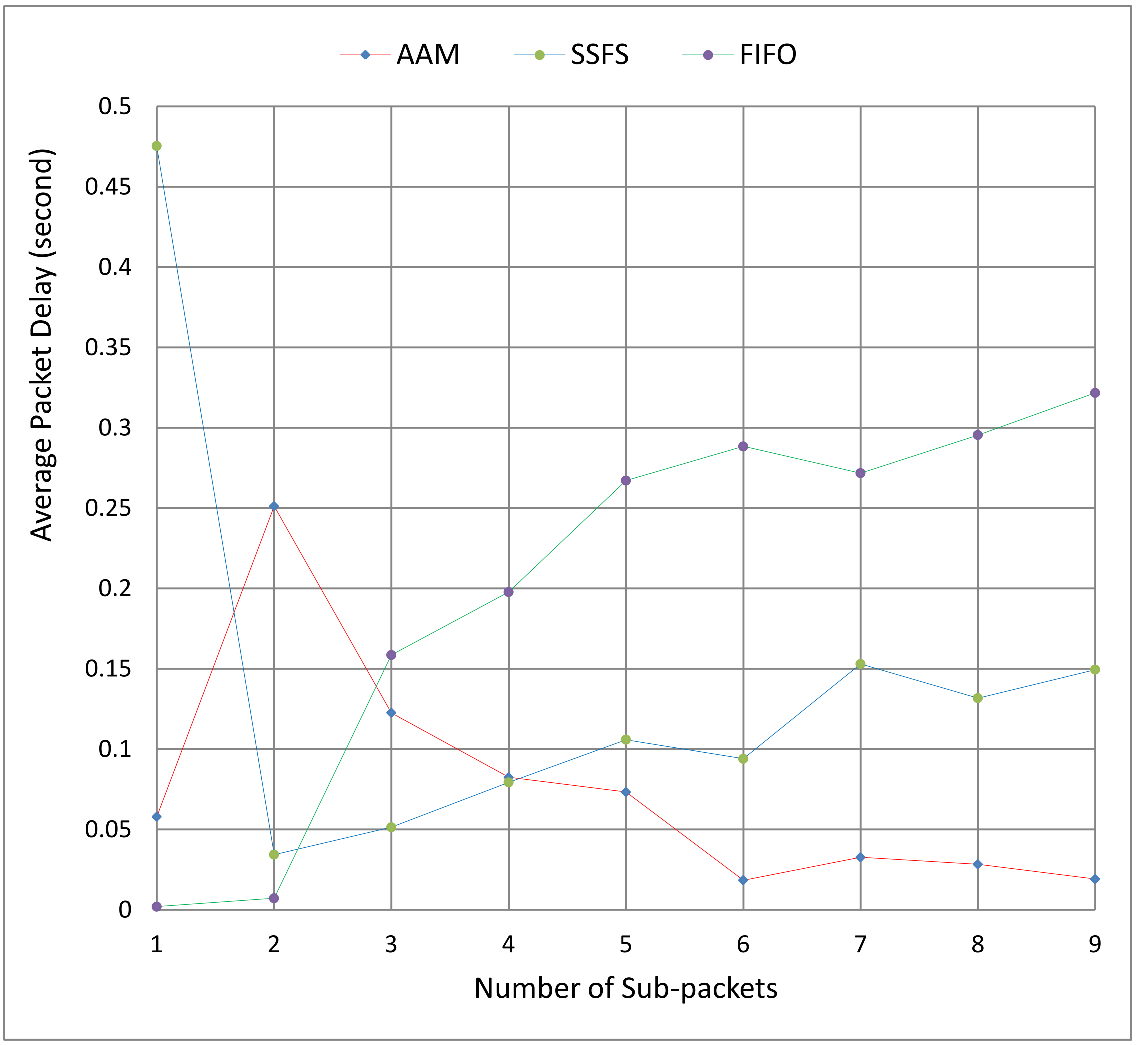

Figure D-5: The number of sub-packets against the average packet delay for the captured traffic trace file 5.

The captured traffic trace file 6

Table D-6: The details for the captured traffic trace file 6

Time	Date	Location	PPS
09:30 – 10:30	19th, June, 2012	*Costa coffee shop,* Dublin	9.4

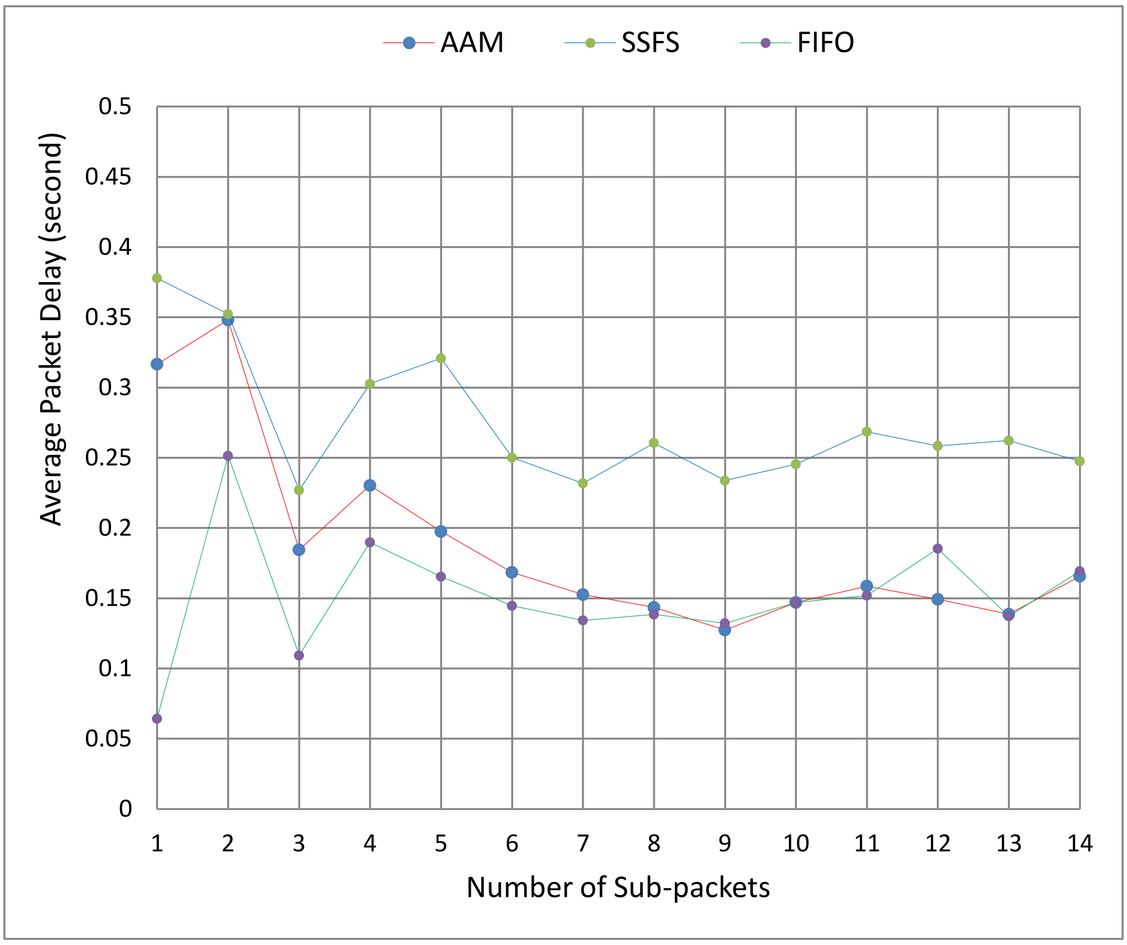

Figure D-6: The number of sub-packets against the average packet delay for the captured traffic trace file 6.

The captured traffic trace file 7

Table D-7: The details for the captured traffic trace file 7

Time	Date	Location	PPS
11:00 –12:00	19th, June, 2012	*Parliament Square,* TCD, Dublin	5.6

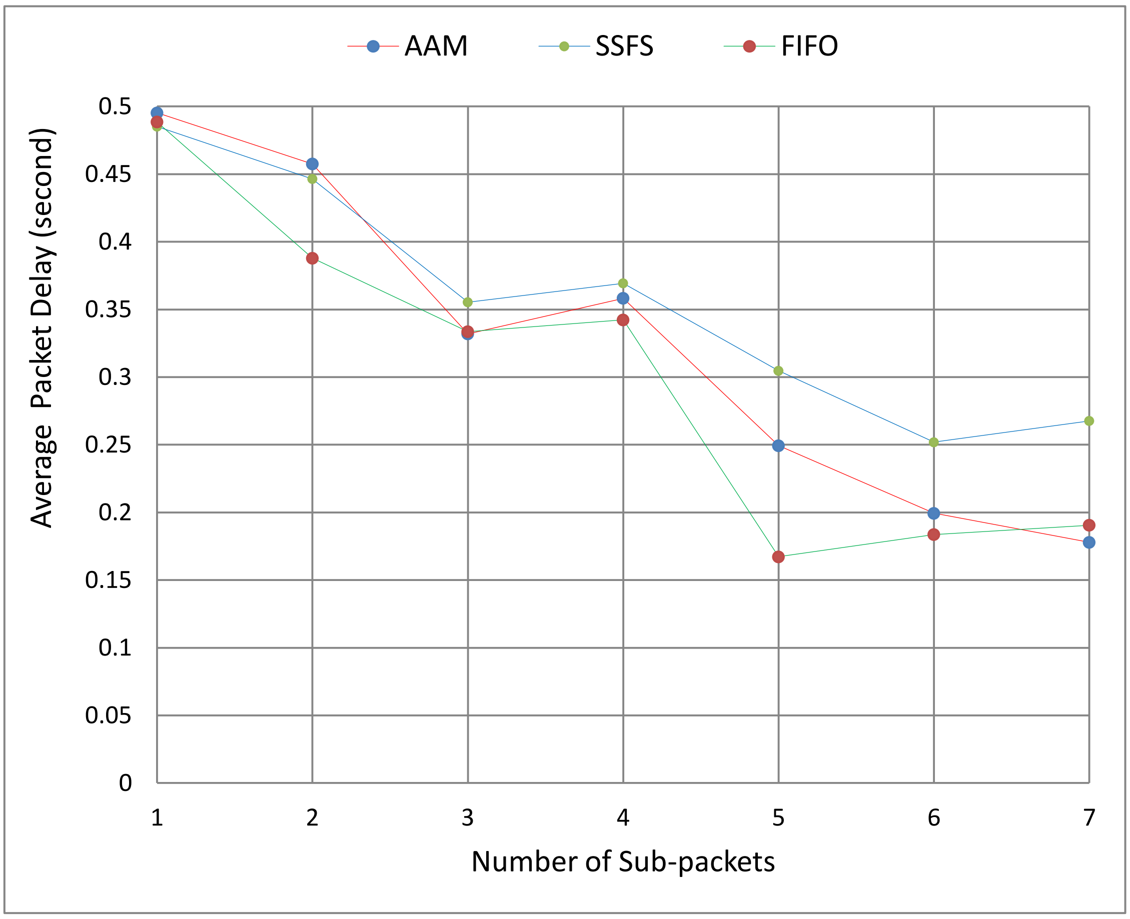

Figure D-7: The number of sub-packets against the average packet delay for the captured traffic trace file 7.

The captured traffic trace file 8

Table D-8: The details for the captured traffic trace file 8

Time	Date	Location	PPS
12:30 – 13:30	19th, June, 2012	*Costa coffee shop,* Dublin	9.2

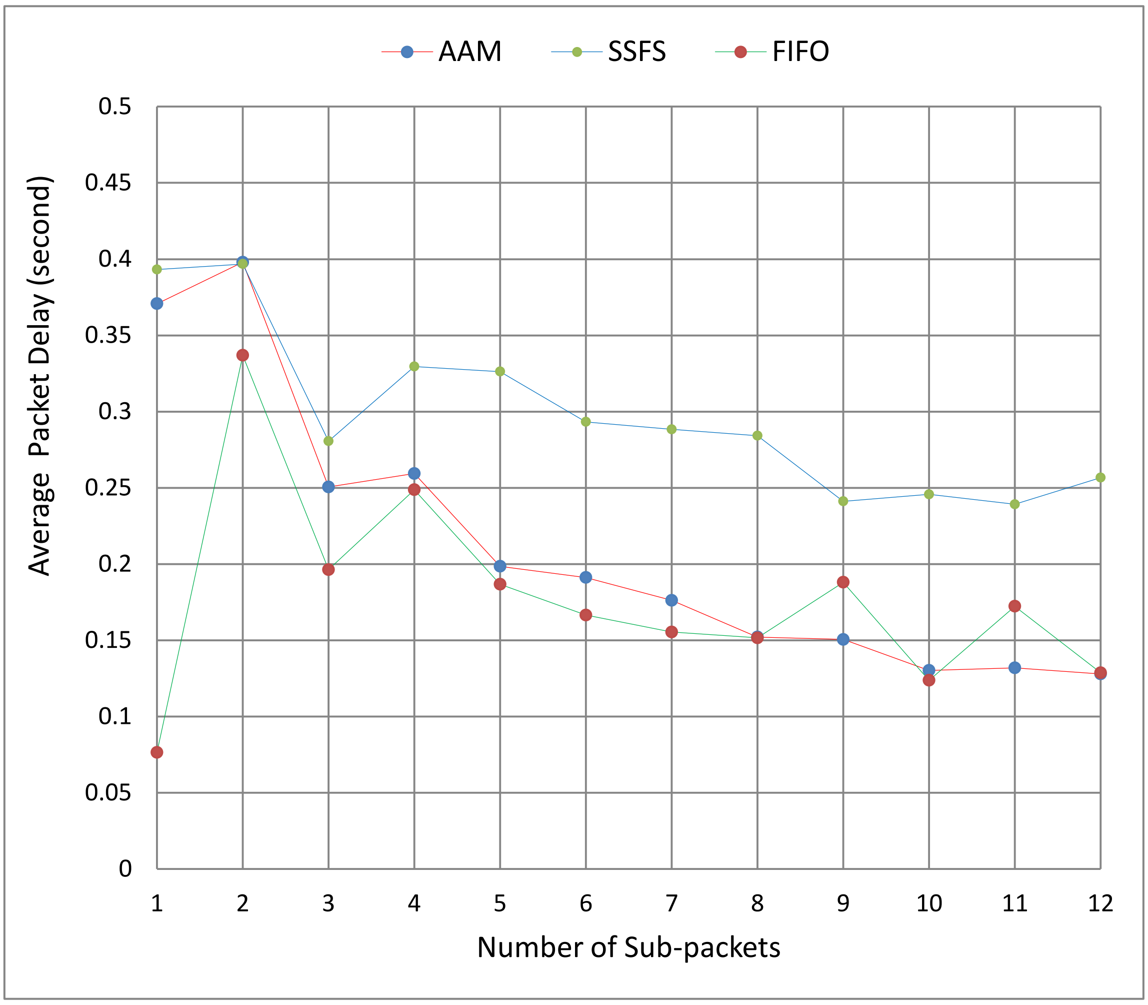

Figure D-8: The number of sub-packets against the average packet delay for the captured traffic trace file8.

The captured traffic trace file 9

Table D-9: The details for the captured traffic trace file 9

Time	Date	Location	PPS
16:00 –17:00	19th, June, 2012	*Parliament Square,* TCD, Dublin	3.6

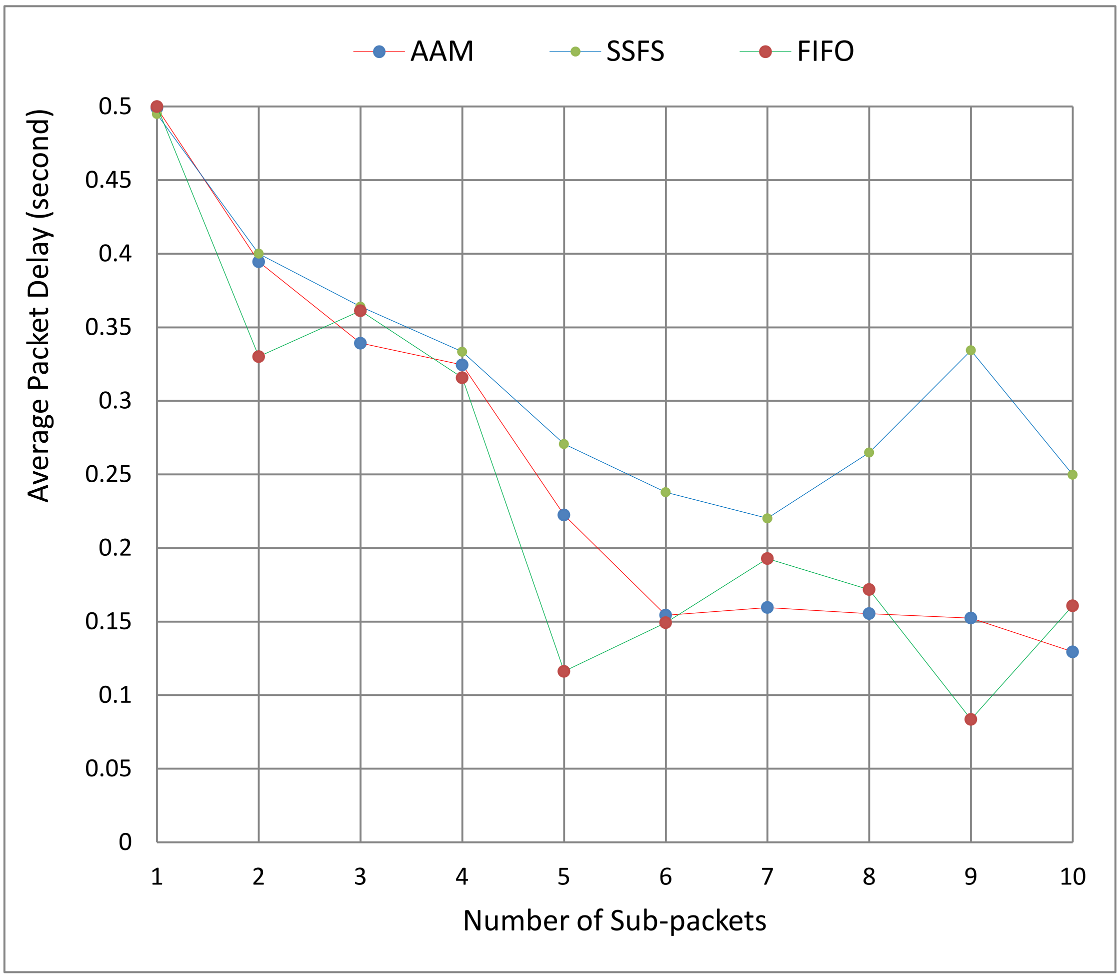

Figure D-9: The number of sub-packets against the average packet delay for the captured traffic trace file 9.

The captured traffic trace file 10

Table D-10: The details for the captured traffic trace file 10

Time	Date	Location	PPS
17:00 – 18:00	19th, June, 2012	*Costa coffee shop,* Dublin	6.6

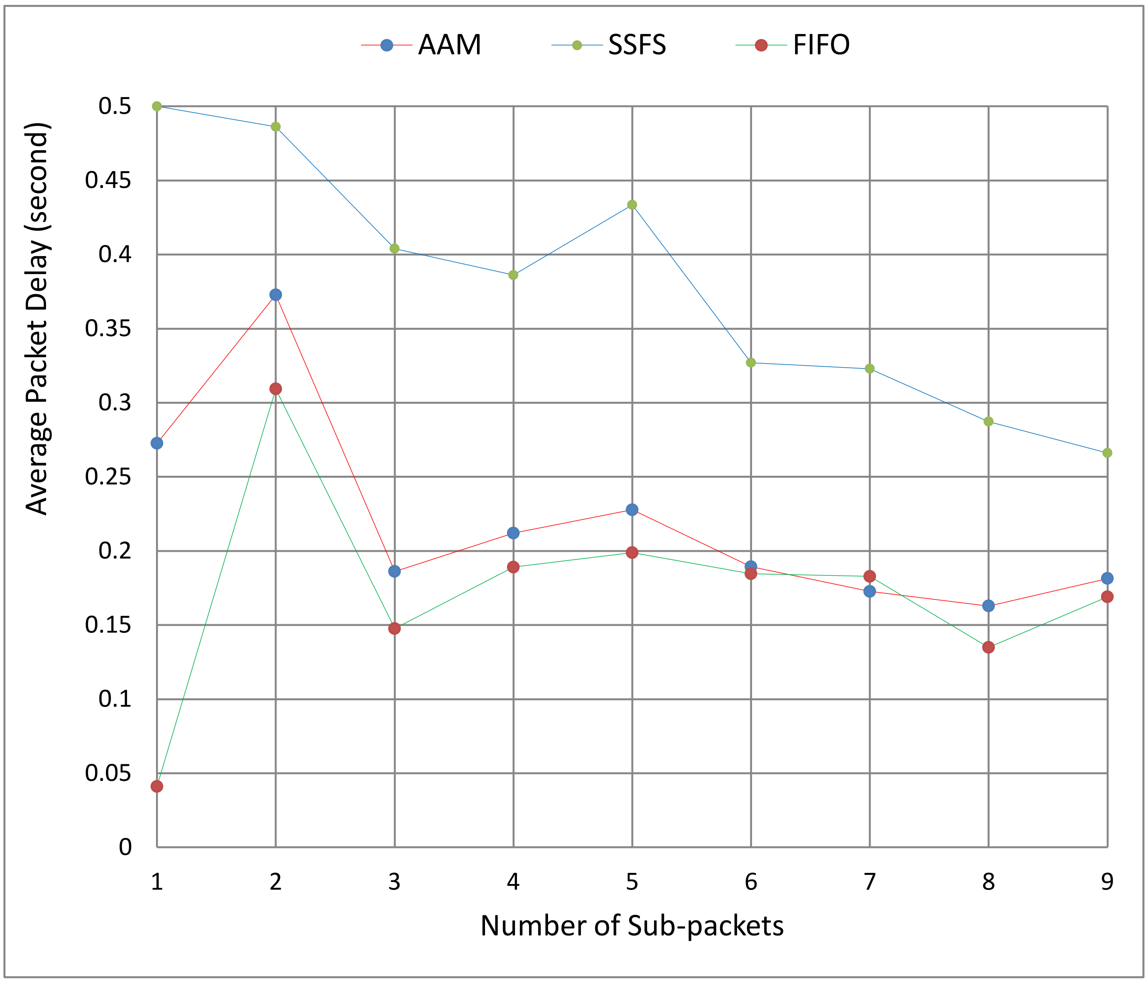

Figure D-10: The number of sub-packets against the average packet delay for the captured traffic trace file 10.

The captured traffic trace file 11

Table D-11: The details for the captured traffic trace file 11

Time	Date	Location	PPS
12:00 –13:00	24th, June, 2012	*Hueston* train station, Dublin	6.87

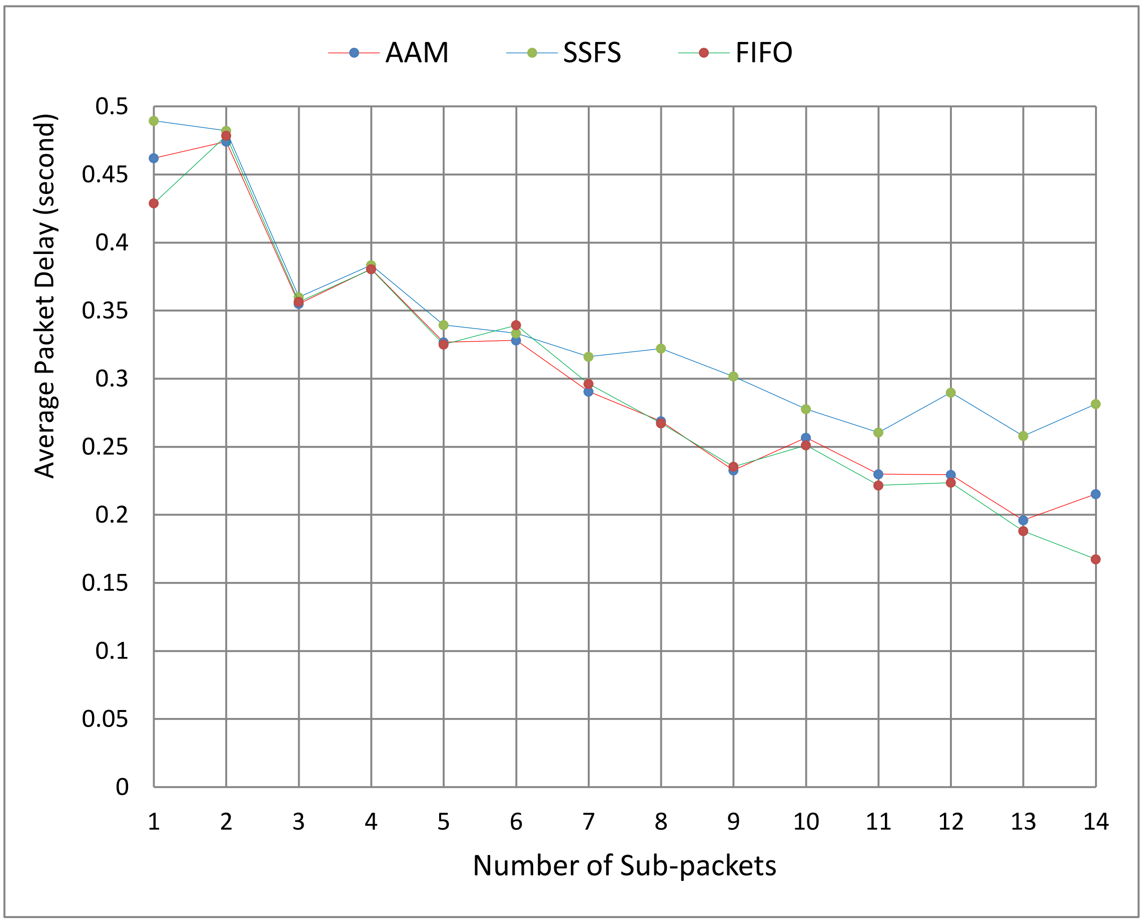

Figure D-11: The number of sub-packets against the average packet delay for the captured traffic trace file 11.

The captured traffic trace file 12

Table D-12: The details for the captured traffic trace file 12

Time	Date	Location	PPS
13:30 – 14:30	24th, June, 2012	*Hueston* train station, Dublin	4.2

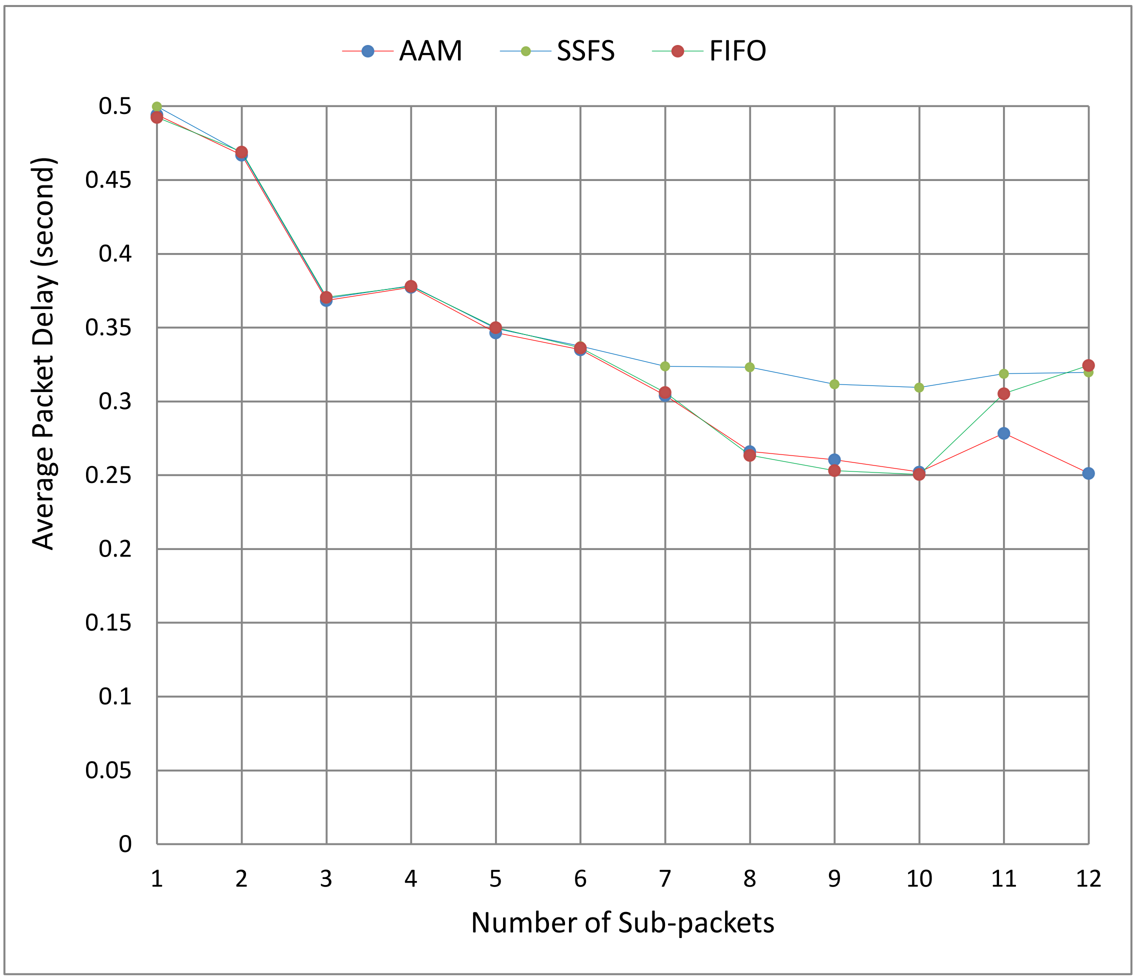

Figure D-12: The number of sub-packets against the average packet delay for the captured traffic trace file 12.

The captured traffic trace file 13

Table D-13: The details for the captured traffic trace file 13

Time	Date	Location	PPS
15:00 –16:00	24th, June, 2012	*Hueston* train station, Dublin	6.2

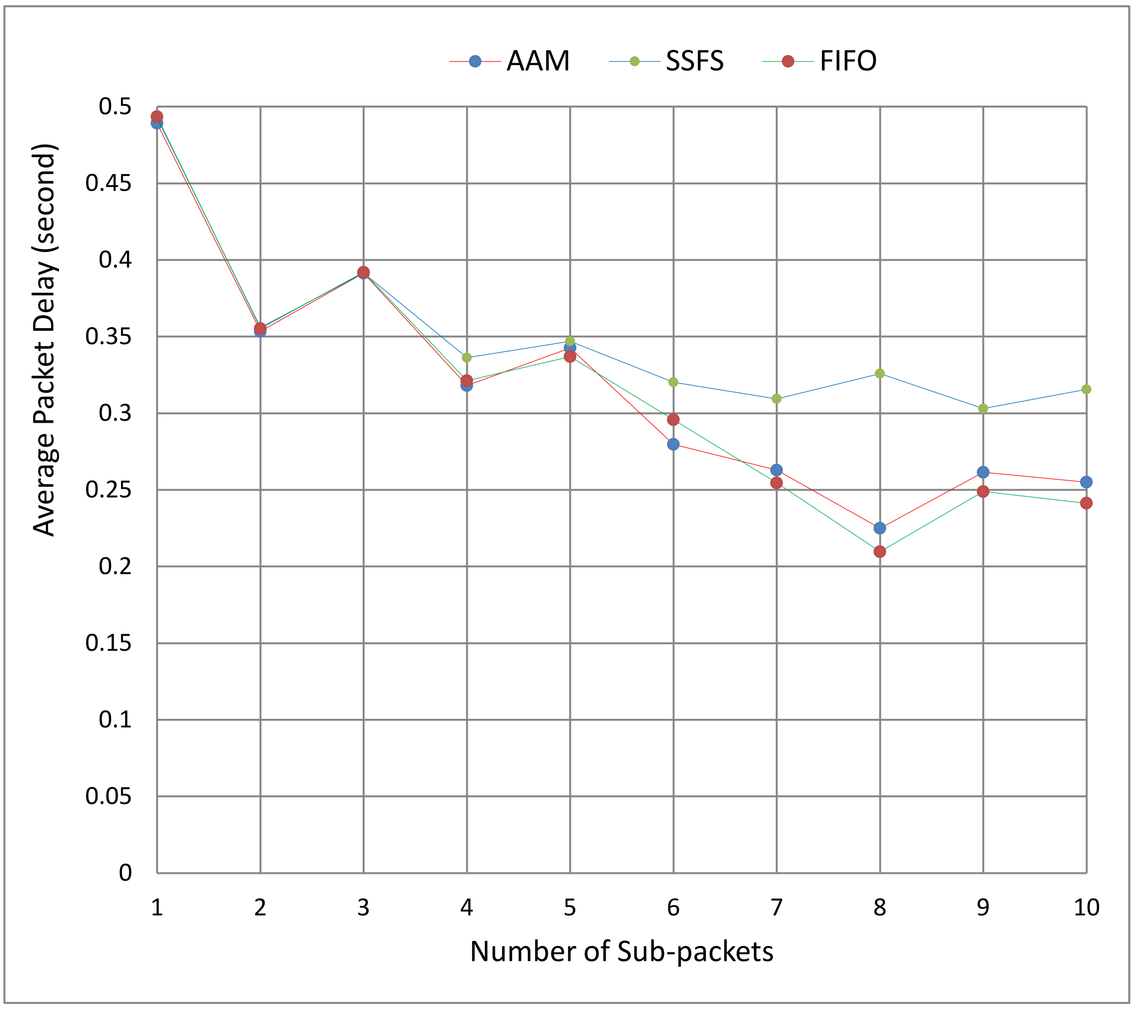

Figure D-13: The number of sub-packets against the average packet delay for the captured traffic trace file 13.

The captured traffic trace file 14

Table D-14: The details for the captured traffic trace file 14

Time	Date	Location	PPS
10:30 –11:30	26th, June, 2012	Library, Kevin street, DIT, Dublin	19.2

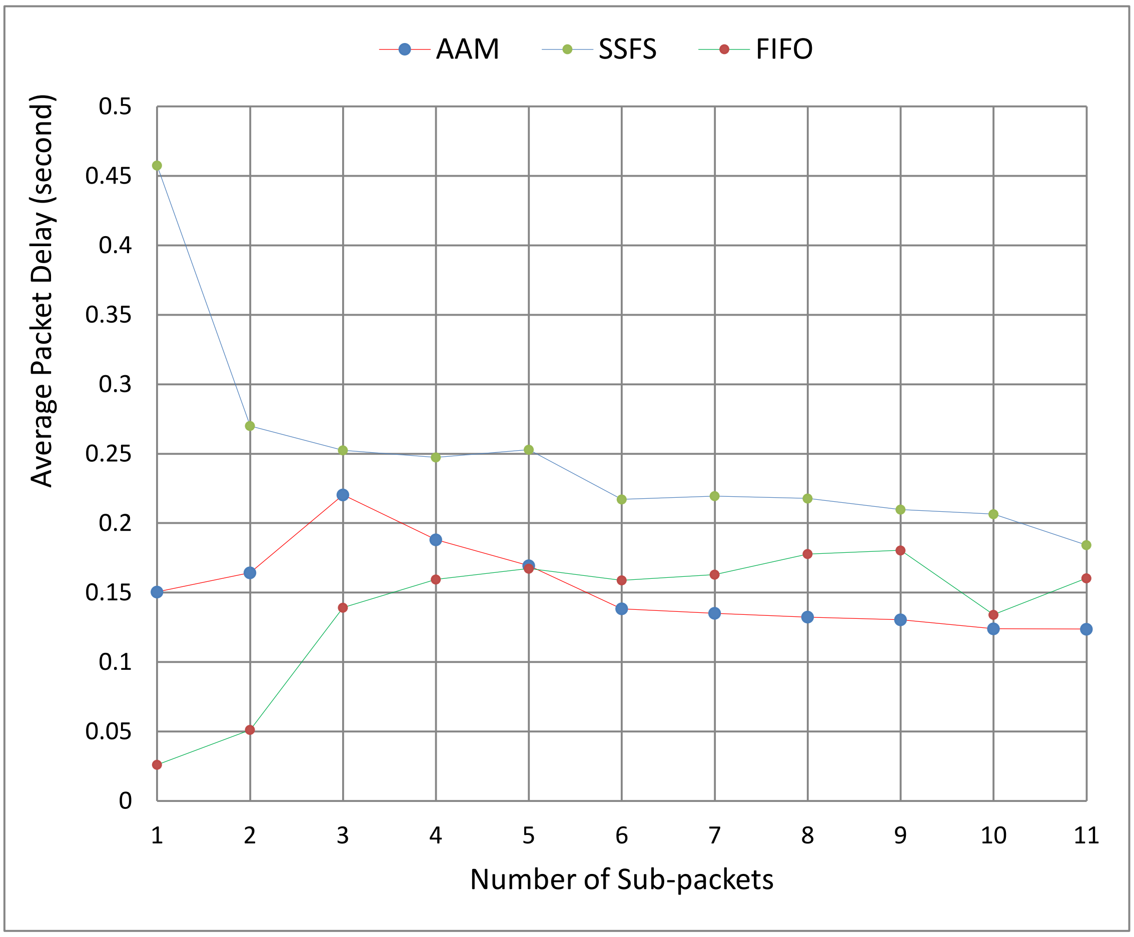

Figure D-14: The number of sub-packets against the average packet delay for the captured traffic trace file 14.

The captured traffic trace file 15

Table D-15: The details for the captured traffic trace file 15

Time	Date	Location	PPS
12:00 –13:00	26th, June, 2012	Library, Kevin street, DIT, Dublin	3.8

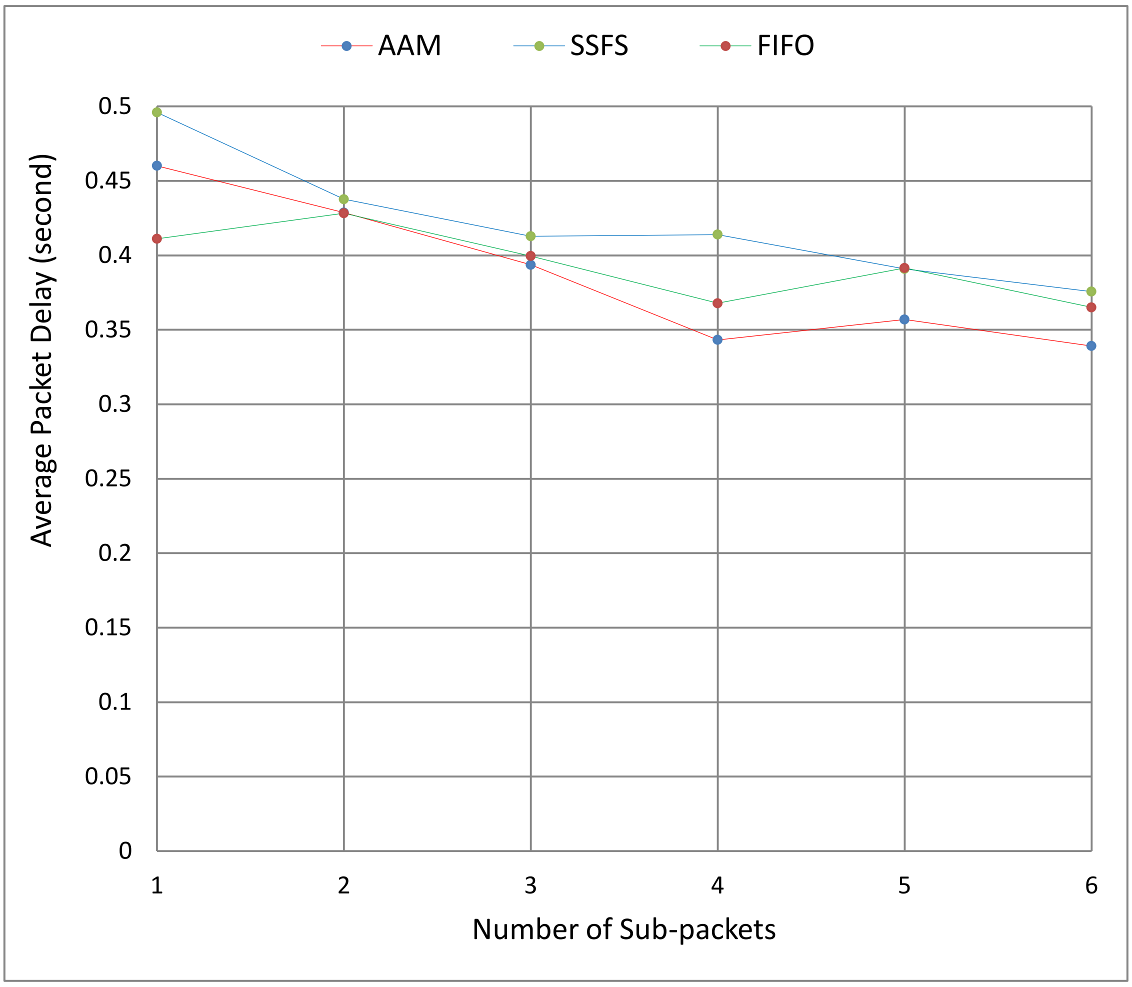

Figure D-15: The number of sub-packets against the average packet delay for the captured traffic trace file 15.

The captured traffic trace file 16

Table D-16: The details for the captured traffic trace file 16

Time	Date	Location	PPS
19:00 –19:50	17th ,July, 2012	*Shuangliu* airport, Sichuan, China	6.9

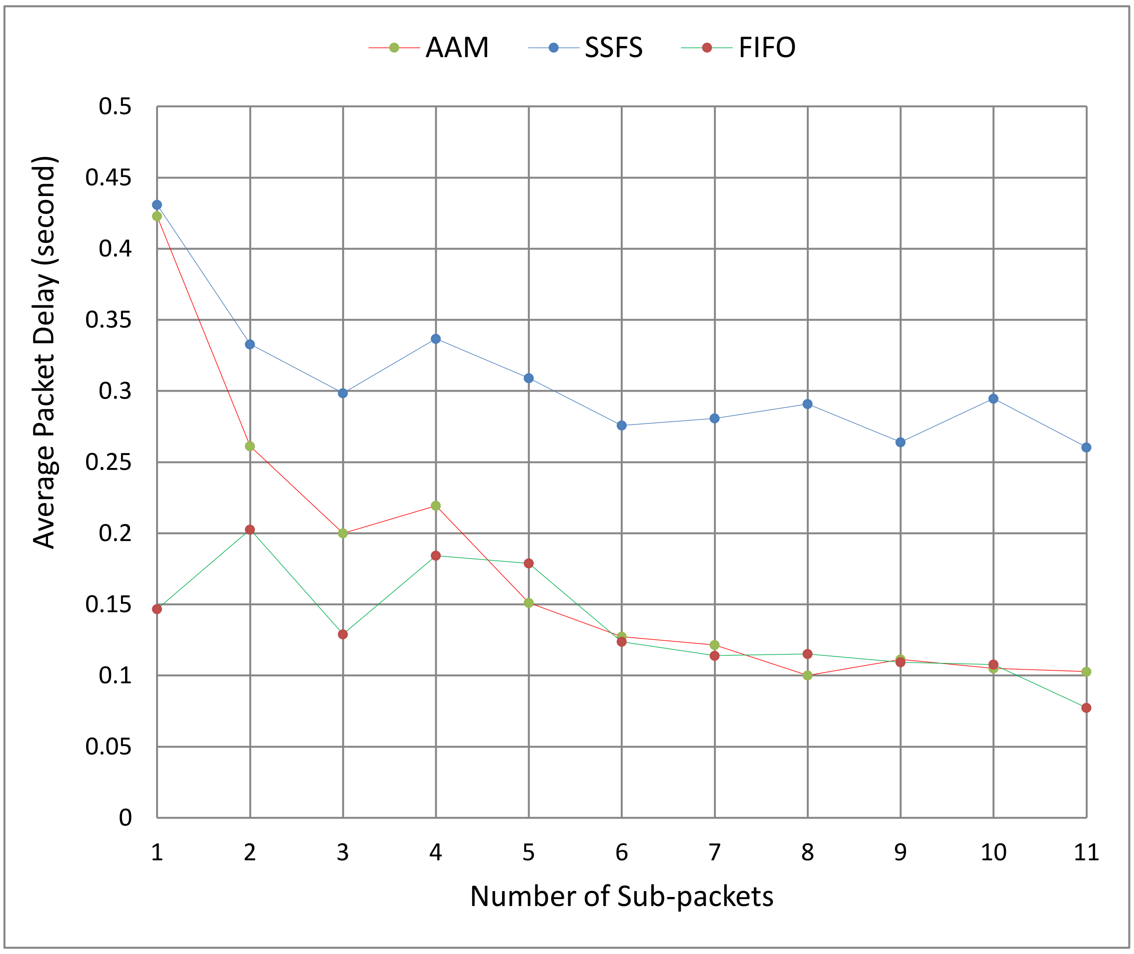

Figure D-16: The number of sub-packets against the average packet delay for the captured traffic trace file 16.