FACULTY OF SCIENCE AND TECHNOLOGY

PERFORMANCE ANALYSIS OF A DEVELOPED ADMISSION CONTROL MODEL IN MOBILE AD-HOC NETWORK (MANET)

FOLAYO AINA

A thesis in partial fulfilment of the requirements of Anglia Ruskin University for the degree of Doctor of Philosophy

Submitted: June 2020

Acknowledgements

First and foremost, I would like to express my profound gratitude to the Almighty God, who has kept and provided for me, throughout the duration of my PhD studies.

I would also like to thank my supervisors, Dr. Sufian Yousef and Dr. Erika Sanchez-Velazquez for their immense contributions, support, guidance and advices. Your inputs and constructive criticisms during the research process, the writing and the submissions are highly appreciated; and this is hereby acknowledged.

To my loving and ever supportive husband, Mr. Mayowa Aina, I am so grateful to you for all the encouragement you gave me when I needed it the most. You have supported me all through the thick and thin of my research journey. Your financial support and words of encouragement has brought me thus far. To my wonderful and precious parents, (Prof. and Dr (Mrs) P.A. Osanaiye), I thank you from the bottom of my heart for the continuous faith you have shown in me, and also for your immense contributions in making me who I am today. Thank you for the financial support you provided even through all difficulties, may the almighty God reward you for all the labour of love you have shown me. Thank you (Engr. (Dr). & Mrs. Aina), your support both financially and morally, is highly appreciated. To my son and daughter, Pires and Stephanie, thanks so much for the support and understanding you demonstrated throughout my research work, your co-operation is highly appreciated. To my siblings and entire family, thank you for all your prayers and words of encouragement all through my PhD journey. Your kind words have kept me going; and I sincerely appreciate it. Thanks Dr. Opeyemi Osanaiye for all your support, most especially assisting with proof reading my Thesis, I owe you a lot brother.

To Mr. Gboyega Adeloye and Late (Mrs) Ibukun Adeloye, my sincere appreciation goes to you, thank you so much for your financial support towards the success of my PhD, your kind heart towards me is highly appreciated.

To all the members of my research groups, thank you all for your immense contributions towards the improvement of the quality of my research output.

To all my friends in Anglia Ruskin University Chelmsford, Colchester, London and all over the world, who have contributed towards my PhD in one way or another, I owe you all a great deal. Thank you all for your support; and God bless you!

ANGLIA RUSKIN UNIVERSITY

ABSTRACT

FACULTY OF SCIENCE AND TECHNOLOGY

DOCTOR OF PHILOSOPHY

PERFORMANCE ANALYSIS OF A DEVELOPED ADMISSION CONTROL MODEL IN MOBILE AD-HOC NETWORK (MANET)

FOLAYO AINA

JUNE 2020

As wireless mobile network becomes widespread, the demand for user applications is higher and the services provided by the wired application is expected to be available in the wireless medium. Therefore, the users of these applications will expect the same Quality of Service (QoS) as obtained in wired network. Providing a reliable QoS in wireless medium, especially MANET, is quite challenging and remains an ongoing research trend. The key issue of MANET is its inability to accurately predict the needed resources, to avoid interference with an ongoing traffic flow. An essential solution to the issues posed by MANET is the introduction of an admission control component for a guaranteed QoS. Admission control helps to control the usage of resources when an additional service is requested. For an admission decision to be made for a new flow, the expected bandwidth consumption must be correctly predicted prior to admission not-withstanding the fact that wireless medium is shared, and nodes contends among themselves to access the medium. Some recent solutions considered the MAC layer back-off impact due to collision as well as the non-synchronization of the sender and receiver during the available bandwidth estimation process.

This thesis therefore has three (3) objectives. Firstly, it investigates the various techniques for estimating bandwidth in MANET that gives better accuracy. Secondly, this thesis develops an efficient bandwidth estimation and admission control mechanism that limits the frequent bandwidth usage for achieving lesser network overhead during the retrieval of neighbouring bandwidth. Finally, this thesis investigates the key metrics necessary to be considered to ensure a guaranteed control of admission within the network.

The novelty of this thesis is the proposed resource allocation and admission control in MANET (RAACM) solution which is an admission control scheme that estimates the available bandwidth needed within a network using a robust and accurate resource estimation technique. Furthermore, the various factors that must be considered for an effective estimation were highlighted and simulations were carried out using Riverbed 17.5.

Results obtained from the simulation, studies and compared the throughput impact of a network that has no admission control implementation with our proposed RAACM protocol. RAACM throughput result shows accuracy in its result with less data drop. The throughput result obtained from RAACM was thereafter compared with other state-of-the art admission control protocol. RAACM's admission control throughput analysis performance with respect to the HCF is high when compared with closely related research work. RAACM's delay and data dropped analysis was also studied and comparison was made with the state-of-the art admission control protocols, RAACM however, provides efficiency and accuracy in its result.

In conclusion, by giving a close attention to the channel idle time estimation between a sender and a receiver for available bandwidth estimation through analysing it with respect to overlapping and non-overlapping period as well as dependant node distribution, network accuracy and efficiency have been realised in this thesis.

Keywords: Admission Control, MANET, QoS, IEEE802.11e, Bandwidth Estimation, Resource Allocation.

Table of Contents

List of Figures

List of Tables

Notations

AABE	Adaptive Available Bandwidth Estimation
AABWM	Analytical Available Bandwidth Estimation Including Mobility
AAC	Adaptive Admission Control
ABCT	Ad-Hoc Network with Concurrent Transmissions
ABE	Available Bandwidth Estimation
AC	Access Categories
ACK	Acknowledgement
ADDTS	Add Traffic Scheme
AIFS	Arbitration Inter-frame Spaces
AIFSN	Arbitration Interframe Space Number
AODV	Ad hoc On-Demand Distance Vector
AP	Access Point
API	Application Program Interface
ARP	Address Resolution Protocol
AS	Autonomous System
BAN	Body Area Network
BandEst	Bandwidth Estimation
BART	Bandwidth Available in Real-Time
BE	Best Effort
BECIT	Cognitive Agent Based Available Bandwidth Estimation using Collision Probability, Idle Period, Synchronization and Random Waiting Time
BK	Background
BS	Base Station

BWER	Bandwidth Efficiency Ratio
CACP	Contention-Aware Admission Control Protocol
CBR	Constant Bit Rate
CCA	Clear Channel Assessment
CDMA	Code Division Multiple Access
CFB	Contention Free Bursting
CITR	Channel Idle Time Ratio cPEAB
CS	Carrier Sense
CSMA/CA	Carrier Sensing Multiple Access with Collision Avoidance
CTS	Clear to Send
CW	Contention Window
DABE	Distributed Available Bandwidth Estimation
DACME	Distributed Admission Control for MANET Environments
DARPA	Defence Advanced Research Projects Agency
DCF	Distributed Coordinating Function
DES	Discrete Event Simulator
DIFS	DCF Interframe Space
DiffServ	Differentiated Services
DLI-ABE	Distributed Lagrange Interpolation Based Available Bandwidth Estimation
DSDV	Destination Sequence Distance Vector
DSR	Dynamic Source Routing
DSSS	Direct Sequence Spectrum
EABRRL	Estimation of Available Bandwidth Ratio of a Remote Link
EDCA	Enhanced Distributed Coordination Access

EDCF	Enhanced Distributed Channel Function
ertPS	Extended Real-Time Polling Service
FAR	False Admission Ratio
FHSS	Frequency Hopping Spread Spectrum
FSM	Finite State Machine
FIFO	First-In-First-Out
FRR	False Rejection Ratio
FSR	Fisheye State Routing
GNAPP	Gaps of Non-Adjacent Probe Packet
GPT	Generic Passive Technique
GUI	Graphic User Interface
HCCA	Hybrid Coordination Function Controlled Channel Access
HCF	Hybrid Coordination Function
HTTP	Hypertext Transfer Protocol
IAB	Improved Available Bandwidth
ICMP	Internet Control Message Protocol
IDE	Integrated Development Environment
IETF	Internet Engineering Task Force
IGI	Initial Gap Increase
IntServ	Integrated Services
IPV6	Internet Protocol Version 6
IR	Infrared
ITU	International Telecommunication Union
KF	Kalman Filters

LAN	Local Area Network
MAC	Media Access Control
MANET	Mobile Ad-hoc Network
MBAC	Measured Based Admission Control
MiBT	Minimal Backlogging Techniques
MIMO	Multiple Input Multiple Output
MPLS	Multiprotocol Label Switching
MPDU	MAC Protocol Data Unit
MR-BART	Multi-Rate Available BE In Real-Time
NEXT	New Enhanced Available Bandwidth Measurement Technique
NEXT-V2	New Enhanced Available Bandwidth Measurement Technique Extension
NICMP	Node Internet Control Message Protocol
NLCPPD	Network Link Characteristics Using Packet Pair Dispersion
nrtPS	Non-Real Time Polling Service
OFDM	Orthogonal Frequency-Division Multiplexing
OFDMA	Orthogonal Frequency-Division Multiple Access
OLSR	Optimized Link State Routing Protocol
OSI	Open Systems Interconnection Model
OSPF	Open Shortest Path First
OSPFv3	Open Shortest Path First Version 3
PAB	Probabilistic Available Bandwidth
PABE	Proactive Bandwidth Estimation
PAN	Personal Area Network
PBAC	Parameter-Based Admission Control

PCF	Point Coordinating Function
PDA	Personal Digital Assistants
PDF	Probability Distribution Function
PDM	Packet Dispersion Measurement
PDU	Protocol Data Unit
PGM	Probe Gap Model
PHY	Physical
PLR	Packet Loss Ratio
PN	Personal Network
PPT	Proactive Passive Technique
PPTD	Packet Pair/Train Dispersion
PRM	Probe Rate Model
PSTN	Public Switched Telephone Network
PTP	Packet Train Pair
QoS	Quality of Service
QAP	QoS Access Point
QoS-AODV	Quality of Service Ad-Hoc On-Demand Distance Vector
RAACM	Resource Allocation and Admission Control in MANET
RBM	Reactive Bandwidth Measurement In 802.11 Networks
RERR	Route Error
RPPAP	Reactive Packet-Pair Active Probing
RREQ	Route Request
RREP	Route Reply
RTO	Retransmission Timeout

rtPS	Real-Time Poling Service
RTS	Request to Send
RTT	Round Trip Time
RTTVAR	Round Trip Time Variation
RT-WABEST	Round-Trip Wireless Available Bandwidth Estimation Tool
SAR	Session Admission Ratio
SC-FDMA	Single Carrier Frequency Division Multiple Access
SIFS	Short Interframe Space
SLDRT	Self-Loading Decreasing Rate Train
SLoPS	Self-Loading Periodic Stream
SRTT	Smoothed Round Trip Time
SS	Subscriber Station
STD	State Transition Diagram
TCP	Transmission Control Protocol
TCP/IP	Transmission Control Protocol and Internet Protocol
TCPV	Transmission Control Protocol Vegas
TCPW	Transmission Control Protocol Westwood
TDMA	Time Division Multiple Access
TOPP	Train of Packet Pairs
TORA	Temporally Ordered Routing Algorithm
TS	Traffic Steams
TTL	Time to Live
TWABE	Two-Way Available Bandwidth Estimation
TXOP	Transmit Opportunity

UDP	User Datagram Protocol
UGS	Unsolicited Grant Service
VANET	Vehicle-To-Vehicle Network
VBR	Variable Bit Rate
VI	Video
Vo	Voice
VoIP	Voice over Internet Protocol
VPS	Variable Packet Size
Wbest	Wireless Bandwidth Estimation Tool
ZRP	Zone Routing Protocol

Copyright Declaration

I declare that this thesis is my own unaided work. Where collaboration with other people has taken place, or material generated by other researchers is included, then parties and/or materials are indicated in the acknowledgement; or they are explicitly stated with references, where appropriate.

This work is being submitted for the degree of Doctor of Philosophy in Engineering and Built-in Environment, at Anglia Ruskin University, Chelmsford. It has not been submitted before for any degree or examination in any other university.

Signature of Author ……………………………………………………………….

Chapter 1: General Introduction

Over the past 15 years, the interest in mobile ad-hoc network (MANET) has significantly increased. Due to its numerous potentials, focus has been shifted towards MANET to provide robust and spontaneous communication in areas where there is limited or lack of centralized infrastructure. One significant difference between the traditional internet access and MANET is that the former can be provided by a gateway node while users of the later are collaborators sharing contents and messages amongst themselves. Application areas of MANET includes battlefields, temporary gathering i.e. conferences, disaster recovery and highly mobile vehicle-to-vehicle networks (VANETs) (Solano & Ordonez, 2017) (See Table 2.2). In the developing world where people live in areas where there are less infrastructure, MANET can be of high benefit. In fact, the project of "One Laptop Per Child" is bringing about the largest real-world MANET like networks till date (Solano & Ordonez, 2017). Most of the currently designed laptops and personal digital assistants (PDA) now comes with 802.11- compliance air interface with the option to operate them in an ad-hoc mode, where 802.11 is the primary enabling technology of MANET. Most literatures on MANET assumes an 802.11-based media access control (MAC) and physical (PHY) layer solutions. According to (Sharma et al., 2018) contention free MAC schemes such as code division multiple access (CDMA) or time division multiple access (TDMA) are difficult to implement for MANET because of its dynamic change in network topology. Therefore, we focus on the 802.11-based MANET.

While MANET provides various benefits to users, there are also underlying quality of service (QoS) issues associated with it. For example, delay, jitter, or packet loss can be experienced by the network with no guaranteed QoS. Therefore, this has necessitated the study of QoS associated with MANET during its design.

QoS is defined by the international telecommunication union (ITU) as the ability to totally satisfy the service users stated needs (ITU QoS manual, 2017). In computer networks, QoS is achieved by service providers by ensuring standard best practice is adhered to with respect to some given metrics such as network availability, delay, bandwidth, packet loss and error rate. The QoS can be classified into hard or soft. Applications that requires a metric to be used for guaranteed QoS and does not tolerate momentary QoS requirement violation is known as hard QoS requirement. Applications that can tolerate momentary QoS requirement violation and the tolerance of the violation does not lead to the malfunctioning of the system is known as the soft QoS requirement. Furthermore, this kind of application can tolerate a limited momentary violation. In general, the QoS solution/model can be classified as per-flow model or class-based model. In a per-flow QoS based model, the QoS are carried out on a per-flow basis, i.e., it ensures that end-to-end QoS requirements of each flow must be guaranteed with minimum or no violation of a QoS flow agreement. In a per-flow QoS model, there is an issue of scalability i.e. in-case there are large number of QoS flow request, there may be insufficient available resources. For the class-based QoS model, the guaranteed QoS are carried out based on aggregates, i.e., this QoS model defines different classes of traffic, and flow groups are mapped to certain class of traffic based on some given criteria such as flow priority, pricing policy, types of application, etc.

Application areas of MANETs as previously discussed are battlefields, temporary gathering i.e. conferences, disaster recovery and highly mobile vehicle-to-vehicle networks (VANETs) (Solano & Ordonez, 2017). Therefore, applications running on such network can generate real-time multimedia data (Hasan & Al-Rizzo, 2017) that requires QoS provision to support its application.

In MANET, bandwidth is shared among transmitting devices and they are often scarce resources. Overhead and the interference associated with the wireless CSMA/CA MAC protocol, and network protocol stack implementation further limits the available bandwidth. This can therefore lead to network congestion even if a nodes transmission rate is below the bandwidth supported by the wireless standard. However, when there is congestion, the rate of delay and loss of packet increases, leading to degradation of performance of real-time multimedia flow within the network. Each node within MANET should therefore estimate the available bandwidth. According to the estimated available bandwidth, an admission control flow algorithm can be used to control the amount of data within the network to satisfy the QoS requirement of the real-time multimedia data. The design of an admission control algorithm for MANET is quite challenging due to the shared nature of the wireless medium (Ambika & Jayachanran, 2017).

The function of a routing protocol is to convey data flow from the source to the destination. The state at which data flow is been conveyed by the nodes in terms of available bandwidth, congestion, and delay during transmission can affect the real-time multimedia application. Therefore, to satisfy the QoS of the real-time multimedia flow, a routing protocol must choose the data forwarding path that best suites the QoS requirement of the real-time multimedia flow.

1.1.Research Motivation

This research is motivated by the fact that MANET is a technology that promises unique communication opportunities. The IEEE 802.11 standard has allowed an affordable MANET to be realised. It is therefore anticipated that the various QoS issues pertaining to the unreliability wireless channel, lack of central co-ordination, contention for channel access and the node mobility must be addressed in order to achieve a general adoption of the technology across board. Unfortunately, due to the nature of MANET, guaranteeing QoS and network resource estimation are non-trivial.

Admission control is one of the most significant system components for QoS provision in MANET. The function of the admission control mechanism is to estimate the network resource states and decide the type of data application session to be admitted into the network without promising more resources than are available, thereby violating any previously made rules. For the admission control to be realised for QoS provision, the estimation of the available resources, such as the available bandwidth, must be carried out to restrict the data traffic in the network based on the available bandwidth and QoS flows requirement. In MANET, the communication medium sharing is done by the MAC layer, therefore the amount of bandwidth available to applications running on MANET is dictated by the MAC layer (Rizal & Bandung, 2017). Also, a routing protocol chooses the data forwarding path. The link state of the forwarding path is selected in terms of the available bandwidth which may impact the performance of a real-time multimedia application (Ambika & Jayachanran, 2017). Addressing the QoS issues in MANET has therefore necessitate a thorough research in this area of study.

1.2.Research Questions

- How can bandwidth estimation for MANET be achieved for a guaranteed QoS?
- How can the bandwidth estimation process be accurate and efficient to limit the overhead generated during the bandwidth retrieval process?
- How can the QoS in MANET be guaranteed when a traffic flow request is initiated?
- What are the measures to be taken in order to analyse the performance of the proposed admission control QoS for bandwidth estimation in MANET?
- What are the measures taken to validate and evaluate the performance of the proposed admission control QoS for bandwidth estimation protocol?

1.3. Research Objectives

The main objective of this Doctoral Degree FHEQ level 8 thesis is to provide resource allocation in MANET in terms of available bandwidth estimation for admission control to achieve better QoS, while guaranteeing a wide bandwidth. We have therefore divided the main objective of this work into sub-objectives for clarity purpose. Therefore, other objectives as identified in this thesis are as follows:

- Objective 1: To develop a bandwidth measurement technique that provides better accuracy when compared with previous bandwidth estimation methods.
- Objective 2: To develop a novel and efficient bandwidth estimation and admission control that prevents frequent bandwidth usage that increases the network overhead during the retrieval of neighbouring bandwidth.
- Objective 3: To develop a mechanism that considers the key metrics necessary to be implemented to enable a guaranteed control of admission within the network.
- Objective 4: To analyse the performance of the throughput, data dropped, and delay of the proposed bandwidth estimation for admission control in order to check for accuracy and efficiency when compared with bandwidth estimation that has no admission control implementation.
- Objective 5: To validate and evaluate our proposed bandwidth estimation for admission control technique for effective comparison with the state-of-the-art bandwidth estimation for admission control protocol.

1.4. A Statement of Original Contribution to Knowledge

1.4.1. Gaps in Knowledge

From the literature, it has been observed that the channel idle time dependency consideration during bandwidth estimation process sensed by the sender and receiver has not been properly addressed. Most of the previous works in the literature did not consider this, while on the other hand, the few works that considered the channel idle time dependency only differentiates the *BUSY* state from the *SENSE* BUSY state. The *IDLE* state that is caused by an empty queue is yet to be addressed. Also, most of the proposed QoS solutions that estimate available bandwidth for admission control broadcasts to two hop neighbours, in order to retrieve the available bandwidth on a carrier sensing region. This tends to create a higher overhead which can possibly be avoided.

This thesis addresses how well the available bandwidth estimation can be carried out by not only differentiating the *BUSY* state from the *SENSE* BUSY state but addressing the *IDLE* state that can be caused by an empty queue when the channel idle time dependency sensed by the sender and the receiver node is considered. This thesis also addresses how well the available bandwidth can be retrieved on the carrier sensing region without flooding the network with enormous broadcast messages. To retrieve the available bandwidth on a carrier sensing region, the HELLO message only advertises to the first-hop range which further propagates to other hops. This technique adopted to retrieve the available bandwidth helps to limit the overhead generated by the network.

1.4.2. Research Contributions

This thesis contributes to knowledge by proposing a resource allocation and admission control in MANET (RAACM) mechanism that estimates the bandwidth for admission control based on the following factors:

- Bandwidth estimation process that considers the channel idle time synchronization and dependency between the sender and the receiver node by differentiating the *BUSY* state from the *SENSE BUSY* states and the *IDLE* state caused by an empty queue.
- HELLO packet propagation to retrieve the available bandwidth on the carrier sensing region.
- A novel, efficient and accurate resource allocation and admission control in MANET (RAACM) that estimates the available bandwidth for the admission controller to either accept or reject a session, when an admission is requested.

1.4.2.1. Bandwidth Estimation Process

This thesis considered the bandwidth estimation process, where channel idle time dependency is incorporated and nodes within the interference range of the sender and the receiving node are not randomly distributed. The bandwidth estimation also provides for synchronization by allowing the sender and the receiver node to witness both common interference (complete overlap) and independent interference (no overlap) during the bandwidth estimation process. collision and back-off have also been highlighted in this thesis as other factors that have an impact on the available bandwidth estimation asides the channel idle time period.

It is important to note that to satisfy a bandwidth requirement, it is essential to estimate the available bandwidth (Milton et al., 2016). If the requirement of an available bandwidth is exceeded, this may result in packet drop, which can increase the delay in a network and eventually decrease the throughput. The bandwidth, as previously mentioned, is a shared resource in wireless medium. The available bandwidth is affected by the channel idle time estimation, intra-flow contention, collision and back-off. Hence, the bandwidth estimation in wireless network differs from that of the wired networks, since bandwidth estimation in wired network does not need to consider the intra-flow contentions. In this thesis, we highlight the different factors to be considered for an effective available bandwidth-based flow admission control in MANET. These are channel idle time dependency which has been modified to enhance proper estimation, intra-flow interference, collision and back-off. Based on these outlined factors, we designed RAACM, a measurement based available bandwidth estimation for admission control algorithm. Our result shows that RAACM significantly outperformed the state-of-the-art available bandwidth-based flow admission control algorithm for MANET.

1.4.2.2. HELLO Packet Propagation

A carrier sensing range that is unsuitable in a network can affect the interference of a mobile ad-hoc network causing a higher collision probability in a channel. When there is a high collision probability in a network, it can cause network overhead which affects the performance of MANET due to the high demand of network reconnection (Aina et al., 2019). Therefore, a challenge is posed to the network designer to implement a network that reduces the network overhead by decreasing the level of collision probability using a suitable carrier sensing (CS) range. There are two ranges around the node. The CS range which covers a double distance circularly around the node and the transmission range which helps to inform us of the node activity through the clear channel assessment (CCA) provided by the MAC layer. Information about the idle/activity time over a predefined time window can be known.

To retrieve the available bandwidth on a carrier sense region, we estimate the serviceable bandwidth. The serviceable bandwidth is defined as the smallest available bandwidth observed on a sensing region. The main idea behind the bandwidth retrieval process is to make use of HELLO message, which is forwarded between nodes for connectivity awareness. The HELLO message only advertises to the first-hop range before it propagates to the rest of the hops in the network. HELLO advertisement to only the first hop range, which has been adopted by our proposed protocol RAACM, has not previously been used in the literature. The serviceable bandwidth calculation remains accurate because the carrier sensing nodes information is propagated in the packet. This further significantly reduce the overhead within a network, since the HELLO packet is extended rather than flooding the information over the retrieval range. The information is however gradually obtained during the network deployment process.

1.4.2.3. Resource Allocation and Admission Control in MANET (RAACM)

RAACM estimates the available bandwidth for the admission controller to either accept or reject a session, when an admission is requested. Admission control is one of the most significant system components for QoS provision in MANET. The function of the admission control mechanism is to estimate the network resources states and decides the type of data application session to be admitted into the network without promising more resources than are available thereby violating any previously made rules. For the admission control to be realised for QoS provision, the estimation of the available resources such as the available bandwidth is carried out based on the factors identified in 1.4.2.1. Thereafter, we restricted the amount of data traffic inside the network based on the available bandwidth and QoS requirement.

1.5.Thesis Organization

The rest of the chapters in this thesis are structured as follows:

Chapter 2: This chapter provides the background information necessary to understand the research carried out. This includes the history and definition of MANET and the characteristics and complexity related to MANET. Furthermore, this chapter also discusses the application area of MANET and the challenges posed by admission control protocols and resource allocation for admission control in MANET. This chapter went further to investigate the bandwidth estimation for admission control in MANET as well as the protocol design consideration for admission control QoS in MANET.

Chapter 3: This chapter is the literature review chapter that discusses the bandwidth estimation for admission control in MANET. The bandwidth estimation for admission control was classified into active technique and passive techniques. Thereafter, the active bandwidth estimation technique was further classified into single pair active technique and packet pair active technique, while the passive bandwidth estimation was classified into generic passive technique and proactive passive technique. Different bandwidth estimation protocols for admission control proposed by different researchers were properly investigated and a table showing the summary of some bandwidth estimation techniques for admission control in MANET and their innovations where outlined in this section. Thereafter, this chapter discusses the admission control in MANET and focused on the distributed admission control and its implementation process. This chapter went further to investigate the routing protocols for MANET design and their classifications as well as the various wireless standards that exist in literature to enhance suitable QoS deployment.

Chapter 4: This chapter provides the detailed simulation tool and the design model used in this thesis. Furthermore, this thesis introduces the optimized network engineering tool (OPNET), its overview, environment, application, and structure. This thesis used OPNET now referred to as Riverbed modeler to carry out the network modelling and simulation. Thereafter, the bandwidth estimation and admission control implementation in OPNET modeler was discussed.

Chapter 5: This chapter analytically described the methodology and implementation of the proposed technique. This technique was explained by describing the bandwidth estimation for admission control in MANET, as well as the key element needed for bandwidth estimation for admission control. Furthermore, the procedure used for the implementation of the distributed admission control was provided in detail.

Chapter 6: This chapter discusses the result analysis, validation and evaluation in this thesis. Also, this chapter discussed the integration of routing protocols into our admission control, the simulation parameters used, and the results obtained. Furthermore, our proposed model was assessed through validation with other state-of-the-art bandwidth estimation and admission control algorithm. Results obtained show that our technique outperformed other proposed state-of-the-art available bandwidth estimation and admission control.

Chapter 7: This chapter discusses the conclusion and the future works. Resource allocation and admission control in MANET (RAACM) for bandwidth estimation measurement was proposed. This is used for admission control purpose to perform some pre-configured checks prior to establishing a connection to know if the current bandwidth resources are sufficient for a proposed connection to guarantee QoS. However, this thesis provided an highlight of its contribution at the concluding section. This thesis also suggested that in the future, it will be of interest to modify the transport protocol used for the communication, routing protocol, and modifying findings of the available bandwidth estimation to see how our estimation can work in different scenarios. Lastly, this thesis suggested the modification of the bit rates for voice and multimedia application.

1.6. Critical Review of the Research Undertaken

Having reviewed the literature, it was observed that the channel idle time dependency sensed by the sender and receiver has not been properly addressed, as most previous work did not consider it. Related works that considered the channel idle time dependency for the estimation of available bandwidth only differentiates the *BUSY* state from the *SENSE* BUSY states, therefore the *IDLE* state caused by an empty queue is yet to be addressed. Furthermore, it was observed that the state-of-the-art available bandwidth for admission control broadcasts to two hop neighbours in order to retrieve the available bandwidth on a carrier sensing region. This tends to create a higher overhead that can possibly be avoided.

This thesis has however, addressed the channel idle time dependency by the sender and receiver by not only differentiating the *BUSY* state from the *SENSE* BUSY states but addressing the *IDLE* state that can be caused by an empty queue during the estimation of available bandwidth. Also, this thesis has been able to address how well the available bandwidth can be retrieved on the carrier sensing region without flooding the network with broadcast messages. To retrieve the available bandwidth on a carrier sensing region, the HELLO message only advertises to the first-hop range which further propagates to other hops. This technique adopted to retrieve the available bandwidth helps to limit the overhead generated by the network.

RAACM addresses the channel idle time dependency between a sender and a receiver by differentiating a nodes *BUSY* state from when it is in a *SENSE BUSY* state and addresses the IDLE state that may be caused by an empty queue. The *BUSY* state is defined as a situation whereby a node is in a transmission or receiving state, while the *SENSE BUSY* state is defined as a situation whereby a node is in the state of sensing. Any other time outside the sensing time, the node will be in an IDLE state. Other state-of-the-art protocols that has been proposed, such as, BECIT, MBA-AODV, AABWM addresses the channel idle time in a different way to RAACM by using an independent node channel idle time method with randomly distributed nodes.

In RAACM, to retrieve the available bandwidth on a carrier sense region, the serviceable bandwidth is estimated (serviceable bandwidth is the smallest available bandwidth observed on a sensing region). The main idea behind the bandwidth retrieval process is to make use of HELLO message, which is forwarded between nodes for connectivity awareness. HELLO message advertisement is sent to the first hop range (1 hop packet propagation). The first hop propagates the HELLO packet to other neighbours in the network to retrieve bandwidth information. The minimum propagation distance of HELLO packets amongst other protocols in the literature such as cognitive agent based available bandwidth estimation using collision probability, idle period synchronization and random waiting time (BECIT) (Chaudhari & Biradar, 2015), (MBA-AODV) (Sharma et al, 2018), and analytical available bandwidth estimation including mobility (AABWM) (Mukta & Gupta, 2019) is 2-hops range. Propagation to more than one hop increases the overhead on a network.

The method adopted by RAACM was comprehensively analysed and a formula was generated. Simulation was carried out using OPNET modeler to evaluate the performance of our proposed novel admission control algorithm. Simulation result obtained was analysed based on admission control throughputs, delay analysis, and data dropped in WLAN 802.11e to check for the QoS. To evaluate the performance of our proposed RAACM, we first compared RAACM with a WLAN that has no admission control and QoS prioritization deployed. We noticed that by allowing three traffic flow (data, voice, and video) to transmit within the network without an admission control, there will be network congestion, as all the traffic flow will attempt to transmit at the same time and with insufficient bandwidth, there will be lots of drop in packet. Secondly, RAACM WLAN throughput access for voice and video admission control was compared with BECIT, MBA-AODV and AABWM. Result obtained shows an efficient throughput, where the average throughput of RAACM for voice traffic tends to be more than BECIT, MBA-AODV and AABWM, used for comparison. The video traffic also shows an effective and reliable result and RAACM ensures that the voice is not giving too much of the priority during transmission and disallowing the video traffic to get hold of a very high portion of the network. However, RAACM provides equal priority for voice and video traffic which enhances accuracy as against other protocol used for comparison. Furthermore, this thesis evaluated the proposed RAACM and the closely related research work with WLAN HCF delay. Result obtained shows that RAACM has the least amount of video delay, while the best effort and voice delay generated by RAACM is less when compared with BECIT and AABWM, with almost similar delay as MBA-AODV. Finally, this thesis evaluated RAACM and the closely related research work with WLAN HCF data dropped. Result obtained shows that RAACM has stability and efficiency in the packet dropped and the amount of packet dropped has been of minimal value as compared with the rest of the other protocol used for comparison.

Chapter 2: General Overview

2.1. Introduction

This chapter will discuss the general overview of resource allocation and admission control in MANET. Furthermore, the history and definition of MANET will be stated before discussing its characteristics and complexity. This chapter will further discuss the application area of MANET and the challenges posed to admission control by the MANET environment. Furthermore, this chapter will discuss the resource allocation for admission control in MANET with emphasis on the bandwidth estimation for administering admission control. Lastly, this chapter highlights protocol design consideration for admission control QoS in MANET.

2.2. History and Definition of Mobile Ad Hoc Networks

The concept of MANET is not new as its origin can be traced back to the defence advanced research projects agency (DARPA) packet radio network project in 1972 (Aina et al., 2019). MANET is characterized by its unique attributes such as flexibility, resilience, mobility, and infrastructure-less which attracted the military, police force and rescue agency. Over the years, research works on ad-hoc network was directed towards the military realm, until the middle 90's when the advent of commercial radio technology came to place and the wireless research community became aware of the advantage and potentials of MANET, aside the military environment and the creation of MANET within the internet engineering task force (IETF) (IETF, 2018). Presently, the research area of MANET is now vibrant and active with the efforts rendered by the research community been able to develop a commercially viable MANET with existing and new application oriented solutions already coming to place.

As opposed to the infrastructural network where each node communicates directly with an access point, MANET does not communicate using a fixed infrastructure as shown in figure 2.1. MANET is an autonomous transitory group of mobile nodes that communicates with one another over a wireless link. Nodes that are located within each other's sending range can directly communicate with one another and have the responsibility of dynamically discover each other. For nodes that are not directly within the same communication range to be granted communication, an intermediate node act as a router that communicates packets generated by other nodes to their destination. These nodes however are battery powered and energy constrained. Also, communication devices can join or leave the network and can also move randomly which may possibly result in an unpredictable and rapid topology changes. Since the nodes in MANET are energy constraint due to the distributed and dynamic multi-hop environment, these nodes therefore need to dynamically organise themselves for proper network functionality in the absence of a fixed network infrastructure or central co-ordination. The specific characteristics and complexity of MANET as summarized in table 2.1 creates more design challenges to the network protocols, coupled with the fact that this network encounter other traditional problems common to wireless communication networks such as having a lower reliability when compared with wired network, limited security measures, interference, time-varying channels, etc.

Despite the various constraints associated with MANET, it still offers numerous advantages. Firstly, the ad-hoc network can be used where fixed infrastructure is unavailable, too expensive, lacks trust or unreliable. Due to its self-organizing nature, self-creation and self-administration capability, MANET can be deployed with limited user intervention as there is no need to outline the necessary base station to be deployed. MANET does not necessarily need to function as a stand-alone, however, they can be deployed by attaching them to the internet, where different devices are integrated together making the service available for other users.

Characteristics and Complexity of MANET
• Device heterogeneity
• Scalable network
• Multi-hop routing
• No infrastructure in place
• Physical security is limited
• Provides for self-organization and self-administration within the network
• Operates within an energy constrained environment
• Dynamic network topology
• Constrained bandwidth with variable link capacity

Table 2.1: Characteristics and Complexity of MANET

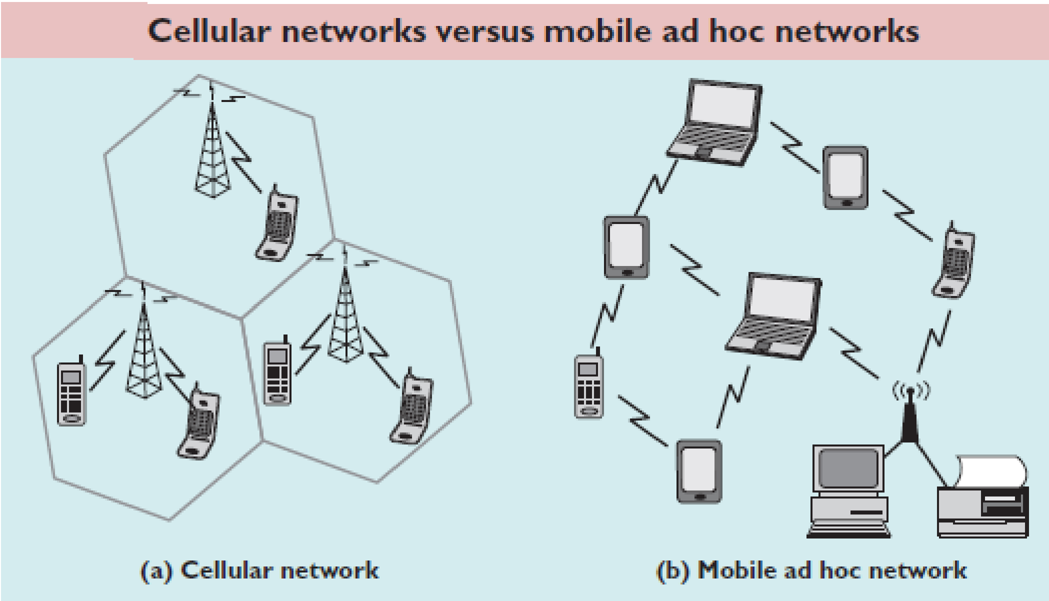


Figure 2.1: Cellular Networks versus Mobile Ad-hoc Network (MANET) (Agarwali et al., 2018)

Applications	Possible scenarios/services
Tactical networks	• Battlefields. • Military operation and communication.
Emergency operation	• Search and rescue services. • Police and fire fighting. • Disaster recovery service. • To support doctors and nurses in the hospital. • Fixed infrastructure replacement in the event of environmental disaster.
Education	• Virtual classrooms • Universities and campuses • Ad-hoc communication during lectures or meetings
Civilian and commercial environment	• Network of visitors at airports • Shopping malls, trade fares, sports stadium. • E-commerce i.e. electronic payments anytime and anywhere. • Vehicle service i.e. taxi network, accident or road guidance, transmission of road and weather condition, intervehicle network. • Business, e.g. mobile offices, dynamic database access.

Entertainment	• Outdoor internet access • Wireless point to point networks • Theme park • Multi-use game • Robotic pets
Enterprise and home networks	• Meeting room and conferences. • Personal network (PN) and personal area network (PAN). • Office or home wireless networks • Construction site networks.
Sensor network	• Body area network (BAN). • Home application e.g. smart sensor and actuators embedded in consumers electronics. • Tracking of environmental condition data, biological/chemical detection, and animal movement.
Extension coverage	• Cellular network access extension. • Internet and intranet linking up
Context aware services	• Information service such as location specific service and time dependent service • Follow-on service such as call forwarding and mobile workspace.

Table 2.2: MANET Application Area (Khan et al., 2018)

Figure 2.2: Applications of MANET (Khan et al., 2018)

While MANET provides various benefits to users, there are also some underlying QoS issues associated with it. For example, delay, jitter, and packet loss can be experienced by the network with no guaranteed QoS.

Providing QoS to MANET users possess concern to the service providers as well as the service users. Tasks, especially real-time applications, require QoS to enhance its communication (i.e. multimedia data). Nodes must therefore cooperate with one another to guarantee effective QoS. The cooperation must include the endpoint flow policing as well as admission control implementation along the route, to prevent network violation of initially made policy. The aim of deployed QoS support is to provide guaranteed application support in terms of delay, jitter, throughput, bandwidth, etc. To ensure this, the MAC layer takes the responsibility of allocating resources at individual nodes, while the network layer must consider resources along the entire communication route. One of the techniques that provide QoS assurance is admission control. The aim of admission control is to decide on the class of data application that can be admitted into the network without having to promise more resources of unavailable spaces, in order not to violate any previously made guarantees. Admission control can be used during the allocation and usage of network resources for several applications that requires additional services.

Admission control can therefore be regarded as a component that needs to allow resources, such as bandwidth, to be used only when it is available (Aina et al., 2019).

2.3. Challenges of Admission Control Protocol in MANET Environment

Due to the probabilistic nature of wireless medium, admission control for MANET have challenging problem. The shared wireless medium characteristics does not provide a unified view of the channel to all the nodes because of the physical difference between wired and wireless communication, since the aim of an admission control protocol is to prevent the over-utilization of network resources and maintain QoS assurance. The challenges of interest are those that prohibit or burdens this operation. Many of these challenges identified are similar to the ones posed to QoS aware routing protocol as discussed in (Sharma et al., 2018), however, there may be some differences in the impact. Therefore, some critical issues to consider when designing an admission control mechanism is as follows:

i. Unreliable wireless channel: Wireless channels are susceptible to bit error because of the interference it has with other transmitting nodes, such as, shadowing, thermal noise, and fading effect (Dixon et al., 2018). This makes it difficult to provide link guarantee for a long period or hard packet delivery ratio.

ii. Channel contention: Nodes in MANET usually communicate on a common channel to discover network topology. This, however, can give rise to interference problem. In the case of a peer-to-peer connection, the contention problem can be avoided in several ways. Firstly, global clock synchronization using TDMA-base system by allocating different time slot to each transmitting node can be considered. For MANET, this is difficult to achieve because of a lack of central coordinator, mobility of node, overhead, and complexity involved (Aina et al., 2019). Secondly, different frequency band such as CDMA, can be used for each transmitter in peer-to-peer connection. However, communication such as neighbour discovery, and other setup must take place through contention.

iii. Lack of central coordinator: Lack of setup planning is the main property of any ad-hoc network. The problem with this is the dynamic change in topology which makes it difficult to provide a centralized coordinator. Therefore. communication protocols that involves the utilization of locally available bandwidth is preferred (Paul et al., 2016). Algorithm complexity and overhead tends to increase because the QoS state need to be disseminated appropriately.

iv. Node mobility: In MANET, nodes may move randomly to any extent. This can cause routing failure which may lead to channel error issues. Mobility can also give rise to the violation of QoS rule assurance previously made. A transmitting node may move into the carrier sensing range of another transmitting node causing interference and reducing the channel access time. Data session that has been admitted based on a given rule may be starved of transmission opportunity (Liao et al., 2017).

v. Issue of connectivity: In mobile device, connectivity may be lost within a group because of distance of nodes from each other or drop in power reservation under a given threshold. A session admitted because of the available route may suffer from transmission opportunity in situation of connection loss among some transmitting nodes. There will therefore be need for re-admission using a new route (Solano & Ordonez, 2017).

A vital component of an admission control mechanism to enhance QoS in MANET is resource allocation.

2.4.Resource Allocation for Admission Control in MANET

Resource allocation aims to dynamically share the physical resources available amongst various transmitting devices. Unlike the cellular system that are organised around the base station and are deployed by the operator, ad-hoc network is relaxed from fixed infrastructures. Therefore, ad-hoc network is a flexible solution for fast and short-lived communication deployment for lots of applications ranging from the military ground or some other critical scenarios to any future smart network. Thus, the lack of infrastructure makes the management of resources difficult when compared with cellular systems that have fixed infrastructure. The key challenge involved in designing MANET is the efficient allocation of network resources and the provision of the necessary QoS required by various users (Yang et al., 2017).

Resource allocation includes several components, such as bandwidth allocation, transmission power control, spectrum allocation, and rate adaptation. Resources allocation within the context of this work will focus on bandwidth estimation for administering admission control.

2.5. Bandwidth Estimation for Admission Control in MANET

The dynamic and decentralized nature of MANET enables network information, such as the topology and the end-to-end flow be passed from one node to another. In addition, distributed bandwidth estimation algorithm is always preferred over the centralized ones. The criteria of the distributed bandwidth allocation are divided into three, namely: the convergence time of the algorithm, the amount of data passed from one node to the other, and finally, the fairness of the algorithm (Jakimoski et al., 2016).

Bandwidth allocation must operate across the MAC and the network layer of the OSI layer model. The responsibility of the network layer is to calculate the end-to-end flow rate while the MAC layer takes the responsibility of scheduling transmission to ensure that each of the flows receive adequate bandwidth over the link (Lei et al., 2015).

2.6. Admission Control to Enhance QoS in MANET

Admission control is a QoS validation that performs some pre-configured checks prior to establishing a connection to know if the current resources are sufficient for a proposed connection. The task of an admission control is to control the allocation and usage of network resources, especially for applications requiring additional service. It does not guarantee any node resources if there are limited available space within the network, to avoid network interference and congestion. Admission control is one of the most crucial mechanism for providing QoS. It is a key component that needs to allow resources like bandwidth to be used when it is available (Solano & Ordonez, 2017).

2.7.Protocol Design Consideration for Admission Control QoS in MANET

2.7.1. Metrics for QoS

Metrics used for measuring QoS was explained in (Sharma et al., 2018), therefore, in this section, we give a brief summary of the QoS metrics:

2.7.1.1.QoS requirement specification: This can be specified based on the traffic result of one or more of the following metrics:

- Average maximum throughput: This is the total maximum amount of packet delivered against a given time interval (Lei et al., 2015).
- Maximum packet loss ratio: This is the maximum data packet that is lost during routing. Loss of data can be caused by buffer overflow when the network is congested, and transmission limit exceeded during mobility or time-out when waiting to discover new route (Kalpana., 2018).
- Maximum delay and jitter: This is the difference between the upper bound delay and the minimum delay determined by cumulative transmission time and packet propagation.

- Maximum packet delay bound: This is the gathering of MAC delay and queuing at each node where the delay propagation is very short (Aina et al., 2019)

2.7.1.2. Quantifying the performance of admission control: By considering the metrics, admission control protocol task act as an essential balance. It also aims to serve many users, thereby admitting as many sessions as possible while effectively and fully making use of the network resources. However, any inaccuracy in making the admission decision can lead to guaranteeing more resources than are available, therefore causing a false admission. It is easier to provide high QoS to admitted session when there is network under-utilization, however, the issue related to this lies around low efficiency in term of overhead, consumption of energy and network resource wastage. Rejecting a session that is supposed to be served without causing any QoS degradation is known as false rejection. In summary, the admission control metric should reflect trade-off between the probability of false admission and false rejection. We can therefore classify the metric based on whether they can utilize resources or their ability to satisfy application requirement of users. Even though most research on admission control show their result based on throughput or delay over time, this tends to show only a small scale of the protocol performance. Metrics therefore, needs to show the quantification of large-scale performance (Hasan et al., 2017). Below are some suitable metrics to be considered:

- Capacity Utilization: This is the average fraction of the network capacity utilized during transmission. Large number of false rejection results in a low network capacity utilization. Therefore, wireless network capacity with random topology tends to be difficult to quantify. Researchers often tend to use aggregated network throughput to reflect the level of capacity utilization, but aggregated throughput is a subjective metric which cannot be used to compare result from different networks, it can only be used to compare result for different protocols within the same network having the same offered traffic load (Ambika & Jayachanran, 2017).
- Session Admission Ratio (SAR): This is the ratio of the data session admitted into the network to the overall number of requesting admission. This metric can be used in cases where difficulty is encountered during the effective capacity utilization estimation. On-like the aggregated throughput, SAR reflects the amount of data session served. It allows admission control mechanism to discover available resources and utilize them. The ability of this protocol to discover a

suitable route may affect the SAR. Therefore, a protocol that achieves a higher SAR that tends not to degrade the QoS experience can be considered. The weakness of this metric is that it depends on the offered traffic load and the entire network capacity. A better measure would be the number of session admitted per unit capacity, if the capacity can be quantified (Solano & Ordonez, 2017).

- False Rejection Ratio (FRR): The value of false rejection can be based on the number of rejected session or admitted requests. Practically, it is difficult to quantify FRR because whether a rejection is false or not depends on the session requirement and state of the resources. Therefore, evaluating FRR will require each admission decision to be compared to the global view of the network resources which cannot be done accurately on a real network, but only by simulation (Solano & Ordonez, 2017).
- False Admission Ratio (FAR): This is the amount of false admission that is normalized by the number of admitted sessions or admitted requests. This is also difficult to quantify just like FRR, but there are lots of method that can help in the indication of the level of over-pledging resources. Measurement can be done based on the average proportion of packet (e.g. delay) or the time fraction (e.g. throughput) for which the required QoS was not upheld (Solano & Ordonez, 2017).

2.7.2. Network Resources

In (Sharma et al., 2018), network resources relevant to QoS have been discussed. This section analyses the resources that are relevant to admission control.

- Channel capacity: It is the most important network resources. If a node's channel indicates busy all the time, the result of other resources will not matter, as there cannot be a good QoS. Low level of residual capacity of the channel give rise to low throughput and delay in channel access of transmitting node. Lots of protocols in the literature have assumed that the rate of a transmitting node is fixed, for ease of analysis and because the rate switching mechanism is not specified in 802.11 standard (Aina et al., 2019). Capacity is therefore mostly expressed in bits per seconds (bps). In situations such as heterogeneous link rate environment, residual capacity may be easily expressed in terms of the fraction of idle time detected by the channel.

- Buffer Space: Buffer space is classified as the second most important network resources. The overall buffer space determines the maximum and actual queue size at relay nodes, which is therefore a major factor in the queuing delay as well as the total end to end packet delay. If the buffer space of a node is filled up, any arriving packet must be dropped. If there is a large maximum queue size, this means that few packets will be lost during congestion, but the end-to-end delay can increase because of longer queuing delay (Aina et al., 2019).
- Battery Charge: Regular access to recharging facilities are always available in MANET, unlike sensor network nodes. Therefore, overhead heavy protocols can probably have an impact on the battery life, thereby necessitating frequent recharging and limits the device usefulness. In terms of fairness, protocols can attempt to balance traffic loads across different routes so that no single user's battery resources are unfairly burdened (Aina et al., 2019).
- Processor time: Processor time is a non-critical resource for admission control since most of the algorithm are computationally simple. However, algorithm such as QoS-aware routing can benefit from abundance processor time (Kalpana & Karthik, 2018).

2.7.3. Estimating Network Resources and QoS Achievable.

A key aspect of admission control mechanism is discovering the state of network resources. By determining this information, admission control decisions can be made. Approaches that have been considered in the literature can be categorised into three namely:

- During route discovery, perform a test of the QoS-related states.
- Deduce the achievable QoS based on the experience of the probe packet sent on a given route.
- Use the QoS experienced by previous node transfer as an indicator for future achievable QoS.

Majority of the solutions from the literature falls under the first category. Therefore, we can estimate the achievable QoS in terms of the following metrics:

- Local residual channel capacity: Monitor the amount of time the channel records idleness by the wireless clear channel assessment (CCA) and virtual carrier-sensing mechanism (Aina et al., 2019). This is also known as channel idle time ratio (CITR). To calculate the maximum transmission rate, observe the amount of capacity consumed during transmission and receiving of packet, then subtract it from the raw channel capacity.

- Link capacity: For the link capacity to be estimated, the delay between transmitting probe packet of known sizes is used, or the minimum of the local residual capacities of the end nodes.
- End-to-end route capacity: This is the minimum of the estimated local residual channel capacity of the nodes on the route, taking inter-route contention into consideration. This also allows probing of end-to-end routes and use the interval between the arrivals of packet to calculate the route capacity.
- End-to-end delay: This can be expressed by sending a probe packet through a route and taking half of the average round trip time experienced by a series of probe packets. An alternate way is that the traversal times of each hop can be individually estimated and summed (Goswami et al., 2017).
- Delay Jitter: Jitter can be estimated based on the delay statistics of existing probe packet or data packet (Aina et al., 2019).
- Packet loss ratio (PLR): The packet loss ratio on a link of periodic beacons with a known frequency can yield an estimate of the PLR (Pinto, 2015). Also, the packet loss ratio of a data or probe packet can be monitored.

2.8.Summary

In this chapter, a general overview of MANET, its services and application area were presented. This chapter also discussed the challenges posed to admission control protocol in a MANET environment. Additionally, the resource allocation for admission in MANET, where bandwidth estimation was given more concentration, was discussed as well as the admission control and protocol design consideration for admission control QoS in MANET.

Chapter 3: Literature Review

3.1. Introduction

This chapter reviews the common bandwidth estimation technique for admission control in MANET and categorize it into active bandwidth estimation technique and passive bandwidth estimation technique. Furthermore, this chapter discussed the active and passive protocol sub-division and highlighted the protocols of each of these techniques. Thereafter, this chapter discussed the admission control in MANET. Admission control was categorized into centralized admission control and distributed admission control. We outlined the characteristics of centralized and distributed admission control separately, based on the characteristics and limitations posed by centralized and distributed admission control mechanism. The distributed admission control mechanism was chosen to be more adequate for the scenario of this thesis, therefore, we focused on using the distributed admission control mechanism. This chapter also discussed the routing protocols for admission control in MANET. Routing protocol in this section was analysed based on its different classification; we choose to deploy AODV routing protocol amongst all other routing protocol because AODV performs well in a more complex scenario (such as high load, high mobility). Finally, we presented a general discussion to enhance our understanding, as well as the chapter summary.

This chapter reviews publications that exist in the literature up until 2020 from ScienceDirect, Springer, IEEE Xplore, Google Scholar and the ACM digital Library, using keywords such as “Admission control in MANET, “Bandwidth estimation for admission control” “QoS in MANET” and “Admission control survey”.

3.2. Bandwidth Estimation for Admission Control

There have been various approaches proposed in the literature to estimate the available bandwidth for admission control. (Liao et al., 2017) and (Dixon et al., 2017) classified the bandwidth estimation techniques into passive (i.e. Non-intrusive) estimation and active (i.e. Intrusive) probing. In (Rizal, 2017), bandwidth estimation techniques were categorized into active probing, mathematical model based, and calculation based passive measurement techniques. (Ni et al., 2016) had a different approach by classifying it into self-congestion and model-based approach, while (Kua et al., 2017) classified the bandwidth estimation technique into algorithm that are designed for specific networks, usually with guaranteed QoS, algorithm that uses probe packet with pre-determined spacing and algorithms targeting video streaming where a client-server is assumed. Researchers have presented different classification of bandwidth estimation techniques, however, they perform the same role notwithstanding their different nomenclatures. We therefore argue that classification of bandwidth estimation into active technique and passive technique as categorized by (Liao et al., 2017) and (Dixon et al., 2017) will simplify the readers understanding of the bandwidth estimation process for admission control.

3.2.1. Active Bandwidth Estimation Technique

In active bandwidth estimation, a dummy packet known as probe packet is transmitted through the network at different traffic rates from the source to the destination node. The available bandwidth along a path is therefore estimated by measuring the inter-arrival times (Jagadev & Pattanayak., 2018). The above technique adds probing traffic and can possibly degrade the existing flow performance (Paul et al., 2016). The main objective of the active technique is to observe the network characteristics by introducing the probe packet.

Most of the previous work classified the active available bandwidth estimation for admission control into single packet/one packet and packet pair (Chaudhari et al., 2015) while other classifications of active technique are different but with the same role. In (Atxutegi et al., 2016), the active available bandwidth estimation technique is classified into isolated probing, direct probing, and iterative probing. (Khangura et al., 2019) classified it into direct probing and iterative probing technique. In (Watson et al., 2019), the active available bandwidth estimation was classified into direct probing, iterative probing, and mixed techniques while (Goebel et eal., 2017) classified it into packet dispersion measurement (PDM), probe gap model (PGM) and probe rate model (PRM). In (Borzemski and Kaminska, 2016), the active technique was classified into variable packet size (VPS) probing, packet pair/train dispersion model (PPTD), self-loading periodic streams (SLoPS) and train of packet pairs (TOPP).

It was observed in (Atxutegi et al., 2016) that isolated probing and probe delay model are the same as single-packet probing technique. In (Yang et al., 2017), the author regards PRM and iterative to be a self-loading technique. Also, (Yang et al., 2017), regards PGM and direct probing to be a packet-pair dispersion technique.

We therefore argue that classification of active bandwidth estimation into single packet/one packet and packet pair, as categorized by (Chaudhari et al., 2015), will simplify the readers understanding of the active bandwidth estimation process. A diagram showing the subdivision of active protocol is shown figure 3.1 below:

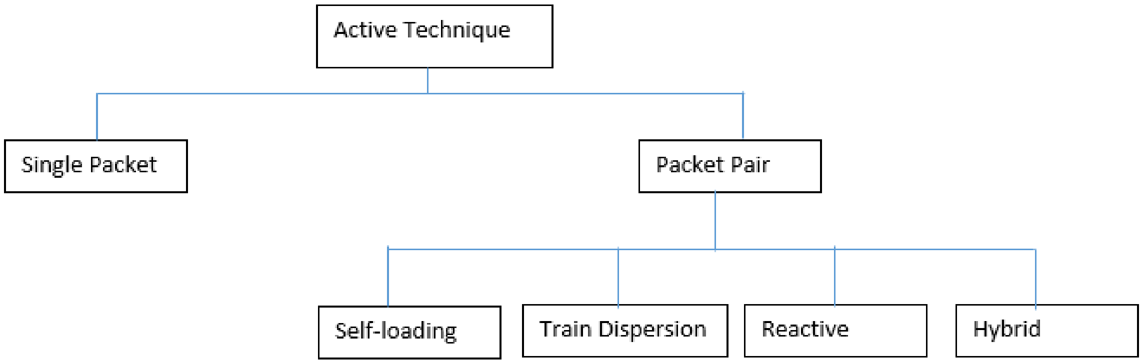


Figure 3.1: Active Bandwidth Estimation

3.2.1.1. Single Packet Active Technique: In this active technique, one probe packet at a predefined time interval is injected into the network in order to measure the delay. Link capacity is measured instead of end-to-end path capacity by using the time difference between the round-trip times of the probe packet from one end of the link to the other. The packet transmission time t =(P/b) +l, where P=packet size, b=link bandwidth, and l is the fixed latency. If the round-trip time and the size of the probe packet is known, the bandwidth can be calculated for a given fixed latency of a link. Tools that have been developed using single packet active probing are as follows; Clink (Salcedo et al., 2018), Pathchar (Kokkonis et al., 2018) Tailgating (Zhou et al.,2016) and Pchar (Abut & Leischner, 2018). The only protocol implemented for the estimation of bandwidth using single probe packet is variable packet size probing (Kirova et al., 2018).

In (Kirova et al., 2018), variable packet size probing was proposed. This protocol measures the capacity along the end-to-end path using the round-trip time from the source through to each hop along the path as a function of the size of the probe packet. In variable packet size probing, a probe packet size is transmitted from a sending host to the network layer along the path of each hop in a continuous way at a predefined interval. The time-to-live field of the IP header in the probe packet forcefully terminate the probe packet when it gets to certain targeted hop. Internet control message protocol (ICMP) time exceeded error message is sent back to the source from each of the hop along the path. The round-trip-time in the probe packet of the target hop is made up of three delay components in the forward and reverse paths which are; serialization delay, propagation delay and queuing delay. Variable packet size probing measures the capacity of the path in the entire network at each hop without making use of any special software along the source and the destination path.

3.2.1.2. Packet Pair Active Technique: In this active technique, two probe packets known as packet pair are sent back-to back towards the target and this echoes the packet back to the sending node. The space between the two packets as shown in the figure 3.2 below is always determined by the bottleneck link which is preserved by a higher bandwidth link (Megyesi et al., 2017). A packet within the packet pair that arrives at the target node have a specific time separation between each packet which is specified by Δin. Having interacted with the cross traffic, from various sources, packets exit the output queue with changed time separation which is stored as Δout. Researchers have tried to formulate models based on the variation between Δin *and* Δout. The packet-pair active probing techniques that have been published differ in the way the packet sequence rate increases and in the metrics measurement of the probe packet flow. The packet-pair active probing technique can therefore be further classified into: self-loading packet-pair active probing, packet pair/train dispersion active probing, reactive packet-pair active probing, and hybrid packet-pair active probing.

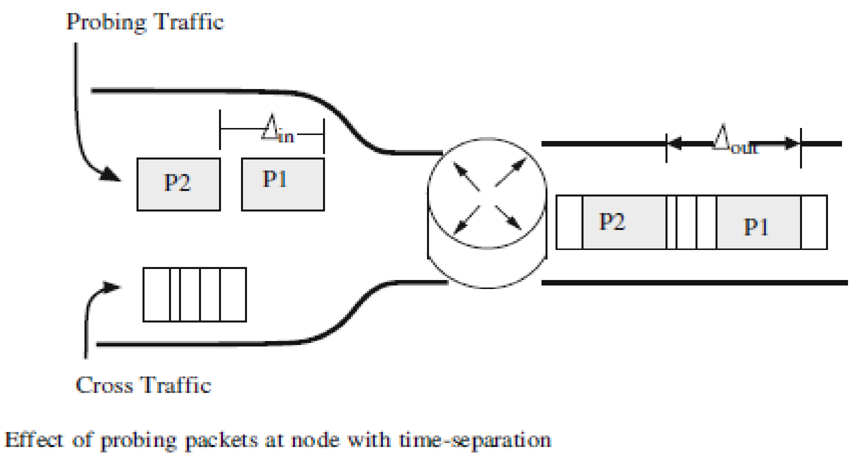


Figure 3.2: Effect of Probing Packets at Node with Time Separation

Self-loading packet-pair active probing: In this packet pair active probing technique, trains of probe packets are iteratively sent into the network at different rate. When compared with a non-iterative packet pair active probing, more probing bits are required which yields a more accurate estimation. The probing bit requirement results in severe intrusiveness and long measurement time. If the sending rate of the probe packet is faster than the available bandwidth, the probe packet will queue, thereby increasing the end-to-end delay (Airon & Gupta, 2017). Delay variation at the receiver is used in estimating the available bandwidth which is the minimum rate of probing that does not saturate the tight link or decide the beginning of a congestion. Protocols that falls within the self-loading packet-pair in literature is as follows:

Self-loading periodic stream (SLoP) (Boz & Manner, 2020), Train of packet pair (TOPP) (Khangura & Fidler, 2017), packet train pair (PTP) (Selin et al., 2011), MinProbe (Wang et al., 2014), self-loading decreasing rate train (SLDRT) (Hu et al., 2012), and probabilistic available bandwidth (PAB) (Thouin et al., 2011).

(Boz & Manner, 2020) proposed a self-loading periodic stream (SLoPS). This protocol estimates the end-to-end available bandwidth path using the probe packet sent with equal packet size. It also measures a one-way delay of the probe packet. This was achieved by sending a packet pair stream at a given rate and carrying out an estimation on the available bandwidth from the interference of self-induced congestion. An initial TCP connection was established between the sender and the receiver. With this connection, the probe packet sender includes the current value of Δin to the receiver. The receiver can then calculate the Δout and send a return message with the value of Δout to the sender. The sender thereafter can determine whether the probe packet is beyond the available bandwidth after having the information about Δout and adjusting the probe rate range. Probe rate is adjusted exponentially to demarcate rough bandwidth ranges, and any constrain to the value of bandwidth. The feature of this protocol includes fewer overhead, speed, more accurate available bandwidth for a short-term decision, and no time-stamp requirement in the packet. Results obtained by using this protocol shows that if there is a fluctuation in the available bandwidth within a short time, there will be increase in the measurement delay, and rise in cost. It cannot gradually adjust its measurement according to the network load. If the stream rate is higher than the available bandwidth, the one-way delay of a periodic stream will show an increase in trend.

(Khangura & Fidler, 2017) proposed a train of packet pair (TOPP). The difference between SLoPS and TOPP is the statistical measurement processing (Khangura & Fidler, 2017). TOPP linearly increases the rate while SLoPS adjust it using a binary search. Another important difference between these protocols is that TOPP can estimate the tight link capacity of a path. This capacity can be higher than the capacity of a path if there are different tight and narrow links. Both SloP and TOPP during measurement can detect changes in the available bandwidth. Both protocols also overload the network path due to self-induced congestion. Both protocols also use a First-in-first-out (FIFO) queuing system along a router path and assumes that the average rate of cross-traffic slowly changes and are constant for a measured duration. TOPP therefore estimates the available bandwidth of a network path by checking the time gap between probe packets using the dispersion technique. This was achieved by iteratively sending a train of packet pair at a gradually increasing rate from source to destination. Rates are thereafter changed by modifying the input gap of each pair. The available bandwidth is estimated as the maximum input rate not higher than the measured rate at the destination. Assuming a packet pair is sent from the source with an initial dispersion Ds and the probe packet size is L byte: therefore, the offered rate of the packet pair will be Ro= L/Ds. If the offered rate is higher than the end-to-end available bandwidth A, the probe packet afterwards will queue behind the one that arrives before it and the receivers measured rate will be R*m*< Ro, otherwise, TOPP assumes that the packet pair will arrive with similar rate it had at the sending node (Khangura & Fidler, 2017).With TOPP, there is an estimation of the hidden bottle neck. TOPP also have a high efficiency without any self-interference. It is network friendly, but accuracy is affected whenever there is a propagation of network error using the theoretical model. Result presented by TOPP shows that it can handle both single hop and multi-hop having different bandwidth and cross traffic.

(Selin et al., 2011) and (Airon & Gupta, 2017) proposed a packet train pair protocol (PTP). This protocol estimates both the link capacity and the end to end available bandwidth of a network with a single measurement that focus mainly on the number of probing bit and the estimation error on the internet, this is achieved by using the impulsive packet probing technique. This technique has a potential to address the iterative probing problem such as increase in the number of probe packet and poor accuracy measurement. Packet train pair makes use of a pair of probe packet train rate of V_1 and V_2, which results in a small number of probing bit. It includes a transmitter generating probe packet trains having multiple packet. The receiver receives those probe trains, measures probe train dispersion, and calculates the available bandwidth. For measurement of accuracy, a technique called sweep was used which updates the estimated available bandwidth value. The technique uses a limited measured result with few probe packets which yields a short time measurement and small number of probing bit, keeping the measured error to a minimum value. This technique works well with a single-hop network. In Packet train pair network, the estimated link capacity in two hop networks only corresponds to a tight link and not a narrow link. Therefore, the available bandwidth measurement of conventional technique requires several measurement cycles for estimating each point of available bandwidth, whereas in packet train pair, only two measurements are required. The available bandwidth is estimated with relatively few errors while the capacity of the link was estimated with lots of error (about 10%), because the link capacity was estimated as the sum of the available bandwidth and cross traffic, load both of which had error in estimation.

MinProbe was proposed by (Wang et al., 2014). This protocol is an active protocol that measures the available bandwidth of a node with high fidelity, minimal cost and in user space.

In (Hu et al., 2012), self-loading decreasing rate train (SLDRT) was proposed. This protocol measures the available bandwidth using a single decreasing rate packet train under a cross-traffic. This was achieved by sending a single decreasing packet train rate with probe packet. These packets are accumulated at the tight link causing the one-way delay of successive packets at the receiver which shows an increase in trend. With the packet-train rate decreasing, the tight link congestion is gradually eliminated, and the one-way delay will show a decrease in trend. Eventually, the one-way delay remains approximately stable when the probe packet trains input rate is the same as the available bandwidth. Self-loading decreasing rate train protocol deduces the available bandwidth by making use of the whole probe packet trains instead of considering the rate of individual packet. By doing this, the bias measurement caused by busty traffic can be efficiently eliminated. In a real network, the one-way delay is interrupted by clock skew between the sender and the receiver. This protocol is said to estimate the available bandwidth in a timely manner, accurately and with less overhead measurement than other existing protocol, but it allows more probe packet within a short interval of time. Results from this protocol is tested under the fluid cross-traffic model and under the non-fluid cross traffic model for both the single hop and multi hop network. Robustness was tested for cross traffic changes during successive measurement, where the amount of cross-traffic remains constant during the process of measurement.

(Thouin et al., 2011) proposed probabilistic available bandwidth (PAB). This protocol estimates the probabilistic available bandwidth of multiple path in a network. This was achieved by determining the highest input rate for which a traffic flow can send to achieve an output rate that is almost equivalent to the input rate with specific probability. PAB executes three tasks; first is to probe a path and present a measured outcome. The strategy of probing should be based on principle of self-induced congestion and selecting the probing rate at every iteration. Secondly is the computation of marginal posterior of the probabilistic available bandwidth path from measurement outcome by running propagation on the factor graph and establish confidence interval for the probabilistic available bandwidth. Finally, it identifies measurement (by choosing the path) at each iteration that minimizes the network overhead. The average amount of measurement and byte per path needed to complete the estimation process is a function of the size of the probe train, which provides a more valid mapping between inferred and measures quantity. When the available bandwidth estimation is done in term of the input and output rates, there will be no need to bridge the gap between packet utilization and dispersion through an invalid general model assumption.

Packet Pair/Train Dispersion active probing: This protocol can also be referred to as direct probing or probe gap model. It is a fast and lightweight available bandwidth measurement. In packet pair/train dispersion active probing, the cross traffic gets in between the probe packet and disperses them as shown in figure 3.2. The available bandwidth estimation is done by mathematically computing the measured sending and receiving gaps between probe packets. There were two assumptions made, which are: 1. Tight and narrow link (bottleneck link) are the same and 2. The capacity C of the bottleneck is known in advance. To minimize the chance that cross traffic disperse the packets further down the path after being dispersed by the bottleneck, lots of packet pair probe is performed and the minimum dispersion is observed. The accuracy of this protocol is typically dependent on cross-traffic relative to the traffic measurement. Though accuracy is achieved in a single queue, it cannot estimate the available bandwidth of a multi-hop path, even if there is only a single bottleneck in that path (Shiobara & Okamawari, 2017). The list of the protocols that falls under packet pair/train dispersion active protocol in literature are; initial gap increase (IGI) (Marttinen et al., 2016), two-way available bandwidth estimation(TWABE) (Li & Chang, 2009), gaps of non-adjacent probe packet (GNAPP) (Li et al., 2013), network link characteristics using packet pair dispersion (NLCPPD) (Dey et al., 2011), adaptive available bandwidth estimation (AABE) (Azevedo et al., 2018), new enhanced available bandwidth measurement technique (NEXT) (Paul et al., 2014), new enhanced available bandwidth measurement technique extension (NEXT-V2) (Paul et al., 2016), new enhanced available bandwidth measurement technique extension with piggybacking (NEXTV2 with piggybacking) (Paul et al., 2016), WBest (Li et al., 2008), and RT-WABEST (Yang et al., 2017), etc.

In (Marttinen et al., 2016) initial gap increase (IGI) was proposed. IGI protocol measures the available bandwidth on a network path by determining the input packet pair gap which produces a high correlation between changes in the packet gap and that of the competing traffic on the bottleneck link. This protocol starts by sending a packet train in sequence with an increasing initial packet gap, to capture the relationship between competing traffic on the bottle neck link and the change in gap between a probe packet pair in a single-hop network. The average input gap and output gap difference for each train is monitored. To obtain the available bandwidth, the estimated competing traffic bandwidth is subtracted from the estimate of the bottleneck link bandwidth. This single-hop model gap is used to understand the relationship between the queue size of router, throughput of competing traffic, and input/output gap. This protocol has a lower error measurement, which helps to accurately estimate the number of competing traffic within the bottleneck router in the single-hop gap model. Thus, accuracy is low for light competing traffic in multi-hop gap model because light competing traffic only increase a small amount of probing gap, and the timing measurement error becomes more significant.

(Li & Chang, 2009) proposed a two-way available bandwidth estimation (TWABE). This protocol estimates the available bandwidth of the up-link as well as the down-link by using the ICMP and traceroute timestamp concept. In this protocol, the ICMP implements the traceroute which is used in calculating the path length from the sender to the receiver. When it gets to the ith round (i= 1, 2, ……., H), an NICMP probe packet with TTL= i is forwarded from the sender to the receiver. All the probe packet has the same size, but the gap between all the successive packets should either be in an increasing format or decreasing format to allow for accuracy in estimation. When a sender receives a returned probe packet, the input and output gaps of consecutive probe packet are computed based on their timestamp, which is used for calculating the capacity C_j and the rate of cross-traffic α_j for tight link j. Therefore, the available bandwidth is $A_j= C_j\alpha_j$. This protocol also works well in packet loss environment, since there is a detection of packet loss by checking the ICMP sequence number of every packet.

In (Li et al., 2013) gaps of non-adjacent probe packet (GNAPP) was proposed. This protocol bi-directionally measures the available bandwidth in a similar way with TWABE. It evaluates the available bandwidth of both the uplink and the downlink tight link path. The application area of most probing techniques does not focus on multimedia network streaming, but only evaluate available bandwidth of a path along one direction, like from the source to the destination. GNAPP makes use of traceroute and modified ping program which employs an ICMP timestamp. Whenever packets from the ICMP return to the client at its local time, the gap between the two-probe packets is calculated by using the time stamp for the available bandwidth estimation. Any two-consecutive probe packet gap remains unchanged if there is no congestion within the link. The accuracy of available bandwidth degrades if the network has a busty cross-traffic. To reduce the impact of two consecutive probe packets from not capturing the cross traffic for accuracy purpose, the non-adjacent probe packet gap, with any consecutive probe packet must be analysed. Moreover, to improve the accuracy of estimation, a moving average must be used with the introduction of a two-stage filtering algorithm. The filtering algorithm helps to reduce the probability of large errors that may be caused by noise estimation. Numerical result on using this protocol shows that the filtering algorithm method can drastically improve the interference accuracy. It is also seen that the reliability of GNAPP estimation technique when the pareto ON-OFF cross-traffic is used is better than when the poison cross traffic is made used of.

(Dey et al., 2011) proposed network link characteristics using packet pair dispersion (NLCPPD). This protocol analyses dispersion of packet based probing technique within a unicast and multicast tree and develops a theoretical model of discrete time queue by considering the characteristics of the link. The method used in NLCPPD allows packet pair probes with a given separation between them to be injected at the source. The probe packet allows a discrete time queue on one path or on the multicast tree. For a single queue with a specific separation between the input probes, the conditional distribution separation between the queue output probes in term of the arrival process distribution is derived. The main result obtained here is the joint distribution of the queue number of arrival and the number of departure from the queue between the slots where the probes are injected. Making use of this, we can obtain the output separation distribution for any given input separation. The joint distribution separation is obtained at all the output for multicast trees. Results show that the available bandwidth estimation fairly works well for two queues serially as well as multicast. By multicasting, the estimation can be improved significantly in the aspect of accuracy and efficiency, even if the depth of the tree is above two.

Adaptive available bandwidth estimation (AABE) was proposed by (Azevedo et al., 2018). This protocol estimates the available bandwidth on a network using active probing with packet pair dispersion with varying overhead. AABE makes use of direct probing in each adaptation period of the video streamer by sending a fixed packet size and a fixed gap size of a single packet train without interrupting the packet transmitted. The packet pair was used in estimating the bottleneck link capacity and IGI formula was used for cross traffic estimation. The cross traffic is deducted from the capacity of the bottleneck link to obtain the available bandwidth estimation. Depending on the probe packet loss rate and the value of the available bandwidth estimate, AABE adjust the number of packets to be used in the packet train of the next period that alleviates the negative effect of active probe packet when there is congestion. Result as presented by the author shows a substantial increase in the performance of streaming with higher perceptual quality.

New enhanced available bandwidth measurement technique (NEXT) was proposed by (Paul et al., 2014). NEXT is a probing scheme and a rate adjustment algorithm used in estimating end-to-end available bandwidth on a network part. Its concept is based on self-inducing congestion and a probe train structure that allows a packet to be frequently sampled on a given region than another region. The highest sample region allows the algorithm to find a more accurate turning point. Whenever the dynamic available bandwidth is outside the highest sampled region, the lower and upper packet stream rate is readjusted to fit the dynamic available bandwidth into the region. The spread factor is used in adjusting the range between the lower and the upper rate to keep the packet number less before measuring the available bandwidth intrusively.

(Paul et al., 2016) proposed a new enhanced available bandwidth measurement technique extension (NEXT-V2) which is an active protocol. This protocol was said to be an extended version of NEXT that effectively measures the end-to-end available bandwidth within a fixed wireless network. The structure of this protocol is more like a packet train with an optimal rate adjustment and a modified excursion detection algorithm which is used to identify the available bandwidth with higher accuracy, less overhead and less convergence time.

New enhanced available bandwidth measurement technique extension with piggybacking (NEXTV2 with piggybacking) was proposed by (Paul et al., 2016). In this protocol, estimation is based on a proxy technique that conveys application data by piggybacking inside the probing packets, resulting in fewer overhead.

(Li et al., 2008) proposed a wireless bandwidth estimation tool (WBest). This protocol has a two-stage algorithm, namely, packet pair technique that estimates the capacity of a link where the last hop is the wireless LAN (WLAN) and a packet train technique that estimates the achievable throughput in order to know the available bandwidth. The parameter of WBest are optimised with trade-off of accuracy, convergence time and intrusiveness. WBest avoids using a search algorithm to detect the available bandwidth by statistically detecting the available fraction of the effective capacity to limit the estimation delay and impact of random wireless channel error.

Round-trip wireless available bandwidth estimation tool (RT-WABest) was proposed by (Yang et al., 2017). It was designed based on round trip time measurement with two stage algorithms. The first stage employs packet pair dispersion technique to estimate path capacity and the second stage sends a packet train to infer available bandwidth.

Reactive packet-pair active probing (RPPAP): This is another available bandwidth estimation technique. In this technique, for a reactive bandwidth measurement to be activated, a probe packet is sent from the source node to the destination node. If the source node does not receive the acknowledged probe packet before the time out period, the source node resends the probe packet to the destination node again. Once the destination node receives the probe packet sent by the source node, the destination node acknowledges the probe packet by periodically sending series of back to back probe packet on all the available paths to the source node. The packet, thereafter, travels along the path from the sending node to the receiving node and produces gaps between them. Those gaps are measured at the receiving node and the receiving node estimates the amount of probe packet lost because a unique sequence number identifies each probe packet. All the raw gaps and the estimated amount of lost probe packet feed into a filtering module removes the biggest and smallest gaps. Gap is calculated as the mean of the remaining gaps used in calculating the available bandwidth. The following protocols are identified to exist under RPPAP which will be discussed in this study: Bandwidth available in real-time (BART) (Jansen et al., 2018), multi-rate available BE in real-time (MR-BART) (Sedighizad et al., 2012), minimal backlogging techniques (MiBT) (Poorzare et al., 2018), distributed admission control for MANET environments (DACME) (Appandairai & Kannan, 2020), and reactive bandwidth measurement in 802.11 networks (RBM) (Shah et al., 2018).

In (Jansen et al., 2018), bandwidth available in real time (BART) was proposed. This protocol enables the end to end available bandwidth of a network path to be continuously estimated based on Kalman filters (KF) bandwidth tool. This protocol relies on congestion that are self-induced. The available bandwidth of the network path is continually sampled with a sequential probe packet sent randomly. A current estimate is maintained and increases with new inter-packet time separation measurement in a probe packet sequence. If there is less rate of probing sequence compared to the available bandwidth, no measurement will be performed on the new available bandwidth. Result gotten in (Jansen et al., 2018) shows that the accuracy of BART is linear with the probe train length and probe packet sizes. Kalman filters promises a fast, reliable, and efficient available bandwidth estimation.

(Poorzare et al., 2018) proposed minimal backlogging technique (MiBT). This protocol estimates the available bandwidth using the statistics of the probing traffic service rate. MiBT avoids the usage of probe gap model and probe rate model. The probing traffic service rate statistics is a consistent estimator of the available bandwidth for a G/G/1 queuing system under a minimal backlogging condition which support MiBT theoretically. For MiBT to be emulated in a real multi-hop network, the minimal backlogging condition or probing rate closer to the available bandwidth based on the length of the busy period is detected. In order to maintain a minimal backlogging condition, the probe rate adaptively changes. A higher level of accuracy of minimal backlogging condition is detected by the busy period of probe packet when compared with the rate response curve or gap response curve. The estimation of the initial probing rate mechanism decreases the gap between the available bandwidth and the initial probing rate. By using the variance and mean of the available bandwidth estimation, a range of available bandwidth for a short interval of time can be obtained.

In (Appandairai & Kannan, 2020), a distributed admission control for MANET environments (DACME) was proposed. This protocol accesses the end-to-end bandwidth path as well as the end-to end delay and jitter within a MANET. Different probes are used to assess the available bandwidth, delay and jitter of a given end-to-end path. A probe packet is sent through a path from a source to access that path. A generated back-to-back probe packet is followed by a reply of probe to the destination. The source agent keeps a timer to detect any probe reply loss. The destination node upon receiving all the probe packet is expected to send a single packet reply with a measured estimated available bandwidth. Result shown by this protocol establishes that DACME offers a new framework for QoS support in MANET which is based on media access control (MAC). The probe set size is a reasonable choice that offers a good balance between the accuracy of available bandwidth and the time for admission control.

(Shah et al., 2016) proposed a reactive bandwidth measurement in 802.11 networks (RBM). In RBM, the available bandwidth is derived reactively from the measured gap between two probe packets at the destination node of the 802.11x network. This protocol is a reactive bandwidth measurement that allows nodes to send probe packets to the destination for activation of the measurement process, and the destination responds by sending lots of probe packet back to the source serially and in a regular interval. For the first successful probe packet to be sent, the source needs to send, the request to send (RTS) to the next hop node before sending of the clear to send (CTS), DATA, and acknowledgement (ACK). The second successfully sent probe packet may immediately follow the first but with delay existing between those two packets especially when there are other nodes using the medium or during contention within the medium. The total time of transmitting the probe packet from a source to its next hop node has a transmitting time RTS (t_{rts}), CTS (t_{cts}), the probe packet itself (t_{data}), and ACK (t_{ack}) for the processing time and probe packet. Therefore, the gap between the first probe packet and its next is GAP= $t_{ack} + t_{othernodes} + t_{rts} + t_{cts} + t_{data} + t_{processingtime}$.

Hybrid packet-pair active probing: The protocol that falls under hybrid packet pair active probing is known as PATHCOS++ (Lin et al., 2010). This protocol integrates the probe rate model and the probe gap model.

In (Lin et al., 2010) PATHCOS++ was proposed. This protocol estimates the available bandwidth of an end to end path by sending a train of time stamp probe packet from a source node to the destination node and integrating the advantages of probe rate model and probe gap model-based techniques. It consists of self-induced congestion mechanism packet gaps for available bandwidth estimation. Changes in one-way delay of the probe packets are monitored by the receiver and analysis are conducted based on the mechanism. This protocol extends the analysis of queuing behaviour of probe packet from single hop scenario to multi-hop scenario. It sends probe packets with rates controlled by a *cos* function and find big bumps in the response from probing curves for available bandwidth to be observed without information on the bottleneck link capacity. This technology does not make fluid cross traffic assumption. PATHCOS++, as described by (Lin et al., 2010), is efficient in providing end-to-end available bandwidth accuracy information. Its accuracy is almost not affected when there are multiple congestion links.

Therefore, according to (Aina et al, 2019), for any wireless ad-hoc network, the active measurement technique is not ideal due to the following reasons:

- In an active estimation technique, probe packet is used to measure the available bandwidth between the source and the destination. If the number of source and destination node is large, there will be many probe packets sent within the network end to end pair, therefore requiring a large amount of bandwidth to be used.
- Due to the time varying nature of wireless links, the network topology is not stable when compared to wired link topology. Therefore, the active bandwidth estimation will have to conduct its estimation at a higher frequency, resulting in extra bandwidth usage.
- The active bandwidth estimation introduces extra overhead, affect the accuracy, and degrades the network performance of the bandwidth estimation. Therefore, the active bandwidth estimation approach is not the best choice for measurement in wireless networks.

3.2.2. Passive Bandwidth Estimation Technique

In (Mukta & Gupta, 2019), passive estimation is referred to as a calculation-based technique. Passive bandwidth estimation technique does not inject any probe packet into the network when estimating the required available bandwidth. Dispersion and delay are used to observe the acknowledgement and data flow and probe packet are not made use of. This form of estimation works with earlier generated information traces collected. The local information on bandwidth utilization is used for calculating the available bandwidth and exchanged via local broadcast. Passive bandwidth estimation can be divided into two, namely; generic passive technique

(GPT) and proactive passive technique (PPT). A diagram showing the subdivision of passive protocol is shown in figure 3.3 below:

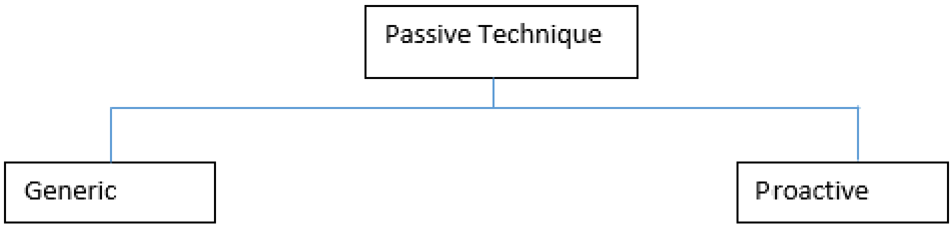


Figure 3.3: Passive Technique

3.2.2.1. Generic Passive Technique (GPT): This protocol requires a probability distribution function (PDF) of a TCP flow packet interarrival time (Liao et al., 2017). The PDF shows the behaviour of spike, spike bump, spike train and train of spike bumps. The characteristics of GPT is described as a bottleneck having no cross-traffic. The three GPT protocol that will be discussed in this section are: estimation of available bandwidth ratio of a remote link or path segments (EABRRL) (Nam et al., 2013), TCP Vegas (TCPV) (Humeida & Nilsson, 2018), and TCP Westwood (TCPW) (Al-Hasanat et al., 2017).

In (Nam et al., 2013), estimation of available bandwidth ratio of a remote link or path segments (EABRRL) was proposed. This protocol estimates the bandwidth ratio at a remote path or link without deploying it at the remote node. EABRRL estimates the delay for a segment path remotely by monitoring the node. Here, two ICMP timestamp streams of packets are sent to both end nodes of a target link in accordance with Poisson process. The one-way delay is measured from the packet sending time difference and the value of timestamp received from remote nodes. It also extracts the queuing delay component from the delay measured and estimates the available bandwidth product ratio of a given path on a link. The available bandwidth is thereafter inferred from the ratio of the available bandwidth products. No condition on the link ratio rate of consecutive link is required. It estimates the available bandwidth ratio of the link beyond the tight link on a given path without overloading any network link. This protocols intrusiveness is very low in the sense that it doesn’t incur any short-term congestion. It can overcome conventional approach limitation such as inability to probe the link beyond the tight link with the minimum available bandwidth. Result shows that the available bandwidth ratio measured value lies within the standard deviation from the average estimation values in most link. When far links are probed, there is an increase in the variance, but the standard deviation is still very small when compared with the average.

(Humeida & Nilsson, 2018) proposed TCP Vegas (TCPV). This protocol estimates the bandwidth as an active throughput during the internet connection time. It computes the difference between the expected flow rate (denoted as cwnd/RTT_{min}) and the actual flow rate (denoted as cwnd/RTT). RTT_{min} is the minimum round-trip time measured by the TCP source, and cwnd denotes the congestion window w size. Based on observation, this protocol adjusts the value of the round-trip time. The actual flow rate tends to be closer to the expected flow rate when we have a non-congested network, but with a congested network, the actual rate tends to be smaller than the expected flow rate. Results from this protocol shows that it achieves 37-71% better throughput. In a homogenous network scenario, it does not achieve fairness because competing connections can converge to different cwnd value of parameter.

(Al-Hasanat et al., 2017) proposed TCP Westwood (TCPW). This protocol estimates the available bandwidth by measuring the acknowledgement rate. The available bandwidth measurement is done by allowing the source node along a TCP path to estimate the measured bandwidth and find the average rate of the returned acknowledgement. The sequence of the bandwidth sample obtained is considered by using the acknowledgement arrivals from which the smoothed value is evaluated using low-pass filtering on the sequence of samples. This kind of bandwidth estimation has been used indirectly before for controlling the TCP window, via the bottleneck backlog estimation. After a consecutive congestion period, the source uses the measured bandwidth to properly set the congestion window, set the slow start threshold and thereafter start a faster procedure recovery. When the source receives an acknowledgement, it specifies that certain amount of data having a specific packet transmission is delivered to the destination. If the transmission process is not affected by any loss, by calculating the average data delivered at a specific time, gives a fair estimation of the current bandwidth used by the source node. In case there is a duplicate acknowledgement received by the source node which indicates an out-of-sequence reception, it is also counted towards the bandwidth estimation and a new estimation is computed after their reception. Result from using this protocol shows that it is not very sensitive to random errors and it is very effective in mixed wired and wireless networks.

3.2.2.2. Proactive Passive Technique (PPT): Proactive passive technique is non-intrusive because there is no frequent exchange of HELLO packet. This protocol only considers the

MAC overhead when calculating the available bandwidth. The available bandwidth is used for the selection of network in heterogeneous network environment. Parts of the protocols that comes under PPT are: Available Bandwidth Estimation (ABE) (Sarr et al., 2008), improved available bandwidth (IAB) (Zhao et al., 2009), cognitive passive estimation of available bandwidth (cPEAB) (Tursunova et al., 2010), accurate passive bandwidth estimation (APBE) (Park & Roh, 2010), distributed available BE (DABE) (Yang & Zhang, 2012), proactive bandwidth estimation (PABE) (Farooq and Kunz, 2013), passive available bandwidth estimation (PABE) (Zhao et al., 2014), distributed LaGrange interpolation based available bandwidth estimation (DLI-ABE) (Chaudhari et al., 2014), available bandwidth estimation method for IEEE 802.11 ad-hoc network with concurrent transmissions (ABCT) (Lei et al., 2015), measured based bandwidth estimation technique and flow admission control (BandEst) (Farooq and Kunz, 2015), cognitive agent based available bandwidth estimation using collision probability, idle period synchronization and random waiting time (BECIT) (Chaudhari & Biradar, 2015), MBA-AODV (Sharma et al., 2018), analytical available bandwidth estimation including mobility (AABWM) (Mukta & Gupta, 2019).

(Sarr et al., 2008) proposed an available bandwidth estimation (ABE) algorithm for wireless network. Estimation of the available bandwidth is done by using the wireless channel sensing mechanism, where consideration is given to both the virtual and physical carrier sensing, and different types of wireless CSMA/CA MAC layer interframe spacing. They argued that measuring the channel activities by considering the amount of time spent in the physical and virtual carrier sensing with different interframe space results in the over-estimation of the available bandwidth. This happens because of the non-synchronization between the sender and the receiver within an ad-hoc network. The authors therefore present a mathematical model that considers the collision probability to estimate the actual available bandwidth. It considers the future back-off overhead through mathematical model. The collision probability is derived from the amount of HELLO messages received by a node over the amount of HELLO packets expected to be received by a node at the previous interval measurement. The admission control flow algorithm makes use of one-hop neighbour and two-hop neighbour information to calculate the intra-flow contention when the maximum intra-flow contention is equal to four. To calculate the inter-flow contention, the minimum available bandwidth within the interference range is determined and used to decide the flows admission request. The drawbacks of this technique are; firstly, if there is an increase in data traffic load within a network, the only consideration is the additional back-off overhead, other important factors like additional retransmission and contention window overheads are ignored. Secondly, the intra-flow contention count calculation does not always provide a right contention count, and the inter-flow contention count appears as been too simple, since it only considers the minimum available bandwidth within the interference range of a node. Lastly, the collision probability is calculated without considering the hidden and exposed node.

(Zhao et al., 2009) proposed an improved available bandwidth (IAB). The improved available bandwidth estimates the available bandwidth of a giving link for QoS in wireless ad-hoc network. It considers synchronization between the source and the destination node by differentiating the busy state caused by the transmitting and receiving nodes from those caused by the sensing nodes. It also improve the accuracy of estimating the overlapping probability of the idle channel time of two adjacent nodes. For a node to be termed BUSY, it must be in either a receiving state or transmission state. A node is termed SENSE BUSY when it is in a sensing state. If the node is not in any of these states, it means the node is IDLE. The drawback of this technique is similar to those mentioned in (Sarr et al., 2008).

Cognitive passive estimation of available bandwidth (cPEAB) was proposed by (Tursunova, 2010). This protocol estimates the available bandwidth of a network in an overlapped WiFi WLANs environment. It considered the additional overhead caused by acknowledgement frames, therefore estimating the available bandwidth by measuring the proportion of waiting and back off delay, packet collision probability, acknowledgment delay, and channel idle time. Furthermore, cPEAB considered the hidden and exposed node to have a more accurate available bandwidth measurement. The drawback of this proposed algorithm is that the intra-flow contention count calculation does not always provide a right contention count. Retransmission and contention window overheads were also ignored in this proposed algorithm. Additionally, to retrieve the available bandwidth on a carrier sensing, HELLO packet is broadcasted to two hop neighbour which tends to flood the network to increase the network overhead. Lastly, the dependency of the channel idle time ratio only differentiates between the *BUS*Y and *SENSED BUSY* and did not consider an empty queue to be an idle channel time period. We define the *BUSY* state as a situation whereby a node is in the state of transmission or receiving. The *SENSE BUSY* state is defined as a situation whereby a node is in the state of sensing. Any other time apart from the sensing time, the node will be in an *IDLE* state. The *IDLE* state means that the node is neither transmitting, receiving nor sensing any packet. For a channel to be idle, the channel does not necessarily have to be sensed idle by both physical and virtual wireless carrier sensing mechanism, but, the interface queue must be empty.

(Park & Roh, 2010) proposed an accurate passive bandwidth estimation (APBE). In APBE, the available bandwidth is estimated by considering request to send (RTS) and clear to send (CTS) overheads. It measures correctly the proportion of DIFS and back-off, the packet collision probability, the acknowledgment delay and the channel idle time. It is calculated as:

R/C_{ohd} = ((RTS + CTS) + (2 X SIFS))/T, if RTS/CTS is used otherwise, it is considered as 0. SIFS therefore represents the short inter-frame space. To estimate the available bandwidth, ABW= (1-K) X (1-R/C) X (1-ACK) X (1-P_c) X (T_i/T) X C.

Where K= proportion of bandwidth consumed during the waiting and back-off time, P_c = packet collision probability from hidden and exposed node, T_i = channel idle time of the wireless medium in a measurement period T, C= maximum channel capacity. Results from using APBE shows a higher accuracy, while its error is lower when compared with cPEAB.

distributed available BE (DABE) was proposed by (Yang & Zhang, 2012) for admission control available bandwidth measurement. It uses a total busy period that includes the frame intervals, transmission time, and back-off duration for channels within the monitoring period in a distributed manner. The drawback of this scheme is that the contention window overhead was not considered with increase in data traffic load inside the network. Also, the assumptions made in the mathematical model may not hold through in the actual network.

(Farooq and Kunz, 2013) proposed a proactive bandwidth estimation (PABE) for IEEE 802.15.1-based network. PABE is a measurement-based enhancement for available bandwidth estimation method and flow admission control algorithm. Instead of using a model to predict the collision and back-off, empirical data gathering was adopted for predicting any additional back-off overhead. Besides, it used the value of the expected future data traffic load to predict additional overhead instead of using the existing one. The drawback of this algorithm is that, if there is an increase in data traffic load within a network, additional retransmission and contention window overheads are ignored. Also, the intra-flow and inter-flow contention count is wrongly calculated. Lastly, to retrieve the available bandwidth on a carrier sensing, HELLO packet is broadcasted to two hop neighbour which tends to flood the network, therefore increasing the network overhead.

In (Zhao et al., 2014), passive available bandwidth estimation (PABE) was proposed. In this protocol, the effectiveness of the link capacity is considered by analysing the random factor in transmission such as back-off time and the frames retransmission. For the channel idle time ratio to be estimated, a new lower threshold is introduced.

(Chaudhari et al., 2014) proposed distributed LaGrange interpolation based available bandwidth estimation (DLI-ABE). In this protocol, the channel idle time synchronisation uses the actual channel utilization and collision rate. Also, the collision probability model uses a separate Lagrange interpolation polynomial at each node; depending on the behaviour of node.

Available bandwidth estimation method for IEEE 802.11 ad-hoc network with concurrent transmissions (ABCT) was proposed by (Lei et al., 2015). This protocol focused on estimating available bandwidth of a medium using the control-gap based concurrent transmission.

Measured based bandwidth estimation technique and flow admission control (BandEst) was proposed by (Farooq and Kunz, 2015). This protocol proactively considers the complete wireless 802.15.4's unslotted CSMA-CA MAC layer overhead and considers the future load. It also considers the estimation of intra-flow contention and estimates contention on non-relaying nodes. Additional MAC layer overhead that is associated with increased data traffic load was considered and an algorithm that deals with concurrent admission request in FIFO was implemented. The drawback of BandEst is that it has a higher overhead because it broadcast to two-hop. BandEst does not also consider the channel idle time dependency. The effect of hidden/exposed collision node on the accuracy of bandwidth estimation has also been neglected by this protocol.

cognitive agent based available bandwidth estimation using collision probability, idle period synchronization and random waiting time (BECIT) was proposed by (Chaudhari & Biradar, 2015). In BECIT, the author adopted an intelligent software agent known as cognitive agent (CA) to estimate the available bandwidth. The author stated that the intelligence is provided similar to the logical human being thinking ability for cognitive agent decision making. The technique uses CA at each node to create mobile agents for the collection and distribution of statistics. The collected statistics are then used by the CA for bandwidth estimation using the distributed LaGrange interpolation estimation. The drawback of BECIT is the estimation process of the channel idle time which tend to affects the bandwidth estimation accuracy. Surrounding nodes in BECIT are assumed to be a random nodes this tends to affect the synchronization process. Also, to retrieve the available bandwidth on a carrier sensing, HELLO packet is broadcasted to two hop neighbour which tends to flood the network, therefore increasing the network overhead.

(MBA-AODV) was proposed by (Sharma et al, 2018). In MBA-AODV, the author proposed an approach for estimating bandwidth by considering node mobility. They highlighted that node mobility usually result in frequent link failures as well as retransmissions. To tackle this they made use of a reactive route discovery process which discovers end-to-end path from the source to the destination node as well as intermediate node route discovery to provide for QoS. The drawback of MBA-AODV is the estimation process of the channel idle time which tend to affects the bandwidth estimation accuracy. Surrounding nodes in MBA-AODV are assumed to be a random nodes this tends to affect the synchronization process. Also, to retrieve the available bandwidth on a carrier sensing, HELLO packet is broadcasted to two hop neighbour which tends to flood the network, therefore increasing the network overhead.

analytical available bandwidth estimation including mobility (AABWM) was proposed by (Mukta & Gupta, 2019). The author stated that node mobility results in link instability which leads to loss of data and delay that have an impact on the available bandwidth. However, they proposed an analytical approach for the estimation of the available bandwidth on a link. The contribution of AABWM are; it uses mathematical model based on renewal theory to estimate the collision probability of packets to enhance simplicity, mobility consideration under 3-D space for predicting link failure and for admission control purpose. The drawback of AABWM is the estimation process of the channel idle time which tend to affects the bandwidth estimation accuracy. Surrounding nodes in AABWM are assumed to be a random nodes this tends to affect the synchronization process. Also, to retrieve the available bandwidth on a carrier sensing, HELLO packet is broadcasted to two hop neighbour which tends to flood the network, therefore increasing the network overhead.

Having reviewed the state-of-the-art passive available bandwidth, it has been observed that the channel idle time dependency sensed by the sender and the receiver has not been properly addressed, as most previous work did not consider it. The related works that considered the channel idle time only considered node independency where all nodes are randomly distributed, this may however affect the accuracy of the bandwidth estimation. This thesis considers the dependency between the source node and the destination node channel idle time and addresses the *BUSY* state, *SENSE BUSY* state and the *IDLE* state caused by an empty queue which has not yet to be addressed. Also, the available bandwidth for admission control broadcasts to two hop neighbours in order to retrieve the available bandwidth on a carrier sensing region. This tends to create a higher overhead that can possibly be avoided.

Year	Protocol	Active	Passive	BWE at Path	BWE at Link	BWE at Node	Accuracy	Overhead	Innovations
2008	WBest (Li et al., 2008)	✓		✓	✓		Medium	Medium	It makes use of packet dispersion model in order not to depend on search algorithm for available bandwidth measurement. It therefore, statistically measures the relative available fraction of the effective capacity, mitigation estimation delay and effect of wireless channel error.
	ABE (Sarr et al., 2008)		✓	✓	✓	✓	Medium	Low	Considered various retrieval range as well as making use of the back-off, waiting time, channel idle time measurement period, maximum capacity, and collision probability during the estimation process.
2009	TWABE (Li et al., 2009)	✓		✓	✓		Varies	Medium	Estimation that considers the uplink, downlink, and packet loss environment.
	IAB (Zhao et al., 2009)		✓		✓	✓	High	Low	It differentiates the busy state of a channel caused by transmission from the sense busy caused by the carrier sensing.

2010	PATHCOS ++ (Lin et al., 2010)	✓		✓			High	Varies	Provides for self-induced congestion mechanism as well as packet gaps.
	cPEAB (Tursunov a et al., 2010)		✓	✓	✓	✓	Varies	Low	Additional overhead caused by acknowledgement frame was considered which was not considered in ABE and IAB.
	APBE (Park and Roh, 2010)		✓	✓	✓	✓	High	Low	Its innovation is similar to cPEAB, but it has an addition of RTS and CTS.
2O11	NLCPPD (Dey et al., 2011)	✓		✓	✓		High	Varies	It considers the conditional distribution of separation between the output probe of a queue.
	PTP (Selin et al., 2011)	✓		✓	✓		High	Medium	The available bandwidth of a path and link are computed at a given time using probe rates and time dispersion.
	PAB (Thouin et al., 2011)	✓		✓			High	Varies	Makes use of weighted entropy/weighted confidence interval on multipath with specific rate of probing.
	QBEM (Ali and Zafar, 2011)		✓	✓	✓		High	Low	Uses HELLO packet available bandwidth from the ratio of busy and free time.

2012	MR-BART (Sedighizad et al., 2012)	✓		✓	✓		High	Varies	It estimates the bandwidth using inter-packet strain and Kalman filters.
	SLDRT (Hu et al., 2012)	✓		✓			High	High	Available bandwidth measurement was done using a stable one-way delay.
	DABE (Yang & Zhang, 2012)		✓		✓		High	Low	It uses a total busy period that includes the frame intervals, transmission time, and back-off duration for channels within the monitoring period in a distributed manner.
2013	GNAPP (Li et al., 2013)	✓		✓	✓		High	High	It is used in both the uplink and downlink and has a novel two stage filtering algorithm for improving the accuracy of the estimation.
	EABRRL (Nam et al., 2013)		✓	✓	✓		High	Low	Queuing delay are computed using two streams of ICMP timestamp packet.
2014	MinProbe (Wang et al., 2014)	✓		✓	✓		High	Medium	It measures the available bandwidth with high fidelity, minimal cost and in user space.
	PABE (Zhao et al., 2014)		✓	✓	✓	✓	High	Medium	The effectiveness of the link capacity is considered by analysing the random factor in transmission such as back-off time and the frames retransmission. For the channel idle time ratio to be estimated, a new lower threshold is introduced.

	DLI-ABE (Chaudhari et al., 2014)		✓	✓	✓	✓	Medium	Low	The channel idle time synchronisation uses the actual channel utilization and collision rate. Also, the collision probability model uses a separate Lagrange interpolation polynomial at each node depending on the behaviour of node.
2015	ABCT (Lei et al., 2015)		✓	✓	✓	✓	Medium	Low	It focused on estimating the availability of a medium using the control-gap based concurrent transmission.
	BandEst (Farook and Kunz, 2015)		✓	✓	✓	✓	High	Low	Proactively considers the complete wireless 802.15.4's unslotted CSMA-CA MAC layer overhead as well as the future load. Estimation of intra-flow contention is carried out on non-relaying nodes.
	BECIT (Chaudhari and Biradar, 2015)		✓	✓	✓	✓	Medium	Medium	Intelligent software called cognitive agent (CA) was used to estimate the available bandwidth. The CA is used for data collection as well as distribution of statistics.
2016	NEXT-V2 (Paul et al., 2016)	✓		✓	✓		Medium	Medium	It has an optimal rate adjustment and a modified excursion detection algorithm is used to identify the available bandwidth with higher accuracy, less overhead and less convergence time.
	NEXTV2 with piggybacking (Paul et al., 2016)	✓		✓	✓		Medium	Low	Estimation is based on a proxy technique that conveys application data by piggybacking inside the probing packets resulting in fewer overhead.

2017	RT-WABest (Yang et al., 2017)	✓		✓	✓		Medium	Low	It was designed based on round trip time measurement with two stage algorithms. The first stage employs packet pair dispersion technique to estimate path capacity and the second stage sends a packet train to infer available bandwidth.
2018	MBA-AODV (Sharma et al., 2018)		✓	✓	✓	✓	Medium	Medium	The bandwidth estimation during node mobility was studied. The author estimated the bandwidth of a mobile node using the reactive route discovery process for the discovery of end-to-end path from the source to the destination node as well as the intermediate node route discovery to enhance QoS.
2019	AABWM (Mukta and Gupta, 2019)		✓	✓	✓	✓	Medium	Medium	Analytical approach was used for the estimation of available bandwidth on a link. This was carried out by using mathematical model based on renewal theory to estimate the collision probability. Also, mobility was considered under 3-D space to predict link failure and for admission control purpose.

Table 3.1: Summary of Some Bandwidth Estimation Technique for Admission Control in MANET (Aina et al., 2018)

3.3. Admission Control in MANET

For QoS to be provided to applications or flows in a network, a group of mechanism that supports admission control, scheduling, resource reservation, classification, policing, or shaping is used. Admission control, which is the subject of the research, is used for controlling the amount of traffic permitted in the network. Admission control were used in the public switched telephone network (PSTN) to establish a call by deciding whether to accept a call (when there are enough available resources in the network) or reject (when there are no available resources within the network) it. In our admission control implementation, our objective is to determine if the network has enough resources for any incoming request. This thesis also determines if a traffic class is accepted or rejected based on the traffic class prioritization utilized.

In a packet switch network, the admission control is used to provide QoS through integrated services (Intserv) (Braden et al., 2018) and Differentiated services (DiffServ) (Blake et al., 2018). The integrated service is based on per-flow reservation in the network for a per-flow QoS to be guaranteed. The Intserv requires the maintenance of individual flow state in the router and its signalling complexity increases with the flow number. Therefore, the admission control decides if a flow is accepted or not. Diffserv, on the other hand, relies on packet markers and, policing at the edge of the router. In Diffserv, the admission control may not be used but its deployment is recommended to control real-time traffic at the ingress node (Budka et al., 2016) (i.e. for traffic classified with an expedited forwarding per-hop-behaviour).

Intserv and Diffserv are not often used in the implementation of IP global network since the internet is composed of multiple AS (autonomous system) with different network administration. Therefore, this implementation is only applied in a set of networks under a unique administration.

The required QoS can also be achieved by using the overprovisioning technique used in the deployment of resources that is enough to handle all the estimated offered traffic. Though the overprovisioning technique allows for network simplicity maintenance, it does not provide the desired QoS level in scenario that involves network congestion or “greedy application” scenario (application that always consumes more network resources that are available). Therefore, the overprovisioning technique cannot provide a guaranteed QoS.

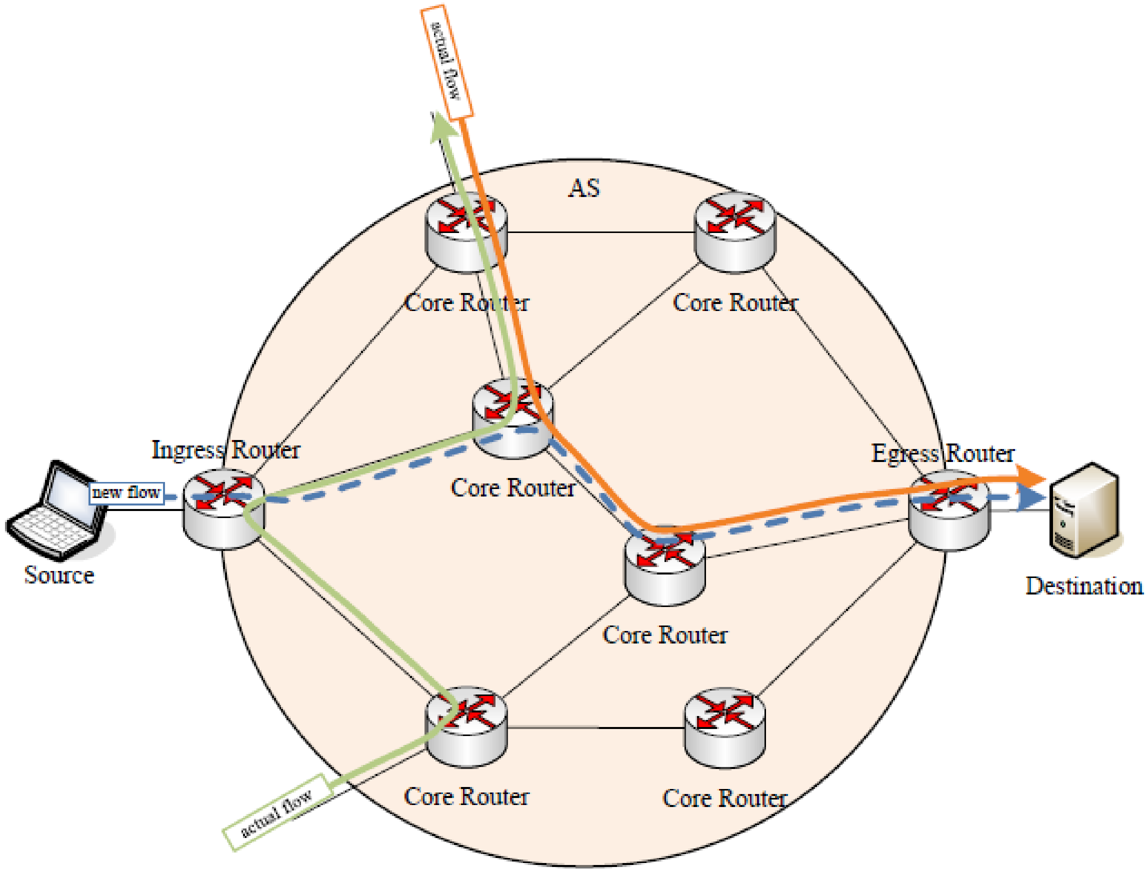


Figure 3.4: Network Topology for Admission Control Provision (Pinto, 2015)

Figure 3.4 presents an example of the topology implemented with an admission control in the AS area. In this topology, a source node flow starts a new flow which is intended to be delivered to the destination node. The new flow enters the AS area through the ingress router. The AS area is made up of one admission control mechanism. The admission control mechanism checks if the network can admit the new flow without disrupting the services already assured. For this decision to be made, admission control considers the characteristics of the traffic and the QoS requirement of the new flow and of the flow along the destination path. This decision is valid through all the path from the ingress router down to the egress router. The ingress router is thereafter given the order to admit or deny the new flow towards the identified destination. An admission controller can make a wrong decision, i.e. to accept a flow that does not meet with the necessary requirement or the admission control mechanism admitting a flow without having enough resources that will accommodate that flow. This decision is referred to as a false negative acceptance.

Describing admission control is based on the location where the admission decision is implemented. This can be categorized into two namely: centralized admission control and distributed admission control. The centralized admission control assumes a unique identity (i.e. one of the core routers shown in figure 3.4). It performs the admission control decision and exchange of signalling with the ingress node at the arrival of a flow. The distributed admission control on the other hand assumes that the decision is performed at multiple points in a distributed manner (i.e. all the routers in figure 3.4 are embedded with an admission control mechanism).

The centralized admission control mechanism assumes the up-to-date complete knowledge of the resource usage and the network topology using a unique identity. In (Oshiba et al., 2016), a centralized admission control was proposed, and decisions were taken based on end-to-end delay in a wireless mesh network. Centralized admission control mechanism, however, may not be adequate for a large and dynamic network because the unique identity will need to process higher information and bottlenecks, which may result in a single point of failure.

The distributed admission control mechanism avoids a single point of failure and the concern posed by centralized approach. Since the distributed admission control is made up of multiple admission control decision points, they may have different views of the resource occupancy and different decisions may be taken for flows that are competing for the same resources. These different decisions may result in QoS violation and inefficiency of resource usage.

Considering the different characteristics and limitations posed by centralized and distributed admission control mechanism, the distributed admission control mechanism appears to be more adequate for this thesis, therefore, we focus on using the distributed admission control mechanism.

3.3.1. Distributed Admission Control

In figure 3.5, the classification of distributed admission control mechanism as proposed in literature is depicted. The distributed admission control has been classified into two groups according to their operation in AS. These are: Edge-to-Edge and Hop-by-Hop operation. Similar classification can be found in (Chromy et al., 2017).

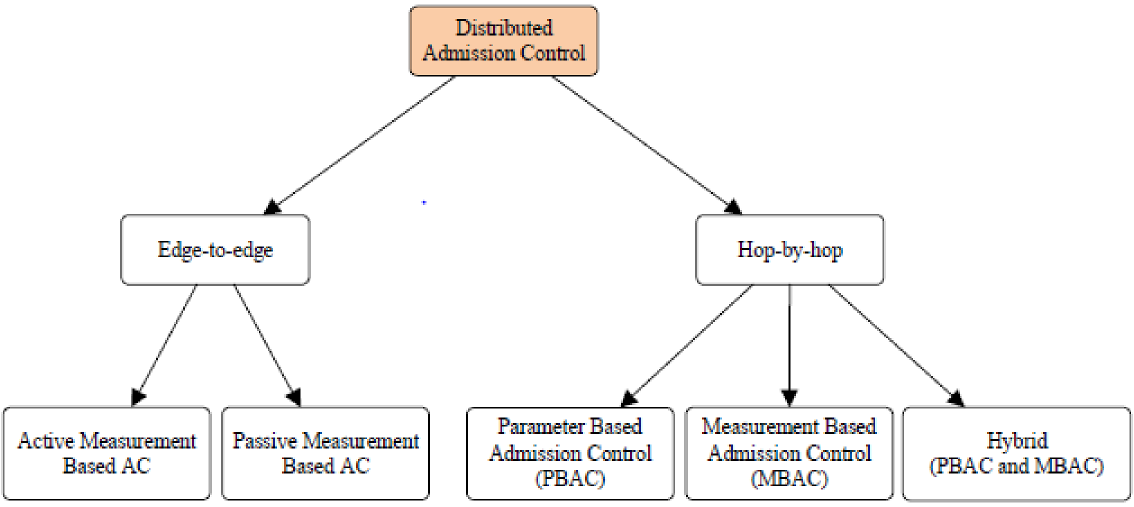


Figure 3.5: Distributed Admission Control Overview (Pinto, 2015)

In the Edge-to-Edge operation, only the ingress and the egress nodes can make admission control decision. The admission control decision is taken on the egress node based on measurement and the decision is sent back to the ingress node (see figure 3.4). Therefore, only these routers exchange control plane packets (signalling or measurement), and the decision made is valid to the entire path from the ingress router to the egress router. One of the advantages of the Edge-to-Edge operation is that the intermediate node does not have to maintain a given reservation state since they do not partake in the admission control decision. The Edge-to-Edge operation mode can therefore be further divided into active measured-based admission control and passive measured-based admission control. The active measurement based uses a probing flow to test the entire path in order to provide an admission control decision for the egress node. In the passive measured based admission control, the QoS aggregate of the accepted flow is measured continually at the ingress and used to provide an admission decision at the egress node.

The Hop-by-Hop allows all the nodes to participate in the admission control decision. Each of the AS router (as shown in figure 3.5) implements an independent admission control mechanism that takes a local decision about a new flow. The local decision is not valid for all the path from the source to the destination. If a specific node accepts a flow, it proceeds towards the next node and it is re-evaluated. Therefore, the complete decision is achieved when the flow gets to the last node (egress router) in case other nodes have accepted the flow. Suppose a router rejects a node, this decision will be propagated to the other nodes, up until it gets to the engress router in order to reject the flow prior to entering the AS area. Therefore, in the

Hop-by-Hop operation, all the nodes on a path communicates with one another and each node must maintain the state for the actual aggregated reservation. In Hop-by-Hop operation, admission control decisions are taken simultaneously by different nodes. This kind of situation can result to concurrent problem such as thrashing (Cheikh, 2016). Thrashing occurs when one flow is accepted in a node and the required resource reservation is done only in that node. If that node later gets rejected, other flows in the previous nodes may have been falsely accepted, since the resources were available.

The Hop-to-Hop operation mode can be further divided into three types, namely: parameter-based admission control (PBAC), measured based admission control (MBAC), and hybrid (combination of measured and parameter-based admission control). PBAC (also known as traffic descriptor-based admission control) proposal assumes that the traffic characteristic of a new flow is already known before it is established, as it does not allow for any measurement or resource estimation. Traffic characteristics are conveyed by traffic descriptors, which is the only input for the admission control mechanism to provide decision. As earlier stated, the PBAC assumes that each node has a complete knowledge of the currently admitted flow request and current available network resources. The disadvantage of this technique is that it is difficult to have an accurate knowledge of each flow service request before it is established. In (Pinto et al., 2015), PBAC mechanism implementation in a peer-to-peer network for real time video streaming was presented. The admission control decision was based on traffic descriptors that characterized the applications and their contract with the service provide and the network resources.

MBAC makes an admission control decision based on real-time network measurement. It attempts to capture the requirement and characteristics of a given admitted flow and bases its decision on that information from the admitted flow. MBAC, when compared with PBAC, has the advantage to dispense knowledge about the characteristics of a flow and find it easier to predict the characteristics of aggregated flows. The disadvantage of MBAC is that the decision whether to accept a flow or not is dependent on the measurement, which may be associated with errors that could possibly lead to false negative or positive error. (Pinto, 2015) proposed a MBAC mechanism for wireless sensor network which is based on direct measurement of the packet loss ratio, throughput, and inter-arrival jitter. The authors used packet probing for measuring the performance of the parameter. (Inaba et al., 2017) implemented two admission control mechanisms; PBAC and MBAC. The authors in this work tested for the busty traffic pattern between the MBAC and PBAC and concluded that MBAC provides more efficient network utilization as compared with PBAC. The authors in (Sanada et al., 2015) proposed an analytical model for node delay distribution in wireless networks. An admission control scheme was developed for traffic with stochastic QoS guarantees to be applied at the source node.

Hybrid admission control was developed to address the problems encountered by PBAC and MBAC. The hybrid admission control uses both the traffic description knowledge submitted and the network measurement to predict future service level required by a flow. (Chamraz, 2015) proposed a hybrid admission control for available bandwidth measurement which provides better network utilization when compared to using PBAC or MBAC. (Pinto, 2015) proposed a hybrid admission control mechanism that directly estimates the bandwidth effectiveness from available trace. These estimations were later used in conjunction with peak rate value (given by the traffic descriptor) to take the admission decision about a given flow. (Cobo et al., 2017) proposed a hybrid admission control mechanism for real time traffic that takes both reliability and delay into account, and a fairness-aware rate control algorithm for non-real time traffic in a wireless sensor network tested in IEEE802.11. The admission control was deployed at the source node.

It is important to note that the kind of admission control implemented is also dependent on the QoS requested for by the application flow which will be transported within the network. The flow requesting a defined service level of an admission control can have diverse QoS requirements in terms of data rate, throughput, etc.

In (Kundu, 2019), the PBAC mechanism is used to provide hard real time services, which is based on a worse case bounds derived from the parameters describing the flow. The algorithm proposed tend to result in low network utilization when there is a busty traffic. MBAC, on the other hand, can use less stringent admission control algorithm, therefore, they are used for providing soft real-time service. In general, the type of admission control implemented should always be adequate to application and network specification as well as to the trade-off between network resource utilization and the conflicting requirements for QoS of the current flow to be maintained. Though in most cases, the admission control makes a per-flow decision, other granularities can be defined for the admission control target decision such as per -packet, per-user, or per-TCP session (Pinto et al., 2015). (Atxutegi et al., 2016) proposed an admission control decision called per-packet admission control and the decision is based on resource token instead of bandwidth measurement.

A single traditional IP network operates in a First-In-First-Out (FIFO) queuing order with tail drop, meaning that the admission control implementation can become more complex and costlier for the operation and administration of network. Therefore, the admission control implementation must maintain efficiency and simplicity. Particularly, the admission control deployment in MANET means that specific challenges will be tackled. When we compare wireless network with structural network, the wireless network normally have less usable spectrum, less reliability, and prone to interference and multipath fading. Therefore, the congestion effect can be more severe in MANET and an admission control will be very helpful. Hence, the admission control mechanism must be carefully designed in term of its efficiency and performance to cope with the wireless network limitation.

However, based on the characteristics of the different admission control outlined in this section, as well as the advantages and disadvantages surrounded by the different admission control protocol, we have chosen to deploy the edge-to-edge admission control technique. Our choice was motivated by the fact that the decision to accept or reject a given flow is made by both the ingress and the egress node only, on-like the hop-by-hop technique that allows all the nodes to make an admission control decision. In a hop-by-hop admission control technique, a flow that may have been previously admitted in a network may later be rejected in that same network because all the nodes participate in the admission control decision. This kind of instance can therefore result in thrashing. Thrashing occur when one flow is accepted in a node and the required resource reservation is done only in that node. If that flow later gets rejected, other flows in the previous nodes may have been falsely accepted, since the resources were available. Therefore, to avoid thrashing, edge-to-edge admission control has been deployed.

3.4. Admission Control Routing in MANET

The routing protocol is responsible for route information exchange, finding a path towards the destination (based on criteria such as length of hop, required minimum power, and lifetime of the wireless link), information gathering about the path that have been broken, mending the broken path, expanding minimum processing power and bandwidth. It is important to note that while designing a routing protocol, there are several challenges that may be encountered due to the nature of MANET. Such challenges include mobility, shared channel and error-prone, bandwidth constraint, and location-dependent contention. Other resource constraints that limits the capacity of a routing protocol are buffer storage, computing power, and battery power.

Routing protocols are therefore essential for ensuring the efficient functionality of MANET (Alabdullah et al, 2019) (Wang & Jiang, 2016). Its functions include regulating and building paths between nodes for packet to travel from a source node to its destination. A MANET path is made up of intermediate node set that allows the transportation of packets across a given network. Once a sender transmits packet to its destination, each intermediate node receives the packet and forwards the packet to other nodes until that packet reaches its destination. Due to the characteristics posed by MANET, such as node mobility, routing becomes a bit complex (Perkins, 2018).

Therefore, MANET environment requires a bandwidth efficient and dynamically adaptable routing protocol. The routing protocols used must be able to adjust to the network topology changes, as well as reducing the routing control overhead, in order to provide for an available bandwidth for data communication.

Research has been carried out to further enhance the MANET routing protocol (Clausen and Jacquent et al., 2018) (IETF, 2018) (Zeb et al., 2016) (Perkins, 2018). There are different methods used in classifying MANET routing protocols; figure 3.6 shows this classification. Common approach for MANET routing protocol classification is according to the discovery of route and the routing information update mechanism. MANET routing protocols are therefore divided into three groups, namely: proactive (or table driven), reactive (or on-demand), and hybrid routing protocol. In a proactive routing protocol, a consistent and up-to-date information is maintained as shown in (Menon, 2019) (IETF, 2018). A reactive protocol, on the other hand, establishes route according to the requirement of a system. A further illustration of this protocol is found in (Perkins, 2018). The hybrid routing protocol is the integration of proactive and reactive routing protocol component. The reactive protocol adjusts easily to topology changes and they consume less bandwidth resources because they avoid unnecessary periodic update of routing information at each node. A distinctive protocol that falls under reactive protocol is ad-hoc on-demand distance vector (AODV) (Perkins, 2018) and dynamic source routing (DSR) (IETF, 2018). In the next section, a general classification of MANET routing protocol will be properly investigated, and we will look into the background of AODV, our routing protocol choice.

3.4.1. Routing Protocol Classifications

There have been several published work and academic papers on the comparative analysis of MANET routing protocols. (Lakshman et al., 2018) (De Amorim et al., 2018) (Guck et al., 2017) presented an overview of the routing solution for ad-hoc network. The routing protocol classification for MANET is based on different criteria. Figure 3.6 shows the MANET routing protocol classification.

It is important to note that there are numerous routing protocols used for specific purpose, therefore, since AODV performs well in a more complex scenario (such as high load, high mobility), we have deployed AODV routing protocol in our work to enhance better performance, therefore we will only be expatiating on AODV in the next section.

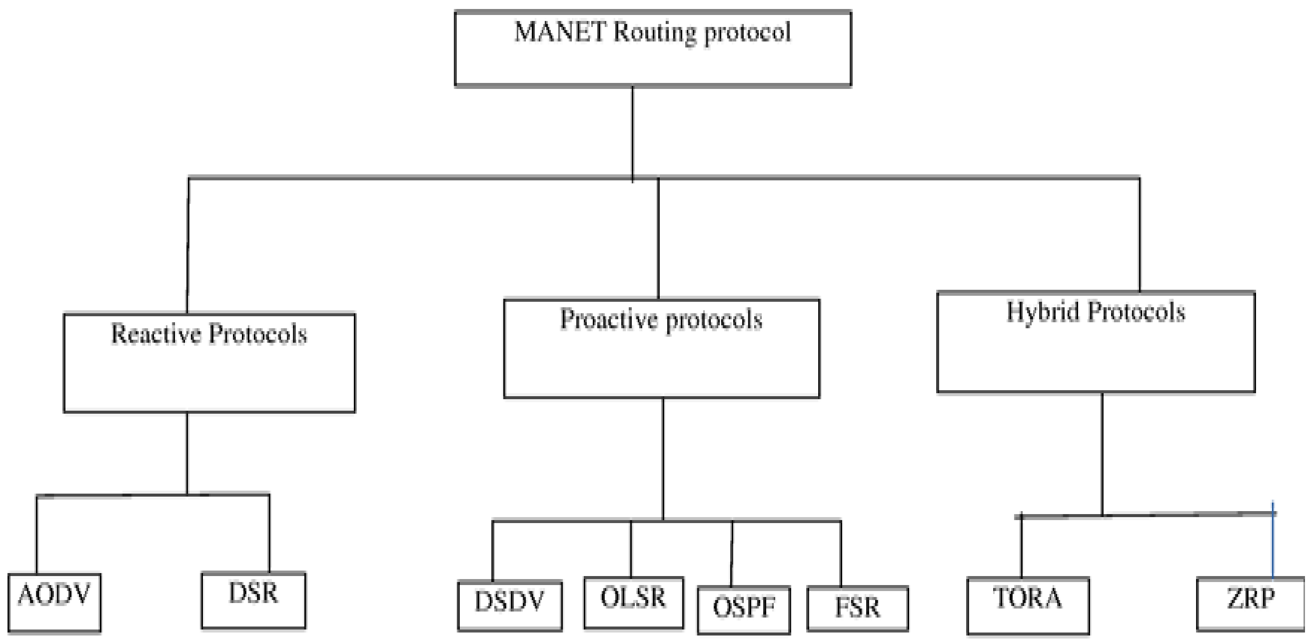


Figure 3.6: Routing Protocol Classification (Panda, 2018)

3.4.1.1. Ad-hoc On-Demand Distance Vector (AODV)

We explain the AODV routing protocol in this section since our RAACM is equipped with the AODV routing protocol.

Among the reactive protocols previously mentioned, AODV is the most popular and highly researched MANET routing protocol (Perkins, 2018). This routing protocol supports a dynamic oriented routing condition and has a minimal memory overhead. It requires a lower processing and network utilization and can determine a unicast route to a given destination within the MANET.

An on-demand algorithm provides for route discovery and maintenance and are controlled by the sender node. Also, sequence numbers are used to ensure route update. AODV is self-starting, loop-free, and able to scale to large numbers of nodes (Perkins, 2018).

Some advantages of AODV that is common to dynamic source routing (DSR) routing protocol, are route creation on-demand and building paths between the sender and receiver through the route discovery mechanism. Also, AODV has a common advantage with the destination sequence distance vector (DSDV) routing protocol, as they both have sequence number for maintaining the latest information between nodes.

AODV builds a route in an on-demand manner. The route built is not updated until there is a route breakage or there is a time out, in order to reduce the network overhead. To minimize the network overhead, each of the node has the responsibility of ensuring connectivity to its local nodes (i.e. one or two hops away) instead of all its routes. Each node is therefore responsible for maintaining updates on the occurrence of broken links or time-out. Therefore, it is possible to control ad-hoc network over a large area because of the minimal network overhead acquired. AODV allows all the nodes to maintain a table containing information about which of the neighbours to send the packet to reach the destination. The sequence number, as mentioned previously, is one of the characteristics possessed by AODV that ensures the freshness of route (Adarbah, 2015). The AODV routing is therefore made up of two phases, namely; route discovery and route maintenance (Perkins, 2018).

Route Discovery

If a source node needs to send a packet to the destination node, the route discovery mechanism is triggered by broadcasting a special control packet known as Route Request (RREQ) to its neighbours and the neighbours further rebroadcast these RREQ to their own neighbours. This process goes on until the RREQ gets to its destination node. The destination node then sends a control packet called Route Reply (RREP) following the path of the RREQ in a reverse order to inform the source node that a route has been established. This type of broadcast is also known as pure flooding (Perkins, 2018).

In AODV, an expanding ring technique is used during the RREQ flooding, Each RREQ has a time to live (TTL) feature that states the number of hops the RREQ should rebroadcast to. If the TTL value exceeds a given threshold, an error will be detected and a unicast route error (RERR) packet will be sent to the source node. Also, the RERR can be issued if the destination node cannot be located. The route followed by the RERR is always the same as the route followed during the first RREQ discovery up until the point of failure in a reverse order (De Amorim et al., 2018). In the event of an error occurring, the source node initiates a new route discovery process with a different sequence number, and this is repeated until a successful route is found. If a route is broken due to network mobility, paths can be rebuilt using additional route discovery technique. If a link to an intermediate node breaks, the local node will try repairing the broken link by creating a new receiver sequence. Thereafter, it floods that sequence to all nodes within a specified area limited to the hop counts of lower value rather than the initial hop count used for network discovery. Suppose the node that detects a broken link cannot find any alternative route to the destination, a RERR will be initiated and transmitted to the sender and the route discovery will be re-initiated over a larger area (higher hop distance) than that of the local node, if they still need the route (Perkins, 2018). The packet exchange length during the process of route discovery is kept small when compared with the data packet, but it remains significant, especially when dealing with multiple route discovery phase (Perkins, 2018).

Route Maintenance

The second and final stage of AODV routing protocol is route maintenance. Route maintenance is the process of responding to change in network topology that occurs after establishing a route. Routes are maintained regularly, especially when their purpose is met. During the process of route maintenance, the intermediate nodes consistently monitor the active link. Each of the nodes also keep an update list of its 1-hop neighbour, which is obtained through the periodic exchange of HELLO packets. The routing table is always made up of a pre-allocated destination, the next forwarding up towards the destination, and a sequence number. A route update highly depends on the sequence number of an incoming message. Updates are therefore performed when the incoming sequence number is more than the existing number. The routing table also maintains a pre-determined time for route that expires. The expiry time is usually updated to the current time plus the time value which is also known as ACTIVE_ROUTE_TIMEOUT which is attached to each route entry. The expiry time is therefore utilized whenever a given route is used for data packet delivery in order to check if the route status is out-dated or not by testing the usage within this time. Once the period specified expires, the routing table will be declared as void. In case of any link breakage or if a node receives a packet with its destination absent from its forwarding route, a RERR must be initiated by the node and sent in form of an immediate response (Saini & Sharma, 2019). Figure 3.7 shows the route maintenance process that occur whenever there is a disruption in the link of a given node. As shown in Figure 3.7, there is a link breakage between node B and node D. Node B therefore generates a RERR message, this RERR is transmitted to S. In AODV, there are two route repair types used in addressing the breakage of link. A new route can be rebuilt from the source node (source repair), or locally repaired by intermediate node (local repair).

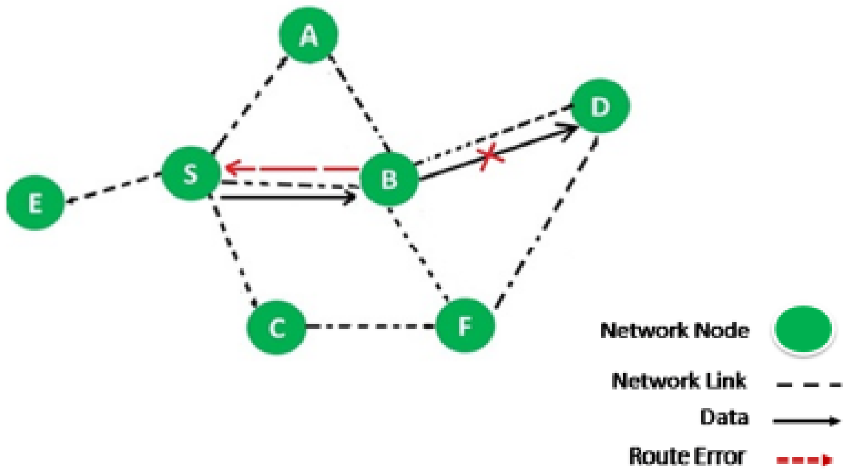


Figure 3.7: Route Maintenance Process in AODV

3.5. Wireless Protocol Standards

As earlier mentioned in chapter 1 of this thesis, our bandwidth estimation for admission control is focused on IEEE 802.11-based MANET. It is however essential to study the wireless environment to understand the type that is essential to be deployed in this thesis.

The wireless protocol IEEE have proposed various standards, This standards cover the functions of both the physical layer and the MAC layer of the OSI layer model. Figure 3.8 shows the architecture of the IEEE wireless standards for the physical layer (PHY) and the MAC layer. The IEEE firstly proposed three PHY layer options which are infrared (IR), frequency hoping spread spectrum (FHSS) and direct sequence spectrum (DSSS) (Yazid et al., 2017). Thereafter, two other bandwidth scheme were proposed in 1999 for wireless local area network (LAN) which are; IEEE 802.11a and IEEE 802.11b (Yazid et al., 2017). IEEE

802.11a operates at a frequency rate of 5GHz with a data rate of up to 54Mbps, and IEEE 802.11b operates at a frequency rate of 2.4GHz with a data rate of up to 11Mbps. In 2003, a newer standard, IEEE 802.11g, was introduced. This standard was an extension of IEEE802.11b and has a data rate of up to 54Mbps with a 2.4GHz frequency. Other standards introduced by IEEE after the introduction of IEE802.11g, includes, but not limited to 802.11n, 802.11f, 802.11i, and 802.11e. IEEE 802.11n aim to improve the performance of IEEE802.11a, by improving the data rate up to 150Mbps. IEEE802.11n uses a multiple input multiple output (MIMO) technology and adaptive OFDM as the main physical layer mechanism. IEEE802.11f is used for inter access standard for accessing wireless stations that are roaming in multivendor access points (AP). IEEE 802.11i is used for security authentication (Sivaram et al., 2019). IEEE 802.11e aim to improve the QoS of a network and will be utilized in this thesis because the aim of this thesis is to improve the QoS of a wireless network.

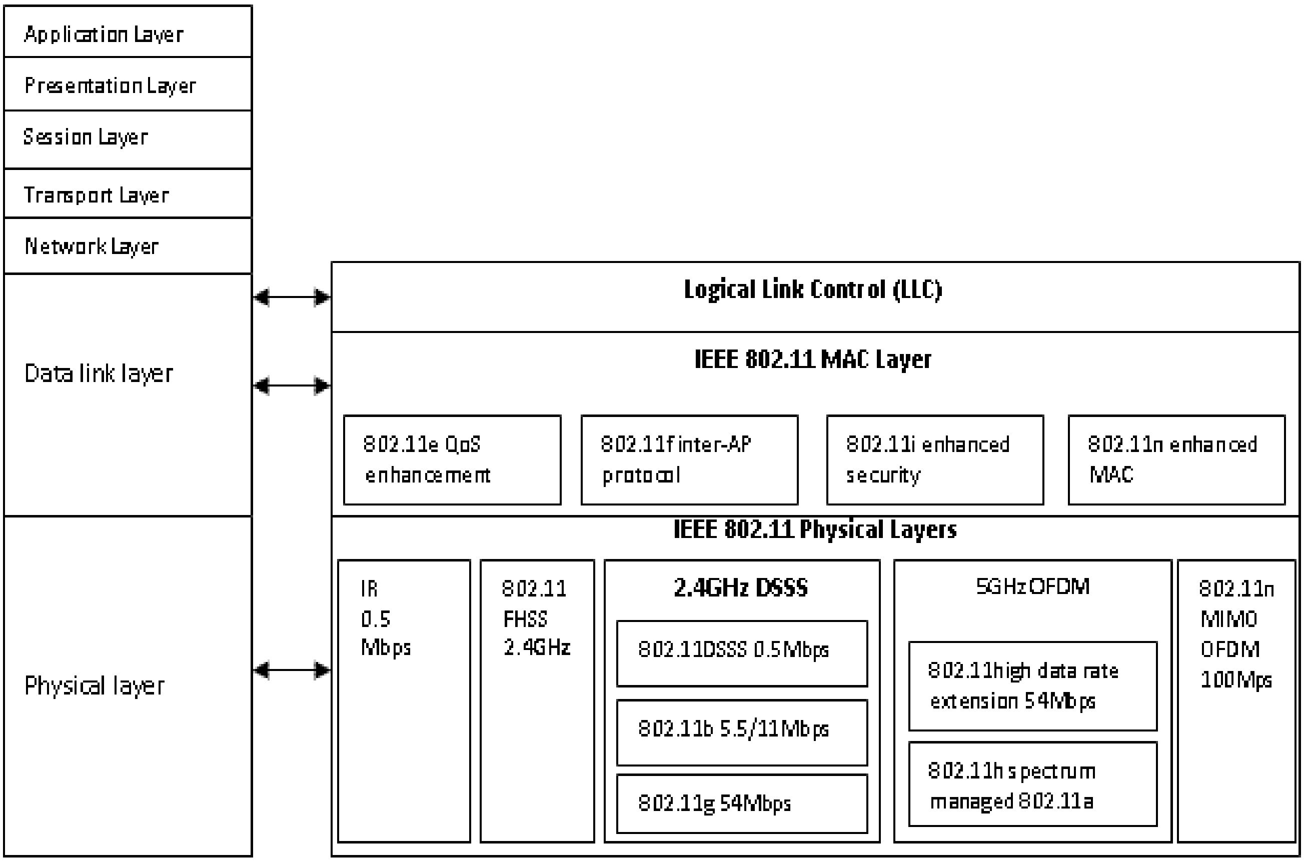


Figure 3.8 Physical and MAC layer IEEE802.11 standards (cisco, 2020).

3.5.1. QoS in Wireless IEEE 802.11

The first version of wireless 802.11 standard is made up of two medium access support functions which are distributed coordinating function (DCF) and point coordinating function (PCF). However, there was an introduction of QoS to the IEEE 802.11 MAC through the enhanced distributed coordination function (EDCF) and the hybrid coordination function controlled channel access (HCCA). Therefore, DCF is the fundamental access method for IEEE 802.11 MAC and the enhanced distributed coordination access (EDCA) is the enhanced variant of the DCF. DCF and EDCA are both designed for ad-hoc and infrastructure networks while the PCF and HCCA are designed for the infrastructure networks (Yazid et al., 2017).

3.5.2. Limitation of IEEE 802.11

It is important to note that DCF does not offer any QoS support. Its mission was to work as a best effort traffic protocol and to provide only best-effort service. The DCF is like the first-in-first-out (FIFO) scheduling task mechanism. Here, the wireless terminal will access the medium with the same priority value. This service, however, does not guarantee bandwidth, jitter, and packet loss for real time applications. This makes bandwidth, loss sensitive application and delay to be treated the same way as the best effort traffic. Therefore, applications requiring QoS support tends to suffer the same treatment as applications that require only best effort support. Because of the behaviour of DCF, all the different types of traffic suffer from the same end-to-end delay, bandwidth variation and, packet loss. DCF has also been observed to be affected by different kinds of overheads which is generated by the physical layer, control frame back-off, etc. The consistency of this problem increases with respect to the increase in data rate. Also, whenever a station accesses a channel, the frame transmission will carry on the same overhead as any previously transmitted frame of similar flow, thereby having a great impact on the network performance (Al-Maqri et al., 2016).

3.5.3. Enhancement of DCF

As previously mentioned, IEEE802.11 was not designed to support QoS, therefore it is deprived of all QoS supports. However, the IEEE802.11e was proposed to overcome the QoS limitation of IEEE802.11 and to provide to an extent the QoS support at the MAC layer. Prior to the advent of IEEE 802.11e, researchers worked immensely on improving the DCF. Various approaches were proposed in the literature for improving the QoS in DCF, however, DCF still posses some limitation as regards to the QoS. Therefore, there was an extension to the DCF known as EDCF which is used in IEEE802.11e. Enhanced Distributed Channel Function

(EDCF) extends the DCF access method by incorporating a MAC layer service differentiation. EDCF defines multiple access categories (AC), with AC-specific contention window (CW) sizes, Transmit Opportunity (TXOP), and Arbitration Inter-frame spaces (AIFS). Access to the medium in EDCF is differentiated by using the priority principle. However, the algorithm of DCF has not been completely changed in EDCF, but its time interval has been customized for each priority. The time interval has been adjusted accordingly to increase/decrease the channel access probability thereby encouraging/discouraging the data flow transmission with high/low priority (Yazid et al., 2017).

3.5.4. QoS in IEEE 802.11e

The IEEE 802.11e was designed to support the issue of QoS that arises from other standards that are commonly used (e.g. IEEE 802.11a, IEEEE 802.11b, and IEEE802.11g), and do not provide for QoS support. However, for QoS to be provided by these previous standards, lots of bandwidth will be utilized. The IEEE802.11e QoS enables the QoS access point (QAP) to schedule resources based on the requirement of the wireless station. These parameter are however optimized based on the different traffic types (Ahmed & Rikli, 2018).

It is important to note that the IEEE802.11e standard are backward compatible with the original IEEE802.11 MAC standard. The contention free (HCCA) and the contention based (EDCA) medium access in QAP is by the HCF. In order to prevent starvation in DCF of IEEE 802.11b, each of the QoS station is provided with a fixed transmitting time interval which is known as transmission opportunity (TXOP). TXOP is obtained by either contending for the medium in EDCA or after receiving a contention free poll frame in HCCA. Once an admission is granted to a QoS station, transmission of streams can be guaranteed based on the QoS requirements (Yazid et al., 2017).

3.5.5. HCF Controlled Channel Access (HCCA)

The QoS requirement in HCCA is provided by using a centralized polling technique. Wireless network load in HCCA are divided in traffic streams (TS) depending on the traffic specification protocol parameter. For a QoS station to be included in a polling list, a QoS reservation request need to be forwarded for each TS. The QoS reservation request is forwarded to the QAP. The request can be sent using the QoS management frame Add Traffic Scheme (ADDTS) (Shi et a., 2019).

3.5.6. Enhanced Distributed Channel Access (EDCA)

EDCA is an IEEE 802.11e protocol that allows access based on contention. The IEEE 802.11e describes four different access categories (AC). An example of these access categories described by this standard is as shown in table 3.2 below. A separate queue and an associated set of access category parameter is maintained by each access category. Different access categories supports queues that are related to different applications as shown in fig 3.9. Incoming packets are therefore allocated to one of the queues. Queuing packets are accessed based on two basic priority mechanisms, arbitrary interframe space number (AIFSN) and CW (CW_{min} and CW_{max}). By making use of the parameters in the QoS station, each of the AC contends in order to access the channel. Once a station detects that a medium is free for transmitting, it will automatically wait for a period called the Arbitration Interframe Space (AIF), this is to avoid colliding with other traffic categories. Before data transmission, each AC begins by counting down additional random number of time slot known as CW. Packets having a higher priority will be allowed more access to the channel as compared with packets having a lower priority (Yazid et al., 2017).

Priority Values	User priority (UP)	Access Category (AC)	Traffic types
Lowest	1	AC_BK	Background traffic
	2	AC_BK	Background traffic
	0	AC_BE	Best effort traffic
	3	AC_BE	Best effort traffic
	4	AC_VI	Video traffic
	5	AC_VI	Video traffic
	6	AC_VO	Voice traffic
Highest	7	AC_VO	Voice traffic

Table 3.2: (cisco, 2020)

Packets having an AC of 3 and AC of 2 (voice and video) are termed as higher priority while frames with AC of 0 and AC of 1 are packets with lower priority which is also known as best-effort (BE) traffic. Figure 3.10 below shows the channel access mechanism of EDCA.

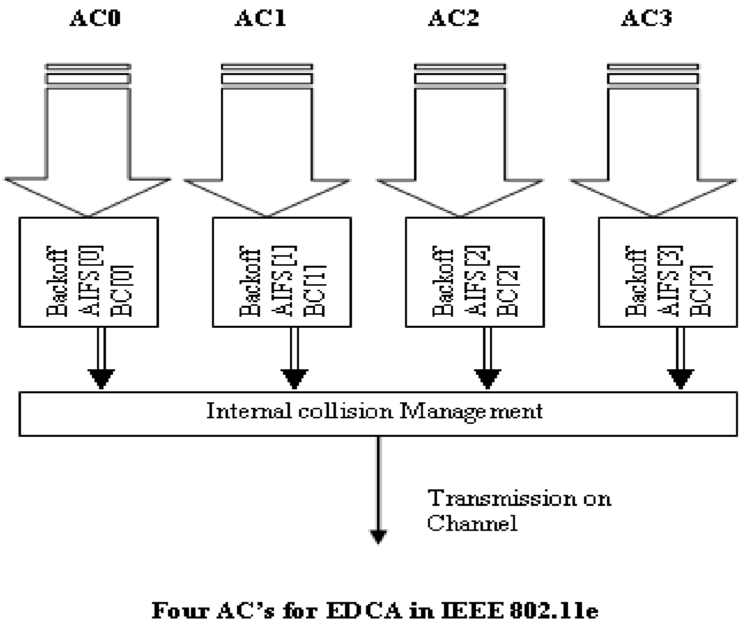


Figure 3.9: IEEE 802.11e Access Category (Vijay & Malarkodi, 2016)

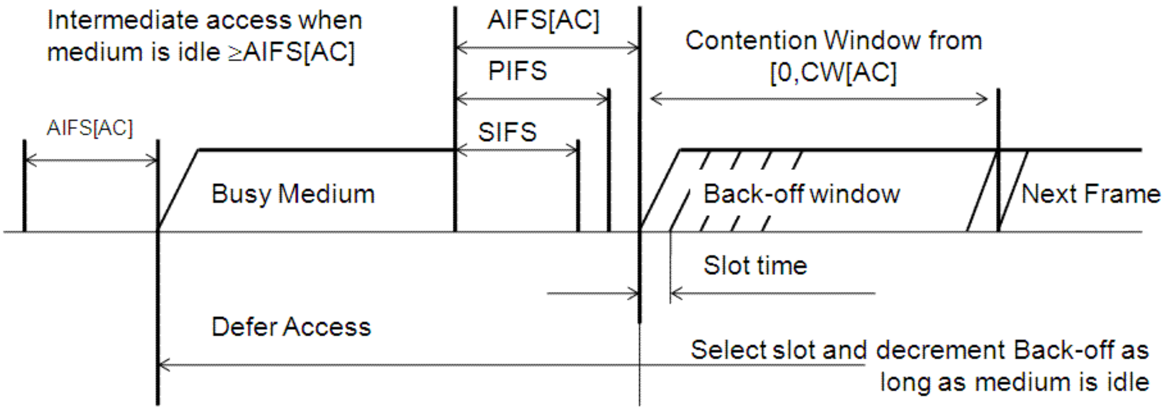


Figure 3.10: IEEE 802.11e EDCA Channel Access Mechanism (Vijay & Malarkodi, 2016)

Once access is granted to a QoS station, packets can be delivered by that station for a duration of the TXOP. The rate of transmitting the packet is determined by the AC's and the physical bandwidth. Therefore, devices that operates at a higher data rate of the physical layer can have a greater number of bits transmission than devices that operates at a lower data rate of the physical layer. The TXOP*Limt,* enables traffic with greater bandwidth requirement to have more channel access (Yazid et al., 2017).

In wireless networks, all stations are allowed to select a random back-off interval which is between 0 and CW, thereafter, the stations are to wait for a certain number of slot times before attempting to access the channel. However, the contention window is set to a minimum value, CWmin, and it doubles its size if there is any collision till it gets to its maximum value CWmax. The scaling and randomization process of CW size enables the minimization of collision.

The AIFS for a higher priority is always shorter when compared with traffic categories with lower priority value. This however makes the traffic that have a lower priority value to wait for a longer period than traffic with a higher priority value. Figure 3.11 below shows the AIFS time period for different AC (ke et al., 2017).

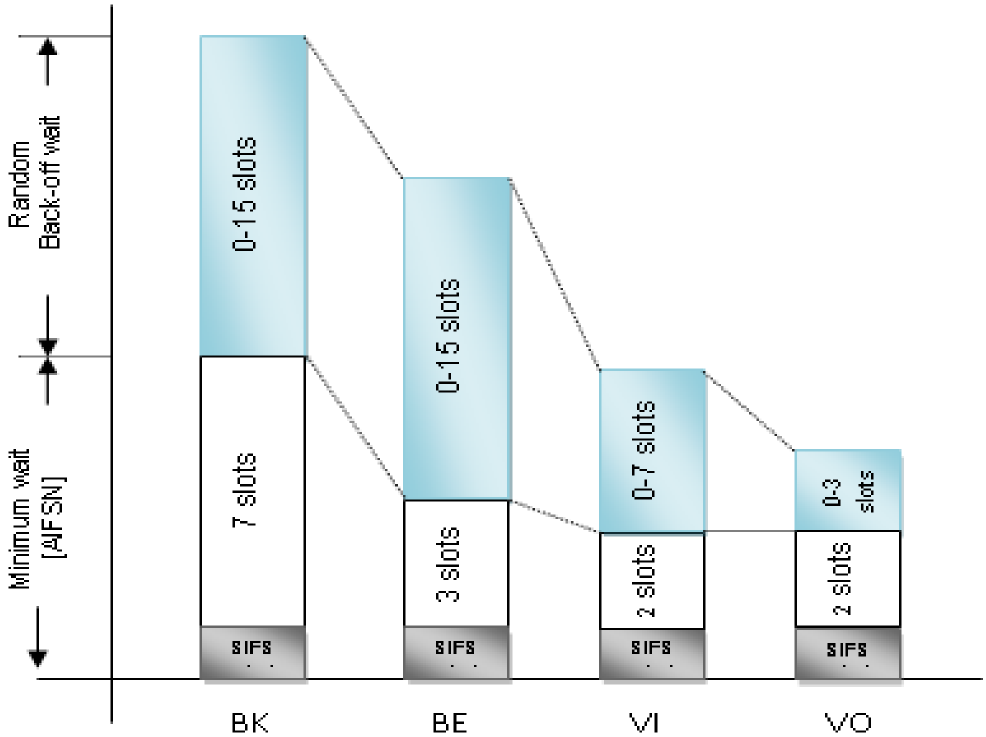


Figure 3.11: AIFS and the CW of different traffics (cisco, 2020)

For a QoS station, the value for the minimum and maximum AIFSN are 2 and 15, and for QAP the values are between 1 and 15. The higher priority value for an AC is due to a lower value for the AIFS, CWmin, and CWmax that give rise to a higher TXOPLimit. Therefore, the major difference between the DCF and EDCA is that EDCA makes use of AC specific parameters (e.g. AIF[AC], CWmin [AC], and CWmax [AC]) while DCF use a fixed value (e.g. DIFS, CWmin, and CWmax). The EDCA are therefore updated to the QAP through periodic communication. Table 3.3 shows the EDCA default value parameter (Yazid et al., 2017).

AC	$CWmin$	$CWmax$	AIFSN	TXOP	
				FHSS	DSSS
AC_VO	$(CW_{min}+1)/4-1$	$(CW_{min}+1)/2-1$	2	6.016msec	3.008msec
AC_VI	$(CW_{min}+1)/2-1$	CW_{min}	2	3.264msec	1.504msec
AC_BE	CW_{min}	CW_{max}	3	0	0
AC_BK	CW_{min}	CW_{max}	7	0	0

Table 3.3: EDCA default value parameter (cisco, 2020)

3.5.7. CWmin and CWmax

CWmin and CWmax in EDCF does not have a fixed value like DCF. Its value is dependent on the AC it occupies. Once a station start to wait for AIFS time, each of the station begins by decreasing its random back-off timer. If another station begins to transmit signal before the back-off timer reaches zero, the station will defer its access until the medium is available again, before it continues to decrease its back-off time from where it previously stopped. Once the timer reaches zero, the station will be permitted to transmit signal. If the CWmin of two or more stations reaches zero at the same time, a collision will occur. In a situation like this, the stations will increase the CW based on the binary exponential algorithm up to the value of CWmax, then it will wait. For an AC, the value of the CW is prioritized and stations with higher priority can wait for a shorter time before transmitting (Yazid et al., 2017). The CW parameter for frequency hopping spread spectrum (FHSS) and direct sequence spread spectrum (DSSS) is shown in table 3.4

	FHSS	DSSS
CWmin	15	31
CWmax	1023	1023

Table 3.4: CW for FHSS and DSSS

3.5.8. Transmission Opportunity (TXOP)

In IEEE 802.11e, once a QoS station is allowed access to a channel successfully, it is permitted to send packet for a period of TXOP. During the TXOP period, a QoS station can send packets as much as it can without having to contend for channel access. It can send multiple MAC protocol data unit (MPDUs) from the same access category after a SIFs delay between subsequent frame transmission and acknowledgement frame. Frames are divided into fragments in-case they are too big to transmit during the TXOP duration. The TXOP provides for both contention free transmission period and collision free transmission period. There is therefore an inefficiency observed in cases where packets in a queue is few. This will result in the channel being idle. TXOP has a maximum value called TXOPLimit, which helps to improve the channel performance. Table 3.3, EDCA default value parameter, shows the default values of different parameters. The frame transmission in EDCA is influenced by TXOP. The overall frame exchange sequence is made up of intermediate SIFS, RTS/CTS and ACK frames. In transmission period, TXOP does not exit its limit (TXOPLimit). If a TXOPLimit is non-zero, this means that the TXOP duration is not exceeded and the EDCA can keep on forwarding its packet. Therefore, we define a contention free bursting as a situation whereby the value of the TXOPLimit is non-zero, which makes the EDCA to forward a number of packets in the TXOP within the available TXOPLimit where the frame belongs to the same AC. In this circumstance, each data block is replaced by SIFS slot instead of AIFS. Post back-off period is also affected as shown in figure 3.12 below. The TXOP, therefore enables the transmission of packet belonging to the same AC but not the same station (Yazid et al., 2017).

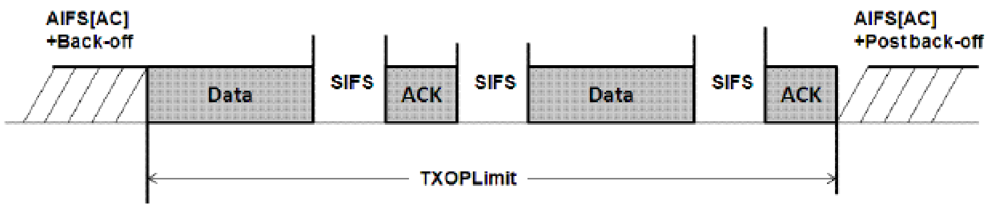


Figure 3.12: Contention Free Bursting (CFB) (cisco, 2020)

The handshake process is performed in the CFB by using only one RTS/CTS frame instead of using every frame in the CFB. When the TXOPLimit is zero, the CFB will be disabled. Suppose the transmission period of the first frame exceeds the TXOPLimit, the frame should be divided into fragment. If the TXOPLimit for access category of best effort (AC_BE) and access category of background (AC_BK) traffic is zero, this means that the CFB is not activated for this kind of AC. When a higher priority traffic is transmitting, it will hold the station for a longer period of time which can however result in the starvation of other AC frames with lower priority (Yazid et al., 2017).

3.6. Discussion

Bandwidth estimation is a vital admission control component that enhance QoS in MANET. In the preceding section, we categorized the bandwidth estimation technique into active and passive technique. Different nomenclatures have been used by researchers to categorize bandwidth estimation, however, all those classifications perform similar functions. The bandwidth related metrics, as identified in the literature, are link/path capacity and available bandwidth estimation at the node (Megyesi et al., 2017). Most research work have used the active bandwidth estimation technique for the estimation of wired link channel while only few attempts have been made using the active measurement technique for estimating wireless network because of its inaccuracy in measurement. Active end-to-end available bandwidth estimation introduces extra overhead, affect the accuracy, and degrades the network performance of the bandwidth estimation. Therefore, the active bandwidth estimation approach is not the best choice for measurement in wireless networks (Oshiba et al., 2016). Passive techniques, on the other hand, uses local information such as loss in packet, delay, and congestion situation to monitor the medium of communication. This is done over a certain period of time without interrupting any existing traffic flow (Aina et., 2019). Passive techniques which was adopted in this thesis prevent extra usage of bandwidth channel which may cause more overhead.

For proper admission control to be carried out, we identified two-admission control types which are, distributed admission control mechanism and centralized admission control mechanism. When compared, the distributed admission control mechanism is suitable for large and highly dynamic network, therefore it was deployed in this thesis. The distributed admission control provides a admission control decision in a distributed manner which mostly implies sending extra control message for distribution of decisions. Therefore, they are not optimized to the constraints of MANET, where nodes use IEEE802.11 standards with limited resources. Also, to avoid processing useless packets, we noted that the admission control mechanism should be deployed at the ingress and the egress of a network node in a distributed edge-to-edge manner as against the hop-by-hop admission control that allows decisions to be taken simultaneously by different nodes. This kind of situation can result in concurrent problems such as: thrashing (Cheikh, 2016). Thrashing occurs when one flow is accepted in a node and the required resource reservation is done only in that node. If that node later gets rejected, other flows in the previous nodes may have been falsely accepted, since the resources were available.

The routing protocols used was also discussed in the previous section. It was noted that the routing protocol adopted in a network must be able to adjust to network topology changes. This work therefore adopted Ad-hoc on-demand routing protocol (AODV) for our admission control implementation of routing protocol in MANET amongst other existing routing protocols. This is because it performs better in complex scenarios such as high load and high mobility. Further discussion on AODV was also provided in section 3.4.1.1.

Finally the wireless protocol standard was discussed, amongst the different wireless protocol that exit, this thesis adopted the IEEE802.11e wireless standards, this is because this standard will enable the QoS that is proposed in this thesis to be achieved. Discussion on the EDCA and HCCA was highlighted as being the source of channel access mechanism for the guarantee of bandwidth to enhance QoS.

3.7. Motivation to Start this Research

Having reviewed the literature, it was observed that the channel idle time dependency sensed by the sender and receiver has not been properly addressed, as most previous work did not consider it. The related works that considered the channel idle time dependency only differentiates the *BUSY* state from the *SENSE* BUSY states, therefore the *IDLE* state caused by an empty queue is yet to be addressed. Furthermore, the state-of-the-art available bandwidth for admission control broadcasts to two hop neighbours in order to retrieve the available bandwidth on a carrier sensing region. This tends to create a higher overhead that can possibly be avoided.

This thesis therefore addresses how well the available bandwidth estimation can be carried out by not only differentiating the *BUSY* state from the *SENSE* BUSY state but addressing the *IDLE* state that can be caused by an empty queue when the channel idle time dependency sensed by the sender and the receiver node is considered. This thesis also addresses how well the available bandwidth can be retrieved on the carrier sensing region without flooding the network with enormous broadcast messages. To retrieve the available bandwidth on a carrier sensing region, the HELLO message only advertises to the first-hop range which further propagates to other hops. This technique adopted to retrieve the available bandwidth helps to limit the overhead generated by the network.

3.8. Summary

This chapter reviewed the academic literature of available bandwidth estimation method and admission control in MANET. Also presented in this chapter is a subdivision of the bandwidth estimation techniques. Active techniques and passive estimation techniques are the two main techniques proposed in this chapter to estimate the available bandwidth for admission control, despite different researchers using different nomenclatures to classify these bandwidth estimation techniques. Classification of bandwidth estimation into active and passive technique helps to simplify the readers understanding of the bandwidth estimation process for admission control. Active end-to-end available bandwidth estimation was thereafter found to introduce extra overhead, affect the accuracy, and degrades the network performance of the bandwidth estimation. Therefore, the active bandwidth estimation approach is not the best choice for measurement in MANET. We have suggested passive available bandwidth estimation for admission control deployment in MANET.

This chapter went further to investigate the admission control that can be deployed for the bandwidth estimation. The admission control was therefore classified into centralized and distributed admission control. Considering the different characteristics and limitations posed by centralized and distributed admission control mechanism, the distributed admission control mechanism appears to be more adequate for the scenario of this thesis, therefore, the distributed admission control mechanism was deployed in this thesis.

Furthermore, we discussed the routing protocols that can be deployed with admission control in MANET. We therefore choose to use an on-demand (reactive) routing protocol AODV because it performs well in complex scenarios such as high load and high mobility.

Lastly, we discussed the wireless standards that will enable the thesis to be carried out appropriately, amongst the different wireless standard that exists, this thesis choose to deploy the IEEE 802.11e standards because of the incorporation of QoS that exist in this standard. In IEEE 802.11e, we looked at the two medium access that exist for proper implementation to be carried out, which are HCCA which is a contention free access and EDCA which is a contention based medium access. AC based on TXOP function and its limit in IEEE802.11e was properly looked into as well as the AIFs waiting time for collision avoidance.

Chapter 4: Research Methodology for MANET Bandwidth Estimation and Admission Control with Simulation and Design Plan

4.1. Introduction

The aim of this chapter is to provide an in-depth description of the simulation tool as well as the design model used in this research work. This thesis used the optimized network engineering tool modeler (OPNET) now referred to as Riverbed modeler (Riverbed 17.5), for network modelling and simulation. Regardless of the change in name of OPNET modeler to riverbed modeler, throughout this work, we will use the name OPNET.

4.2. Modelling and Simulation

Simulation in this thesis, is used to evaluate the network performance based on the following:

- Simulating different network scenario is cost effective with regards to timing.
- For better understanding, simulations can be repeated easily.
- When a network is simulated, it provides better understanding of the network.
- It makes it possible to simulate the network with different performance metrics and parameter.

4.2.1. Simulation Tools

There are different network simulation tools used in evaluating network performance. Examples of some common network simulation tools are: Network simulator 2 (NS-2), NS-3, GloMoSim, OMNeT ++, QualNet, J-Sim, OPNET, etc. As earlier mentioned, we used OPNET in this research work because of the availability of the licenced version at Anglia Ruskin University (ARU). Also, OPNET is easy to deploy when compared with other simulation tools. Other simulation tools will be briefly discussed and compared with OPNET.

1. **Network Simulator 2 (NS-2):** NS-2 is an open source simulation tool that can be freely downloaded and installed on the computer system. It consists of two simulation tools, Network Simulator (NS), which contains IP protocols and Network animator (nam) for visualizing the simulation. NS-2 supports two languages; C++ and Tcl. It has many built-in simulation modules with several features (Kumar, 2009). Key among these features are,
    - Provision of network environment for ad-hoc network.
    - Multiple path routing.
    - Wireless channel modules

- Mobile host for wireless cellular network
- It can be installed on multiple platforms such as Ubuntu, Windows, UNIX, etc.

2. **Network Simulation 3 (NS-3):** This is not an extension of NS-2. NS-3 is a new simulation tool that supports two languages, C++ and python. In NS-3, windows platform is partially supported because some aspects of NS-3 is dependent on Unix/Linux support.
3. **OMNeT++:** OMNeT is used for simulating power consumption problem, it is easy to trace and debug any problem encountered. It support C++ programming language. OMNet simulation tool has limited routing protocols available.
4. **GloMoSim:** GloMoSim is a network simulation tool freely available online and can be downloaded and used for research and educational purpose. It provides simulation environment for wireless and wired network. GloMoSim uses a set of library modules to simulate a given routing protocol in the protocol stack. C language and PARSEC are used for developing the library. The QualNet version of GloMoSim is used for commercial basis.
5. **J-Sim:** J-Sim is an open source simulation tool and can be freely downloaded and installed on computer system. It has a reusability and interchangeability model which makes it easy to trace and debug programs. J-Sim supports two programming languages, Java and Tcl.

Simulator Comparison

Simulator	Language	Advantages	Disadvantages
OPNET	C, C++	Support large number of customer. Provides professional support. Well documented.	It is costly but provides suitable price for universities. OPNET is more suitable to managers than for researchers because of their generic performance
NS-2	C++, Otcl, TCL	New protocols are added easily. Large number of protocols are available. It is an open source.	Takes a longer time to learn. Poorly documented.

		Visualisation tools are available. It has many user-groups.	
NS-3	C++, Python	It is not an extension of NS-2. It is a new simulator.	It fairly supports windows platform because some of the aspects of ns-3 is dependent on Unix/Linux support.
OMNET++	C++	Used for simulating power consumption problem. Easy to trace and bug.	There are limited available routing protocol. It has no compatibility (not portable)
QualNet	C++	Supports distributed computing and multi-processor systems. It has animation capability. GloMoSim is an open source of QualNet which is freely available and are specialized for ad-hoc network. It is important to note that GloMoSim lacks some of QualNet facilities.	Problem with installation on Linux. It is costly. Slow Java-based user level.

J-Sim	Java, Tcl	It is open source. It is reusable and has an interchange ability model. Trace and program debug are easy.	Simulation efficiency is low. Run-time overhead. Only one MAC protocol is provided for wireless network.

Table 4.1: Comparison of Different Simulation Tools

4.3. Introduction to OPNET Modeler

The OPNET technology was acquired by the riverbed technology in 2012 (Riverbed, 2018) and it is a commercial product for providing network modelling and simulation software solution for Engineering, Science, and Research students. OPNET is a dynamic discrete event simulator (DES) made up of an easy to use graphic user interface (GUI) which supports analytical simulation, hybrid simulation and a 32-bit/64-bit fully parallel simulation (Lu and Yang, 2012). OPNET is a logical or mathematical model of a physical system comprising of specific point in simulated time. It provides an extensive development environment with tools used for simulation, model design, data collection and analysis. It also provides support for communication network model such as MANET modelling and distribution system. OPNET enables the simulation of all kinds of networks and technology such as VoIP, OSPFv3, IPv6, TCP, MPLS, MANET, etc. The key features of OPNET are:

- Fastest discrete event simulation engine among leading industry solutions.
- Object-oriented modelling.
- Integrated GUI-based debugging and analysis
- 32-bit and 64-bit fully parallel simulation kernel.
- Hundreds of vendors’ device and protocol model with source code.
- Hierarchical modelling environment.
- Open interface for integrating external object files, library and other simulations.
- Optional system-in-to-loop to interface simulations with live systems.
- Discrete event, hybrid, and optional analytical simulation.

The OPNET simulator are usually carried out in four different ways. Firstly, users create the network model, also known as modelling. In statistics, users will have to select values according to the required results and then simulate the network. The last step involves the user analysis of results as shown in the flow chart below:

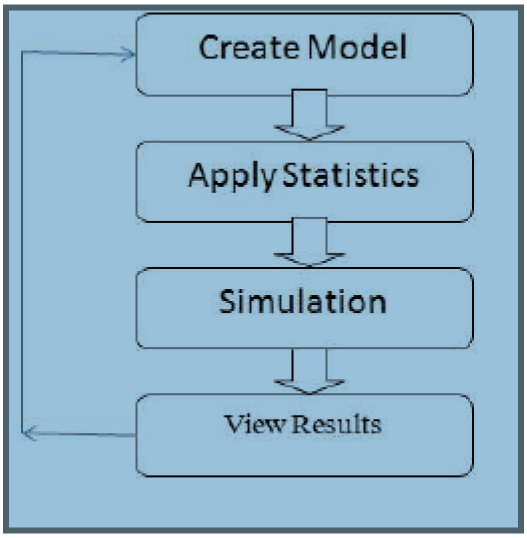


Figure 4.1: OPNET Simulator Flow Chats.

4.4. OPNET Environment

As earlier mentioned, the OPNET modeler provides a development environment with modelling features, however, within the context of this research work, our focus will only be on the OPNET modeler editors.

4.4.1. OPNET Modeler Editors

In OPNET Modeler, there are different types of editors used to simplify modelling and simulation tasks that have a graphic user interface feature (GUI). This research work therefore focus on using the following editors: project editor, node editor, process editor and open model source code. Each of these editors enables some related functions within a window as contained in the general graphical environment of OPNET. Table 4.3 provides an explanatory summary of the editors used.

4.4.1.1. Project Model

The MANET network model as well as any network model in OPNET is created within the project editor. The project editor provides the tool-box and platform for modelling and setting up of any network architecture. It has a drag and drop button to add network objects such as switches, base station (BS), subscriber (SS) workstations, servers, links, etc. Figure 4.2 below depicts an example of a modelled wireless network.

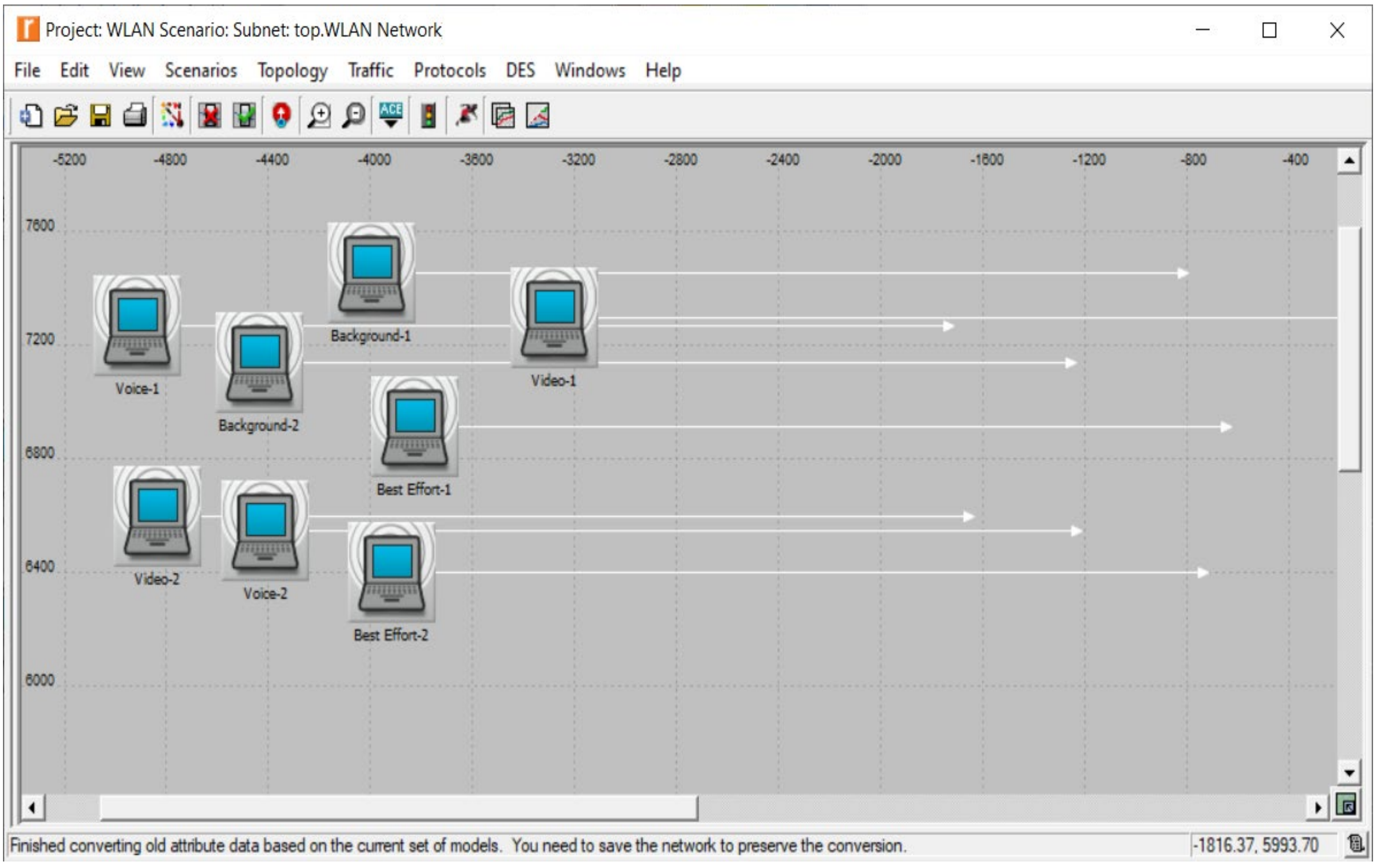


Figure 4.2: WLAN Project Editor Environment

4.4.1.2. Node Mode

The node model in OPNET is represented by a blocked structure and represents the internal functionality of a network node object. The interconnected blocks of nodes are referred to as *modules*. Each of the modules is made up of a set of input, output, and some state memory. The node models determine the way in which the input and output modules are connected using objects known as *connections*. There are two basic types of connection; connections that carries data packet and connections that conveys individual values e.g. statistical values. Figure 4.3 depicts the node model structure of MANET subscriber workstation. It represents a predefined protocol which is based on wireless standards. The protocol includes the application layer, UDP, TCP/IP, MAC, ARP, and PHY. It is also used for modelling transceiver antenna for subscriber station.

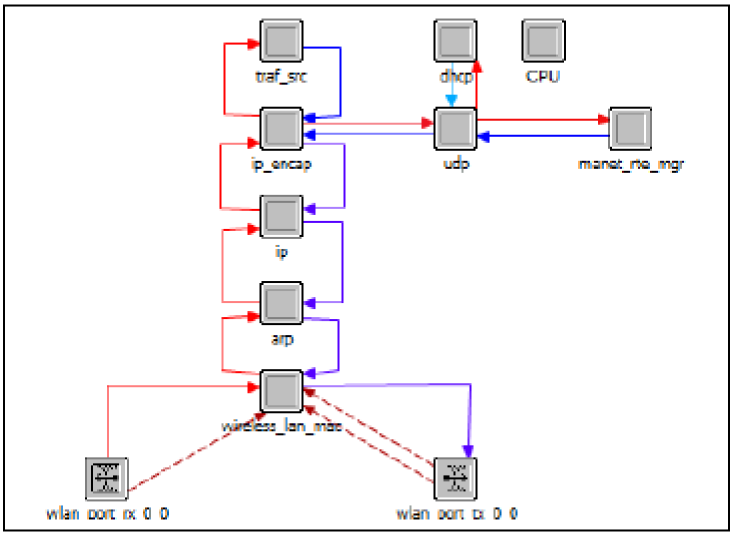


Figure 4.3: MANET Node Model

4.4.1.3. OPNET Process Model

The process model of OPNET are created and edited in the process editor. OPNET process models are used for expressing the *modules* behaviour within a node model. They are also used for setting up and modelling an extensive range of software and hardware subsystem such as: shared resources, operating systems, algorithms, custom statistics, communication protocols, etc. For proper development and definition of these subsystem, OPNET uses only one programming language known as *Proto-C*. In OPNET, the process editor supports *Proto-C* and it is fully incorporated in the OPNET modeler application. The *Proto-C* is a programming language that incorporate the C and C++ features within it. The finite state machine (FSM) used for describing the process model behaviour are state transition diagram (STD). *States* are used to represent the top-level modes that a process can enter, while *transitions* are the changes in state possible for the process. Both the state and transitions are components of STD and *Proto-C* are represented in the process editor graphically. Table 4.2 shows the object used for building the process models.

While STD graphically represents the process model, the *Proto-C* language supports the textual representation of all parts of a process model. The textual representation is done inside the state and transition objects which is established on a robust library of simulation distributed system related procedures or OPNETs unique application program interface (APIs). The logic of *Proto-C* can be easily converted into C and C++ programming language with less overhead for efficient performance. Therefore, state information access, and control flow statement (e.g.

structure statement selection like *if-else* statements that sequence through multiple operation) are implemented in a direct manner to prevent any performance issue.

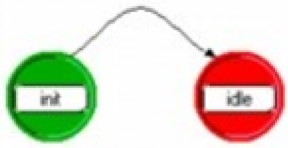
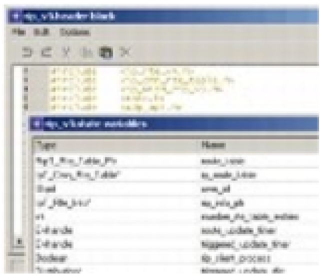
Object Type	Definition	Representation
State	It characterises a mode of the process which has been achieved as a result of preceding stimuli and analogous decisions. States comprise of code conveying processing that is done soon after they are entered, or immediately before being exited.	
Transition	Specifies a likely path that a process can take from a source state to a destination state. Every state could be the source and destination of any number of transitions. A transition has a condition statement which stipulates the prerequisites for the process to occur after the transition.	
Model Level Information Blocks	There are quite a few blocks of text that specify additional components of the process and they include: declaration of state, and temporary variables; user-defined functions that can be called by the process' states and transitions; code to be implemented upon the termination of a process; and the declaration of global variables, data structures, etc.	

Table 4.2: Process Model Object

Forced and Unforced States

There are two states in OPNET process model: *forced* and *unforced* state with both states having varying execution time. Forces state in Proto-C are always symbolised by green circle colour while the unforced state is always symbolised by the red circle colour. Figure 4.4 below shows an example of both the forced state and unforced state.

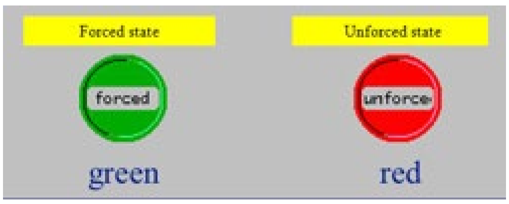


Figure 4.4: Forced and Unforced States

An unforced state allows a pause in between the state execution, therefore, it can be used to measure the true state of a system. In forced state, there is no waiting once there is a process execution, hence, it cannot be used to represents a system model that persist for a particular duration.

Editor	Purpose
Project Editor	Edits the topology of communication network model and used for basic analysis and simulation.
Node Editor	Used for specifying the device model arrangement. Device model can be represented in form of an object node in the network domain (e.g. work station, switches, bridges, etc.)
Process Editor	Used for specifying the process model activities. Process models are represented as a process in the node domain and exist inside the processor, queue, and external system modules. The process models uses a finite state machine (FSM) paradigm to express actions that are subject to current state and new stimuli.

Table 4.3: Types of Editors

4.5. OPNET MANET Model Suite Overview

An explanation of the OPNET MANET model suite's basic feature is given in this section. This suite includes a discrete event simulator that analyse the network performance in a wireless network. It is important to note that MANET suite comes with a licensing permit that allows for viewing and modification of MANET process models. Sub-sequent sections discuss some of the important features of OPNET MANET model suite.

4.5.1. Model Features

The OPNET MANET model provides various important functionalities as specified in the IEEE802.11 standards. This includes MAC messages such as bandwidth requests and MAC protocol data unit (PDU). It has a radio link for static burst configuration of all BS and SS connection. It supports QoS flow as well as scheduling and QoS categories. Table 4.4 summarizes these features and other related features of OPNET MANET model.

Model Features	Description
MAC messages	802.11e MAC PDU, 802.11e management message (bandwidth request), admission request messages, ranging messages and mobility messages.
Radio link control	This enables the configuration of static burst profile for each connection.
Mobility	To enhance mobility support, the mobility features are modelled to include: dynamic selection, neighbour advertisement of predefined scanning, base station scanning or mobile subscriber.
Scheduling service	This model feature supports unsolicited grant service (UGS), extended real-time polling service (ertPS), real-time poling service (rtPS), non-real time polling service (nrtPS) and bandwidth estimation
Packet loss modelling	This is the feature model packet loss which is caused by the effect of the physical layer.
Bandwidth allocation and request mechanism	This feature model includes aggregated request, piggybacked bandwidth request, etc.
PHY modelling	It models the PHY layer overheads as well as the PHY profiles for direct sequence. It models co-channel interference and path-loss.
Broadcast and multicast traffic	It supports broadcast and multicast traffic when the modelling of the physical layer is enabled.
Quality of service (QoS)	It supports QoS features such as admission control, active service flow, service class name parameter, queuing/buffer, global service class name, etc.

Initial SS and BS association	Specifies the base station a subscriber is connected to.

Table 4.4: Summary of OPNET Model Features and Description.

4.5.2. OPNET Mobility Configuration

OPNET mobility configuration is required to set the mobility pattern for all the mobile nodes used within the simulation. Random way point is seen as one of the mobility models used in OPNET simulation. Figure 4.5 shows the mobility configuration

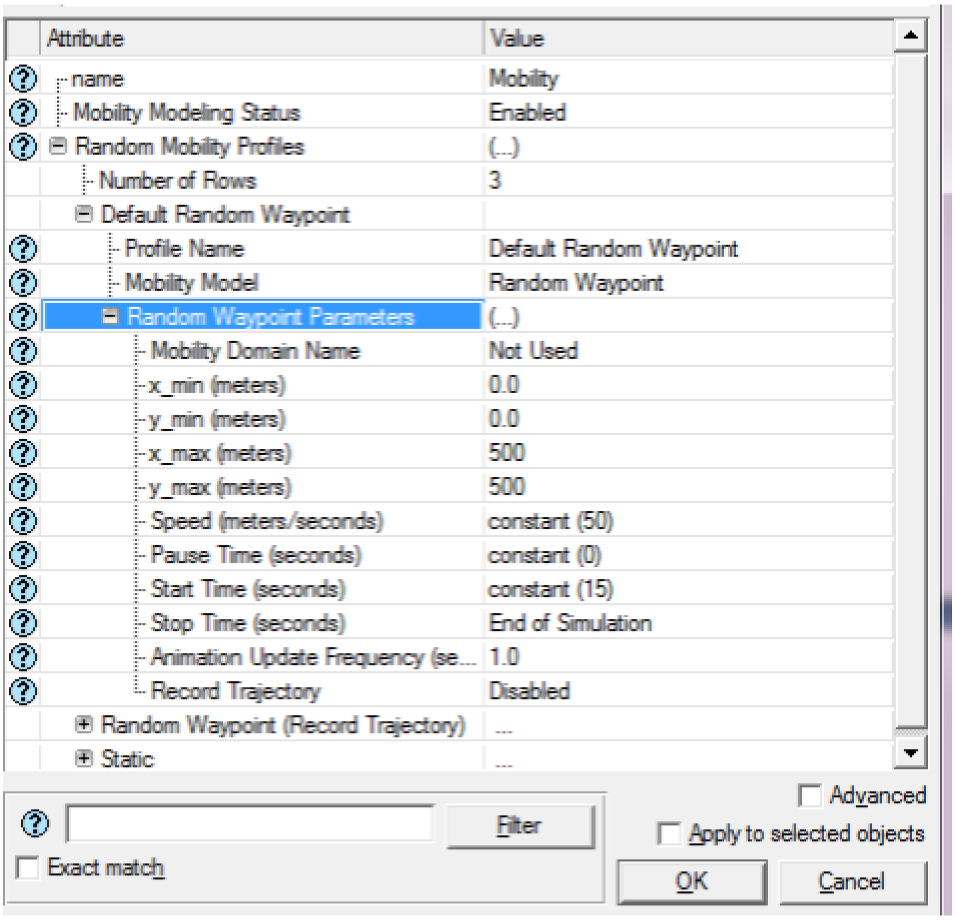


Figure 4.5: Mobility Configuration.

4.5.3. OPNET IEEE 802.11E HCF and EDCA Configuration

To allow QoS for admission control and prioritisation, the EDCA parameter is essential to be configured which provides access for different traffic specification. EDCA is an IEEE 802.11e protocol that allows access based on contention. The IEEE 802.11e describes four different access categories (AC), an example of the access category described by this standard is as shown in figure 4.6 below. A separate queue and an associated set of access category parameter is maintained by each access category. Different access category supports queues that are related to different application. Incoming packets are therefore allocated to one of the queues. Queuing packets are accessed based on two basic priority mechanisms, which are arbitrary interframe space number (AIFSN) and CW (CW_{min} and CW_{max}). By making use of the parameters in the QoS station, each of the AC contends in order to access the channel.

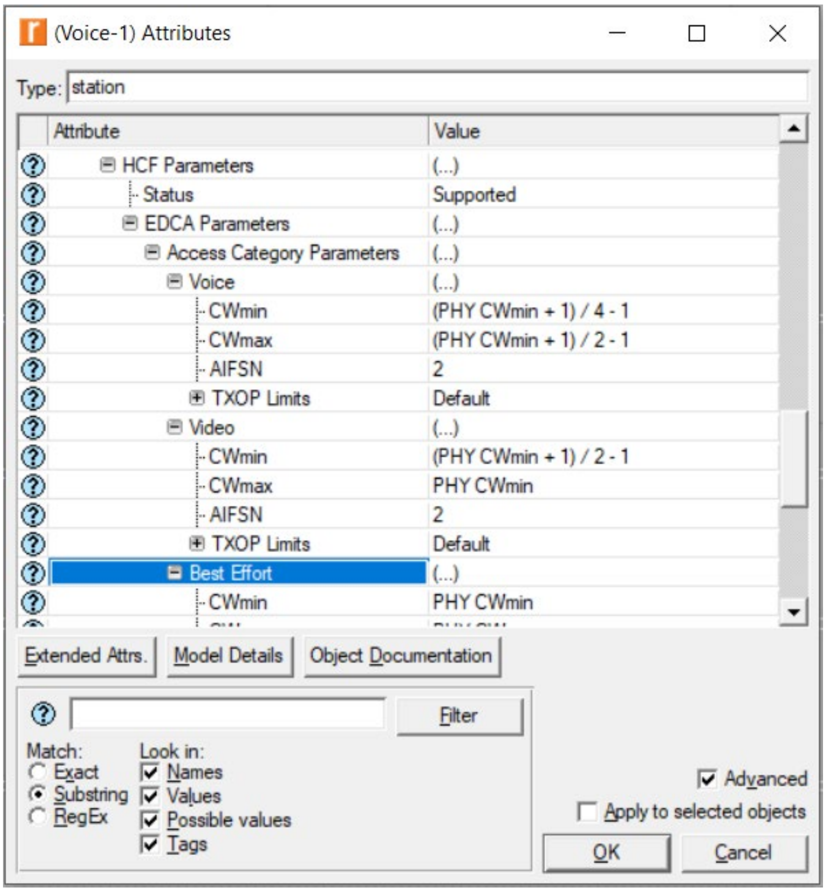


Figure 4.6: HCF and EDCA Configuration

4.5.4. OPNET Traffic Types of Services

The traffic types of services in OPNET is used to specify what types of services that is provided by the network. OPNET is a configuration standard used for implementing different traffic such as Background, Best Effort, Interactive Voice, Interactive Multimedia, etc. The generated traffic creates demand on the network bandwidth and underlying network technology. The traffic also adds up to the server loads. Figure 4.7 below shows the traffic type of service that can be deployed by a network in OPNET. This thesis deployed the best effort, voice, and video traffic for the purpose of QOS and admission control implementation on the network.

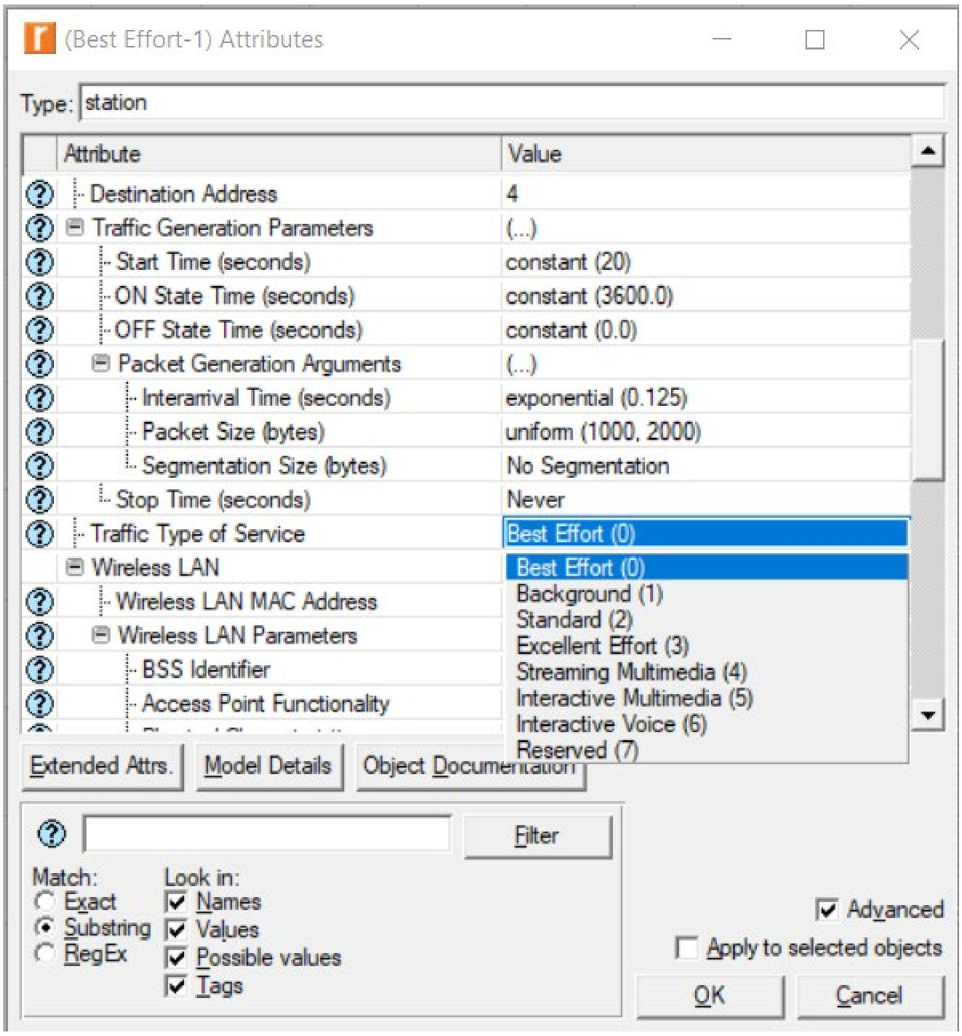


Figure 4.7: OPNET Traffic Types of Service

4.6. Bandwidth Estimation based Admission Control Set-up in OPNET

This section describes how the bandwidth-based admission control was setup in OPNET. As shown in figure 4.8, there are few sources, an admission control module, and a sink.

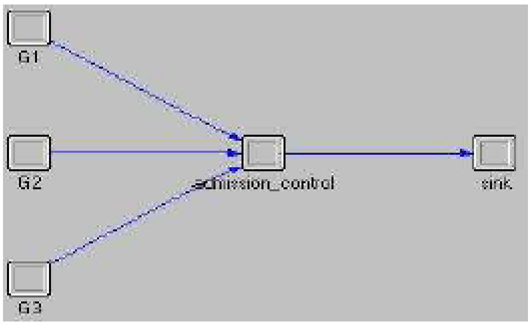


Figure 4.8: Admission Control Node Model Setup at the Source Node

The function of the node model is to imitate the real process offered to a connection request such as admission control. The source in figure 4.8 is used for generating connection request using OPNETs *simple_source* process model and our designed packet format. As shown in figure 4.9, our packet format is made up of three fields (*groupID, rate* and *service time*). This fields are given different values at different sources for different packets. *groupID* states the group the connection request belongs to while, *rate* states the required bandwidth of this request, *Service time* states the amount of time the connection will last for and it is a random variable of exponential distribution.

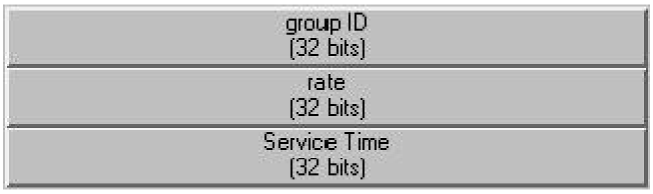


Figure 4.9: Packet Format

The packet generated from all the sources are sent to the *admission_control* module. On receiving the packet, the *admission_control* module extracts the information of a certain connection request from the received packet and executes the admission control algorithm. If the connection request is rejected, the packet of the connection request is sent to the source node or deleted. In the case of an accepted connection request by the source (ingress) node, there will be an engress admission control check as well, since we are using the distributed edge-to-edge admission control strategy (In the Edge-to-Edge operation, only the ingress and the egress nodes can make admission control decision). Therefore, if there is an accepted connection at both the ingress and the egress admission controller, the *admission_control* module must reserve corresponding bandwidth resources for it and keep the information of the alive connection before it is out of service time and deleted. Figure 4.10 shows the process model used by the *admission_control* module.

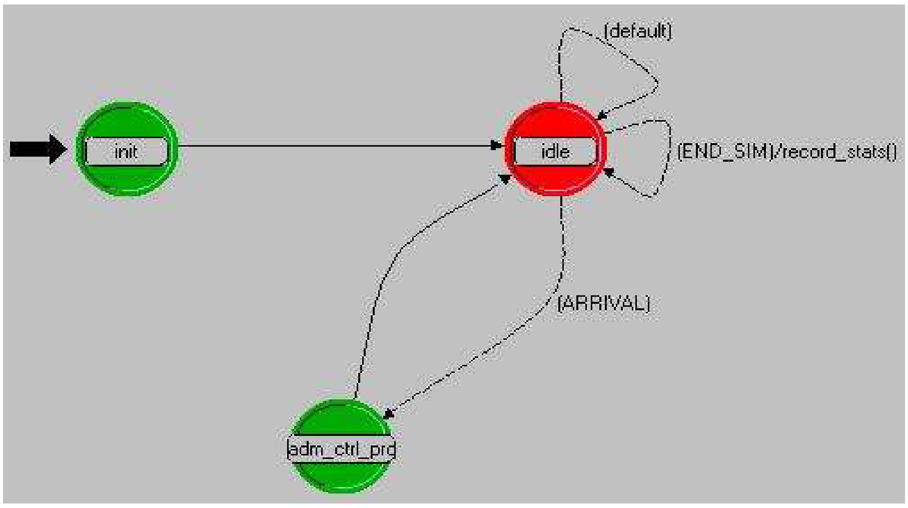


Figure 4.10: Process Model of Admission Control

The *admission_control* process model starts by executing the *init* state. The *init* state is used to run the start-up initialization before passing the control to the *idle* state. The *idle* state is executed whenever an interruption generated. When a *stream* interrupt is generated, the control is passed on to the *admission_control_process* state which is used to implement the admission control algorithm.

4.7. Summary

This chapter has discussed the different simulation tools that can be deployed for simulation of our proposed bandwidth estimation for admission control. Simulation tools are important for performance evaluation of a network. A table showing the comparison of all the simulation tool was outlined based on the languages it supports, their advantages, and disadvantages. Optimized network engineering tool modeler (OPNET) now referred to as Riverbed modeler (Riverbed, 2018) was described as a research modelling tool for simulating and designing of communication network. OPNET was therefore used for carrying out the network modelling and simulation done in this research work because it can best simulate our proposed technique and its availability. Also, OPNET was used because it is easy to deploy when compared with other simulation tools. OPNET's environment was thereafter studied and its basic features. It was pointed out in this chapter that the OPNET Modeler environment is made up of four editors namely: project editor, node editor, process editor, and open model source-code. The OPNET modeler 17.5 which was used in this work has a MANET network module for modelling MANET networks and it enables the addition of new features to existing design. At the heart of every node object in OPNET Modeler is the node model and process model comprising of state transition diagrams that are programmed with *Proto-c* programming language which can be modified into C/C++ programming languages. This section also discussed the admission control in OPNET modeler with the design of a new network object having its own node model and process model. One of the advantages associated with using OPNET for designing is that it provides professional support and it is well documented with support for large numbers of customers.

The next chapter presents the proposed scheme, RAACM, as well as the key element needed for the design of a proper bandwidth estimation and admission control.

Chapter 5: Implementation of a New Bandwidth Estimation For Admission Control in MANET

5.1. Introduction

This chapter begin by defining the available bandwidth of a channel and the different states of wireless network. Thereafter, the proposed approach used in this thesis to estimate the available bandwidth for admission control was presented. Finally, in this section, a description of how the proposed approach used differs from the original/standard and the state-of-the-art available bandwidth measurement was highlighted.

5.2.Definition of the Available Bandwidth

In a wireless medium, the available bandwidth is a network characteristic that is dependent on both the link and the link direction. The available bandwidth that is associated with different link directions with the same node will have different values. This is mainly because the available bandwidth of different links or directions will generate different interference.

The available bandwidth of a network is therefore defined as the maximum transmission throughput between two neighbour node in a given transmission direction, having a condition that the QoS of any ongoing packet within that medium is not disrupted. From the transmission point of view, the bandwidth used by any ongoing transmission and the transmission QoS requirements should be considered during the available bandwidth estimation. From the channels point of view, the available bandwidth is associated with the effective channel capacity on the available idle time of the channel at a given period of time.

Therefore, it is important to note that the standard available bandwidth estimation formula of a wireless channel is majorly based on the effective channel capacity and the available channel idle period.

The channel capacity in the available bandwidth estimation does not only depends on the data rate set by the physical interface card, but also on other random factors in the network. For the available channel idle period to be accurately determined, the meaning of *availability* in IEEE 802.11e will be clarified.

Before defining the concept of *availability,* it is essential to be familiar with the following three fundamental terms with respect to the available bandwidth range distance. These are; carrier sensing range (*CSR*), transmission range (*TR*), and outside zone range (*OZR*). The carrier sensing range, which covers a double distance circularly around the node, is the distance through which a node can sense the transmission of another node based on carrier sensing threshold. The transmission range gives information about the node activities through the clear channel assessment (*CCA*) provided by the MAC layer and it is the distance through which a node can receive and decode a signal correctly, given that there is no interference around. The outside zone range covers the distance from the sending node to any node beyond the carrier sensing node, i.e., any transmission occurring outside the transmission range and the carrier sensing range.

The *availability* channel period between the source and the destination link of a wireless channel is hereby achieved given that the following five conditions are satisfied.

i. No node within the *CSR* of the source node *S* is sending packets. This condition, however, meets the wireless channel requirement, i.e., the wireless channel should be idle before transmitting a packet.
ii. No node within the *CSR* of the source node *S* is receiving packets. For example, within the *CSR o*f *S*, if *S* sends a data, and other nodes simultaneously receives data, this will result in collision. Therefore, the channel occupied by the receiving nodes within the *CSR* of the *S* is not available for *S.*
iii. No node within the *CSR* of the destination *D* is sending packets. This condition guarantees that the packet sent by *S* will be received by *D* without collision.
iv. No node within the *CSR* of the destination *D* is receiving packet.
v. No node within the *CSR* of the source/destination (*S/D)* has a non-empty queue.

Once all the above five conditions are met, then it is guaranteed that there is channel *availability* for a given link. However, the throughput of this channel *availability* is said to be the available bandwidth that can be used without obstructing any existing flow. The *link carrier sensing area* is defined as the union between the carrier sensing area of two ends of the link. Giving the topology below as in Figure 5.1, the dashes round about the circular lines represents the *link carrier sensing area* of link *S* and *D.* Therefore, any frame sent within the *link carrier sensing area* of *S* and *D* will make the channel *S* and *D* to be unavailable, thereby violating the above stated condition of i and iii. Given that the outer area (OZR) of the *link carrier sensing area* of *S* and *D* sends a packet to the destination located within the dashes round about the circular lines (CSR), the available bandwidth will be affected, thus violating the condition stated in ii and iv. For example, the data received by 5(2) from 6(3) may collide with the data sent by *S* (*D).* Also, giving that any of the nodes within the *CSR* of *S/D* has a queue of packet within it, this may cause the channel to be unavailable in-case there is a trigger which may cause a shift to enhance the node in the *CSR* to start sending packet. This, therefore, will tend to violate the condition stated in v. All these complications however make it more difficult to accurately estimate the available bandwidth on a network.

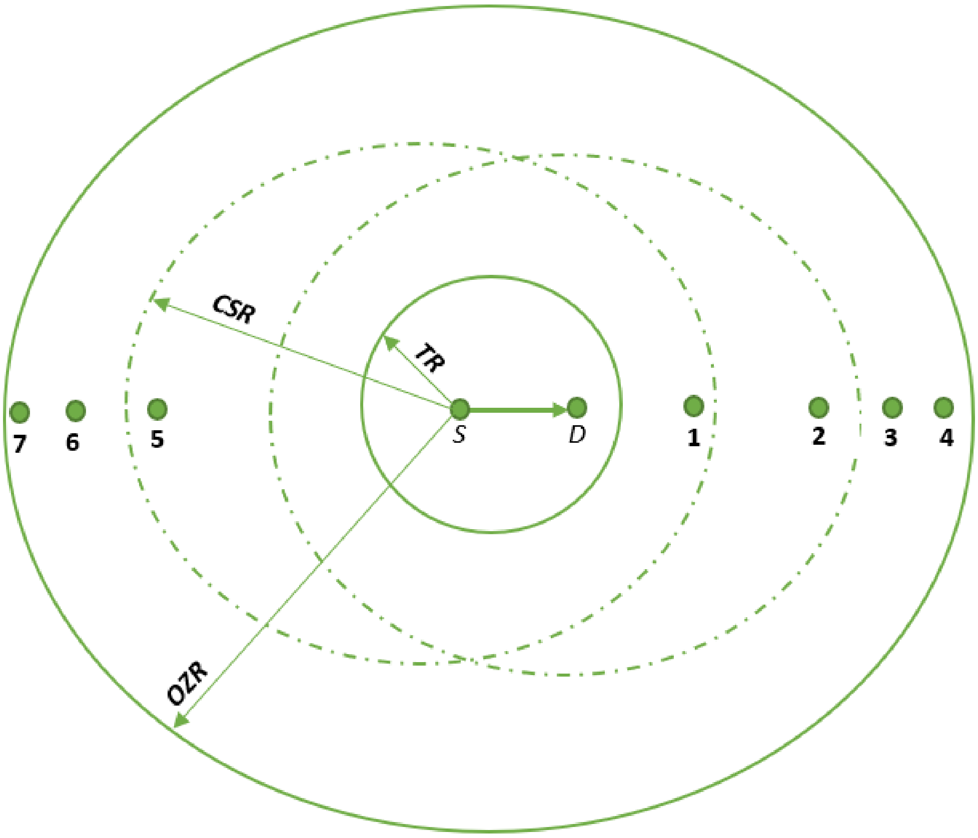


Figure 5.1 Availability Definition

5.3. Definition of the Different Wireless Network States

A node is said to be in a state of transmission, only if it is currently emitting signals through its antenna. A node is said to be in a receiving state if there are nodes transmitting within its transmission range. A node is said to be in a sensing state if the medium is sensed busy but not receiving frames because the energy is below the receiving threshold. A node is said to be in an idle state if it is not transmitting, receiving, or sensing any packet.

It is very important to note that the BUSY state of a channel, as defined in this thesis, is a situation whereby a node is in the transmission or receiving state, while the SENSE BUSY state is defined as a situation whereby a node is in the sensing state. Any other time outside the sensing time, the node will be in an IDLE state. The IDLE state, however, is defined as a situation whereby a node is neither Busy, Sense busy or has a non-empty queue.

5.4. Proposed Approach to Estimate the Available Bandwidth for Admission Control: RAACM (Resource Allocation and Admission Control in MANET)

This thesis propose a Resource Allocation and Admission Control in MANET (RAACM) estimation method to improve the accuracy and efficiency of the available bandwidth estimation. RAACM therefore considers the following: (i) The channel idle time synchronization and dependency between the sender and the receiver by differentiating the *BUSY* state from the *SENSE BUSY* state and the *IDLE* state caused by an empty queue. (ii) HELLO packet propagation to retrieve the available bandwidth on the carrier sensing region.

Based on the definition of the available bandwidth given in section 5.2, the approach used in this thesis to accurately and efficiently estimate the available bandwidth is presented

As highlighted in section 5.2, it is important to note that the standard formula for the available bandwidth estimation in a wireless channel is majorly based on the effective channel capacity and the available channel idle period. Therefore, the available bandwidth of link l_{sd} (denoted as AB_{sd}) is originally computed using equation 1 below;

$$AB_{sd} = \frac{T_{idle(sd)}}{T} \times C \qquad (1)$$

Where $T_{idle(sd)}$ is the available channel idle time period and C is the maximum effective channel capacity.

Below is the proposed algorithm used to accurately estimate the available channel idle time period and the effective channel capacity of a link.

5.4.1. Estimating the Available Channel Idle Time Period in RAACM

Based on the definition given for the idle state in section 5.3, RAACM estimates the channel idle time as;

$$T_{idle(sd)} \approx T - T_B - T_{SB} - T_E \qquad (2)$$

Where T is the total time period, T_B is the time duration when the station is *BUSY*, T_{SB} is the time duation when the staion is *SENSING BUSY,* and , T_E is the time duration when the station has an *EMPTY QUEUE.*

Therefore, the available channel idle time between link *s,d* at a given period can be rewritten as;

$$\frac{T_{idle(sd)}}{T} \approx \frac{T-T_B-T_{SB}-T_E}{T} \qquad (3)$$

Equation 3 means that for a channel to be regarded as idle, the channel must not be busy, sensing busy and must not have any pending frame for transmission within the interface (i.e. the interface queue must be empty).

The five conditions previously stated in section 5.2 that guarantees the availability of the channel further buttress this explanation. For example, in order for a channel to be regarded as idle, the channel must not be busy and sense busy which satisfies bullet points i, ii, iii, and iv. Also, for a channel to be regarded as idle, the channel must not have any pending frame for transmission within the interface (i.e. the interface queue must be empty), which satisfies bullet point v.

5.4.1.1. Illustration showing the BUSY state and the SENSE BUSY state and IDLE state of a channel link

To illustrate the relationship between BUSY state, the SENSE BUSY state and the IDLE state, let us consider the scenario given in the figure 5.2 below; where S is the sender and D is the destination (S is transmitting to D). Table 5.1 shows the different channel states sensed by all nodes in figure 5.2. It is important to note that all nodes within the transmission range of S are regarded to as been busy because they can decode any packet and are able to know the time a given transmission will finish. At this time, they are in the receiving state, which is BUSY.

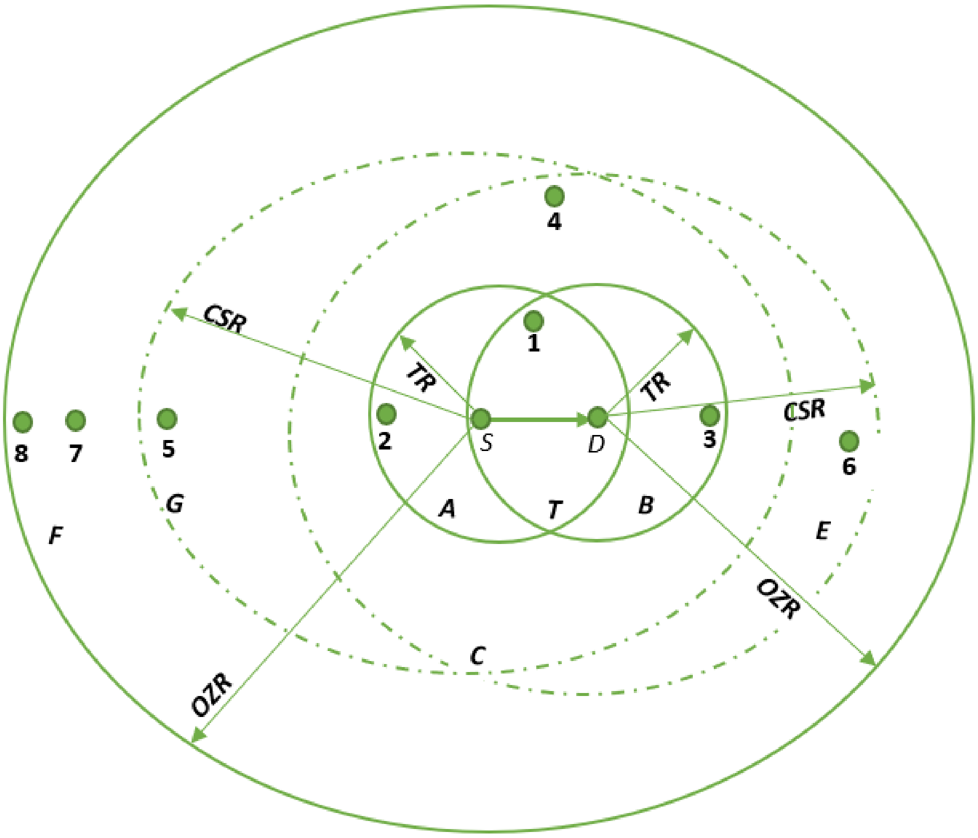


Figure 5.2 Transmission Range and Carrier Sensing Range

CHANNEL NODES (S -> D)	BUSY	SENSE BUSY	IDLE
Sender (S)	✓		
Receiver (D)	✓		
1	✓		
2	✓		
3		✓	
4		✓	
5		✓	
6			✓
7			✓
8			✓

Table 5.1: Scenario showing the different channel states sensed by nodes in Figure 5.2

5.4.1.2.Channel idle time synchronization and dependency between the sender and the receiver

It is important to note that the synchronization of the channel idle time can be analysed based on node dependant (nodes that are not randomly distributed) and non-dependant (nodes that are randomly distributed). It is also of importance to note that synchronization between a sending node and a receiving node can be analysed based on non-overlapping channel and complete overlapping channel idle time, an example of this is shown in Figure 5.3 below.

This thesis therefore incorporates a dependant channel idle time synchronization, where nodes within the interference range of the sender and the receiving node are not randomly distributed. Also, the synchronization is considered by allowing the sender and the receiver node to witness both common interference (complete overlap) and independent interference (no overlap).

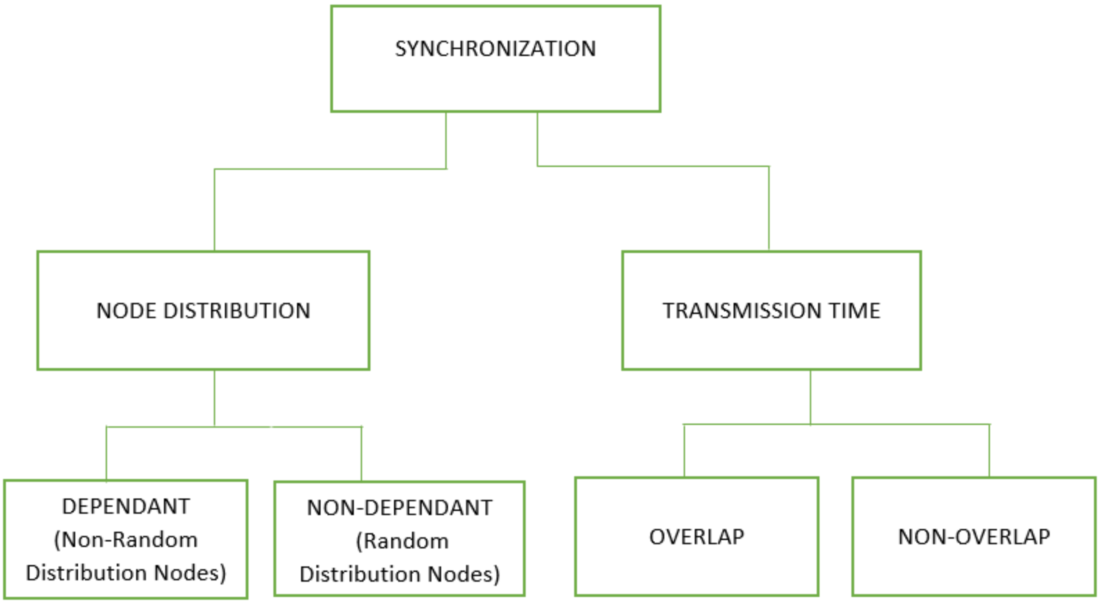


Figure 5.3 Synchronization Classification

Figure 5.4a and 5.4b, represents the medium availability along the sending and receiving node. In both scenarios, the channel idle time ratio measured at each of the node is 50%.

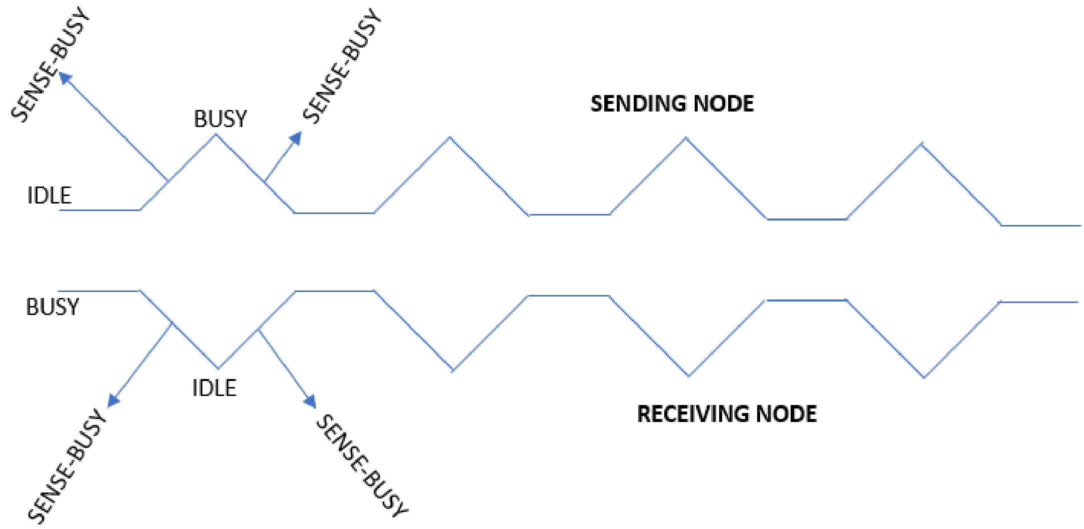


Figure 5.4a Synchronization between the Sending node and the Receiving node with no-overlap

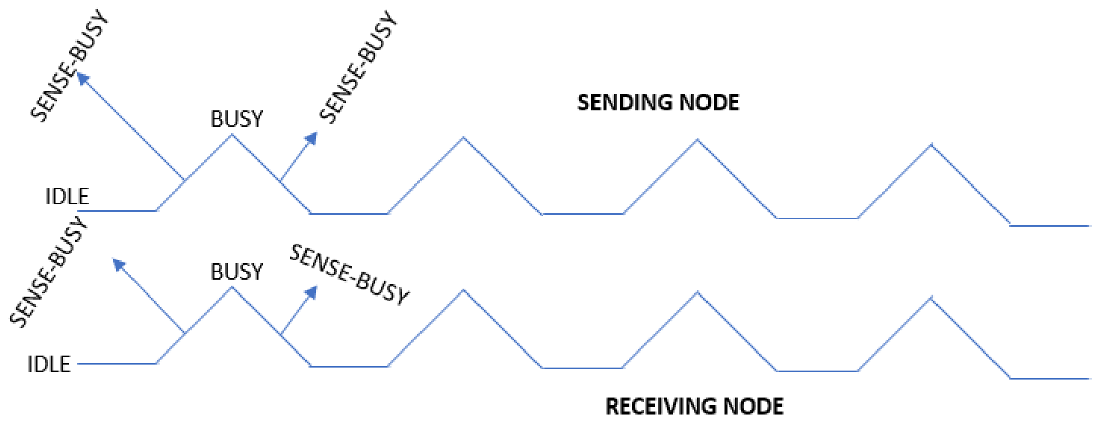


Figure 5.4b Synchronization between the Sending node and the Receiving node with Complete Overlap

In Figure 5.4a, the synchronized idle channel time does not overlap, and the available bandwidth on the link is null. This is because the medium available period between the sender and the receiver does not overlap each other. Figure 5.4b depict a complete opposite scenario as Figure 5.4a. Here, the synchronized idle channel time is completely overlapped (i.e. the sending node and the receiving node are both idle at the same time. The same principle is applied to the busy and the sense busy node) and the available bandwidth on the link is 0.5 given that the total channel capacity is 1.

However, considering the transmission time synchronization with respect to complete idle channel period overlapping in Figure 5.4b, this will result in the overestimation of the available bandwidth. Also, if the transmission time synchronization with respect to the non-overlapped idle channel time is considered in Figure 5.4a, this will result in the underestimation of the available bandwidth. Therefore, combining the common interference (complete overlap) and independent interference (no overlap) will provide an efficient and accurate bandwidth estimation in terms of the transmission time.

It is very important to note that considering the transmission time synchronization alone for the channel idle time measurement will still result in an inaccurate estimation of the available bandwidth. Therefore, this thesis considers a dependant node distribution synchronization which does not support a random distribution of node between the sending node and the destination node to provide an efficient and accurate bandwidth estimation.

An illustration showing the need to consider the dependency of sender idle time and receivers idle time is depicted in Table 5.2a and Table 5.2b below. Using Figure 5.2, Table 5.2a shows all the possible communications between the sender (*S*) and the receiver (*D*), considering other neighbouring nodes within the network. Also, all the possible state of *S* and *D* based on Table 5.2a are sketched in Table 5.2b below. It is important to note that there may be a shift in the time duration/intervals of the communication between the source and destination.

In Table 5.2a, the interval row represents the different period of occupancy observed by the source (*S*) and the destination node (*D*). The sender/receiver's communication row represents the behaviour of the source and destination at different intervals. As previously mentioned Table 5.2a was generated from Figure 5.2 and only *S* and *D* behaviour is observed. Therefore, in interval 1, *S* can send a packet to *D* and *D* can receive a packet from *S*. In interval 2, *S* may not send packet to *D* (*S* may send a packet to any other node in its transmission range). In interval 3, *S* is a member of *G* (*S* is in the carrier sensing range of *G*). In interval 4, *S* and *D* is a member of *C* (link of the carrier sensing range of *S* and *D*). In interval 5, *D* may not receive from *S* (*D* may receive from any node in the transmission range of *S*). In interval 6, *D* is a member of *E* (*D* is in the carrier sensing range of *E*). In interval 7, *S* and *D* is not a member of *F*.

Let us take a close look at Table 5.2b (derived from using Table 5.2a), it is clearly seen that, intervals 1, 4, and 7 totally overlapped and synchronized. However, intervals 2 (5) and 3(6) will affect the bandwidth estimation and can cause the bandwidth to be under-estimated because at these intervals, their synchronization does not totally overlap. During this period, one of the nodes is BUSY while the other one is SENSING BUSY (Intervals 2 and 5) and also, one of the nodes is SENSING BUSY while the other one is IDLE (Intervals 3 and 6). These two non-overlapped synchronized periods therefore need to be properly addressed to guarantee proper bandwidth estimation.

i. *To address the period where one node is BUSY and the other one is SENSING BUSY (Interval 5 and 2);*

Let M_1 be the probability that S is SENSING BUSY and D is BUSY and M_2 be the probability that D is SENSING BUSY and S is BUSY. The probability that intervals 5 and 2 will occur are *M (interval 5 appears)* = $M_1 \left(\frac{T_{SB}^S}{T} \times \frac{T_B^D}{T}\right)$; *M (interval 2 appears)* = $M_2 \left(\frac{T_{SB}^D}{T} \times \frac{T_B^S}{T}\right)$. Where T_{SB}^X denotes the SENSE BUSY interval sensed by a given node x at a given point in time and T_B^X denotes the BUSY interval sensed by a given node x at a given point in time.

Therefore, the synchronised and dependency between a sending node and the destination node with respect to addressing the period where one node is BUSY and the other one is SENSING BUSY are shown in equations 4 to 7 below;

$$T_{SB(S)} \approx \frac{T_{SB}^S \times (1 - M\ (Interval\ 5\ appears))}{T} \approx \frac{T_{SB}^S \times (1 - M_1 \times \frac{T_B^D}{T})}{T} \tag{4}$$

$$T_{SB(D)} \approx \frac{T_{SB}^D \times (1 - M\ (Interval\ 2\ appears))}{T} \approx \frac{T_{SB}^D \times (1 - M_2 \times \frac{T_B^S}{T})}{T} \tag{5}$$

$$T_{B(S)} \approx \frac{T_B^S \times (1 - M\ (Interval\ 2\ appears))}{T} \approx \frac{T_B^S \times (1 - M_2 \times \frac{T_{SB}^D}{T})}{T} \tag{6}$$

$$T_{B(D)} \approx \frac{T_B^D \times (1 - M\ (Interval\ 5\ appears))}{T} \approx \frac{T_B^D \times (1 - M_1 \times \frac{T_{SB}^S}{T})}{T} \tag{7}$$

ii. *To address the period where one node is IDLE and the other one is SENSING BUSY (intervals 3 and 6);*

Let M_3 be the probability that S is SENSING BUSY and D is IDLE and M_4 be the probability that D is SENSING BUSY and S is IDLE. The probability that Interval 3 and 6 will occur are

M (intervals 3 appears) = $M_3 \times \frac{T_{SB}^{S}}{T}$; *M (interval 6 appears)* = $M_4 \times \frac{T_{SB}^{D}}{T}$. Remember that T_{SB}^{X} denotes the SENSE BUSY interval sensed by a given node *x* at a given point in time.

Therefore, the synchronised and dependency between a sending node and the destination node with respect to addressing the period where one node is IDLE and the other one is SENSING BUSY is shown in equation 8 and 9 below;

$$T_{idle(S)} \approx \frac{T_{idle}^{S} \times (1 - M\ (Interval\ 6\ appears))}{T} \approx \frac{T_{idle}^{S} \times (1 - M_4 \times \frac{T_{SB}^{D}}{T})}{T} \quad (8)$$

$$T_{idle(D)} \approx \frac{T_{idle}^{D} \times (1 - M\ (Interval\ 3\ appears))}{T} \approx \frac{T_{idle}^{D} \times (1 - M_3 \times \frac{T_{SB}^{S}}{T})}{T} \quad (9)$$

Based on the pattern followed by i, and ii, we assume that by considering the dependency and synchronization of the time duration when the station has an *EMPTY QUEUE* within its interface, equation 10 and 11 is derived. Where M_5 is the probability that *S* is SENSING BUSY and *D* is having an EMPTY *QUEUE* and M_6 be the probability that *D* is SENSING BUSY and *S* is EMPTY QUEUE.

$$T_{E(S)} \approx \frac{T_{E}^{S} \times (1 - M_6 \times \frac{T_{SB}^{D}}{T})}{T} \quad (10)$$

$$T_{E(D)} \approx \frac{T_{E}^{D} \times (1 - M_5 \times \frac{T_{SB}^{S}}{T})}{T} \quad (11)$$

Intervals	1	2	3	4	5	6	7
Sender/Receiver's Communication	S<=>D or (S⊂T)	S <≠> D or (S⊂A)	S ⊂ G	S, D ⊂ C	D <≠> S or (D⊂ B)	D ⊂ E	S/D ≠> F

Table 5.2a: llustration showing Dependency Consideration

Intervals	1	2	3	4	5	6	7
SOURCE (S)	BUSY	BUSY	SENSE-BUSY	SENSE-BUSY	SENSE-BUSY	IDLE	IDLE
DESTINATION (D)	BUSY	SENSE-BUSY	IDLE	SENSE-BUSY	BUSY	SENSE-BUSY	IDLE

Table 5.2b: Interpretation of Table 5.2a

Recall that $\frac{T_{idle(sd)}}{T} \approx \frac{T-T_B-T_{SB}-T_E}{T}$, based on this we therefore have;

$$\frac{T_{idle(sd)}}{T} \approx \frac{T-\left[\left(\frac{T_B^S \times (1-M_2 \times \frac{T_{SB}^D}{T})}{T}\right)\left(\frac{T_B^D \times (1-M_1 \times \frac{T_{SB}^S}{T})}{T}\right)\right]-\left[\left(\frac{T_{SB}^S \times (1-M_1 \times \frac{T_B^D}{T})}{T}\right)\left(\frac{T_{SB}^D \times (1-M_2 \times \frac{T_B^S}{T})}{T}\right)\right]-\left[\left(\frac{T_E^S \times (1-M_6 \times \frac{T_{SB}^D}{T})}{T}\right)\left(\frac{T_E^D \times (1-M_5 \times \frac{T_{SB}^S}{T})}{T}\right)\right]}{T} \quad (12)$$

5.4.1.3.Estimating the Maximum Sensing Range of the Available Channel Idle Time Period

In RAACM, the sending node, the destination node and the transmission of other neighbouring node will contend for the channel. It is therefore necessary to take note of the maximum range a node is allowed to sense. We therefore propose and use a new sensing threshold that not only monitor the transmission range or the carrier sensing range, but will monitor any neighbouring transmission outside the carrier sensing range that may possibly affect the estimation of the channel idle time.

With reference to section 5.2, i, ii, iii, iv, and v will have an impact on the channel availability, however, their impact may vary. There is therefore a need to properly choose the range of sensing to prevent improper network estimation of the channel idle time. For a source to detect any nearby transmission, any of the three ranges are to be monitored from the source/destination to the Transmission range, Carrier Sensing range, or Outside zone range (check the diagram in Figure 5.2). Therefore, since the outside zone range tends to be the maximum distance range amongst the three ranges, our proposed RAACM selects the outside zone range to be the peak a node is allowed to sense in order to check that the channel is idle. The outside zone range is denoted as OZR, hence, the corresponding sensing threshold is represented as OZR-threshold. However, just like the carrier sensing range threshold (CSR-threshold), OZR-threshold has an additional feature to sense the medium range of nodes that are beyond the carrier sensing range. Hence, using the OZR-threshold, the source of the target link can sense the transmission and determine whether the medium is available for the transmission to be granted.

At a given time period T_p, let T_{busy}^{ozr} denote the channel busy time that is sensed by a sender when an OZR-threshold is used and let T_{busy}^{csr} denote the channel busy time that is sensed by a sender when CSR-threshold is used. However, since we are making use of a larger sensing range, the channel can claim to be too busy and the channel busy time sense by the OZR-threshold may under-estimate the available idle channel time. A typical example of this is as follows; considering the diagram in Figure 5.1, when the transmission are located outside the dashes round about the circular lines, this means that the transmitting nodes (e.g. transmission from node 7 to node 6 or transmission from node 4 to node 3) are not within the channel link of *S* and *D*. In RAACM, this kind of transmission is denoted *as no-impact occasion.* Since it is not possible for the passive sensing to detect the *no-impact occasion,* the OZR-threshold approach will therefore cause the channel idle time to under-estimate. In Figure 5.1, let us denote the outside of the dashes round about the circular lines as a *special area (SA)* of link S and *D*. Therefore, if a node in the *SA* of link *S* and *D* sends a packet, and the destination is located outside the link carrier sensing area of *S* and *D*, the transmission will not affect the available channel. However, the two cases given below will have an effect on the channel availability;

- If the destination of a transmitting frame is located in the carrier sensing range of node *S* (e.g., if node 6 is transmitting to node 5), the receiving frame (node 5) may interfere with the packet sent by node *S*.
- If the destination of a transmitting frame is not located in the carrier sensing range of node *S*, but it is located in the carrier sensing range of node *D* ((e.g. if node 3 is transmitting to node 2), the receiving frame (node 2) may interfere with the acknowledgement frame sent by node *D*).

Therefore, based on the above analysis, the *no impact occasion* happens to be the only reason in which a channel will lead to under-estimation if an OZR-threshold is used for monitoring the idle channel time. By using the OZR-threshold to monitor the channel idle time, we assume that the channel time used for transmission in the *SA* at a given time period T_p is T_{SA}. We therefore define *Standard probability* of a given link (denoted by P_c) as the probability that the destination of a transmission in the *SA* is not situated within the link carrier sensing of *S* and *D*. Therefore, the available channel idle time underestimated which is sensed by the *OZR-threshold* can be compensated by estimating the channel time occupied through the *no impact occasion.* Therefore, the range of the channel idle time when node *S* is transmitting to node *D* is denoted as $R_{S,D}$ and is computes as;

$$R_{S,D} = \frac{P_C\,(T_{OZR} - T_{SA})}{T_p} \tag{13}$$

$$T_{SA} = \left(\frac{T_{OZR} - T_{CS}}{\pi R_{OZR}^2 - \pi R_{CS}^2}\right) A_C \tag{14}$$

Where A_c denotes the area of SA. Therefore, the range of the channel idle time when node S is transmitting a packet to node D is finally computed as;

$$R_{S,D} = \frac{P_C\,\left(T_{OZR} - \left(\frac{T_{OZR} - T_{CS}}{\pi R_{OZR}^2 - \pi R_{CS}^2}\right) A_C\right)}{T_P} \tag{15}$$

Finally, the channel idle time with respect to the range propagation is expressed as;

$$\frac{T_{idle(sd)}}{T} \approx \left(\frac{T - \left[\left(\frac{T_B^S \times (1 - M_2 \times \frac{T_{SB}^D}{T})}{T}\right)\left(\frac{T_B^D \times (1 - M_1 \times \frac{T_{SB}^S}{T})}{T}\right)\right] - \left[\left(\frac{T_{SB}^S \times (1 - M_1 \times \frac{T_B^D}{T})}{T}\right)\left(\frac{T_{SB}^D \times (1 - M_2 \times \frac{T_B^S}{T})}{T}\right)\right] - \left[\left(\frac{T_E^S \times (1 - M_6 \times \frac{T_{SB}^D}{T})}{T}\right)\left(\frac{T_E^D \times (1 - M_5 \times \frac{T_{SB}^S}{T})}{T}\right)\right]}{T} \right) \times R_{S,D} \tag{16}$$

5.4.2. Effective Channel Capacity

As previously highlighted in section 5.2, the available bandwidth estimation in a wireless channel is also dependent on the effective channel capacity. It is therefore important to note that the value of the channel capacity will not be the total/raw value of the medium capacity as given by the standard (IEE802.11e), but some fixed overheads (e.g. headers, acknowledgement, rts and cts) that are introduced by the MAC protocol need to be accounted for.

A typical example of this is as follows; Given that the total capacity assigned by a wireless standard to a medium is 54-Mbps, the throughput delivered by this capacity is not expected to be higher than 33.2Mbps.

It is also very important to note that the available bandwidth estimation is dependent on some other factors by the reason of its standards asides the channel idle time and the capacity for proper functionality and estimation to be carried out. These factors are, collision and back-off. For the purpose of clarity, let us consider the frame exchange sequence for each attempt of packet transmission in wireless 802.11e EDCA as in Figure 5.5. For a transmission to occur between station 1 and station 2, there are various factors that will automatically have an impact (i.e. consume the bandwidth) on the available bandwidth of both stations, these are; AIF, back-off, RTS, CTS, and Acknowledgement.

However, for the purpose of this thesis, the RTS, CTS, and Acknowledgement will be included in the calculation of capacity C (C denotes the capacity of the available bandwidth as shown in equation 1). The AIF will be included in the calculation of the collision probability, and lastly, the back-off duration will also be addressed as the other major factor that have an impact in the available bandwidth estimation process.

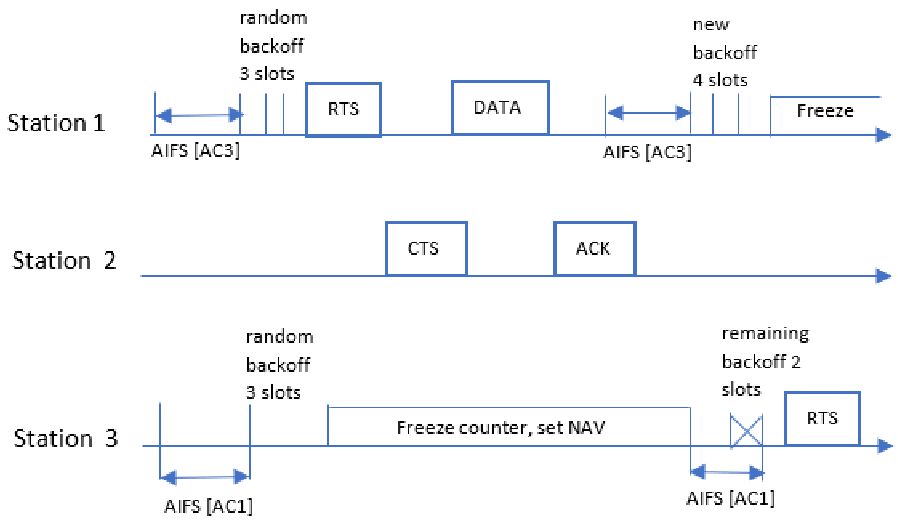


Figure 5.5: IEEE802.11e EDCA frame exchange sequence (Peng, 2012)

5.4.3. *Other Factors that have an impact on the Available Bandwidth Estimation*

As previously mentioned in section 5.4.2, the available bandwidth estimation is dependent on some other factors by the reason of its standards asides the channel idle time and the capacity for proper functionality and estimation to be carried out. These factors are; collision and back-off. It is important to note that the reason for considering collision in our estimated available bandwidth is that due to the nature of wireless, there may still be some slight level of inaccuracy during the available bandwidth measurement which may sometimes result in collision. Therefore, it is wise to include the collison probability in our estimated available bandwidth process in case it occurs. Once there is a collision this will eventually lead to a backoff. To calculate the collision probability and the back-off duration, the formula in the work of (Sharma et al, 2018) is used.

5.5.Estimation of Available Bandwidth Using RAACM

The available bandwidth (AB_{sd}) between link s and d (l_{sd}) proposed in this thesis is based on consideration of the channel idle time synchronization and dependency between the sender and the receiver, through the differentiation of the *BUSY* state from the *SENSE BUSY* state and the *IDLE* state caused by an empty queue. This is estimated in equation 17 below;

$$AB_{SD} = (1 - Cl) \times (1 - Bo) \times \left(\frac{T - \left[\left(\frac{T_B^S \times (1 - M_2 \times \frac{T_{SB}^D}{T})}{T}\right)\left(\frac{T_B^D \times (1 - M_1 \times \frac{T_{SB}^S}{T})}{T}\right)\right] - \left[\left(\frac{T_{SB}^S \times \left(1 - M_1 \times \frac{T_B^D}{T}\right)}{T}\right)\left(\frac{T_{SB}^D \times (1 - M_2 \times \frac{T_B^S}{T})}{T}\right)\right] - \left[\left(\frac{T_E^S \times (1 - M_6 \times \frac{T_{SB}^D}{T})}{T}\right)\left(\frac{T_E^D \times (1 - M_5 \times \frac{T_{SB}^S}{T})}{T}\right)\right]}{T} \right) \times C \times R_{S,D} \quad (17)$$

Where *Cl* denotes the collision probability, *Bo* denotes the Back-off duration of bandwidth consumed and *C* denotes the capacity of the link, $R_{S,D}$ is the maximum propagation range.

5.5.1. Description of How Our Proposed RAACM Formula Differs from the Original Bandwidth Estimation Measurement

As earlier mentioned, the contribution of this thesis is focused on the channel idle time measurement, which is part of the available bandwidth estimation formula.

Recall that the available bandwidth estimation used in this thesis is a passive bandwidth estimation which is generally estimated as the product of the channel idle time period and the capacity. Therefore, the standard available bandwidth between the source and destination link of a channel is expressed in the equation below;

$$AB_{sd} = \frac{T_{idle(sd)}}{T} \times C_{sd} \quad (19)$$

In this thesis, we modified how the channel idle time ($T_{idle(sd)}$) is computed. The modification and explanation of the channel idle time period, as used in this thesis, can be found in section 5.4.1.

5.5.2. Difference between our Proposed RAACM Modification and BECIT, MBA-AODV, and AABWM

It is important to note that BECIT (Chaudhari & Biradar, 2015), MBA-AODV (Sharma et al, 2018), and AABWM (Mukta & Gupta, 2019) which is the state-of-the-art available bandwidth estimation for admission control technique estimates the channel idle time in a similar way, and we will be describing the way their channel idle time was formulated and how it differs from our proposed RAACM in this section.

In BECIT, MBA-AODV, and AABWM, the authors estimate the channel idle period using the overlapping probability of two ends idle time in order to consider the synchronization between the sender and the receiver. However, for the synchronization to occur, each surrounding node around the sender and the receiver views each nodes of the channel as being a uniformly random distribution and are independent of one another. This assumption is however very crucial and vital to the calculation of the channel idle time in order to give a better accuracy during the bandwidth estimation.

For better illustration, we analyse the synchronization of the channel idle time by dividing it into synchronization that is based on node dependant (nodes that are not randomly distributed) and synchronization based on node that are non-dependant (nodes that are randomly distributed).

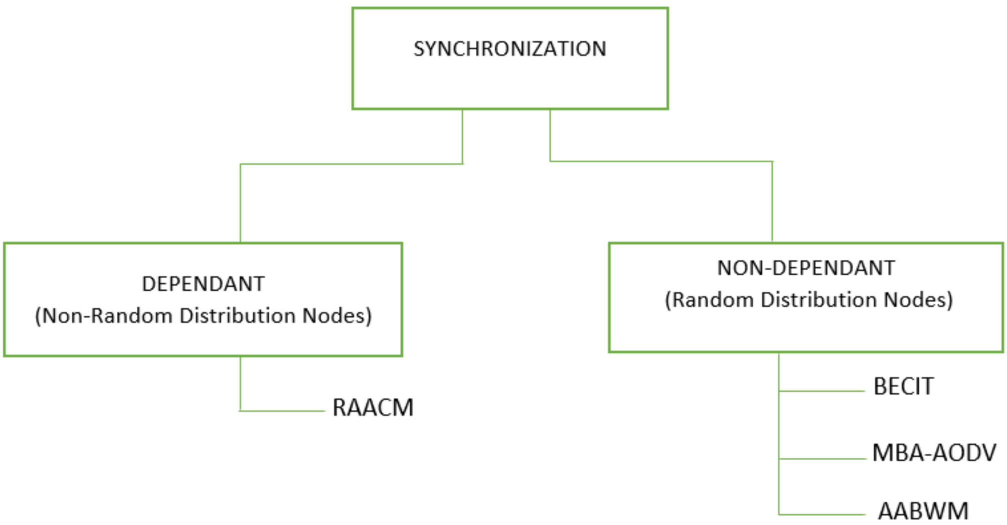


Figure 5.6: Synchronization

Therefore, what makes our proposed RAACM different from BECIT, MBA-AODV and AABWM, is that in RAACM, nodes are dependent on one another for proper channel idle time bandwidth estimation unlike BECIT, MBA-AODV and AABWM where nodes are randomly distributed which may result in an inaccurate available bandwidth estimation.

For example, considering the previous scenario used in Figure 5.4b, the medium availability of this scenario given that the total channel capacity is 1 will be equal to 0.5. However, since the available bandwidth measurement with respect to the channel idle time dependency for BECIT, MBA-AODV and AABWM have a non-dependant node feature and, the nodes are randomly distributed, the channel idle time between a source and destination was calculated as a product of the idle time of the source node and the idle time of the destination node. Therefore, the source node idle channel time is uniquely estimated as well as the destination idle channel time. Based on this, the available bandwidth between link *s* and *d* is estimated as;

$$AB_{SD} = \frac{T_{idle(s)}}{T} \times \frac{T_{idle(d)}}{T} \times C \qquad (20)$$

The channel idle time is then estimated as $T_{idle(sd) \approx T_{idle(s)} \times T_{idle(d)}}$, which is equivalent to 0.5 × 0.5 = 0.25. Giving that T (period) is 1.

On the other hand, since RAACM's channel idle time period synchronization have a dependant node feature and are not randomly distributed, this will enable the source node to be dependent on the destination node. Therefore, RAACM considers and estimate their channel idle time dependant synchronization by allowing the sender and the receiver node to witness both common interference and independent interference. The channel idle time synchronization and dependency between the sender and the receiver in RAACM is therefore estimated by differentiating the *BUSY* state from the *SENSE BUSY* states and the *IDLE* state caused by an empty queue. This is achieved with respect to the common and independent interference between the source and the destination node.

Further analysis and estimation of the way our proposed RAACM considers the sender and the receiver's channel idle time dependency is found in section 5.4. However, channel idle time formula for RAACM is estimated as; $T_{idle(sd) \approx T - T_B - T_{SB} - T_E}$

Based on this, the available bandwidth proposed for RAACM's available bandwidth between link *s* and *d* is estimated as;

$$AB_{SD} = \frac{T - T_B - T_{SB} - T_E}{T} \times C \qquad (21)$$

Where T is the total time period, T_B , is the time duration when the station is *BUSY*, T_{SB} is the time duation when the staion is *SENSING BUSY,* and , T_E is the time duration when the station has an *EMPTY QUEUE* and *C* is the capacity (this formula does not include the other factors that affect the available bandwidth).

5.6.**Distributed Flow Admission Control**

The previous section described the process of estimating the available bandwidth for admission control in MANET. We identified the important factors to be considered for available bandwidth estimation for admission control design.

The admission control mechanism is deployed in order to decide if a flow should be transmitted or not based on the available bandwidth estimated. Therefore, a procedure must be followed as to whether to accept a flow or reject it if, the estimated available bandwidth as shown in equation 17 is greater than the requested bandwidth. This decision is taken at the designated mobile node using a distributed admission control mechanism.

RAACM admission control proposed in this thesis includes two processes which are performed at the source node and the intermediate node. Having estimated the available bandwidth using equation 17, suppose a node needs to forward a packet frame, the admission controller at the ingress and egress node determines whether the available resources can meet the requirement of a new flow.

The procedure followed by the source node is shown in Figure 5.7a below. Once the new packet flow is generated at the application layer, the packet is classified into real-time traffic and best effort traffic and prioritization is assigned to the packet based on its classification. The packet are allowed transmission based on their priority. Thereafter, the source node performs a check to see if the available bandwidth estimated using equation 17 is able to meet the demand of the newly requested flow. If the demand of the requesting flow cannot be met, that flow will be rejected. However, if the demand can be met, the flow is accepted, and a broadcast message is forwarded to the intermediate node with a timer issued to indicate the time the transmission intends to finish. Additionally, there is also a part for acknowledgement reception from the destination node. If a message received indicates an acknowledgement message, this allows any intending packet to be forwarded as indicated. Figure 5b shows the procedure followed by the intermediate node. At the intermediate node, packets are accepted, and a check is performed to enquire the type of packet. If the packet is a resource request flow, the flow parameter can be obtained, else flow packets are handled based on their request. Thereafter, another admission control request takes place using the same equation 17. If the requested bandwidth is less than the available estimated bandwidth, packet flow is forwarded to its destination node, and an acknowledgement is sent back to the source node. If otherwise, packet flow stop forwarding.

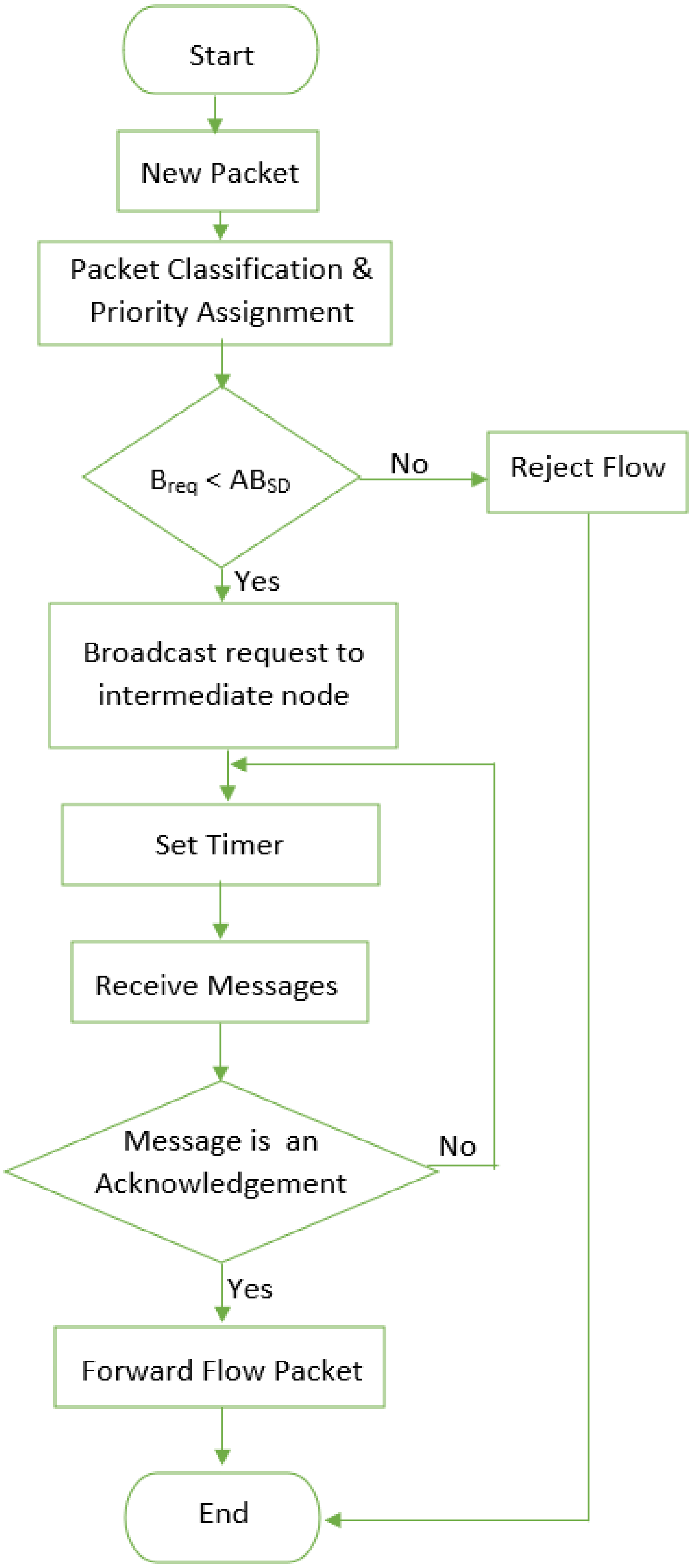


Figure 5.7a: Admission Control Protocol at the Source node for Bandwidth Estimation in RAACM

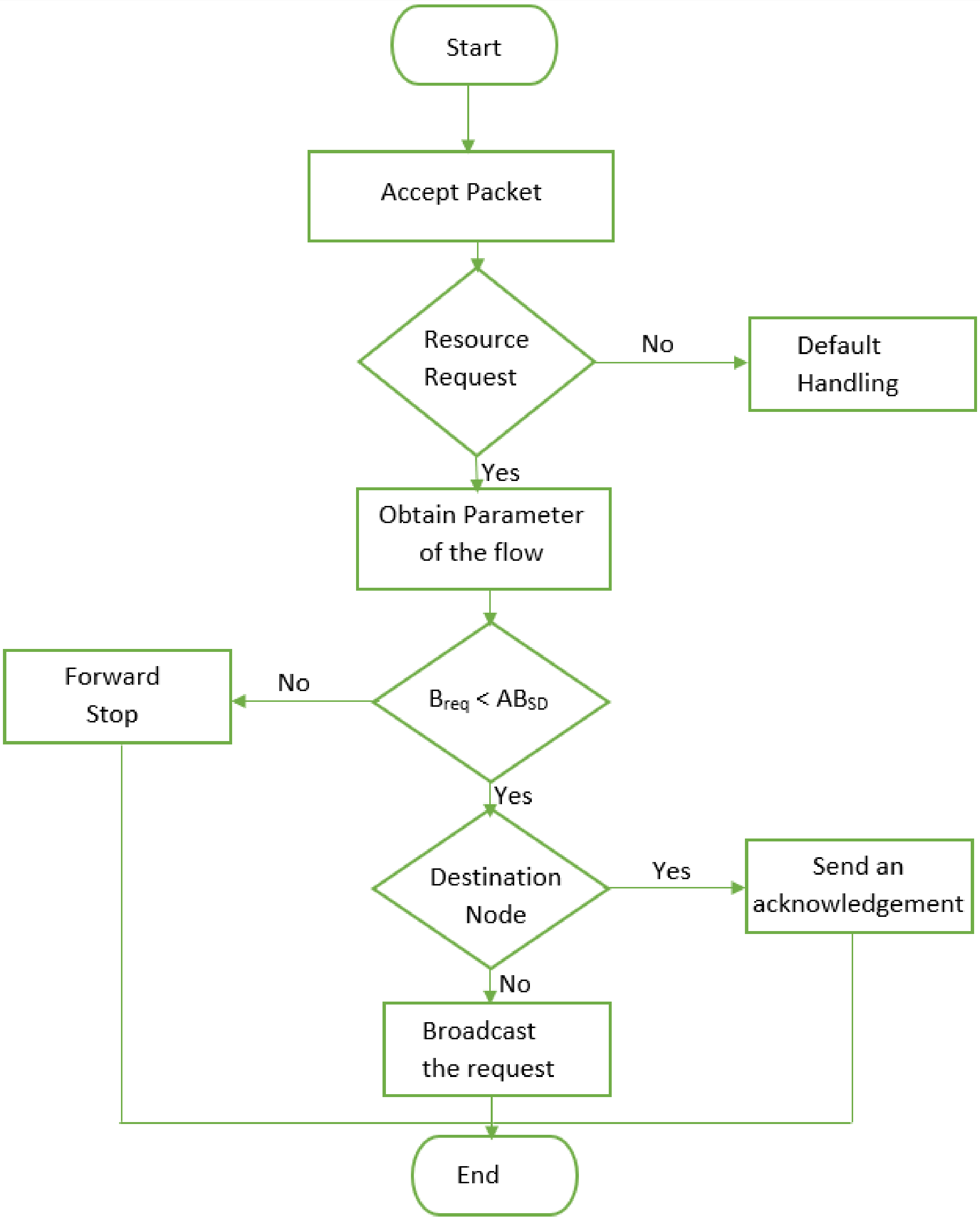


Figure 5.7b: Admission Control Protocol at the Intermediate node for Bandwidth Estimation in RAACM

5.7. HELLO packet propagation to retrieve the available bandwidth on the carrier sensing region.

The HELLO bandwidth retrieval technique allows every node within the network to input their current bandwidth usage into the *HELLO platform*, therefore, this enables any hosts that need to estimate the available bandwidth to do so by checking the bandwidth consumption indicated in the *HELLO platform* from its second hop neighbours. In a wireless medium, when a host wants to gain access to the medium, it will first check if the medium is idle before gaining access to the medium. The medium availability detection has been previously discussed in section 5.2. Each host can therefore deduce its bandwidth information based on the information obtained from its two-hop neighbour. However, it is important to know how well this information from the two hops neighbour can be retrieved without inducing too much overhead on the network. The first hop neighbour information is retrieved directly by sending the HELLO message to the first hop. Table 5.3 shows the structure of the *HELLO platform* providing information about the neighbour bandwidth consumption and the time unit.

Host Address	**Bandwidth Consumption**	**Time Unit**
Neighbour Address (N_1)	Bandwidth Consumed by N_1	Time Unit
-		
-		
-		
-		
Neighbour Address n (N_n)	Bandwidth Consumed by N_n	Time Unit

Table 5.3 HELLO Platform Structure

In Table 5.3 above, the bolded items in the table on the first row represents the host's information while the other rows represents the neighbours information. The host Address node specifies the address of the host and the address of its neighbours. The bandwidth consuption column provides information about the bandwidth consumed by the host and its neigbour and it is periodically updated. The time unit of both host and its neighbour specifies the time all the information was registered unto the *HELLO platform.* The second hop neighbour bandwidth information cannot be obtained directly, therefore, this thesis propose to use HELLO packet propagation to retrieve the second neighbour host information. This is achieved by sending the HELLO message to the first hop range, thereafter, the HELLO packet propagates to the rest of the hops on the network to enquire the available bandwidth and the neighbour's information.

From the discussion above, it is observed that instead of broadcasting the HELLO packet to all the various hops to enquire the available bandwidth of a two hop neighbour information, RAACM only send one HELLO message to the first-hop range. This later extends to the rest of the hops on the network in order to learn about the available bandwidth information within the carrier sensing range of a node. This technique helps to significantly reduce the overhead within a network, since the HELLO packet is overextended rather than flooding the information over the retrieval range. However, the information is gradually obtained during the network deployment process. Once the neighbour node registers its bandwidth consumption, the HELLO platform indicates if it is an updated one based on the timestamp indicated.

5.7.1. Step by Step Analysis of the Available Bandwidth Retrieval

Step 1: Source A input its bandwidth information (IP/Host address, bandwidth consumption and its current time) into the HELLO platform.

Step 2: Source A sends HELLO message to the 1st hop neighbour to retrieve the bandwidth information (IP/Host address, bandwidth consumption and its current time).

Step 3: Source A gets the available bandwidth information of the 1st hop neighbour and inserts it into the HELLO platform.

Step 4: Source A extends the HELLO message in the 1st hop to the 2nd hop neighbour to retrieve the available bandwidth information of the second hop.

Step 5: Source A gets the available bandwidth information of the 2nd hop neighbour and inserts it into the HELLO platform.

Step 6: Source A extends the HELLO message in the 1st hop to the n^{th} hop neighbour to retrieve the available bandwidth information of the n^{th} hop.

Step 7: Source A gets the available bandwidth information of the n^{th} hop neighbour and inserts it into the HELLO platform.

Step 8: HELLO Platform can therefore retrieve the total available bandwidth consumed by each node.

5.8.Discussion

The need for effective estimation of bandwidth for admission control in MANET is important for proper functionality and management of the network. The estimation task however encounters its own challenges due to the nature of MANET. In this chapter, we propose a resource allocation and admission control in MANET (RAACM) mechanism that estimate the available bandwidth for admission control.

The main contribution of this thesis is based on the modification of the channel idle time that is used for the bandwidth estimation process. RAACM which is our proposed admission control protocol, considers a channel idle time dependency between a sender and a receiver by differentiating a nodes BUSY state from when it is in a SENSE BUSY state and addresses the IDLE state that may be caused by an empty queue to guarantee an accurate and efficient bandwidth estimation. RAACM also estimate the channel idle time dependency by allowing the sender and the receiver node to witness both the common interference and the independent interference to prevent under-estimation and over-estimation of the available bandwidth estimation.

In RAACM, to retrieve the available bandwidth on a carrier sense region the HELLO message advertisement is sent to the first hop (1 hop packet propagation). The first hop propagates the HELLO packet to other neighbours to retrieve bandwidth information. This process prevents unnecessary use of available bandwidth which may result in network overhead.

5.9. Summary

One of the notable challenges faced by the current bandwidth estimation techniques for admission control in MANET is the implementation of an accurate measurement to enhance good QoS. Various estimation techniques have been proposed but none of the existing solutions have properly addressed the channel idle time dependency between the sender and the receiver by differentiating the *BUSY* state from the *SENSE BUSY* states and the *IDLE* state caused by an empty queue during the bandwidth estimation process. Also, none of the previous work proposed a technique that sends HELLO packet to its one-hop neighbours which further propagates to the rest of the nodes to retrieve the available bandwidth on a carrier sensing region as this technique helps to limits the impact of the additional overhead on the carrier sensing multiple access with collision avoidance (CSMA/CA).

Therefore, in this thesis, we propose a resource allocation and admission control in MANET (RAACM) mechanism that estimates the bandwidth for admission control. RAACM adopted the following;

i. Bandwidth estimation process that considers the channel idle time synchronization and dependency between the sender and the receiver node by differentiating the *BUSY* state from the *SENSE BUSY* states and the *IDLE* state caused by an empty queue.
ii. HELLO packet propagation to retrieve the available bandwidth on the carrier sensing region.
iii. A novel, efficient and accurate resource allocation and admission control in MANET (RAACM) that estimates the available bandwidth for the admission controller to either accept or reject a session, when an admission is requested.

Chapter 6: Result Analysis and Validation of RAACM

6.1. Introduction

This chapter presents the result analysis, validation, and evaluation of our proposed RAACM. For comparison, we have integrated the proposed bandwidth estimation for admission control RAACM into AODV routing protocol and implemented using OPNET simulation tool. This simulation tool has been selected because it can best simulate our proposed technique and also its availability. Therefore, RAACM protocol is based on route request message issued by a transmitter, a distributed admission control and a route reply message that is unicasted by the destination node. Thus, we study the impact of our bandwidth estimation technique for admission control and compared our proposed RAACM protocol with those that closely exist in the literature.

6.2. RAACM Features

In RAACM-AODV, neighbour nodes exchange their locally computed available bandwidth using the *HELLO* messages. Every sample period (T) seconds, each node will locally estimate its medium occupancy ratio and includes this information in the *HELLO* packet.

The *HELLO* technique generates an additional overhead depending on how often it is sent. Normally, the frequency of the *HELLO* packet emission should be adapted to the dynamic flow. For a meaningful comparison, *T* is set to 1 second in RAACM-AODV.

6.3. Simulation Parameters

In this section, OPNET modeler 17.5 was used to simulate the design in order to evaluate the performance of RAACM. 100 nodes have been deployed in a 1200x1200m area. Furthermore, other network parameters have been set accordingly (i.e. Data rate of 11Mbps). *T* is set to 1 second while 8 sender and receiver nodes, which are dependent on one another, are selected among the 100 nodes to transmit traffic. The other nodes are either acting as relay nodes or idle. Simulation was carried out for 60 seconds and each simulation was repeated 10 times. Table 6.1 and 6.2 presents the parameters used for the simulation.

Parameter	**Value**
Number of nodes	100
Total network area	1200 X1200m
Wireless Standard	IEEE802.11e
Physical Characteristics	Direct Sequencing
Data rate	11Mbps
Packet size	1000bytes
Number of sender-receiver	8
T	1sec
Number of simulation (repetition)	10 times
Simulation time	60s
SIFS	10µs (microsecond)
Slot time	20µs
CWmin	31
CWmax	1023
Traffic type of service	Best effort, Voice and Video
Transport Protocol	UDP
Traffic Bit Rate	CBR

Table 6.1: Simulation Parameter of the Physical Characteristics

EDCA Parameter	CWmin	CWmax	AIFSN
AC_VO	(CWmin +1)/4-1	(CWmin +1)/ 2-1	2
AC_VI	(CWmin +1)/ 2-1	CWmin	2
AC_BE	CWmin	CWmax	3

Table 6.2: EDCA QoS Parameter

6.4. Simulation Model, Validation and Evaluation of RAACM

The performance of RAACM estimation technique was simulated and compared with three other protocols, namely, BECIT, MBA-AODV and AABWM. The simulation was carried out using OPNET 17.5. The simulation parameters used, which depicts the average results obtained over 10 simulation trials are shown in table 6.1 and 6.2.

Different estimation parameters such as WLAN throughput, HCF access category throughput, delay, and data dropped encountered were used to evaluate the performance of RAACM.

Note that BECIT, MBA-AODV and AABWM have been modified to function in IEEE802.11e QoS setting, since they were previously configured for IEEE802.11. AODV routing protocol has also been used for simulation for the purpose of comparison.

6.4.1. Performance Evaluation

In order to demonstrate the performance of RAACM, we transmitted three different traffic flows simultaneously and viewed their behaviour. For each flow, the source and destination node are selected, and dependent on each other. Each of the flow is made up of 1000bytes frames with a data rate of 11Mbps. The duration for the simulation is 1 hour. Figure 6.1(a) and (b) depicts the throughput of three different network traffic types with and without the implementation of admission control and network prioritization. Note that without an admission control as shown in Figure 6.1 (a), there is a network congestion, as all the flows will attempt to transmit at the same time. All the flows will be granted access irrespective of the available network resources. This will however lead to a lot of packet drop. To prevent loss of packet, it is important to introduce an admission control protocol as well as traffic prioritization to ensure that network traffic are well structured during transmission. This work has proposed a RAACM protocol for accurate and efficient transmission across the network. The result of deploying RAACM is shown in Figure 6.1 (b). When an admission control (RAACM) is introduced, there is a difference in the throughput result, as the traffic is more structured and any flow that will result in the network degradation is not allowed in the network. In Figure 6.1 (b), the best effort traffic is not granted access while the voice and video traffic were allowed to transmit. From the graph we observe that the transmission flow starts as normal, thereafter, on getting to the admission controller, RAACM denied the best effort flow access, therefore terminating the transmission and the throughput diminishes till there is no throughput at all. The voice and video flow are, however, allowed transmission access to the network because they meet the networks specification and has a higher priority than the best effort traffic.

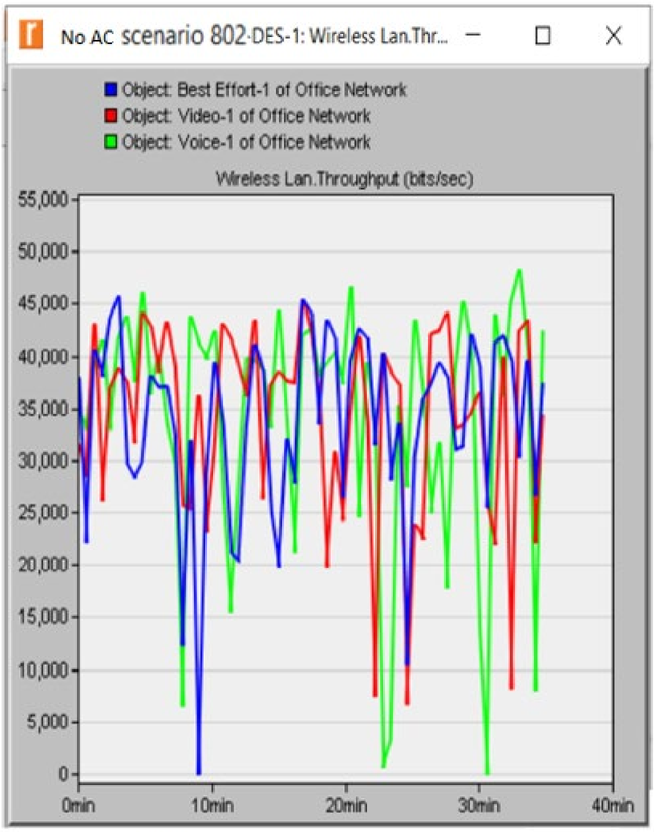


Figure 6.1 (a): WLAN without QoS Admission Control and Prioritization

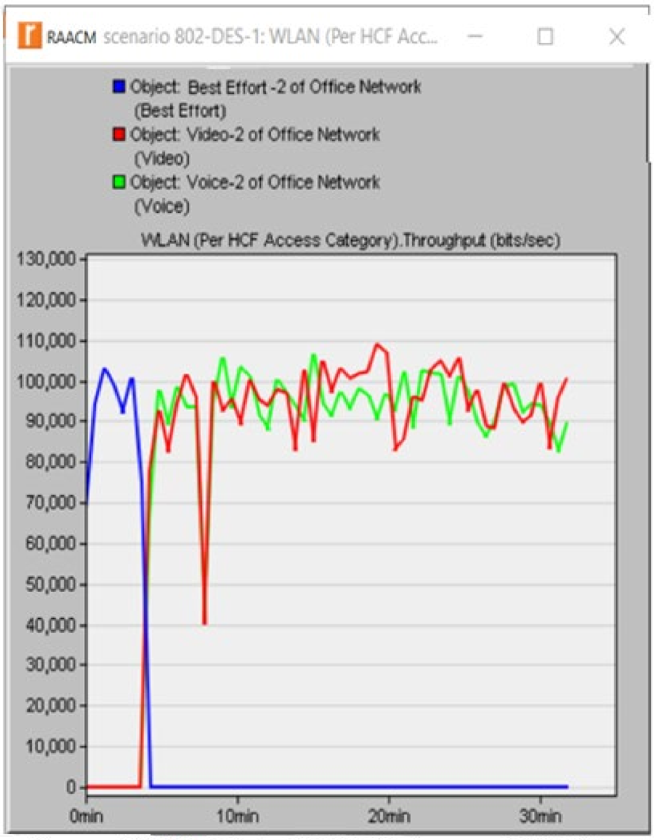


Figure 6.1 (b): RAACM HCF Access WLAN

6.4.2. QoS HCF Admission Control Evaluation

6.4.2.1. HCF Throughput Analysis and Comparison

In this section, we first present the HCF WLAN throughput admission control comparison for voice and video traffic. Thereafter, we present the WLAN HCF admission control protocol throughput comparison showing how the best effort was denied access by all the admission control protocols.

6.4.2.1.1. HCF WLAN Throughput Admission Control for Voice and Video Traffic

Figure 6.2 shows the comparison of WLAN HCF voice throughput analysis of our proposed RAACM with other state-of-the-art admission control protocol. Result from the graph shows that the average voice throughput of RAACM tends to be more than BECIT, MBA-AODV, and AABWM. Table 6.3 shows the raw values of the average throughput analysis of voice traffic and how RAACM voice throughput traffic result surpasses the rest of the admission control protocols used for comparison. This analysis makes our proposed RAACM more efficient and accurate, as the higher average value realised by RAACM voice traffic shows that it is more reliable and has a faster and higher voice throughput, when compared to the rest of the protocol considered.

Figure 6.3 shows the comparison of the WLAN HCF video throughput analysis of our proposed admission control protocol RAACM with BECIT, MBA-AODV, and AABWM. Results from the graph shows that AABWM and BECIT has the highest throughput results in term of video traffic. The result from AABWM and BECIT video throughput may appear to be more than RAACM, but the realistic aspect of it is that, AABWM and BECIT is operating as a greedy protocol during the video traffic transmission as it allows a lot of video traffic to gain access to the network channel and does not bother about giving equal access to the voice traffic which is of more importance. The average video throughput of MBA-AODV is less as compared with RAACM. Table 6.3 gives the average data analysis of the video throughput realised.

In summary, based on the analysis in Figure 6.2, 6.3 and Table 6.3, the average throughput of RAACM for voice traffic tends to be more than BECIT, MBA-AODV and AABWM, while the average throughput of video traffic for AABWM tends to be more compared to RAACM, BECIT, and MBA-AODV. This analysis makes our proposed RAACM more functional, as the higher average value realised by RAACM voice traffic shows that it is more reliable and has a faster and higher voice throughput, when compared to the rest of the protocol considered. We have also ensured that the voice is prioritized for transmission by disallowing the video traffic to get hold of a very high portion of the network.

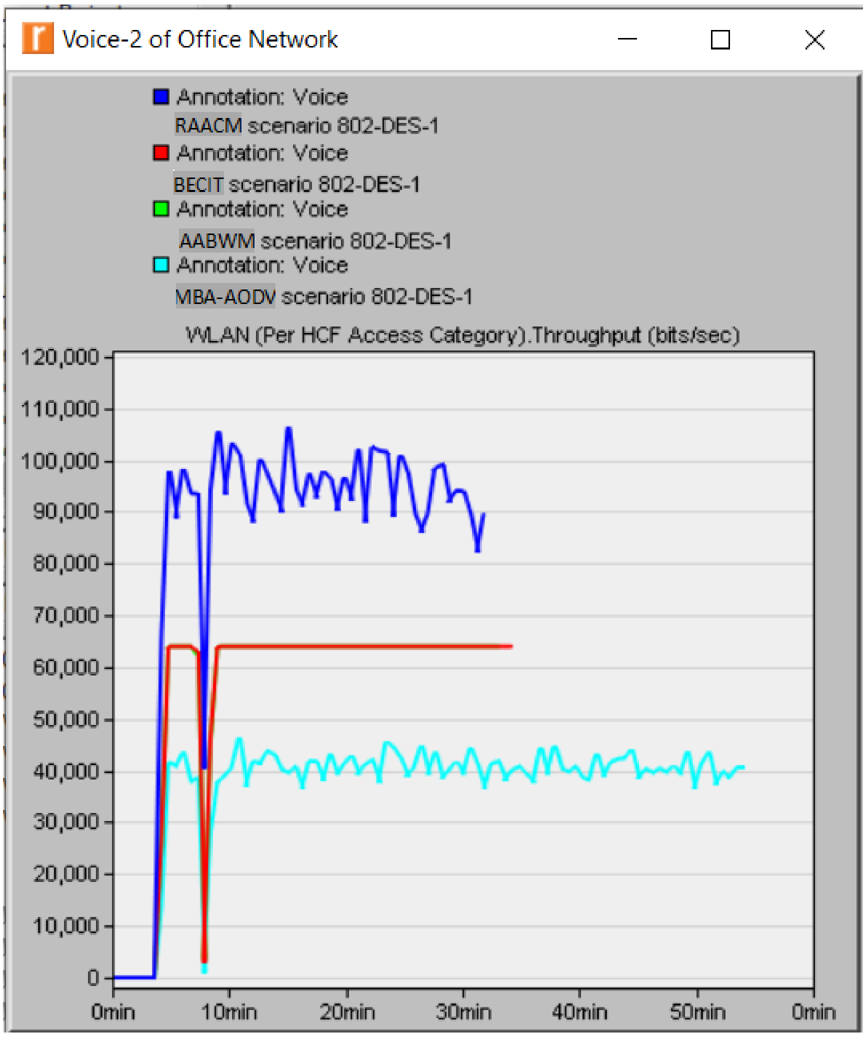


Figure 6.2: WLAN HCF Throughput Access for Voice Admission Control Comparison

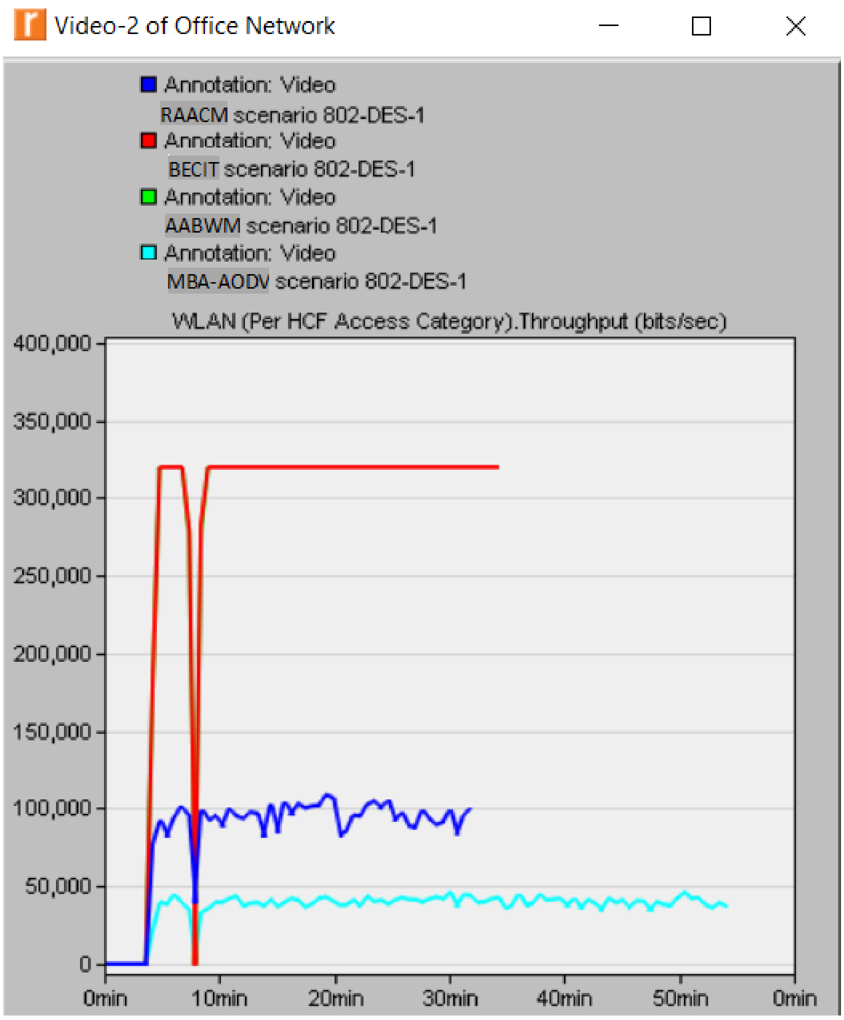


Figure 6.3: WLAN HCF Throughput Access for Video Admission Control Comparison

Protocol	Average Best Effort Throughput (bit/sec)	Average Video Throughput (bit/sec)	Average Voice Throughput (bit/sec)
RAACM	12,195	82,186	81,221
BECIT	7,421	272, 161	54,261
MBA-AODV	3,091	36,464	36.812
AABWM	7,496	270,317	53.925

Table 6.3: WLAN HCF Access Category Average Throughput Report Protocol Comparison

6.4.2.1.2. WLAN HCF Throughput Comparison Showing Best Effort Traffic

Figure 6.4a shows the comparison of best effort throughput with voice throughput of all the admission control protocol. While, figure 6.4b shows the comparison of the best effort throughput with video throughput of all the admission control protocol. From the graphs in 6.4 a and b, it is seen that the best effort was not granted access to the network and its flow was dropped when it got to the admission controller, this is because the voice and the video traffic were prioritised against the best effort traffic, that was why the throughput level of best effort diminished to zero all through the transmission process. Therefore, all the admission control protocols (RAACM, BECIT, MBA-AODV and AABWM) prevent the best-effort flow from transmitting in order to limit network congestion and to provide for network prioritization.

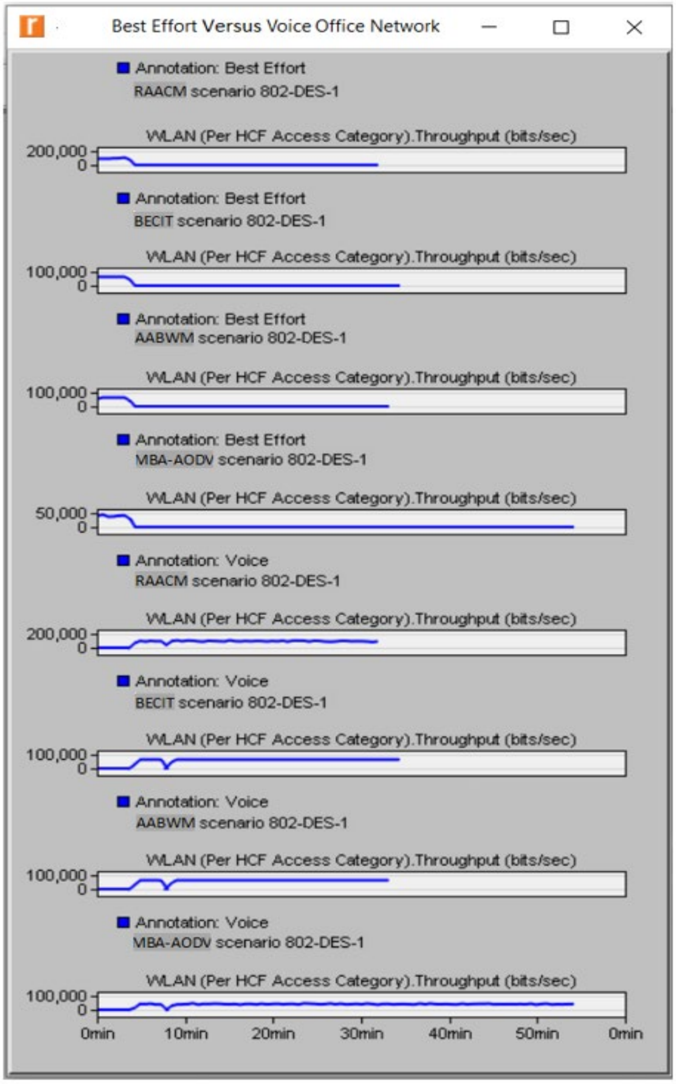


Figure 6.4a: Best Effort Versus Voice

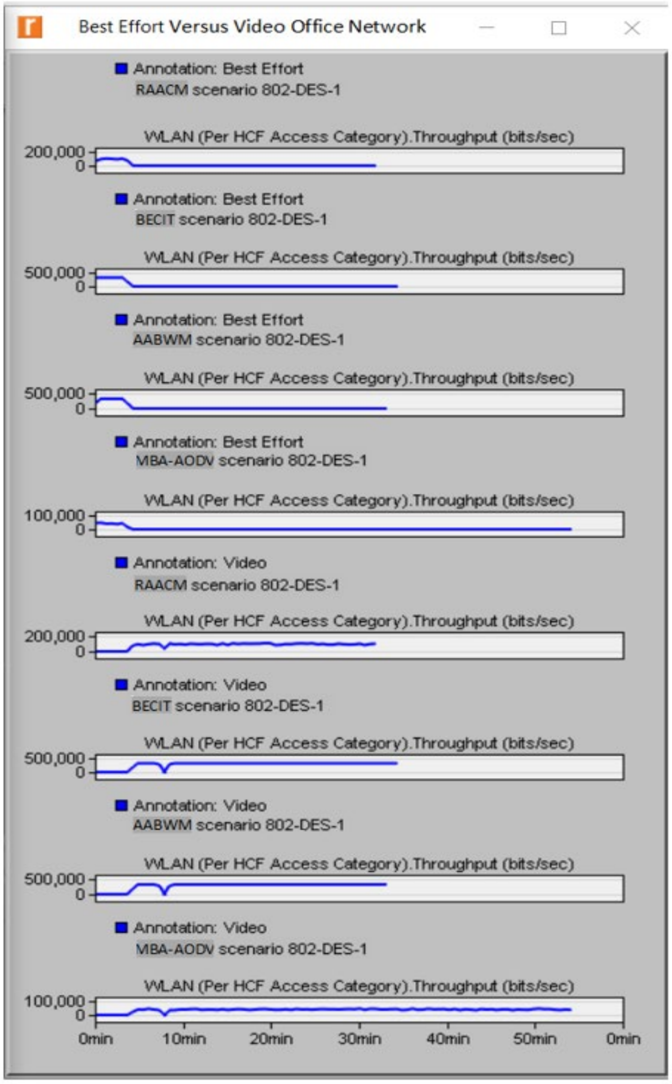


Figure 6.4b: Best Effort Versus Video

Part of the throughput success in voice and video traffic of RAACM is attributed to its assumption of the channel idle time, where the sender and receiver are dependent on each other for proper transmission and network nodes are not randomly distributed. While in BECIT, MBA-AODV, and AABWM, the channel idle period of each surrounding node around the sender and the receiver view each node of the channel as being a uniformly random distribution. They are independent of one another, therefore there is no proper estimation that is been carried out.

It is important to note that this throughput analysis is for a contention-based access for HCF, resulting into best effort traffic been denied access, based on its access category. The video and voice traffic were prioritized against best effort traffic. Also, by checking through the RAACM average throughput result in table 6.3, it is seen that the value of voice and video traffic throughput are quite close. This is due to the act that both traffic were given the same access priority. The video traffic throughput, however, tends to be slightly higher than the voice traffic throughput because the video traffic access request arrives earlier than the voice traffic access request.

To create a fairer network access, the IEEE802.11e protocol also provides for a contention free access through the TXOP. Therefore, during the TXOP period, the best effort traffic can access the medium based on the duration given to it. Once it gets to its limit, the contention free period for the best effort traffic will be terminated.

6.4.2.2 WLAN HCF Delay

Figures 6.5 to 6.7 show the delay comparison of the admission control protocols for each traffic (best effort, voice and video). Table 6.4 presents the data collected from the graphs in Figure 6.5 to 6.7.

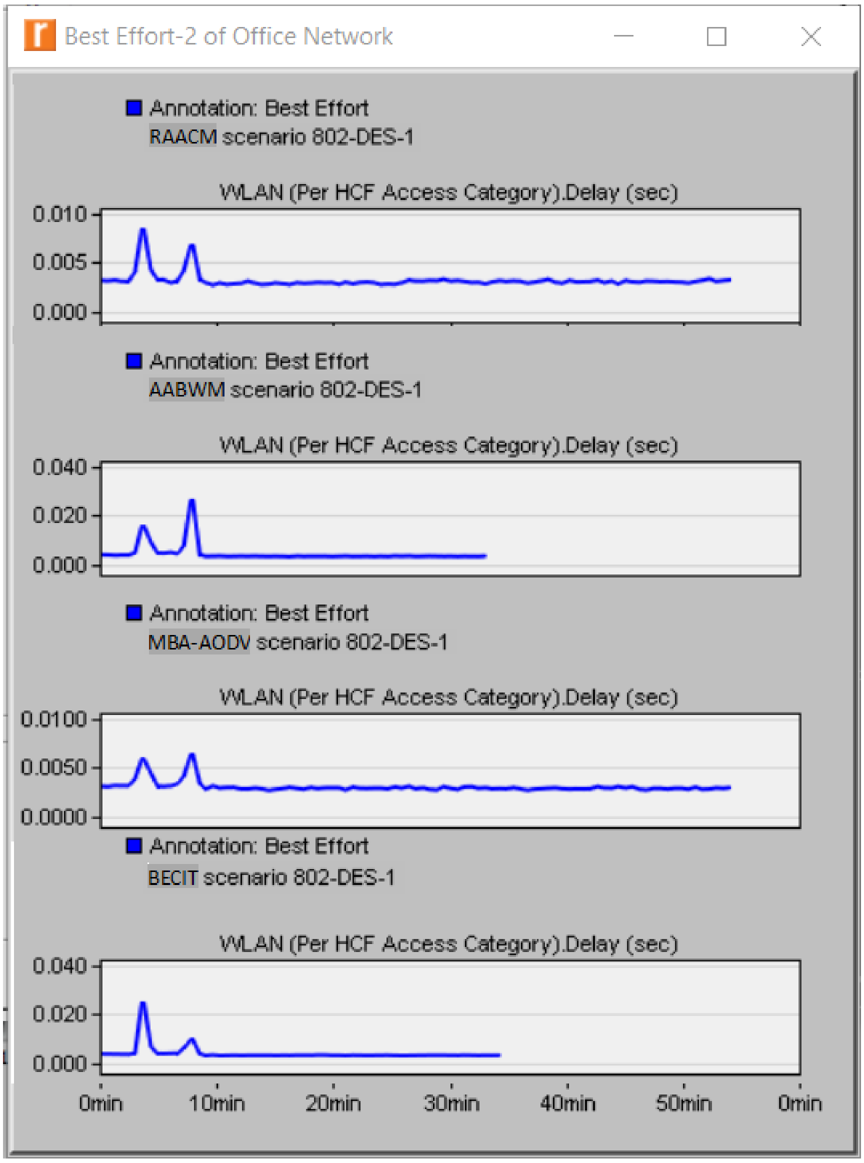


Figure 6.5: WLAN HCF Admission Control Delay Protocol Comparison of Best Effort Traffic

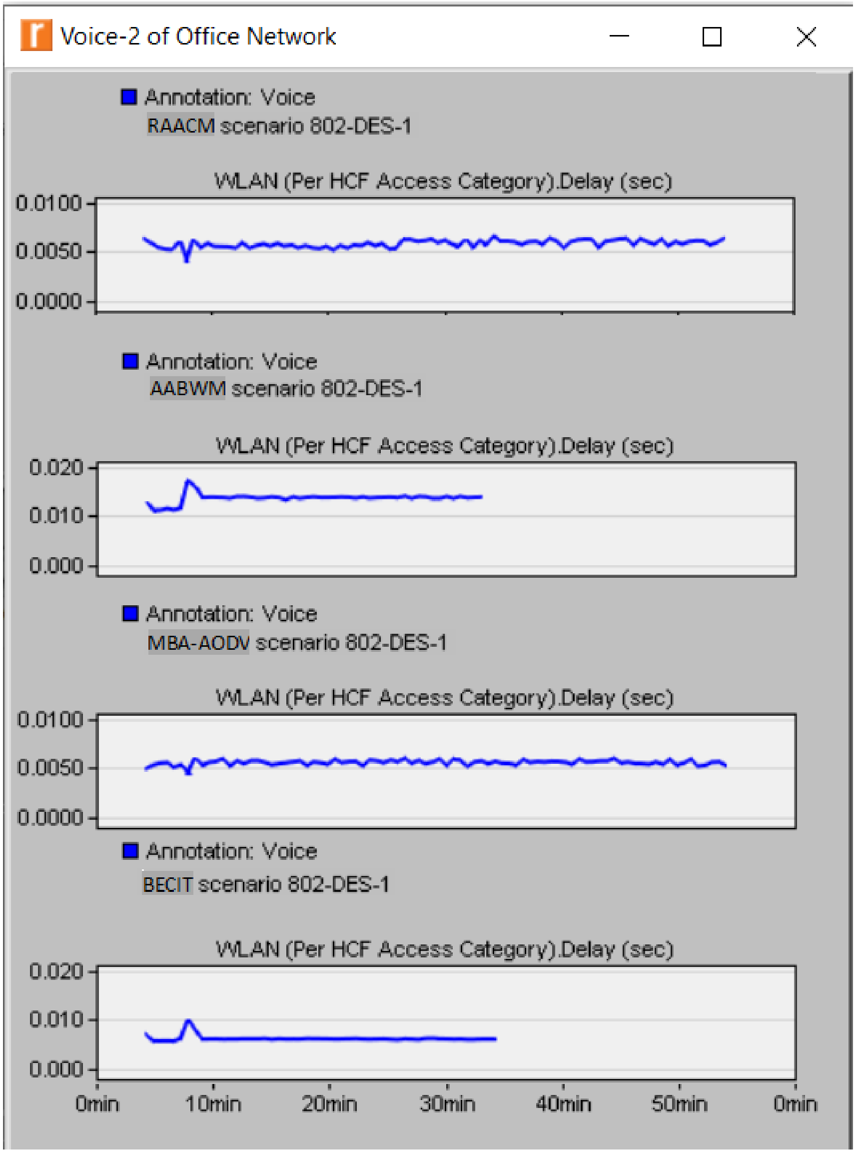


Figure 6.6: WLAN HCF Admission Control Delay Protocol Comparison of Voice Traffic

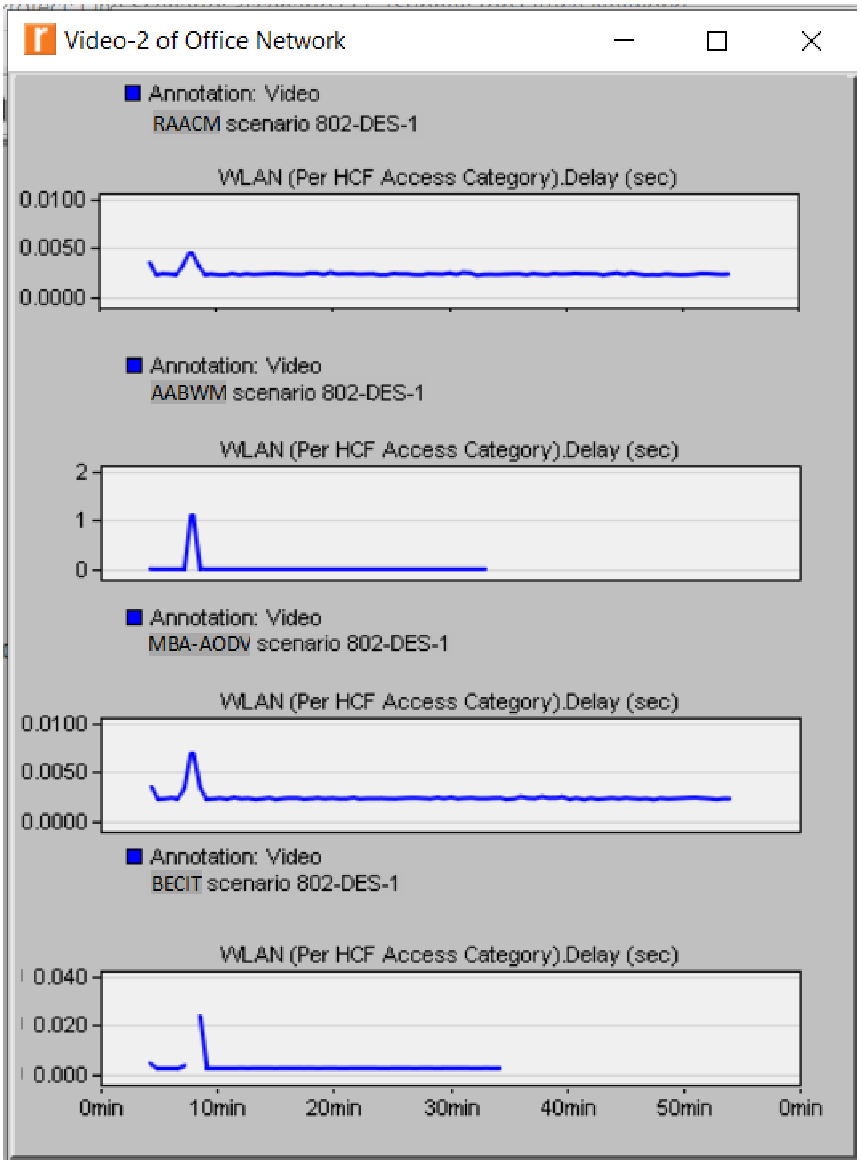


Figure 6.7: WLAN HCF Admission Control Delay Protocol Comparison of Video Traffic

Protocol	Average Best Effort Delay (Sec)	Average Video Delay (Sec)	Average Voice Delay (Sec)
RAACM	0.0032	0.0024	0.0058
BECIT	0.0041	0.0030	0.0062
MBA-AODV	0.0031	0.0024	0.0056
AABWM	0.0045	0.0278	0.0137

Table 6.4: WLAN HCF Access Category Delay Report Protocol Comparison

From Table 6.4, it is seen that RAACM has the least amount of video delay (same value as MBA-AODV) compared with other protocols except MBA-AODV. Also, the best effort and voice delay generated by RAACM is less when compared with BECIT and AABWM. Though the value of the voice and best effort delay generated by MBA-AODV is less as compared to RAACM, there difference is minute, therefore it can be said that the RAACM and MBA-AODV operate at almost the same level of delay.

6.4.2.3 WLAN HCF Data Dropped

Figures 6.8 to 6.10 show the data dropped in the designed admission control. Packets are dropped due to buffer overflow and retry threshold exceeded. This thesis however, consider the result of the data dropped due to buffer overflow. Buffer overflow is the amount of higher layer packets that are dropped because no acknowledgement is received for packets that are sent. Figures 6.8 to 6.10 denote the data dropped buffer overflow for best effort traffic, voice traffic, and video traffic. In figure 6.8, the rate of data dropped for the best effort traffic in RAACM as compared with the rest of the protocols (i.e., BECIT and MBA-AODV) is very close. It is seen that AABWM maintains a higher and steady data dropped compared with all other protocols. However, RAACM best effort traffic does not cause a lot of data drop as compared with the rest of the admission control protocol.

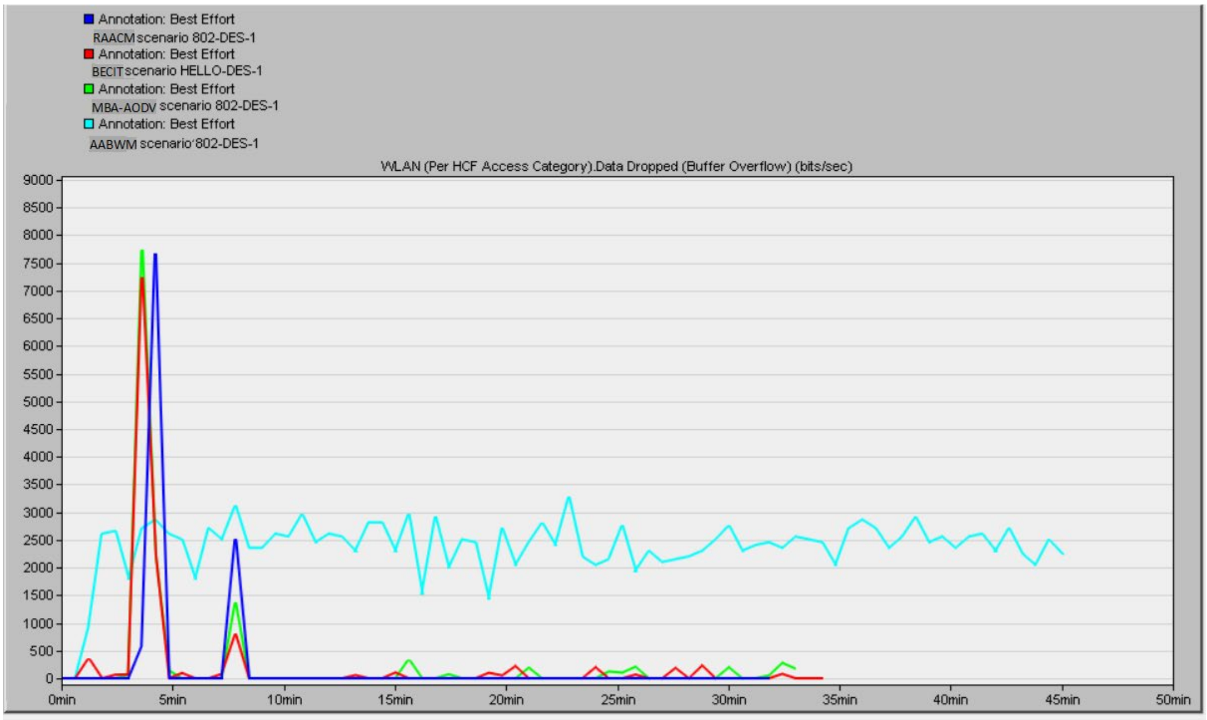


Figure 6.8: WLAN HCF Admission Control Protocol Data Dropped Comparison for Best Effort

Figure 6.9 shows the WLAN HCF admission control protocol data dropped comparison for voice traffic. All the admission control protocol for voice traffic also shows similar results in the packet dropped, even though the RAACM video encountered a trigger at some point and was able to adjust to the traffic and then stopped dropping packet.

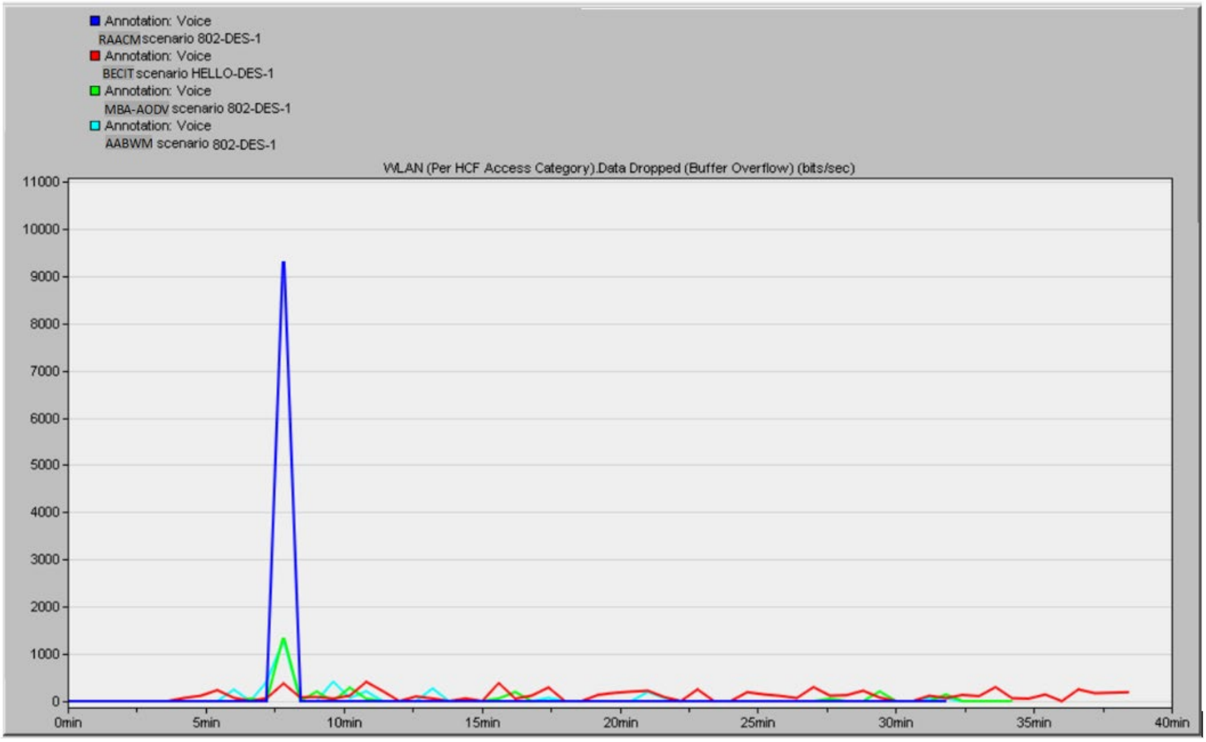


Figure 6.9: WLAN HCF Admission Control Protocol Data Dropped Comparison for Voice

Figure 6.10 also shows the WLAN HCF admission control protocol data dropped comparison for video traffic. All the admission control protocol for video traffic also shows similar results in the packet dropped except for BECIT that maintains a higher drop in video packet.

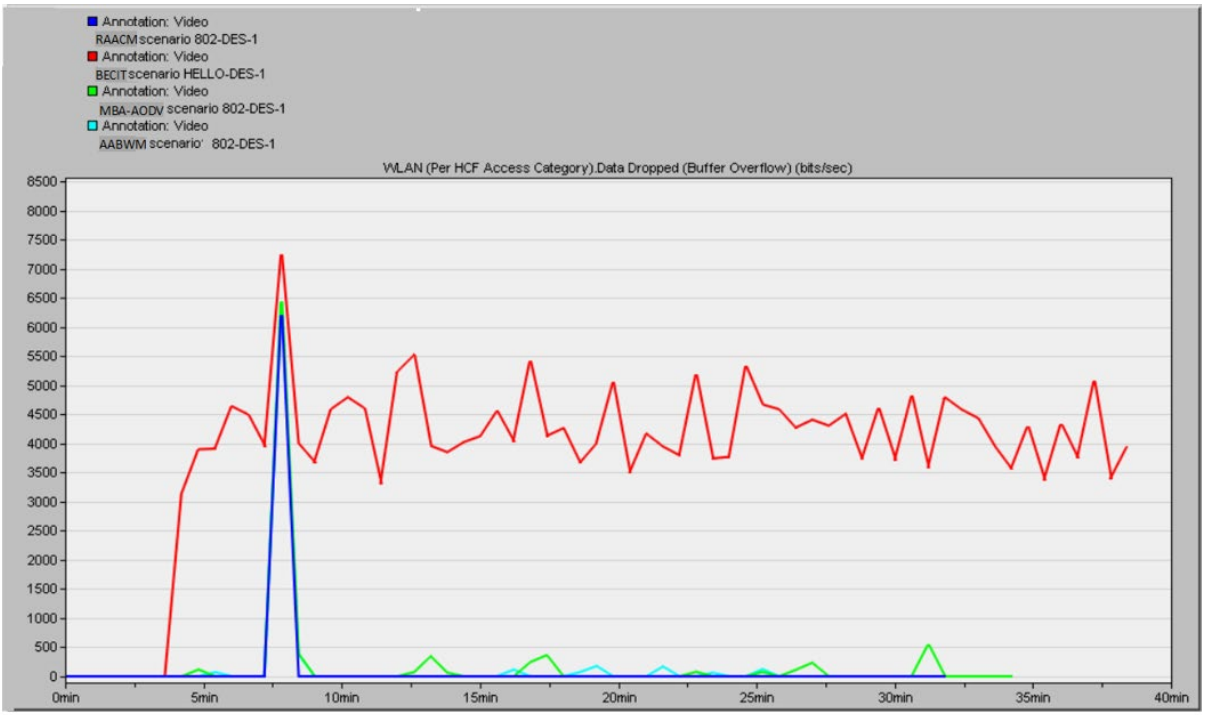


Figure 6.10: WLAN HCF Admission Control Protocol Data Dropped Comparison for Video

However, in summary, the results for data dropped in best effort, voice and video varies as some of the related works used for comparison dropped more data in either video, voice and best effort, but all through the data drop comparison, RAACM has shown stability in the packet dropped and the amount of packet dropped has been of minimal value as compared with the rest of the other protocol used for comparison.

6.5. Discussion

For an appropriate estimation of bandwidth for an admission control in MANET to be achieved, it is important to consider the key factors necessary to be developed to enhance good QoS. This thesis has therefore proposed an RAACM admission control protocol that incorporates nodes dependency into its design were nodes are not randomly distributed within the network and allows for both complete and independent node interference to be observed by the sending and the receiving node.

To evaluate the performance of our proposed RAACM, we first compared RAACM with a WLAN that has no admission control and QoS prioritization deployed. Results obtained shows that without the deployment of an admission control, no procedures will be followed by any transmitting nodes and this will lead to a huge amount of drop in packet. RAACM WLAN throughput access for voice and video admission control was thereafter compared with BECIT, MBA-AODV and AABWM. Result obtained shows an efficient throughput, where the average throughput of RAACM for voice traffic tends to be more than BECIT, MBA-AODV and AABWM, used for comparison. The video traffic also shows an effective and reliable result and RAACM ensures that the voice is not giving too much of the priority during transmission and disallowing the video traffic to get hold of a very high portion of the network. However, RAACM provides equal priority for voice and video traffic which enhances accuracy as against other protocol used for comparison. Furthermore, this thesis evaluated the proposed RAACM and the closely related research work with WLAN HCF delay. Result obtained shows that RAACM has the least amount of video delay, while the best effort and voice delay generated by RAACM is less when compared with BECIT and AABWM, with almost similar delay as MBA-AODV. Finally, this thesis evaluated RAACM and the closely related research work with WLAN HCF data dropped. Result obtained shows that RAACM has stability and efficiency in the packet dropped and the amount of packet dropped has been of minimal value as compared with the rest of the other protocol used for comparison.

6.6. Summary

This chapter has been able to provide results obtained when our proposed RAACM is deployed for measurement of bandwidth estimation and admission control. We compared the throughput results obtained when RAACM is deployed, with no admission control implementation, and result from the graph show accuracy and effectiveness of our proposed RAACM. RAACM was also compared with the state-of-the-art available bandwidth and admission control techniques based on other evaluation criteria and the results recorded shows that RAACM outperforms the state-of-the-art bandwidth estimation for admission control.

Chapter 7: Conclusion and Future Work

As the final concluding chapter of this thesis, this section presents the overall contributions as against the set objectives by presenting a chapter summary. Thereafter, recommendations and suggestions for future work are discussed.

7.1. Conclusion

This thesis entitled "Performance Analysis of a Developed Admission Control Model in Mobile Ad-Hoc Network" expound the overall goal of this study. The aim of the resource allocation and admission control is to ensure an effective and efficient sharing of resources to ensure proper network utilization and functionality. Admission control decides whether to admit data sessions that satisfies a given QoS requirement (i.e. bandwidth), without violating any previously made rules or reject sessions.

The work presented in this thesis therefore focused on QoS provision in a wireless MANET. In this context, we proposed a resource allocation and admission control for MANET, by presenting a bandwidth estimation measurement technique. This is used for admission control purpose to perform some pre-configured checks prior to establishing a connection to know if the current bandwidth resources are sufficient for a proposed connection to guarantee QoS. We therefore highlighted the following contributions;

(i) Bandwidth estimation process, where channel idle time dependency is incorporated and nodes within the interference range of the sender and the receiving node are not randomly distributed. The bandwidth estimation also provides for synchronization by allowing the sender and the receiver node to witness both common interference (complete overlap) and independent interference (no overlap) during the bandwidth estimation process. collision and back-off have also been highlighted in this thesis as other factors that have an impact on the available bandwidth estimation asides the channel idle time period.

(ii) HELLO packet propagation to retrieve the available bandwidth on the carrier sensing region.

(iii) A novel, efficient and accurate resource allocation, and admission control (RAACM) that estimates the available bandwidth for the admission controller to either accept or reject a session, when an admission is requested.

For the admission control to be realised for QoS provision, the estimation of the available resources, such as the available bandwidth, was carried out, where we modified the channel idle time to enhance efficiency and accuracy. Also, to retrieve the available bandwidth on a carrier sense region, we estimate the serviceable bandwidth. The serviceable bandwidth is therefore defined as the smallest available bandwidth observed on a sensing region. The main idea behind the bandwidth retrieval process is to make use of HELLO message, which is forwarded between nodes for connectivity awareness. The HELLO message only advertises to the first-hop range before it propagates to the rest of the hops on a network. HELLO advertisement to only the first hop range, which has been adopted by our proposed protocol RAACM, has not been previously used in literature. The serviceable bandwidth calculation remains accurate because the carrier sensing nodes information allows packet propagation. This further helps to significantly reduce the overhead within a network, since the HELLO packet is over extended rather than flooding the information over the retrieval range. The information is gradually obtained during the network deployment process. RAACM has been proposed to estimate the available bandwidth for the admission controller to either accept or reject a session when an admission is requested.

In summary, this thesis began by presenting the introductory chapter that introduced the main aim and focus of this thesis. This thesis discussed the original research contributions, research questions and objectives. Thereafter, the thesis presented the general overview of MANET, its services, application area, and challenges posed by the admission control protocol. We discussed the challenges the MANET environment poses to admission control protocol. Additionally, the resource allocation for admission in MANET was discussed as well as the admission control and protocol design consideration for admission control QoS in MANET. The literature review chapter presented a detailed review of the academic literature on the available bandwidth estimation methods and admission control in MANET published between 2008 and 2020. Bandwidth estimation was sub-divided into active and passive estimation techniques. The passive bandwidth technique was thereafter proposed to be the most suitable available bandwidth estimation method for the proposed admission control model. Furthermore, research gaps in the current proposed bandwidth estimation for admission control was identified. Furthermore, investigation was carried out on the admission control scheme that can be deployed for bandwidth estimation. Distributed admission control was proposed to be accurate for the scenario in this thesis. Also, this thesis discussed the routing protocol that can be deployed with admission control in MANET as well as the wireless protocol standard to enhance QOS. The AODV routing protocol was proposed as well as the IEEE802.11e. Further discussion on the different simulation tools that can be deployed for the simulation of the designed resource allocation and admission control was also presented in this thesis. A table showing the comparison of all the simulation tools was outlined based on the languages it supports, their advantages as well as disadvantages. Optimized network engineering tool modeler (OPNET) now referred to as Riverbed modeler (Riverbed, 2018) was deployed for modelling and simulation due to its numerous network features, and it can best simulate our proposed technique. It was also deployed because of the availability of the version at Anglia Ruskin University (ARU). This thesis thereafter presents our proposed scheme, RAACM, and its deployment. A comprehensive detail of the steps that RAACM complied with for accurate and effective available bandwidth estimation was discussed.

Finally, this thesis presented the results obtained when our proposed RAACM is deployed for bandwidth estimation and admission control. A comparison was carried out between the throughput results achieved when RAACM is deployed and the throughput result when there is no admission control implementation. The result obtained show the accuracy and effectiveness of our proposed RAACM. RAACM was also compared with the state-of-the-art available bandwidth and admission control based on other evaluation criteria. The results obtained show that RAACM outperforms the state-of-the-art bandwidth estimation for admission control.

7.2. Future Work

Since this thesis proposed a QoS technique for the estimation of bandwidth in MANET, the results achieved was based on the set of protocols used in our design. However, in the future, it will be of interest to modify the transport protocol used for the communication, routing protocol, bit rates for voice and multimedia application implemented as this modification will have an impact on the result gotten. Also, the available bandwidth of a link between two nodes, where the estimation of the available bandwidth will be based on the bandwidth consumed by another variable bandwidth link/ flow will be considered and measurement will be taken. However, detailed explanation of our intention for future work is detailed below;

- In future, different routing protocols will be integrated with RAACM as against AODV used in this thesis, and metrics like end-to-end delay or metrics that trades-off end-to-end available bandwidth will be used to observe their performance.

- RAACM’s flow admission control algorithms control messages and lost or corrupt control messages can impact the performance of RAACM. Therefore, in the future, we will be interested in evaluating the performance of RAACM using different bit error rate values.
- We observed that having multiple transmitters in the network will have an impact on the additional overhead. Therefore, in the future, this can be incorporated to analyse the possible impact. This is not trivial as the estimate of the number of transmitters/flows within the interference range of the node estimating the MAC layer overhead are required.
- To allow proper functionality of RAACM, there is need to deploy a transport protocol, the transport protocol provides a way at which packets from a host/computer gets to another host/computer. However, this thesis has deployed a user datagram protocol (UDP) for its transport protocol communication where the throughput, delay, data dropped WLAN analysis, and results has been recorded, our intention in the future, is to modify the transport layer by using the transmission control protocol (TCP) protocol and then analyse its result.
- In future it will be of interest to deploy a variable bit rate (VBR) for the voice and multimedia application that will be used in RAACM, as against the constant bit rate (CBR). It will be of utmost interest to see results from using the CBR for encoding and comparing them with the results obtained in VBR.
- In future, we will be interested in modifying the findings of the available bandwidth estimation and see how our estimation can work in different scenarios. Since this thesis was able to generate a formula to estimate the available bandwidth of a link (RAACM). Using RAACM, the future plan however is to incorporate a topology which is made up of six nodes where each node has a source and a destination as in Figure 7.1. We will therefore, estimate the available bandwidth of a link between two nodes (node 3 and node 4), the estimation of the available bandwidth will be based on the bandwidth consumed by another variable bandwidth link flow (F1). Where F2 has a constant bandwidth.

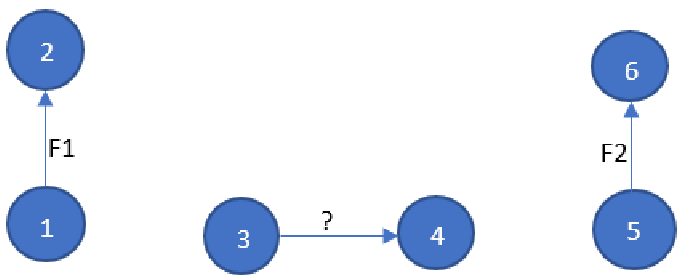


Figure 7.1

References

Abut, F, & Leischner, M, 2018, ‘ An experimental evaluation of tools for estimating bandwidth-related metrics’, *International Journal of Computer Network and Information Security*, vol. 10, no. 7, pp. 1-11.

Adarbah, H, 2015, *Bandwidth and Energy-Efficient Route Discovery for Noisy Mobile Ad-Hoc Networks*. DeMontford University, Leicester.

Agarwal, S, Kaur, R, Agarwal, T, 2018, ‘MANET Expansion Security Challenges Attacks and Intriguing future Trends’ *International Journal of Computer Sciences and Engineering,* pp. 906-915.

Ahmed, N, Rikli, N, 2018, ‘A QoS Based Algorithm for the Vertical Handover between WLAN IEEE 802.11 e and WiMAX IEEE 802.16 e’, *International Journal of Computing and Digital Systems,* vol.7, no.1, pp. 11-22.

Aina. F, Yousef, S, Osanaiye, O, 2018, ‘Bandwidth Estimation for Admission Control in MANET: Review and Conceptual MANET Admission Control Framework, *vol. 881 of the Advances in Intelligent Systems and Computing series*.

Aina, F, Yousef S, & Osanaiye, O, 2019, ‘RAAC: A Bandwidth Estimation Technique for Admission Control in MANET’, *Series: Electronics and Energetics Facta Universitatis,* vol. 32, no. 3, pp. 463-478.

Aina, F, Yousef S, & Osanaiye, O, 2019,’RAACM: Resource Allocation for Admission Control in MANET’, *International Journal of Wireless Information Networks*, vol.26, pp. 243-256.

Airon, N, &Gupta, N, 2017, ‘Bandwidth estimation tools and techniques: A review’, *International Journal of Research*, 10.20944/preprints201710.0060.v1.

Alabdullah, M, Atiyah, B, Khalaf, K, Yadger, S, 2019, ‘Analysis and simulation of three MANET routing protocols: A research on AODV, DSR & DSDV characteristics and their performance evaluation’, *Periodicals of Engineering and Natural Sciences*, vol. 7, no. 3, pp. 1228-1238.

Al-Hasanat, M, Al-Hasanat, A, Althunibat, S, Alsbou, A, 2017, ‘Performance evaluation of selected e2e TCP congestion control mechanism over 4G networks’, *International Journal of Wireless & Mobile Networks (IJWMN)* Vol. 9, no. 6, pp. 71-79.

Ali, R, & Zafar, F, 2011, ‘Bandwidth estimation in mobile ad-hoc network (MANET)’. *International Journal of Computer Science*, vol. 8, no. 5, pp. 331–337.

Al-Maqri, M, Othman, M, Ali, B, Hanapi, Z, 2016, 'Adaptive multi-polling scheduler for QoS support of video transmission in IEEE 802.11 e WLANs', *Telecommunication systems, Springer,* vol. 61, pp. 773-791.

Ambika, I, Jayachandran, A, 2017, 'Efficient Admission Control Mechanism for Multiple Traffic Flows in Mobile Ad Hoc Network', *Journal of Computational and Theoretical Nanoscience,* vol. 14, no.12, pp. 5915-2921.

Appandairaj, P, Kannan, K, 2020, 'Software-Defined Multilayered Admission Control for Quality of Service Assurance in Mobile Ad-hoc Networks', *Wireless Communications and Mobile Computing, hindawi,* pp. 1-23.

Atxutegi, E, Liberal, F, Saiz, E, & Ibarrola, E, 2016, 'Toward standardized internet speed measurements for end users: current technical constraints', *IEEE Communications Magazine*, vol. 54, no. 9, pp. 50-57.

Azevedo, D, Bonfim, M, Lima, L & Fernandes, S, 2018, 'Towards an Accurate Bandwidth Estimation Tool for 802.11n Wireless Networks', *IEEE Symposium on Computers and Communications (ISCC), Natal*, pp. 00486-00491.

Blake, S, Black, D, Carlson, M, Davies, E, Wang, Z & Weiss, W, 'An Architecture for Differentiated Services', RFC 2475 (Informational), Internet Engineering Task Force, May. 2018, updated by RFC 3260. [Online]. Available: http://www.ietf.org/rfc/rfc2475.txt.

Borzemski, & AKamińska-Chuchmała, L, 2016, Distributed web server's data performance processing with application of spatial econometrics models. In: Grzech A., Borzemski L., Świątek J., Wilimowska Z. (eds) Information Systems Architecture and Technology: Proceedings of 36th International Conference on Information Systems Architecture and Technology – ISAT 2015 – Part II. Advances in Intelligent Systems and Computing, vol 430. Springer.

Boz, E, Manner, J, 2020, 'A hybrid approach to QoS measurements in cellular networks'. *Computer Networks, Elsevier,*vol.172.

Braden, R, Clark, D & Shenker, 'Integrated Services in the Internet Architecture: an Overview', RFC 1633 (Informational), Internet Engineering Task Force, May. 2018. [Online]. Available: http://www.ietf.org/rfc/rfc1633.txt.

Budka, K, Deshpande, J, & Thottan, M, 2016, *Communication networks for smart grids*.

Chamraz, F, & Baroňák, I, 2015, ‘Impact of admission control methods to the traffic management’, *Information And Communication Technologies And Services,* vol. 13, no. 4, pp. 280-288.

Chaudhari, S, & Biradar,R, 2015, ‘Available bandwidth estimation using collision probability, idle period synchronization and waiting time’, *Wireless personal communication an international journal,* vol. 85, pp. 597-621.

Chaudhari, S & Biradar, R, 2014, ‘Collision probability based Available Bandwidth estimation in Mobile Ad Hoc Networks’, *The Fifth International Conference on the Applications of Digital Information and Web Technologies (ICADIWT 2014)*, Bangalore, 2014, pp. 244-249.

Cheikh, H, 2016, ‘Integration of streaming and elastic traffic: Modelling and Performance Analysis’, *Proceedings of the 9th EAI International Conference on Performance Evaluation Methodologies and Tool*, pp. 65-68.

Chromy, E, Holiencin, M, Baronak, I, & Behan,L, 2017, New Approach in Admission Control Realisation, International Journal of Control Systems and Robotics, pp.155-162.

Cisco, WLAN Quality of Service. [Accessed: 14-04-2020].

Clausen, T & Jacquet, P, “Optimized Link State Routing Protocol (OLSR),” *RFC 3626, IETF Network Working Group*, 2003. [Online]. Available: http://www.ietf.org/rfc/rfc3626.txt. [Accessed: 25-Feb-2018].

Cobo, L, Castro, H, & Mejía, C, 2017, ‘A Distributed Connection Admission Control Strategy for Wireless Multimedia Sensor Networks’, 2017, In: Solano A., Ordoñez H. (eds) Advances in Computing. CCC 2017. Communications in Computer and Information Science, vol 735. Springer, Cham.

Dey, B, Manjunath, D, & Chakraborty, S, 2011, ‘Estimating network link characteristics using packet-pair dispersion: A discrete-time queueing theoretic analysis’. *Computer Networks* vol. 55, no. 5, pp. 1052–1068.

Farooq, M, & Kunz, T, 2014, ‘BandEst: Measurement-Based Available Bandwidth Estimation and Flow Admission Control Algorithm for Ad Hoc IEEE 802.15.4-Based Wireless Multimedia Networks’, *International Journal of Distributed Sensor Networks,* pp. 1-15.

Farooq, M & Kunz, T, 2013, ‘Proactive Bandwidth Estimation for IEEE 802.15.4-Based Networks’, *IEEE 77th Vehicular Technology Conference (VTC Spring)*, Dresden, pp. 1-5.

Goebel, N, Krauthoff, T, Graffi, K, & Mauve, M, 2017, ‘Moving measurements: Measuring network characteristics of mobile cellular networks on the move’, *Computer Communications*, vol. 97, pp. 81-95.

Goswami, S, Joardar, S, Das, C, Kar, S, & Pal, D, 2017, *Performance Analysis of three routing protocols in MANET using NS-2 and ANOVA test with varying speed of nodes*, Ad-hoc Networks, Jesus Hamilton Ortiz and Alvaro Pachon de la Cruz, IntechOpen, DOI: 10.5772/66521. Available from: https://www.intechopen.com/books/ad-hoc-networks/performance-analysis-of-three-routing-protocols-in-manet-using-the-ns-2-and-anova-test-with-varying.-

Guck, J, Bemten, A, & Reisslein, M, 2017, ‘Unicast QoS routing algorithms for SDN: A comprehensive survey and performance evaluation’, *IEEE Communications Surveys & Tutorials*, vol. 20, No. 1, pp. 388-415.

Hasan, MZ, Al-Rizzo, H, 2017, ‘A survey on multipath routing protocols for QoS assurances in real-time wireless multimedia sensor networks’, *IEEE Communications Surveys & Tutorials,* vol. 19, no. 3, pp. 1424-1456.

Hu, Z, Zhang, D, Zhu, A, Chen, Z, & Zhou, H, 2012, ‘SLDRT: A measurement technique for available bandwidth on multi-hop path with bursty cross traffic’, *Computer Networks*, vol. 56, no.14, pp. 3247–3260.

Humeida, Y, Nilsson, M & Appleby, S, 2018, ‘TCP Congestion Response for Low Latency HTTP Live Streaming’, *IEEE 19th International Symposium on "A World of Wireless, Mobile and Multimedia Networks" (WoWMoM)*, Chania, pp. 14-21.

IETF, “The Dynamic Source Routing Protocol for Mobile Ad Hoc Networks.” [Online]. Available: http://www.ietf.org/proceedings/43/I-D/draft-ietf-manet-dsr-00.txt. [Accessed: 7-May-2018].

Inaba, T, Ozera, K, Sakamoto, S, Oda, T, Ikeda, M, & Barolli, L, 2017, ‘A testbed for admission control in WLANs: effects of RSSI on connection keep-alive time’, 2017, *31st International Conference on Advanced Information Networking and Applications Workshops (WAINA)*, Taipei, pp. 722-729.

ITU Quality of Service Regulation Manual, 2017. https://www.itu.int/dms_pub/itu-d/opb/pref/D-PREF-BB.QOS_REG01-2017-PDF-E.pdf.

Jagadev, N, & Pattanayak, B, 2018, ‘A survey on Bandwidth Management in MANET, 2018’, *International Journal of Engineering and Technology*, vol. 7, no.3.4. pp.38-41.

Jakimoski, K, Aresenovski, S, Gorachinova, L, Chungurski, S, IIiev, O, Djinevski, L, Kamcheva, E, 2016, 'Measurements of Available Bandwidth in Computer Networks', *International Journal of Grid and Distributed Computing*, vol. 9, no. 4, pp. 201-210.

Jansen, B, Goodwin, T, Gupta, V, Kuipers, F, & Zussman, G, 2018, 'Performance Evaluation of WebRTC-based Video Conferencing', *ACM SIGMETRICS*, vol. 45. No. 3, pp. 56-68, https://doi.org/10.1145/3199524.3199534.

Kalpana, V, Karthik, V, 2018, 'Bandwidth Constrained Priority Based Routing Algorithm for Improving the Quality of Service in Mobile Ad hoc Networks', *International Conference on Soft-computing and Network Security (ICSNS),* Coimbatore, pp. 1-8.

Ke, CH, Yang, CY, Chen, JL, 2017, 'A novel mapping mechanism for MPEG-4 video delivery over IEEE 802.11 e networks', *Multimedia Tools and Applications*, vol. 76, pp. 5937-5949.

Khan, B, Olanrewaju, R, Anwar, F, Najeeb, A, & Yaacob, M, 2018, 'A Survey on MANETs: Architecture, Evolution, Applications, Security Issues and Solutions', *Indonesian Journal of Electrical Engineering and Computer Science,* vol. 12, no. 2, pp. 832-842.

Khangura, S, & Fidler, M, 2017, 'Available bandwidth estimation from passive tcp measurements using the probe gap model', *IFIP Networking Conference (IFIP Networking) and Workshops*, Stockholm, pp.1-9.

Khangura, S, Fidler, M, & Rosenhahn, B, 2019, 'Machine learning for measurement-based bandwidth estimation', *Computer Communications*, vol. 144, pp.18-30.

Kirova, V, Siemens, E, Kachan, D, Vasylenko, O, & Karpov, K, 2018, 'Optimization of Probe Train Size for Available Bandwidth Estimation in High-speed Networks',- *MATEC Web of conferences*, https://doi.org/10.1051/matecconf/201820802001.

Kokkonis, G, Psannis, K, & Roumeliotis, M, 2018, 'Transferring wireless high update rate supermedia streams over IoT', *New Advances in the Internet of Things. Studies in Computational Intelligence*, vol 715, pp. 93-103.

Kua, J, Armitage, G, & Branch, P, 2017, 'A Survey of Rate Adaptation Techniques for Dynamic Adaptive Streaming Over HTTP', *IEEE Communications Surveys & Tutorials*, vol. 19, no. 3, pp. 1842-1866.

Kumar, V, 2009, ‘simulation and comparison of aodv and dsr routing protocols in manets”, *computer science and engineering department the par university Patiala.* http://hdl.handle.net/10266/84.5

Kundu, A, Mahindru, R, Salapura, V, 2019, ‘Admission control based on the end-to-end availability’,

Lakshman, L, Khan, RU, & Mishra RB, 2018, ‘MANETs: QoS and investigations on optimized link state routing protocol’, *I. J. Computer Network and Information Security*, vol. 10, pp. 26-37.

Lei, L, Zhang, T, Zhou, L, Chen, X, Zhang, C, & Luo, C, 2015, ‘Estimating the Available Medium Access Bandwidth of IEEE 802.11 Ad Hoc Networks With Concurrent Transmissions’, *IEEE Transactions on Vehicular Technology*, vol. 64, no. 2, pp. 689-701.

Li, M, Wu, Y.-L., & Chang, C.-R, 2013, ‘Available bandwidth estimation for the network paths with multiple tight links and bursty traffic’, *Journal of Network and Computer Applications*, 36(1), 353–367.

Li, M, & Chang, C, 2009, ‘A two-way available bandwidth estimation scheme for multimedia streaming networks adopting scalable video coding’. *IEEE Sarnoff symposium*, pp. 1–6.

Li, M, Claypool, M & Kinicki, R, 2008, ‘WBest: A bandwidth estimation tool for IEEE 802.11 wireless networks’, *33rd IEEE Conference on Local Computer Networks (LCN)*, Montreal, Que, pp. 374-381.

Liao, X, Yang,Z, Yuan, P, 2017, ‘An unscented Kalman filter based available bandwidth estimation algorithm for space bundle links’, *9th International Conference on Wireless Communications and Signal Processing (WCSP)*, Nanjing, pp. 1-6.

Lin, H, Liu, M, Zhou, A, Liu, H, & Li, Z, 2010, ‘A novel hybrid probing technique for end-to-end available bandwidth estimation’. *In IEEE 35th conference on local computer networks*, pp. 400–407.

Lu, Z, & Yang, H, 2012, ‘Unlocking the power of OPNET modeler’, 2012, *New York: Cambridge University Press*.

Marttinen, A, Heikkinen, TP, Carlen, P, Luoma, M, Peuhkuri, M, Manner, J, & Jantti, R, 2016, "Estimating KPIs in deployed heterogeneous networks," *IEEE Communications Magazine*, vol. 54, no. 10, pp. 158-165.

Megyesi, P, Botta, A, Aceto, G, Pescapé, A, 2017, 'Challenges and solution for measuring available bandwidth in software defined networks', *Computer Communications*, vol. 99, pp.48-61.

Menon, V, 2019, 'Optimized Opportunistic Routing in Highly Dynamic Ad hoc Networks', Preprint, 2019020130 (doi: 10.20944/preprints201902.0130.v1).

Milton,J, Subhash, J, & Premkumar, R, 2016, 'Bandwidth Estimation with Admission Control Scheme for QOS Routing in Mobile Adhoc Networks', *Australian Journal of Basic and Applied Sciences,* vol. 1, no.1, pp. 443-450.

Mukta, Gupta, N, 2019, 'Analytical Available Bandwidth Estimation in Wireless Ad-Hoc Networks Considering Mobility in 3-Dimentional Space'*, Scalable Computing: Practice and Experience,* vol. 20, no.2, pp. 317-333.

Nam, S, Kim, S, Lee, S, & Kim, H, 2013, 'Estimation of the available bandwidth ratio of a remote link or path segments'. *Computer Networks*, vol. 57, no. 1, pp. 61–77.

Ni, Y, Shuai, Q, Yang, C,& Liu, J, 2016, 'A Context-Aware Adaptive Streaming Media Distribution System in a Heterogeneous Network with Multiple Terminals'. *Mathematical Problems in Engineering*, https://doi.org/10.1155/2016/9576502.

Oshiba, T Nogami, Nihei, K, 2016, 'Robust available bandwidth estimation against dynamic behavior of packet scheduler in operational LTE networks', *IEEE Symposium on Computers and Communication (ISCC)*, pp. 1276-1283.

Panda, N, Pattanayak, B, 2018, 'Analysis of Blackhole Attack in AODV and DSR', *International Journal of Electrical and Computer Engineering (IJECE)*, vol. 8, no.5, pp. 3092-3102.

Park, H, & Roh, B, 2010, 'Accurate passive bandwidth estimation (APBE) in IEEE 802.11 wireless LANs', *Proceedings of the 5th international conference on ubiquitous information technologies and applications*, Sanya, pp. 1–4.

Paul, A, Tachibana, A & Hasegawa, T, 2016, 'Implementation Design of Available Bandwidth Measurement Scheme: A Proxy based Approach. *MOBIQUITOUS '16 Adjunct Proceedings, hiroshima, Japan,* pp. 257-262.

Paul, A, Tachibana, A, & Hasegawa, T, 2016. 'An Enhanced Available Bandwidth Estimation Technique for an End-to-End Network Path', *IEEE Transactions on Network and Service Management*, vol. 13, no. 4, pp. 768-781.

Paul, A, Tachibana, A & Hasegawa, T, 2014, 'NEXT: New enhanced available bandwidth measurement technique, algorithm and evaluation', *IEEE 25th Annual*

International Symposium on Personal, Indoor, and Mobile Radio Communication (PIMRC), Washington, DC, pp. 443-447.

Perkins, C, "Ad hoc On-Demand Distance Vector (AODV) Routing." [Online]. Available: https://www.ietf.org/rfc/rfc3561.txt. [Accessed: 07-May-2018].

Pinto, P, 2015, *Admission Control based on End-to-end Delay Estimation to Enhance the Support of Real-time Traffic in Wireless Sensor Networks*, Ph.D. dissertation, University of Porto.

Poorzare, R, Poorzare, A, Abedidarabad, S, 2018, 'Optimizing optical networks by using CWN algorithm', *Review of Computer Engineering Studies,* pp. 98-102.

Riverbed, "Riverbed Technologies - Network Simulator," [Online]. Available: http://www.riverbed.com/products/steelcentral/opnet.html?redirect=opnet. [Accessed 19 April 2018].

Rizal, A, Bandung, Y, 2017, 'Passive Available Bandwidth Estimation Based on Collision Probability and Node State Synchronization in Wireless Networks' *Journal of ICT Research and Applications,* vol.11, no. 2, pp. 131-150.

Saini, T, Sharma, S, 2019, 'Prominent unicast routing protocols for Mobile Ad hoc Networks: Criterion, classification, and key attributes', *Ad-Hoc Networks Elsevier,* vol. 89, pp. 58-77.

Salcedo, D, Guerrero, C, & Martinez, R, 2018, 'Available bandwidth estimation tools metrics, approaches and performance', *International Journal of Communication Networks and Information Security (IJCNIS)*, vol.10, no. 3, pp. 580-587.

Sanada, K, Shi, J, Komuro, N, & Sekiya,H, 2015,' End-to-end delay analysis for IEEE 802.11 string-topology multi-hop networks', *The Institute of Electronics, Information and Communication Engineers,* vol. E98, no. 7, pp. 1284-1293.

Sarr, C, Chaudet, C, Chelius, G & Guérin Lassous. I, 2008, 'Bandwidth Estimation for IEEE 802.11-Based Ad Hoc Networks', *IEEE Transactions on Mobile Computing,* vol.7, no.10, pp. 1228-1241.

Sedighizad, M, Seyfe, B, & Navaie, K, 2012, 'MR-BART: Multi-rate available bandwidth estimation in real-time', *Journal of Network and Computer Applications*, vol. 35, no. 2, pp. 731–742.

Selin, P, Hasegawa, K, & Obara, H, 2011, 'Available bandwidth measurement technique using impulsive packet probing for monitoring end-to-end service quality on the internet', *17th Asia- Pacific conference on communications,* pp. 518–523.

Shah, S, Ahmed, E, Xia, F, Karim, A, & Noor, R, 2018, ‘Adaptive beaconing approaches for vehicular ad hoc networks: A survey’. *IEEE Systems Journal*, vol. 12, no. 2, pp. 1263-1277.

Sharma, L, Lal, C, Sharma, D, & Kaliyar, P, 2018, *Enhancing QoS for Multimedia Services Using Mobility-Aware Bandwidth Estimation Algorithm in MANETs*, Springer Singerpore, Optical and Wireless Technologies.

Shi, C, He, S, Deng, Z, Zhang, X, Shen, C, 2019, ‘Adaptive access mechanism with delta estimation algorithm of traffic loads for supporting weighted priority in IEEE 802.11e WLANs’, *Journal of Ambient Intelligence and Humanized Computing*. vol. 10, pp. 1693–1702.

Shiobara, S, Okamawari, T, 2017, ‘A novel available bandwidth estimation method for mobile networks using a train of packet groups’, *Proceedings of the 11th International Conference on Ubiquitous Information Management and Communication,* vol. 59, pp. 1-7.

Sivaram, M, Porkodi, V, Mohammed, A, Manikandan, V, & Yuvaraj, N, 2019, ’Retransmission DBTMA Protocol With Fast Retransmission Strategy to Improve the Performance of MANETs’, *IEEE Access,* vol. 7, pp. 85098-85109.

Solano, A, Ordoñez, 2017, *Advances in Computing,* Springer, 12th Colombian Conference, Columbia.

Thouin, F, Coates, M, & Rabbat, M, 2011, ‘Large scale probabilistic available bandwidth estimation’, *Computer Networks*, vol. 55, no.9, pp. 2065–2078.

Tursunova, S, Inoyatov, K, & Kim, Y, 2010, ‘Cognitive passive estimation of available bandwidth (cPEAB) in overlapped IEEE 802.11 WiFi WLANs’, *IEEE network operations and management symposium,* pp. 448–454.

Vijay, B, & Malarkodi, B, 2016, Improved QoS in WLAN using IEEE 802.11e, Twelfth International Multi-Conference on Information Processing, pp. 17-26.

Wang, Q, & Jiang, J, 2016, ‘Comparative examination on architecture and protocol of industrial wireless sensor network standards’, *IEEE Communications Surveys & Tutorials*, vol. 18, no. 3, pp. 2197-2219.

Wang, H, Lee, K, & Li, E, 2014, ‘Timing is Everything: Accurate, Minimum Overhead, Available Bandwidth Estimation in High-speed Wired Networks’ *Proceedings of the 2014 Conference on Internet Measurement*, pp. 407-420.

Watson, B, Ma, Y, Cahill, J, &, Doughty, B, 2019, ‘Probing ligand removal and ordering at quantum dot surfaces using vibrational sum frequency generation spectroscopy’, *Journal of Colloid and Interface Science*, vol. 537, pp. 389-395.

Yang, P & Zhang, Y, 2012, ‘Available bandwidth estimating method in IEEE802.11e based mobile ad hoc network’. *9th international conference on fuzzy systems and knowledge discovery,* pp. 2138–2142.

Yang, T, Jin, Y, Chen, Y, & Jin, Y, 2017, ‘RT-WABest: A novel end-to-end bandwidth estimation tool in IEEE 802.11 wireless network’, *International Journal of Distributed Sensor Networks*, vol. 13, no. 2.

Yazid, M, Aïssani, D & Bouallouche-Medjkoune, L, 2017, ‘Modeling and Analysis of the TXOP Limit Efficiency with the Packet Fragmentation in an IEEE 802.11e-EDCA Network Under Noise-Related Losses’, *Wireless Personal Communications,* vol. 9, no. 2.

Zeb, A, Islam, A, Zareei, M, Mamoon, I, Mansoor, N, Baharun, S, Katayama, Y, & Komaki, 2016, ‘Clustering analysis in wireless sensor networks: The ambit of performance metrics and schemes taxonomy’, *International Journal of Distributed Sensor Networks*, vol. 12, no. 7.

Zhao, P, Yang, X, Yu, W, Dong, C, Yang, S, Bhattarai, S, 2014, ‘Towards efficient estimation of available bandwidth for IEEE 802.11-based wireless networks’ *Journal of Network and Computer Applications*, vol. 40, pp.116-125.

Zhao, H, Garcia-Palacios, E, Wei, J, & Xi, Y, 2009, ‘Accurate available bandwidth estimation in IEEE 802.11-based ad hoc networks’, *Computer Communications*, vol. 32, no.6, pp. 1050–1057.

Zhou, H, Ye, C, Duan, Y, Qi, Q, & Zhang, Y, 2018, ‘Estimating End-to-End Available Bandwidth for Cyber-Physical Applications in Hybrid Networks’, *Wireless Internet Conference, WICON, Lecture Notes of the Institute for Computer Sciences, Social Informatics and Telecommunications Engineering*, vol. 214, pp. 197-206.

APPENDIX