# Medium Access Control Mechanisms for Quality of Service in Wireless Computer Networks

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# DECLARATION

This is to certify that I am responsible for the work submitted in the thesis, and that neither the thesis nor the original work contained therein has been submitted to this or any institution for a higher degree.

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# ABSTRACT


## Medium Access Control Mechanisms for Quality of Service in Wireless Computer Networks

The fast growth and development of wireless computer networks and multimedia applications means it is essential that these applications can be transmitted over the standard IEEE 802.11 Medium Access Control (MAC) protocol with high Quality of Service (QoS). The lack of QoS in the standard IEEE 802.11 Distributed Coordination Function (DCF) results in applications with considerably different QoS requirements receiving similar network services. This means, the performance of time-sensitive applications with stringent delay, jitter, and packet loss requirements will not be met. Even time-insensitive applications will be treated unfairly because access to the medium is on a random basis. Therefore, the main aims of this thesis are: (i) to investigate the limitations and performance of the IEEE 802.11 DCF, (ii) to develop a comprehensive solution that provides effective and efficient QoS provisioning in IEEE 802.11 DCF scheme in a fair, scalable, and robust manner. The latter is achieved by developing novel MAC mechanisms for providing QoS in the IEEE 802.11 DCF for multimedia transmission.

The scarcity of channel capacity, unfairness and hidden terminal problems, multiple hops, and other conditions and parameters that affect QoS in a wireless network are issues which require in depth investigations and analysis. In this thesis, extensive investigations using the network simulator 2 (*NS-2*) package were carried out to examine the impact of these issues on the main QoS parameters (throughput, delay, jitter, packet loss and collision). The findings revealed that the IEEE 802.11 DCF protocol performed inadequately when transmitting various applications due to the limitations inherent in its operation. The performance of the IEEE 802.11 DCF protocol was also investigated by studying the impact of varying the values of minimum Contention Window ($CW_{min}$) and the Distributed Inter Frame Space (*DIFS*). The study shows that inappropriate values of $CW_{min}$ and *DIFS* resulted in significant network performance degradations and demonstrated that it was important to select an appropriate set of MAC protocol transmission parameters in order to provide improved QoS.

Artificial Intelligence (*AI*) techniques using fuzzy logic and Genetic Algorithms (*GAs*) for assessing and optimising MAC protocol transmission parameters were developed and their effectiveness evaluated. The study confirmed that the application of *AI* techniques significantly improved the QoS for audio and video applications by more than 50% and fairly shared the channel access among the contending stations as compared to the standard IEEE 802.11 DCF scheme. Ratio based and Collision Rate Variation (*CRV*) schemes were developed to dynamically adjust the *CW* and *DIFS* values according to the current and past network conditions. Using these schemes significant improvements with service differentiation were achieved in an Independent Basic Service Set (*IBSS*). A queue status monitoring technique was devised for the intermediate stations. This provided QoS differentiation at the MAC layer for multi-hop networks. Autoregressive (*AR*) models that accurately predicted the network parameters were also developed. These enabled the MAC protocol transmission parameters to be adjusted in an improved manner. Using these models, average QoS was improved by more than 60%; average delay, packet loss and collision were reduced by more than 50% compared to IEEE 802.11 DCF scheme.

This led to the development of novel MAC mechanisms to provide QoS in IEEE 802.11 MAC protocol. The mechanisms support multiple QoS metrics and consider traffic history and predict future network conditions. The schemes also are characterised by the simplicity, robustness, and ease of implementation. The contribution of this thesis is the development of a comprehensive solution to provide effective and efficient QoS differentiation in IEEE 802.11 DCF scheme for multimedia transmission in a distributed, fair, scalable, and robust manner. Furthermore, through the use of these approaches, the findings of this study provide a framework that also contributes to the knowledge concerning the QoS over the IEEE 802.11 MAC protocol.

# LIST OF PUBLICATIONS

In the course of completing this study, the following articles were published.

1. M. Saraireh, R. Saatchi, U. Shur, and R. Strachan. " Fuzzy Logic Based Evaluation of Quality of Service for Multimedia Transmission". In Proceeding of PREP 2004, EPSRC, University of Hertfordshire, Page(s): 13-14, 5-7 April 2004, United Kingdom.

2. M. Saraireh, R. Saatchi, S. Alkhayatt and R. Strachan. " IEEE 802.11 MAC Protocol Quality of Service Performance Analysis". In Proceeding of PREP 2005, EPSRC, Lancaster University, Page(s): 171-172, 30$^{th}$ March - 1$^{st}$ April 2005, United Kingdom.

3. M. Saraireh, R. Saatchi, S. Alkhayatt and R. Strachan. "A Performance Comparison of IEEE 802.11 MAC Access Mechanisms for Different Traffic Types. ", In the Proceeding of the 6$^{th}$ Annual Post Graduate Symposium on the Convergence of Telecommunications, Networking and Broadcasting, Liverpool Johm Moores University, Page(s): 313-318, ISBN: 1-902-56011-6, 27 & 28 June, 2005, United Kingdom.

4. M. Saraireh, R. Saatchi, S. Alkhayatt and R. Strachan. " Impact of Varying the Minimum Value of Contention Window (CW*min*) of the IEEE 802.11 MAC Protocol on the QoS Parameters ", In Proceeding of 10th IFIP International Conference on Personal Wireless Communications PWC'05, Page(s): 219-226, ISBN: 1-86094-582-1, August 25-27, 2005, France.

5. M. Saraireh, R. Saatchi, S. Alkhayatt and Rebecca Strachan. "A Comparative Study of IEEE 802.11 MAC Access Mechanisms for Different Traffic Types. ". In Proceeding of ICETE, 2$^{nd}$ International Conference on E-business and Telecommunication Networks, Microsoft Convention Centre at Reading, vol. 2, Page(s): 28-35, ISBN:972-8865-32-5, October 3-7, 2005, United Kingdom.

6. M. Saraireh, R. Saatchi, S. Al-khayatt and R. Strachan , "A Comparative Study of IEEE 802.11 MAC Access Mechanisms for Different Traffic Types", 2$^{nd}$ International Conference on E-business and Telecommunication Networks ICETE 2005 Best paper, A book will be published by Springer with ICETE 2005 best papers, 2005, United Kingdom.

7. M. Saraireh, R. Saatchi, S. Al-khayatt and R. Strachan, "Development and Evaluation of a Fuzzy Inference Engine System to Incorporate Quality of Service in IEEE 802.11 Medium Access Protocol", IEEE International Conference on Wireless and Mobile Communications ICWMC'06, Page(s): 29-34, ISBN: 0-7695-2629-2, July 29-31, 2006, Romania.

8. M. Saraireh, R. Saatchi, S. Al-khayatt, R. Strachan, and Z. Abo-Hammour, "Optimisation of IEEE 802.11 MAC Protocol Parameters Using a Hybrid Genetic-Fuzzy Approach", Proceedings of the IEEE Systems, Man and Cybernetics Society United Kingdom & Republic of Ireland Chapter 5$^{th}$ Conference on Advances in Cybernetic Systems AIC2006, Sheffield Hallam University, Page(s): 253-258, ISSN 1744-9170, September 7-8, 2006, United Kingdom. *This paper was awarded the best paper prize.*

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# DEDICATION

I dedicate this work to:

- *My parents, for their invaluable support throughout the years, who devoted their life to the achievement of this dream.*
- *My lovely wife, Seham, whose unconditional love makes everything possible.*
- *My son Ahmad and my daughters Leena and Layan, who make every day new and precious.*
- *My brothers and sisters who shared with me my dream.*

# GLOSSARY TERMS

| | |
|---|---|
| **AC**: | Access Category. |
| **ACK**: | Acknowledgement. |
| **ADIFS**: | Adaptive Distributed Inter Frame Space |
| **AEDCF**: | Adaptive Enhanced Distributed Coordination Function. |
| **AI**: | Artificial Intelligence. |
| **AIFS**: | Arbitration Inter-frame Space. |
| **AM**: | Amplitude Modulation. |
| **AODV**: | Ad-hoc On-demand Distance Vector. |
| **AP**: | Access Point. |
| **AR**: | Autoregressive. |
| **ARC**: | Adaptive Rate Controller. |
| **ARQ**: | Automatic Repeat and Request. |
| **ARMA**: | AutoRegressive Moving Average. |
| **ARME**: | Assured Rate MAC Extension. |
| **ATM**: | Asynchronous Transfer Mode |
| **BEB**: | Binary Exponential Backoff. |
| **BI**: | Backoff Interval. |
| **BS**: | Base Station. |
| **BSS**: | Basic Service Set. |
| **CAC**: | Call Admission Control. |
| **CBR**: | Constant Bit Rate. |
| **CF-End**: | Contention Free End. |
| **CFP**: | Contention Free Period. |
| **CoG**: | Centre of Gravity. |
| **CoGS**: | Centre Of Gravity method for Singletons. |
| **CP**: | Contention Period. |
| **CP_Threhold**: | CaPtured effect Threshold. |
| **CRA**: | Contention Resolution Algorithm. |
| **CR**: | Collision Rate. |
| **CRV**: | Collision Rate Variation. |
| **CS_Threshold**: | Carrier Sensing Threshold. |
| **CSMA/CA**: | Carrier Sense Multiple Access with Collision Avoidance. |
| **CSMA/CD**: | Carrier Sense Multiple Access with Collision Detection. |
| **CTS**: | Clear To Send. |
| **CW**: | Contention Window. |
| **$CW_{max}$**: | Maximum Contention Window. |
| **$CW_{min}$**: | Minimum Contention Window. |
| **CWD**: | Contention Window Differentiation. |
| **CWS**: | Contention Window Separation. |
| **DA**: | Demand Assignment. |
| **DCF**: | Distributed Coordination Function. |
| **DIFS**: | Distributed Inter Frame Space. |
| **DLL**: | Data Link Layer |
| **DS**: | Distribution System. |
| **DSF**: | Distributed Fair Scheduling. |
| **DSP**: | Digital Signal Processing. |
| **DSSS**: | Direct Sequence Spread Spectrum. |
| **DWFQ**: | Distributed Weighted Fair Queuing. |
| **EDCA**: | Enhance Distributed Channel Access |
| **EDCF**: | Enhanced Distributed Coordination Function. |

| | |
|---|---|
| **EIED:** | Exponential Increase Exponential Decrease. |
| **EIFS:** | Extended Inter frame Space. |
| **EM:** | Electromagnetic. |
| **ESS:** | Extended Service Set. |
| **ETSI:** | European Telecommunication Standards Institute. |
| **EWMA:** | Exponentially Weighted Moving Average. |
| **FAMA:** | Floor Acquisition Multiple Access. |
| **FCR:** | Fast Collision Resolution. |
| **FDD:** | Frequency Division Duplexing. |
| **FEC:** | Forward Error Correction. |
| **FER:** | Forward Error Correction. |
| **FHCF:** | Fair-scheduling for HCF. |
| **FHSS:** | Frequency Hopping Spread Spectrum. |
| **FIFO:** | First In First Out. |
| **FIS:** | Fuzzy Inference System. |
| **FLC:** | Fuzzy Logic Control. |
| **FM:** | Frequency Modulation. |
| **FTP:** | File Transfer Protocol. |
| **FuRWA:** | Fuzzy Reasoning for Wireless Awareness. |
| **GA:** | Genetic Algorithms. |
| **GDCF:** | Gradual Distributed Coordination Function. |
| **GloMoSiM:** | Global Mobile Information Systems Information Library. |
| **HCF:** | Hybrid Coordination Function. |
| **HIPERLAN:** | High Performance European Radio LAN. |
| **IBSS:** | Independent Basic Service Set. |
| **IF_range:** | Interference range. |
| **IFQ:** | Interface Queue. |
| **IFS:** | Inter Frame Space. |
| **IR:** | Infrared. |
| **ISI:** | Inter-Symbol Interference. |
| **ISM:** | Industry, Science and Medical. |
| **LLC:** | Logical Link Control. |
| **LM:** | Leftmost maximum. |
| **LMILD:** | Linear/Multiplicative Increase and Linear Decrease. |
| **MA:** | Moving Average. |
| **MAD:** | Mean Absolute Deviation. |
| **MAE:** | Mean Absolute Error |
| **MAC:** | Medium Access Control. |
| **MACA:** | Multiple Access with Collision Avoidance. |
| **MACAW:** | Multiple Access with Collision Avoidance for Wireless. |
| **MANET:** | Mobile Ad-Hoc Networking. |
| **MILD:** | Multiplicative Increase and Linear Decrease. |
| **MIMO:** | Multi Input Multi Output. |
| **MoM:** | Mean Of Maxima. |
| **MRE:** | Mean Relative Error. |
| **MS:** | Master Scheduler. |
| **MSE:** | Mean Square Error, |
| **MMPDU:** | Management MAC Service Data Unit. |
| **MSDU:** | MAC Service Data Unit. |
| **NAM:** | Network AniMator. |
| **NAV:** | Network Allocation Vectors. |
| **NS-2:** | Network Simulator 2. |

| | |
|---|---|
| **OFDM**: | Orthogonal Frequency Division Multiplexing. |
| **OPNET**: | Optimised Network Engineering Tools. |
| **OSI**: | Open System Interconnection. |
| **OTCL**: | Object Tool Command Language. |
| **PAN**: | Personal Area Network. |
| **PC**: | Point Coordinator. |
| **PCF**: | Point Coordination Function. |
| **PCS_range**: | Physical Carrier Sensing Range. |
| **PDA**: | Personal Digital Assistant. |
| **PF**: | Persistence Factor. |
| **PHY**: | PHYsical layer. |
| **PIFS**: | Point Inter Frame Space. |
| **PLCP**: | Physical Layer Convergence Procedure. |
| **PMD**: | Physical Medium Dependent. |
| **QoS**: | Quality of Service. |
| **RERR**: | Route ERRor. |
| **RM**: | Rightmost Maximum. |
| **RMSE**: | Root Mean Square Error |
| **RRA**: | Random Reservation Access. |
| **RREP**: | Route REPly. |
| **RREQ**: | Route REQuest. |
| **RSSI**: | Received Signal Strength Indicator. |
| **RT_Threshold**: | Received Threshold. |
| **RTS**: | Request To Send. |
| **SBA**: | Sensing Backoff Algorithm. |
| **SCFQ**: | Self-Clocked Fair Queuing. |
| **SD**: | Slow Decrease. |
| **SIFS**: | Short Inter Frame Space. |
| **SISO**: | Single Input Single Output. |
| **SS**: | Slave Scheduler. |
| **TBTT**: | Target Beacon Transition Time. |
| **TC**: | Traffic Category. |
| **TCL**: | Tool Command Language. |
| **TCMA**: | Tiered Contention Multiple Access. |
| **TCP**: | Transmission Control Protocol. |
| **TDD**: | Time Division Duplexing. |
| **ToS**: | Type of Service. |
| **TTL**: | Time-To-Live value. |
| **TX_range**: | Transmission range. |
| **TXOP**: | Transmission Opportunity. |
| **UDP**: | User Datagram Protocol. |
| **UHF**: | Ultra High Frequency. |
| **VBR**: | Variable Bit Rate. |
| **VMAC**: | Virtual MAC. |
| **WLAN**: | Wireless Local Area Network. |
| **WMAN**: | Wireless Metropolitan Area Network. |
| **WPAN**: | Wireless Personal Area Network. |
| **WWAN**: | Wireless Wide Area Network. |

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# *CHAPTER 1*

# Introduction

## 1.1 Background and Motivation

Wireless communications is a technology that is becoming an important feature of many aspects of our daily life. Not only are computer networks becoming mobile, many devices will have one or several wireless interfaces such as laptops, cameras, and phones (Bray, 2001). In some cases these devices and even fixed stations wish to communicate with each other without requiring an infrastructure. In these cases, there is a need for ad-hoc wireless networks to provide an effective network communication between different wireless devices. This type of network has a number of applications such as conferences, emergency operations, and military operations (Ilyas and Qazi, 2003) and (Tsai and Tu, 2004). In ad-hoc networks, the devices need to be within the transmission range of each other in order to be able to establish a direct communication and to compete with each other to access the wireless medium. If the devices are out of the transmission range of each other due to lack of transmission power, long distance between the wireless devices, interference, noise or due to mobility, it becomes essential to have an intermediary node between them (Tsai and Tu, 2004). This node results in a multi-hop ad-hoc network. In this type of network a routing protocol is required and the Medium Access Control (MAC) protocol has to share the media with fairness between different devices for different applications (IEEE, 1999) and (Tsai and Tu, 2004).

Wireless networks face a large number of challenges that the conventional wired networks do not. Wired networks can easily communicate over wires with small probability of error. Nodes in wired networks are capable of listening to the medium while they are transmitting and can use collision detection procedures. The wired topology rarelychanges since nodes are usually fixed. This simplifies the routing of data packets between the communication parties. These features do not exist in a wireless network. Here bandwidth is scarce and expensive, and there are unpredictable wireless link properties. Wireless networks communicate over a highly error prone medium (Xin et *al.*, 2006). Collisions become harder to detect since in wireless systems it is very difficult for any node to listen to the medium while transmitting on the same channel. In wireless networks the transmission power is much greater than the received signal power since if a node attempts to listen to the channel while transmitting, it would need to receive what is being transmitted by itself (i.e. self interference). As a result, collision detection is not applicable to wireless networks. Consequently, wireless networks adopt a collision avoidance mechanism and a lack of acknowledgement reply is the only indication of collision occurrence. The wireless topology can change randomly and rapidly and node membership within the network is also not fixed. The wireless channel has lower capacity than a wired network due to bandwidth-constrained and hence congestion is more problematic. These are just some of the challenges that wireless networks experience. This study concentrates on the issue of Medium Access Control (MAC) in wireless networks where the properties of the wireless medium such as half-duplex mode operation, unfairness, hidden terminal, exposed terminal, and capture effect make the operation of MAC protocol very challenging (Gummalla, 2000) and (Chakrabarthi and Mishra, 2001).

The function of the MAC protocol is to provide efficient and fair sharing of medium among all stations in the network. In wireless networks, MAC protocols can be categorised either as distributed or centralised protocols (Crow et *al.*, 1997). Distributed wireless networks, also called ad-hoc networks, are wireless stations communicating with one another without any need for central administration. Centralized wireless networks, are extensions to wired networks and have Access Points (*AP*) that act as the interface between wireless and wired networks. The *AP* polls the stations before assigning access rights in turn and a station is only permitted to send when it is allocated the right to do so. An example of a centralised protocol is the IEEE 802.11 Point Coordination Function (PCF). Distributed protocols are contention algorithms that permit stations in ad-hoc networks to be able to communicate according to the Carrier Sense Multiple Access with Collision Avoidance (*CSMA/CA*) mechanism. The IEEE 802.11 Distributed Coordination Function (DCF) is one example of a distributed protocol. Other examples include *ALOHA* (Roberts, 1975), *HIPERLAN* (ETSI, 2000), *MACAW* (Bharghavan et *al.*, 1994).

The fast growth of wireless technology has been accompanied by the rapid growth of multimedia applications. Those applications have strict requirements on network parameters, particularly, Quality of Service (QoS) parameters such as throughput, delay, delay variation (jitter), packet loss, and collision. Therefore, exceeding these requirements either decreases the communication quality or degrades it completely, since multimedia quality is governed by QoS offered by the network.

Quality of Service is defined as a set of service requirements to be met by the network while transporting data packets from source to destination (Crawley, 1998). Therefore, assuring or guaranteeing the QoS requirements in wireless networks is very demanding because the wireless channel has variable characteristics due to bandwidth limitation, interference, noise, signal attenuation and signal fading in addition to the above-mentioned problems. In order to deal with these problems, many wireless networks schemes have been defined. Some of these enhance the QoS of the whole system; others differentiate between the priorities of each station to give it different QoS parameters (Sobrinho, 1999). The IEEE 802.11 was proposed with minimal QoS provision and supports a best effort service. This minimal support is insufficient for transmitting multimedia applications such as real time audio/video over the wireless channel. Thus, providing QoS to wireless networks has become an area of active research.

The DCF is the main function of the standard IEEE 802.11 protocol that operates in a distributed manner (IEEE, 1999). In this function all wireless devices compete between each other to access the channel without the presence of a centralised controller. The competition process in a heavily loaded medium may lead to unpredictable network performance. In the standard IEEE 802.11 MAC, the centralised scheme (Point Coordination Function, PCF) was an attempt to provide reliable access to the wireless medium (IEEE, 1999). But problems with this scheme directed the IEEE standards group to propose a newer version called IEEE 802.11e (IEEE, 2004). The aim of this proposed scheme is to provide QoS guarantees to IEEE 802.11 based networks by considering the medium access control mechanisms. This version is not finalised yet but has drawn much attention from both the research community and industry. Therefore, the distributed function of the standard IEEE 802.11 MAC is considered for this study.

For a system with little complexity (i.e. little uncertainty) mathematical equations can provide precise representation of its operation. For more complex systems where significant data is available, model-free techniques such as Artificial Intelligence (*AI*) effectively reduce the complexity. For the most complex systems where few numerical data exist and only imprecise information is available, *AI* provides an effective way for understanding them (Ross, 1995). Realisation of medium access control that caters for QoS is a complex process as it involves imprecise information from the measured data (i.e. delays, jitter, packet loss, throughput, and collision). Furthermore, the dynamics of the channel vary in space and time in a complex manner. Therefore, the applications of fuzzy logic and Genetic Algorithms (*GAs*) besides other conventional approaches as part of IEEE 802.11 medium access control are valuable.

The main area of concern in this study is to investigate the limitations of the DCF medium access control protocol, improve its operation, and provide QoS when transmitting various applications. This can be achieved by incorporating novel MAC mechanisms that are based on both traditional and artificial intelligence techniques to adjust the main MAC protocol transmission parameters and to predict the network conditions.

## 1.2 Research Aim and Objectives

The aim of this research is to propose techniques which will result in improvements to the operation of IEEE 802.11 based Medium Access Control (MAC) mechanism and data transmission process in wireless computer networks. The study leads to improvements in the QoS provided by wireless computer networks when transmitting different applications. The overall objectives of this study are to:

(i) Investigate the limitations of the medium access control mechanism (MAC) currently used in IEEE 802.11. In particular, analyse the performance of the legacy system which is based on the Distributed Coordination Function (DCF) for transmitting different applications.

(ii) Investigate the impact of various MAC protocol transmission parameters such as; minimum Contention Window ($CW_{min}$), Distributed Inter Frame Space (*DIFS*), and the number of retry limits on the QoS parameters e.g. delay, jitter, throughput, packet loss, and collision for *CBR* and *VBR* traffic.

(iii) Quantitatively, evaluate, assess, and analyse the QoS performance of wireless networks for transmission of multimedia applications.

(iv) Investigate the effectiveness of intelligence techniques such as fuzzy logic, and Genetic Algorithms (*GAs*), for determining the optimum transmission conditions such as optimum *CW* size and optimal *DIFS* value when transmitting various applications.

(v) Develop novel approaches based on traditional techniques to dynamically adjust the main MAC protocol transmission parameters for various traffic types and different operation conditions.

(vi) Investigate the possibility of predicting network conditions such as the number of collisions when stations attempt to access the channel and whether this prediction ability can be used to improve the MAC operation. This investigation should be achieved for various topologies, operating conditions and application types.

## 1.3 Research Contributions

Research on improving the performance of the MAC protocol through developing novel MAC mechanisms and providing QoS is critical in the case of wireless networks. The schemes proposed in this study contribute in expanding the boundaries of knowledge within the IEEE 802.11 areas. A brief outline of each contribution is given below for each of the objectives outlined above; and a detailed description of each item will be discussed later in relevant chapters.

(i) A detailed investigation on the limitations and performance of the basic IEEE 802.11 DCF scheme was carried out for different operation conditions, network sizes, traffic types, number of connections and for both MAC protocol access mechanism. This included studying the impact of MAC protocol transmission parameters such as $CW_{min}$, *DIFS*, and the number of retry limits on the QoS parameters. These evaluations formed suitable baselines for this study.

(ii) According to these evaluations, new mechanisms based on *AI* and traditional techniques were developed. A fuzzy logic based approach was proposed to assess the QoS provided by the network for various applications. It combines several QoS parameters to obtain one output that represents the QoS of the transmitted application. So far, there have been no known studies to assess the QoS for various applications using the fuzzy logic approach. Fuzzy logic and genetic algorithms were also developed to adjust the main MAC protocol transmission parameters. A fuzzy logic controller was devised to adjust the $CW_{min}$ size according to the assessed QoS, other network conditions and parameters such as collision rate, and previous $CW_{min}$ values. A genetic algorithm optimisation technique was also developed to optimise the $CW_{min}$ and *DIFS* for different network configurations. The use of these techniques provided valuable tools for improving the performance of the IEEE 802.11 MAC protocol. Using *AI* techniques for adjusting multiple MAC protocol transmission parameters according to the assessed QoS and other network parameters have rarely been carried out for the IEEE 802.11 MAC protocol. Most of the previous work focused only on one or two QoS parameters without combining them together in one quantity. Other studies have not used fuzzy logic and *GAs* mechanisms for optimising multiple MAC protocol transmission parameters to provide QoS.

(iii) Ratio based and Collision Rate Variation (*CRV*) schemes were developed for online adaptations of *CW* and *DIFS* MAC protocol transmission parameters. These were based on the current and previous collision rate and collision rate variation values. Ratio based and Collision Rate Variation (*CRV*) schemes were developed to extend the operation of IEEE 802.11 DCF scheme in order to improve its performance and to provide service differentiation. Different from other studies, Ratio based and *CRV* schemes considered the QoS parameters and the QoS assessed by fuzzy logic approach in the adaptation process. In other studies, although there were many efforts dedicated to improve the protocol performance, they only adjusted one parameter such as *CW*, rarely adjusted the *DIFS*, and nor combined them at runtime (i.e. online adaptation). Adaptive service differentiation schemes for providing service differentiation among time-sensitive and time-insensitive applications in single-hop networks were developed. These schemes are a continuation of the Ratio based and *CRV* schemes. The values of collision rate, collision rate variation and packet loss were considered for service differentiation under various network conditions and operations. Incorporating the queue status monitoring approach proposed in this study with the adaptive service differentiation scheme, service differentiation in multi-hop networks was achieved.

(iv) Regression models were developed that predict the collision rate, collision rate variation, queue status occupancy and *CW*. These schemes incorporated the collision ratio, collision rate variation, queue status ratio and packet loss values to adjust the MAC protocol transmission parameters and to provide service differentiations. The regression models developed in this study predicted the collision ratio, collision rate variation, *CW*, and queue occupancy which were not often discussed in the literature.

## 1.4 Thesis Organisation

The outline of the thesis is schematically shown in Figure 1.1. Chapter 2 covers the theory of general aspects of wireless technologies, the electromagnetic spectrum, and the development of Wireless Local Area Networks (*WLANs*) technology. The IEEE 802.11 standards are briefly outlined including a description of the Physical (*PHY*) layer. A detailed description of the IEEE 802.11 MAC protocol is provided. This includes the protocol architecture, functions, limitations, and access mechanism and a detailed review of the current state of the art. This identifies the main drawbacks of the existing mechanism and shows how this can be mitigated and addressed by new MAC mechanisms. This chapter also outlines the QoS aspects from the perspective of medium access control. This is then followed by reviewing the current state of the art in the area of QoS in MAC protocol including its limitations and how these can be minimised by the new service differentiation schemes proposed for this study.

Chapter 3 provides an examination of the relevant background for the area of artificial intelligence. An introduction to fuzzy logic system, its structure, and an illustrative example are presented. A description of the use of genetic algorithms as an optimisation technique is provided. The main steps of the conventional genetic algorithm technique are also presented. The applications of *AI* techniques in wireless domain, especially in the IEEE 802.11 MAC protocol area are identified. The justifications of using the *AI* techniques are provided.

Chapter 4 outlines the research methodology being employed to meet the research objectives. Network simulation overview and topologies, assumptions made and how they affected the network performance are discussed. The simulation and measurement models that are used throughout this thesis are explained.

Chapter 5 outlines the limitations and performance of the IEEE 802.11 DCF scheme. It investigates the limitation of the protocol such as unfairness, hidden terminal problem, and multi-hop networks through simulation under different network condition operations, various traffic types, and both MAC protocol access mechanisms. The performance of the protocol under different operating conditions and the impact of MAC protocol transmission parameters are investigated. This chapter represents the baseline of the proposed approaches for this study.

Chapter 6 introduces the use of *AI* techniques in the area of IEEE 802.11 MAC protocol. Fuzzy logic is used to assess QoS for various multimedia applications. Fuzzy logic and genetic algorithms are used to adjust and optimise the main MAC protocol transmission parameters.

Chapter 7 introduces the Ratio based and the Collision Rate Variation (*CRV*) schemes. It reviews the state of the art in the field of MAC protocol adaptation mechanisms, and provides a full description of the operation of the Ratio based and *CRV* schemes. The results obtained when the performance of these schemes was evaluated for different operating conditions are discussed.

In Chapter 8, the features of the Ratio based and *CRV* schemes to provide service differentiation in single and multi-hop networks at runtime are provided. A review of the current state of the art is introduced. A description of the proposed adaptive service differentiation schemes and the results obtained are discussed.

In Chapter 9, the use of Autoregressive (*AR*) prediction models to improve the performance of IEEE 802.11 MAC protocol and to provide service differentiation is introduced. The relevant background for the area of linear regression and the related work in the area of wireless networks, in particular in the IEEE 802.11 MAC protocol are provided. A description of the proposed *AR* models and the results obtained by employing these methods are discussed.

Chapter 10 provides the overall findings of this research followed by the conclusions drawn and points towards future research directions.

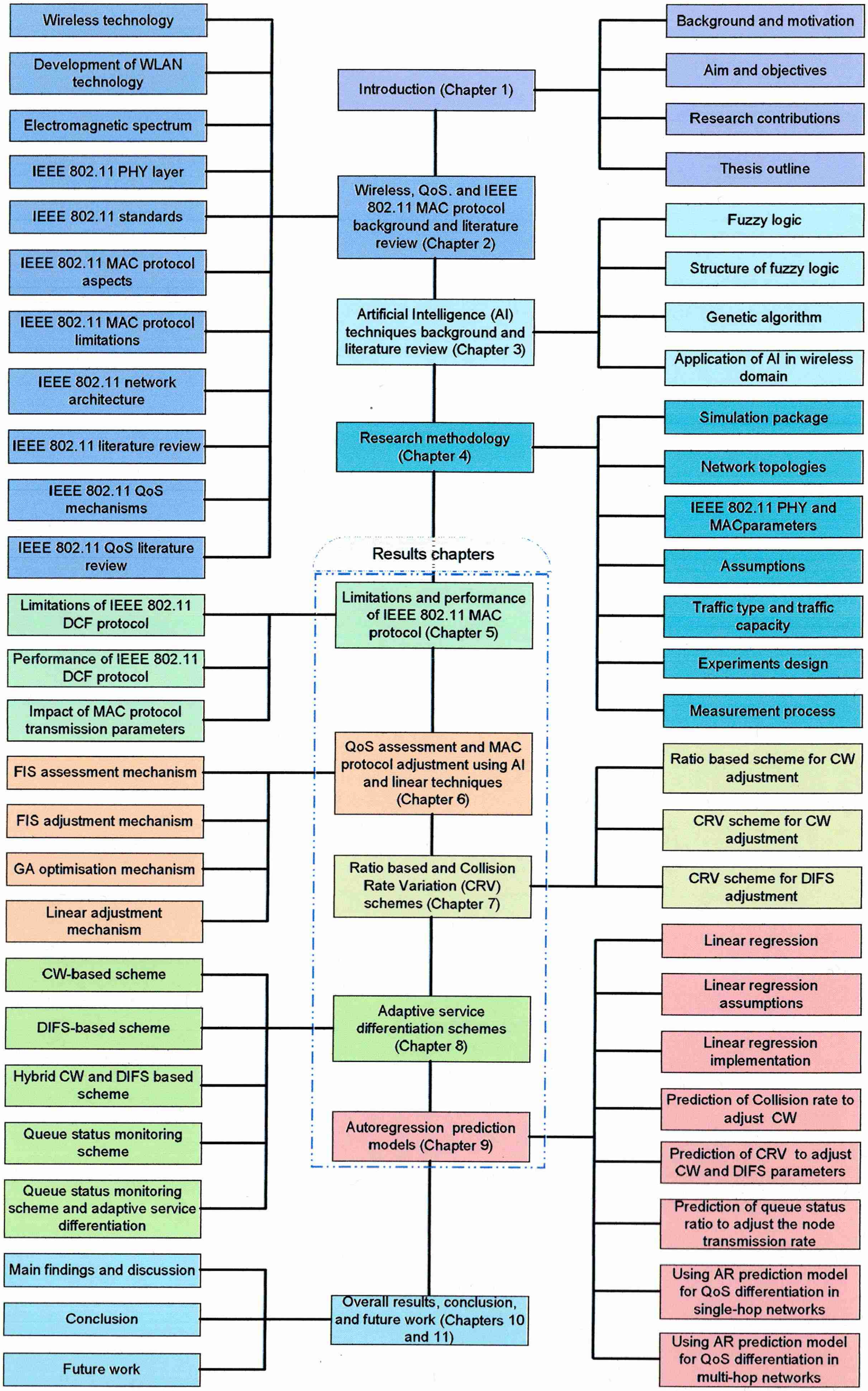



**Figure 1.1:** Schematic overview of the thesis.

# *CHAPTER 2*

# IEEE 802.11 DCF Protocol and Literature Review

## 2.1 Introduction

A variety of communication media are used for transmitting applications such as voice, video and data. Fibre optics cable can be used to carry light between communication parties, water waves can carry sound signals, while different type of cables such as coaxial and Unshielded Twisted Pairs (*UTP*) can be used to carry radio or Electromagnetic (*EM*) waves between the two ends of communication. Each medium has different propagation properties which makes it convenient for some applications and not for others. Moreover, light signals, sound waves and *EM* radiation can be used with different applications (voice, video, and data) through the free space.

The random transmission of applications over the wireless medium may lead to incomprehensible or unpredictable results. Therefore, a controller which manages access to the medium of the sharing resources is an essential tool for achieving a successful transmission process between the communication parties, and ensuring access is fair and suitable.

The MAC protocol in wireless networks is the protocol that controls access to the shared medium by applying rules and procedures that permit the communication pairs to communicate with each other in an efficient and fair manner. The IEEE 802.11 protocol covers the physical layer (*PHY*) and the Medium Access Control (MAC) sub-layer of the Open System Interconnection (*OSI*) (Cisco, 2003).

The rest of the chapter is organised as follows: The next section provides general description about wireless technologies. The electromagnetic spectrum is introduced in section 2.3. The development of *WLAN* and the *WLAN* standards are introduced in sections 2.4 and 2.5, respectively. The IEEE 802.11 network architectures are discussed in section 2.6. The *WLAN* technology is introduced in section 2.7. The physical layer characteristics of the IEEE 802.11 MAC protocol is presented in section 2.8. A detailed description of the IEEE 802.11 MAC protocol is outlined in section 2.9. Section 2.10 discusses the limitations of the IEEE 802.11 protocol. The state of the art in the area of IEEE 802.11 DCF scheme and its performance is discussed in section 2.11. An overview of Quality of Service (QoS), QoS components, QoS parameters, and IEEE 802.11 QoS mechanisms are discussed in section 2.12. In the last section, issues that need further investigations are introduced.

## 2.2 Wireless Technologies

Wireless technologies have shown a rapid growth during recent years. They include Wireless Wide Area Network (*WWAN*), Wireless Metropolitan Area Network (*WMAN*), *WLAN*, Wireless Personal Area Network (*WPAN*), and Ubiquitous technology. This classification of wireless technologies is based on the coverage area and the data rate as shown in Figure 2.1 (Hännikäinen$_{(a)}$ et *al.*, 2002). *WWAN* corresponds to digital mobile phone networks, such as Global System for Mobile telecommunications (*GSM*). They have a wide coverage area. Wireless Metropolitan Area Networks (*WMANs*) are emerging for data transmission in municipal areas. They are limited to fixed point-to-point or point to multipoint connections with restricted mobility (Hännikäinen$_{(b)}$ et *al.*, 2002). *WLAN* has been designed to replace and expand legacy *LAN*s. *WLAN* supports a large number of services, for instance to cover broadband wireless Internet access in hot spot areas, as well as short range serial cable replacement (Ala-Laurila et *al.*, 2001). *WLAN* enables a fast network installation and easy topology changes. Therefore, *WLAN* can be established on purely temporary basis, for example conferences, meetings, colleges and universities, and even emergency operations. *WLAN* also makes possible data networking in places without an existing wired *LAN* infrastructure, such as in military operations (IEEE, 1999). *WPAN* technology is aimed at connecting different personal devices, such as a mobile phone, laptop computer, and Personal Digital Assistant (*PDA*). *WPAN* has a smaller working area, smaller data rate, and less number of devices per network compared to *WLAN*. Ubiquitous technologies have the smallest coverage area and they are proposed for various control and automation applications.

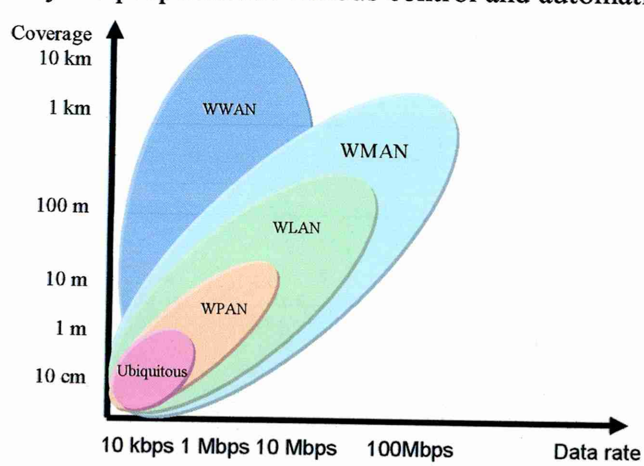



**Figure 2.1:** Wireless communication technologies, coverage area and data rate.

## 2.3 Electromagnetic Spectrum and Radio Environment for WLAN

The electromagnetic wave is an energy radiated in the form of a wave as a result of the motion of electric charges (Nasa, 2006). The various types of electromagnetic radiation differ only in wavelength and frequency; they are similar in all other respects. The Ultra High Frequency (*UHF*) waves are produced by short antennas such as those used by wireless devices. These waves travel at the speed of light approximately $3 \times 10^8$ metre/second. Most wireless devices are located within the *UHF* band except the infrared rays which occupy that part of the *EM* spectrum with a frequency less than that of visible light and greater than that of most radio waves.

The characteristics of the wireless medium make wireless networks fundamentally different from wired networks. The wireless networks use electromagnetic waves to propagate in free space between the transmitter and the receiver instead of propagating inside any material medium such as cables in the wired networks. This means that the wireless channel is unprotected from the outside signals which make it less reliable, less deterministic, and more erroneous than the wired networks. The propagation of the electromagnetic waves in the vacuum is not concentrated in one direction but is propagated in different directions. Therefore, the signals may travel straight forward to the recipient and may be bent by the effect of obstacles and then received with the other signals. This phenomenon is called the multi-path effect in which the receiver receives multiple copies of the same signal with a delay of time between them. The arrival of the transmitted signals is proportional to the length of the path the signal follows. Different paths for the transmitted symbol may lead to an overlap with itself or with other symbols depending on the delay between the symbols and the symbol period causing an Inter-Symbol Interference (*ISI*). Another major problem that affects the propagation of the electromagnetic waves is the attenuation of the transmitted signals. Attenuation is a reduction of the signal strength during transmission. The reduction of the signal strength to low values leads to bit errors at the receiving IEEE 802.11 radio. The occurrence of bit errors forces the transmitter to retransmit the frame resulting in system degradation. The amount of attenuation increases as either the frequency or the distance between the pair of communication entities increases.

## 2.4 Development of WLANs and Standardisation

The development of *WLAN*s has assumed a great importance in communication field. Wireless *LAN* networks are superior to wired networks with regards to ease of installation and flexibility, though they do suffer from lower bandwidth and variable delay due to variability of the wireless medium in time and space and due to fading and multi-path effects. In the development of *WLAN*s, two competing standards have emerged: The European Telecommunications Standards Institute (*ETSI*) that comes with the High Performance European Radio *LAN* (*HIPERLAN*) and the Institute of Electrical and Electronics Engineers (*IEEE*) that presents the IEEE 802.11 *WLAN*. *HIPERLAN* which operates in 5GHz frequency band originally provided a higher data rate than the IEEE 802.11 standard (ETSI, 2000). Other parts of the standard, IEEE 802.11a, and IEEE 802.11g provide an equivalent data rate (IEEE, 1999).

The development of IEEE 802.11 *WLAN*s started in 1988 as IEEE 802.4L (part of the IEEE 802.4 Token Bus Wired *LAN* standard). In 1990 it changed its name to IEEE 802.11 to form *WLAN* standard and became part of the IEEE 802 *LAN* Standards Organisation. In 1997 the IEEE 802.11 standard was presented and defined the Physical (*PHY*) and medium access control (MAC) layers to support data rate of 2 Mbps in the 2.4 GHz Industry, Science and Medical (*ISM*) license free frequency band. A new version of the standard has been presented to produce IEEE 802.11b, which increases the data rate from 2 Mbps up to 11Mbps, IEEE 802.11g, which provides up to 54Mbps, and the IEEE 802.11a version which provides a high data rate up to 54 Mbps. A summary of the standard technologies of the *WLAN*, their frequency band and data rates is listed in Table 2.1 (Philip, 2005).

**Table 2.1:** Summary of standard technologies of the *WLAN*.

| Standard | Data rate | Frequency | Transmission scheme | Details |
|---|---|---|---|---|
| **IEEE 802.11** | 2 Mbps | 2.4 GHz | DSSS, FHSS | IEEE specification extended into 802.11b |
| **IEEE 802.11b** | 11 Mbps | 2.4 GHz | DSSS | Average actual throughput 5 Mbps. |
| **IEEE 802.11a** | 54 Mbps | 5.0 GHz | OFDM | Less potential for RF interference at 5 GHz than 2.4 GHz. Shorter range than, and incompatible with 802.11b. Average actual throughput 10-25 Mbps. |
| **IEEE 802.11g** | 54 Mbps | 2.4 GHz | DSSS, OFDM | Compatible with 802.11b. Uses additional OFDM modulation technique above 20 Mbps. Average actual throughput 10-25 Mbps. |
| **IEEE 802.11n** | 200+ Mbps | 2.4/5 GHz | MIMO | Incomplete yet, expected by the end of 2006, backward compatible with 802.11a/b/g. It is to achieve that by adding MIMO (multiple-input, multiple-output), using multiple antennas. Average throughput of 100+ Mbps. |
| **Bluetooth** | 1 Mbps | 2.45 GHz | FHSS | Best suited to connect PDAs, mobile phones and short range wireless devices. |
| **HiperLAN/1** | 24 Mbps | 5 GHz | *CSMA/CA* | Only used in Europe. Support for real time services. |
| **HiperLAN/2** | 54 Mbps | 5 GHz | OFDM | Only used in Europe. It can carry ATM cells, IP packets, and digital voice (cellular phones). |

## 2.5 IEEE 802.11 WLAN Standards

The IEEE 802.11 is the first standard for *WLAN* that uses the RF radiation with 1 and 2 Mbps data rates and it is currently the most widely used *WLAN* (IEEE Group, 2006). In 1999, the IEEE approved the extension of the previous standard. The new IEEE 802.11b extension defines a standard for products of wireless networks working at 11 Mbps. The need for wireless access to local networks grows with the number of mobile devices such as notebooks and *PDA*s, as well as the desire of users to be connected to a network with higher data rate. As a result, other extensions to the standard have been approved such as IEEE 802.11g and IEEE 802.11a. A number of others are also reported. A brief description of these protocols is as follows:

***IEEE 802.11b***: It operates at 2.4 GHz of the *ISM* band. The main improvements of this version were the standardisation of the *PHY* layer to support higher speeds up to 5.5 and 11 Mbps (3Com, 1998). In this way, IEEE 802.11b is designed to be backwards compatible and can interoperate at 1 and 2 Mbps with the original standard in *DSSS* technique. The IEEE 802.11b provides 11 Mbps over distances up to 300 - 400 metres in an outdoor environment and 30 - 50 metres in an indoor environment with low noise (Zahariadis, 2004). The IEEE 802.11b with noisy environments uses dynamic rate degradation in which the protocol degrades transmission to lower speeds declining to 5.5, 2 and 1 Mbps, and then returning back to its highest speed automatically when there is no interference (Heegard et *al.*, 2001).

***IEEE 802.11a***: This is similar to IEEE 802.11b, but offers higher data rate up to 54 Mbps in a distance up to 50 metres. IEEE 802.11a operates at 5 GHz spectrum range of the *ISM* band. This high frequency range provides the IEEE 802.11a with less interference than the 2.4 GHz spectrum (IEEEa, 1999). The *PHY* layer of IEEE 802.11a is based on *OFDM* technology which provides an efficient communication in time varying environments, where the transmitted signals are reflected from many points, resulting in different propagation times before they ultimately arriving at the receiver.

***IEEE 802.11g***: It is an extension of the IEEE 802.11b. It extends the *PHY* layer of IEEE 802.11b protocol to achieve high data rate up to 54 Mbps. IEEE 802.11g operates at 2.4 GHz band and it uses *DSSS* technology as in IEEE 802.11b to support up to 11 Mbps. Moreover, it uses *OFDM* technology as the case in IEEE 802.11a to support higher data rate up to 54 Mbps. However, IEEE 802.11g is backwards compatible with

IEEE 802.11b products (IEEEg, 2002). These characteristics give IEEE 802.11g advantageous over IEEE 802.11a protocol.

***HiperLAN2***: The IEEE 802.11, IEEE 802.11b, IEEE 802.11g, and IEEE 802.11a standards do not support any QoS mechanism (IEEE, 1999). Consequently, Europe proposed the High Performance Radio Local Area Network Type 2 (HiperLAN2) standard, as it guarantees QoS (ETSI, 2000). HiperLan2 is designed to support high speed access to a variety of networks including 3G mobile networks, ATM networks, and *IP* based networks (Yousef and Strange, 1998). It operates in the 5 GHz band and provides up to 54 Mbps data rate.

***IEEE 802.11e***: This was approved at the end of 2005 as a standard that defines a set of Quality of Service enhancements for the IEEE 802.11 standard by improving the efficiency of the MAC protocol and providing service differentiation between different types of data traffic (Networkworld, 2005) and (IEEE, 2004). The IEEE 802.11e specifications allow packets to gain priority by defining four traffic classes, each with its own queue. Differentiation enables enhanced multimedia capabilities by giving higher priority for time-sensitive data packets such as video and audio over general data packets whose delivery time is less significant such as e-mail, *HTTP*, and File Transfer Protocol (*FTP*). In the IEEE 802.11e, a new access method called Hybrid Coordination Function (*HCF*) is introduced. The main feature of *HCF* is the definition of four Access Categories (*AC*) (i.e. four queues) and eight Traffic Streams (*TS*) at MAC layer. Therefore, the *HCF* is a queue-based service differentiation scheme that incorporates both DCF and PCF enhancements. Enhanced DCF (*EDCF*) or Enhance Distributed Channel Access (*EDCA*) is the contention based *HCF* channel access. *EDCF* supports different classes, by allocating different *CW* sizes (i.e. different $CW_{min}$ and $CW_{max}$) and different *DIFS* values called Arbitration Inter Frame Space (*AIFS*) for each class. Smaller values of *CW* and *AIFS* correspond to a higher priority and eight priorities are supported. Each station can have many flows, which may belong to different classes.

***IEEE 802.11n***: It is the next generation extension of the physical layer. It is expected that IEEE 802.11n will support throughput (useful data rates) of over 100 Mbps. The standard is still in its early development stage. Among the proposed approaches to provide such high data rates are: smart antenna technology, enhanced modulation, and increased channel bandwidth (using both 2.4 and 5GHz bands). It is projected that IEEE 802.11n will also offer a better operating distance than current networks (IEEE, 2006).

## 2.6 IEEE 802.11 Network Architecture

The Basic Service Set (*BSS*) is the fundamental building block of the wireless network architecture. A *BSS* is defined as a group of stations that are under the direct control of a single coordination function (distributed scheme or polling scheme). The transmission medium of a wireless network is a shared radio channel. However, according to the control scheme used, two main wireless topologies can be defined: distributed wireless topology (ad-hoc) and infrastructure based topology. These are discussed below.

### 2.6.1 Distributed Wireless Networks

Distributed wireless networks are wireless stations that communicate with each other without the presence of an infrastructure in the location. In the ad-hoc topology or the Independent Basic Service Set (*IBSS*), all terminals access the medium independently. Therefore, control and management of an ad-hoc topology are distributed among the contending stations (Hännikäinen et *al.*, 2002). Data transfer takes place directly between terminals on a peer-to-peer mode. Access to outside network resources can be available through an intermediary station. An example of a typical ad-hoc wireless network is depicted in Figure 2.2a. An ad-hoc wireless network consists of a group of fixed or mobile stations, where stations operate as sources of data packets, destinations or routers between a source and a destination.

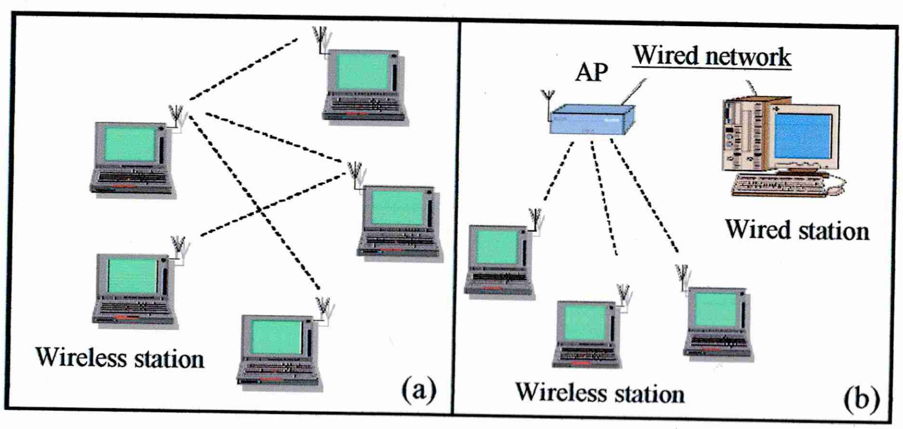



**Figure 2.2:** Wireless networks, (a) Ad-hoc network, and (b) Infrastructure network.

In a distributed wireless network, each wireless node has a wireless interface which could be an RF interface or infrared interface. This interface allows the node to communicate and exchange packets between each other in a distributed way. This type of network does not have a central controller. This means that the wireless stations act independently from each other. Another issue with wireless ad-hoc networks is that all data transmission and reception must be in the same frequency band, because there are no specific nodes that interpret the transmission between different frequency bands (Hsieh and Sivakumar, 2001).

### 2.6.2 Centralised Wireless Networks

Centralised wireless networks, or infrastructure networks are extensions to wired networks as presented in Figure 2.2b. This type of network is different from a distributed network since it has a Base Station (*BS*) or Access Point (*AP*) that operates as an interface between the wired and wireless networks. This administration unit (*BS*) gives a high degree of flexibility in the design of MAC protocols (Gummalla and Limb, 2000). The downlink transmissions (the link from the base station to the wireless nodes) are broadcast and can be heard by all wireless devices on the network. The uplink (the link from wireless node to the base station) is shared between all nodes which provide a multiple access channel. The *BS* in infrastructure networks can administer the uplink transmission by giving access to the channel with respect to the QoS requirements. The infrastructure networks, the down link and the uplink operate in a half duplex mode.

In the infrastructure topology, the Access Point (*AP*) carries out the polling services among wireless stations and the backbone wired networks. Additionally, the management and control of the infrastructure topology are commonly integrated into the *AP*. The *AP* can thus centrally control the transmissions of wireless stations and forward data packets between them (Pahlavan and Levesque, 1994), (LaMaire et *al.*, 1996), and (Pahlavan et *al.*, 1997). The *AP* supports range extension by providing the integration points necessary for network connectivity between multiple Basic Service Set (*BSS*), thus forming an Extended Service Set (*ESS*). The *ESS* consists of multiple BSSs that are integrated together using a common Distribution System (*DS*). The *DS* can be thought of as a backbone network that is responsible for MAC-level transport of MAC Service Data Units (*MSDUs*).

## 2.7 WLAN Technologies

*WLAN* protocols cover the physical (*PHY*) and the data link layers of the Open System Interconnection (*OSI*) as shown in Figure 2.3. The data link layer has been divided by IEEE 802 *LAN/MAN* standards committee into Medium Access Control (MAC) and

Logical Link Control (*LLC*) sub layers (IEEEStd, 2001). This thesis concentrates on the MAC layer as a separate layer. It is the key protocol layer for managing and controlling *WLAN* and therefore the data transfer service. The *LLC* layer, in contrast, is not addressed.

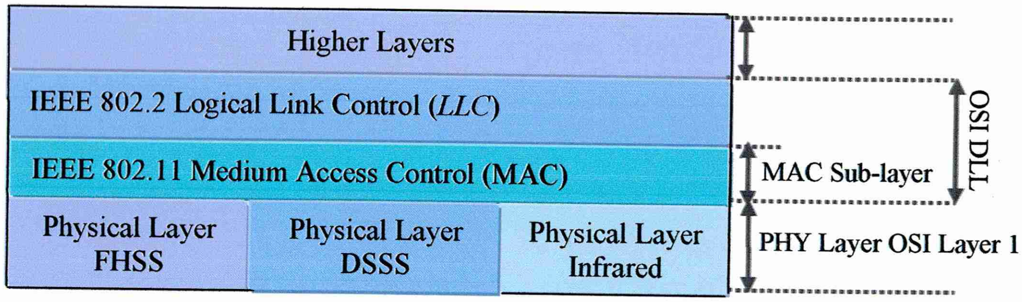



**Figure 2.3:** IEEE 802.11 standards regarding to the *OSI* reference model.

## 2.8 IEEE 802.11 Physical (PHY) layer

The major task of the physical layer is to transmit data bits from the sender to the receiver with minimum bit errors (Lynch, 2006). It is the interface between the MAC sub-layer and the wireless medium. It offers three phases of functionality (Lee et *al.*, 2006): First, it offers a frame exchange between the MAC and *PHY* under the control of the Physical Layer Convergence Procedure (*PLCP*) sub-layer. Secondly, it uses signal carrier and spread spectrum modulation to send data packets over the medium under the control of the Physical Medium Dependent (*PMD*) sub-layer. Finally, it provides a carrier sense indication back to the MAC to check the medium activity (IEEE, 1997).

According to the standard, different thresholds are used to specify whether a sent packet is correctly received (Takai et *al.* 2001) and (Wu, 2004). These are: the receiving threshold (*RXThreshold*), the carrier sensing threshold (*CSThreshold*), and the capture threshold (*CPThreshold*). The *RXThreshold* determines the signal strength of the packet received by the receiver. If the received signal strength is greater than this threshold, the packet is received correctly. Otherwise, the packet is tagged as corrupted and the MAC layer will discard it. The *CSThreshold* determines whether the packet is detected by the receiver. If the received signal strength is greater than this threshold, the packet transmission can be sensed. However, the packet cannot be decoded unless signal strength is greater than *RXThreshold*. The *CPThreshold* refers to the capture phenomenon. If two packets are received simultaneously (i.e. collided) it is still possible to receive the stronger packet if its signal strength is higher than the *CPThreshold* of the other packet.

The original IEEE 802.11 provided data rates of either 1 Mbps or 2 Mbps, through the free licensed frequency *ISM* band applications between 2.4 - 2.4835 GHz. The IEEE 802.11 standard defines the transmission over three different physical layers (PHYs), Infrared (*IR*), Frequency Hopping Spread Spectrum (*FHSS*), and Direct Sequence Spread Spectrum (*DSSS*) (Crow et *al.*, 1997). The following sections provide a brief description of the *FHSS*, *DSSS*, the Infrared, and the *OFDM* transmission techniques.

### 2.8.1 Frequency Hopping Spread Spectrum (FHSS)

This technology operates at 2.4 GHz of the *ISM* frequency band. It is used with different applications due to its robustness against interference. The transmission and reception occur on one frequency for a short period of time then they jump to another frequency. The jumping pattern is usually based on some pseudo-random hopping algorithm. Frequency hopping provides good scalability, where several base stations could be located in the same coverage area and able to have ongoing transmission at the same time. Within the *ISM* band there are 79 frequencies ranging from 2.402 GHz to 2.480 GHz which are available for IEEE 802.11 standard (Crow et *al.*, 1997). Frequency hopping allows only a small bandwidth (maximum bandwidth is about 1 MHz) which, in turn, leads to small data rates only 1 or 2 Mbps.

### 2.8.2 Direct Sequence Spread Spectrum (DSSS)

The *DSSS* technique operates at the same frequency band as the *FHSS* technique. It provides higher data rate than the one offered by the *FHSS* mechanism. IEEE 802.11, IEEE 802.11b and IEEE 802.11g, all operate on this technology. The IEEE 802.11b and the IEEE 802.11g versions support data rate up to 11 Mbps and 54 Mbps, respectively. The operation of this technique is based on breaking the original data bit into multiple "sub-bits" or chips; each chip is represented by a 1 or 0. These chips are transmitted over broader frequency range. A receiver which should be at the same chipping code of the transmitter takes all the received chips and runs them through a decoder to retain the original data stream. The amplitude of the transmitted signal using *DSSS* technique is very small which looks like noise in the radio spectrum. This feature provides a reasonable level of immunity against jamming and interference with other signals. In terms of data transmission rate, the *DSSS* technique offers higher data rate than the *FHSS* technique. While in terms of scalability, the *FHSS* provides higher scalability and wider coverage area than the *DSSS* technique (Crow et *al.*, 1997).

### 2.8.3 Infrared (*IR*)

The IEEE 802.11 standard also includes the IEEE 802.11 Infrared (*IR*) Physical Layer. It is one of the main *PHY* layers supported in IEEE 802.11 protocol (IEEE, 1999). IEEE 802.11 *IR* has different characteristics from the *FHSS* and *DSSS* layers, since it uses near visible light as its transmission medium. Moreover, the *IR* transmission depends on the light energy and it requires line of sight between the pair of communication. The *IR PHY* layer is only used with line of sight applications since its transmitted signal cannot go through the obstacles. In *FHSS* and *DSSS* the signal can penetrate the walls, which give them the ability to be used indoor and outdoor environments.

### 2.8.4 Orthogonal Frequency Division Multiplexing (OFDM)

*OFDM* is a transmission technique being applied in broadband wireless access systems as a way to eliminate wireless transmission problems and to improve the channel data rate. It is used in the IEEE 802.11a and the HiperLAN/2 standards with data rate up to 54 Mbps. This technique operates in the 5 GHz *ISM* frequency band. In *OFDM* technique the data is divided into several interleaved parallel bits stream where each stream modulates a separate sub carrier (i.e. the channel frequency is divided into several independent sub channels). The receiving system reconstructs the message from the separate carriers. One of the main advantages of *OFDM* is the efficient use of the radio spectrum, since all sub channels are packed closed together. On the other hand, the transmission range or the coverage area is very small which is due to the higher frequency band (Cimini, 1985).

## 2.9 IEEE 802.11 Wireless Medium Access Control Protocols

The original IEEE 802.11 standard was developed in 1997 and then amended in 1999. The aim was to develop a specification for *WLAN* which is split into the MAC and the *PHY* layers. The *PHY* layer has been discussed in section 2.8. The MAC layer is described in detail in the following sections. This includes MAC QoS mechanisms and extensive review of the contributions that have been made to the literature in the area of improving the performance of IEEE 802.11 standard and providing QoS differentiation.

### 2.9.1 Medium Access Control Protocols Aspects

The wireless MAC protocols have been studied widely since 1970s. They have different characteristics to their counterparts in the wired networks. The MAC protocol in wireless networks operates in a half-duplex mode. This is because a large part of the power dissipates into the receiver path due to self interference (Nasa, 2006). The dissipation of the power is much higher than the received power, which makes it impossible to detect a received signal while transmitting information. This means collision detection is impossible while transmitting data. For this reason, most of the proposed MAC protocols try to reduce the number of collision over the wireless link by using collision avoidance.

The received power signal varies with time. This is due to the multiple path propagation effects in which the received signal is a superimposition of a time shifted and attenuated copies of the transmitted signal. As long as the received signal is higher than the *RXThreshold*, good link quality is achieved. When the strength of signal is below the *RXThreshold*, the receiver is in fade (Stuber, 2001). To alleviate such a problem, an exchange of small frames between the parties of communications is performed to test the wireless channel. Once these frames are successfully exchanged this gives a good indicator for the wireless link between the pair of communications.

The time variation of the channel and the variation of the signal strength also affect the number of errors in wireless networks. This increases the bit error rate of the wireless link which in turn increases the probability of packet loss. Another factor that increases the packet loss is the long bursts when a station is in fade. This can be minimised by transmitting small packets using Forward Error Correction (*FEC*) codes and using the retransmission process. In most protocols, they return back an immediate positive acknowledgement to the source as the packet is received. If the positive acknowledgement is not received, the source assumes errors have occurred and retransmits the affected packets.

As mentioned earlier, *RXThreshold*, *CSThreshold* and *CPThreshold* define the signal levels above which the signal can still interfere with other transmission and above which level the signal can be sensed by other wireless stations. Only stations within a specific radius of the sender can detect the carrier on the channel. As a result of carrier sensing range which is location dependent, three types of problems can result: hidden terminal problem, exposed terminal problem, and capture effect. Hidden terminal and exposed terminal problems will be discussed later in this chapter. Regarding capture effect, it takes place when a receiver can receive a strong signal from two simultaneous transmissions both within its transmission range. In some cases capture phenomenon can improve the MAC protocol performance, but it may result in unfair sharing of channel bandwidth with preference to the closest station. More details about this phenomenon can be found in (Goodman and Saleh, 1987) and (Goodman et *al.*, 1989).

### 2.9.2 IEEE 802.11 MAC Functions and Access Mechanisms

The IEEE 802.11 standard (IEEE, 1997) defines the MAC layer, a set of protocols for controlling the access to the wireless medium between stations in a competent manner. The IEEE 802.11 standard identifies a *CSMA/CA* protocol. In *CSMA/CA*, when a station has a packet to be transmitted, it first listens to the media to ensure no other on going transmission is currently taking place. If the channel is idle, it then transmits the packet. Otherwise, it selects a random "backoff interval" which determines the period of time the station has to wait until it is allowed to transmit its packet. During the idle period of the channel, the transmitting station decrements its backoff counter. During the busy period the station suspends its backoff counter. During the idle period, when the backoff counter reaches zero, the station transmits the packet. This method minimises the probability that two stations pick the same backoff interval and transmit simultaneously and leads to collision avoidance rather than the collision detection employed by the wired IEEE 802.3 Ethernet protocol.

The IEEE 802.11 standard defines two channel access mechanisms, called coordination functions (IEEE, 1999). These coordination functions determine when a station is permitted to transmit, and when it must be prepared to receive data. The mandatory function is the DCF which adopts the *CSMA/CA* mechanism to provide services for asynchronous data transmission. The optional function is the PCF which incorporates a polling coordinator that is located at the *AP*, and is proposed for use with real time traffic. It is worth noting that the compulsory DCF function is used throughout this study. It was chosen because it provides scalability, simplicity and availability in the market. In contrast, the PCF function is barely implemented in current products due to its complexity and inefficiency for normal data transmission (Ziao and Pan, 2005). Although, the DCF is not designed for time-sensitive applications, it is a robust protocol. It provides a reliable error control mechanism of failed packets through the transmission of the positive acknowledgement after each successful transmission. The following sections describe both the DCF and PCF operation functions in details.

#### 2.9.2.1 Distributed Coordination Function (DCF)

The IEEE 802.11 MAC DCF defines two access methods (IEEE, 1999). The first one is known as the basic access method which is based on a two-way handshake procedure

(*DATA/ACK*). The second one adopts a four-way handshake procedure, where the *DATA/ACK* phase is preceded by control frames called *RTS* and *CTS*. Both access methods use a *CSMA/CA* mechanism for accessing the wireless medium.

Under the basic access method, when a station has a packet to transmit, it first senses the channel status. If the channel is busy, the station backs off its transmission and persistently monitors the channel until it is measured as idle for a period of time called the Distributed Inter Frame Space (*DIFS*). At this point, the station generates a random backoff interval before transmitting in order to minimise the probability of multiple stations concurrently starting transmission. The time after the *DIFS* period is slotted to a number of time slots (backoff time). Each slot time has a duration which is at least equal to the time required for a station to measure an idle channel plus the time required for switching from listening mode to transmitting mode. The backoff counter is decreased by one for each idle slot, suspended if the channel is sensed busy, and then reactivated if the channel is idle again and remains idle for more than a *DIFS* time duration. When the backoff timer reaches 0, the data packet is transmitted. The winning station is only allowed to transmit at the beginning of each time slot. Figures 2.4 and 2.5 illustrate the timeline of the basic access mechanism and the operation of the *CSMA/CA* protocol, respectively.

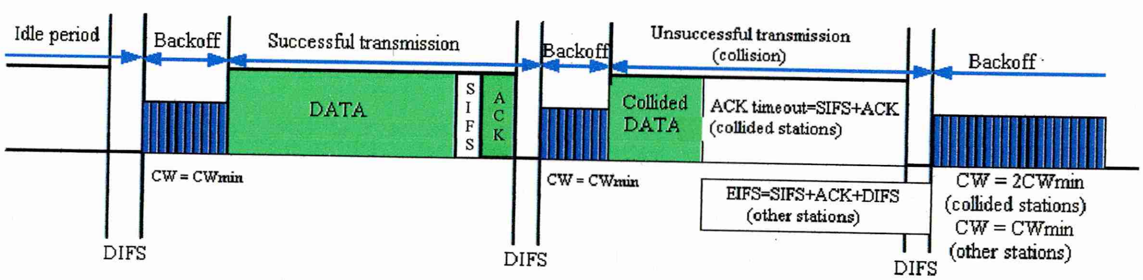


**Figure 2.4:** Timeline of the basic access mechanism in DCF.

The selection of the random number of the backoff timer is based on the binary exponential backoff algorithm. The competing stations choose a random number between 0 and ( $CW-1$ ) with equal probability. The $CW$ is set to the minimum Contention Window size ($CW_{min}$) for every new data frame transmission. If the data packet is successfully transmitted, the backoff counter of the transmitted station resets to $CW_{min}$ and then the station starts to compete with the other stations for accessing the wireless medium. After successfully receiving a data packet, the receiving station replies with a positive acknowledgement (*ACK*) after waiting for a Short Inter Frame Space (*SIFS*) period. If the transmitter still has packets queued for transmission after successful transmission it must execute the backoff process (IEEE, 1999). If the *ACK* frame is not received by the sender within a *SIFS* period and after the completion of the data frame transmission, the transmission is assumed to be unsuccessful, and a retransmission is scheduled according to the specified backoff rules.

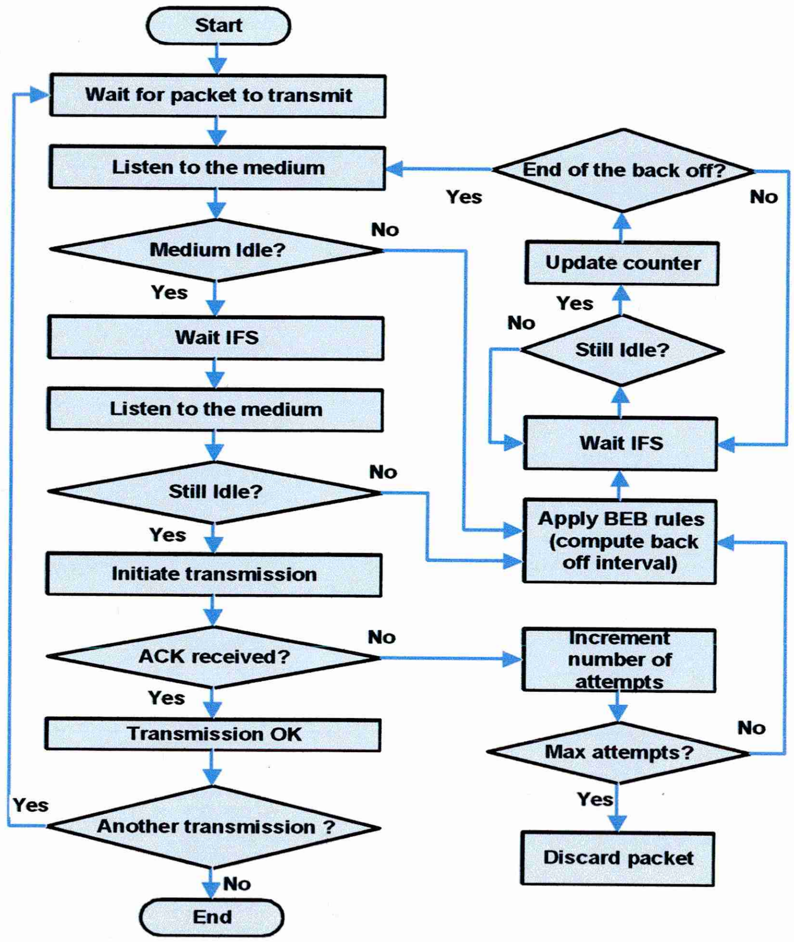



**Figure 2.5:** *CSMA/CA* backoff procedure of the basic access mechanism of DCF.

Unsuccessful transmission could be due to a collision over the wireless link. If a collision has occurred, the *CW* is doubled according to ( $2^m CW_{\min} - 1$ ) where $m$ represents the number of retransmission attempt. The *CW* is doubled until it reaches the maximum *CW* size ($CW_{max}$) and remains at its maximum value ($CW_{max}$) until the number of retransmission attempts is exceeded as shown in Figure 2.6. Every station maintains a Station Short Retry Count (*SSRC*) as well as a Station Long Retry Count (*SLRC*) and both have an initial value of 0. The *SSRC* indicates the maximum number of transmission attempts before a data packet is discarded. This SSRC is applicable to the data packet where the exchange of *RTS/CTS* (discussed later) is not required. The *SLRC* indicates the maximum number of transmission attempts before a data packet is discarded. This retry limit is relevant to the data packet that requires exchange of *RTS/CTS* control frames. The minimum value of $CW_{min}$ is set to 32, the $CW_{max}$ is set to

1024 and the number of attempts is set to 7 and 4 for the *SSRC* and *SLRC*, respectively, as defined by the standard (IEEE, 1999). In order to avoid channel capture by the winning station, a station has to wait a random backoff time between two consecutive packet transmissions. This ensures that all stations are capable of accessing the wireless medium in a fair manner.

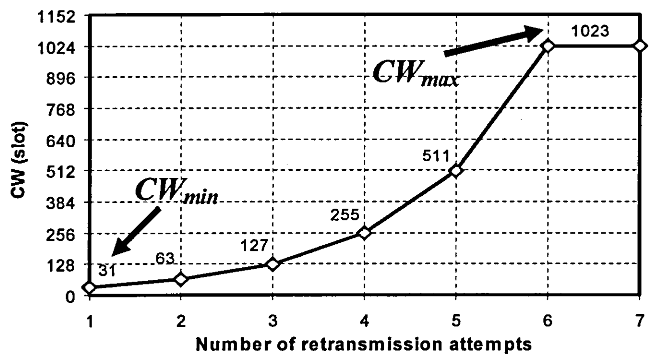



**Figure 2.6:** *CW* selection procedure of the IEEE 802.11.

In IEEE 802.11, DCF also identifies another mechanism which is an optional access method for transmitting data packets. The optional mechanism is involved in transmission of special short control messages called *RTS* and *CTS* frames. These short messages are always transmitted prior to the transmission of the data packet.

The *RTS/CTS* access mechanism is mainly used to minimise the amount of time wasted when collisions occur. Collision occurs in *RTS* and *CTS* control messages instead of data packets. The *RTS/CTS* access mechanism is also used to combat the hidden station problem. Before commencing the transmission of a data packet, the source station sends a short control frame, called *RTS*, declaring the duration of the forthcoming transmission. When the destination station receives the *RTS* frame, it replies with a *CTS* frame after a *SIFS* interval, with the duration of the future transmission. Upon hearing *RTS* and *CTS*, all other stations in the vicinity of the sender and the receiver update their Network Allocation Vectors (*NAV*), *NAV* is a counter residing at each station that represents the time in milliseconds and indicating the length of the current transmission burst, with the information about the duration for which the channel is going to be busy. Therefore, the *NAV* is essentially a channel reservation vector. As a condition for accessing the channel, the MAC Layer examines the value of its *NAV*. Before a station can attempt to send a packet, the *NAV* must be zero. Preceding a packet transmission, a station computes the amount of time necessary to send the packet based on the packet's length and data rate. The station places a value representing this time in the duration field in the header of the packet. When stations receive the whole packet, they check this duration field value and use it as the basis for setting their corresponding *NAVs*.

This process reserves the medium for the sending station. Thus, all stations in the neighbourhood of the sender and receiver defer their transmissions and receptions to avoid collisions. After the successful *RTS/CTS* exchange, the source station transmits the data packet. The receiver responds with an *ACK* packet to acknowledge a successful reception of the data packet. Figure 2.7 depicts the timeline of the *RTS/CTS* access mechanism and Figure 2.8 illustrates how other stations in the vicinity of the pair of communication update their *NAVs* with regard to the exchange of the *RTS/CTS* messages and the transmission of data packets. The use of the *RTS/CTS* access mechanism is determined by the value of *RTS_Threshold*. If the data packet has a size greater than the *RTS_Threshold* then *RTS/CTS* access mechanism is used; otherwise, the basic access mechanism is employed. The value of *RTS_Threshold* should be correctly chosen in order to get the advantages of the *RTS/CTS* access mechanism.

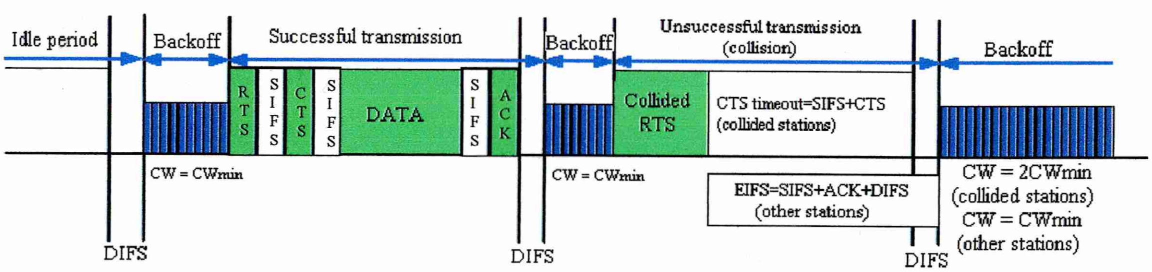



**Figure 2.7:** Timeline of *RTS/CTS* access mechanism in *DCF*.

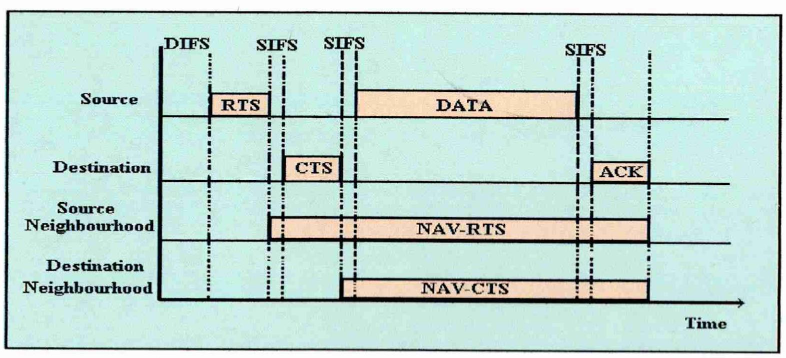



**Figure 2.8:** IEEE 802.11 DCF scheme with *RTS/CTS* access mechanism.

Fragmentation of a large packet is one of the main operations of the IEEE 802.11 MAC protocol. The MAC protocol can fragment packets in an attempt to increase the probability of sending them without errors caused by the interference. The number of fragments to be transmitted is computed based on the *MSDU* size and the fragment threshold. The destination station receives these fragments and stores them in the reassembly buffer until all fragments have been received. Using fragmentation for a large packet size improves the reliability of data exchange between the pair of communication, since each fragment requires an acknowledgement.

The transmission of data packets using the *RTS/CTS* access mechanism also follows the same rules of the Binary Exponential Backoff (*BEB*) that is used with the basic access mechanism as discussed in earlier part of this section. All the exchanged frames between the sender and the receiver (*RTS*, *CTS*, *DATA*, and *ACK*) are separated by a *SIFS* as depicted in Figures 2.7 and 2.8. The Inter Frame Space (*IFS*) time intervals between the transmissions of control frames or data packets are used to provide priority and control access to the channel. These intervals can be defined as specified in the IEEE 802.11 standard as follows (IEEE, 1997): (i) Short Inter Frame Space (*SIFS*) which is specified for control frames such as acknowledgment (*ACK*) frame, (ii) Point Inter Frame Space (*PIFS*) which is used with the PCF function, (iii) Distributed Inter Frame Space (*DIFS*) which is used with data packets in the DCF function, and (iv) Extended Inter Frame Space (*EIFS*). *SIFS* is less than *PIFS* and *DIFS*, since control messages must have higher priority in order to initiate the handshake communication between connections. *PIFS* is less than *DIFS*, which is allocated to time-sensitive applications; whereas *DIFS* is assigned for time-insensitive applications. The *EIFS*, unlike *SIFS*, *PIFS* and *DIFS*, has a variable value and is only used when there has been an error in frame transmission.

#### 2.9.2.2 Point Coordination Function (PCF)

The IEEE 802.11 standard defines the Point Coordination Function (PCF) to allow stations priority access to the wireless medium (support time-sensitive services). Unlike the DCF mode, the PCF method requires the presence of a station called Point Coordinator (*PC*) to coordinate the access to the medium. The PCF has higher priority than the DCF, because it starts transmissions after a *PIFS* duration which is shorter than *DIFS* and longer than *SIFS*. Time is divided into repeated periods, called super frame. With PCF, a Contention Free Period (*CFP*) and a Contention Period (*CP*) alternate over time, in which a *CFP* and the following *CP* form a super frame as depicted in Figure 2.9 (Crow et *al.*, 1997). During the *CFP*, the PCF is used for accessing the medium, while the DCF is employed during the *CP*. It is compulsory that a super frame includes a *CP* of a minimum length that allows at least one MAC Service Data Unit (*MSDU*) delivery under DCF. A super frame begins with a beacon frame, regardless if PCF is active or not. The beacon frame is a management frame that maintains the synchronisation of the local timers in the stations and delivers protocol related parameters. The *PC*, which is typically co-located with the *AP*, generates beacon frames at regular beacon frame intervals, thus every station knows when the next beacon frame will arrive; this time is called Target Beacon Transition Time (*TBTT*) and is declared in every beacon frame. The beacon frame is required in pure DCF even if there is only contending traffic. The *PC* at any point during the *CFP* period can suspend the PCF mode and return back to DCF mode by sending Contention Free End *CF-End* message.

As in the DCF, all stations in the same *BSS* update their *NAV* to the maximum value of the *CFP* period at the beginning of each *CFP* repetition interval. Moreover, all stations are inhibited from transmission during the *CFP* unless they responded to a poll frame from the *PC* or transmission of an *ACK* for the previous reception. Hence after, the *PC* senses the channel, if the channel is idle and remains idle for a *PIFS* interval; the *PC* commences transmission by sending a beacon frame, after that the *PC* waits for a *SIFS* interval, then it transmits a *CF-poll*, data, or *CF-poll + data* frame as shown in Figure 2.9 (Crow et *al.*, 1997). When the network is lightly loaded, the *PC* transmits a *CF-End* frame to terminate the *CFP* if there is no more data is buffered. If the wireless station receives a *CF-poll* from the *PC*, the station can reply to the *PC* request *CF-ACK* or *data + CF-ACK* frame after a *SIFS* period. If the station responded with *data + CF-ACK* and received successfully by the *PC*, the *PC* can transmit a *data + CF-ACK + CF-poll* frame to another station where the *CF-ACK* part of the frame is used to confirm receipt of the previous data packet. If the *PC* transmits a *CF-poll* frame and the destination station does not have data packet to transmit it responds with a null function to the *PC* (Crow et *al.*, 1997). If the *PC* is unable to receive an *ACK* for a transmitted packet, it waits a *PIFS* period and then initiates transmission to the next station in the polling list.

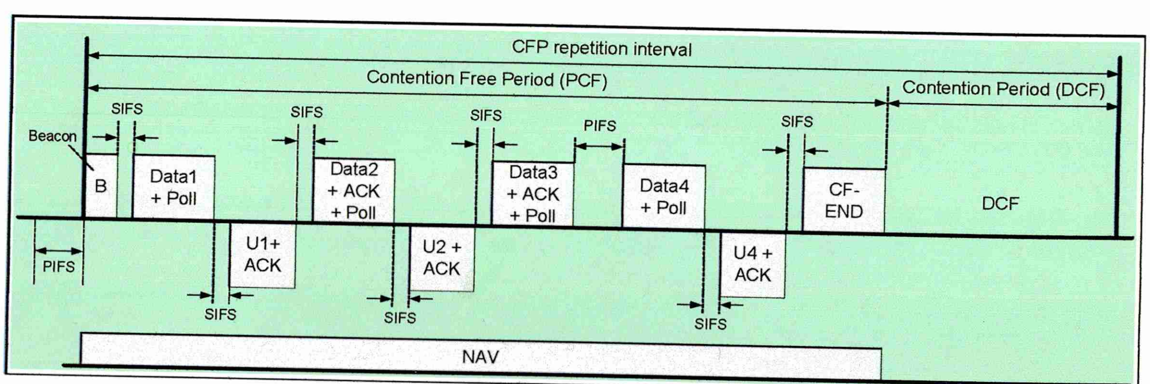



**Figure 2.9:** The *CFP* repetition interval with PCF and DCF cycle.

## 2.10 IEEE 802.11 MAC Protocol Limitations

Although DCF is a simple and effective mechanism, there are a number of limitations inherent in both the DCF and the PCF schemes. They do not support QoS, nor guarantee to meet the multimedia applications requirements. The DCF scheme only supports best-effort service and it does not provide any guarantee of QoS. There are some other problems experienced with the performance of the IEEE 802.11 DCF protocol related to the lack of bandwidth (i.e. limited resource availability), mobility (i.e. dynamic varying network topology), error prone shared medium (i.e. vulnerability to the interference), hidden and exposed terminal problems. Subsequently, these limitations lead to unfairness problems among the contending stations that cause significant performance degradations particularly for the starved stations.

### 2.10.1 Distributed Coordination Function Limitations

The shared medium imposes critical challenges to the protocol operation of the IEEE 802.11 MAC protocol. The hidden terminals may cause collisions and the exposed terminals may result in throughput degradation. Additionally, the unfairness problem is a significant factor that deserves consideration. Actually, the impact of unfairness problem becomes more significant in multi-hop networks and may lead to throughput and QoS degradation and starvation of some stations. It has been shown in several studies that multi-hop ad-hoc networks perform poorly with *TCP* traffic and heavy *UDP* traffic (Broch et *al.*, 1998), (Fu et *al.*, 2003), (Li et *al.*, 2001), and (Xu and Saadawi, 2001). This is because all the wireless links in the vicinity share the same wireless resources. All the traffic flows crossing these links need to contend for the channel before transmission. Hence, severe MAC contention and collision can result in contention among transmitting stations and can lead to network performance degradation. The hidden terminal, exposed terminal, and unfairness and multi-hop problems are briefly discussed below.

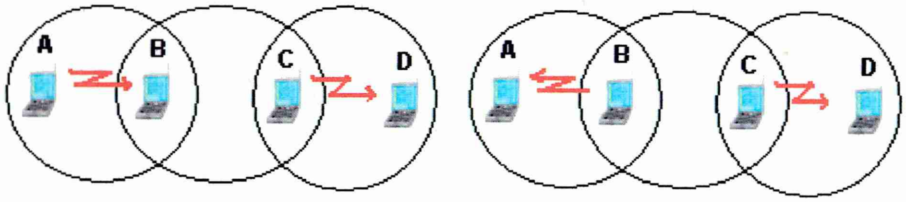
#### 2.10.1.1 Unfairness Problem

Unfairness can result from different prospects of channel access. There are two main causes of unfair channel access: (i) the backoff mechanism and (ii) the capture effect phenomenon. The IEEE 802.11 MAC protocol uses the *BEB* algorithm to manage the competition between stations to access the medium. In the *BEB* algorithm after a successful transmission each station decreases the backoff interval to minimum value ($BI_{min}$). After a collision or unsuccessful transmission each station doubles its backoff interval up to the maximum backoff interval ($BI_{max}$) by duobling the value of its Contention Window minimum ($CW_{min}$). Consequently, the decrease of the *BEB* to the minimum value or the increase of the *BEB* to the maximum value determines the priority of the station for accessing the channel. In some scenarios, this algorithm is poorly performed (Xu and Saadawi, 2002), (Haas and Deng, 2003). It always prefers the station that just successfully seized the channel. This can be explained by considering a simple network with two stations competing with each other. Each station has enough data traffic to flood the medium. When one of these stations for example, station 1, successfully transmits its data packet, it decreases its backoff interval to the minimum value. Given that, station 2 does not successfully transmit its data packet, its backoff interval increases. Henceforth, it has to compete with station 1 with a larger backoff interval. As a result of that, station 1 will frequently gain access to the medium, while station 2 will frequently double its backoff interval until it reaches the maximum value. Therefore, station 1 takes over the channel, while station 2 experiences starvation. Subsequently, different stations may use different *CW* sizes, leading to different transmission probabilities and hence short-term unfairness as well as long-term unfairness. Meanwhile, the capture effect phenomenon might also lead to unfairness access to the medium due to its effect on the MAC operation. It refers to the mechanism where a receiver can receive one of two simultaneously arriving packets if being allowed by their received power (Lau and Leung, 1992). The destination can receive the packet from the sending station within the highest transmitting power. A transmission power control can be used to minimise the effect of power capture (Pahlavan and Levesque, 1994), (Tannenbaum, 1996), (Ahmadi et *al.*, 1996), and (Bing, 2000).

The unfairness problem is an important issue in the MAC protocol. It is possible that it can be solved by enhancing the MAC protocol in a correct manner to achieve the required fairness. Several enhancements have been proposed to the MAC protocol such as Distributed Fair Scheduling (*DSF*) (Vaidya et *al.*, 2000) and *MACAW* protocol (Bharghavan et *al.*, 1994).

#### 2.10.1.2 Hidden and Exposed Terminals

A hidden terminal occurs when there are at least three basic service sets in a wireless network. Collisions can occur at stations which are located in the common boundary of two basic sets. Figure 2.10a depicts a typical hidden terminal situation where there is an ongoing communication between the stations *A* and *B*. If the station *C* does not have any information about the ongoing communication between the stations $A$ and $B$, it can commence transmitting to station $D$ causing a collision in station $B$.




(a) Hidden terminal (b) Exposed terminal

**Figure 2.10:** An example of hidden and terminal problems, (a) hidden terminal and (b) exposed terminal.

In Figure 2.10b the exposed terminal problem is depicted. An exposed terminal is one that is located within the sensing range of the sender (i.e. it senses the transmission of a transmitter) but not within that of the receiver (i.e. cannot interfere with the reception at the receiver). If we assumed that station $B$ is sending to station $A$, then, when station $C$ is ready to transmit for example to station $D$, it senses station $B$'s carrier and therefore defers transmission. However, there is no reason for station $C$ to defer its transmission to station $D$ or other stations rather than station $B$ since station $A$ is out of station $C$'s range. Station $C$'s carrier sense did not provide the necessary information because it is exposed to station $B$ even though it will not collide or interfere with station $B$ transmission.

In general, the hidden terminal problem reduces the capacity of a network due to the increased number of collisions, while the exposed terminal problem reduces the network capacity due to unnecessarily deferring stations from transmitting. In order to minimise the impact of the hidden terminal problem, the IEEE 802.11 standard has introduced the *RTS/CTS* access mechanism. This allows for large packets to be transmitted with fewer collisions.

Several studies have been carried out to investigate the hidden terminal problem. In (Xu and Saadawi, 2001) experiments have been carried out to point out the consequences of this problem in multi-hop wireless ad-hoc networks performance. In (Fullmer and Garcia, 1997) a solution to the hidden terminal problem was proposed. This is called Floor Acquisition Multiple Access (*FAMA*) protocol. The objective of this protocol was for a station that had data to send to acquire control of the channel before sending any data packet, and to ensure that no data packet collided with any other packets. Due to the mobility feature of wireless networks, stations may move out of the transmission range of other stations at their vicinities, causing a pair of communication to be hidden from each other. A hidden terminal can be caused by fixed obstacles, which prevent signals of one station from getting the other. In (Khurana et *al.*, 1998) and (Kleinrock$_{(a)}$ and Tobagi, 1975) a framework was proposed for modelling hidden terminals which could handle both mobility and static obstructions. The study revealed that the hidden terminal problem could have a very negative effect on the performance of the IEEE 802.11 MAC protocol. Additionally, the study showed that the delay increased significantly in the presence of hidden terminals; using *RTS* helps alleviate the effect of hidden terminal.

#### 2.10.1.3 IEEE 802.11 MAC Protocol in Multi-hop Ad-hoc Wireless Networks

In wireless ad-hoc networks all links between stations are wireless and there is no need for a control or administration unit. A multi-hop wireless network is an example of an ad-hoc network where the intermediary stations act as routers between the source and the destination. The MAC protocol in wireless networks is a shared and limited facility, thus the control of accessing the wireless media is a sophisticated task. A number of studies have investigated into the behaviour of the MAC protocol in wireless multi-hop networks (Broch et *al.*, 1998), (Fu et *al.*, 2003), (Li et *al.*, 2001), (Zhai et *al.*, 2004), (Kanodia et *al.*, 2001), and (Zhai et *al.*, 2006). Two examples as follow: In (Xu and Saadawi, 2001), the performance of the IEEE 802.11 MAC protocol over multi-hop wireless networks was investigated. They concluded that the IEEE 802.11 was inappropriate for multi-hop wireless networks. Their work only evaluated the performance of IEEE 802.11 with *TCP* transport protocol and did not consider the application type. Hung and Sivakumar (2002) argued that certain improvements in terms of reducing the MAC protocol band of contention and fairness model were necessary in order to realise efficient and fair medium access. Through simulation results they concluded that the IEEE 802.11 MAC protocol did not perform well in multi-hop wireless environments.

### 2.10.2 PCF Limitations

The PCF scheme is based on the polling mechanism, an insufficient and complex scheme that deteriorates the performance of the high priority traffic at heavy loaded network (IEEE, 1999), (Zhu et *al.*, 2004), (Lindgren et *al.*, 2001), and (Mangold et *al.*, 2002). It experiences substantial delay at low load; stations have to wait for polling even the channel is idle. The incompatibility between the Contention Free Period (*CFP*) and Contention Period (*CP*) leads to unpredictable beacon delay. Moreover, the transmission time of the polled station is unknown which may prevent the Point

Coordinator (*PC*) to providing guaranteed QoS to other stations that are polled during the remaining *CFP*. The alternating *CFP* and *CP* introduce significant overheads particularly if the superframe size becomes short. Additionally, PCF scheme limits the scalability of the network since it is designed for infrastructure networks. In addition, PCF is a centralised approach that suffers from location-dependent errors (Zhu et *al.*, 2004). Thus, PCF has not drawn noticeable attention from the research community.

## 2.11 MAC Protocol Transmission Parameters Adjustments

IEEE 802.11 MAC performance analysis has been a very active research field in recent years. The performance of many characteristics of IEEE 802.11 MAC protocol has been explored in detail. However, there are still many issues in this area that need further investigations, such as how to improve the performance of the medium access control mechanisms for transmission of various applications and QoS provisioning for time-sensitive applications. As the main focus of this study is on how to offer good performance to time sensitive applications such as audio and video, it is essential to review the previous efforts that have been made to the knowledge in the area of performance of IEEE 802.11 MAC protocol.

### 2.11.1 Development of IEEE 802.11 Access Schemes

One of the most primitive MAC protocols was *ALOHA* (Roberts, 1975) and (Tanenbaum, 2003). In pure *ALOHA*, stations send their packets without regard to the current condition of the medium. A successful transmission is determined upon receiving a positive acknowledgment. A collision is assumed if no *ACK* returns. A performance analysis confirmed that pure *ALOHA* utilises at most 18% of the channel capacity. Accordingly, a simple extension to pure *ALOHA* named slotted *ALOHA* was proposed (Roberts, 1975). Slotted *ALOHA* was able to achieve a maximum channel utilisation of 36% compared to pure *ALOHA*.

To further improve the channel utilisation, Carrier Sense Multiple Access (*CSMA*) protocols with two categories known as non-persistent *CSMA* and p-persistent *CSMA* were developed (Kleinrock and Tobagi, 1975). In non-persistent *CSMA*, if the medium is idle, the station transmits instantly. However, if the medium is busy, it waits a random amount of time and then senses the channel again. In p-persistent *CSMA*, the node constantly senses the medium until it is idle. It then transmits at a given slot with probability ($p$) and defers transmission to the next slot with probability ($1-p$). In the case of a collision, the node waits a random delay before transmitting. If the appropriate *p* value is chosen, a channel utilization of almost 100% is possible when *p*-persistent *CSMA* is in use. However, the low values of *p* required to obtain high values of throughput for a particular node also results in high values of delay for other waiting nodes. This does not provide acceptable levels of QoS. Kleinrock and Tobagi (1975) showed that *CSMA* protocols outperformed both pure and slotted *ALOHA* in terms of delay and throughput. Given this, most of the subsequent protocols have been based on CSMA protocols. Another important work in the area was the development of the Multiple Access with Collision Avoidance (*MACA*) scheme (Karn, 1990). This employed the exchange of *RTS/CTS* and Binary Exponential Backoff (*BEB*) algorithm prior to data transmission. Therefore, *RTS/CTS* functionality was used to both avoid collisions on the medium, and to alleviate the hidden node problem.

The detailed simulation study by Bharghavan et *al.* (1994) showed that the binary backoff algorithm used in *MACA* and IEEE 802.11 caused suboptimal performance for several reasons as discussed in (Kwon et *al.*, 2003), (Bharghavan et *al.*, 1994), and ($\text{Cali}_{(b)}$ et *al.*, 2000). The *BEB* exposes large variations in the *CW* size, since after unsuccessful transmission the *CW* size is doubled and reset to $CW_{min}$ after successful transmission. This behaviour leads to an undesirable increase in the risk of collision. This motivated Bharghavan et *al* (1994) to modify the *MACA* protocol and to propose the Multiple Access with Collision Avoidance for Wireless (*MACAW*). *MACAW* scheme used a new backoff strategy to reduce the oscillation in the *CW* size called Multiplicative Increase Linear Decrease (*MILD*) scheme. In the case of collision, the collided stations increase their *CW* multiplicatively, while in successful transmission the stations decrease their *CW* linearly instead of resetting to $CW_{min}$. Specifically, the Multiple Increase Linear Decrease (*MILD*) scheme in (Bharghavan et *al.*, 1994) modifies the *BEB* by multiplying the *CW* size by 1.5 on a collision and decreasing it by one on successful transmission as follows: in case of collision the *CW* increases multiplicatively $CW = \min(1.5 * CW, CW_{\max})$ and decreases linearly in case of successful transmission $CW = \max(CW - 1, CW_{\max})$.

The *MILD* performed well when the network load was steadily heavy but did not provide a good performance when the network load was light. This was because it wasted a large number of idle slots resulting from the long time to recover from the backoff caused by occasional collisions. Furthermore, *MILD* could not adjust the *CW* size quickly when the number of stations varied sharply, since it decreased the *CW* size by 1 after successful transmission. This is insufficient in heavily loaded networks. Another feature of *MACAW* was the use of a Data Sending (*DS*) packet to replace the need for carrier sensing as an attempt to improve the protocol performance.

### 2.11.2 Analytical Models for IEEE 802.11 MAC Protocol

There have been many studies on IEEE 802.11 MAC protocol to improve QoS. The IEEE 802.11 DCF has been modelled in several studies. The analysis models proposed in (Bianchi, 1996), (Bianchi, 1998), and (Bianchi, 2000) derived simple analytical models to compute the saturation throughput of the IEEE 802.11 DCF. They showed that the performance of DCF depended on the protocol parameters, particularly, the $CW_{min}$ and the number of contending stations. Bianchi and Tinnirello also proposed an analytical model to estimate the number of contending station at saturation conditions based on an extended Kalman filter ($\text{Bianchi}_{(b)}$ and Tinnirello, 2003). Their extended scheme proved its effectiveness in both saturation and non saturation conditions. Their models mostly indicated that the throughput is highly dependent on the value of *CW*. For instance, the study in (Bianchi, 2000) showed that throughput was highly dependent on the $CW_{min}$ size and the optimal value of $CW_{min}$ depended on the number of contending stations in the network, where small contending stations required small $CW_{min}$ size. Cali et *al.* (1998) proposed an analytical model to estimate the theoretical upper bound for the capacity of IEEE 802.11b protocol. According to the proposed analytical formula, their results showed that the protocol operated at much lower level than the theoretical capacity. $\text{Cali}_{(a)}$ and Gregori (2000) derived a theoretical upper bound by approximating DCF with a p-persistent protocol. They proposed a dynamic and distributed algorithm, IEEE $802.11^{+}$ (IEEE 802.11 plus), which permitted each station to estimate the number of contending stations and to tune its *CW* to the optimal size at runtime. $\text{Cali}_{(a)}$ and Gregori (2000) assumed that stations were able to obtain perfect feedback about the network and channel conditions. Ziouva and Antonakopoulos (2002) improved Bianchi model proposed in (Bianchi, 1998) by considering the impacts of busy channel conditions on the backoff algorithm in order to calculate the delay bound at saturation conditions. Utilising (Bianchi, 1998), (Bianchi, 2000), and (Ziouva and Anotonakopoulos, 2002), Xiao (2003) proposed a priority model for real-time applications. The priority scheme involved differentiating the $CW_{min}$ size, an increasing factor for the value of *CW*, and the maximum backoff stage. Accordingly, delay and throughput parameters of different priority classes at saturation condition were estimated analytically. The scheme provided reasonable service differentiation with regards to delay and throughput. Bianchi$_{(a)}$ and Tinnirello (2003) proposed another analytical model for evaluating the throughput and delay performance of the *CSMA/CA* mechanism. Further, their results showed that using different *IFS* values, QoS differentiation could be provided. Other analytical models such as (Zhao et *al.*, 2002), (Banchs et *al.*, 2003), and (Zhu and Chlamtac, 2003) were also proposed to improve the performance of the IEEE 802.11e protocol and to provide service differentiation according to the values of $CW_{min}$, *BI*, and *IFS*.

In (Li and Battiti, 2003) an analytical model to compute the throughput and packet transmission delays to support service differentiation has been proposed. The proposed approach was based on adjusting the $CW_{min}$ value and the packet length according to the priority of each traffic flow. The simulation results of the proposed approach showed that good accuracy of the performance evaluations could be achieved by using the proposed analysis model. A simple performance analysis that calculated throughput, packet delay, packet drop probability and packet drop time for the IEEE 802.11 protocol has been proposed in (Chatzimisios et *al.*, 2004). According to their results the $CW_{min}$ size, the $CW_{max}$ size, and the data rate considerably affected the performance of IEEE 802.11 protocol for channel access. Also their results showed that high values of $CW_{min}$ improved the performance in terms of lower packet drop probability and higher throughput values but caused an increase in packet delay in certain cases. The study indicated that an increase in the $CW_{max}$ size enhanced transmission performance since the number of packet collisions was significantly decreased. Furthermore, increasing the data rate in which packets were transmitted resulted in a considerable degradation of packet delay. An analytical model proposed in (Li et *al.*, 2004) indicated that throughput, delay and fairness were sensitive to the chosen system parameters such $CW_{min}$, $CW_{max}$ and the number of packet retransmissions. Therefore, adjusting the system parameters may achieve an optimal performance.

Although the development of analytical models for the IEEE 802.11 MAC protocol provided a good knowledge to understand the behaviour of the protocol, they were only validated for saturation conditions (i.e. at the maximum load traffic). Furthermore, these models have focused on specific complex problems associated with the IEEE 802.11 protocol operations which limited the scope of analysis.

### 2.11.3 Adaptations of the IEEE 802.11 Transmission Parameters

The standard IEEE 802.11 MAC protocol was developed without any consideration of QoS characteristics and offers only a basic best effort service. However, in order to improve the protocol performance and to provide QoS in the original IEEE 802.11 DCF scheme, several studies based upon numerous different MAC transmission parameters have been proposed.

Several recent studies (Kang and Mutka, 2001), (Peng et *al.*, 2002), (Kuo and Kuo, 2003), (Kwon et *al.*, 2003), (Deng et *al.*, 2004), (Kuppa and Prakash, 2005), and (Sung and Yun, 2006) have proposed an improvement to the performance of IEEE 802.11 DCF scheme by either modifying the backoff mechanism (mainly *CW*) or adjusting the value of Inter Frame Space (*IFS*). In (Kang and Mutka, 2001), the authors proposed a differentiation scheme for ad-hoc networks which offered three levels of flow, gold, silver, and bronze. Each class had individual backoff algorithm that split from the standard *BEB*. For instance, the gold class (i.e. the higher priority) had the shortest computed backoff interval. This implied that the higher priority classes had the higher probability to gain access to the medium first.

The work in (Peng et *al.*, 2002) showed that adjusting the $CW_{min}$ value improved the throughput of the transmitting stations. The proposed algorithm required each station to calculate a parameter that was closely related to the wireless link collision and periodically refreshed the $CW_{min}$ value according to the current value of the calculated parameter. Thereafter, the transmitted data packets were used to piggy-back the just tuned $CW_{min}$ value to refresh all other stations of the single-hop network. In this approach, although there was an improvement in the throughput of the transmitting stations, all stations still had the same right to access the medium and it did not provide any type of service differentiation between different traffic types.

In (Haitao et *al.*, 2002), a contention window resetting scheme in DCF was proposed to enhance the throughput. The proposed scheme was based on modifying the operation of the IEEE 802.11 standard, whenever the retry limit is reached; the *CW* is kept fixed without resetting it to $CW_{min}$ as the case in the IEEE 802.11 standard. In a departure from the standard, after successful transmission of the packet, the *CW* value was set to the maximum of [$CW/2$ or $CW_{min} + 1$]. While after unsuccessful transmission the proposed scheme followed the IEEE 802.11 standard and doubled the *CW* size. The proposed scheme showed that modifying the MAC protocol transmission parameters improved the performance of the legacy protocol.

The Fast Collision Resolution (*FCR*) proposed in (Kwon et *al.*, 2003) aimed to provide an appropriate scheduling algorithm where a station transmits with probability 1 if it is scheduled to transmit and with probability 0 otherwise. In *FCR* scheme, when an active station detects a collision; it doubles its *CW* size and chooses a new backoff value, regardless of whether it is involved in the collision. This in turn reduces the probability of collision. Their results showed that the *FCR* scheme outperformed the IEEE 802.11 DCF scheme in terms of throughput on the cost of fairness. Several other algorithms that dynamically changed the value of *CW* have been proposed. In (Kuo and Jay, 2003) a dynamic resetting scheme of *CW* size was proposed. The proposed scheme was based on varying the *CW* size after successful and unsuccessful transmission. After successful transmission the *CW* was reset to the maximum value of ($CW/2$ , $CW_{\min}+1$), however, the *CW* size was set to value equal to the $\min(2W, CW_{\max})$ after unsuccessful transmission. Their results showed that the performance of the basic access mode strongly depended on the *CW* parameter while that of the *RTS/CTS* access mode was less affected by varying this parameter.

Zhao et *al.* (2003) described an adaptive step size algorithm to get the optimal $CW_{min}$ that increased the throughput. Romdhani et *al.* (2003) proposed an adaptive algorithm for the IEEE 802.11.e Enhanced Distributed Coordination Function (*EDCF*) called the Adaptive Enhanced Distributed Coordination Function (*AEDCF*). The approach adjusted the *CW* size for each queue (or Traffic Category *TC*) after successful and unsuccessful transmission. In *AEDCF*, after successful transmission the *CW* size in each queue decreased based on a multiplication factor rather than resetting to $CW_{min}$. Similarly when collision occurred, instead of doubling the *CW* size for that queue, it increased by another multiplication factor, taking into account the number of collision the queue experienced. In (Qiang et *al.*, 2003), a Slow *CW* Decrease (*SD*) mechanism was proposed to ease the level of contention for channel access. This scheme was based on a collision rate to estimate the level of contention in the network. Based on that, the scheme attempted to reduce the number of collisions and to enhance the achieved throughput by selecting a preferred backoff stage (previous stage - factor *g*) after successful transmission, and incrementing current stage by one after collision. Packet length was also considered to improve the throughput in heavily loaded network by adjusting the backoff timer as reported in (Bononi et *al.*, 2000).

In (Chen et *al.*, 2003), the actual *CW* value was kept fixed instead of being exponentially increased. According to the network conditions, a linear feedback model was employed to compute the throughput and the delay for the IEEE 802.11 MAC protocol. These were used to get the optimal contention window size. Their results showed that the optimal contention window scheme improved the performance compared to the IEEE 802.11 MAC protocol. The Exponential Increase Exponential Decrease (*EIED*) scheme proposed in (Song et *al.*, 2003) improved the performance of IEEE 802.11 DCF by using an exponential increase and exponential decrease functions to adjust the *CW* size after successful and unsuccessful transmissions, respectively. The *CW* size was increased by a static factor ($ri$) when the station was involved in a collision ($CW = \min[ri * CW, CW_{\max}]$) and it was decreased by a static factor ($rd$) when the station transmitted the packet successfully ($CW = \max[CW/rd, CW_{\max}]$). The *EIED* scheme did not consider the network transmission conditions since the amount of increase and amount of decrease were based on a static factor ($ri$ and $rd$). It only considered the current transmission situation and did not consider the past history of successful and unsuccessful transmission.

In (Deng et *al.*, 2004), the Linear/Multiplicative Increase and Linear Decrease (*LMILD*) backoff algorithm was presented. In this scheme, colliding stations increased their *CW* multiplicatively, while other stations overhearing the collisions increased their *CW* linearly. After successful transmission, all stations decreased their *CW* linearly. The *LMILD* outperformed the legacy IEEE 802.11 DCF at a steadily heavy loaded network, but at a lightly loaded network it did not provide a good performance because a large number of idle time slots were wasted as a result of slow linear decrease of the *CW* value. In (Qixiang et *al.*, 2004), a simple self-adaptive contention window adjustment algorithm for IEEE 802.11 MAC protocol has been proposed. Their results showed that the performance of the legacy IEEE802.11 MAC protocol was sensitive to the initial parameter settings.

Accordingly, the current best effort for maintaining QoS over the IEEE 802.11 DCF protocol is being replaced by the IEEE 802.11e. The IEEE 802.11e is proposed by the IEEE group "E" and is able to provide service differentiations (IEEE, 2004). The traffic is differentiated by having different access categories (*AC*). The priorities are determined with different parameters such as $CW_{min}[AC]$, $CW_{max}[AC]$, different arbitration inter-frame spaces (*AIFS*) and transmission opportunities (*TXOPs*[1]). IEEE 802.11e can give low average delay to high priority traffic; but at higher loads, low priority traffic suffers from starvation. Therefore, it is not advantageous to starve low priority traffic, but rather to give acceptable or relative differentiation. In (Gannoune and Robert, 2004), a dynamic tuning for the $CW_{min}$ value in the IEEE 802.11e protocol was proposed. The results obtained using the proposed approach indicated that $CW_{min}$ adjustment improved the channel utilisation and throughput and reduced the values of delay and jitter for the high priority traffic.

The Gradual Distributed Coordination Function (*GDCF*) was another example of changing the behaviour of the backoff algorithm of the basic IEEE 802.11 DCF scheme. The *GDCF* in (Wang et *al.*, 2004) changed the manner the *CW* is decreased, after successful packet transmissions. Instead of resetting *CW* to $CW_{min}$, an exponential decreasing algorithm using steps was introduced. However, their approach did not support traffic type differentiation. An adaptive DCF scheme was proposed in (Kuppa and Prakash, 2005). The proposed approach was based on adjusting the backoff procedure based on knowledge of the collision and the freeze time of the backoff timer (the time when the channel is busy). Their results demonstrated that the proposed scheme outperformed the IEEE 802.11 DCF scheme in terms of throughput. In a distributed network, in order to provide QoS and to reduce the probability of collision, Ziao and Pan (2005) proposed a QoS differentiation scheme. The proposed scheme was based on allocating smaller *CW* size for real-time traffic and considering only one transmission attempt for real-time packets after collision. Their results showed that the real-time traffic had a much better opportunity of being transmitted without a collision and therefore received a better level of service. When combined with admission control, the proposed scheme could guarantee the QoS in terms of throughput, delay.

Several studies have dynamically adjusted $CW_{min}$, $CW_{max}$, the backoff interval, or the *DIFS* values for the IEEE 802.11 DCF protocol. For example, the proposed approach in (Aad and Castelluccia, 2001) combined three MAC parameters to achieve service differentiation between different priority classes. *DIFS* was one of these parameters. It

[1] *TXOP*: is one of the crucial features of the IEEE 802.11e MAC protocol and it is defined as an interval of time when the station has the right to initiate transmissions, defined by a starting time and a maximum duration.

was statically assigned for each traffic class. Ksentini et *al.* (2004) presented the Adaptive Inter Frame Space (*AIFS*) technique for providing service differentiation between multiple Traffic Categories (*TCs*) in the IEEE 802.11e protocol. Their results indicated that the adjustment of the *IFS* led to better performance and achieved service differentiation. In (Zhang and Ye, 2004), the length of *DIFS* was adopted as a differentiation mechanism. The *DIFS* length was calculated based on the ratio of estimated transmission rate to the total transmission rate. Their scheme imposed major modifications to the IEEE 802.11 DCF scheme in which the single queue was split into two queues. Their results showed that using variable length of *IFS*, service differentiation could be achieved. In (Pattara-atikom et *al.*, 2004), the *IFS* parameter and other parameters such as quantum rate and deficit counter were used to provide absolute and relative throughput for real-time and non real-time applications, respectively. The results of their approach showed that using an adjusted *IFS* value, QoS could be supported rather than using fixed *IFS*. The results of the proposed approach in (Robinson and Randhawa, 2004) indicated that the variation of the *AIFS* between stations led to a lower probability of collisions and a faster progressing of backoff counter. Since when the wireless medium became idle, a shorter value of *DIFS* allowed a packet to decrement its backoff counter earlier than packets with longer *DIFS*. However, the different values of *DIFS* could be employed to reduce the probability of collisions.

Several studies such as (Pong and Moors, 2003), ($Gu_{(a)}$ and Zhang, 2003), (Xiao et *al.*, 2004), and (Malli et *al.*, 2004) have focused in improving the performance of the enhanced version of IEEE 802.11 protocol (i.e. IEEE 802.11e MAC). For instance, the ability of IEEE 802.11e to provide service differentiation was investigated in (Xiao, 2004). Sung and Yun (2006) proposed a method for optimising MAC parameters in the *EDCF* protocol, such as *CW* and *DIFS*. Although, the proposed method provided better performance in terms of throughput and delay than the IEEE 802.11e, it was based on storing several network configurations using Pareto database. For each new configuration the proposed scheme required comparing the current configuration with the already stored in the Pareto curve. This resulted in poor performance. Since the IEEE 802.11e MAC (IEEE, 2004) is in the final stage and not standardised yet, therefore, the focus of this study is based on the original IEEE 802.11 standard (IEEE, 1997).

## 2.12 Quality of Service

The term QoS is widely used but with a variety of meanings and perspectives. For instance, RFC 2386 defines QoS as a set of service requirements to be met while transmitting data packets from the source to the destination (Crawley et *al.*, 1998). Another definition is that QoS refers to the ability to provide a level of assurance of data delivery and to provide a set of measurable service attributes in terms of delay, jitter, throughput, and packet loss over the network. Quality of service can also be defined as the ability of network components, such as a host and application to provide some consistent level of ensuring data delivery over the network with different levels for different classes of traffic (Antonio et *al.*, 2003). Others offer "*QoS represents the set of those quantitative and qualitative characteristics of a distributed multimedia system necessary to achieve the required functionality of an application*" (Caprihan et *al.*, 1997). In this study, QoS refers to the ability of the network of providing the desired QoS requirements in terms of delay, jitter, throughput, and packet loss for the transmitted applications.

### 2.12.1 Differentiated and Integrated Services

Several research efforts were presented to improve QoS in the Internet. These efforts produced two QoS architectures known as Differentiated Service (*DiffServ*) (Blake et *al.*, 1998) and Integrated Service (*IntServ*) (Braden et *al.*, 1994). *IntServ* aimed to offer end-to-end per-flow QoS guarantees (Ghanwani et *al.*, 2000) and (Seaman et *al.*, 2000). *DiffServ* aimed to provide simple service differentiation according to differentiate the processing priority of data packets (Cisco, 2001). Although the *IntServ* QoS architecture presented a potential QoS guarantee at *IP* layer, it still has some limitations to be used in wireless networks such as scalability, practicality and complexity. Conversely, *DiffServ* as a simple architecture can easily interact with the MAC sub-layer. In the *DiffServ* approach, each station sets a small bit pattern in each packet in the *IP* version 4 (*IPv4*) Type of Service (*ToS*) field or Traffic Class (*TC*) field in *IP* version 6 (*IPv6*), to mark a packet to receive a special forwarding behaviour based on the priority level that has been assigned (Nichols et *al.*, 1998) and (Rodriguez et *al.*, 2001). The use of the *DiffServ* or priority-based approach integrated with the dynamic adjustment of MAC parameters such as *CW* and *DIFS*, service differentiation among different traffic types can be achieved at the MAC level.

### 2.12.2 QoS Components

Quality of Service in *WLAN* has several components including QoS mapping, resource allocation, prioritisation ($Gu_{(b)}$ et *al.*, 2003) and (Sheu et *al.*, 2004), traffic flow identification, policing, and admission control (Crawley et *al.*, 1998). The conversion of QoS representation between layers is referred to QoS mapping. Specifying the network resources to a requested QoS is referred to the resource allocation. In admission control, it determines the ability of the network to support the demanded traffic with the requested network level QoS parameters (Chiang and Carlsson, 2001), (Dong et *al.*, 2003), and (Ahn et *al.*, 2002).

Resource reservation is usually used in centralized mechanisms. The prioritisation scheme is normally employed in distributed mechanisms. In the differentiation scheme, traffic is classified based on the application and resources are assigned according to classes of priority depending on the availability and demand (Iftikhar et *al.*, 2003). In the reservation scheme, network resources are reserved according to signalled requests initiated from applications. In differentiation schemes, packet transmissions are defined with a priority level that defines the treatment with respect to other priorities. The priority value can be used to label the packet to belong in a specific class. Each traffic class can have a predefined QoS support (Ferguson and Huston, 1998), (QoSForum, 1999), and (Kilkki, 1999).

Traffic flow identification is used to assign the configured priority or reservation to proper packets. This can be achieved either by assigning a separate label of the flow or by checking the header information of a network packet (QoSForum, 1999), (Cisco, 2001), (Rodriguez et *al.*, 2001), and (Kilkki, 1999). Policing and shaping are other two important QoS components used to limit traffic flow. Policing refers to monitoring of the delivered traffic by dropping or remarking traffic that exceeds limits in order to protect a network from malicious behaviour. Shaping regulates the traffic back to a defined rate by delaying the traffic in order to meet the specified reservation. Admission control component is the first step that determines whether to accept or reject the QoS requests. Admission control verifies the ability of the network to support the demanded traffic with the requested network level QoS parameter (QoSForum, 1999) and (Cisco, 2001). Queuing with different queue management schemes (Shenker and Wroclawski, 1997) and (Stallings, 1998) is a widely used method for differentiating the waiting time of data packets at the queue of node. Queues of different sizes can be used to assign levels of importance according to traffic class of service designations. Queues that overflow typically discard packets to reduce network congestion (Kakaraparthi, 2000).

### 2.12.3 Quality of Service Parameters

This section presents a brief description of the main QoS parameters that were used in this thesis.

#### 2.12.3.1 Throughput and Channel Utilisation

Throughput and channel utilisation are related to each other and they are commonly employed metrics when studying QoS. Throughput is defined as the rate of successful data transmission per unit of time and it is given in Equation 2.1 (Wang et *al.*, 2000).

$$Th_i(t) = \sum S_i(t)/t_i \tag{2.1}$$

Where $Th_i$ is the throughput (bits/s or bps) during the $i^{th}$ sampling interval, $S_i(t)$ is the total bits of all successfully received packets within the $i^{th}$ interval, and $t_i$ is the time duration of the $i^{th}$ interval.

Maximum throughput can be also defined as the maximum data transfer that can be sustained between two endpoints for an application's traffic to be carried by the network. Channel utilisation is defined as the total achieved throughput with respect to the channel bandwidth.

#### 2.12.3.2 Average Delay and Cumulative Distribution of Delay

Average delay is another popular QoS performance metric. It imposes strict QoS requirements for time-sensitive applications such as audio and video. It is defined as the waiting time from when a packet enters the interface queue until the packet is successfully acknowledged or it is the amount of time needed by a packet to be transmitted and completely received by the destination (Almes, 1999) and (Michaut and Lepage, 2005). In this study delay and average delay are given in Equations 2.2 and 2.3, respectively (Wang et al., 2000).

$$D_i = r_i - s_i \tag{2.2}$$

$$Average\ delay = \frac{1}{n}\sum_{i=1}^{n} D_i \tag{2.3}$$

Where $D_i$ is delay (in second) of the $i^{th}$ packet arrived, and $r_i$ and $s_i$ are the timestamps of the arrival and departure of the $i^{th}$ packet, and $n$ is the number of received packets.

Cumulative distribution of delay is an important metric that needs to be considered particularly for QoS differentiation schemes. It provides a general overview about the probability of packets that have delay below a certain threshold (e.g. probability of packets that have delay less than or equal to 400 msec in which QoS can be achieved).

#### 2.12.3.3 Average Jitter

Jitter is one of the most important characteristics of networks supporting time dependent applications. It is used for determining QoS for video and audio applications. It refers to the variation in delay between consecutive packets. In this study, jitter and average jitter are computed according to Equations 2.4 and 2.5 (Michaut and Lepage, 2005).

$$J_i = |D_i - D_{i-1}|, \qquad i \rangle 0 \tag{2.4}$$

$$Average\ jitter = \frac{1}{n}\sum_{i=1}^{n} J_i \tag{2.5}$$

Where $J_i$ is the absolute values of jitter in second of the $i^{th}$ packet, $D_i$ and $D_{i-1}$ are the delays of two consecutive packets obtained from Equation 2.2, and $n$ is the number of successfully received packets.

#### 2.12.3.4 Packet Loss

Packet loss is important for certain applications. For example in data transmission, packet loss has to be 0 to achieve QoS (ITU, 2001). In wireless networks packet loss may occur for different reasons, it occurs either due to error introduced by the physical transmission medium, link errors between two wireless endpoints such as interference, link failure, buffer overflow, or due to collisions. It is defined as the percentage of packets discarded by the wireless station (source, intermediary, or destination) due to collision and due to buffer overflow. In this study the packet loss ratio is given in Equation 2.6 (Michaut and Lepage, 2005).

$$PL_i = 100 * \left(1 - \sum R_i / \sum S_i\right) \tag{2.6}$$

Where $PL_i$ is the total packet loss ratio in percent during the $i^{th}$ interval and $\sum R_i$ and $\sum S_i$ are the total number of received and transmitted packets with the $i^{th}$ interval, respectively.

#### 2.12.3.5 Collision Rate and MAC Protocol Efficiency

Collision rate and MAC efficiency are related to each other. Collision rate is a primary performance metric used in this study since collision is one of the main challenges in wireless networks. MAC efficiency represents the percentage of total successfully acknowledged packets to the total number of sent packets at MAC level.

The IEEE 802.11 MAC protocol can support a wide range of applications with various traffic characteristics, such as video, audio and data transfers. The transmission of these applications over the wireless channel is characterised by the QoS performance. Various applications have various QoS requirements or sensitivity. A summary of QoS requirements for the applications recommended by ITU can be found in (ITU, 2001).

### 2.12.4 Multimedia QoS Requirements over the IEEE 802.11 Protocol

With an increasing amount of audio and video being sent over computer networks, the ability to provide QoS guarantees for these applications are important. The challenges associated with providing service guarantees for multimedia transmission are various. For instance, in wired networks the main and critical challenge is network congestion. Several challenges exist for wireless networks in addition to those in wired networks. For this reason and in order to provide QoS for multimedia transmission, QoS parameters have to suit the application requirements. The transmission of these applications over the wireless channel is characterised by four primary QoS metrics (throughput, delay, jitter, and loss). These metrics were discussed earlier. According to these parameters, multimedia applications have different QoS requirements. For instance, a video conferencing service requires, low jitter, low delay and higher bandwidth but can tolerate some packet loss. Other applications also have various QoS requirements. The sensitivity of these applications to the QoS parameters can be seen in Table 2.2 (Abdullah et *al.*, 2003).

**Table 2.2:** Examples of common applications and the sensitivity of their QoS requirements.

| Applications | | Sensitivity | | | |
|---|---|---|---|---|---|
| | | Loss | Delay | Jitter | Bandwidth |
| **Data traffic** | E-mail | High | Low | Low | Low |
| | File transfer | High | Low | Low | Low, Medium, High |
| | Audio on demand | Low | Low | High | Medium |
| | Video on demand | Low | Low | High | High |
| **Real time** | Telephony | Low | Low | High | High |
| | Videoconferencing | Low | High | High | High |
| | Confidential Videoconferencing | Low | High | High | High |

### 2.12.5 IEEE 802.11 QoS Classifications

Several studies were carried out for supporting QoS in ad-hoc networks. This includes QoS-based routing protocols (Chen and Nahrstedt, 1999), (Lin and Liu, 1999), and (Baoxian and Mouftah, 2005), resource reservation schemes (Mirhakkak et *al.*, 2000), and a MAC protocol (Sobrinho and Krishnakumar, 1999). Generally, these schemes work together to achieve specific goals that are specified by a QoS service model

(Braden et *al.*, 1994), (Blake et *al.*, 1998), and (Xiao et *al.*, 2000). The focus of this study is the QoS mechanisms in the standard IEEE 802.11 MAC protocol.

IEEE 802.11 MAC protocol can be categorised into distributed and centralised control schemes. The centralised scheme is not considered due to its complexity and lack of scalability which made the use of this scheme limited (i.e. it requires a controller which limited the coverage area). However, the distributed scheme is considered due to its simplicity, scalability, robustness and ease of implementation (i.e. each node can operate as a source, destination, or a router). For the distributed scheme, QoS support can be categorised into priority-based and fair scheduling-based as shown in Figure 2.11 (Pattara-atikom et *al.*, 2003). The priority-based scheme is the aim of this study.

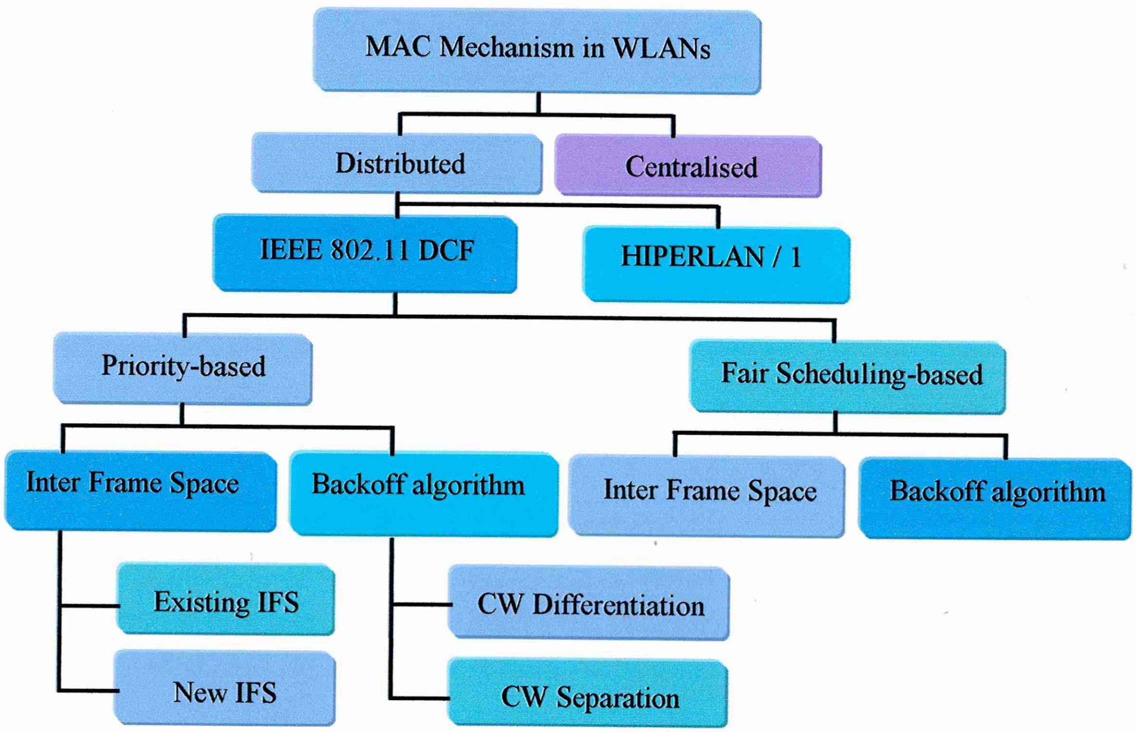



**Figure 2.11:** Taxonomy of Distributed QoS in IEEE 802.11 MAC.

As depicted in Figure 2.11, the IEEE 802.11 MAC protocol parameters such as *CW*, *DIFS*, and Backoff Interval (*BI*) have been proposed for supporting QoS and performing service differentiation between different traffic types for the DCF scheme.

### 2.12.5.1 QoS Priority-based Mechanisms

In the priority-based scheme, most of the proposed approaches were aimed to support service differentiation by providing different MAC parameters values that enabled high priority classes to access the medium faster than low priority classes. For instance, faster access can be provided by assigning a smaller *CW* that results in a smaller

Backoff Interval (*BI*) as reported in (Sobrinho and Krishna, 1996), (Benveniste, 2002) (Chen et *al.*, 2002) (Veres et *al.*, 2001) (Kim et *al.*, 2001) (Barry et *al.*, 2001) (Kang and Mutka, 2001) (Sheu et *al.*, 2004) and (Gannoune, 2006) or by assigning smaller Inter Frame Space (*IFS*) as reported in (Deng and Chang, 1999), (Aad and Castelluccia, 2001), and (Ksentini et *al.*, 2004).

Service differentiation through allocating different *CW* values was based on two techniques: (1) *CW* differentiation (*CWD*) and (2) *CW* separation (*CWS*). The work proposed in (Ayyagari et *al.*, 2000) and (Gannoune, 2006) was based on *CWD* scheme. In (Ayyagari et *al.*, 2000), the small values of $CW_{min}$ and $CW_{max}$ were assigned to high priority traffic where the $CW_{min}$ of high priority class is less than the $CW_{min}$ of low priority class and the $CW_{max}$ of high priority class is also less than the $CW_{max}$ of low priority class. Because the *BI* is a random integer that is uniformly distributed between [0, *CW*], the two priorities were differentiated by the average *BI* values. In *CWD* scheme, the *CW* values between different classes may overlap, consequently, low priority traffic can sometimes gain access to the medium earlier than high priority traffic and this depends on the amount of overlap among the contention window values of the traffic classes. For instance, the results of using narrow overlap between the $CW_{min}$ and $CW_{max}$ of high and low classes as in (Barry et *al.*, 2001) showed that the delay between these classes was clearly differentiated. Another example that used *CWD* scheme to provide service differentiation was proposed in (Chen et *al.*, 2002). The proposed approach was based on a dynamic adjustment of the *CW* range with respect to the variation in the number of active stations. It used a priority reference value called priority limit that was piggy-backed with the transmitted packets to aid each individual station to compute its *CW*.

In *CWS* scheme, the $CW_{min}$ and $CW_{max}$ values of high priority traffic are completely separated from the $CW_{min}$ and $CW_{max}$ of low priority traffic. In this case, the high priority traffic is more likely to be served before low priority traffic if they arrive at the same time. If they arrive in two different times, the low priority traffic may be transmitted before high priority traffic even if the ranges are completely separated; therefore the *CW* sizes of these classes may still overlap. The proposed scheme in (Deng and Chang, 1999) is an example of the *CWS* scheme. The proposed scheme was based on specifying two different *CW* values, $CW_{high-priority}$ for high and $CW_{low-priority}$ for low priorities which were completely separated.

Using *IFS* is another technique for providing service differentiation in IEEE 802.11 MAC protocol. This technique provides different waiting times for different priority classes when the medium is sensed idle by the contending stations. Subsequently, service differentiation can be performed by allocating smaller *IFS* values to high priority traffic in order to gain earlier access to the medium than low priority traffic.

Service differentiation through using the *IFS* values was based on: (i) using the existing *IFS* values defined by the standard such as *SIFS*, *PIFS*, and *DIFS* (see section 2.9.2.1) and (ii) using new *IFS* values. Different schemes were proposed based on the already available *IFS* values (i.e. *SIFS*, *PIFS* and *DIFS*). For instance, the proposed approaches in (Deng and Chang, 1999), (Shue and Shue T., 2001), and (Banchs et *al.*, 2001) were used *PIFS* and *DIFS* values to differentiate between time-sensitive and time-insensitive applications. On the contrary, some other approaches used new *IFS* values to differentiate between high and low priority traffic. These new *IFS* values were based on allocating the low priority traffic longer *IFS* value than the *IFS* value of high priority traffic (i.e. $IFS_i < IFS_j$), where $i$ and $j$ represent the high and low priority traffic, respectively. For instance, in (Aad and Castelluccia, 2001), different schemes to provide service differentiation were used; these were: different backoff interval, different $CW_{min}$, different *IFS*, and different frame length. They also added a small random time at the end of *IFS* to mitigate collisions with packets in the same priority class. Note that, QoS differentiation parameters of each class were statically assigned.

### 2.12.5.2 Fair Scheduling mechanisms

Other schemes such as Distributed Fair Scheduling (*DFS*) were proposed to achieve fair access to the wireless medium among the contending stations (Vaidya et *al.*, 2000). These schemes were based on the fair queuing mechanism in the wireless domain. In this category, the channel is allocated based on the flow requirements since some flows may require higher throughput than other flows. Therefore, a scheduling mechanism is used to provide a fair resource allocation according to flow requirements.

Several approaches were proposed and adopted the idea of fair scheduling. For instance, Vaidya et *al.* (2000) used the IEEE 802.11 backoff algorithm to determine which station should send first based on the specified weight for each station. The longer the backoff interval was, the lower was the weight of the sending station. The backoff interval range was determined by the frame length. So, a packet with the smallest ratio between its length and its weight gained a faster access to the channel. In this case, the high priority traffic was assigned a higher weight which provided it with a shorter backoff interval.

Some other examples were also proposed to achieve fair distribution of channel capacity such as Distributed Weighted Fair Queuing (*DWFQ*) proposed by Banchs$_{(a)}$ and Perez (2002) and the Assured Rate MAC Extension (*ARME*) schemes proposed in (Banchs$_{(b)}$ and Perez, 2002) and the absolute and relative throughput proposed in (Pattara-atikom et *al.*, 2004). In *ARME*, two types of services were proposed; the assured rate service and the best-effort service. The assured service was obtained according to the calculated *CW*, while the best-effort service followed the operation of the IEEE 802.11 MAC protocol. The computed *CW* size of assured service was associated with the bandwidth request of the assured station. Therefore, *ARME* adjusted *CW* for the assured class until the bandwidth requested was achieved. In this case, the smaller the *CW* of the assured station the higher the probability of gaining access to the channel.

In (Kanodia et *al.*, 2001), relative priorities for delay and throughput in multi-hop networks were proposed. This approach aimed to send back the scheduling information into *RTS*/*DATA* packet and then used this information to modify the backoff intervals. This approach required all stations to monitor all transmitted packets in order to obtain the scheduling information which in turn increased the overhead in the network. Some other studies such as (Barry et *al.*, 2001), (Ayyagari et *al.*, 2000), (Imad and Castelluccia, 2001), and (Vaidya et *al.*, 2000) were proposed to provide service differentiation based on the distributed function of the standard. These schemes were based on modifying the backoff intervals of the IEEE 802.11 MAC protocol. These studies and the already discussed ones were also considered to be the basis of this work.

Although, the fair scheduling mechanisms achieved fair access to the medium and assured service differentiation such as the work carried out in (Vaidya et *al.*, 2000), (Banchs$_{(a)}$ and Perez, 2002), and (Banchs$_{(b)}$ and Perez X., 2002), they imposed some major modifications to the IEEE 802.11 standard. These modifications increased the system complexity in addition to extra computational cost. For instance, the *DWFQ* mechanism proposed by Banchs$_{(a)}$ and Perez (2002) increased the complexity and the overhead by exchanging a special frame to broadcast the fairness ratio between stations. Therefore, the use of fair scheduling mechanisms were not the purpose of this study, instead, a priority-based schemes were used due to their simplicity, less computations and ease of implementation.

## 2.13 Issues to be Addressed

Many efforts have been dedicated on improving the performance of the standard IEEE 802.11 DCF scheme, particularly by tuning the MAC protocol transmission parameters. These studies experienced several drawbacks. For instance, some of the discussed schemes required exchange of control messages between stations; others imposed sophisticated computations and major modifications to the structure of the IEEE 802.11 DCF scheme. These only depended on the current network conditions without considering the past history the network experienced. Most of these studies only considered one or two QoS parameters when one or two applications were transmitted. Therefore, the directions of interest considered in this study are: the number of MAC transmission parameters and the combination of these parameters which can be adjusted in order to improve the performance of the protocol and to provide QoS differentiation. The number of QoS parameters that are considered include delay, jitter, throughput, packet loss, and collision. The combination of these parameters in one evaluation system to quantify the QoS according to the application type is investigated. The study also considers the number and type of applications such as video, audio, and data that need to be transmitted and supported with a desirable level of QoS. Finally, the new approaches need to be simple so they can be implemented without major modifications to the original IEEE 802.11 standard.

Although significant research efforts were made to support service differentiation in the IEEE 802.11 DCF by adopting the priority-based scheme, several issues have not yet been considered. These include:

(i) Most of the proposed approaches were implemented for the enhanced version of the IEEE 802.11 DCF scheme (i.e. IEEE 802.11e where each station has four different traffic categories).

(ii) Most of the proposed approaches statically set the differentiation parameters and these parameters were only used at the initial transmission. The parameters settings were not based on the variations of the traffic load and this did not lead to service differentiation. If the difference between the parameters was very small, QoS differentiation can not be guaranteed and if the difference was too large, this affected the low priority since they have to wait longer when there was no high priority traffic to be transmitted. Some other approaches were based on adjusting one parameter.

(iii) Most of the proposed approaches only considered one or two QoS metrics particularly delay and throughput.

(iv) None of the proposed approaches evaluated the QoS of multimedia applications.

In this thesis the following issues were considered for providing service differentiation in the basic IEEE 802.11 DCF scheme:

(i) MAC protocol parameters such as *CW* and *DIFS* were dynamically adjusted based on the current, the past, and the future network conditions.

(ii) Several QoS metrics such as delay, jitter, throughput, packet loss, MAC efficiency, collision rate, and cumulative distribution of delay and QoS were considered.

(iii) This study uses fuzzy logic approach instead of analytical modelling to assess the QoS by combining different QoS metrics according to the application type.

(iv) In this study, in order to avoid the drawbacks of *CWD*, *CWS* and *IFS* differentiation, the adaptive differentiation scheme combined these parameters when they were dynamically adjusted.

(v) The *CW*-based differentiation, *DIFS*-based differentiation, adaptive service differentiation, and queue status monitoring schemes provided service differentiation in both single and multi-hop networks.

## 2.14 Summary

The main objective of this chapter was to provide an extensive background about wireless technologies, QoS, and the original IEEE 802.11 standard. The chapter first outlined a general overview about wireless technologies, the development of *WLAN* standardisation, *WLAN* network architectures, the electromagnetic spectrum, and *WLAN* technology. Hence after, the IEEE 802.11 *PHY* layer was outlined. Since the main focus of this study is on the IEEE 802.11 MAC protocol, sections 2.9, 2.10, 2.11 provided extensive detail about its functionalities, operation, and limitations. The IEEE 802.11 DCF scheme only provides best effort service which is insufficient for multimedia transmission, therefore, the notion of QoS, its component and the taxonomy of QoS mechanism for the IEEE 802.11 standard were discussed. Additionally, in this chapter, an extensive literature review for previous studies in the area of performance and QoS differentiation of the IEEE 802.11 MAC protocol particularly the DCF function was introduced. Afterwards, the issues that need further investigations were discussed.

Due to the limitations in the DCF and PCF schemes and the growth of multimedia applications, QoS support in the IEEE 802.11 MAC protocol is needed. This demand is commonly recognised by the IEEE 802.11 standardisation and specification groups, as well as research institutions. The QoS support proposals for the IEEE 802.11 standard are dependent on the network conditions, application type, network topology, and MAC protocol transmission parameters. The Task Group e (*TGe*) of the IEEE 802.11 in their proposal has come up with a new access mechanism, *EDCF*, in order to give priority based distributed channel access for the stations. The natural shortcomings in this proposed *EDCF* mechanism are that it tends to give better service to high priority class while offering a minimum service for low priority traffic. Although this improves the QoS but the performance is not optimal since *EDCF* parameters can not be adapted to the network conditions and the MAC transmission parameters are set statically at the initial transmission duration. Although, the upcoming IEEE 802.11e will support QoS in both centralised and distributed topologies, it has resulted in relatively complex functionality. As it has been also discussed in section 2.13, each of the related works found in the literature has its shortcomings. This provided us with sufficient motivation and justification for developing new MAC mechanisms to overcome some of these shortcomings to improve the protocol performance and to provide service differentiation in single and multi-hop networks.

# *CHAPTER 3*

# Artificial Intelligence Techniques

## 3.1 Introduction

This chapter introduces the basic concepts of the two important paradigms in Artificial Intelligence (*AI*) system: fuzzy logic, and Genetic Algorithms (*GAs*). The definition of fuzzy logic is presented with its foundations and its basic concepts and operations. The fundamental principles of conventional *GAs* and its main operational steps are explained. The applications of these two techniques in the area of wireless network, in particular in the IEEE 802.11 MAC protocol are reviewed.

The study within this chapter is twofold: (i) use Fuzzy Inference System (*FIS*) to combine QoS parameters and (ii) use the *GA* to optimise the IEEE 802.11 MAC protocol transmission parameters.

## 3.2 Fuzzy logic

Fuzzy logic has proved popular by the scientific community for a variety of applications and it has a long tradition with respect to its scientific development. Initially, it was used in system theory to describe and implement uncertain ideas and general concepts. Lotfi Zadeh in 1960s applied fuzzy set theory to logic and linguistics, and then to artificial intelligence (Zadeh, 1965).

Fuzzy logic has many similarities with human knowledge and reasoning. Its robustness due to the direct expression of the input/output relationship without a physical derivation of the rules, its simplicity and less complexity provide it with a growing interest in the engineering community (Pedrycz, 1993) and (Yager and Filev, 1994). It has been used in several areas such as control, decision-making, optimisation, graphical theory, and evaluating systems. It was created to describe the real-world slope that exists between true and false. Instead of having an absolute true or absolute false as the case in binary logic, fuzzy logic deals with degrees of membership and degrees of truth between the extremes. Figure 3.1 shows how fuzzy logic implements a gradient of possible states between true and false as opposed to a binary logic which has only one or zero values (Franklin et *al.*, 1998).

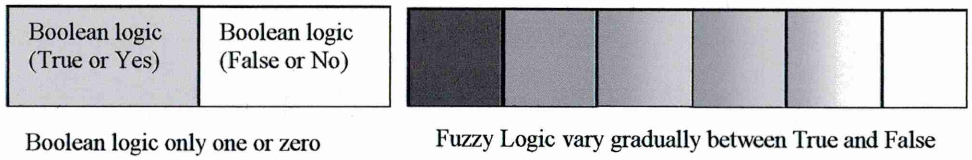



**Figure 3.1:** Binary logic vs. Fuzzy logic.

Fuzzy logic provides a feasible solution for controlling complex systems that cannot be analysed with traditional techniques. A fundamental concept of fuzzy logic is the mathematical description of linguistic uncertainty using fuzzy sets. Some popular elements that contribute to uncertainty in decision making are imprecise, missing, inaccurate, and conflicting information (National, 1997) and (Raju et *al.*, 1991). Fuzzy logic is based on the theory of fuzzy sets where variables can have different degrees of membership of sets. This is different from the well-known logic theory where a variable is either a full member of a set or it is not a member of that set. The degree of membership of a variable in a fuzzy set can vary between 0 and 1. This can allow for gradual transition from a membership function to a non-membership function instead of abrupt transition as the case in the traditional theories.

Fuzzy logic uses *IF* (antecedent) - *THEN* (consequent) rules to generate conclusions or outputs from input variables. The antecedents are the "*IF*" part of the rule (inputs) that are used in the decision-making process. The consequents are the "*THEN*" part or the implications of the rules. If the number of inputs is large, the number of rules in the rule set can become unmanageable, as they increase exponentially with the number of inputs (Raju et *al.*, 1991).

## 3.3 Structure of a Fuzzy System

A block diagram of a fuzzy system is presented in Figure 3.2. The fuzzy controller is located between a pre-processing block and a post-processing block (Jantzen, 1998), (National, 1997) and (Franklin et *al.*, 1998). The following subsections outline each block briefly.

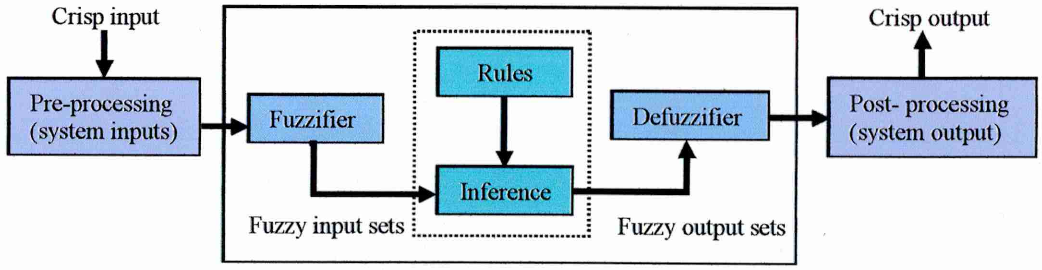



**Figure 3.2:** Block diagram of a fuzzy inference system.

### 3.3.1 Fuzzification

Fuzzification converts each input data to degrees of membership by using one or several membership functions. The fuzzification step matches the input data with the conditions of the rules to determine how well the condition of each rule matches that particular input instance. There is a degree of membership for each linguistic term that applies to that input variable (Jantzen, 1998 and National, 1997).

### 3.3.2 Rule Base

The rules of fuzzy logic may use several variables both in the condition and the output of the rules. The fuzzy controllers can therefore be used in both Single Input Single Output (*SISO*) problems and Multi Input Multi Output (*MIMO*) problems. The rules in a fuzzy system are represented in linguistic variables. A linguistic variable is used to combine multiple subjective categories describing the same context using an *IF_THEN* format. In Figure 3.3 the terms (*Low*), (*Normal*), (*Raised*), and (*High Fever*) are employed to identify the uncertain and subjective category of body temperature. These terms are named linguistic terms and correspond to values of the linguistic variable (body temperature). Each linguistic term is represented by a fuzzy set defined by a membership function. The Y-axis corresponds to the degree of membership and the X-axis is called the universe of discourse (U) (Jantzen, 1998).

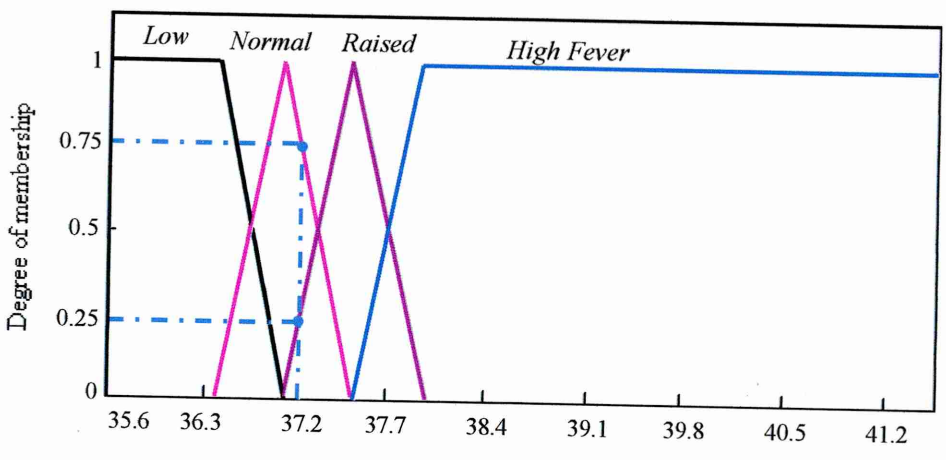



**Figure 3.3:** Body temperature variable represented using fuzzy terms (membership functions).

The linguistic variable in Figure 3.3 allows for the translation of crisp measured temperature, given in degrees centigrade, into its linguistic description. This process is called fuzzification. A body temperature of 37.2 C°, for example, might be evaluated as a raised temperature with degree of membership equal to 0.25, or as normal with degree of membership equal to 0.75. The degree of membership to the fuzzy set raised temperature can be interpreted as degree of truth associated to the statement 'the patient suffers from raised temperature'. Thus, using fuzzy sets defined by membership functions within logical expression leads to the notion of fuzzy logic.

A typical rule example that represents the QoS when multiple input variable are combined is as follows:

"***IF*** *throughput* is **high** and *delay* is **low** and *jitter* is **low** ***THEN*** *QoS* is **excellent**"

Each rule could have one or more connectives. In fuzzy, statements are connected with ***and***, ***or***, ***IF-THEN***, ***IF and only IF***, and ***NOT***. The most connective used is the multiplicative ***and*** instead of minimum. A rule could have more than one connective in the same rule, for instance: "***IF*** *loss* is **low** ***and not*** zero ***or*** *delay* is **low** ***and*** *jitter* is **medium** ***THEN*** *QoS* is **good**". Another important factor that should be taken into account is the universe or a universe of discourse. The universe contains all elements before designing the membership functions. This means, it is necessary to consider the universes for the inputs and outputs before setting up the membership functions. For example in the following rule: "***IF*** *throughput* is **high** ***and*** *delay* is **medium** ***THEN*** *QoS* is **good**", the membership functions for high and medium have to be defined for all possible values of throughput and delay, and a standard universe may be suitable.

### 3.3.3 Membership Functions

Every element in the universe is a member of a fuzzy set with a degree of membership between zero and one. The degree of membership for all its members describes a fuzzy set, such as *high* label of the input variable throughput. In fuzzy sets, each element of the universe of discourse is assigned a degree of membership (Jantzen, 1998). The switch from a membership to a non-membership occurs gradually rather than a sharp move. For instance, the transition from the membership function *low* to the membership function *medium* occurs gradually according to the overlap between these membership functions. All the elements that have a degree of membership are called the support of the fuzzy set (Jantzen, 1998). Each element ( $x$ ) in the universe has a number that can be obtained by a function called a membership function ( $\mu(x)$ ). The element $x$ at the universe and the membership function $\mu(x)$ can be expressed in a single pair of fuzzy sets. In the fuzzy set $S = \{(x, \mu(x))\}$, $x$ belongs to the universe and $\mu(x)$ is its degree of membership in the fuzzy set $S$. The pair $(x, \mu(x))$ is a fuzzy singleton; singleton output means replacing the fuzzy sets in the output by numbers (Jantzen, 1998).

Different shapes of membership functions can be employed when modelling linguistic terms. This includes Trapezoidal, Gaussian, Triangular, Sigmoid, Crisp, Singleton, and Bell-shaped. Figure 3.4 depicts a Gaussian membership function example of the output variable QoS.

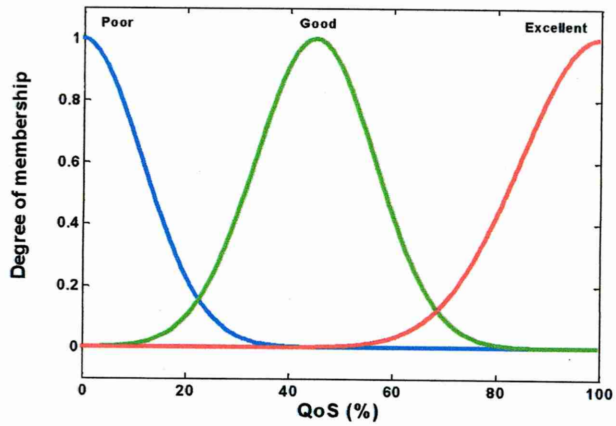



**Figure 3.4:** Gaussian membership functions for the output variable QoS.

### 3.3.4 Fuzzy Inference Engine

The *FIS* uses the fuzzified inputs together with the rules to perform inferencing. This operation determines the degree that the premise of each rule is satisfied. If the premise of a given rule has more than one clause then fuzzy operators are applied to combine them. The two commonly used operators are *AND* (i.e. minimum value) and *OR* (i.e. maximum value) as given in Equation 3.1. The values obtained are used to shape the membership functions of the output fuzzy sets (i.e. the *conclusion* part of the rules). A commonly used approach is to truncate the membership functions of the output fuzzy sets by using the results obtained when evaluating the premise parts of the rules.

The fuzzy inference consists of two components: Firstly, aggregation, which is evaluation of the ***IF*** part of each rule, and secondly, composition, which is evaluation of the ***THEN*** part of each rule. The conventional linguistic operators used for two-valued logic is not applicable with fuzzy sets; thus, with fuzzy logic, different sets of operators are defined as given in Equation 3.1 (National, 1997).

$$\left[\begin{array}{c} \text{AND: } \mu A \bullet B = \min(\mu A, \mu B) \\ \text{OR: } \mu A + B = \max(\mu A, \mu B) \\ \text{NOT: } \mu \neg A = 1 - \mu A \end{array}\right] \tag{3.1}$$

The degree of truth of the ***IF*** condition is computed using the linguistic operator to indicate how sufficiently each rule describes the current condition. More than one rule might be triggered at the same time describing the current situation presented by the input variables. Each of these rules defines an action (consequent or conclusion) to be taken in the ***THEN*** condition. The degree to which the consequent is valid is given by the adequateness of the rule to the current situation. This adequateness is computed by the aggregation phase as the degree of truth of the ***IF*** condition (Jantzen, 1998).

In the composition stage, the final linguistic value of the output variable (conclusion) is obtained by using the maximum (MAX) operator on the possible consequents from the rule-base. This type of fuzzy inference is called MAX-MIN Inference (Mamdani, 1977) and (Mamdani and Gaines, 1981). However, other implication methods can be used, such as Sum-Product method (Kaufman, 1975).

### 3.3.5 Defuzzification

Defuzzification is the process that coverts the output of the fuzzy set (inference engine) into numeric values (crisp value). There are several defuzzification methods these include Centre of Gravity (*CoG*), Centre of Area (*CoA*), Centre of Maximum (*CoM*) and Mean of Maximum (*MoM*). A common approach for obtaining the *FIS* output is centroid or CoG which calculates the weighted mean of the membership function for the fuzzy set *i* (Ross, 1995).

## 3.4 Genetic Algorithm (GA)

Genetic algorithms, *GAs*, were developed by John Holland and are modelled on the Darwinian concepts of natural selection and evolution (Holland, 1975). Genetic algorithms are a relatively new class of optimisation techniques, which are generating growing interest in the engineering community. They are well suited for a broad range of problems encountered in science and engineering and have performed efficiently in a number of diverse applications in electrical engineering, e.g. system identification (He et *al.*, 2002) and (French et *al.*, 1997), neural networks (Blanco et *al.*, 2000) and (Lopez et *al.*, 1999), fuzzy systems (Wong and Hamouda, 2000) and (Baron et *al.*, 2001), image processing (Pastorino et *al.*, 2000), signal processing (Liu, 2001), and wireless network, (Yener and Rose, 1997), (Sherif et *al.*, 1999), (Ozugur et *al.*, 2001), (Turgut et *al.*, 2002), (Barolli et *al.*, 2003), (Zdarsky et *al.*, 2005).

Genetic algorithms are based on principles inspired from the genetic and evolution mechanisms observed in natural systems (Goldberg, 1989). Their fundamental principle is the maintenance of a population of solutions to the problem that evolves toward the global optimum. They are based on the triangle of genetic reproduction, evaluation and selection (Goldberg, 1989). Genetic reproduction is performed by means of two main genetic operators: crossover and mutation. Evaluation is performed by means of the fitness function that relies on the specific optimisation problem. Selection is the mechanism that chooses parent individuals with probability proportional to their relative fitness for mating process (Abo-Hammour et *al.*, 2004).

Genetic algorithms can be distinguished from, calculus-based and enumerative methods, for optimisation by the following characteristics (Goldberg, 1989):

- *GAs* search for optimal solution using a population of individuals, not a single individual. This very important characteristic gives *GAs* much of their search power and also points to their parallel nature.
- *GAs* use only objective function information. No other auxiliary information is required. Much of the interest in genetic algorithms is due to the fact that they belong to the class of efficient domain-independent search strategies that are usually superior in performance to traditional methods without the need to incorporate highly domain-specific knowledge.
- *GAs* use probabilistic transition rules, not deterministic rules in contrast to calculus based and enumerative methods.

The construction of a genetic algorithm for the solution of any optimisation problem can be separated into five distinct but, related operations (Goldberg, 1989): (i) the genetic representation of potential problem solutions, (ii) a method for creating an initial population of solutions, (iii) the design of the genetic operators, (iv) the definition of the fitness function, and (v) the setting of the system parameters, including the population size, probabilities by which genetic operators are applied. Each of these components greatly affects the solution obtained and the performance of the genetic algorithm. These five factors have resulted in the availability of numerous variants of *GAs* reported in the literature. The genetic algorithm used in this study consists of the following steps (Goldberg, 1989) and (Abo-Hammour et *al.*, 2002):

(i) ***Initialization***: An initial population comprising of $N_p$ individuals is created in this phase at the genotype level by filling the bit strings randomly with 1 or 0 values. The coding process is then used to find phenotype values of the population.

(ii) ***Evaluation*:** The fitness, a nonnegative measure of quality used as a measure to reflect the degree of goodness of the individual, is calculated for each individual in the population according to its phenotype structure.

(iii) ***Selection*:** In the selection process, individuals are chosen from the current population to enter a mating pool devoted to the creation of new individuals for the next generation such that the chance of a given individual to be selected to mate is proportional to its relative fitness. This means that best individuals receive more copies in subsequent generations so that their desirable traits may be passed onto their offspring. This step ensures that the overall quality of the population increases from one generation to the next.

(iv) ***Crossover:*** Crossover provides the means by which valuable information is shared among the population. It combines the features of two parent individuals to form two children individuals that may have new phenotype structures compared to those of their parents and plays a central role in *GAs*. Conventional crossover involves exchanging genes (bits) between each pair of parents selected from the mating pool. It is generally applied with a relatively high probability of crossover, $P_c$. Three known schemes are generally used which include the single-point crossover, the multi-point crossover and the uniform crossover schemes. In the single-point crossover method, a crossover point is randomly selected along the parent strings and the crossover operator exchanges the characters after the crossover point between the two-selected parent strings. In the multi-point crossover method, $m$ different random crossover points across the chromosome are selected first, splitting the genotype string of each chromosome into $m+1$ parts. The offspring are created by choosing genotype fragments from each parent alternately; that is, swapping partial strings with the same size and position between the two selected chromosomes. In the uniform crossover method, each chromosome position (gene) is crossed with some probability, which is typically one-half, i.e. each corresponding pair of genes exchange their values independently with a probability of 0.5. As a result, a random crossing mask is implicitly generated with the probability of one at any position typically being set to one-half. Characters from the parental strings having ones at the corresponding positions in the crossing mask are swapped while generating the offspring strings, and the remaining characters remain intact.

(v) ***Mutation:*** Mutation is often introduced to guard against premature convergence. Generally, over a period of several generations, the gene pool tends to become more and more homogeneous. The purpose of mutation is to introduce occasional perturbations to the variables to maintain the diversity within the population. In a conventional mutation operator, the bitwise complement mutation is applied at the gene level with some low probability of mutation, $P_m$. It is realized by performing bit inversion (flipping) on some randomly selected bit positions of offspring bit strings.

(vi) ***Replacement:*** After generating the offspring population through the application of the genetic operators to the parent population, the parent population is totally or partially replaced by the offspring population depending on the replacement scheme used. This completes the "life cycle" of the population.

(vii) ***Termination:*** *GA* is terminated when some convergence criterion is met.

The fact that the conventional genetic algorithm uses both genotype and phenotype data presentation, requires some coding process that relates both schemes. If the optimisation problem consists of $N_v$ variables and each variable is represented by a substring of $N_s$ characters or genes, then chromosomes or individuals are formed by cascading the genes of $N_v$ variables, forming a longer string of length $L=N_vN_s$ genes. In this way, the population may be viewed as a vector of $N_p$ elements where each element consists of $L$ genes. The coding process requires the user to specify the desired accuracy or resolution to be used, which specifies the number of genes per substring. In addition, the lower and upper bounds of the variables are needed. The decoding process is governed by the following equation (Abo-Hammour et *al.*, 2002):

$$x = x_{lower} + \frac{x_{dec}}{2^{N_s}-1}(x_{upper} - x_{lower}) \tag{3.2}$$

Where

$x$ represents the value of the variable
$x_{dec}$ represents the decimal decoded value of the variable
$x_{lower}$ is the lower bound of the variable
$x_{upper}$ is the upper bound of the variable.

To summarise the evolution process in a conventional genetic algorithm, an individual is a candidate solution of the variables to be optimised; whereby, each individual consists of a string of $L=N_vN_s$ genes. Initially, $N_p$ individuals are randomly generated representing the initial population. The population undergoes the selection process, which results in a mating pool among which pairs of individuals are crossed with probability $P_c$. This process results in an offspring generation where every individual child undergoes mutation with probability $P_m$. After that, the next generation is produced according to the replacement strategy applied. This process is repeated till the convergence criterion is met where the $N_v$ variables of the best individual are required unknown values. The block diagram of the *GA* is given in Figure 3.5 (Abo-Hammour et *al.*, 2004).

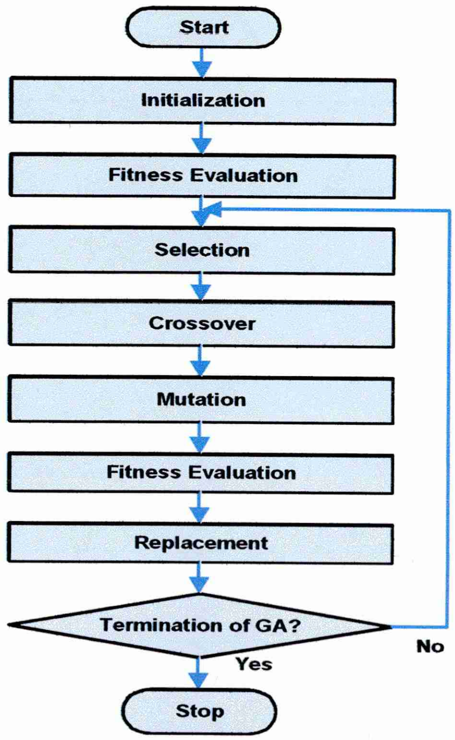



**Figure 3.5:** Block diagram of the typical genetic algorithm.

## 3.5 Application of AI techniques in IEEE 802.11 MAC protocol

Artificial intelligence techniques can be used for various tasks: analysis, evaluation, design, control, monitoring, supervision, optimisation, etc. Throughout this study, fuzzy logic will be used as an evaluation technique to assess the QoS and as a controller to adjust the $CW_{min}$ size of the IEEE 802.11 DCF scheme. The *GA* approach as an optimisation technique will be used for optimising the $CW_{min}$ and *DIFS* values of the IEEE 802.11 DCF scheme. The applications of these techniques in wireless network, particularly in the IEEE 802.11 MAC protocol are presented in this section.

A number of studies have used fuzzy logic in the area of computer networks. Fuzzy logic was used to assess QoS for multimedia transmission. In (Saraireh et *al.*, 2004), the

QoS parameters, throughput, delay, jitter and packet loss were fed into a fuzzy system and the output represented an estimate of the QoS for the transmitted application. An Adaptive Rate Controller (*ARC*) based on fuzzy inference system was proposed in (Sheu et *al.*, 2003). According to the input parameters of the fuzzy system which were: received signal strength indicator (*RSSI*), the frame error rate (*FER*) and the MAC delay, the *ARC* intelligently selected the transmission rate for frame transmissions. Their results showed that the proposed fuzzy controller improved the wireless network throughput and reduced the access delay. A dynamic contention window selection scheme to achieve a theoretical throughput limit based on fuzzy reasoning approach was proposed in (Chen et *al.*, 2004). The proposed mechanism was developed based on observing the degree that the medium was busy and the number of neighbours to generate a suitable back-off window in order to reduce the number of collisions and improve the throughput. The method provided significant improvements in terms of throughput and fairness between stations. However, the proposed approach for controlling the *CW* assumed that the channel status was known in advance and both fuzzy input parameters were based on estimating the number of active nodes.

In (Cheng and Marsic, 2001) a fuzzy reasoning for wireless awareness (*FuRWA*) was proposed. The proposed approach aimed to embed intelligence into the application to detect wireless links by using fuzzy logic reasoning. According to the simulation and experimental results, *FuRWA* was a feasible way to enhance QoS in hybrid communication environments.

The use of fuzzy logic for multimedia transmission over wireless networks was also explored. In (Ma and Gunawan, 2002) fuzzy rules were constructed to dynamically share the common resources between voice and data. Their simulation results showed that the proposed fuzzy scheme reduced the data delay at high voice loading conditions.

Fuzzy logic has been used in the area of routing protocol in wireless ad-hoc networks. In (Alandjani and Johnson, 2003), a new routing protocol that applied fuzzy logic to differentiate resource allocation according to traffic importance and network state was proposed. Their simulation results indicated that they succeeded in providing a higher reliability and a lower delay for important traffic than the previous protocols.

The fuzzy logic system has been also used for service differentiation and call admission control. In (Khoukhi and Cherkaoui, 2005), a fuzzy logic system for call admission control and service differentiation in wireless ad-hoc network was proposed. The proposed model, called *FuzzyMARS*, included a set of mechanisms: admission control for real-time traffic, a fuzzy logic system for best-effort traffic regulation, and three schemes for real-time traffic regulation. *FuzzyMARS* architecture supports both real-time *UDP* traffic and best-effort *UDP* and *TCP* traffic. Their simulation results showed that the use of fuzzy logic in wireless ad-hoc networks might add more flexibility and capability of operating with the imprecise information due to the mobility of nodes.

Chen and Hsiao (2005) proposed a distributed fuzzy control algorithm to reserve bandwidths. The proposed fuzzy control algorithm was based on controlling the delay of the mobile nodes in order to reserve the bandwidth and to achieve service differentiation between different classes. Their simulation results showed that the fuzzy controller was capable of meeting the QoS constraints for different transmission classes. Moreover, fuzzy logic has also been used in the area of QoS. By considering multiple queues for different packets, a fuzzy controller was proposed in (Zhang and Ma, 2000). In this approach two queues were considered to achieve class differentiation. However, the transmission of data packets was based on the queue status without considering the wireless MAC transmission parameters. Liu and Hsu (2005) proposed two distributed random access protocols for multi-channel *WLAN*s. These approaches were based on major modifications in the operation of the IEEE 802.11 standard protocol. The first approach was named the *CSMA/CA* protocol which was a modified version of the IEEE 802.11 standard. The second approach was called fuzzy logic control protocol (*FLC*) which used a simple fuzzy logic controller to tune the actual value of *CW*. The obtained results showed that the *FLC* protocol significantly reduced the probability of collisions, increased the channel utilization and alleviated the fairness problem inherited in the IEEE 802.11 standard. Although there was an improvement, the proposed approaches were based on multi-channel *WLAN*s and therefore required major modifications to the standard. They did not also have service differentiation between different traffic types.

Several studies were carried out into the applications of fuzzy logic into computer networks, wired, wireless, Asynchronous Transfer Mode (*ATM*) and packet switched networks. The literature contains much discussion about the use fuzzy logic as a controller, evaluation, and call admission control systems. These studies can be found in

(Ascia et *al.*, 2001), (Hu and Petr, 2000), (Ascia et *al.*, 2002), (Kazemian and Meng, 2003), and (Soud and Kazemian, 2003).

The previous work involving the use of *AI* techniques and genetic algorithms to solve problems in the wireless domain include (Yener and Rose, 1997), (Sherif et *al.*, 1999), (Ozugur et *al.*, 2001), (Turgut et *al.*, 2002), (Barolli et *al.*, 2003), and (Zdarsky et *al.*, 2005). The main consideration of these studies has been the optimal utilisation of scarce and hence costly wireless resources such as bandwidth. In (Zomaya, 2002) and (Subrata and Zomaya, 2003), the *GA* has been used to find an optimal location management strategy for cellular networks. Therefore, no work has been yet addressed by using a hybrid genetic-fuzzy mechanism for optimising multiple MAC protocol transmission parameters to provide QoS for various applications that use the same wireless medium.

## 3.6 Summary

This chapter provides an introduction to the use of *AI* techniques such as fuzzy logic and *GAs* for evaluating QoS and adjusting IEEE 802.11 MAC protocol transmission parameters. The concept of a fuzzy system is first explained. Afterwards, the structure of fuzzy logic is highlighted. A rule-based fuzzy model is described that uses the linguistic (Mamdani) model. In this model, the structure of the rules, the inference, the membership functions, and defuzzification methods are presented. Illustrative examples are given throughout the text. Additionally, the use of *GAs* as an optimisation technique in the area of IEEE 802.11 MAC protocol is presented. An introduction about *GAs* is provided. Afterward, the main steps used in the conventional *GA* are also introduced. The applications of these techniques in the wireless domain are outlined.

The use of *AI* techniques in the wireless domain as discussed in section 3.5 showed that the *AI* techniques provide an effective means to reduce the complexity. For the most complex systems where few numerical data exist and only imprecise information is available, artificial intelligence provides an effective way for understanding them (Ross, 1995). Realisation of medium access control which caters for QoS is a complex task which involves imprecise information from the measured data (i.e. delays, jitter, and throughput). The dynamics of the channel varies in space and time in a complex manner. Therefore, the use of fuzzy logic and genetic algorithms as part of the IEEE 802.11 MAC protocol are valuable tools. Subsequently, these techniques can be used for evaluating QoS and optimising the main MAC protocol transmission parameters.

# CHAPTER 4

# Experimental Procedure

## 4.1 Introduction

The aim of this chapter is to provide details of the general experimental procedures and evaluation methods that were commonly used throughout this study. The details of the procedures which are specific to individual studies are provided in the relevant chapters. Data and information sources used in this research are outlined. A detailed explanation of the procedures, tools, and data processing techniques used in this research are also discussed. An explanation of the network simulation environments, experimental design and the measurement process are also provided.

## 4.2 Simulation Overview

There are three mechanisms for performance evaluation, these are: simulation, analytical modelling, and measurement (Jain, 1991). It was not practical to use the analytical modelling technique in this research because of the nature of wireless medium that varies in time and space. Measurements from real systems are also excluded since the implementation of the proposed approaches in real networks would have been too time consuming for this study. Therefore, simulation was chosen as the most appropriate approach. Within the simulation process, data were collected from simulation runs then quantitatively analysed. The analysis and critical evaluation of data were based on two criteria: (i) the results collected from the standard protocol and (ii) the findings from recent literature search or work within the field of study.

In order to simulate wireless networks with realistic topologies a simulation tool was required. There were several simulation tools that could have been used. The most well-known tools are Global Mobile Information Systems Information Library (*GloMoSim*) (GloMoSim, 2006) and (Zeng et *al.*, 1998), Optimised Network Engineering Tools (*OPNET*) (OPNET, 2006) and Network Simulator (*NS-2*) (NS, 2006). These tools have oriented to wireless domain. *GloMoSim* was not used for purpose of this study since its model libraries are not open source and so cannot be easily modified. Although *OPNET* is well oriented to wireless models particularly the new versions, it is not open source and it imposes high level of complexity when modifications are required. *NS-2* is an open source and freely available simulation tool that runs on different platforms such as

Linux and windows[2]. *NS-2* is a comprehensive platform and can deal with number of network issues e.g. different applications, protocols, and traffic models. It can be extended either by modifying the Object Tool Command Language (*OTcl*) or *C++* code. It has also been widely validated; giving confidence in most of the functionality of the simulator. For these features, *NS-2* was used for this study. A brief review of *NS-2* tool is provided in appendix A.

In *NS-2*, a simulation task is specified by a simulation script written in the Tool Command Language (*Tcl*) (Ousterhout, 1990). This script illustrates the network topology (nodes, their configurations, locations, and interconnection), transport protocols (*UDP* and *TCP*), traffic type (*CBR*, *VBR*, and exponential traffic), and simulation events (start and stop time). This can be specified by the following steps:

(i) Create an object for the *NS-2* simulator (i.e. *C++* and *TCL* object).
(ii) Create trace files to store the simulation results.
(iii) Specify the node configurations e.g. interface queue and routing protocol.
(iv) Create objects for network nodes, locations, and links and specify their parameters, hence creating the network topology.
(v) Create objects for the *UDP* source and *UDP* destination.
(vi) Create objects for the sending and receiving applications (*CBR* and *VBR*) and attach them to the *UDP* source and Destination objects, respectively.
(vii) Schedule events, such as start and end times of data packet transmission, trace the required information about the network objects for analysis, and when the simulation should terminate.
(viii) Start the simulation.

A typical simulation process using *NS-2* is depicted in Figure 4.1. It consists of generating the input file (i.e., *TCL* script that contains the network components, the traffic and the communication between them). This input file is then used for the simulation. The output of the simulation is two trace files, the data file and the visualisation file. Data file contains all the required information about the parameters that are selected prior to the simulation. Visualisation file (*NAM*) shows the simulation run components such as the node distribution, node movement, packet sent, packet received, packet drops, and link. Other tools such as AWK (Aho et *al.* 1988), Perl (Wall

[2] Under window platform, cygwin software is required (Hunt H., 2006)

and Schwatz 1993), and *MATLAB* (MATLAB, 2004) were also used for data processing, analysis, and evaluation.

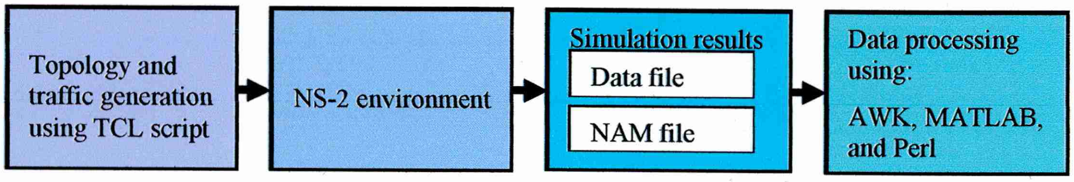
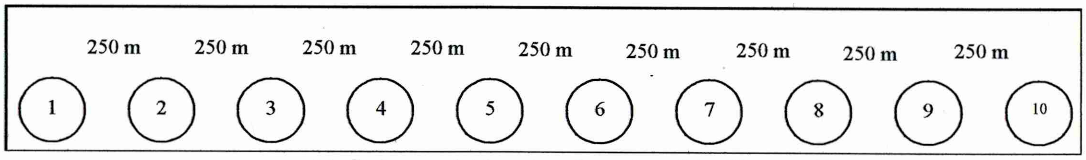


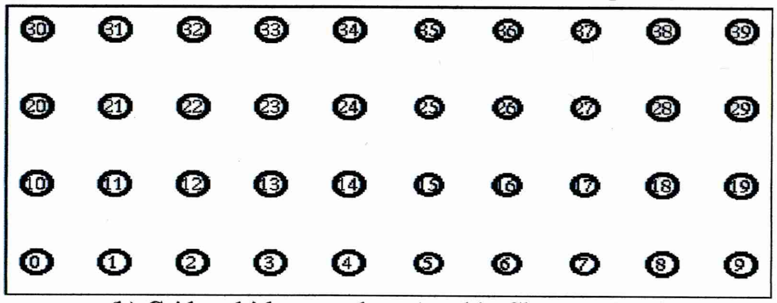

**Figure 4.1:** Atypical simulation example.

### 4.2.1 Network Topologies

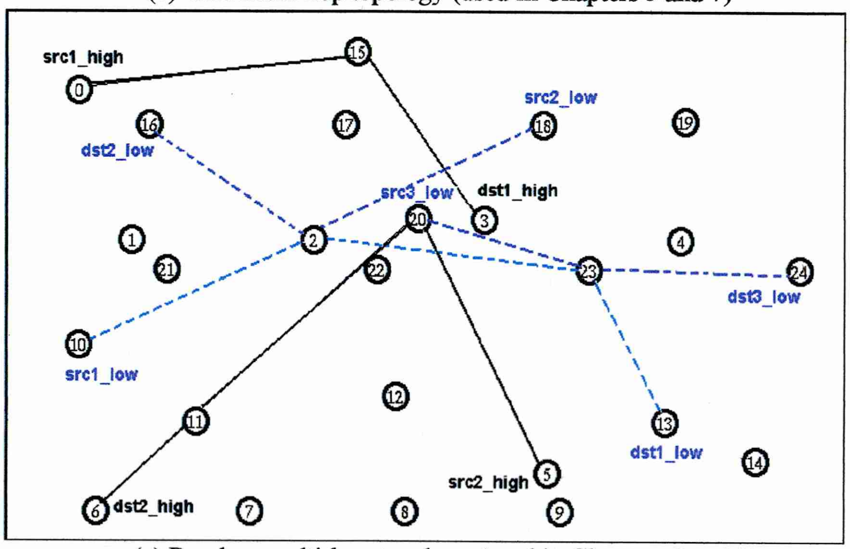
This section presents the network models used for this study. Four network models with different scenarios were used for the simulations. These models were selected since they covered the main configurations of both the single-hop and the multi-hop networks. The first model was a string topology with 10 fixed stations positioned in an area of $1800m \times 500m$ with a distance not less than 250 metres between any two adjacent stations to perform multi-hop network as shown in Figure 4.2a. This model was used to examine the limitations of the standard IEEE 802.11 DCF scheme (i.e. Chapter 5). The second model represents a grid shape topology with 40 fixed stations distributed in an area of $1800m \times 1800m$ with a distance equal 250 metres between any two adjacent stations as shown in Figure 4.2b. This model performed as a multi-hop network and was used to investigate the impact of the presence of multiple hidden terminal problems (i.e. Chapter 5). This third network model consisted of 20 stations randomly distributed and generated a multi-hop network as shown in Figure 4.2c. The third network model covered the area of $1000m \times 1000m$ with distance not less than 250 metres between any two adjacent stations. The stations in this model were fixed and were classified as high priority stations and low priority stations. The solid line represents a possible route between high priority stations while the dotted line represents a possible route for low priority stations. This model was employed with the adaptive service differentiation schemes and autoregressive models for providing service differentiation in multi-hop networks (i.e. Chapters 8 and 9). The fourth model used 40 fixed stations randomly deployed in an area of $100m \times 100m$ to represent a wireless ad-hoc network (*IBSS*) as shown in Figure 4.2d. All stations in this model could hear each others transmission. The maximum distance between the furthest stations was less than 100 metres. This model was employed for performance evaluation of the basic IEEE 802.11 DCF scheme and when studying the impact of MAC protocol transmission parameters on the QoS metrics (i.e., Chapter 5). It was also used for validating the performance of the *AI* techniques, Ratio based, *CRV*, adaptive service differentiation, and prediction approaches and used to compare them with the basic IEEE 802.11 DCF scheme (i.e., Chapters 5, 6, 7, 8, and 9). The scenarios that required specific topologies will be discussed in the relevant chapters.



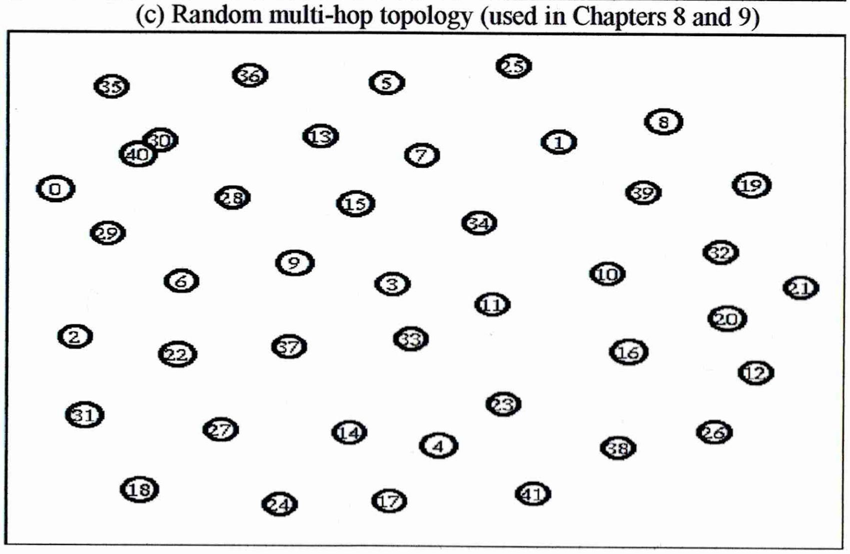

(a) String multi-hop topology (used in Chapter 5)



(b) Grid multi-hop topology (used in Chapters 5 and 7)




(c) Random multi-hop topology (used in Chapters 8 and 9)



(d) IBSS single-hop topology (used in Chapters 5, 6, 7, 8, and 9)

**Figure 4.2:** Network topologies, (a) string multi-hop topology (Chapter 5), (b) grid multi-hop topology (Chapters 5 and 7), (c) random multi-hop topology (Chapters 8 and 9), and (d) random *IBSS* topology (Chapters 5, 6, 7, 8, and 9).

### 4.2.2 Physical Layer (PHY) Parameters

The main physical layer parameter considered in this study is the channel bit rate. Two bit rates were considered. One for data transmission called data rate and the second for control frames transmission called basic rate. For most simulations, 2 Mbps data rate was assumed. Some selected simulations used a data rate up to 11Mbps. A 1 Mbps basic rate was assumed for control frame transmission. Other parameters that were implemented in the simulation are listed in Table 4.1.

### 4.2.3 Medium Access Control (MAC) Layer Parameters

The Distributed Coordination Function (DCF) of IEEE 802.11 protocol was only considered in this research study, and several approaches were proposed to extend the operation of this scheme. The aim was to minimise changes to the standard when developing the approaches. The IEEE 802.11 standard was modelled to work as a Lucent *WavLan* at frequency 914 MHz and *DSSS* radio interface card (NS, 2006). Most simulations were carried out based on this model. However, selected simulations were based on the IEEE 802.11b protocol (IEEE, 1999). The IEEE 802.11b version was modelled according to the technical specifications of an *ORiNOCO* IEEE 802.11b card. A summary of MAC and *PHY* parameters of these models is listed in Table 4.1.

**Table 4.1:** IEEE 802.11 and IEEE 802.11b simulation settings (NS, 2006) and (IEEE, 1999).

| Parameter | Typical Values IEEE 802.11 (Lucent Wavlan) | Typical Values IEEE 802.11b (ORiNOCO) |
|---|---|---|
| *DIFS* | 50 μsecs | 50 μsecs |
| *SIFS* | 10 μsecs | 10 μsecs |
| $CW_{min}$ | 31 slots | 31 slots |
| $CW_{max}$ | 1023 slots | 1023 slots |
| Slot time | 20 μsec | 20 μsec |
| RTS | 20 bytes + PHY header | 20 bytes + PHY header |
| CTS | 14 bytes + PHY header | 14 bytes + PHY header |
| ACK | 14 bytes + PHY header | 14 bytes + PHY header |
| UDP header | 8 bytes | 8 bytes |
| IP header | 20 bytes | 20 bytes |
| MAC header | 28 bytes | 28 bytes |
| PHY header | 24 bytes | 24 bytes |
| Short Retry Limit | 7 | 7 |
| Long Retry Limit | 4 | 4 |
| Modulation Technique | DSSS | DSSS |
| RTS-threshold | 2300 bytes | 2300 bytes |
| Data Rate | 2.0 Mbps | 2, 5.5, 11 Mbps for data |
| Basic Rate | 1.0 Mbps | 1.0 Mbps |
| Offered Load | 2.0 Mbps | 2.0 Mbps |
| Capture Threshold | 10.0 dB | 10.0 dB |
| Carrier sensing Threshold | 1.559e-11 W | 5.011872e-12 (W) |
| Receiving Threshold | 3.652e-10 W | 5.82587e-9 (W) |
| Transmission Power | 0.2818 W | 0.03162277 (W) |
| Frequency band | 914 MHz | 2.472 GHz |
| IFQ size | 50 packets | 50 packets |
| Routing Protocol | *AODV* | *AODV* |

### 4.2.4 Assumptions

Once the simulation environment including the topologies and the other simulation parameters were determined, some assumptions were required: an error free channel was used and collision was the only cause of transmission failure over the channel. The capacity of the channel was assumed to be 2 Mbps. This helped to minimise the simulation time into a manageable duration and allowed the behaviour of the protocol to be investigated at heavy loaded conditions regardless of the channel capacity size. The propagation times were assumed to be negligible with respect to the packet transmission time. Each station transmitted one type of traffic to its corresponding destination and stations were positioned anywhere within the specified area. The position was fixed during each simulation. The transport layer protocol in all stations was the User Datagram Protocol (*UDP*). The basic access and the *RTS/CTS* access mechanisms with Direct Sequence Spread Spectrum (*DSSS*) *PHY* were considered throughout the evaluation. The basic access mechanism with Direct Sequence Spread Spectrum (*DSSS*) *PHY* was considered as a main scheme throughout the evaluation. However, selected simulations considered both the basic access and the *RTS/CTS* access mechanisms. Queue size was equal to 50 packets for single hop scenarios (i.e. when stations were located in the same *IBSS*) and it was varied for multi-hop scenarios. This will be highlighted for each scenario in the relevant chapters. All simulations were performed in *WLAN* environments with different number of connections.

### 4.2.5 Number of Connections

The number of stations in a network plays an important role in the protocol performance particularly at high competition among stations. The number of stations in the network under investigation was varied from 2 to 40 stations according to the selected scenario. The number of connections was 2, 3, 4, 5, 6,7, 8, 9, 10, and 20. These connections were also varied according to the selected scenario.

### 4.2.6 Traffic Type and Traffic Capacity

Two traffic types were considered: Constant Bit Rate (*CBR*) and Variable Bit Rate (*VBR*). *CBR* traffic was adopted to model the multimedia type applications such as audio, video, and data. The audio packet size was 160 bytes and its inter-packet interval was 20 msec. Corresponding to 64 Kbps Pulse Code Modulation (PCM) audio flows (i.e. encoded as a G.711 voice encoding scheme) (Markopoulou et *al.*, 2003). Video has been modelled with different packet sizes. A 1280 bytes packet size and 10 msec inter-packet interval was used to generate 1 Mbps data rate, whereas, a packet size of 512 bytes and inter-packet interval of 10 msec was employed to generate 384 Kbps data rate. Data were also modelled by *CBR* traffic with different packet sizes and different transmission rates, e.g. a 200 bytes packet size and 12.5 msec inter-packet interval was modelled to deliver 128 Kbps bulk data. Larger packet sizes such as 1500 bytes with different inter packet interval was also employed to generate 960 Kbps and/or 480 Kbps data rate.

*VBR* traffic was derived from a H.263 encoded Jurassic Park I movie file (TKN, 2005). Each record of the file consisted of the time interval, frame type and frame length. The total number of frames was 50200 with an overall size of 200.5 Mbytes and 3600 seconds duration (TKN, 2005). Part of the trace file was used with 800 bytes mean frame length and 289 bytes standard deviation. Another *VBR* trace file with 60 minutes length was also considered in this study. It had variable frame size and variable interval. The packet size had a mean variation equal to 3993 and standard deviation equal to 2541 bytes.

*CBR* and *VBR* traffic were generated in various traffic capacities, such as light, medium, and heavy load. Light load traffic represented 25% of the channel capacity that was assumed to be 2 Mbps. Medium load traffic corresponded to 60% of the channel capacity. Heavy load traffic related to more than 80% of the channel capacity. The amount of the delivered traffic in each case was equally distributed among stations in the network.

In Chapters 8 and 9, *CBR* traffic was classified into high and low priority traffic. Each high priority station was capable of generating high bit rate with 384 Kbps and low bit rate at 192 Kbps. Low priority stations were able to generate either 480 Kbps or 160 Kbps as a high and low bit rate, respectively.

### 4.2.7 Quality of Service Parameters and Performance Metrics

Throughput, channel utilisation, delay, jitter, packet loss, MAC protocol efficiency, collision rate, and Cumulative Distribution of delay are considered to be the main QoS parameters for this thesis. These parameters were briefly described in section 2.12.3. According to the application type, delay, jitter and packet loss are considered the main QoS parameters for time-sensitive applications such as audio and video. However, throughput, collision rate, and MAC efficiency parameters are considered for QoS for time-insensitive applications. A summary of QoS requirements for these applications as recommended by ITU group is provided in Table 4.2.

**Table 4.2:** QoS requirements for audio, video, and data as recommended by ITU group ($ITU_{(a)}$, 2001)

| Medium | Application | Typical Data rates | Parameter | | |
|---|---|---|---|---|---|
| | | | Delay | Jitter | Packet loss |
| **Audio** | Conversational voice | 4 - 64 Kbps | less than (150 msec) preferred<br>less than (400 msec) minimum limit | less than (1 msec) preferred | less than 3% preferred |
| **Video** | Videophone | 16 -384 Kbps | less than (150 msec) preferred<br>less than 400 msec limit | less than (50 msec) preferred | less than 3% preferred |
| **Data** | Bulk data | 10 KB-10 MB | less than 10 sec | N.A | Zero |

## 4.3 Experimental Design

According to the topologies discussed in section 4.2.1, the simulations were performed for several scenarios and configurations in order to evaluate the performance of the developed schemes by means of comparison with the basic IEEE 802.11 DCF and/or the Exponential Increase Exponential Decrease (*EIED*) schemes. These scenarios include varying the network type (single-hop and multi-hop), changing network size (small, medium, and large network sizes), various traffic type (*CBR* and *VBR*), altering traffic capacity (light, medium, and heavy load traffic), specifying traffic priority (high priority or time-sensitive application and low priority or time-insensitive application), and finally when the number of active stations varied over time. A complete description of each scenario will be highlighted in the relevant chapter.

Simulations were repeated 10 times, each time using a different seed which introduced a random element in the network starting condition. This randomness for example defined which node managed to transmit first when several nodes were requested to transmit at a given time and avoided the bias of random number generation. The results of the 10 simulations were then averaged. The simulation time was 300 seconds in order to obtain accurate and consistent results in a steady state condition. This period was considered sufficient to examine the long term behaviour of the IEEE 802.11 MAC protocol. Selected simulations considered 200 and 400 seconds simulation duration and this will be highlighted for each scenario. Ad-hoc On-demand Distance Vector (*AODV*) was considered for all simulations as the routing protocol since it has proven to be efficient as opposed to proactive protocol in Mobile Ad-hoc Networks (*MANET*) (Broch, 1998).

## 4.4 Measurement Process

The simple measurement model shown in Figure 4.3 was employed for the evaluation of the developed schemes. Performance metrics such as delay, jitter, throughput, packet loss, MAC efficiency, and collisions discussed in (Chapter 2 section 2.12.3) were considered. During runtime, the simulation results were stored in a data file for analysis. AWK (Aho et *al.*, 1988) and Perl (Wall and Schwatz, 1993) scripting languages were used to extract these QoS parameters from the currently generated trace files during the same run of the simulation.

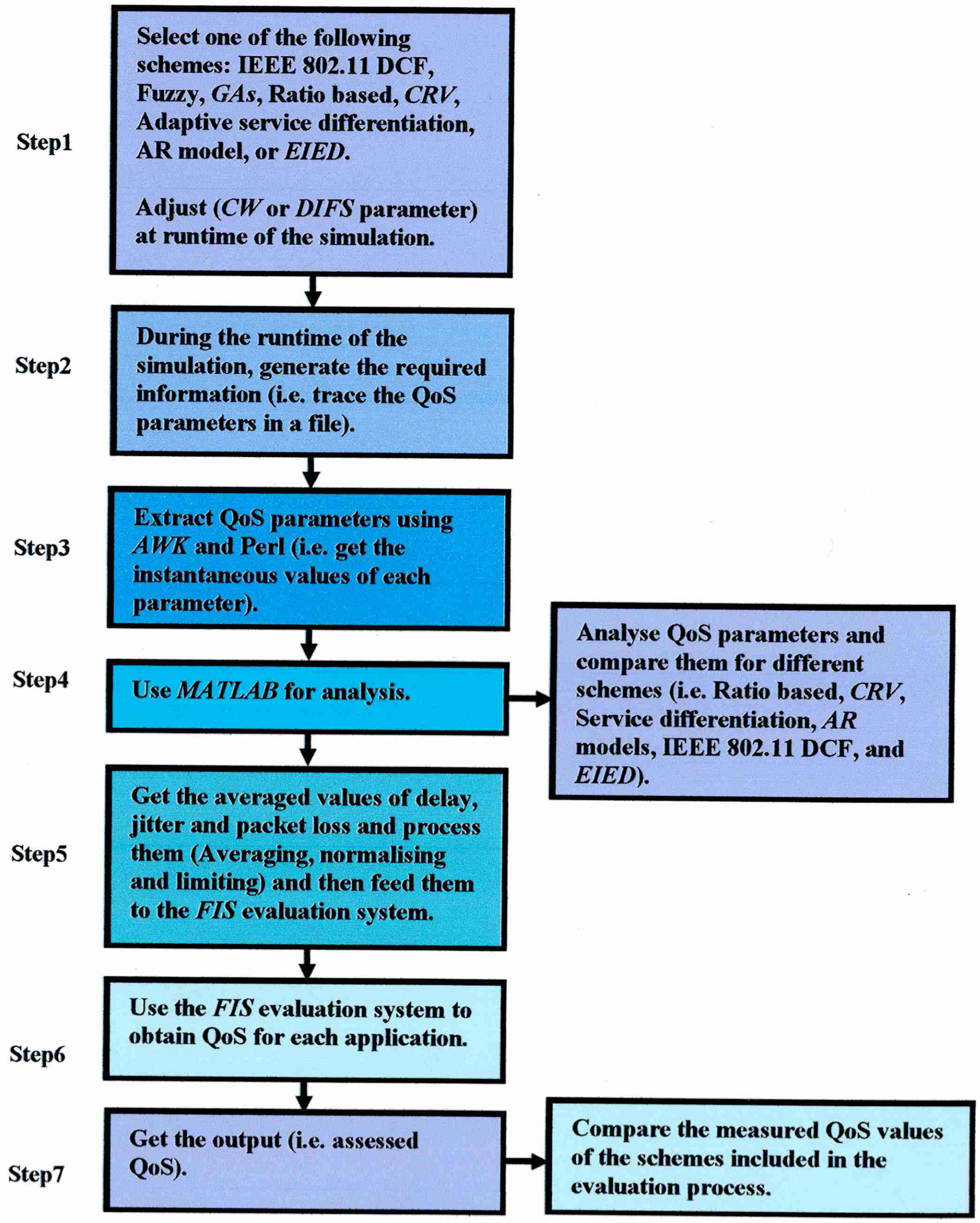



**Figure 4.3:** Measurement process including *FIS* evaluation system.

Delay and jitter were computed by considering three main fields: the sent time, the received time and the packet ID of the successfully received packet. Subtracting the sent time from the received time for the same packet ID gave the delay. Jitter was computed by measuring the absolute value of the delay difference between any two consecutive packets. Throughput was calculated at the receiving station by counting the number of successfully received bytes with respect to the elapsed time. Basically, packet loss was a function of throughput and was calculated by counting the number of sent packets and the number of received packets at the destination. The difference gave the number of lost packets. In this study, packet loss rate was considered which represented the percentage of number of packet lost with respect to the total packet generated by the source. Collision and MAC protocol efficiency were seen to be related. MAC efficiency represented the percentage of total successfully acknowledged packets to the total number of sent packets at the MAC level. Collision rate was obtained by counting the number of retransmitted packets to the total sent packets by MAC protocol.

Once the simultaneous values of the performance metrics were obtained, they were input to *MATLAB* program which split them into a number of blocks based on the selected window size (i.e. gathering every *n* consecutive packets in one window or block) (MATALB, 2004). The values in each block were then averaged for each parameter separately. These values were normalised and limited before they were fed to the Fuzzy Inference System (*FIS*) in order to ensure that all the QoS parameters have the same contribution in the assessment process. The output of the *FIS* was the assessed QoS. To guarantee that the performance comparison between the developed and other schemes was fair, the comparison was carried out for QoS parameters (i.e., without the *FIS* system) and for the assessed QoS (i.e., after the *FIS* system) as shown in Figure 4.3.

## 4.5 Summary

This chapter has described the procedure used to investigate the limitations and the performance of the basic IEEE 802.11 DCF scheme. It outlined the methods used to validate the performance of the proposed approaches and to compare their performance with the basic IEEE 802.11 DCF and *EIED* schemes. Most of this study is based on simulations carried out using *NS-2*; so its features were discussed in section 4.2. The data collected by the simulation are quantitatively analysed to validate the performance of the proposed approaches and compare their performance with the standard IEEE 802.11 DCF scheme. The measurement process was supported by the AWK and Perl scripting language and *MATLAB* since the latter had the ability to provide the fuzzy tool box in addition to its ease of development.

# *CHAPTER 5*

# IEEE 802.11 MAC Performance Analysis

## 5.1 Introduction

IEEE 802.11 is a relatively new standard for wireless networks (IEEE, 1999). Its need started from the many differences between traditional wired and wireless networks and the increased need for interoperability among different manufacturers. In this chapter the limitations and the performance of the Distributed Coordination Function (DCF) within the IEEE 802.11 MAC protocol are reviewed and evaluated.

The limitations of the IEEE 802.11 DCF protocol are discussed in section 5.2 for several scenarios. These include unfairness, the hidden terminal problem, and its use in multi-hop networks. Performance results for the IEEE 802.11 MAC protocol are provided in section 5.3. This section includes the variation of, the number of active stations in the same Independent Basic Service Set (*IBSS*), the minimum *CW* size ($CW_{min}$), the *DIFS* length and finally the variations in the number of retry limits.

## 5.2 Analysis of the IEEE 802.11 DCF Limitations

QoS support in wireless networks is more complicated than in the wired networks since bandwidth is more limited, unfairness, hidden terminal, delay and bit error rate are high and characteristics of the wireless channel vary over time and space. The standard IEEE 802.11 does not provide QoS for the increasing number of multimedia applications (IEEE, 1999). Thus, many studies were carried out to investigate the limitations and the performance of the protocol in order to enhance its performance. This section demonstrates through simulations the limitations of the IEEE 802.11 MAC protocol.

### 5.2.1 Unfairness Problem of IEEE 802.11 MAC Protocol

The unfairness issue in the MAC protocol has a negative impact on the behaviour of higher layer protocols and on multimedia applications such as audio and video that are time-sensitive applications. It can increase delay and reduce throughput.

In this section the results obtained for the unfairness problem experienced in the IEEE 802.11 MAC protocol when transmitting applications using *UDP* transport protocol are provided. These applications used *CBR* and *VBR* traffic types. Also, the impact of this problem on QoS parameters was examined. The effect of this problem was also investigated in both the basic access and the *RTS/CTS* access mechanisms for light and heavy *CBR* traffic.

A simple chain topology with a distance of 250 metres between any two adjacent stations was employed to ensure that every station only reached its nearby station directly as shown in Figure 5.1. This topology provided a suitable multi-hop connectivity to carry out the study. The arrows indicate the direction of transmissions between stations. *CBR* traffic was generated with 512 bytes packet size and two different inter-packet intervals to provide the network with light and heavy loads. For this scenario the channel bandwidth was 1 Mbps. The simulation time was 300 seconds. This period was sufficient to examine the behaviour of the IEEE 802.11 MAC protocol. Other simulation settings are as listed in Table 4.1 (see Chapter 4).

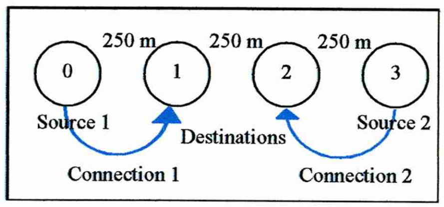



**Figure 5.1:** Unfairness problem with MAC protocol.

#### 5.2.1.1 Unfairness Problem with *CBR* Traffic

The sending stations 0 and 3 in the network of Figure 5.1, started transmission at two different rates: Station 0 started its transmission at the 30th second of the simulation to station 1 with transmission rate equal to 0.4 Mbps. Station 3 initiated its transmission to station 2 at the 90th second of the simulation with transmission rate equal to 0.8 Mbps.

As shown in Figure 5.2, the average throughput of the first connection degraded by 98% during the period (90 to 141 seconds) and by 80 % of its peak value (0.4 Mbps) during the period (141 to 250.5 seconds). The sharp drop of average throughput of the first connection was due to the activation of the second connection at the 90th second.

Here, the IEEE 802.11 backoff algorithm performed poorly. When station 3 gained access to the channel and successfully transmitted its data packets, its backoff timer was reset in order to initiate another transmission by competing with other stations. In contrast, station 0 failed to transmit its data packets because it was deferred by its destination due to the exchange of *RTS* and *CTS* message of the second connection. Therefore, its backoff interval doubled.

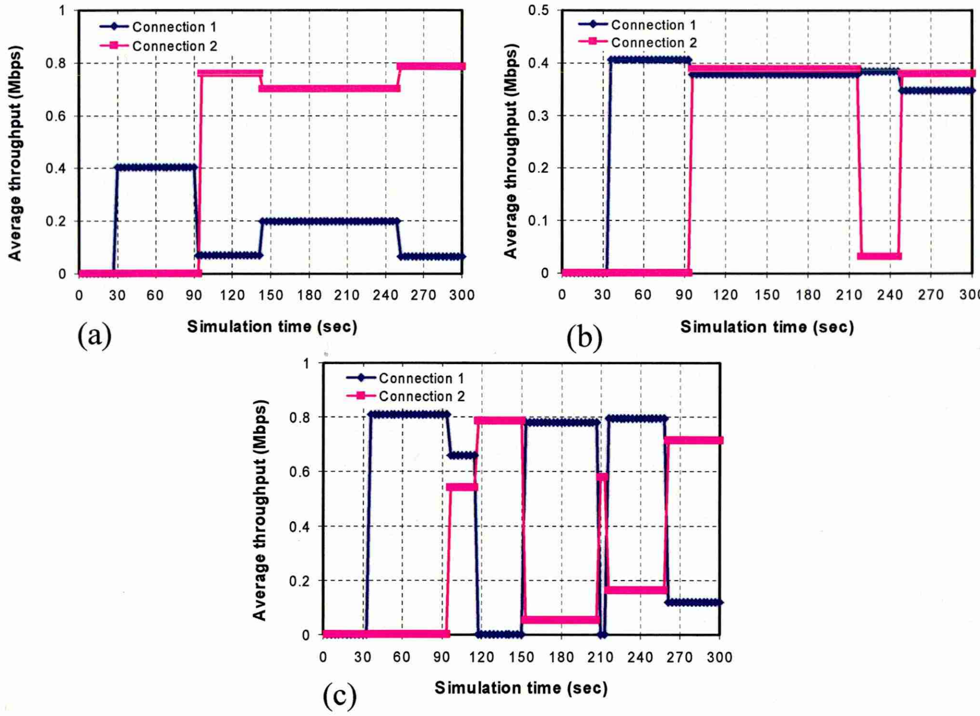



**Figure 5.2:** Unfairness problem of MAC protocol, (a) two different transmission rates (0.4 and 0.8 Mbps), (b) both at low transmission rate (0.4 Mbps) and (c) both at high transmission rate (0.8 Mbps).

Station 3 had a small backoff interval since it successfully transmitted its data packets. Accordingly, station 3 repeatedly gained access to the channel and transmitted its data packets while all the transmission attempts by station 0 failed during the period that station 3 was transmitting. The prevention of transmission from station 0 caused a large increase in its backoff interval. Thus, when station 3 completed its transmission and had no more to send, the timer for station 0 may stay backed off, however during the time the station could still be transmitting. When the two sending stations transmitted with the same transmission rates (0.4 Mbps each), the unfairness problem affected both connections. The impact was less than the case where the two sending stations transmitted at different transmission rates, high (0.8 Mbps) and low (0.4 Mbps). This was because the channel capacity was sufficient to simultaneously serve the two connections without any significant effect on each others transmissions (see Figure 5.2b).

In Figure 5.2b, during the period 216 to 246 seconds of the simulation, the average throughput of the second connection degraded by 93%. This was because the first connection captured the channel continuously after successful transmission. As a result, station 3 failed to transmit to station 2 and this enforced the MAC protocol to report a call back message to the network layer indicating route failure. As a result, station 3 dropped all the buffered data packets. Once the link was established with station 2, station 3 continued its transmission.

The impact of unfairness problem and the scarcity of channel capacity were obvious when the sending stations transmitted at high data rate (0.8 Mbps each) as shown in Figure 5.2c. At the receiving station, the average throughput achieved when there was one connection over the channel was 0.8 Mbps. At the time the second connection was activated, a significant reduction of average throughput took place for both connections. The reduction of average throughput exceeded 22% for each connection. This reduction was due to the shortage of channel bandwidth (being set to 1 Mbps for this scenario), the hidden terminal problem, and the incapability of the MAC protocol to share the channel in a fair manner at heavy loads.

#### 5.2.1.2 Unfairness Problem with *VBR* Traffic

The simulation discussed in section 5.2.1.1 was carried out when the sources transmitted *VBR* traffic. The results obtained showed that the impact of the unfairness problem on the QoS parameters still exists. More information about the scenario and the complete findings is provided in Appendix B (see Appendix B.1)

#### 5.2.1.3 Unfairness Problem with Simple Priority Scheme

In this scenario, the effect of the unfairness problem on the network performance was investigated. A simple priority scheme was proposed to deal with this effect. It was based on specifying different $CW_{min}$ size for each connection. The values were chosen according to several simulations. The results obtained were compared with the findings of the default settings (i.e., fixed $CW_{min}$ size equal 31). Altering the $CW_{min}$ size for the transmitting stations resulted in variations of the backoff intervals for these stations. A large $CW_{min}$ size leads to a longer backoff interval, while a small $CW_{min}$ size directs to a shorter backoff interval.

The simulation was carried out using the settings listed in Table 4.1. Station 0 transmitted *CBR* traffic to station 1 using 1 Mbps load. Station 3 transmitted to station 2 a *VBR* traffic using a 1 Mbps load. The network was fully loaded (i.e. traffic was set to maximum channel capacity 2 Mbps). *CBR* traffic packet size was equal to 1000 bytes. *VBR* traffic had a variable packet size and a variable inter-packet interval. Both MAC protocol mechanisms were used. The simulation time was 300 seconds.

When the basic access mechanism was used, the average throughput of *CBR* and *VBR* traffic degraded by 58% and 15%, respectively from their peak values (1 Mbps) as shown in Figure 5.3a. When the *RTS/CTS* access mechanism was used, the average throughput of *CBR* traffic degraded by 86%, while the *VBR* traffic only degraded by 12%. This was because the IEEE 802.11 backoff mechanism gave priority to the latest successful station which was the *VBR* source. In order to reduce the effect of the unfairness problem in the selected scenario, *CBR* traffic was assigned a small $CW_{min}$ size equal to 15 while *VBR* traffic was specified a large $CW_{min}$ size equal to 127.

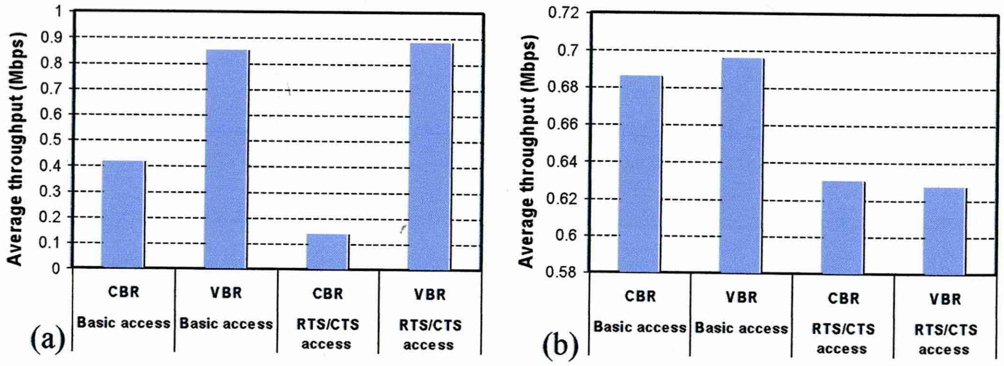



**Figure 5.3**: Average throughput of *CBR* and *VBR* traffic with basic and *RTS/CTS* access mechanisms, (a) average throughput with unfairness problem (without priority scheme), and (b) average throughput with a simple priority scheme.

The results obtained revealed that the simple priority scheme shared the medium equally between the *CBR* and *VBR* connections as depicted in Figure 5.3b. Using the priority scheme, the reduction in the average throughput of *CBR* traffic was equal, 31% and 40%, in the basic and the *RTS/CTS* access mechanisms, respectively. A reduction of 58% and 86% in the average throughput was observed in the basic and *RTS/CTS* access mechanisms, respectively when the default settings of the protocol were used.

### 5.2.2 Hidden Terminal Problem

The aim of this section is to demonstrate the effect of hidden terminal problem on the QoS parameters when transmitting different traffic types. The traffic related to applications such as, audio, video and data. The effect of hidden terminal problem was investigated in both the basic access and the *RTS/CTS* access mechanisms. Moreover, the investigations included the absence and the presence of one hidden terminal and multiple hidden terminals problems[3] (i.e. 6 hidden terminals).

The chain topology shown in Figure 5.4 was used. Two sources (stations 0 and 2) transmitted simultaneously to stations 1 and 3. The arrows represent the direction of data flow e.g. station 0 transmits to station 1 while station 2 transmits to station 3.

[3] In these experiments multiple hidden terminals mean six hidden terminals problems, unless specified.

Three simulations were carried out for *CBR* traffic which modelled audio, video and data. All simulations were carried out with the absence of the hidden terminal problem, with the presence of one hidden terminal and with the presence of multiple hidden terminals problems. With multiple hidden terminals, the network topology shown in Figure 4.2b (see Chapter 4) was used in order to provide multiple communications pairs.

The results with the presence of hidden terminals were compared with the results at the absence of hidden terminals in both MAC protocol access mechanisms. In each experiment, the QoS parameters were evaluated in terms of average throughput, average delay, average jitter, and average packet loss. The packet size was 160 bytes for audio, 1280 bytes for video and 1500 bytes for data. These packet sizes represent the default values of each traffic type. The active stations generated *CBR* traffic with 64 Kbps, 1 Mbps and 960 Kbps for audio, video, and data, respectively. Other simulation parameters were set as presented in Table 4.1.

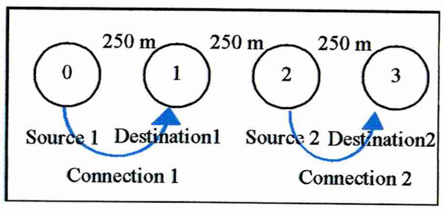



**Figure 5.4:** Chain topology, hidden terminal problem

### 5.2.2.1 Audio Transmission

In this simulation, audio sources started their transmission at the first second of the simulation with a 64 Kbps bit rate for each source. Station 1 was located within the transmission range of transmitters, station 0 and station 2, while the sources were located out side the transmission range of each other. At the time station 0 transmits packets to station 1, collision occurs at station 1. This was because station 0 did not have any knowledge about the ongoing transmission between stations 2 and 3. The average throughput at station 1 remained at 64 Kbps in the presence of one hidden terminal since it was transmitted at low bit rate (i.e. 64 Kbps) and the channel was capable of serving all transmitted traffic. Average throughput degraded slightly by 1.6% in the presence of multiple hidden terminals in both MAC protocol access mechanisms and was still within the desired value of the QoS for audio applications.

The measured values of average delay (i.e., 29 ms) in both MAC protocol access mechanisms were kept within the QoS requirements. The presence of hidden terminals revealed an increase in the measured value of average jitter of the received packets at the destinations. There was no loss of audio packets due to collisions and buffer overflow by the absence of hidden terminal problem. 1.8% and 8% losses were observed by the presence of one hidden terminal and multiple hidden terminals, respectively. The high value of packet loss (i.e. 8%) prevented the QoS for audio to be achieved (Alcatel, 2003).

#### 5.2.2.2 Video Transmission

Each video source transmitted 1 Mbps with 1280 bytes packet size and 10 msec inter-packet interval. For the basic access mechanism, average throughput degraded by 19% when one hidden terminal was present and degraded by 34% when multiple hidden terminals were present. For the *RTS/CTS* access mechanism, average throughput degraded by 27% for one hidden terminal and 40% for multiple hidden terminals (see Table 5.1). The degradation of average throughput in both access mechanisms was due to two reasons: the presence of the hidden terminals and the unfairness of the MAC protocol (when there were multiple transmissions over multi-hop wireless ad-hoc networks). Moreover, for the *RTS/CTS* access mechanism the degradation was due to the exchange of *RTS* and *CTS* control messages prior the transmission of actual data.

Video traffic has strict QoS requirements in terms of delay and jitter. For instance, in the absence of hidden terminals, average delay remained within the QoS requirements recommended by ITU group for video transmission ($\text{ITU}_{(a)}$, 2001). The existence of one hidden terminal caused the average delay to exceed the minimum limit (i.e. 400 msec) by 43% when the basic access mechanism was used and by 60% when the *RTS/CTS* access mechanism was used. The presence of multiple hidden terminals significantly degraded the network performance.

The packet loss rate was also affected by the hidden terminal problems. For the basic access mechanism, it increased from 17.5% at the presence of one hidden terminal to 34.5% when six hidden terminals were present. For the *RTS/CTS* access mechanism, up to 36.9% of packet loss rate was observed at the presence of multiple hidden terminals.

#### 5.2.2.3 Data Transmission

During the last set of simulations, each data source transmitted 960 Kbps. Data transmission is a time-insensitive application. However, it is strictly sensitive to the packet loss parameter ($\text{ITU}_{(a)}$, 2001). Therefore, in order to provide QoS for data transmission, packet loss rate should be 0. The presence of hidden terminals resulted in a large number of packet loss due to collision and buffer overflow in both MAC protocol access mechanisms. Packet loss rate due to buffer overflows was more than 30% which significantly degraded the QoS for data transmission. A summary of the protocol performance for audio, video and data transmission in the presence and absence of the hidden terminal problems is provided in Table 5.1.

**Table 5.1:** QoS parameters measurements when transmitting three *CBR* applications, with the absence and presence of hidden terminal problem.

| Application | Access mechanism | No hidden terminal | | One hidden Terminal | | Six Hidden Terminals | |
|---|---|---|---|---|---|---|---|
| | | Average throughput (Kbps) | Average delay (msec) | Average throughput (Kbps) | Average delay (msec) | Average throughput (Kbps) | Average delay (msec) |
| Voice | Basic | 64 | 1.4 | 64 | 15 | 63 | 7.6 |
| | RTS/CTS | 64 | 2.2 | 64 | 16.2 | 63 | 15 |
| Video | Basic | 1024 | 6 | 810 | 572 | 660 | 1810 |
| | RTS/CTS | 1024 | 6.6 | 730 | 639 | 600 | 1640 |
| Data | Basic | 935 | 6.8 | 830 | 636 | 640 | 1550 |
| | RTS/CTS | 935 | 7.5 | 760 | 717 | 580 | 1250 |
| **Application** | **Access Mechanism** | **No hidden terminal** | | **One hidden Terminal** | | **Six Hidden Terminals** | |
| | | **Average jitter (msec)** | **Packet loss (%)** | **Average jitter (msec)** | **Packet loss (%)** | **Average jitter (msec)** | **Packet loss (%)** |
| Voice | Basic | 0.2 | 0 | 1.4 | 1.9 | 2.03 | 7.8 |
| | RTS/CTS | 0.21 | 0 | 1.8 | 0.6 | 3.2 | 1.1 |
| Video | Basic | 0.21 | 0 | 5.3 | 17.5 | 60 | 34.5 |
| | RTS/CTS | 0.21 | 0 | 6.1 | 25.6 | 47 | 36.9 |
| Data | Basic | 0.22 | 0 | 6.13 | 10.1 | 52 | 29.9 |
| | RTS/CTS | 0.21 | 0 | 6.8 | 17.8 | 31 | 32.5 |

### 5.2.3 IEEE 802.11 MAC Protocol in Multi-hop Ad-hoc Wireless Networks

In this section, the impact of increasing the number of hops on the network performance is investigated. *CBR* and *VBR* traffic transmission were considered when they delivered packets into the network with 100% of channel capacity. The impact of increasing the number of hops was also investigated for the basic and *RTS/CTS* access mechanisms.

In this section, the network shown in Figure 4.1a (see Chapter 4) was used. Station 1 represented the source of data packets and the destination station was defined by the number of hops for different simulation runs. For example at one hop, station 2 was the sink of data packets, and at two hops station 3 was the sink for the transmitted traffic and so on until the last hop in the string (i.e. station 10). The intermediary stations acted as routers of data packets to the intended destination. Two sets of simulations were carried out. In the first, station 1 transmitted *CBR* traffic and in the second, the source transmitted *VBR* traffic. Both were carried out in the basic access and the *RTS/CTS* access mechanisms. For *CBR* traffic the packet size was 1000 bytes, while *VBR* traffic had a variable packet size and a variable inter-packet interval. Other simulation parameters were set as listed in Table 4.1 (see Chapter 4).

In an ad-hoc wireless network, data packets move along a string of intermediary stations toward the destinations. The successive data packets of a single connection interfere with each other as they move down the string, forcing contention in the MAC protocol. The ideal MAC protocol can attain string utilisation as high as 33% (Li et *al.* 2001). In the selected topology of Figure 4.1a (see Chapter 4), where station 1 was the source and station 10 was the final destination. Stations 1 and 2 could not transmit simultaneously, because station 2 could not receive and transmit at the same time (it operates in half duplex mode). Stations 1 and 3 could not transmit simultaneously because station 2 could not accurately hear station 1 if station 3 was sending. Stations 1 and 4 could send simultaneously. This led to a channel utilisation of 33%. Using the *RTS/CTS* access mechanism, station 4's packet transmissions interfered with *RTS* packets sent from station 1 to station 2, preventing station 2 from correctly receiving station 1's *RTS* transmissions or sending the corresponding *CTS*. In this case, station 5 could transmit to its intended destination while station 1 was transmitting to station 2 without interfering with successful reception at station 2. This provided a channel utilisation equal to 25% of the effective channel capacity (i.e. 1.6 Mbps).

Figures 5.5a and 5.5b depict the average throughput and channel utilisation for *CBR* traffic, respectively, when both MAC protocol access mechanisms were used. The maximum average throughput achieved was 1.6 Mbps when the basic access mechanism was used and 1.4 Mbps when the *RTS/CTS* access mechanism was used for one-hop. The minimum average throughput values obtained for 10 hops were 0.247 Mbps and 0.155 Mbps for the basic access and the *RTS/CTS* access mechanisms, respectively. It can also be observed that for the selected scenario the basic access a mechanism achieved better channel utilisation than the *RTS/CTS* access mechanism. For single-hop network, 80% channel utilisation was obtained using the basic access mechanism. This value was 12.5% higher than the value obtained when the *RTS/CTS* access mechanism was used. In the *RTS/CTS* access mechanism, the intermediate station was incapable of forwarding the receiving packets due to the exchange of *RTS* and *CTS* messages which caused buffer overflows.

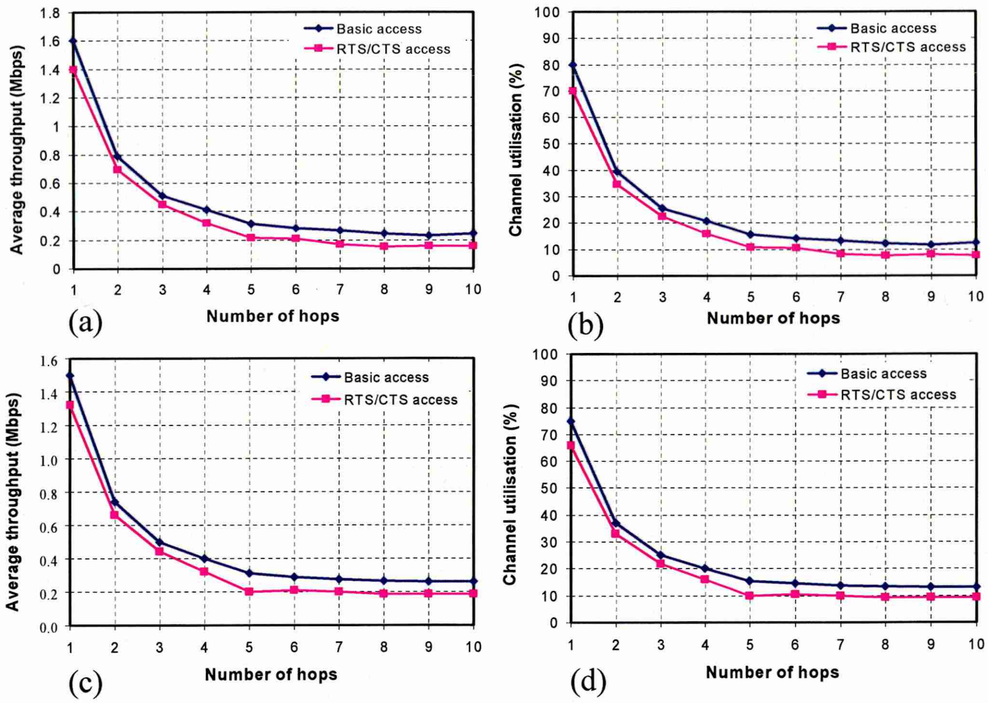



**Figure 5.5**: QoS parameters vs. number of hops for *CBR* and *VBR* traffic in both MAC protocol access mechanisms, (a) average throughput vs. number of hops for *CBR* traffic, (b) channel utilisation vs. number of hops for *CBR* traffic, (c) average throughput vs. number of hops for *VBR* traffic, and (d) channel utilisation vs. number of hops for *VBR* traffic.

When *VBR* traffic was transmitted, the achieved throughput and channel utilisation were as presented in Figures 5.5c and 5.5d. The peak values of average throughput were achieved for a one-hop network. In a 10-hop network, less than 13% of the transmitted data packets were successfully received in both MAC protocol access mechanisms. This implied that the transmission over multi-hop networks was demanding particularly for multimedia transmission, since it resulted in a large QoS degradation for *CBR* and *VBR* traffic in both MAC protocol access mechanisms.

Figures 5.6a, 5.6b, 5.6c, and 5.6d show that average delay and jitter were also large when the number of hops was increased. This was because the packets took longer to be sent crossing through number of intermediary stations.

As the number of nodes was increased, the average throughput for *CBR* and *VBR* traffic degraded to 0.25 Mbps and 0.16 Mbps for both MAC protocol access mechanisms, respectively. The degradation was due to the high drops in each intermediary station in the path of the communication parties.

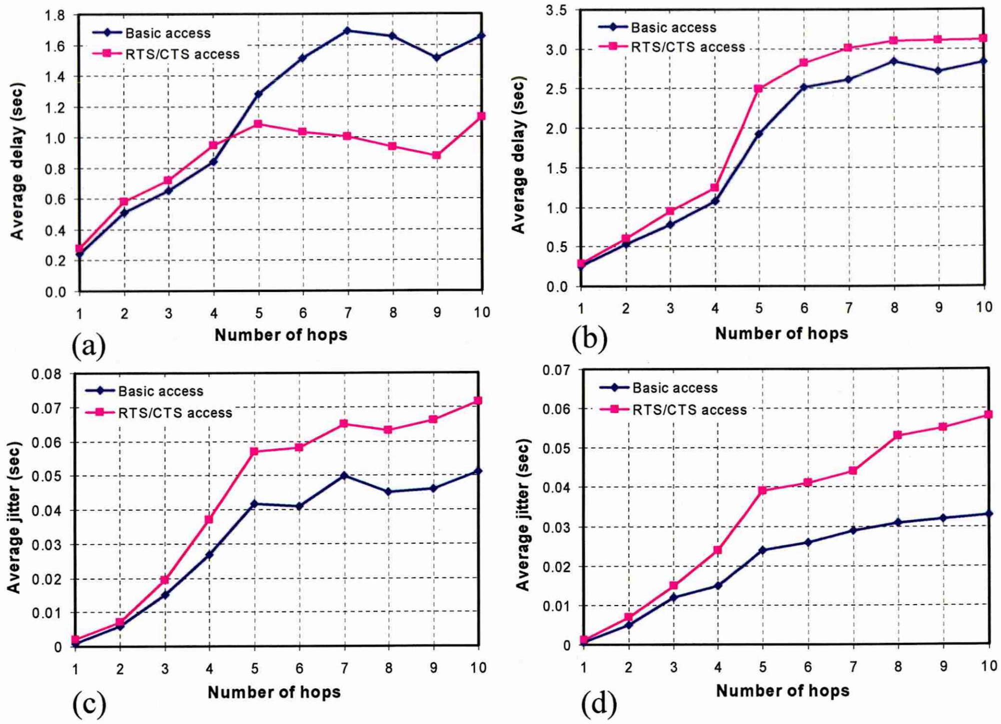



**Figure 5.6**: QoS parameters vs. number of hops for *CBR* and *VBR* traffics in both MAC protocol access mechanisms, (a) average delay vs. number of hops for *CBR* traffic, (b) average delay vs. number of hops for *VBR* traffic, (c) average jitter vs. number of hops for *CBR* traffic, and (d) average jitter vs. number of hops for *VBR* traffic.

As previously discussed, in multi-hop networks, the minimum expected channel utilisation has to be around 25% of the peak value that was achieved when only one pair of communication used (i.e., 25% of the effective channel capacity). The channel utilisation values obtained in these simulations were less than 25%. The minimum channel utilisation was less than 11% and 13% for *CBR* and *VBR* traffic, respectively. A number of simulations were carried out to explain the cause of this reduction with the increase of number of hops. Through these experiments, throughput was measured as a function of offered load. These findings agreed with the results obtained by (Li et *al.*, 2001). The *CSMA/CA* mechanism in the IEEE 802.11 protocol can schedule data packets successfully at light load. This was apparent through the linear relationship between the offered load and the average throughput for both the *CBR* and *VBR* traffic as shown in Figures 5.7a and 5.7b.

In Figures 5.7a and 5.7b, when the offered load was equal to 0.35 and 0.275 Mbps, the network transmitted and received data packets successfully with fewer packet drops in both MAC protocol access mechanisms and for *CBR* and *VBR* traffic. When the offered load exceeded these values (i.e. 0.35 Mbps and 0.275 Mbps) average throughput degraded as shown in Figures 5.7a and 5.7b.

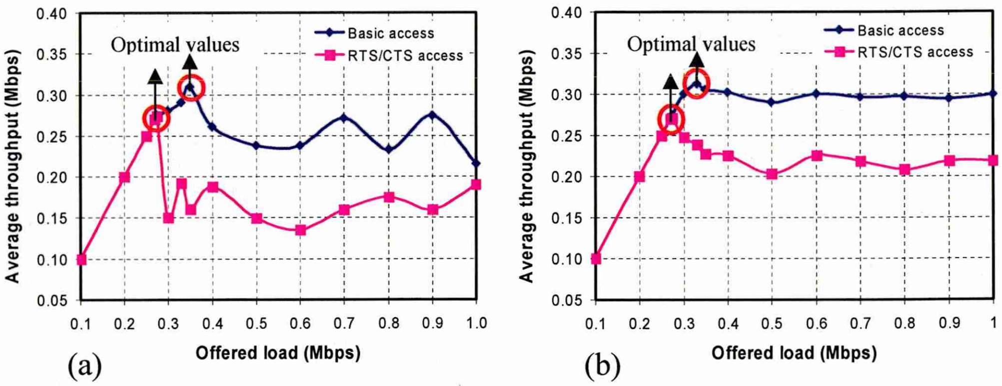



**Figure 5.7:** Average throughput vs. Offered load on an 8 hops string, (a) average throughput for *CBR* traffic, and (b) average throughput for *VBR* traffic.

In this scenario the basic access mechanism achieved better throughput and better channel utilisation for the *CBR* and *VBR* traffic than the *RTS/CTS* access mechanism. This was because the basic mechanism resulted in less overhead than the *RTS/CTS* access mechanism.

It can be concluded that the standard IEEE 802.11 DCF scheme is incapable of performing well in multi-hop networks with the default settings particularly when the network is congested. Therefore, new MAC mechanisms are required to improve network performance and to provide service differentiation in single and multi-hop networks. This will be presented and evaluated in Chapters 8 and 9.

## 5.3 IEEE 802.11 MAC Protocol Performance

In this section a performance evaluation of the legacy IEEE 802.11 DCF scheme is carried out. Through extensive simulations the performance of the IEEE 802.11 DCF scheme when, the number of stations, $CW_{min}$ size, *DIFS* length, and the number of retry limits are varied is analysed. Further, the impact of this variation on the QoS parameters is determined. A comprehensive comparison of the access methods provided by the IEEE 802.11 MAC protocol is carried out and suggestions are made as to when each should be employed.

### 5.3.1 Varying Number of Active Stations on the Network Performance

The aim of this section is to investigate the impact of increasing the number of active stations and data rate on QoS parameters. The performance of MAC protocol access mechanisms for *CBR* and *VBR* traffics is also analysed. The performance of the IEEE 802.11 MAC protocol was investigated when the number of active stations in the same *IBSS* was increased. Two different channel data rates were chosen; a low data rate equal to 2 Mbps and a high data rate equal to 11 Mbps. Control frames were transmitted at 1 Mbps to ensure that all the stations in the interference range could overhear the control frames clearly (Li et *al.*, 2005). The IEEE 802.11b standard was used since it offers multi data rates (IEEE, 1999). The protocol parameter settings were as shown in Table 4.1 (see Chapter 4) and the network topology shown in Figure 4.2d (see Chapter 4) was used. The network load was 100% of channel capacity for each simulation. Each connection was specified as a "source – destination" pair in which the number of connections was varied for each simulation. The simulations were carried out for both *CBR* and *VBR* traffic and both MAC protocol access mechanisms. The *CBR* traffic packet size was 1280 bytes while the *VBR* traffic packet size was variable (mean packet size 3993 bytes with standard deviation equal 2541).

Simulations were repeated 10 times in order to avoid the bias of random number generation. The results of the 10 simulations were averaged to determine the general behaviour of the network. Each simulation was run for a duration of 300 seconds.

Figures 5.8a and 5.8b show the relationship between the active stations and the channel utilisation (channel utilisation is the ratio of the received bits to the channel data rate). When the number of active stations was increased, the channel utilisation decreased slightly for the case of *RTS/CTS* access mechanism compared with the basic access mechanism. In the *RTS/CTS* access mechanism, collisions only involved control frames which were relatively small in size; hence the bandwidth wasted in collisions was less than the basic access mechanism. This explained the slight rate of decrease in the channel utilisation curve for the *RTS/CTS* access mechanism.

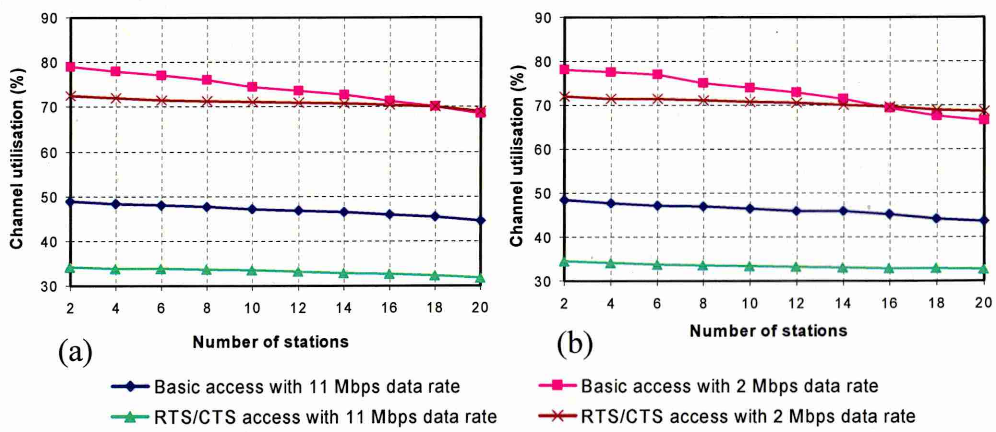



**Figure 5.8:** Channel utilisation vs. number of stations for two different data rates and for both MAC protocol access mechanisms, (a) channel utilisation for *CBR* (b) channel utilisation for *VBR*.

The impact of RTS and CTS overhead on the average throughput became very small when data packet sizes were very large (above 2000 bytes) as shown in Figure 5.9. At small packet sizes, the basic access mechanism outperformed the *RTS/CTS* access mechanism, while at large packet sizes, the *RTS/CTS* access mechanism outperformed the basic access mechanism.

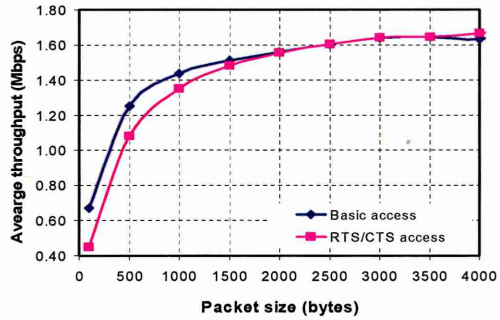



**Figure 5.9:** Average throughput vs. packet size at basic and *RTS/CTS* access mechanisms.

The packet delay from end-to-end should not exceed 400 ms for time sensitive applications in order to achieve the required QoS (Coverdate, 2000). As shown in Figures 5.10a, 5.10b, 5.10c and 5.10d, at a low data rate (2Mbps), both MAC access mechanisms did not meet this QoS requirement when the number of active stations was increased to more than 4. High data rates achieved better performance than low in terms of delay and average delay also slightly increased. This met the QoS requirements for multimedia transmission up to 10 stations and then started to exceed the limit as the number of active stations was increased.

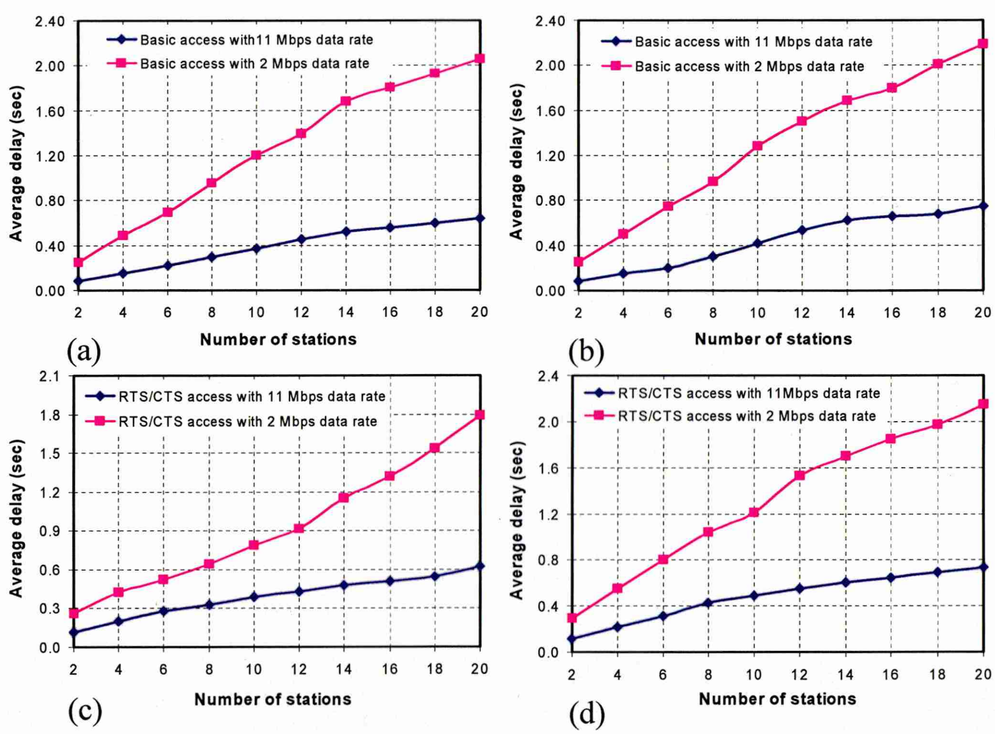



**Figure 5.10:** Average delay vs. number of stations, (a) *CBR* using basic access, (b) *VBR* using basic access, (c) *CBR* using *RTS/CTS* access, and (d) *VBR* using *RTS/CTS* access.

For high data rate, with the basic access mechanism, the average delay was reduced by 69% and 66% compared to low data rate for *CBR* and *VBR* traffic, respectively. When the *RTS/CTS* access mechanism was considered, average delay at high data rate was also reduced by 58% and 63% for *CBR* and *VBR* traffic, respectively.

The values of average delay in both MAC access mechanisms were more than the desired range for QoS (i.e., 150 ms for high QoS and 400 ms for the minimum limit) when the low data rate was used (Coverdate, 2000). Conversely, high data rate could provide acceptable QoS requirements in terms of average delay. One of the major roles of QoS is to keep delay, jitter and packet loss within the QoS requirements of the transmitted traffic (Coverdate, 2000). For instance, to achieve high QoS for multimedia applications, average jitter should not exceed 20 msec. The average jitter increased as the number of active stations increased. In other words, as the number of stations increased; the probability of collisions increased due to a high degree of competition between stations. This forced the MAC protocol to retransmit the collided packets. When the collided packets were successfully received, they experienced large jitter; this variation depended on the number of retransmitted packets.

As shown in Figures 5.11a, 5.11b, 5.11c and 5.11d, the transmission of data packets with the high data rate had a positive impact on the achieved value of average jitter.

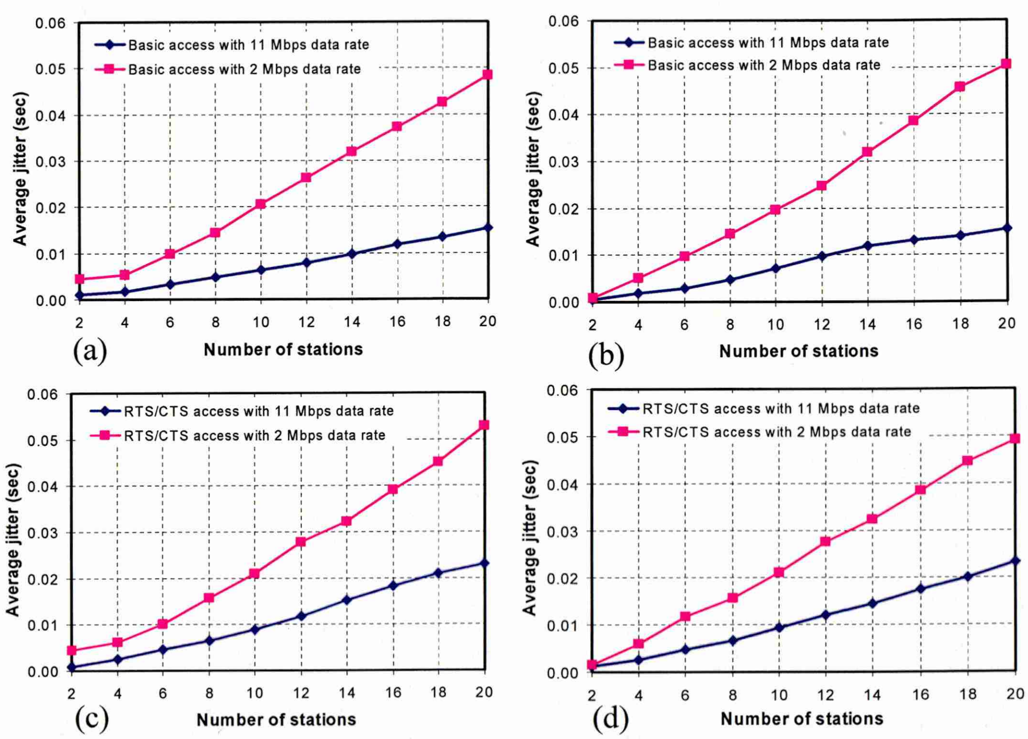



**Figure 5.11:** Average jitter vs. number of stations, (a) *CBR* using basic access, (b) *VBR* using basic access, (c) *CBR* using *RTS/CTS* access, and (d) *VBR* using *RTS/CTS* access.

A high data rate resulted in small average jitter values. This was because the transmission time of data packets at the high data rate was smaller. The results obtained at 11 Mbps indicated that the values of average jitter for *CBR* and *VBR* traffic in both MAC access mechanisms were kept within the acceptable range of QoS (less than 20 ms); whereas, the low data rate resulted in large values for average jitter.

Figure 5.12 shows the packet loss as a function of number of stations for high and low data rates. High data rates (11 Mbps) resulted in larger packet loss for both *CBR* and *VBR* traffic and for both MAC protocol access mechanisms. For the basic access mechanism and as shown in Figure 5.12, packet loss rate for *CBR* and *VBR* was 30% and 23% larger than the values obtained for low data rates (2 Mbps). When the *RTS/CTS* access mechanism was employed, the packet loss ratio was larger by 36% and 35% than the values obtained for low data rates for *CBR* and *VBR* traffic, respectively.

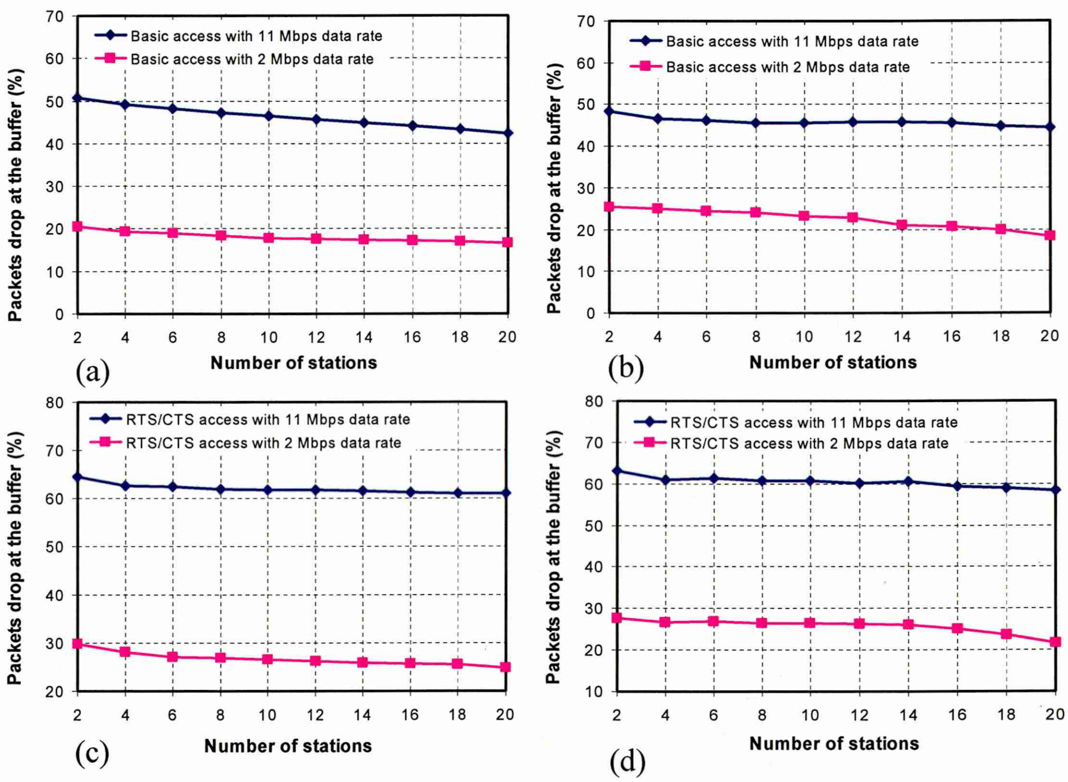



**Figure 5.12:** Packet drops using the buffer vs. number of stations, (a) *CBR* using basic access, (b) *VBR* using basic access, (c) *CBR* using *RTS/CTS* access, and (d) *VBR* using *RTS/CTS* access.

In this scenario, the basic access mechanism outperformed the *RTS/CTS* access mechanism in terms of packet loss when the number of stations was small. For large networks, the *RTS/CTS* access mechanism outperformed the basic access mechanism because of collisions. With the *RTS/CTS* access mechanism, collisions occurred in control frames while in the basic access mechanism collisions occurred in data packets.

### 5.3.2 Impact of Varying MAC Protocol Transmission Parameters

The performance of IEEE 802.11 DCF scheme can be improved and optimised by considering some of IEEE 802.11 DCF tuneable parameters (Gast, 2002). In this section, the following three parameters are investigated: the minimum Contention Window ($CW_{min}$) size, Distributed Inter Frame Space (*DIFS*) length and the number of retry limits. The effects of these parameters are investigated in terms of QoS parameters such as delay, jitter, throughput, and packet loss. The aim of studying the impact of these MAC protocol transmission parameters is to develop adaptive schemes that can be used to dynamically optimise their values. In the following section the effect of $CW_{min}$ is examined for various network sizes and different packet sizes for two traffic types.

#### 5.3.2.1 Varying the Minimum Contention Window ($CW_{min}$) Size

In the standard IEEE 802.11 DCF protocol, with successful transmission, the backoff algorithm reduces the contention window size to $CW_{min}$. Conversely, it is doubled if collision occurs until it reaches its maximum limit (i.e. $CW_{max}$). The magnitude of $CW_{min}$ size has a significant impact on the network performance. Large values of $CW_{min}$ may lead to a long delay of data packets which may lead to high drops at the buffer and long delays and small values may cause a high number of collisions. Therefore, controlling the $CW_{min}$ size may improve the network performance.

The simulations outlined in this section are to investigate the impact of varying the $CW_{min}$ size on the network performance, in particular on the QoS parameters. Including the number of stations and the packet size may critically help in determining the optimal $CW_{min}$ size. Since small network and small packet size may require a small value of $CW_{min}$; whereas, large network and large packet size may require a large value of $CW_{min}$. The applications type such as *CBR* and *VBR* also has an influence on determining the optimal $CW_{min}$ size. Consequently, the three factors were investigated with the variation of $CW_{min}$ size.

##### *5.3.2.1.1 Impact of Varying the $CW_{min}$ size with VBR Traffic*

Four *VBR* connections, with inter-packet interval equal to 0.89 second and standard deviation equal to 0.56 second, (all within the transmission range of each other) in the network shown in Figure 4.2d (see Chapter 4) provided the network with its load. Each connection was specified as a source - destination pair. Each source was associated with a *VBR* traffic generator, which transmitted packets at a fixed packet size and variable interval. The simulations were carried out for different packet sizes (500 bytes, 900 bytes and 1400 bytes). The reason different packet sizes were selected was that the optimal $CW_{min}$ for small packet sizes should have a small $CW_{min}$ and for large packet sizes should have a larger $CW_{min}$. The channel capacity was 2 Mbps. Other simulation settings are as presented in Table 4.1 (see Chapter4).

Figure 5.13a shows the relationship between the average throughput and the $CW_{min}$ size. The simulations were carried out with packet sizes equal to 500, 900, 1400 bytes. Small values of $CW_{min}$ resulted in small average throughput for all packet sizes. For example, when $CW_{min}$ size was 15, for a packet size equal to 500 bytes, the average throughput was 1.19 Mbps. At large value of $CW_{min}$, for instance when $CW_{min}$ was equal to 255 and packet size was equal to 500 bytes, the average throughput was 30% less than its peak value at the optimal $CW_{min}$ size (i.e. 47).

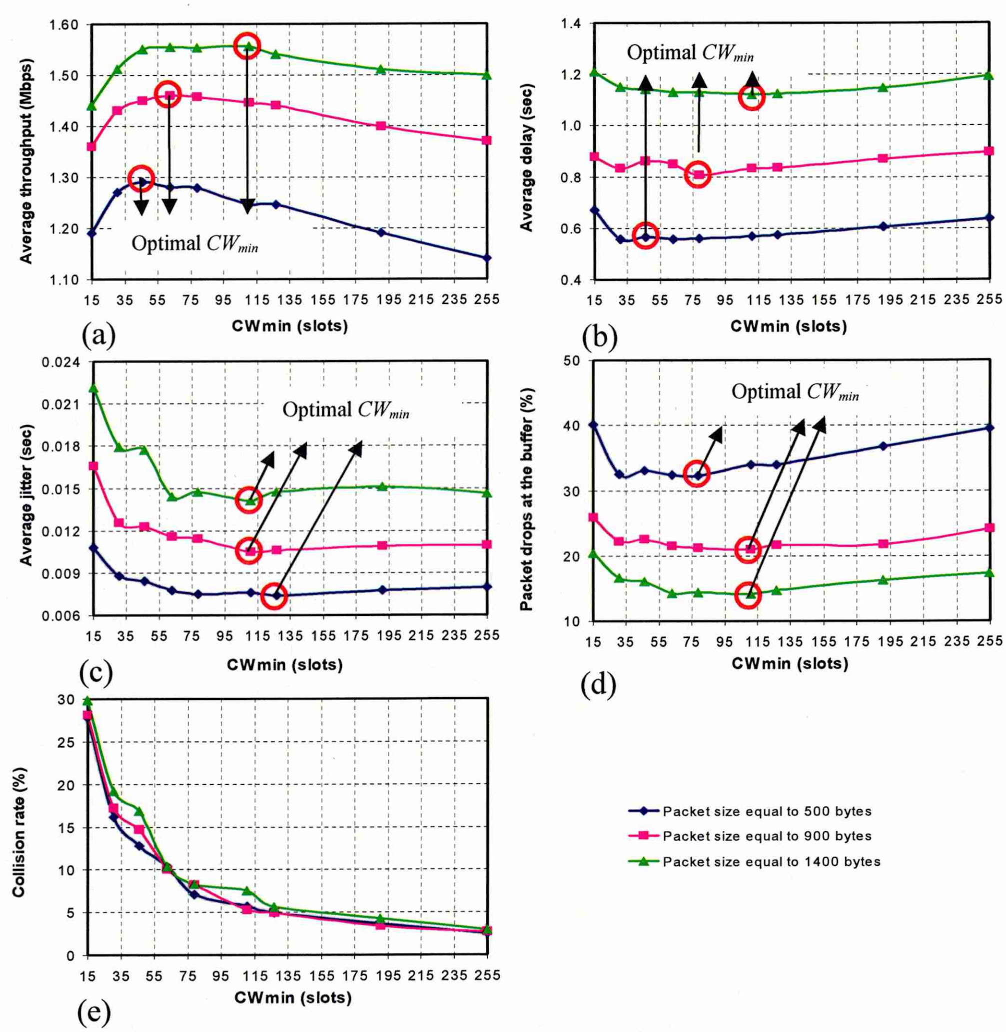



**Figure 5.13:** QoS parameters vs. $CW_{min}$ on the for *VBR* traffic (different packet sizes), (a) average throughput, (b) average delay, (c) average jitter, (d) average packet drops at the buffer, and (e) average collision rate.

High values of delay and jitter were observed at small $CW_{min}$ sizes for all packet sizes as depicted in Figure 5.13b. For small values of $CW_{min}$, high collisions over the network took place. The cause of collisions was due to inability of MAC protocol to serve all generated data packets within a short period of time (small number of time slots).

Regarding average delay, an increase of $CW_{min}$ size resulted in a gradual decrease in the average delay as shown in Figure 5.13b. Smallest values of average delay were obtained for the following $CW_{min}$ values: For a packet size equal to 500 bytes, the optimal value of $CW_{min}$ was 63 (i.e., average delay equal 570.5 ms). For packet size equal to 900 bytes, the optimal $CW_{min}$ was 79 (average delay 878.1 ms) and for packet size equal to 1400 bytes the optimal value of $CW_{min}$ was equal 111 (average delay 1.12 second). Any further increase of the $CW_{min}$ value above the optimal values led to an increase in average delay.

Small values of average jitter were obtained for $CW_{min}$ equal to 111 for packet sizes 900, 1400 bytes, while the smallest value of average jitter for a packet size equal to 500 byte was obtained when $CW_{min}$ size was 127. As shown in Figure 5.13c, for $CW_{min}$ equal to 15, large values of average jitter were obtained for packet sizes equal to 500, 900, 1400 bytes. At small values of $CW_{min}$, high collisions between data packets took place.

Data packets that were dropped due to collisions and due to buffer overflow were significantly affected by the variation of the $CW_{min}$ size as shown in Figures 5.13d and 5.13e. A small $CW_{min}$ size resulted in large packet drops due to collisions. This was because; the deferring period of each active station was very short. As a result, the possibility that two or more stations accessed the channel at the same time was very high. This simultaneous transmission increased the number of collisions which led to a large packet drop. In contrast, large values of $CW_{min}$ resulted in small collisions. This was because each active station was deferred for a sufficient period of time before transmitting its data packet. Very small and large values of $CW_{min}$ resulted in large packet drops due to buffer overflow. At small values of $CW_{min}$, the data packets drop due to collisions was very large. Accordingly, the MAC protocol was busy from retransmission of the collided data packets, which resulted in a long defer of data packets at the queue. When the buffer exceeded its maximum limit (it was set to the default value of 50 packets during this experiment) the source will simply drop all the incoming packets until space is available at the buffer or all packets passed down to

MAC protocol. At large values of $CW_{min}$, the data packets had to wait for a long period of time at the queue to be served by MAC protocol. This long defer increased the total number of data packet drops.

The relationship between the data packet drop due to buffer overflow and $CW_{min}$ is shown in Figure 5.13d. The simulations were repeated with packet sizes equal to 500, 900, and 1400 bytes. It was observed in all cases that packet drop gradually reduced to smallest as the $CW_{min}$ size increased. Thereafter any further increase in $CW_{min}$ increased the packet drop rate. The reason the curves followed the same trend was that very small values of $CW_{min}$ resulted in more collisions and very large values of $CW_{min}$ resulted in a high waiting time for the stations to transmit. A summary of the QoS parameters obtained is provided in Table B.1 (see Appendix B.2).

##### *5.3.2.1.2 Impact of Varying the $CW_{min}$ size with Different Network Sizes*

In this investigation, a network with 5 and 20 connections, based on the topology shown in Figure 4.2d (see Chapter 4) was simulated. The active sources transmitted *CBR* traffic with full load (i.e. 100% of channel capacity).

Figure 5.14 examines the dependency of the QoS parameters on the $CW_{min}$. The figures demonstrate a small network size consisting of 5 connections and a large network size consisting of 20 connections. Figures 5.14a and 5.14b show average delay and jitter against $CW_{min}$ for small and large network sizes, respectively (i.e. 5 and 20 connections). Large values of average delay were observed for both small and large $CW_{min}$ sizes in the small network. For small $CW_{min}$ size, collision was the main cause of higher values of average delay and jitter, since collision imposed extra overhead on the network by retransmission of the collided packet. As the $CW_{min}$ size increased, average delay and jitter reduced to a minimum average delay at $CW_{min}$ size equal to 47 and minimum average jitter at $CW_{min}$ size equal to 127. Afterwards, average delay increased gradually with the increase of $CW_{min}$ above the optimal value. This was because of increasing idle time slots at large $CW_{min}$ sizes. Average jitter was not greatly affected by the increase of $CW_{min}$ size above 47 due to the reduction in the retransmission of data packets.

In large network scenarios, although a $CW_{min}$ value less than 47 resulted in small values for average delay, they still caused a serious reduction in average throughput, considerable increase in packet loss rate and large number of collisions as shown in

Figure 5.14d and 5.14f. Figures 5.14c and 5.14d show that average throughput increased and packet loss rate decreased as $CW_{min}$ size was increased. The effect is justifiable since an increase in $CW_{min}$ causes the number of collisions to decrease and the system throughput to become larger and packet loss smaller.

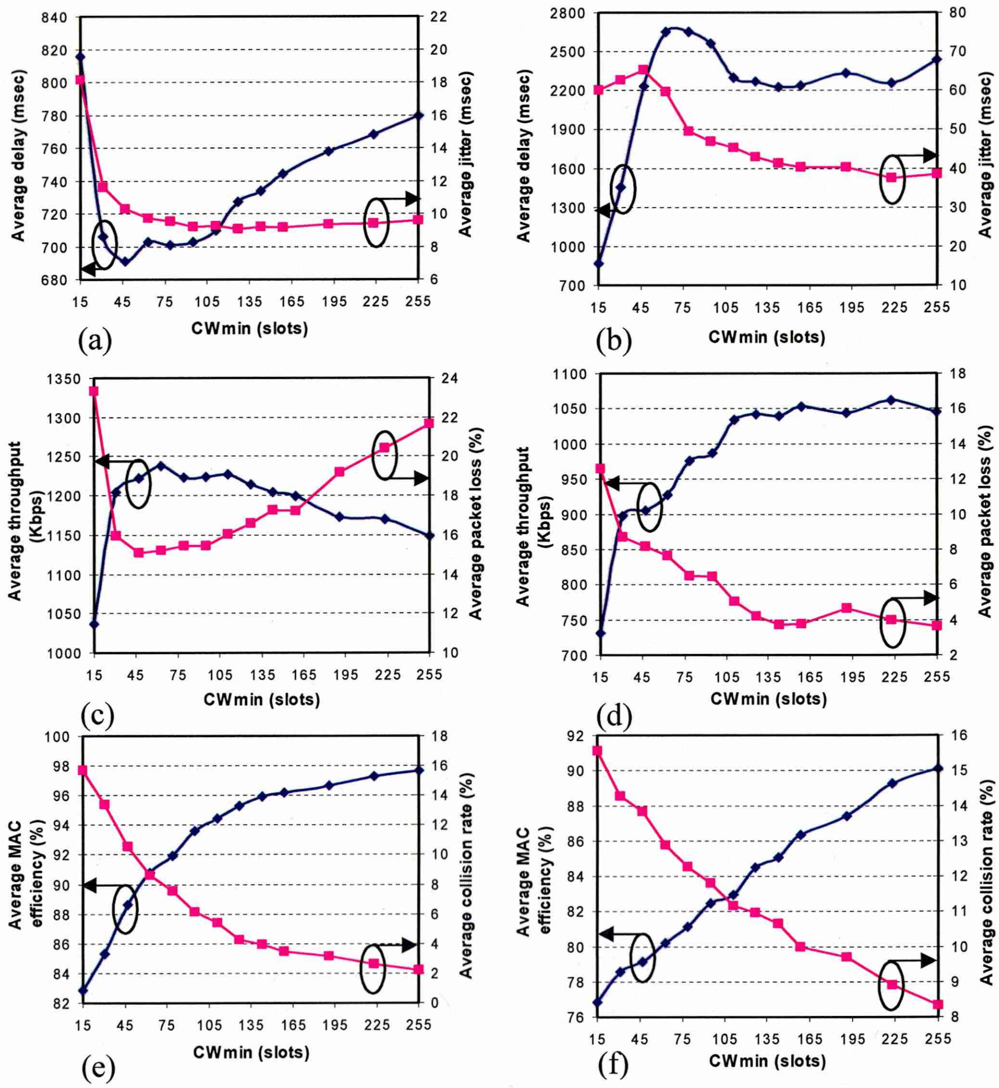



**Figure 5.14**: QoS parameters vs. $CW_{min}$ for two network sizes, (a) average delay and average jitter for 5 connections, (b) average delay and average jitter for 20 connections, (c) average throughput and average packet loss rate for 5 connections, (d) average throughput and average packet loss for 20 connections, (e) average MAC efficiency and average collision rate for 5 connections, and (f) average MAC efficiency and average collision rate for 20 connections.

As shown in Figures 5.14e and 5.14f, large values of $CW_{min}$ caused a reduction in the number of collisions and improved the protocol efficiency for small and large network sizes, since the possibility of simultaneous transmission was very low.

Figure 5.14 indicates that the optimal $CW_{min}$ size was different for different QoS parameters in the same scenario and for various network sizes. It also confirms that a very small $CW_{min}$ size was not effective for large networks due to the increased number of collisions. Conversely, a large $CW_{min}$ size was inappropriate for a small network size due to many idle slots.

#### *5.3.2.1.3 Impact of $CW_{min}$ in the Basic Access and RTS/CTS Access Mechanisms*

In this simulation the topology shown in Figure 4.2d (see Chapter 4) was used when 8 stations transmitted *CBR* traffic. The active sources delivered data packets into the network with full load (i.e. 100% of channel capacity).

Figure 5.15 shows the impact of $CW_{min}$ for both MAC protocol access mechanisms. For the *RTS/CTS* access mechanism, the maximum value of average throughput was achieved at a $CW_{min}$ equal to 47, while the minimum value was achieved at a $CW_{min}$ equal to 255 as shown in Figure 5.15a. The average throughput achieved by the basic access mechanism was 14.5% higher than the *RTS/CTS* access mechanism for a $CW_{min}$ equal to 79. This improvement was due to the *RTS/CTS* access mechanism requiring exchange the *RTS* and *CTS* frames prior to the transmission of data packets.

For the basic access mechanism, small values of $CW_{min}$ resulted in high values of average delay and average jitter. This was due to an increase in the number of collisions. As the $CW_{min}$ size was increased, the probability of collisions decreased resulting in smaller values of average delay and jitter. A further increase in the $CW_{min}$ size caused longer defers of data packets at the buffer which eventually increased average delay. The long defer of data packets increased the delay, therefore, small values of $CW_{min}$ caused a large number of collisions and large $CW_{min}$ sizes led to high drops at the buffer. As a result of this, high values of average delay were observed. When $CW_{min}$ was equal to 63, the optimal value of average delay was obtained.

For the *RTS/CTS* access mechanism, as the value of $CW_{min}$ increased the collision probability in control frames decreased. This in turn reduced the number of exchanged *RTS* and *CTS* messages and resulting in a smaller average delay. In contrast, a large $CW_{min}$ size increased the waiting time for data packets at the buffer and increased the number of idle slots that finally led to higher delay. The overhead of *RTS* and *CTS* messages added extra waiting time for the packets at the buffer, which also enlarged the average delay. The Smallest value of average delay was obtained for a $CW_{min}$ size equal to 63 in the basic access and the *RTS/CTS* access mechanisms as shown in Figure 5.15b.

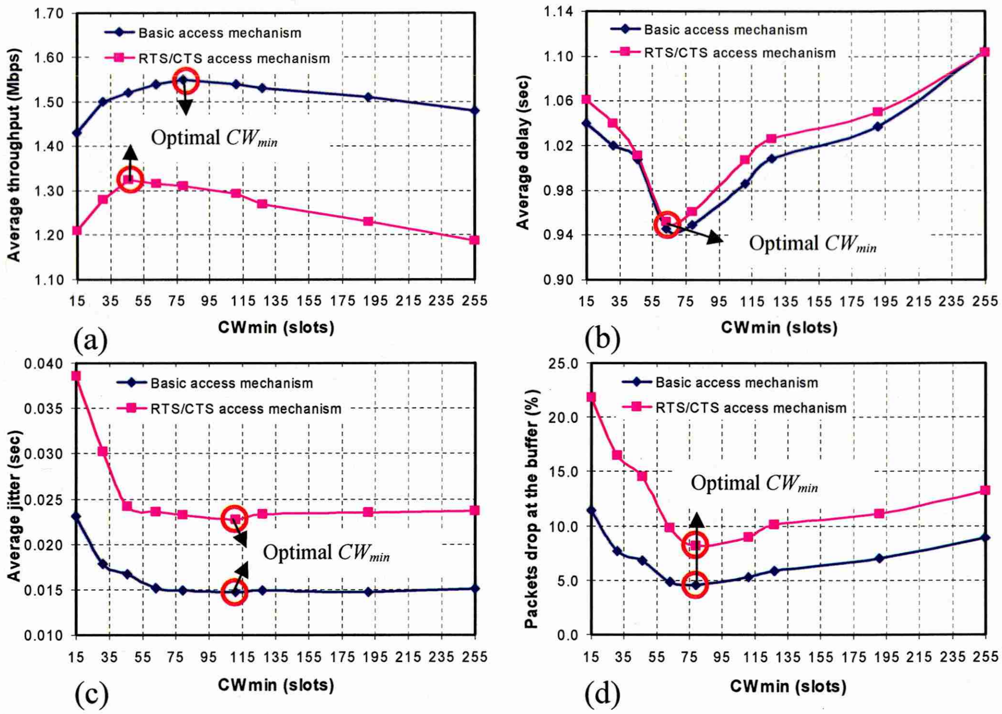



**Figure 5.15**: Impact of $CW_{min}$ on both the basic access and the *RTS/CTS* access mechanisms, (a) average throughput, (b) average delay, (c) average jitter, and (d) average packet drops at the buffer.

Large average jitter values were observed for small values of $CW_{min}$, since there were a high number of collisions in both access mechanisms. As the $CW_{min}$ size increased, the number of collisions became less, which in turn reduced the variation in time of the received packets at the destinations. Hence after, the average jitter was decreased gradually to have its smallest value at $CW_{min}$ equal to 111 in both MAC protocol access mechanisms as shown in Figure 5.15c. Any further increase in the $CW_{min}$ size above 111 did not significantly affect jitter in both access mechanisms. Smaller values of average jitter were obtained when the basic access mechanism was used. A reduction by 35% in average jitter was observed compared to the values obtained when the *RTS/CTS* access mechanism was employed. The difference was due to the exchange of control messages prior to the transmission of data packets.

Data packets drop was affected by the fluctuation of the $CW_{min}$ size above and below the $CW_{min}$ optimal value (i.e. 79) as shown in Figure 5.15d. For small values of $CW_{min}$, stations were deferred for a short period of time resulting in a large number of collisions. The large number of collisions led to large drops at the buffer and to a busy medium for retransmissions of collided packets. The exchange of *RTS* and *CTS* messages caused larger drops at the buffer as compared to the basic access mechanism. This was because in the *RTS/CTS* access mechanism the transmission of each packet was preceded by the exchange of *RTS* and *CTS* control messages. Moreover, failure in exchanging these control messages forced the MAC protocol to retransmit them again. This delayed the actual data packets at the buffer resulting in high drops at the queue.

#### 5.3.2.2 Impact of *DIFS* on the QoS parameters

In this section, the network topology shown in Figure 4.2d (see Chapter 4) was used. Two network sizes were adopted for the simulations. These were: a small network with 5 connections and a large network with 20 connections. Each data source transmitted *CBR* traffic with a packet size equal to 512 bytes. The network was loaded with 80% of the channel capacity (i.e. $0.80*2000 Kbps = 1600 Kbps$).

The $CW_{min}$ size was kept fixed at 31 slots in these scenarios and other MAC and *PHY* parameters are as provided in Table 4.1 (see Chapter 4). The *DIFS* values were varied over the range 20 to 100 $\mu$sec with an increment of 10 $\mu$sec. In order to achieve accurate results and to avoid the bias of random number generation, the simulation time was 300 seconds and each simulation was repeated 10 times with average values used.

This section demonstrates the results obtained through *DIFS* variation. Figure 5.16 shows the impact of the variation of *DIFS* on the QoS parameters, delay, jitter, throughput, packet loss and collision in small and large networks.

Figures 5.16a and 5.16b show the relationship between average delay and jitter with a variation in the *DIFS* value. Average delay and jitter increased with an increase in the *DIFS* values for both small and large networks. Consequently, larger values of *DIFS* caused the delay and jitter to increase. Before packet transmission, each station is required to wait for a *DIFS* when it senses an idle channel. Thus, a shorter *DIFS* is preferable in order to enable the contending station to count down their backoff timer sooner and to gain more advantages to access the channel. This is achieved when the number of active stations is small. When the number of contending stations is large, the probability of collisions is higher leading to performance degradation.

Average throughput was also affected by the variation of the *DIFS* values as shown in Figure 5.16c. For instance, average throughput degraded from 1245 Kbps at *DIFS* equal to 20 $\mu$sec to 1132 Kbps at *DIFS* equal to 100 $\mu$sec in the small network. This was expected, since large *DIFS* values imposed longer waiting times for data packets at the buffer which, eventually discarded these packets due to the buffer overflows, and hence dramatically magnified the total packet drops as depicted in Figure 5.16d. It can be observed that the number of packet drops at the buffer increased with larger *DIFS* values. This was due to the longer waiting time of data packets at the buffer.

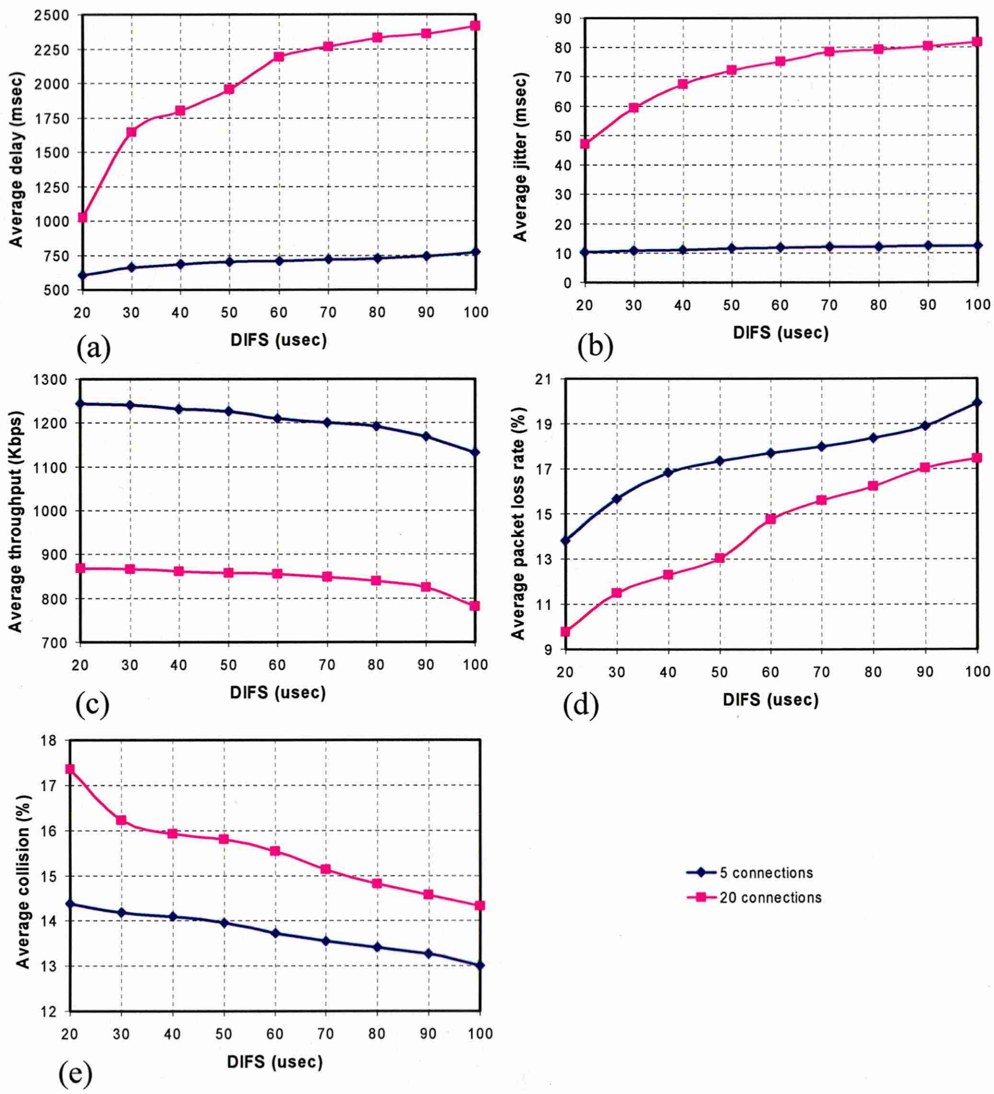



**Figure 5.16:** QoS parameters vs. *DIFS* in small and large networks, (a) average delay, (b) average jitter, (c) average throughput, (d) average packet loss rate, and (e) average collision rate.

A positive effect of increasing the *DIFS* value was a noticeable change in the average collision rate. Figure 5.16e shows that a significant reduction of average collision rate was observed at large values of *DIFS*. For example, collision rate reduced by 17% when *DIFS* value was changed from 20 $\mu$sec to 100 $\mu$sec in the large network.

### 5.3.2.3 Impact of Varying the Number of Retry Limits

This section discusses the impact of varying the number of retry limits on the QoS parameters. The results revealed that retry limits of less than 4 caused large packet losses for all network sizes. Values greater than 7 resulted in high values of delay; therefore, the values between 4 and 7 were capable of providing desirable performance.

In the IEEE 802.11 DCF scheme, the number of retry limits has two values. The short retry limit which is equal 7 and the long retry limit which is equal 4 (IEEE, 1999). Thus, as indicated in the obtained results, these two values provided a trade off performance between average delay and packet loss rate. Accepting these values as they are defined by the IEEE 802.11 standard is more convenient since the range of variation is very small (IEEE, 1999). More information about the scenarios and the complete findings is provided in Appendix B (see Appendix B.3).

## 5.4 Summary

In this chapter extensive simulations were carried out to investigate the limitations and performance of the standard IEEE 802.11 MAC protocol. The results revealed that these limitations led to a significant degradation in the QoS parameters and in overall network performance. Additionally, an inappropriate selection of the $CW_{min}$ and *DIFS* values led to a high number of collisions and large packet drop rates at the buffer. Based on these findings, the standard IEEE 802.11 DCF protocol performed inappropriately for certain applications. Improving the performance of IEEE 802.11 MAC protocol will be an aim of this study. The simulations carried out so far have helped determine the baseline of developing new MAC protocol mechanisms for improving the network performance and provisioning QoS in wireless networks.

# *CHAPTER 6*

# Development and Evaluation of Artificial Intelligence Techniques to Incorporate QoS into IEEE 802.11 MAC Protocol

## 6.1 Introduction

The problem of selecting an appropriate set of MAC protocol transmission parameters and QoS mechanism to provide predictable QoS using the IEEE 802.11 DCF scheme is an important issue in ad-hoc networks. The aim of this chapter is to simulate suitable wireless networks, develop a Fuzzy Inference System (*FIS*) to intelligently assess the QoS for video and audio applications, and to use a second *FIS* mechanism to adjust the size of $CW_{min}$ in such a way to significantly improve QoS for the applications. In addition, to develop a hybrid genetic-fuzzy approach to optimise two main MAC protocol transmission parameters these are $CW_{min}$ and *DIFS* according to the application type. Finally, to use a linear adjustment method to optimise the $CW_{min}$ size when various multimedia applications were transmitted.

Section 6.2 presents a discussion of the related work. The proposed fuzzy logic, linear adjustment, and hybrid genetic-fuzzy approaches including the simulation of experiments and traffic models are discussed in section 6.3. In section 6.4 the findings of these approaches compared with the legacy IEEE 802.11 DCF scheme are presented.

## 6.2 Previous Studies to Incorporate AI into IEEE 802.11 Protocol

The effect of adjusting the value of the contention window and/or *DIFS* on the network performance has been analysed in a number of studies. In (Qixiang et *al.*, 2004) a simple self-adaptive contention window adjustment algorithm for IEEE 802.11 MAC protocol has been proposed. This demonstrated that the performance of the legacy IEEE 802.11 MAC protocol was sensitive to the initial parameter settings. In (Gannoune and Robert, 2004) a dynamic tuning for the $CW_{min}$ value was proposed to improve the performance of the enhanced version of the IEEE 802.11 (i.e., IEEE 802.11e *EDCF*).

*DIFS* parameter has been studied for providing service differentiation among different traffic priorities (Aad and Castelluccia, 2001) and (IEEE, 2004). The value of *DIFS* in these studies was statically assigned for each class. However, less efforts has been made on tuning the *DIFS* for various traffic types. For instance, the proposed approach in (Aad and Castelluccia, 2001) combined three MAC parameters to achieve service differentiation among the different priority classes. *DIFS* was one of these parameters which were statically assigned for each traffic class.

A number of studies have used fuzzy logic in the area of computer networks. Fuzzy logic was used to assess the QoS for multimedia transmission (Saraireh et *al.*, 2004). A dynamic contention window selection scheme to achieve a theoretical throughput limit in wireless networks based on fuzzy reasoning approach was proposed in (Chen et *al.*, 2004). Liu and Hsu (2005) proposed two distributed random access protocols for multi-channel *WLAN*s. The proposed approaches were based on major modifications in the operation of the IEEE 802.11 standard protocol. In (Zomaya, 2002) and (Subrata and Zomaya, 2003), the *GA* has been used to find an optimal location management strategy for cellular networks. Therefore, no work has been provided a hybrid genetic-fuzzy mechanism to optimise multiple MAC protocol transmission parameters.

Although the discussed studies in this section and in sections 2.11 and 3.5 (see Chapters 2 and 3) have reported an improvement in the network performance when the values of *CW* and/or *DIFS* were appropriately set, however, these methods did have limitations. For example, they did not assess the network QoS, used one or two QoS parameters, relied on estimation for the number of contending stations, caused major modifications to the structure of the standard, and only considered one application type. Therefore, the use of *FIS* and the hybrid genetic-fuzzy approaches enabled (i) transmission parameters (delay, jitter and packet loss) to be integrated to indicate the QoS for the applications, (ii) the $CW_{min}$ and *DIFS* values to be intelligently adjusted based on the assessed QoS, previous $CW_{min}$, and other QoS parameters such as collision.

## 6.3 Description of the Approach

In the following sections, in addition to the network simulation and traffic models the four main approaches to incorporate QoS in IEEE 802.11 DCF scheme are described. These are: an approach based on fuzzy inference system to assess the QoS, a second approach based of fuzzy inference system to adjust the $CW_{min}$ size, a hybrid genetic-fuzzy approach to adjust the $CW_{min}$ and the *DIFS* parameters and a simple linear adjustment approach to adjust the $CW_{min}$ size.

### 6.3.1 Network Simulation and Traffic Models

In this chapter, the results for the network shown in Figure 4.2d (see Chapter 4) which consisted of 40 nodes and transmitted audio, video, and data applications are reported. However, consistent results were obtained for several other topologies. The audio and video traffic of packet sizes 160 bytes and 512 bytes, respectively were modelled using Constant Bit Rate (*CBR*). The transmission rate for each audio source was 64 kbps and for the video sources were 1 Mbps and 384 kbps alternatively. Data stations generated data packet streams with fixed size of 1500 bytes, corresponding to File Transfer Protocol (*FTP*). *UDP* was used as the main transmission protocol for most scenarios. Because the *UDP* protocol is simpler, faster and cheaper than *TCP* protocol, it does not require an acknowledgment (this less overhead); so it does not yield retransmission time, which makes it suitable to time-sensitive applications (Zheng and Boyce, 2001). Transmission Control Protocol (*TCP*) was used with *FTP* scenarios, because *FTP* traffic does not tolerate any packet loss and if any occurs the *TCP* protocol can control it through applying its congestion control mechanism (Zheng and Boyce, 2001). The simulation period was 300 seconds. Simulations were repeated 10 times, each time used a different seed that introduced randomness in the network starting condition in order to avoid the bias of random number generation. The results of the 10 simulations were averaged to determine the general behaviour of the network.

### 6.3.2 QoS Assessment Fuzzy Inference System

A fuzzy logic approach for assessing the QoS for audio, video, and data traffic has been developed. The structure of the developed approach is shown in Figure 6.1. The *FIS* system consists of four main processes; fuzzy inputs, fuzzy rules, fuzzy inferencing, and fuzzy outputs. The inputs to the system were delay, jitter and packet loss for time-sensitive application such as audio and video. However, time-insensitive application, packet loss, collision, and MAC efficiency (see section 2.12.3 in Chapter 2) were considered as the fuzzy input parameters. According to the QoS requirements for each parameter, each fuzzy input was represented by three fuzzy sets to generate the required membership functions. The position and geometry of each membership function was calculated according to the degree of overlap between these membership functions as shown in Figure 6.1. A typical example of the Gaussian membership parameters and the position of these values (as indicated by the deviation from the mean value) are summarised in Table 6.1. These parameters were determined according to the application QoS requirements. The relationships between the inputs and the related QoS achieved by the applications were expressed by fuzzy rules. The number of fuzzy rules is related to the number of input variables and the number of sets associated with each input variable. The devised QoS assessment approach has nine rules formed from a combination of three input variables, each input represented by three fuzzy sets. Typical rule examples for audio and video application that include these three inputs are as shown in Figure 6.2. The rules were written after considering the ITU recommended ranges for QoS parameters for video and audio applications ($ITU_{(a)}$, 2001).

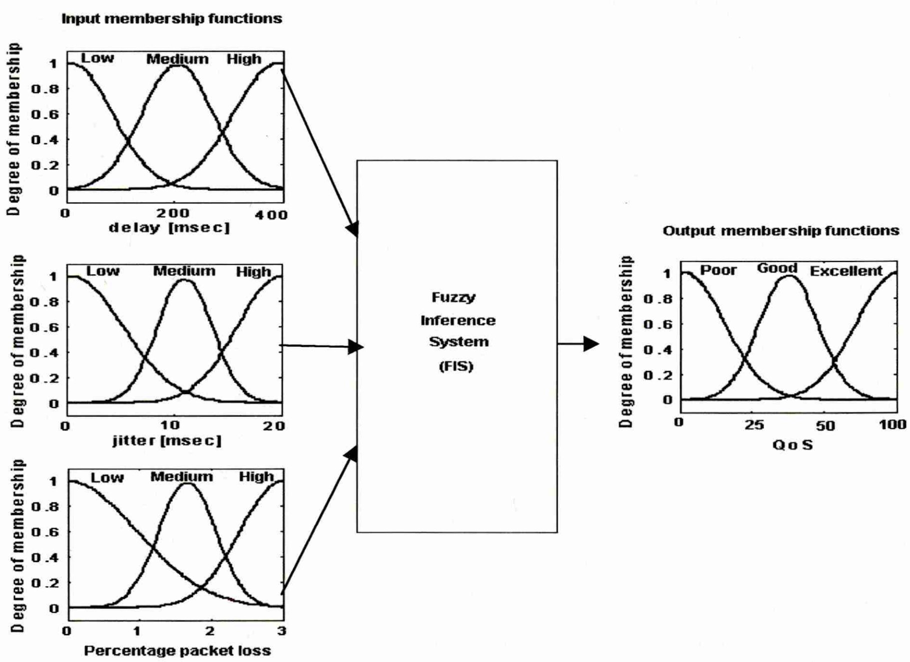



**Figure 6.1:** *FIS* structure for QoS assessment.

**Table 6.1:** Input and output Gaussian membership functions settings that were used for QoS assessment *FIS* system (video application).

| Label | Input variables | | | | | | Label | Output variable | |
|---|---|---|---|---|---|---|---|---|---|
| | Delay (msec) | | Jitter (msec) | | Packet loss (%) | | | QoS (%) | |
| | Mean | SD | Mean | SD | Mean | SD | | Mean | SD |
| **Low** | 150 | 15 | 15 | 1.44 | 0 | 0.57 | **Poor** | 0 | 11.5 |
| **Medium** | 355 | 70 | 33 | 10 | 1.4 | 0.4 | **Good** | 48 | 11.5 |
| **High** | 600 | 104 | 50 | 8.6 | 3 | 0.87 | **Excellent** | 100 | 11.5 |

"*IF Delay* is Low AND *Jitter* is Low AND *Loss* is Low *THEN QoS* is Excellent"
"*IF Delay* is Medium AND *Jitter* is Low AND *Loss* is Low *THEN QoS* is Good"
"*IF Delay* is Low AND *Jitter* is Low AND *Loss* is Medium *THEN QoS* is Good"
"*IF Delay* is Medium AND *Jitter* is Low AND *Loss* is Medium *THEN QoS* is Good"
"*IF Delay* is Low AND *Jitter* is Medium AND *Loss* is Low *THEN QoS* is Good"
"*IF Delay* is Medium AND *Jitter* is Medium AND *Loss* is Low *THEN QoS* is Good"
"*IF Delay* is Low AND *Jitter* is Medium AND *Loss* is Medium *THEN QoS* is Good"
"*IF Delay* is Medium AND *Jitter* is Medium AND *Loss* is Medium *THEN QoS* is Good"
"*IF Delay* is High OR *Jitter* is High OR *Loss* is High *THEN QoS* is Poor"

**Figure 6.2:** Typical set of rules for the QoS assessment *FIS* mechanism

As indicated in Table 6.1, the three inputs, delay, jitter and packet loss were represented by high, medium, or low while only one single fuzzy output that represented the QoS by poor, good, or excellent. In this approach, the output variable was split into three singleton fuzzy sets, these are labelled as linguistic variables (Poor, Good, and Excellent QoS levels). The input and output variables employed several membership functions such Triangular, Trapezoidal, and Gaussian. Gaussian membership functions were used for the devised fuzzy system, because the tests showed that they provided best results, and had the capability of smooth transition from one membership function to another membership function, their short notation, and Gaussian membership functions proved effective in other studies in the area of networking such the work presented in (Saraireh et *al.*, 2004), (Saraireh et *al.*, 2006) and (Oliveira and Braum, 2004). The gaussian membership function is given in Equation 6.1, where $c_i$ and $\sigma_i$ are the centre and width of the $i^{\text{th}}$ fuzzy set $A^i$, respectively.

$$\mu_{A^i}(x)=\exp(-\frac{(c_i-x)^2}{2\sigma_i^2}) \tag{6.1}$$

In the *FIS* technique, the fuzzy input (crisp input) values were mapped into membership functions (fuzzification process) and assessed according to the rules considered. The output of each fired rule was aggregated and the output was used as an input to the defuzzifier. The defuzzifier converts the inferred fuzzy control action into a nonfuzzy control action (i.e. QoS) under a defuzzification strategy. In this study the centroid defuzzification method was used since it provided the best results. The output range (i.e. QoS) was 0 – 100, which classified symmetrically into three classes. These classes were defined as the QoS levels which were Poor, Good, and Excellent. The values less than 33% were classified as a poor QoS level, the values between 34% - 66% were categorised as a Good QoS level, whereas, the values greater than 66% were considered as an Excellent QoS level.

After setting up the network topology, selecting the appropriate MAC parameters, and configuring the network traffic, QoS metrics were quantitatively assessed. This was carried out by following the steps depicted in Figure 6.3. After simulating the selected network, a data file was generated (this is referred to as a trace file in *NS-2* simulation package). The QoS metrics such as throughput, delay, jitter, packet loss, and collision were extracted for each traffic type as discussed in section 4.4 (see Chapter 4). So, for the assessing process of the QoS, the measurements of the QoS parameters were taken either with respect to the simulation time (i.e. averaging the value of throughput, delay, jitter and packet loss after every one second of the simulation) or with respect to the blocking process in which the generated packets were divided into groups of packets called blocks. The number of packets in each block was equal to the number of packets during one second of the simulation unless the number of blocks was specified.

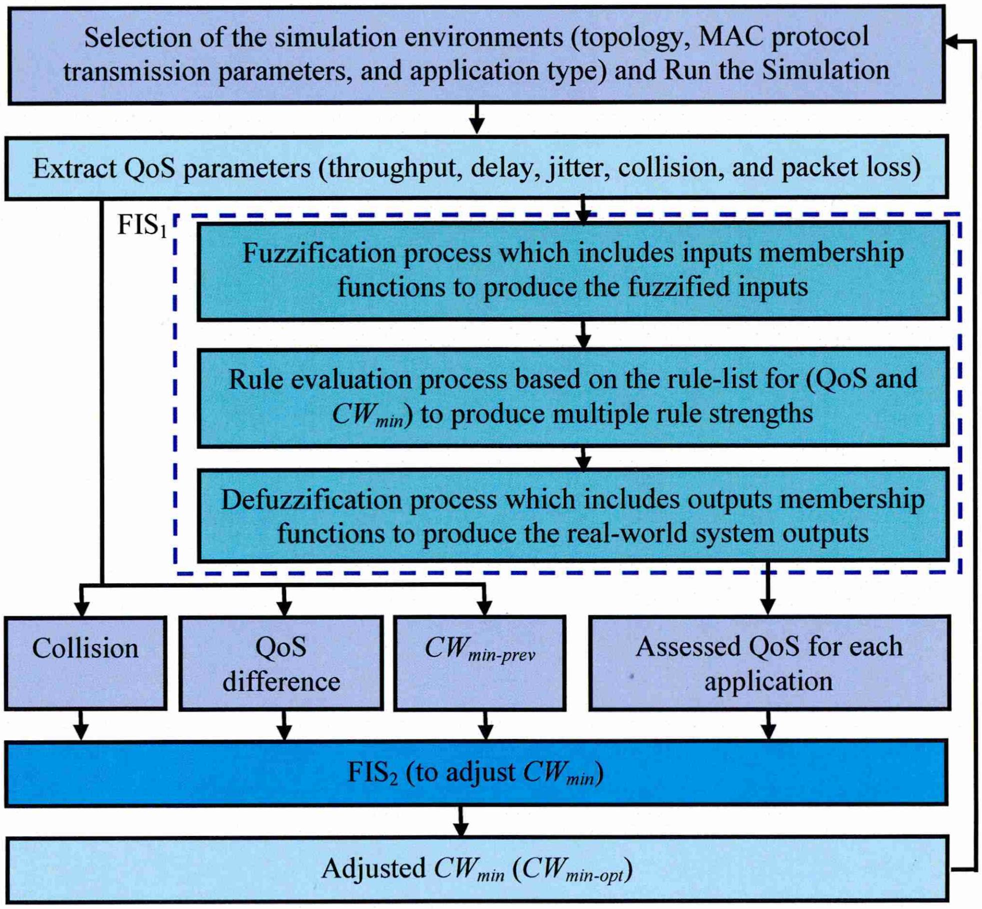



**Figure 6.3:** Fuzzy logic model for assessing QoS and optimising the $CW_{min}$ value.

### 6.3.3 Contention Window Adjustment Using a Fuzzy Inference System

A second *FIS* based on Mamdani fuzzy inference systems was implemented to adjust the $CW_{min}$ size (Mamdani, 1977). The QoS metrics throughput, delay, jitter, packet loss, MAC efficiency, and collisions were averaged and used by the QoS assessment *FIS* to assess the QoS for the transmitted application as discussed in section 6.3.2. The assessed QoS was fed into the second *FIS* together with the previous $CW_{min}$ size, MAC efficiency, average collision rate, and QoS difference (this was the difference between the current assessed QoS and the previous assessed QoS for the same type of traffic, and it was only used to determine whether the current QoS was improved or not when adjusting the $CW_{min}$). These parameters were considered as the input variables for the $CW_{min}$ adjustment *FIS*. Concerning the $CW_{min}$ input variable, each application had a different $CW_{min}$ size range. For instance, the $CW_{min}$ range for audio was from 7 to 31, for video the range was from 15 to 64, while for data the range was from 127 to 255. The selection of these ranges was based on the application type and its QoS requirements. Each input variable had a different number of membership functions. E.g., the previous value of $CW_{min}$ had seven Gaussian membership functions which were labelled as extremely low (*Elow*), very low (*Vlow*), *Low*, *Medium*, *High*, very high (*Vhigh*), and extremely high (*Ehigh*). The other input variables had a smaller number of membership functions as shown in Figure 6.4. The locations and the degree of overlap between these membership functions were chosen as indicated in Table 6.2 since these values provided best results. The second *FIS* processed these inputs by following the processes discussed in section 6.3.2 to provide a new $CW_{min}$ size for each application. The new $CW_{min}$ size was used for the next simulation run. Examples of the rules used are presented in Figure 6.5 and the complete set is provided in Appendix C.1.

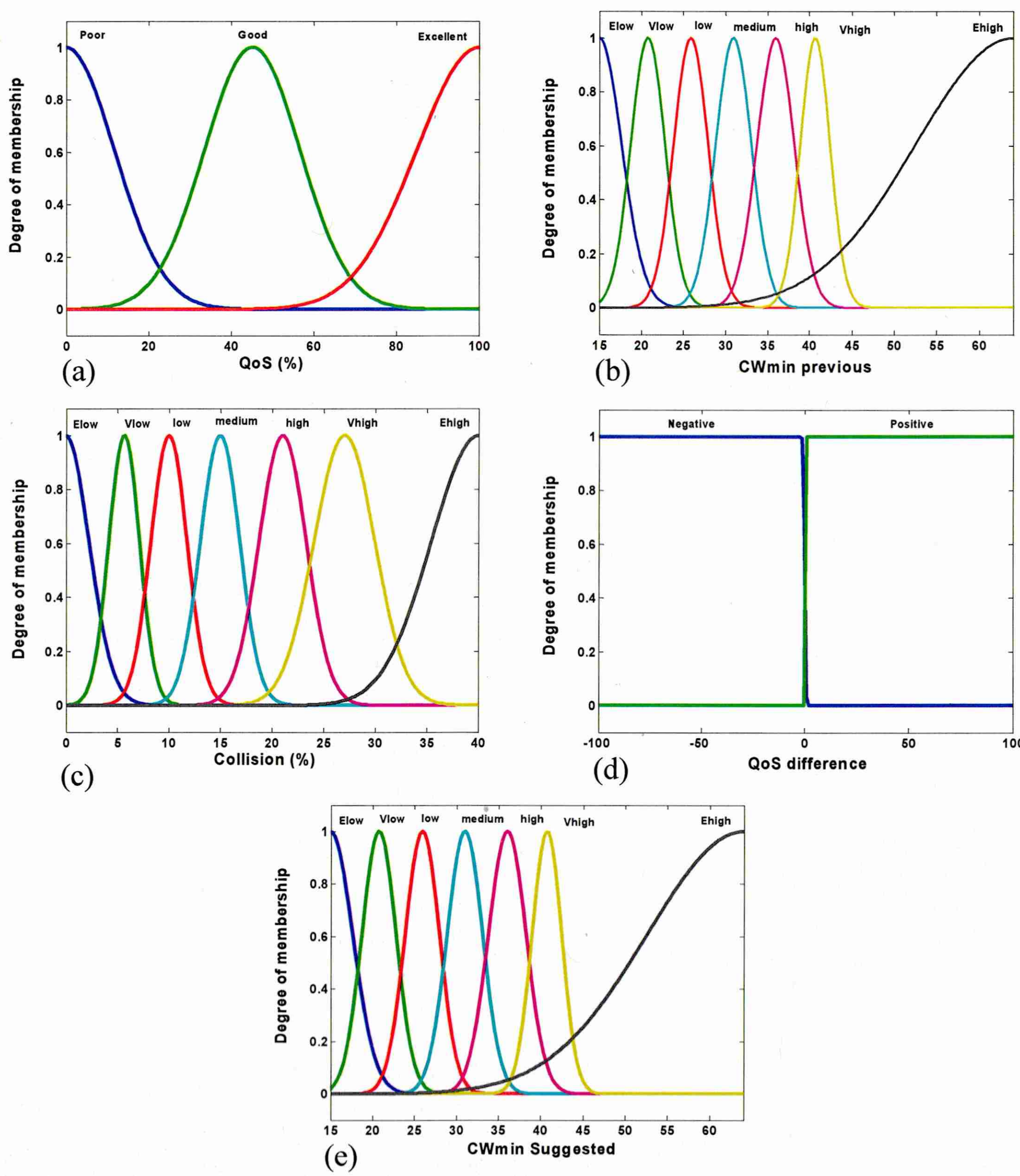



**Figure 6.4:** Example of input and output membership functions used in the second *FIS* for controlling the $CW_{min}$ size, (a) QoS input variable, (b) previous $CW_{min}$ input variable, (c) collision input variable, (d) QoS difference input variable, and (d) adjusted $CW_{min}$ output variable.

When the system outlined in Figure 6.3 was used to control the value of $CW_{min}$, the measurements of the QoS parameters were averaged for the whole period of the simulation time and then fed to the first *FIS* to assess the QoS. Thereafter, the assessed QoS, the previous $CW_{min}$, collision rate, and QoS difference were fed to the second *FIS* to obtain the controlled $CW_{min}$. The adjusted $CW_{min}$ was fed to the network for the next simulation run. This implied that the adjusted $CW_{min}$ was only fed to the network once at the beginning of the simulation and remained constant until the end of simulation.

**Table 6.2:** Input and output Gaussian membership functions settings used for $CW_{min}$ adjustment of *FIS* system (video application).

| Input variable | Label | Mean | Standard deviation |
|---|---|---|---|
| **$CW_{min}$ previous** | Extremely Low | 15 | 3 |
| | Very Low | 21 | 2 |
| | Low | 26 | 2.05 |
| | Medium | 31 | 2.14 |
| | High | 36 | 2.25 |
| | Very High | 41 | 1.75 |
| | Extremely High | 64 | 11.5 |
| **Current QoS** | Poor | 0 | 11.5 |
| | Good | 48 | 11.5 |
| | Excellent | 100 | 11.5 |
| **QoS difference** | Positive | -100 $\rightarrow$ 0 | - |
| | Negative | 0 $\rightarrow$ +100 | - |
| **Collision** | Extremely Low | 2.29 | 0 |
| | Very Low | 5.7 | 1.49 |
| | Low | 10 | 1.76 |
| | Medium | 15 | 1.91 |
| | High | 21 | 2.29 |
| | Very High | 27 | 2.95 |
| | Extremely High | 40 | 4.6 |
| **Output variable** | **Label** | **Mean** | **Standard deviation** |
| **$CW_{min}$ suggested** | Extremely Low | 15 | 3 |
| | Very Low | 21 | 2 |
| | Low | 26 | 2.05 |
| | Medium | 31 | 2.14 |
| | High | 36 | 2.25 |
| | Very High | 41 | 1.75 |
| | Extremely High | 64 | 11.5 |

"***IF*** $QoS_{prev}$ is Excellent AND $CW_{min\_prev}$ is Low AND *collision* is High AND $QoS_{difference}$ is positive ***THEN*** $CW_{min\_new}$ is Medium"

"***IF*** $QoS_{prev}$ is Good AND $CW_{min\_prev}$ is VLow AND *collision* is High AND $QoS_{difference}$ is Negative ***THEN*** $CW_{min\_new}$ is Low"

"***IF*** $QoS_{prev}$ is Poor AND $CW_{min\_prev}$ is High AND *collision* is Low AND $QoS_{difference}$ is Positive ***THEN*** $CW_{min\_new}$ is Medium"

**Figure 6.5:** Typical set of rules for the *FIS* adjustment mechanism.

### 6.3.4 Hybrid Genetic-Fuzzy System

A hybrid genetic-fuzzy approach for adjusting the $CW_{min}$ and *DIFS* according to the assessed QoS for audio, video, and data traffic has been developed. The input data to the genetic algorithm was divided into two main parts: the *GA*-related parameters and the problem-related parameters. The *GA* related parameters included the population size, $N_p$; the selection scheme used; the replacement method; the crossover probability, $P_c$; the mutation probability, $P_m$; the immigration threshold value and the corresponding number of generations; and finally, the termination criterion. The problem related parameters included the number of contending stations, *X*; application type, MAC protocol transmission parameters such as $CW_{min}$ and *DIFS*, and the measured QoS value that was used to determine the value of the fitness function for the *GA* mechanism.

The settings of the *GA*-related parameters are as follows: the population size was 500 individuals. The rank-based selection strategy was used where the rank-based ratio was 0.5. The crossover probability was 0.7; the mutation probability was 0.005. The generational replacement scheme was applied where the number of elite parents that were passed to the next generation was one-tenth of the population. Extinction and immigration operator was applied when the improvement in the fitness value of the best individual over 400 generations was less than 0.01. The *GA* was stopped when one of the following conditions was met. First, the fitness of the best individual of the population reached a value of 0.8 (i.e. an excellent QoS level). Second, a maximum number of 1000 generations was reached. Third, the improvement in the fitness value of the best individual in the population over 200 generations was less than 0.01.

The settings of problem-related parameters are as follows: the number of connections was varied (i.e., 3, 8, and 20 connections were considered), and the algorithm could be conducted for several network configurations. The application types were audio, video, and data. A range for the MAC protocol transmission parameters was specified according to the application type. The $CW_{min}$ ranges were [15-60], [7-38] and [128-252] time slots with step or accuracy equal to 3, 2 and 4 for video, audio and data, respectively. The *DIFS* ranges were [40-75 $\mu$sec], [20-55 $\mu$sec], and [70-140 $\mu$sec] with steps equal 5, 5, and 10 for video, audio, and data, respectively. The selection of these ranges was based on the QoS requirements for these applications. With regard to the accuracies or steps, they were chosen to be compatible with the operation and requirements of the genetic that agreed with $(2^n)$ values, where ($n$ = 0, 1, 2, 3, …).

Once both the *GA* parameters and the problem related parameters were set the *GA* was started. The *GA* generated solutions were fed into the network simulation package that simulated the selected wireless network. At the end of each simulation run, the QoS parameters were extracted and fed into the developed *FIS* system to intelligently assess the QoS for multimedia applications and provide the fitness value of the candidate solution to the genetic algorithm. The rules of the *FIS* System were written by considering the ITU recommended ranges for QoS parameters for video, audio, and data applications. The output of the *FIS* was the assessed QoS for each connection. The fitness function was calculated according to the *FIS* output. It was based on the minimum QoS achieved by any application and the maximum required QoS for all application as given in Equation 6.2. This process was automated and repeated for all individuals in a certain generation and for a number of generations until a convergence criterion was obtained where the parameters of the best individual so far were found. These were the optimal set for $CW_{min}$ and *DIFS* values.

$$fitness\ function = \frac{Minimum\ QoS\ for\ application}{Maximum\ overall\ QoS} \tag{6.2}$$

### 6.3.5 Linear Adjustment of Contention Window Minimum ($CW_{min}$)

A simple linear scheme for $CW_{min}$ adjustment was developed. The scheme was based on a simple linear function of the $CW_{min}$ according to the assessed QoS for audio, video and data traffic. Firstly, the proposed scheme examined the measured QoS of the transmitted traffic and then slowly increased the $CW_{min}$ by a small number of slots (e.g., 4 slots). Then, the QoS measurement was based on whether it had improved or degraded. If the measured QoS had improved a further increase to the $CW_{min}$ was applied, else a linear decrease in the $CW_{min}$ was performed. The process was repeated until QoS for the transmitted applications did not improve any further. This method was devised to investigate the possibility of using simple linear techniques combined with the *FIS* assessment system in adjusting MAC protocol transmission parameters.

## 6.4 Results and Discussion

The results and discussions are divided into five sections: *FIS* QoS assessment mechanism, linear adjustments of $CW_{min}$, fuzzy logic adjustment of $CW_{min}$, hybrid genetic-fuzzy adjustment of $CW_{min}$ and *DIFS* values, and finally the implication of the proposed approaches in a real system.

### 6.4.1 QoS Assessment of the Basic IEEE 802.11 DCF Scheme

The IEEE 802.11 DCF scheme is only able to support best-effort service without any QoS guarantees or differentiation. In this section the accuracy of the *FIS* assessment system is discussed. Further, the QoS for audio, video, and data applications is assessed with the current settings supported by the IEEE 802.11 standard (see Table 4.1 in Chapter 4). It also outlines how the IEEE 802.11 DCF scheme with these settings is incapable of effectively utilising the channel capacity.

A typical set of results obtained using the QoS assessment *FIS* system is provided in Table 6.3. It can be observed that the mechanism has successfully processed the QoS requirements of the video and audio applications. For instance, high values of delay, jitter, or packet loss resulted in a poor QoS. However, medium and low values of these parameters resulted in good and excellent QoS levels, respectively.

**Table 6.3:** Typical set of results for the QoS assessment *FIS* mechanism.

| Application type | Inputs (QoS Parameters) | | | Output (Assessed QoS) | |
|---|---|---|---|---|---|
| | Delay (msec) | Jitter (msec) | Loss (%) | QoS (%) | Linguistic Term |
| Video | 112 | 40 | 3.9 | 15.8 | Poor |
| | 600 | 22 | 4 | 10.2 | Poor |
| | 300 | 20 | 0.9 | 44.8 | Good |
| | 330 | 21 | 1.8 | 40.1 | Good |
| | 10 | 2.3 | 0 | 89.6 | Excellent |
| | 130 | 13.3 | 1.7 | 79.2 | Excellent |
| Audio | 600 | 8 | 1.3 | 9.5 | Poor |
| | 300 | 1.7 | 5.1 | 26.4 | Poor |
| | 261 | 2.1 | 1.0 | 55.1 | Good |
| | 263 | 0.8 | 2.9 | 53.1 | Good |
| | 15.6 | 1.3 | 2 | 89.7 | Excellent |
| | 70 | 2.4 | 0.9 | 78.6 | Excellent |

Each traffic type has different delay requirements. Video traffic for example requires the following: low delay (less than 400 msec) ($\text{ITU}_{(a)}$, 2001), a packet loss rate less than 3% (Boyce and Gaglianello, 1998), and jitter has to be less than 50 msec. Therefore, these parameters should be kept small (Dalgic and Tobagi, 1996).

Average delay for the three video connections using the standard IEEE 802.11 DCF scheme exceeded the minimum QoS requirements for video transmission. The values of average delay were 421.2, 441.5, and 620.7 msec for the first, second, and third video connections, respectively. The causes of this increase in the delay for video connections were: all stations started with the same $CW_{min}$ size. This implied that they had the same chance to access the channel; thus, the probability of simultaneous transmission was very high which in turn increased the probability of collisions. The collided packets in this case required retransmission by the MAC protocol which in turn led to late arrivals of these packets at the destination as well as a higher drop of the waiting packets at the buffer. Further, the default size of $CW_{min}$ was not optimal. When the $CW_{min}$ size was too high, a number of empty time slots were wasted and resulted in an unjustified waiting time of packets at the buffer. This led to high values of delays and smaller throughput. When the $CW_{min}$ size was too small, this increased the probability of collisions which in turn increased the delay for the transmitted packets (i.e., increased the number of retransmissions of the collided packets).

The reduction in average throughput and the increase of packet loss ratio for video connections were due to the high competition among the active stations in the same *IBSS*. These stations had the same chance to access the channel (same $CW_{min}$ size), which led to high collisions between them. Moreover, the high number of collisions reduced the MAC protocol efficiency which in turn increased the number of retransmissions of the collided packets resulting in high delay and high drops of the packets that were waiting in the buffer.

In order to provide an excellent QoS for audio traffic, delay, jitter, and packet loss have to meet strict QoS requirements. Average delay has to be between 100-400 msec ($\text{ITU}_{(a)}$, 2001). Jitter has to be limited to less than 5 msec to ensure smooth playback at the receiver. Since packet loss in the wireless networks was high, the probability that consecutive audio packets were lost was significant. Therefore, packet loss rate has to be at very low levels. For audio connections, average values of delay and packet loss remained within the QoS requirements. Average delay values were 32.9, 79.78, 121.8 msec for the first, second and third audio connections, respectively. Packet loss had average values less than 1.5%. However, the average values of throughput and jitter were outside the desired range of QoS. The mean values of these parameters are summarised in Table 6.4.

**Table 6.4:** Assessed QoS for audio and video traffic using the standard IEEE 802.11 DCF scheme.

| Connection | Delay (msec) | Jitter (msec) | Throughput (Kbps) | Packet loss (%) | QoS (%) | QoS level |
|---|---|---|---|---|---|---|
| | Mean | Mean | Mean | Mean | Mean | |
| **Video connection 1** | 421.2 | 5.6 | 337.7 | 4.9 | 49.7 | Good |
| **Video connection 2** | 441.5 | 6.4 | 336.8 | 5.1 | 42.2 | Good |
| **Video connection 3** | 620.7 | 9.9 | 284.7 | 26.4 | 18.3 | Poor |
| **Audio connection 1** | 32.9 | 3.9 | 63.5 | 0 | 59.7 | Good |
| **Audio connection 2** | 79.78 | 12.5 | 59.6 | 0 | 17.7 | Poor |
| **Audio connection 3** | 121.8 | 13.6 | 55.6 | 1.1 | 15.8 | Poor |

The QoS parameters for each connection were averaged and fed into the QoS assessment *FIS* to assess their QoS. The assessed QoS for video and audio traffic according to the basic IEEE 802.11 DCF scheme resulted in poor QoS levels for some connections (i.e., they achieved QoS less than 33%). The third video connection had a poor QoS with mean value equal to 18.3%. The second and third audio connections also experienced a poor QoS with mean values equal to 17.7% and 15.8%, respectively. The degradation of the QoS for video and audio connections was due to the high values of delay and jitter which also resulted in high fluctuations in the assessed QoS.

Therefore, the standard IEEE 802.11 protocol with the default settings defined by the standard was incapable of achieving the minimum QoS requirements for multimedia transmission. Furthermore, the protocol was unable to utilise the channel capacity when the protocol operated at heavy load traffic conditions as illustrated in Figure 6.6.

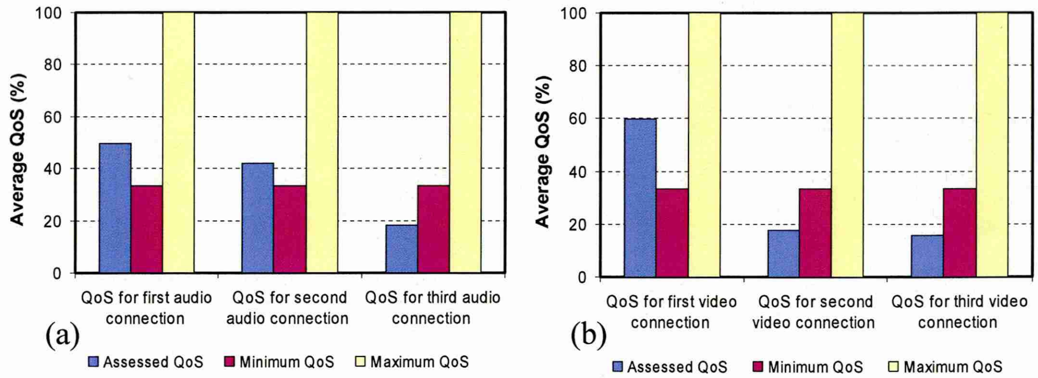



**Figure 6.6:** Assessed QoS for video and audio vs. the minimum and maximum QoS, (a) assessed QoS for video connections, and (b) assessed QoS for audio connections.

In Figures 6.6a and 6.6b, the average QoS achieved by each application was compared with the minimum and maximum QoS requirements for video and audio transmission. It can be observed that three of the six connections were incapable of meeting even the minimum QoS for these applications (i.e., 33%) and only 17.3% average QoS was achieved. This confirmed that the standard IEEE 802.11 DCF scheme was incapable of utilising the channel capacity.

In this section, a *FIS* system was used to assess the QoS for multimedia transmission over the standard IEEE 802.11 MAC protocol. The proposed approach demonstrated the limitations of the distributed function of the IEEE 802.11 protocol. It showed how the QoS for the transmitted applications varied with the variation of the network conditions. The *FIS* assessment system showed that the default settings of MAC parameters in the IEEE 802.11 DCF scheme were unable to achieve the minimum QoS level for multimedia transmission and were unable to utilise the available channel capacity in a best manner. Therefore, the results indicated the need for adjusting the main MAC protocol transmission parameters in order to improve its performance. In the following sections, the three proposed approaches (i.e., linear adjustment, *FIS* adjustment, and the hybrid genetic-fuzzy adjustment) are used to accomplish this objective.

### 6.4.2 Simulation Results with Linear Scheme

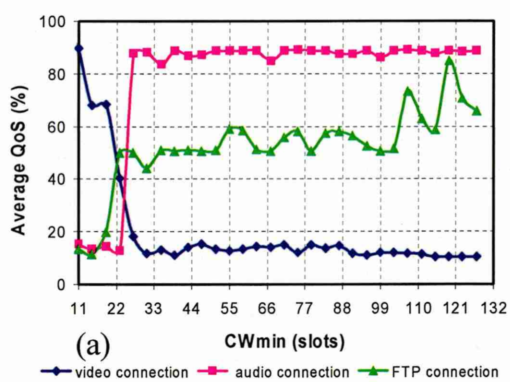
In this section, the QoS was assessed for three traffic types, video, audio and *FTP* using the linear increase/decrease approach. Initially, the $CW_{min}$ size was increased linearly from 7 to 127 slots for one source while it remained fixed at the default value (i.e., $CW_{min}$ = 31) for other sources. Thereafter, the proposed method was used when the $CW_{min}$ size was varied linearly for the three sources simultaneously. In addition, the same approach was used when the number of connections was increased to three video and three audio connections.

The main QoS parameters considered for video and audio traffic were delay, jitter, and packet loss. The MAC efficiency metric was the most important parameter for assessing the QoS for *FTP* traffic, as it provided the *FIS* system with the global knowledge about the network conditions.

### 6.4.2.1 Varying the $CW_{min}$ Size Linearly for each Application Independently

#### *6.4.2.1.1 Video Traffic*

As shown in Figures 6.7a and 6.7b, the assessed QoS for video traffic had a mean value of 62.5% at small $CW_{min}$ sizes e.g. less than 31. When the $CW_{min}$ size was 23, the QoS started to degrade. This was due to the increase in the QoS for audio and *FTP* traffic, since their $CW_{min}$ sizes were close or equal to the $CW_{min}$ size for video traffic, and had the same chance to access the channel. A summary of QoS values for the transmitted applications in both MAC protocol access mechanisms is listed in Table 6.5.



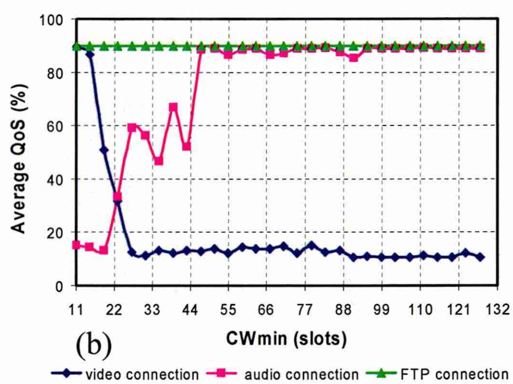




**Figure 6.7:** QoS for video, audio and *FTP* traffic when the value of $CW_{min}$ was varied for video connection in the range of (7 - 127 slots), while it kept fixed for audio and *FTP* traffic, (a) assessed QoS for the basic access mechanism, and (b) assessed QoS for the RTS/CTS access mechanism.

The assessed QoS for video traffic degraded to a poor level at a $CW_{min}$ value above 23. Simultaneously, the QoS for audio and *FTP* traffic moved from poor level to excellent and good levels, respectively. E.g., when the value of $CW_{min}$ was 27, a high QoS with mean value equal to 87.8% (i.e. excellent QoS) was achieved. *FTP* traffic achieved an acceptable QoS with a mean value of 56.3% (i.e. good QoS level). For the *RTS/CTS* access mechanism, the behaviour of the network was similar to the basic access mechanism except that the assessed QoS for *FTP* traffic increased. This was because the MAC efficiency parameter was the main input to the *FIS* assessment system to assess the QoS for *FTP* traffic. The MAC efficiency parameter was computed according to the number of collisions in data packets. Since there were no collisions in data packets when the *RTS/CTS* access mechanism was used, this resulted in high values of this parameter which in turn resulted in high QoS for *FTP* traffic.

**Table 6.5:** QoS statistics for video, audio and *FTP* traffic when the $CW_{min}$ size was varied for video connection in the range of (7-127), while it was kept fixed for audio and *FTP* traffic using $CW_{min} = 31$.

| $CW_{min}$ (slots) | Basic access mechanism | | | RTS/CTS access mechanism | | |
|---|---|---|---|---|---|---|
| | QoS for video (%) | QoS for audio (%) | QoS for FTP (%) | QoS for video (%) | QoS for audio (%) | QoS for FTP (%) |
| 15 | 52 | 13.2 | 11.3 | 86.5 | 14.5 | 89.7 |
| 31 | 11.7 | 88 | 44 | 11.3 | 12.3 | 89.7 |
| 47 | 15.2 | 87.2 | 50.3 | 12.7 | 88.5 | 89.7 |
| 63 | 14.3 | 88.7 | 51 | 13.7 | 88.8 | 89.7 |
| 79 | 15 | 88.7 | 50.3 | 15.2 | 88.7 | 89.7 |
| 95 | 11 | 88.8 | 52.3 | 10.8 | 88.7 | 89.7 |
| 111 | 11.2 | 88.8 | 62.8 | 11.3 | 89 | 89.7 |
| 127 | 10.5 | 88.8 | 65.8 | 10.5 | 88.8 | 89.7 |

Some fluctuations were observed in the QoS curve for audio traffic at small $CW_{min}$ sizes, e.g. less than 43. This was due to the impact of other connections that caused a minor increase in packet loss ratio above the minimum limit of QoS requirements for audio traffic. When the value of $CW_{min}$ for video traffic was increased, the fluctuations in the QoS curve for audio traffic became less and the assessed QoS for audio became high QoS (mean value of 88.3%).

The increase in the value of $CW_{min}$ increases the packet waiting time of video traffic at the buffer and this in turn reduced the achieved throughput at the destination and increasing the amount of delay and jitter for video packets in spite of the reduction in the number of collisions over the medium.

#### *6.4.2.1.2 Audio Traffic*

In this investigation, the value of $CW_{min}$ was varied linearly for audio traffic from 7 to 127 and was kept fixed at 31 for video and *FTP* traffic.

As shown in Figures 6.8a and 6.8b, the audio traffic achieved an excellent QoS level at small $CW_{min}$ sizes (i.e., less than 31). The mean values were 84% and 89.5% for the basic access and *RTS/CTS* access mechanisms, respectively. Excellent QoS was obtained for audio traffic due to its high accessibility to the medium and at the cost of other connections. The QoS parameters obtained for audio traffic for these small $CW_{min}$ sizes met the preferred QoS requirements (see Table 4.2 Chapter 4). E.g., packet loss ratio was zero, and the observed average delay and jitter were less than 36 msec and 10 msec, respectively, additionally, video traffic had a poor QoS with a mean value of 12.6% in both MAC protocol access mechanisms as shown in Figure 6.8. Poor performance for video traffic was due to the impact of audio and *FTP* connections at small $CW_{min}$ values especially at $CW_{min}$ size equal 31. When $CW_{min}$ size was 31, all sources had the same chance to access the medium and the channel was completely captured by *FTP* traffic particularly when the *RTS/CTS* access mechanism was used.

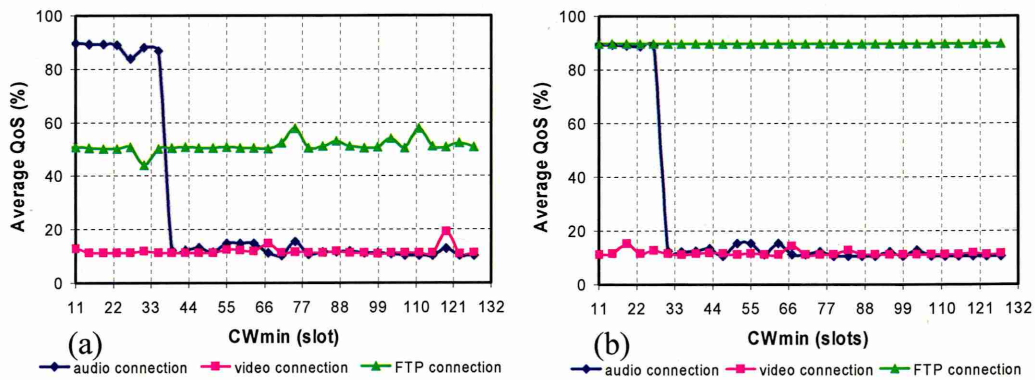



**Figure 6.8:** QoS for video, audio and *FTP* traffic when the value of $CW_{min}$ was varied for audio connection in the range of (7 - 127 slots), while it was kept fixed for video and *FTP* traffic, (a) assessed QoS for the basic access mechanism and (b) assessed QoS for the RTS/CTS access mechanism.

#### *6.4.2.1.3 File Transfer Protocol (FTP) Traffic*

In this section the QoS was assessed when the $CW_{min}$ size was varied linearly for *FTP* traffic within the range of (7 - 127 slots), while the $CW_{min}$ size for audio and video traffic were kept fixed at the default value (i.e., $CW_{min}$ equal 31).

*FTP* traffic was transmitted using the *TCP* protocol with packet size equal to 1500 bytes since a large packet size may achieve better throughput especially on light load networks. The assessed QoS against the $CW_{min}$ size for *FTP* traffic in both MAC protocol access mechanisms is shown in Figures 6.9a and 6.9b. The QoS for *FTP* traffic for the basic access mechanism was very poor for small values of $CW_{min}$. For small $CW_{min}$ sizes, the competition between stations was very high; therefore a high number of collisions occurred. This in turn reduced the average MAC protocol efficiency (i.e., high number of retransmissions). MAC efficiency was considered the main QoS parameter when assessing the QoS for *FTP* traffic instead of packet loss. This was because this parameter reflected the network conditions (e.g., the number of collision in the network). Moreover, *FTP* was transported using *TCP* protocol which was a connection oriented protocol; therefore, part of *FTP* packets still resided in the *TCP* queue, aborting or ending the simulation prevented transmission of these packets and they were considered lost. For these reasons MAC efficiency was considered as a main parameter when assessing the QoS for *FTP* traffic.

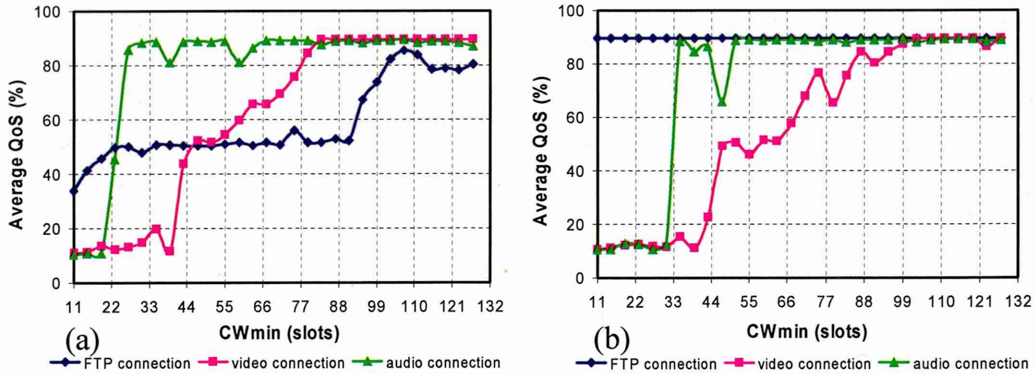



**Figure 6.9:** QoS for video, audio and *FTP* traffic when the value of $CW_{min}$ was varied for *FTP* connection in the range of (7 - 127 slots), while it was kept fixed for video and audio traffic, (a) assessed QoS for the basic access mechanism, and (b) assessed QoS for the RTS/CTS access mechanism.

Poor QoS levels were observed for video and audio traffic in both MAC protocol access mechanisms when the $CW_{min}$ sizes for *FTP* traffic were very small (i.e., less than 31). Any further increase in the $CW_{min}$ size for *FTP* traffic resulted in better QoS for audio, video, and *FTP* traffic in both MAC protocol access mechanisms. Since increasing the $CW_{min}$ for *FTP* traffic provided the video and audio traffic, that had fixed $CW_{min}$ size (i.e. 31), more chances to access the channel. For instance, audio traffic had a mean QoS equal to 75.5% and 70.3%; video traffic had a mean QoS equal to 59% and 54.7%; and *FTP* traffic had a mean QoS equal to 53.3% and 89.5% for the basic access and the *RTS/CTS* access mechanisms, respectively. The QoS for all traffic improved with the increase in the $CW_{min}$ size for *FTP* traffic as shown in Figures 6.9a and 6.9b. This was because the *FTP* traffic was less sensitive to the variations of the $CW_{min}$ size.

### 6.4.2.2 Varying the $CW_{min}$ Size Linearly for All Applications

Here, the $CW_{min}$ size was varied for the three applications together simultaneously. For Audio traffic the $CW_{min}$ was varied from 7 to 31, for video traffic the $CW_{min}$ size was varied from 15 to 90, while for the *FTP* traffic the $CW_{min}$ size was altered from 63 to 255 slots. The selection of these ranges was based on the QoS requirements and the sensitivity of each traffic type (audio has high priority, video has medium priority and *FTP* has low priority) to the variation of the $CW_{min}$ size as discussed in the previous section.

As shown in Figure 6.10a and summarised in Table 6.6, the QoS for the three connections improved with variation of the $CW_{min}$ value for each simulation run. Audio and *FTP* traffic achieved better QoS than video traffic. E.g., the mean QoS for audio and *FTP* traffic were equal to 80% while 60.5% mean QoS was achieved for video traffic. Small fluctuations were noticed in the QoS for video traffic especially at the 8$^{th}$ run of the simulation. This was due to an increase in the $CW_{min}$ value. After decreasing the $CW_{min}$ value for video traffic (i.e. resetting to the previous $CW_{min}$ size), the QoS improved and became steady for the rest of the simulation runs as shown in Figures 6.10a and 6.10b.

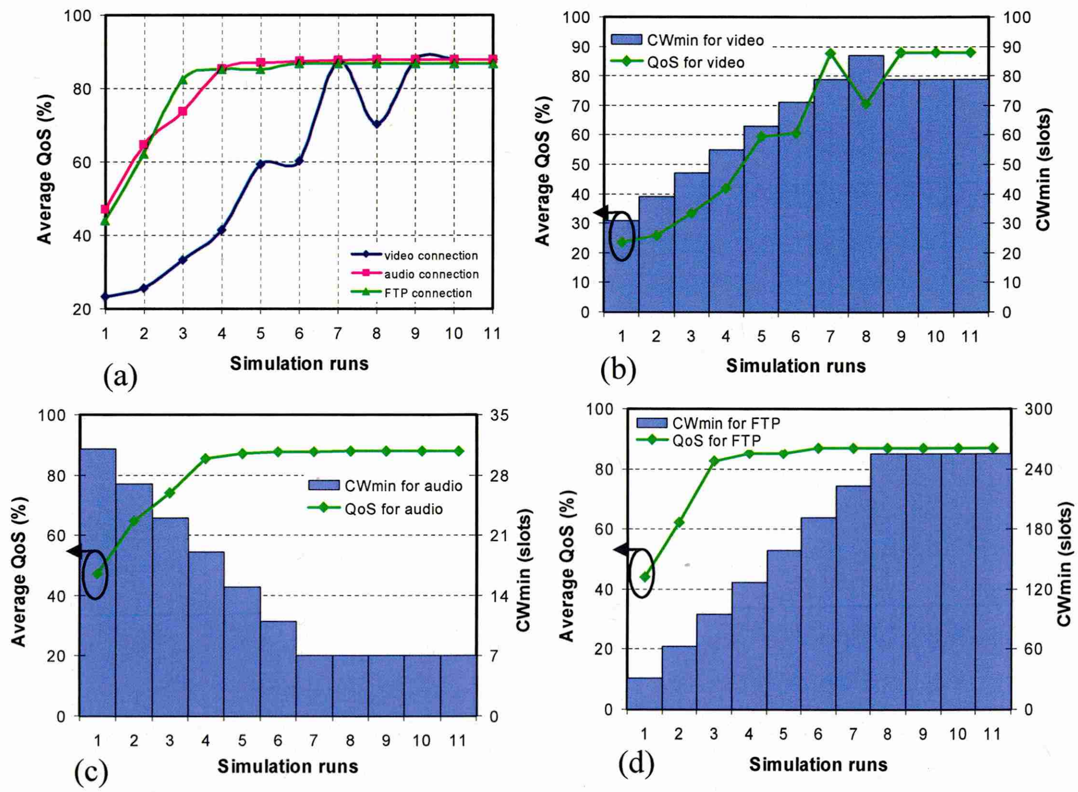



**Figure 6.10:** (a) QoS for video, audio and *FTP* traffic when the $CW_{min}$ size was varied for the three connections together simultaneously. Each traffic type had a specific range (audio (7-127), video (15-64), *FTP* (127 - 255)), (b) the relationship between the QoS and the $CW_{min}$ size for video traffic, (c) the relationship between the QoS and the $CW_{min}$ size for audio traffic and (d) the relationship between the QoS and the $CW_{min}$ size for *FTP* traffic.

In Figures 6.10b, 6.10c and 6.10d, the relationship between the linear variations (increase/decrease) of $CW_{min}$ sizes for three applications is depicted. For video traffic (see Figure 6.10b), the $CW_{min}$ value was increased linearly by 8 slots for each simulation run. The achieved QoS for video traffic improved with this increase up to $CW_{min}$ size equal to 79 slots. After increasing the $CW_{min}$ value to 87 slots, the QoS degraded from 87.5% to 70.2% at the 8$^{th}$ run of the simulation. As mentioned earlier in this section, after decreasing the $CW_{min}$ value by 8 slots, the QoS improved from 70.2% to 87.8%.

The $CW_{min}$ size for audio traffic was linearly decreased by 4 slots for each simulation run. As a result, the QoS improved providing a mean value of 87.8% at $CW_{min}$ value equal to 7 slots as shown in Figure 6.10c. Figure 6.10d shows the relationship between the QoS and the $CW_{min}$ variation (increase/decrease) for *FTP* traffic. As discussed at the beginning of this section, *FTP* traffic had a wider range of $CW_{min}$ values. Therefore, its $CW_{min}$ size was increased linearly by 32 slots for each simulation run. The average QoS for this connection improved with the linear increase in the $CW_{min}$ size to have an excellent level equal to 86.6% at $CW_{min}$ value equal to 255 slots due to a small number of collisions.

**Table 6.6:** QoS obtained for video, audio, and *FTP* traffic using simultaneous variation of the $CW_{min}$ size.

| Video traffic | | Audio traffic | | FTP traffic | |
|---|---|---|---|---|---|
| $CW_{min}$ (slots) | QoS (%) | $CW_{min}$ (slots) | QoS (%) | $CW_{min}$ (slots) | QoS (%) |
| 31 | 23.3 | 31 | 47 | 31 | 44 |
| 39 | 25.7 | 27 | 64.7 | 63 | 62.2 |
| 47 | 33.3 | 23 | 73.8 | 95 | 82.3 |
| 55 | 41.5 | 19 | 85.3 | 127 | 85.2 |
| 63 | 59.3 | 15 | 87 | 159 | 85.2 |
| 71 | 60.3 | 11 | 87.5 | 191 | 86.7 |
| 79 | 87.5 | 7 | 87.7 | 223 | 86.7 |
| 87 | 70.2 | 7 | 87.8 | 255 | 86.7 |
| 79 | 87.8 | 7 | 87.8 | 255 | 86.7 |

The effectiveness of the linear approach was apparent when the number of connections was increased. In this scenario, the number of connections was increased to six connections (three video and three audio connections). During the first run of the simulation all connections transmitted at $CW_{min}$ size equal 31 (i.e., default value). At $CW_{min}$ size equal 31, the 3$^{rd}$ video, the 2$^{nd}$ and the 3$^{rd}$ audio connections experienced poor QoS values as shown in Figure 6.11. Thereafter, increasing the $CW_{min}$ size for video connections by 8 slots for each simulation run improved the QoS for the 3$^{rd}$ video connection on the cost of the QoS for the 2$^{nd}$ and 3$^{rd}$ video connections as shown in Figure 6.11. Decreasing the $CW_{min}$ value for audio connections by 4 slots for each simulation run enhanced the QoS for all audio connections as shown in Figure 6.11.

As shown in Figures 6.10 and 6.11, the linear approach proved its capability of achieving an acceptable range of QoS for the transmitted traffic. Although, there were some fluctuations in the achieved QoS, the results indicated its effectiveness in meeting the QoS requirements for multimedia transmission. In the following section, the performance of the *FIS* adjustment system is discussed.

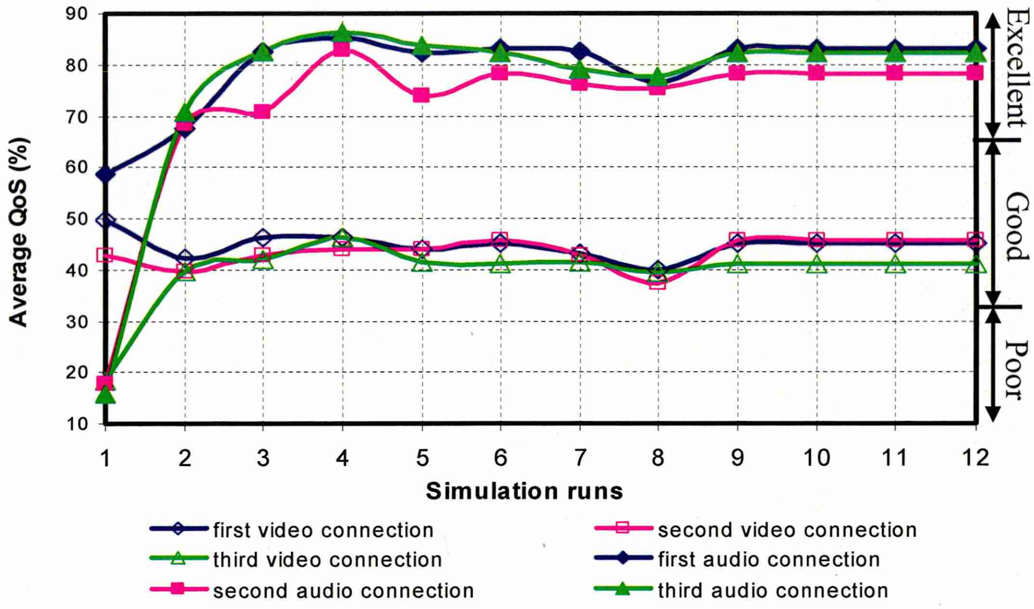



**Figure 6.11:** (a) The QoS for three video connections using linear increase/decrease approach. (b) The QoS for three audio connections using linear increase/decrease approach. (c) The variation of $CW_{min}$ for video connections. (d) The variation of $CW_{min}$ for audio connections.

### 6.4.3 Fuzzy Logic Adjustment of the Minimum CW size

Fuzzy logic was used to assess the QoS and to adjust the $CW_{min}$ value in order to improve the network performance and to achieve an excellent level of QoS when transmitting different traffic types. To achieve these objectives, several scenarios were considered that included various traffic types such as audio, video, and data.

According to the results discussed in section 6.4.2.1, the audio and video traffic were more sensitive to a variation in the $CW_{min}$ size than *FTP* traffic. Therefore, the audio traffic was given a higher priority over the video and *FTP* traffic by assigning smaller $CW_{min}$ sizes in the range of (7 - 31 slots). Video traffic was given a medium priority through specifying higher $CW_{min}$ sizes in the range of (15 - 64 slots); whereas the *FTP* traffic was specified lower priority by assigning a higher range of the $CW_{min}$ sizes (127 - 255 slots).

As shown in Figure 6.12, during the first run of the simulation, one video application was in the poor (i.e. 0 to 33%) QoS range while one audio and *FTP* connections were in the good range (i.e. 34% to 66%). Following the application of the developed method, all three connections had an excellent QoS (i.e. 67% to 100%). Consequently, using different $CW_{min}$ ranges for audio, video, and *FTP* traffic resulted in a reduction in number of packet collisions and allowed for creating priorities for their transmission. During the first simulation run, the $CW_{min}$ size was set to the default value (i.e., 31) for all traffic. For the rest of simulations, the $CW_{min}$ values were adjusted by the *FIS* system according to the current QoS, previous $CW_{min}$ size ($CW_{min\text{-}prev}$), QoS difference, and collision parameters. The selection of these four input parameters was due to their close relationship with the adjusted $CW_{min}$ ($CW_{min\text{-}opt}$) size. For example, the current QoS input parameter for each type of traffic was chosen to determine the current network performance. However, the previous value of $CW_{min}$ ($CW_{min\text{-}prev}$) was selected in order to help in a decision making process i.e., what the next $CW_{min}$ size (i.e., $CW_{min\text{-}opt}$) should be, lower or higher than the previous one. The third input variable was collision; this was added to provide the *FIS* system with a global knowledge about the network condition by determining the amount of competition among these active stations. The last parameter was the QoS difference and was used to track the QoS variation by providing a positive or a negative sign to the controller. Consequently, combining these parameters together resulted in an accurate adjustment of the $CW_{min}$. The results obtained for this investigation are shown in Table 6.7.

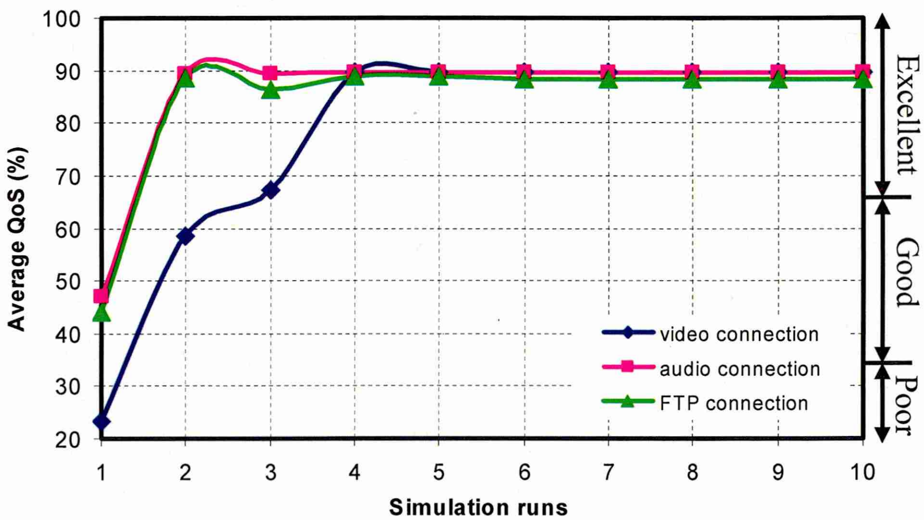



**Figure 6.12:** QoS for video, audio and *FTP* traffic according to the adjusted $CW_{min}$ using *FIS* system.

**Table 6.7:** QoS and the $CW_{min\text{-}opt}$ obtained using the *FIS* system.

| Video | | Audio | | FTP | |
|---|---|---|---|---|---|
| $CW_{min\text{-}opt}$ (slots) | QoS (%) | $CW_{min\text{-}opt}$ (slots) | QoS (%) | $CW_{min\text{-}opt}$ (slots) | QoS (%) |
| 31 | 23.3 | 31 | 47 | 31 | 44 |
| 37 | 58.5 | 19 | 89.5 | 207 | 88.6 |
| 52 | 4.04 | 24 | 89.5 | 200 | 86.3 |
| 52 | 67.3 | 24 | 89.5 | 203 | 88.8 |
| 48 | 89.5 | 25 | 89.5 | 204 | 88.8 |

As indicated in Figure 6.12, the *FIS* adjustment approach reached the maximum QoS levels for the three connections with a small number of simulation runs. However, the linear approach attained the maximum QoS limit (i.e. steady state) with a larger number of iterations. This implied that the *FIS* system was a better control mechanism.

#### 6.4.3.1 Quality of Service Improvements for Multimedia Transmission

In this section multiple video and audio traffic were transmitted. Two main scenarios were discussed when the network was heavily loaded. In the first scenario, three audio and three video connections were considered; and two video and five audio connections were employed in the second scenario.

The QoS achieved for the transmitted audio and video is shown in Figure 6.13. Initially, the video connection was in the poor QoS range (i.e. 0 to 33%) while the other two were in the good range (i.e. 34% to 66%). Following the application of the developed *FIS* method, all three video connections were within the good QoS range with mean QoS equal to 49.2%, 45.7%, and 36.7% for the first, second, and third connections, respectively. Regarding audio connections, primarily one audio application was in the good QoS range and the other two were in the poor range. The method managed to adjust the $CW_{min}$ so that all three audio applications had an excellent QoS range (i.e. 67% to 100%) with an average QoS equal to 75%, 70%, and 75%, for the first, second, and third audio connections, respectively. The reason that the audio applications managed to achieve better QoS is that the specified $CW_{min}$ range for audio applications was lower ($CW_{min}$ range 7 to 31) than that for the video applications ($CW_{min}$ range 15 to 63). Using different $CW_{min}$ ranges for audio and video applications resulted in a reduction in the number of packet collisions and allowed them to establish priorities for their transmission.

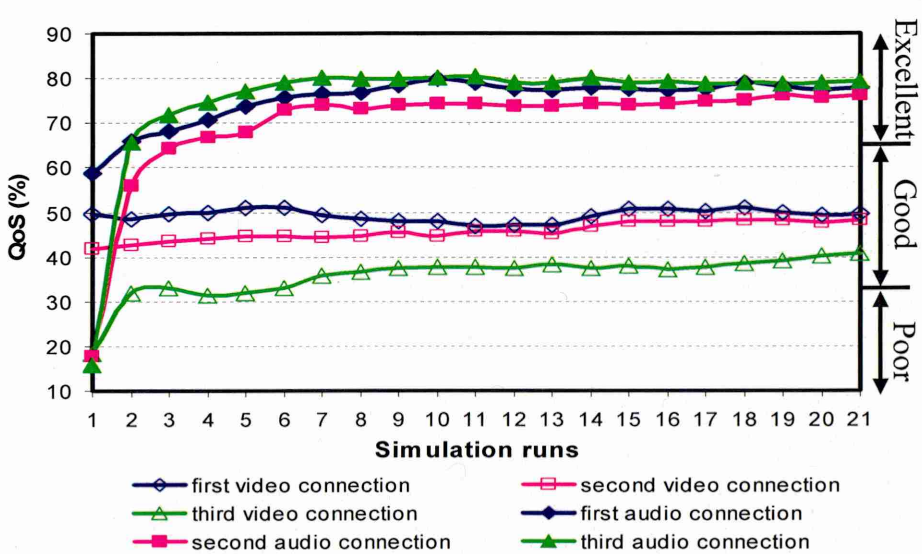



**Figure 6.13:** QoS improvements for a network consisting of three video and three audio connections.

The effectiveness of the developed method is further illustrated in Figure 6.14. Here, a comparison of the percentage QoS achieved by the applications using the standard IEEE 802.11 DCF scheme, linear adjustment method and the *FIS* mechanism is compared. It can be observed that the *FIS* mechanism improved the achieved QoS for 5 out of the 6 applications. The *FIS* system outperformed both, the standard IEEE 802.11 DCF scheme by 45.7% and the linear approach by 9.7%. Additionally, the linear adjustment method outperformed the standard IEEE 802.11 DCF scheme by 36.7%. In the IEEE 802.11 DCF scheme, the three video connections had unequal access to the channel. However, with linear and *FIS* approaches, the three video connections had roughly equal access to the channel. This implied a fairer distribution of channel access. The same observation was made for the three audio connections.

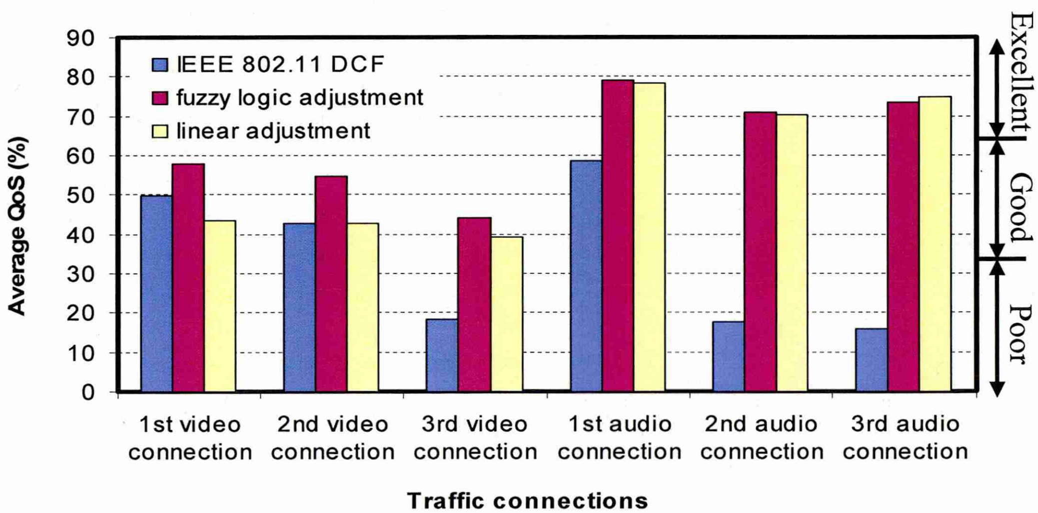



**Figure 6.14:** Assessed QoS for video and audio connections for three schemes, the basic IEEE 802.11 DCF scheme, linear adjustment method, and *FIS* adjustment system.

To further illustrate the effectiveness of the adjustment *FIS* system, another simulation with a larger number of sources was used. This included five audio connections and two video connections with the existence of two background sources. As shown in Figure 6.15, the average QoS for audio and video connections increased gradually for each new simulation run with the new suggested $CW_{min}$ size. Video connections remained in the good QoS range with an improvement of 13%. In contrast, all audio connections began with a good QoS range and then managed to achieve excellent QoS levels with an improvement of 36%. The reason being the specified $CW_{min}$ range for audio connections was lower than that for the video connections.

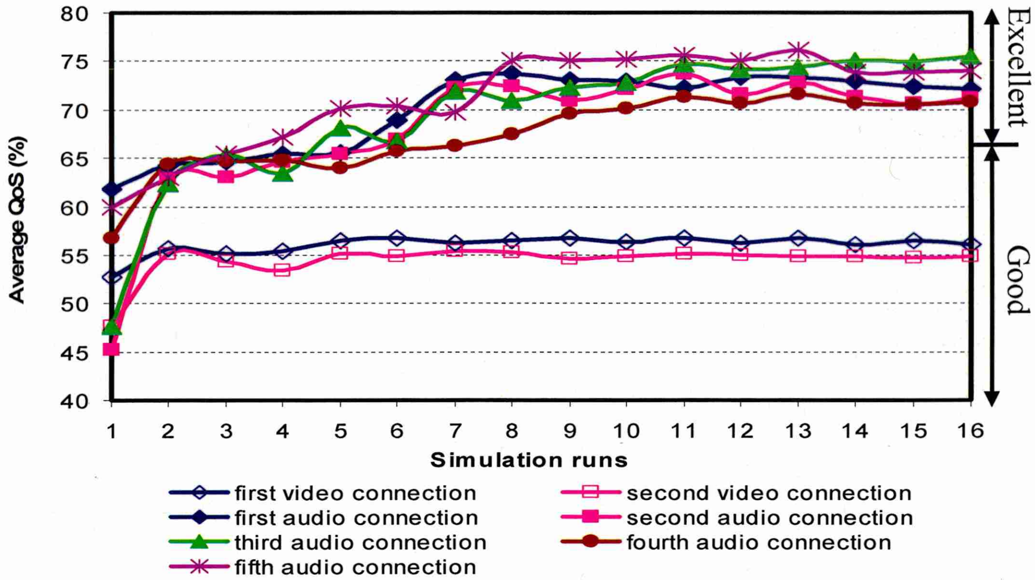



**Figure 6.15:** QoS improvements for a network consisting of two videos, five audios and two background traffic connections.

This previous section demonstrated that the *FIS* system was capable of adjusting the $CW_{min}$ size for the video and audio applications for a small and large number of connections. In the following section, the hybrid genetic-fuzzy approach is discussed when the two MAC protocol transmission parameters i.e. $CW_{min}$ and *DIFS* are adjusted.

### 6.4.4 Hybrid Genetic-Fuzzy Adjustment System

In this section, the findings of a hybrid genetic-fuzzy approach for the optimisation of the $CW_{min}$ and the *DIFS* parameters of the IEEE 802.11 DCF scheme for several configurations are discussed.

The accuracy of the *FIS* assessment approach was examined using typical values for video and audio QoS parameters as given in Table 6.3 (see section 6.4.1). It can be observed that the *FIS* mechanism successfully represented the QoS requirements of the video and audio applications. This confirmed the capability of the *FIS* assessment system for providing the appropriate fitness function for the hybrid genetic-fuzzy approach.

Figures 6.16a and 6.16b show the QoS achieved for the applications when the hybrid genetic-fuzzy was used to adjust the $CW_{min}$ and the *DIFS* for multimedia transmission such as video, audio, and data. Each value represented the average value of QoS obtained by repeating the experiment 10 times with a different starting seed. A comparison of the percentage QoS achieved by the applications using the legacy IEEE

802.11 DCF scheme and that which used the hybrid genetic-fuzzy technique was also provided in Figure 6.16. It can be observed that the hybrid genetic-fuzzy mechanism has improved the QoS for the transmitted application.

In Figure 6.16a, the average QoS for video, audio, and data connections was improved by 67.8%, 39.5% and 40.5%, respectively compared to average QoS obtained when the legacy IEEE 802.11 DCF scheme was used. When the number of connections was increased to 8 (i.e., 3 video and 5 audio) significant improvements were also observed. An improvement of 21.1% was observed in the video connections and 57.1% for audio connections. These improvements in the QoS for a small and medium number of multimedia connections were due to an appropriate selection of $CW_{min}$ and *DIFS* of the proposed genetic-fuzzy approach. This confirmed that the genetic algorithm was capable of providing effective solution for identifying the best $CW_{min}$ and *DIFS* values. The proper selection of $CW_{min}$ and *DIFS* ranges according to the application type and the fitness value provided by the *FIS* assessment mechanism aided this process. The achieved QoS and the optimised $CW_{min}$ and *DIFS* values are shown in Table 6.8.

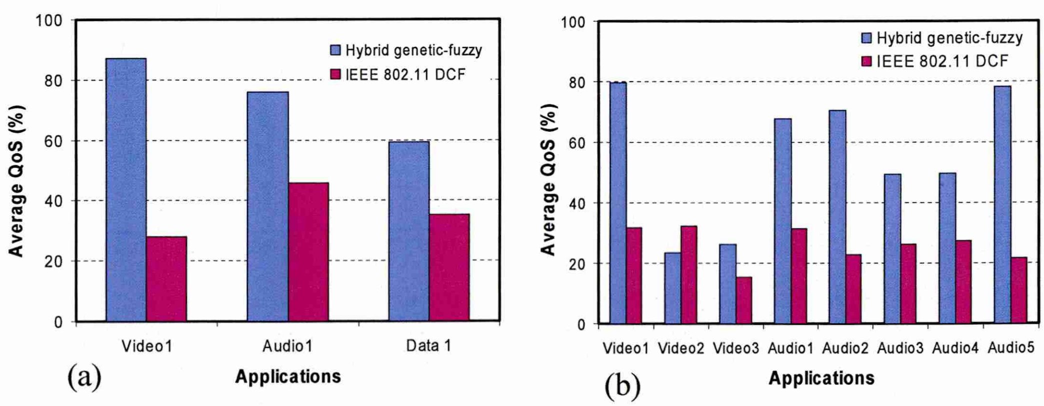



**Figure 6.16:** QoS improvement for a hybrid genetic-fuzzy approach vs. IEEE 802.11 DCF scheme, (a) three multimedia connections, and (b) eight multimedia connections.

To further illustrate the effectiveness of the proposed genetic-fuzzy approach, the number of multimedia connections was increased to 20. The results obtained highlighted the potential improvements in the network performance as depicted in Figure 6.17. The average overall QoS for video, audio and data connections improved by 44.9%, 69.2%, and 55.6%, respectively in comparison with the QoS obtained for the standard IEEE 802.11 DCF scheme. The hybrid genetic-fuzzy approach reduced the probability of collisions and provided a fair access among the contending stations, thus improving the overall network performance. This reduction allowed the QoS for video connections to move from good in case of IEEE 802.11 DCF scheme to the excellent

QoS level when the hybrid genetic-fuzzy system was used. Audio and data connections also managed to move from a poor QoS level to good and excellent QoS levels as shown in Figure 6.17.

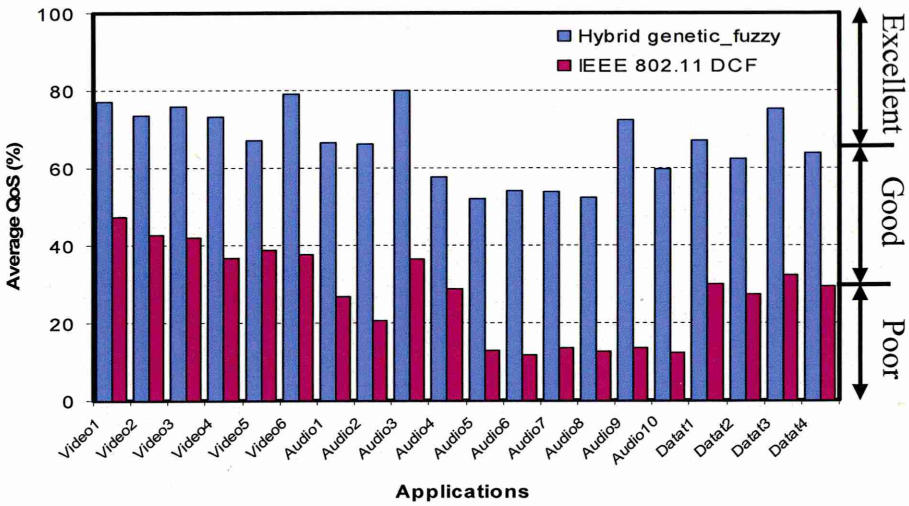



**Figure 6.17:** QoS improvement for a hybrid genetic-fuzzy approach vs. IEEE 802.11 DCF scheme in a network consisting of 20 multimedia connections.

As mentioned earlier in this section, the key issue for these enhancements was the appropriate range of values for the $CW_{min}$ and *DIFS*. Furthermore, the selection of fitness function also played a major role in obtaining a good solution for these two parameters. The fitness function was carefully chosen in order to avoid starvation and to maintain a satisfactory level of QoS. It was represented by the ratio of the minimum QoS for any connection in the network to the maximum required QoS as given in Equation 6.2. As indicated in Table 6.8, the hybrid system was capable of maintaining smaller values of $CW_{min}$ and *DIFS* for audio connections over other connections since audio traffic imposes strict QoS requirements. Although, audio connections had small $CW_{min}$ values, some audio connections achieved smaller average QoS compared with video and data connections. This was because of the small overlap in $CW_{min}$ and *DIFS* ranges between audio and video connections. The average QoS and the adjusted $CW_{min}$ and *DIFS* parameters are summarised in Table 6.8.

The hybrid genetic-fuzzy approach showed a significant improvement in the QoS for the transmitted applications. It was observed that using the proposed *GA* approach to optimise the $CW_{min}$ and *DIFS* values reduced the probability of collisions and provided fair access among the contending stations. This therefore, improved the overall network performance.

**Table 6.8:** QoS, $CW_{min}$, and *DIFS* obtained using the developed genetic-fuzzy system.

| No. of connections | Application | Schemes | | | |
|---|---|---|---|---|---|
| | | Hybrid genetic-fuzzy | | | IEEE 802.11 DCF |
| | | Average QoS (%) | $CW_{min}$ (slots) | *DIFS* ($\mu$sec) | Average QoS (%) |
| 3 | Video1 | 87.1 | 30 | 45 | 28 |
| | Audio1 | 75.9 | 19 | 55 | 45.9 |
| | Data 1 | 59.3 | 176 | 110 | 35.2 |
| 8 | Video1 | 79.6 | 21 | 60 | 31.8 |
| | Video2 | 23.3 | 36 | 50 | 32.2 |
| | Video3 | 26.2 | 27 | 50 | 15.5 |
| | Audio1 | 67.7 | 14 | 30 | 31.4 |
| | Audio2 | 70.6 | 22 | 40 | 22.8 |
| | Audio3 | 49.3 | 22 | 35 | 26.2 |
| | Audio4 | 49.8 | 38 | 30 | 27.5 |
| | Audio5 | 78.2 | 12 | 45 | 21.7 |
| 20 | Video1 | 77.1 | 33 | 60 | 47.3 |
| | Video2 | 73.6 | 57 | 70 | 42.8 |
| | Video3 | 75.8 | 30 | 55 | 42 |
| | Video4 | 73.3 | 42 | 50 | 36.8 |
| | Video5 | 67 | 24 | 60 | 38.7 |
| | Video6 | 79 | 39 | 45 | 37.8 |
| | Audio1 | 66.6 | 8 | 55 | 26.8 |
| | Audio2 | 66.3 | 12 | 55 | 20.5 |
| | Audio3 | 80.1 | 20 | 40 | 36.4 |
| | Audio4 | 57.7 | 24 | 55 | 28.9 |
| | Audio5 | 52 | 30 | 30 | 13.1 |
| | Audio6 | 54 | 28 | 25 | 11.8 |
| | Audio7 | 53.9 | 26 | 30 | 13.4 |
| | Audio8 | 52.3 | 8 | 35 | 12.8 |
| | Audio9 | 72.3 | 10 | 35 | 13.4 |
| | Audio10 | 59.7 | 12 | 30 | 12.3 |
| | Datat1 | 66.9 | 152 | 105 | 30 |
| | Datat2 | 62.5 | 156 | 95 | 27.4 |
| | Datat3 | 75.4 | 244 | 145 | 32.3 |
| | Datat4 | 63.8 | 240 | 85 | 29.5 |

### 6.4.5 Implication of the Developed Approaches in Real System

The ability of implementing the proposed system in a real system is discussed in this section. The measurements of QoS parameters and the adjusted MAC protocol transmission parameters such $CW_{min}$ and *DIFS* can be measured in both, the sender and the receiver. This can be conducted by exchanging a small number of control messages between the communicating pair at specific time duration. The source sends a control message to the destination to start measuring the QoS parameters such as the time received for the sent packets. Hence, the receiver starts recording a sample of traffic that is sufficient to represent the whole population. After a predefined time interval, the source sends a control message to the receiver to send the recorded QoS information. During this period, the source has, the time and the number of sent packets, and the time and the number of received packets. Based on this information, the source is capable of determining the QoS and is able to adjust the required MAC protocol transmission parameters. If the control message is lost between the communicating pair, the source waits for an expiry period and then sends another control message to the destination to send the recorded information again or to start new measurements.

The difference between the actual data and the sampled one was statistically analysed using the t-test (Graph, 2005). The results indicated that the sampling method could represent the whole population since there was no statistical difference between the parent population and the sample version. More information about the implementation and the statistical analysis are provided in Appendix C (see Appendix C.2).

## 6.5 Summary

In this chapter a *FIS* mechanism was proposed to assess the QoS for multimedia transmission over wireless networks. Three schemes, fuzzy logic, linear, and hybrid genetic-fuzzy were proposed to adjust two main MAC protocol transmission parameters $CW_{min}$ and *DIFS*. The key issue of these approaches was to guarantee the different QoS requirements for different traffic classes, while simultaneously ensuring that the limited channel bandwidth is utilised efficiently. This chapter first reviewed the state of the art in section 6.2, followed by a detailed description of the proposed approaches including the simulation model. A full description of the main findings was given in section 6.4.

The study indicated that the developed *FIS* system was capable of assessing the network QoS for multimedia applications. The linear scheme and the application of fuzzy logic and genetic-fuzzy methods resulted in significant improvements in the network QoS. The implication of the proposed schemes in real networks has been examined. However, using a systematic sampling method there was no significance statistical discrepancy between the actual data and the sampled version.

Combining the *FIS* assessment mechanism with other techniques such as the linear scheme will be valuable methods in providing further enhancements in the protocol performance and in the QoS over the IEEE 802.11 MAC protocol. This will be discussed in the following chapters.

# *CHAPTER 7*

# Collision Ratio and Collision Rate Variations Schemes to Improve QoS in the IEEE 802.11 MAC Protocol

## 7.1 Introduction

This chapter proposes two mechanisms for improving the IEEE 802.11 DCF protocol performance and enhancing QoS in IEEE 802.11 DCF scheme. These are Ratio based and Collision Rate Variation (*CRV*) schemes. The Ratio based scheme uses the collision rate value of the current and the past history of the network conditions to adaptively adjust the Contention Window (*CW*) size for each individual station. The *CRV* scheme employs the collision rate and collision rate variation values to dynamically adjust the *CW* and the *DIFS* values locally for each individual station according to the current and previous network conditions. The aim of developing these approaches is to reduce the probability of collisions among the contending stations in a heavily loaded network in an attempt to improve QoS. The proposed schemes are evaluated and compared with the standard IEEE 802.11 DCF and the Exponential Increase Exponential Decrease (*EIED*) schemes.

The relevant studies are described in the next section. Section 7.3 introduces a detailed description of the Ratio based and *CRV* schemes. The simulation model is presented in section 7.4. The results obtained are analysed and discussed in section 7.5. A chapter summary is given in section 7.6.

## 7.2 Previous Studies for Adjusting MAC Protocol Parameters

Several algorithms that dynamically changes the value of *CW* to improve the performance of the IEEE 802.11 DCF protocol have been proposed and are described in (Bharghavan et *a.l*, 1994), (Baldwin et *al.*, 1999), (Sobrinho and Krishnakumar, 1999), (Cali$_{(a)}$ and Gregori, 2000), (Bianchi$_{(a)}$ and Tinnirello, 2003), (Qiang et *al.*, 2003), (Kuo and Jay, 2003), (Kwon et *al.*, 2003), (Zhao et *al*, 2003), (Gannoune, 2004), (Deng et *al.*, 2004), and (Kuppa and Prakash, 2005). For example, in (Deng et *al.*, 2004), the Linear/Multiplicative Increase and Linear Decrease (*LMILD*) backoff algorithm is presented. In the *LMILD* scheme, colliding stations increase their *CW* multiplicatively, while other stations overhearing the collisions increase their *CW* linearly. After successful transmission, all stations decrease their *CW* linearly. An adaptive DCF scheme was proposed in (Kuppa and Prakash, 2005). The proposed approach is based on adjusting the backoff procedure based on the knowledge of collision and the number of freezes time the backoff timer of the station experiences. The study showed that, the proposed scheme outperformed the IEEE 802.11 DCF scheme in terms of throughput.

Several recent methods have improved the performance of IEEE 802.11 DCF by either modifying the *CW* or adjusting the value of Inter Frame Space (*IFS*) (see section 2.11 in Chapter 2). For instance the variation of the Arbitrary Inter Frame Space (*AIFS*) between stations leads to a lower probability of collisions and a faster progressing of the backoff counter as reported in (Robinson and Randhawa, 2004). Ksentini et *al.*, (2004) presented the *AIFS* as a technique for providing service differentiation between different classes in the IEEE 802.11e protocol. In (Zhang and Ye, 2004), the length of *DIFS* was adopted as a differentiation mechanism. In their scheme, the *DIFS* length was calculated based on the ratio of estimated transmission rate to the total transmission rate. Their scheme imposed major modifications to the IEEE 802.11 DCF scheme in which the single queue was split into two queues. Their results showed that using a variable length *IFS*, service differentiation can be achieved. In (Pattara-atikom, 2004), the adjusted *IFS* parameter with other parameters such as quantum rate and deficit counter was used to provide QoS mechanism. Their results showed that QoS can be supported using an adjusted *IFS* length. In (Sung and Yun, 2006), the authors proposed a method for optimising MAC parameters in the *EDCF* protocol, such as *CW* and *DIFS*. The proposed method improved throughput and delay as compared with the IEEE 802.11e. However, it was based on storing several network configurations using a database which imposed high computational overhead.

Most of the discussed schemes require an exchange of information between stations. They also require sophisticated computations as the case in (Bianchi et *al.*, 1996), ($Cali_{(a)}$ et *al.*, 2000), (Qiao and Shin, 2003), and (Sung and Yun, 2006). Other schemes impose major modifications to the structure of the IEEE 802.11 DCF as the case in (Choi et *al.*, 2005). Most studies only consider one or two of the QoS parameters. They only depend on the current conditions of the network without considering the past history. In this chapter, a Ratio based and *CRV* schemes are proposed to overcome these shortcomings. They are as simple as the *BEB* to implement while significantly outperformed the IEEE 802.11 DCF and the *EIED* schemes.

## 7.3 Description of the Approach

The standard IEEE 802.11 DCF protocol adjusts its *CW* value based on the current state of transmission, i.e. it doubles the *CW* value upon unsuccessful transmission and resets to the $CW_{min}$ upon successful transmission (IEEE, 1999). The DCF scheme does not consider the past history of the network or the readily available information. For the *EIED* scheme, *EIED* $(ri, rd)$ is used to denote the amount of increase and decrease in the *CW* size after successful and unsuccessful transmission. If collisions occurred, the new *CW* is increased by the multiplication factor $ri$ and after successful transmission the new *CW* is decreased by the multiplication factor $rd$. In this chapter, the value of 2 is chosen for each $ri$ and $rd$ as one of the possible cases of the *EIED* scheme (Song et *al.*, 2003). The *EIED* scheme is used in the performance comparison, because it is not as aggressive as the standard IEEE 802.11 DCF when the *CW* is reset to $CW_{min}$ after successful transmission.

In order to make the protocol behave correctly, the *CW* and the *DIFS* should be adaptively adjusted to adapt to the dynamic changes in the number of contending stations and in the amount of traffic over time. This can be achieved by tuning the *CW* and *DIFS* values after each successful and unsuccessful transmission. These adjustments are carried out locally for each station at runtime. A detailed explanation of how the *CW* and *DIFS* values are adjusted is given in the following sections.

### 7.3.1 Ratio Based Scheme

#### 7.3.1.1 Case for Successful Transmission

After each successful transmission, the DCF mechanism resets the *CW* of the station to its $CW_{min}$ (i.e. $CW_{new} = CW_{min}$) ignoring the network conditions. This action by the successful station causes frequent collisions especially when the network is very large and heavily loaded because of a small value of *CW*. This agrees with the fact that when a collision occurs, a new one is likely to take place in the near future since the collided packet requires retransmission which causes extra overhead. For this reason Ratio based and the *CRV* schemes are proposed in order to mitigate burst collisions. In the Ratio based scheme, the *CW* size is adaptively adjusted as follows:

The *CW* size is adjusted after computing the current collision ratio for each station, since collisions can provide a good indication about the level of contention in the network. The current collision ratio is computed using the number of collisions and the number of successfully acknowledged transmissions extracted from the history window ($wi$) as shown in Equation 7.1. The history window ($wi$) is a sliding array that contains number of sent packets including part of the history.

$$R_{current}^{wi}[N] = \frac{Num(collisions_{wi}[N])}{Num(collision_{wi}[N]) + Num(successful_{wi}[N])} \qquad (7.1)$$

Where, $Num(collisions_{wi}[N])$ is the number of collisions for station $N$ that is extracted from the history window $wi$, $Num(successful_{wi}[N])$ represents the number of packets that have been successfully acknowledged for station $N$ that is extracted from the same history window $wi$, $R_{current}^{wi}[N]$ is the current collision ratio of station $N$. The $R_{current}^{wi}[N]$ value is computed based on the number of collided packets and the number of successfully received packets that are extracted from the history window $wi$. The $R_{current}^{wi}[N]$ value is always in the range of [0, 1].

In order to maintain a continuous knowledge about the past history of the transmission, the sliding window $wi$ is adopted. To reduce or to alleviate the random fluctuations in the computed $R_{current}^{wi}[N]$ an Exponentially Weighted Moving Average (*EWMA*) (Crowder, 1989) is used to smooth the series of collision ratios (i.e. $R_{current}^{wi}[N]$ value) as given in Equation 7.2.

$$R_{average}^{wi} = (1-\lambda) * R_{current}^{wi} + \lambda * R_{average}^{wi-1} \qquad (7.2)$$

Where $R_{current}^{wi}$ denotes the current or instantaneous collision ratio for station $N$ ; $\lambda$ stands for a weighting factor which determines the memory size used in the average process; $R_{average}^{wi-1}$ represents the previous average collision ratio that is computed from the previous history window ($wi-1$); while $R_{average}^{wi}$ is the average collision ratio at the current history window $wi$.

The instantaneous collision ratio $R_{current}^{wi}$ and the average collision ratio $R_{average}^{wi}$ are calculated based on the size of the total number of packets sent in $wi$. The size of $wi$ is selected not to be so large as to obtain a reasonable estimation about the network status. However, it should not be too small in order to get sufficient knowledge about the readily available information of each individual station. Using a sliding window ensures that the system always keeps a continuous tracking for the history of the total number of packets sent. However, the size of $wi$ and the weighting factor $\lambda$ are selected according to an extensive set of simulations carried out with several network topologies and different traffic loads. This was done in order to achieve a trade-off value between throughput and delay and in order to provide a good balance between removing short term fluctuations impact and capturing long term trends. Upon obtaining the value of $R^{wi}_{average}$ described in Equation 7.2, the new $CW$ size for station $N$ after successful transmission is computed based on Equation 7.3:

$$CW_{new}[N]=CW_{new-1}[N]\left(1-\frac{R^{wi}_{average}}{f}\right) \tag{7.3}$$

Where $CW_{new}[N]$ is the new computed contention window, $CW_{new-1}[N]$ is the previous computed $CW$, and $f$ a scaling factor (the impact of this factor is discussed in section 7.5.1). Hence after, the $CW$ size is selected by the station is obtained using Equation 7.4. Equation 7.4 also guarantees that the $CW_{new}[N]$ size does not go below the minimum contention window (i.e. $CW_{\min}[N]$).

$$CW[N] = Max(CW_{\min}[N], CW_{new}[N]) \tag{7.4}$$

#### 7.3.1.2 Case for Collision

In the legacy IEEE 802.11 DCF (IEEE, 1999), the $CW$ is doubled after each collision. If the maximum limit known as the maximum Contention Window ($CW_{max}$) is reached, the collided station remains at $CW_{max}$. In the Ratio based scheme, after each collision, the new $CW$ of the collided station is computed according to Equation 7.5:

$$CW_{new}[N]=CW_{new-1}[N]\left(1+f*R^{wi}_{average}\right) \tag{7.5}$$

Where $CW_{new}[N]$, $f$, $CW_{new-1}[N]$ and $R^{wi}_{average}$ are as discussed in section 7.3.1.1 (successful transmission case). The selected $CW$ value for the station is obtained using Equation 7.6. Note that $CW_{\max}[N]$ is the maximum contention window for station $N$. Equation 7.6 also ensures that the $CW_{new}[N]$ size does not exceed the maximum contention window (i.e. $CW_{\max}[N]$).

$$CW[N]=Min(CW_{\max}[N], CW_{new}) \tag{7.6}$$

### 7.3.2 Collision Rate Variation Scheme

The Collision Rate Variation ($CRV$) scheme is based on the variation in the collision ratio that was discussed in the Ratio based scheme. Therefore, this scheme is introduced in order to obtain further knowledge about the changes in the network conditions by monitoring the variations in the current and previous values of the collision ratio. In the *CRV* scheme, the *CRV* value of each station is calculated based on Equation 7.7

$$CRV[N] = R_{current\_average}[N] - R_{previous\_average}[N] \tag{7.7}$$

Where, $CRV[N]$ is the collision ratio variation for station $N$, $R_{current\_average}[N]$ and $R_{previous\_average}[N]$ the current and the previous average collision ratio.

According to Equation 7.7, the *CRV* values are allowed to vary between -1 to 1. The variation of *CRV* values is used to adjust the *CW* size and the *DIFS* of each station. This is discussed for each individual parameter in the following two sections.

#### 7.3.2.1 Contention Window Adjustment Using Collision Rate Variation Scheme

In this section the operation of *CRV* scheme to adjust the *CW* size for each individual station based on the variation in the *CRV* value is explained. Using the *CRV* scheme, the new *CW* size (i.e. $CW_{new}[N]$) is updated using Equation 7.8.

$$CW_{new}[N] = CW_{new-1}[N](1 + f * CRV[N]) \tag{7.8}$$

Where, $f$ refers to a scaling factor (see section 7.5.1 for more details), $CW_{new}[N]$ is the computed $CW$ for a station $N$, $CW_{new-1}[N]$ stands for the previous $CW$ size, and $CRV[N]$ is the computed collision ratio variation.

If the computed value of *CRV* using Equation 7.7 is negative, this implies that the current number of collisions is less than previous number of collisions, therefore, the new *CW* size (i.e. $CW_{new}[N]$) is used by the station after each successful transmission and called as $CW_{success}[N]$. To ensure that the new $CW$ for the successful station (i.e. $CW_{new}[N]$) does not go below the minimum contention window of that station (i.e. $CW_{\min}[N]$), the *CW* size for the successful station is limited by Equation 7.9.

$$CW[N] = Max(CW_{new}[N], CW_{\min}[N]) \tag{7.9}$$

If the computed *CRV* value is positive, this implies that the current number of collisions for the station is more than the previous number of collisions for the same station. As a result, the new *CW* size (i.e. $CW_{new}[N]$) is used by the station after each collision and is called $CW_{collision}[N]$. To ensure that the *CW* for the collided station (i.e. $CW_{new}[N]$) does not exceed the maximum contention window size of that station (i.e. $CW_{\max}[N]$), the *CW* size for a station involved in a collision is limited by using Equation 7.10.

$$CW[N] = Min(CW_{new}[N], CW_{max}[N]) \quad (7.10)$$

#### 7.3.2.2 Adaptive Distributed Inter Frame Space Adjustment

The *CRV* value is used to adjust the *CW* size after a successful and unsuccessful transmission. In this section the *CRV* scheme is used to adjust the *DIFS* length, since it provides negative and positive values within [-1 to 1] range. The new scheme is called Adaptive Distributed Inter Frame Space (*ADIFS*). The variation in the *CRV* value leads to a variable length *DIFS* between 20 $\mu$sec and 140 $\mu$sec. The minimum *DIFS* value is 20 $\mu$sec and the maximum *DIFS* is 140 $\mu$sec. The minimum value is selected to be longer than the Short Inter Frame Space (*SIFS*) that is specified for control frames such as acknowledgment frame (IEEE, 1999), while the maximum value is chosen in order to minimise the wasted time slots by avoiding an excessively long defer of data packets. The length of *DIFS* is calculated according to Equation 7.11.

$$ADIFS_{new}[N] = ADIFS_{new-1}[N] + (CRV[N] * ADIFS_{new-1}[N]) / f \quad (7.11)$$

Where, $ADIFS_{new}[N]$ is the new calculated *DIFS* of a station $N$, $ADIFS_{new-1}[N]$ is the previous *ADIFS*, $CRV[N]$ represents the computed collision rate variation value and $f$ is a scaling factor that is used to assign a proper value of *DIFS* within the specified range.

In Equation 7.11, when the $CRV < 0$, it implies that the current collision ratio is smaller than the previous collision ratio, thus the $CRV$ value is considered as a deescalating factor that leads to a reduction in the $ADIFS_{new}[N]$ length in order to reduce the waiting time. When the $CRV > 0$, it implies that the current collision ratio is greater than the previous one, therefore the $CRV$ value is considered as escalating factor that leads to an increase in the $ADIFS_{new}[N]$ length in order to reduce the probability of collisions.

In the $CRV$ scheme, as the $CRV$ value of each station is likely to be different, the computed value of $ADIFS_{new}[N]$ is also different. According to this, contending stations have different opportunities to access the medium. Moreover, the $CRV$ and the scaling factor $f$ values have a direct impact on the calculated $ADIFS_{new}[N]$, therefore, having different values of $CRV$ and $f$, service differentiation could be applied among stations located either in the same Independent Basic Service Set (*IBSS*) or within multi-hop networks. This issue will be discussed in the Chapter 8.

The first term of Equation 7.11 determines the minimum and maximum lengths of the calculated $ADIFS_{new}[N]$. The maximum length is obtained when the $CRV$ value is equal to 1; whereas the minimum length of $ADIFS_{new}[N]$ is obtained when $CRV$ is equal to -1. However, any arbitrary value of $CRV$ in the range of [-1 to 1] can be employed to achieve fair access to the medium.

When a collision occurs the length of $ADIFS_{new}[N]$ increases because of a positive $CRV$ value. Thus, the higher the value $CRV$ is, the longer the length of $ADIFS_{new}[N]$ and the waiting time. This indicates that more stations are contending to access the medium. A shorter length of $ADIFS_{new}[N]$ can be obtained after consecutive successful transmission and when $CRV < 0$. This provides the wireless medium access through earlier decrease of the backoff counter.

Unlike the legacy IEEE 802.11 DCF protocol where the *DIFS* value is fixed regardless of the number of collisions and how much data has been sent; the length of *DIFS* in the *ADIFS* scheme is calculated dynamically and instantaneously for each station locally and independently at runtime of the simulation. The variable length of $ADIFS_{new}[N]$ can help in a significant reduction in the number of collisions. It can also reduce the value of the average delay of time-sensitive application through reducing the waiting time before packet transmission.

In the second term of Equation 7.11, the value of $CRV$ can determine the length and the variation of $ADIFS_{new}[N]$, since it has a positive and a negative values between 1 and -1. The scaling factor $f$ can be used to keep the $ADIFS_{new}[N]$ length within the specified range, i.e. between 20 and 140 $\mu$sec.

In the implementation of Ratio based and $CRV$ schemes, each station adjusts its *CW* and *DIFS* values locally and independently. Some of these stations may have small values of *CW* and *DIFS* that cause a selfish access to the medium. This can occur after obtaining a short period of *DIFS* and a small duration of backoff interval for several consecutive times. This causes starvation for other stations in the network. In such a case, a monitoring mechanism is used to observe the collision ratio and $CRV$ values of past transmission of these selfish stations and it also observes the *CW* and *DIFS* of the starved stations. If the selfish stations have small values of *CW* and *DIFS* while starved stations still have large values, a penalty is applied for the selfish stations from accessing the medium by resetting the *CW* and *DIFS* values of the starved station to the minimum. Therefore, fair access to the medium can be achieved. Part of the pseudo code of the Ratio based and *CRV* schemes are illustrated in Figures 7.1, 7.2 and 7.3.

```
Ratio based scheme when collision occurs
[wi] = 20, flag = 0; f = 3, CWmin = 31; CWmax =1023;
If ( history window4 == [wi] packets) {
        Count the number of collided packets from wi
        Count the number of successfully received acknowledgment packets from wi
        Compute the current collision ratio using Equation 7.1;
        Compute the average collision ratio using Equation 7.2;
        //to avoid starvation for some stations monitor the behaviour of each individual station.
        If (CW size is greater than > (f +1) * CWmin ) {
                Increment flag;
                If(flag ==f +1) {
                        CW = CW min ;
                } else {
                        flag = 0;
                        Compute CW size using Equation 7.5;
                        Apply Equation 7.6;
                }
        }
}
```

**Figure 7.1:** Ratio based scheme in case of unsuccessful transmission.

```
Ratio based scheme when successful transmission occurs
[wi] = 20, flag = 0; f = 3, CWmin = 31; CWmax =1023;
If ( history window == [wi] packets) {
        Count the number of collided packets from wi;
        Count the number of successfully received acknowledgment packets from wi;
        Compute the current collision ratio using Equation 7.1;
        Compute the average collision ratio using Equation 7.2;
        Reset the collision counter;
        Reset the success counter;
        //to avoid starvation for some stations monitor the behaviour of each individual station.
        If (CW size is greater than > (f +1) * CWmin ) {
                Increment flag;
                If(flag ==f +1) {
                        CW = CW min ;
                } else {
                        flag = 0;
                        Compute CW size using Equation 7.3;
                        Apply Equation 7.4;
                }
        }
}
```

**Figure 7.2:** Ratio based scheme in case of successful transmission.

---

[4] A sliding window is considered for each update which provides smooth variation of $R^{wi}_{current}$ value.

```
Collision Rate Variation (CRV) scheme for adjusting CW size

[wi] = 20, flag = 0; f = 3, CW_min = 31; CW_max =1023; CRV =0 ;
If (history window == [wi] packets) {
        Count the number of collided packets from wi;
        Count the number of successfully received acknowledgment packets from wi;
        Compute the current collision ratio using Equation 7.1;
        Compute the average collision ratio using Equation 7.2;
        Reset the collision counter;
        Reset the success counter;
        Compute the collision rate variation value for each station using Equation 7.7;
        If (CRV[N] < 0) {
                //to avoid starvation for some stations monitor the behaviour of each station.
                If (CW size is greater than > (f +1) * CW_min ) {
                Increment flag;
                        If(flag ==f +1) {
                                CW = CW_min ;
                        } else {
                                flag = 0;
                                Compute CW size using Equation 7.8;
                                Apply Equation 7.9;
                                Use the computed CW size after successful transmission as
                                 Follows: CW_success[N] = CW[N];
                        }
                }
        }
        else If (CRV[N] > 0) {
                //to avoid starvation for some stations monitor the behaviour of each station.
                If (CW size is greater than > (f +1) * CW_min ) {
                Increment flag;
                        If(flag ==f +1) {
                                CW = CW_min ;
                        } else {
                                flag = 0;
                                Compute CW size using Equation 7.8;
                                Apply Equation 7.10;
                                Use the computed CW size after unsuccessful transmission as
                                follow:  CW_collision[N] = CW[N];
                        }
                }
        }
}
```

**Figure 7.3:** Collision Ratio Variation (*CRV*) scheme.

In the Ratio based and *CRV* schemes, the *CW* value does not reset to $CW_{min}$ after successful transmission except in the following cases: (i) at the beginning of the transmission where the station starts with its initial *CWmin*, (ii) when the station experiences a large *CW* size (i.e., $CW[N] > (f+1)*CW_{\min}[N]$), in order to avoid starvation, and (iii) when the number of retransmission attempts of the collided packets reaches the maximum limit (i.e. station experiences high value of *CW*).

In the *CRV* scheme, the *CW* and *DIFS* values are adjusted according to the variations in the collision ratio. A negative *CRV* determines whether the station experiences less collision, indicating a more successful transmissions taking place while a positive value determines that the station experiences more collisions. Therefore, the use of the *CRV* scheme provides a good guide for selecting the *CW* value (i.e. $CW_{success}$ *or* $CW_{collision}$) and a good indicator to the amount of increase or decrease of the *DIFS* length. Therefore, this feature combined with the dynamic adjustments of the *CW* and *DIFS* values will be used for providing service differentiation in single and multi-hop networks. This will be discussed in Chapter 8.

## 7.4 Simulation Model

To evaluate the validity of the Ratio based and *CRV* schemes and compare their performance with the standard IEEE 802.11 DCF and *EIED* schemes in terms of QoS parameters, the *NS-2* simulation package was used (NS, 2006). Two network models with different scenarios were used for the simulation. The first model used the topology shown in Figure 4.2d (see Chapter 4). The stations in this model transmitted different *CBR* and *VBR* traffic based on the selected scenario. The second network model employed the topology shown in Figure 4.2b (see Chapter 4). The parameters used in these simulations were based on the IEEE 802.11 network configurations (Choi, 2001) and they are provided in Table 4.1 (see Chapter 4).

In all simulations, *CBR* and *VBR* traffic sources were employed. The packet sizes used for *CBR* traffic were 512, 160 and 200 bytes. The packet generation rates were 384 Kbps, 64 Kbps and 128 Kbps. The *VBR* traffic had a variable frame size with a mean value of 800 bytes and a variable inter-packet interval with a mean value of 4.3 msec.

The simulations were performed for several scenarios in order to critically evaluate the performance of the proposed schemes by means of a comparison with the IEEE 802.11 DCF and the Exponential Increase Exponential Decrease (*EIED*) schemes. These scenarios included varying the network size (small, medium, and large network sizes), traffic load (light, medium, and heavy load), traffic type (*CBR* and *VBR*), and the variation in the number of active stations over time.

## 7.5 Results and Discussion

This section provides the simulation results for the proposed methods and compares them with both, the *EIED* with *ri* and *rd* factors equal to 2 and the standard IEEE 802.11 DCF schemes. The simulations include: different traffic load (light, medium, and heavy loads), various traffic types (*CBR* and *VBR*), different network sizes (5, 10, and 20 connections), both MAC protocol access mechanisms (basic and *RTS/CTS*) and different topologies (single-hop and multi-hop).

This section is split into four main subsections. Section 7.5.1 introduces the sensitivity analysis and parameter tuning of the Ratio based and *CRV* schemes. Section 7.5.2 discusses the results obtained when the Ratio based and *CRV* schemes were used to adjust the *CW* size. Section 7.5.3 outlines the results obtained when the *CRV* scheme was used to adjust the *DIFS* length. Section 7.5.4 shows the performance of Ratio based scheme when the *RTS/CTS* access mechanism was considered.

### 7.5.1 Parameters Tuning

In this section, the influence of the history window size ($wi$), scaling factor ($f$) and weighting factor ($\lambda$) is investigated. The results demonstrated that the right combination of these parameters could lead to better performance. In order to ensure that these parameters were appropriately selected, several simulations were carried out with different topologies and different traffic loads. In this discussion, a network with 10 stations transmitted *CBR* traffic to 10 destinations, and heavy and medium load cases was used. Average throughput and average delay were considered as the main metrics.

The weighting factor $\lambda$ was varied over the range 0 to 1. Figures 7.4a and 7.4b depict the average delay and average throughput as a function of the weighting factor $\lambda$, respectively. Every single point of the results obtained represent an average of 10 simulations in order to avoid the bias of random number generation. It can be observed from Figure 7.4a that the lowest average delay corresponded to $\lambda = 0.65$. Figure 7.4b shows that the highest average throughput corresponded to $\lambda = 0.7$. Consequently, the value of $\lambda = 0.6$ was selected since it achieved a trade-off between average throughput and average delay.

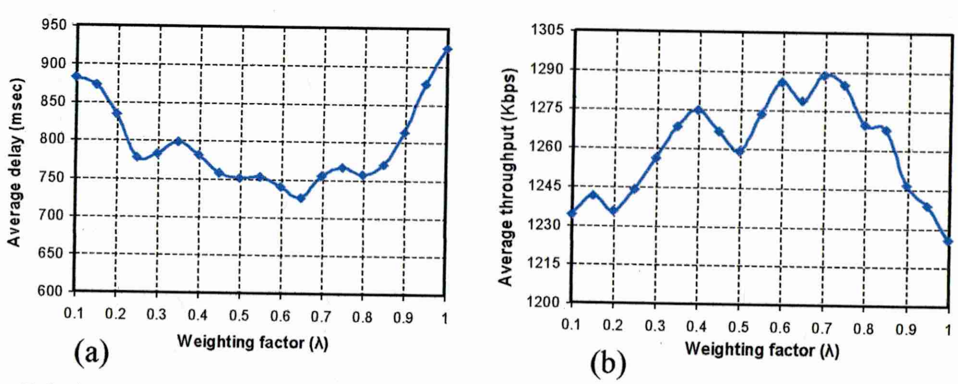



**Figure 7.4:** Average delay and average throughput as a function of weighting factor ($\lambda$), (a) average delay and (b) average throughput.

Figures 7.5a and 7.5b depict the average delay and average throughput as a function of the scaling factor $f$ for the heavy and medium load cases. The value of $f$ was varied over the range 1 to 10 with an increment of 1. A significantly low value of $f$, e.g. $f=1$, resulted in high values of delay and low values of average throughput in heavy and medium load cases. A significantly high value of $f$, e.g. $f=10$, resulted also in high values of average delay and a low value of average throughput. According to Figures 7.5a and 7.5b, a value of $f$, i.e. around 3, provided a trade-off between the average delay and average throughput. The value of $f$, i.e. around 3, provided the lowest average delay and the highest average throughput in heavy and medium load cases as depicted in Figures 7.5a and 7.5b. A variation in the scaling factor $f$ could result in a significant variation in the network performance. Therefore, this feature will be used with other parameters such as packet loss rate for providing service differentiation in single and multi-hop networks (see Chapter 8). In this chapter, the value of $f=3$ was considered for the following simulations.

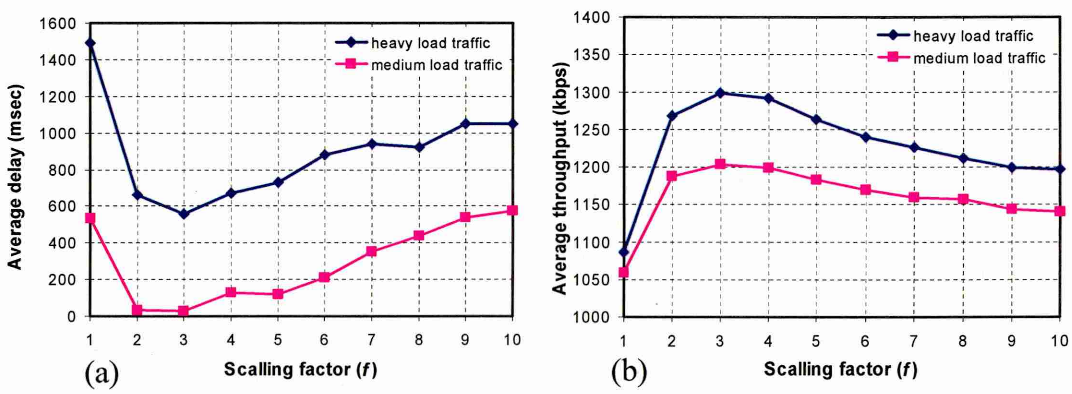



**Figure 7.5:** Average delay and average throughput as a function of scaling factor ($f$), (a) average delay and (b) average throughput.

Figures 7.6a and 7.6b demonstrate the effect of the history window $wi$ on the average delay and average throughput in heavy and medium load cases. The size of the history window $wi$ was varied over the range 5 to 50 with an increment of 5. The simulation results of average delay are shown in Figures 7.6a for heavy and medium load cases. A value of $wi$ around 20 provided the lowest values of average delay in heavy and medium load cases. According to Figure 7.6b, the highest average throughput was achieved when the $wi$ size was around 25 in the heavy load case, while the highest average throughput was achieved when the ($wi$) size was around 20 for the medium load case. Thus, a history window $wi$ of around 20 was selected for the following simulations as it provided a balance between the heavy and medium load cases. Furthermore, it provided a trade-off between the average delay and average throughput.

Note that, the $wi$ values around 20 could maintain small values of average delay compared to smaller values of $wi$. Therefore, a $wi$ size equal to 20 was used in this chapter as indicated in the previous paragraph. Too smaller value of $wi$ were probably insufficient to provide adequate information about the current and the previous network conditions. Moreover, too large values of $wi$ could cause late updates in the adaptation process (i.e. when adjusting *CW* and *DIFS*) in which performance degradation might occur. Subsequently, choosing an appropriate size of $wi$ resulted in minimum average delay and maximum average throughput.

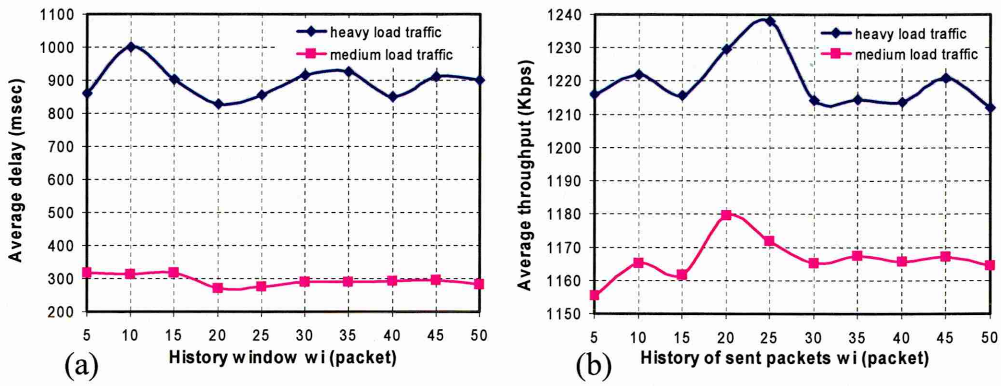



**Figure 7.6:** Average delay and average throughput as a function of history window ( $wi$ ), (a) average delay and (b) average throughput.

## 7.5.2 Performance Evaluation of CW Adjustments Using CBR Traffic

### 7.5.2.1 Light Load Traffic

Here, a 25% of the channel capacity (i.e., 2 Mbps) was considered. This was represented by 500 kbps equally split between the connections in the network. Each connection transmitted 100 Kbps, 50 Kbps, and 25 Kbps corresponding to 5, 10, and 20 connections, respectively.

In order to illustrate the relationship between the four approaches, the values of QoS parameters during the initial period of the network operation are different to those when the network is settled. This is because during the initial period, more control and management frames are transmitted. As indicated in Table 7.1, the four schemes in five connections scenario were able to achieve a comparable set of results; as the protocol was able to provide sufficient capacity and a fair share of the network resources between the contending stations. Similarly, in the case of 10 and 20 connections, there were no significant differences between the performances of the four schemes (see Figure D.1 in Appendix D.1). This implied that the Ratio based and the *CRV* schemes were able to perform well in a lightly loaded network.

**Table 7.1:** Statistical results for three different schemes for light load *CBR* traffic.

| Parameter | No. of connections | Ratio based scheme | IEEE 802.11 DCF scheme | EIED scheme |
|---|---|---|---|---|
| Average delay (msec) | 5 | 2.8 | 2.8 | 2.9 |
| | 10 | 3.3 | 3.3 | 3.3 |
| | 20 | 4.0 | 4.0 | 4.2 |
| Average jitter (msec) | 5 | 0.21 | 0.21 | 0.22 |
| | 10 | 0.53 | 0.57 | 0.55 |
| | 20 | 0.23 | 0.22 | 0.22 |
| Average throughput (Kbps) | 5 | 479.0 | 479.0 | 479.0 |
| | 10 | 470.0 | 470.0 | 470.0 |
| | 20 | 470.0 | 470.0 | 470.0 |
| Average collision rate (%) | 5 | 0.33 | 0.3 | 0.12 |
| | 10 | 0.65 | 0.86 | 0.9 |
| | 20 | 0.65 | 0.87 | 1.1 |
| Average QoS (%) | 5 | 88.1 | 88.1 | 88.1 |
| | 10 | 88.0 | 88.0 | 88.0 |
| | 20 | 88.0 | 88.0 | 88.0 |

#### 7.5.2.2 Medium Load Traffic

Here approximately 60% of the channel capacity was input to the network (i.e. 1.2 Mbps). Each connection transmitted 240 Kbps, 120 Kbps and 60 Kbps corresponding to 5, 10 and 20 connections, respectively. The results obtained for QoS parameters and the assessed QoS for all schemes were comparable when 5 connections were active. As indicated in Table 7.2, there were minor differences between the proposed schemes and the schemes for the 10 and 20 connections (see Appendix D.2 for the simulation results and discussion). This difference was considered insignificant as all QoS parameters values were within the range of the QoS requirements for video application as defined by (ITU, 1996) and ($\mathrm{ITU}_{(a)}$, 2001). Furthermore, the measured QoS values for all schemes were excellent (i.e. more than 86%).

**Table 7.2:** Statistical results for three different schemes for medium load *CBR* traffic.

| Parameter | No. of connections | Ratio based scheme | IEEE 802.11 DCF scheme | EIED scheme |
|---|---|---|---|---|
| Average delay (msec) | 5 | 2.9 | 2.8 | 2.8 |
| | 10 | 12.2 | 7.3 | 4.7 |
| | 20 | 13.0 | 6.3 | 4.5 |
| Average jitter (msec) | 5 | 0.3 | 0.2 | 0.2 |
| | 10 | 3.3 | 1.5 | 1.2 |
| | 20 | 4.1 | 2.7 | 1.0 |
| Average throughput (Kbps) | 5 | 1182.0 | 1182.0 | 1175.0 |
| | 10 | 1122.0 | 1057 | 1121.0 |
| | 20 | 1033.0 | 923.0 | 996.0 |
| Average collision rate (%) | 5 | 0.1 | 0.6 | 0.1 |
| | 10 | 2.7 | 2.5 | 2.0 |
| | 20 | 3.3 | 4.2 | 1.7 |
| Average QoS (%) | 5 | 88.0 | 88.0 | 88.0 |
| | 10 | 87.0 | 88.0 | 88.0 |
| | 20 | 86.0 | 88.0 | 88.0 |

### 7.5.2.3 Heavy Load Traffic

As in previous sections (section 7.5.2.1 and section 7.5.2.2) the Ratio based and *CRV* schemes were validated for heavy load traffic. Different network topologies sizes (5, 10 and 20 connections, i.e. small, medium and large networks) were considered for this purpose. Other scenarios were also introduced to critically investigate the behaviour of the Ratio based and *CRV* schemes. This included the case where the sources transmitted different traffic types and when the number of sources was increased over time.

The offered load delivered into the network for the heavy load traffic was approximately 80% of the channel capacity. This implied that 1.6 Mbps was transmitted by all active senders in the network. Therefore, the transmission rate of each source was equal ($1.6Mbps/n$), where $n$ represents the number of connections.

#### *7.5.2.3.1 Small Network Size (5 Connections)*

Here the simulation consisted of 40 stations as shown in Figure 4.2d (see Chapter 4); only 5 connections transmitted at heavy load to 5 destinations. Around 80% of the channel capacity (more than 1600 Kbps) was offered to the network. The transmission rate of each source was 320 Kbps *CBR* traffic.

Figure 7.7a shows the average delay for the four schemes. The Ratio based and the *CRV* schemes were able to maintain low average delay at heavily loaded traffic. It can be seen that the average delay was 57% and 50% smaller than that for the legacy IEEE 802.11 DCF and the *EIED* scheme, respectively when the Ratio based scheme was employed. Similarly, the *CRV* scheme outperformed the IEEE 802.11 DCF and *EIED* schemes by 55% and 48%, respectively. Indeed, the two proposed schemes provide a smaller delay (less than 400 msec). The *CRV* scheme showed more variations especially at the beginning and at the end of the simulation. This was due to the lack of network history as the number of active stations in the network was very small and each station transmitted large number of packets necessitating several adjustments of $CW_{success}$ and $CW_{collision}$. These fluctuations decreased once the system selected proper values of $CW_{success}$ and $CW_{collision}$. Furthermore, the fluctuations became less when the number of active stations increased as discussed later.

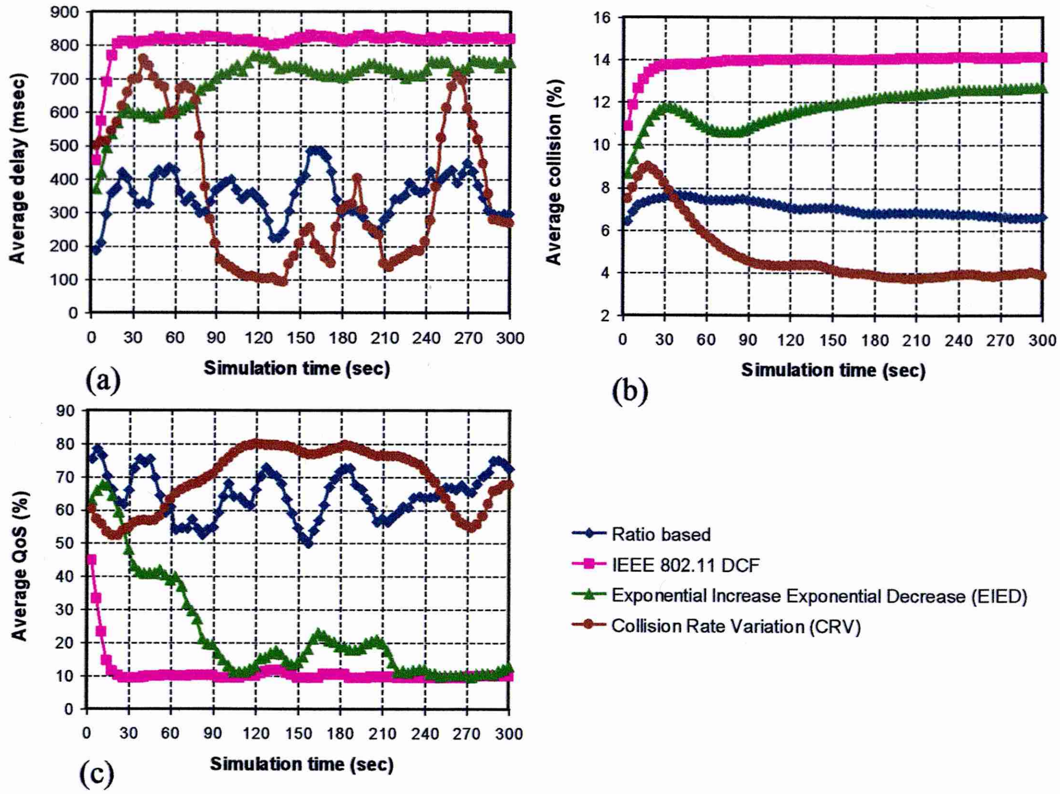



**Figure 7.7:** QoS parameters and the assessed QoS for 5 connections and the heavy *CBR* traffic, (a) average delay (msec), (b) average collision rate (%), and (c) average QoS (%).

As indicated in Table 7.3, the Ratio based and *CRV* schemes achieved small values of average jitter (less than 10 msec). The improvements reached 22% and 15% when the Ratio based scheme was used and reached 41% and 36% when the *CRV* scheme was employed over the IEEE 802.11 DCF and *EIED* schemes, respectively. Moreover, the *CRV* had a mean jitter 24% less than the Ratio based scheme. High value of jitter in the IEEE 802.11 DCF was caused by the sharp variations in the *CW* size, which resulted in a large variation in the Backoff Interval (*BI*), and consequently led to large values of delay and jitter.

As discussed earlier, the MAC efficiency and the collision rate were related to each other. This implied that the reduction in the number of collisions led to an improvement in the performance of the protocol (i.e. improve the MAC efficiency by increasing the number of successful packets transmission with respect to the total number of packets sent). In terms of collision rate, the *CRV* scheme was superior compared with other schemes. The collision rate obtained was considerably lower than the values for the IEEE 802.11 DCF and *EIED* schemes as shown in Figure 7.7b. The transmitted packets were dealt as video applications. Therefore, the QoS requirements recommended by the ($ITU_{(b)}$, 2001) for video application were considered. The QoS parameters obtained for each connection were fed to the *FIS* mechanism. The output of the *FIS* mechanism was the measured QoS for each connection as described in Figure 4.3 (see Chapter 4). Hence after, the average values of QoS and QoS parameters of the whole connections were considered (i.e., every point in the graph represented the average of 5 connections, 10 connections or 20 connections based on the network size).

The QoS obtained is shown in Figure 7.7c. The QoS achieved by the Ratio based scheme had a mean value of 65% (good QoS). Here, QoS was 83% and 65% higher than the QoS obtained when the IEEE 802.11 DCF and *EIED* schemes were used, respectively, and 6% less when the *CRV* scheme was employed. The *CRV* scheme had the best achieved QoS (69%, i.e., excellent QoS). It had a mean QoS 84%, 67% and 6% higher than the IEEE 802.11 DCF, *EIED* and the Ratio based schemes, respectively. The quantitative statistics for a small network scenario in heavy load network are summarised in Table 7.3.

**Table 7.3:** Statistical results obtained for four different schemes using the small size network (5 connections) and heavy load *CBR* traffic.

| Parameter | Statistic measure | Ratio based scheme | IEEE 802.11 DCF scheme | EIED scheme | CRV scheme |
|---|---|---|---|---|---|
| Average delay (msec) | Mean | 345.4 | 810.1 | 699.5 | 365.8 |
| | Stdev | 63.5 | 48.2 | 78.1 | 220.8 |
| Average jitter (msec) | Mean | 9.9 | 12.8 | 11.8 | 7.6 |
| | Stdev | 1.1 | 0.4 | 0.9 | 2.9 |
| Average throughput (Kbps) | Mean | 1297.6 | 1227.2 | 1252.4 | 1227.6 |
| | Stdev | 47.1 | 31.8 | 74.9 | 81.2 |
| Average packet loss (%) | Mean | 7.8 | 21.1 | 16.4 | 9.0 |
| | Stdev | 1.8 | 1.7 | 2.5 | 6.2 |
| Average MAC efficiency (%) | Mean | 92.9 | 85.7 | 86.8 | 95.4 |
| | Stdev | 0.6 | 0.4 | 1.5 | 2.1 |
| Average collision rate (%) | Mean | 7.0 | 13.9 | 11.9 | 4.9 |
| | Stdev | 0.3 | 0.4 | 0.8 | 1.5 |
| Average QoS (%) | Mean | 64.7 | 10.8 | 22.2 | 69.3 |
| | Stdev | 6.7 | 4.7 | 15.4 | 9.4 |
| | QoS level | Excellent | Poor | Poor | Excellent |

#### *7.5.2.3.2 Medium Network Size (10 Connections)*

In this scenario, the topology shown in Figure 4.2d (see Chapter 4) was used and the offered load was 80% of the channel capacity which was equally distributed among the 10 connections. Each source transmitted 160 Kbps to its corresponding destination.

The values of average delay and jitter for all schemes were increased when the number of active stations was increased from 5 to 10 connections with the same amount of traffic. For example, the values of average delay were increased by 44%, 55%, 54% and 47% in the Ratio based, IEEE 802.11 DCF, *EIED*, and *CRV* schemes, respectively. This implied that the contention between stations was a significant factor. In Figure 7.8a, the Ratio based and the *CRV* schemes displayed a smaller mean delay. A reduction of 65% and 59% was observed in the average delay when the Ratio based scheme was employed compared to the IEEE 802.11 DCF and *EIED* schemes, respectively. Similarly, the *CRV* scheme achieved a small average delay which was 9% higher than that obtained for the Ratio based scheme. Note that in case of 5 connections, the Ratio based and the *CRV* schemes experienced high fluctuations. In the case of 10 connections, these fluctuations became less, because of the proper selection of the *CW* size when the network became busier.

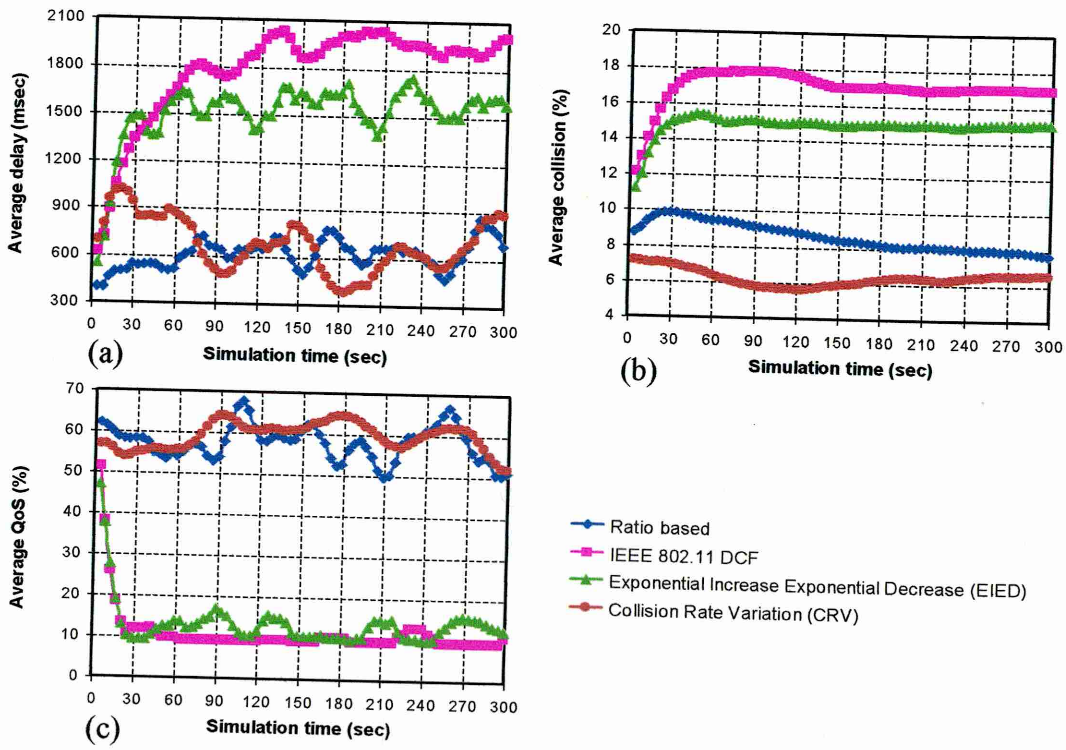



**Figure 7.8:** QoS parameters and the assessed QoS for 10 connections using heavy *CBR* load, (a) average delay (msec), (b) average collision rate (%), and (c) average QoS (%).

According to Figure 7.8b, high values of collision rate were observed especially during the first 30 seconds of the simulation. This was due to the impact of routing information exchange during this period.

A good level of QoS was obtained (i.e., 59% mean value) when the *CRV* scheme was used which was 63%, 58% and 2 % higher the QoS obtained when the standard IEEE 802.11 DCF, *EIED* and Ratio based schemes were used, respectively as shown in Figure 7.8c. However, the Ratio based scheme also achieved a good level of QoS with 58% mean value. The quantitative results of the assessed QoS and other QoS parameters for medium size network are summarised in Table 7.4.

**Table 7.4:** Statistical results for four different schemes using the medium size network (10 connections) and heavy load *CBR* traffic.

| Parameter | Ratio based scheme | IEEE 802.11 DCF scheme | EIED scheme | CRV scheme |
|---|---|---|---|---|
| **Average delay (msec)** | 621.6 | 1788.0 | 1520.3 | 681.9 |
| **Average jitter (msec)** | 21.3 | 40.1 | 36.3 | 16.5 |
| **Average throughput (Kbps)** | 1240.7 | 1025.5 | 1089.1 | 1160.3 |
| **Average packet loss (%)** | 7.9 | 27.1 | 19.5 | 9.6 |
| **Average MAC efficiency (%)** | 91.4 | 81.1 | 83.7 | 92.7 |
| **Average collision (%)** | 8.6 | 17.0 | 14.9 | 6.3 |
| **Average QoS (%)** | 57.8 | 11.5 | 13.7 | 59.4 |

#### *7.5.2.3.3 Large Network Size (20 Connections)*

The performance of the four schemes was affected when the network size was increased from a small network size (i.e., 5 connections) to a medium network size (i.e., 10 connections). This could be significant as the network size became larger. In this section, the performance of the four schemes was evaluated when the number of active stations was increased to 20 connections. The volume of *CBR* traffic represented 80% of channel capacity and each source transmitted 80 Kbps.

Figures 7.9a, 7.9b and 7.9c show that the performance of the four schemes was degraded in the network with 20 connections. However, the Ratio based and *CRV* schemes still performed better than the other two schemes. The mean delay was 57% less than the obtained delay when the IEEE 802.11 DCF and *EIED* schemes were used. The mean jitter value for the Ratio based schemes were 47% less than the values obtained when the IEEE 802.11 DCF and *EIED* schemes were used. Similarly, the average throughput improved by 19% and 15% compared to the IEEE 802.11 DCF and *EIED* schemes, respectively. This was due to the reduction in the number of packets dropped. This was less than 11% when the Ratio based and *CRV* schemes were used. On the other hand, more than 24% of packets were lost when the IEEE 802.11 DCF and *EIED* schemes were used.

As indicated in Table 7.5, the IEEE 802.11 DCF had a mean MAC efficiency of 78% which was 11% less than the one obtained for the Ratio based scheme. The other schemes also showed improvements in their MAC efficiency values, implying that each scheme was able to adjust its backoff timer and particularly the *CW* size, until the behaviour of each scheme became stable. However, the Ratio based and the *CRV* schemes achieved MAC efficiency 10% higher than the other two schemes. The increased efficiency of the Ratio based and *CRV* schemes was because they considered part of the network history to tune their backoff timer by getting a suitable *CW* size after successful and unsuccessful transmission; whereas the other two schemes only considered the current network state.

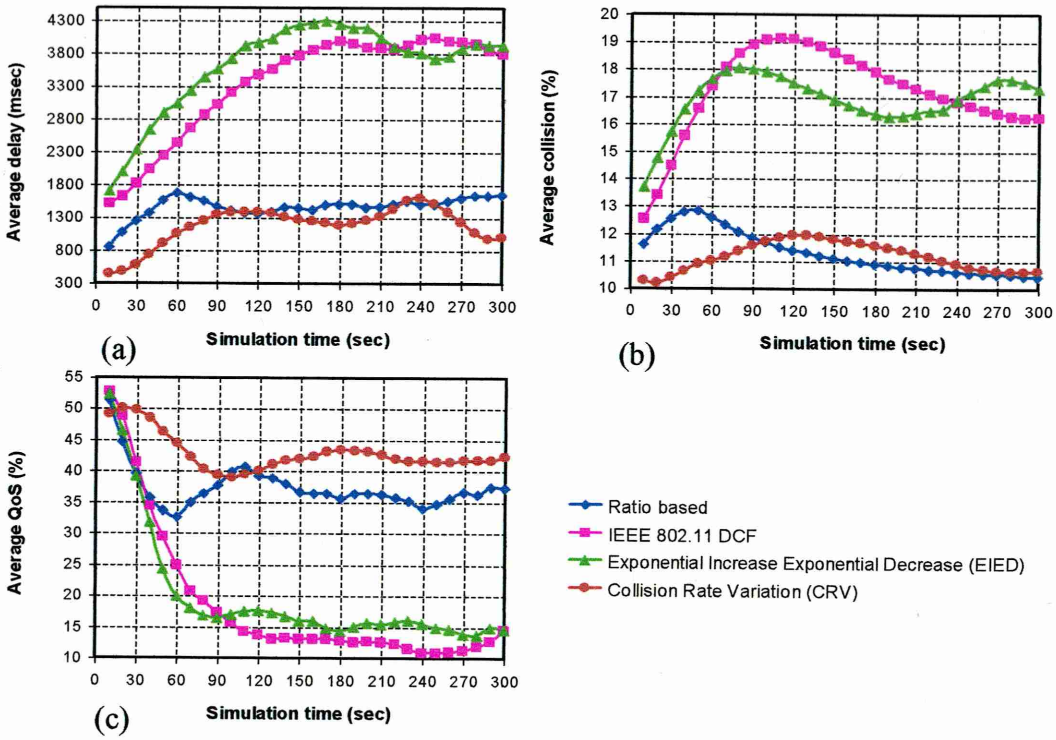



**Figure 7.9:** QoS parameters and the assessed QoS for 20 connections at heavy *CBR* load case, (a) average delay (msec), (b) average collision rate (%), and (c) average QoS (%).

**Table 7.5:** Statistical results using the large network (20 connections) and heavy load *CBR* traffic

| Parameter | Statistic measure | Ratio based scheme | IEEE 802.11 DCF scheme | EIED scheme | CRV scheme |
|---|---|---|---|---|---|
| Average delay (msec) | Mean | 1465.6 | 3348.7 | 3635.0 | 1180.4 |
| | Stdev | 170.6 | 803.8 | 685.9 | 298.8 |
| Average jitter (msec) | Mean | 48.2 | 95.2 | 92.7 | 33.9 |
| | Stdev | 3.4 | 22.5 | 16.7 | 5.2 |
| Average throughput (Kbps) | Mean | 1147.9 | 928.6 | 972.0 | 974.4 |
| | Stdev | 133.6 | 97.9 | 83.9 | 173.5 |
| Average packet loss (%) | Mean | 11.0 | 24.7 | 23.9 | 8.0 |
| | Stdev | 2.2 | 8.8 | 6.3 | 2.7 |
| Average MAC efficiency (%) | Mean | 88.7 | 78.7 | 80.4 | 88.1 |
| | Stdev | 1.8 | 3.4 | 2.1 | 2.0 |
| Average collision rate (%) | Mean | 11.3 | 17.2 | 16.9 | 11.2 |
| | Stdev | 0.8 | 1.6 | 1.0 | 0.5 |
| Average QoS (%) | Mean | 37.4 | 18.6 | 19.8 | 42.94 |
| | Stdev | 3.5 | 11.4 | 9.7 | 2.94 |
| | QoS level | Good | Poor | Poor | Good |

The collision rate obtained is shown in Figure 7.9b; with the collision rates for the four schemes being similar when the traffic load was light (see Appendix D.1). However, at heavy load traffic, the Ratio based and the *CRV* schemes were able to maintain a lower collision rate than the IEEE 802.11 DCF and *EIED* schemes. This behaviour can be explained by the fact that the Ratio based and *CRV* schemes used an adaptive mechanism to adjust the *CW* size based on the collision rate history which the stations experienced. As a result, a considerable reduction in the collision rate values was obtained which improved the network performance. This can be observed in the QoS curve shown in Figure 7.9c. The Ratio based and the *CRV* schemes achieved good levels of QoS with mean values of 37% and 43%, respectively; whereas, the IEEE 802.11 DCF and *EIED* schemes had poor levels of QoS with mean values of 19% and 20%, respectively. The statistical results for this scenario are summarised in Table 7.5.

It is worth noting that, the trend of the QoS curves for all schemes was smoother when the network size increased. This was due to the reduction in the number of packets sent by each station which required smaller adjustments of the *CW* size for each station.

The network performance was affected when the number of transmitting stations was increased. This implied that the network size and the offered load played a major role in the performance of ad-hoc networks. Figures 7.10a and 7.10b depict the collision rate and the evaluated QoS as a function of the network size for the four schemes.

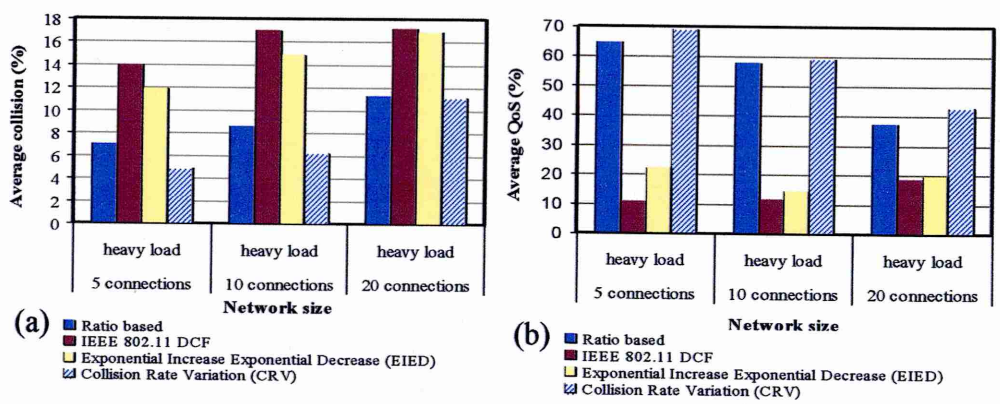



**Figure 7.10:** Collision rate and QoS as a function of network size for the heavy load *CBR* traffic, (a) average collision rate and (b) average QoS.

In Figure 7.10, the performance of a small network (i.e. 5 connections) was better than the medium and large ones. As expected, delay, jitter and packet loss for all schemes increased with an increase in the network size because of large *CW* values (high competition between contending stations). Additionally, the network with 20 connections caused large number of collisions due to the high competition which also led to less MAC efficiency. However, the proposed schemes achieved better performance than the two other schemes, whatever the number of connections was. For instance, at heavy load case for 5, 10 and 20 connections, the IEEE 802.11 DCF and

*EIED* achieved poor QoS (i.e. less than 33%), while the Ratio based and the *CRV* schemes, switching from an excellent level (i.e. more than 66%) of QoS on a small network to a good level (i.e. between 34 - 65%) in case of medium and large networks.

In conclusion the Ratio based and *CRV* schemes were capable of adaptively adjusting the *CW* values after a successful and unsuccessful transmission based on the history of each individual station. Moreover, both schemes were able to achieve an efficient trade off between collision decrease and idle time slot increase to achieve QoS for the transmitted application.

In the previous sections, the performance of Ratio based and the *CRV* schemes was evaluated and compared with the IEEE 802.11 DCF and *EIED* schemes when the number of contending stations was fixed. The following section presents a performance evaluation when the number of active stations varied over time. Moreover, it shows how the Ratio based scheme behaves when the number of active stations and traffic load changed abruptly over time.

#### *7.5.2.3.4 Ratio Based and Collision Rate Variation in a Realistic Scenario*

The performance of the Ratio based and *CRV* schemes was evaluated against an increasing number of active stations over time. This was carried out in order to examine the performance of these schemes when the network was experienced highly changing configurations. Here, the network topology shown in Figure 4.2d (see Chapter 4) was used with twenty stations transmitting *CBR* traffic, using a 512 bytes packet size, to twenty different destinations using the basic access mechanism. The simulation time was 400 seconds. Every 5 seconds a new *CBR* source with 80Kbps generation rate started its transmission. At the 100$^{th}$ second of the simulation, 20 sources were active in the network (i.e. contending to access the channel) and sending video packets to 20 destinations. These 20 *CBR* sources remained active to the 300$^{th}$ second in order to sustain heavy load throughout the 200 seconds (i.e. from 100 to 300 seconds). At the 300$^{th}$ second, the number of active stations was reduced by one every 5 seconds until all sources stopped their transmission at the 400$^{th}$ second.

According to Figure 7.11a, the value of average delay increased with the the simulation time (due to the increase of the number of active stations). However, the Ratio based and *CRV* schemes maintained 50% of average delay when the IEEE 802.11 DCF and

*EIED* schemes were employed. The maximum values of average delay were observed between the 200$^{th}$ and 300$^{th}$ second because of high competition between the contending stations. In the IEEE 802.11 DCF and *EIED* schemes each station selected a large *CW* size in order to avoid collisions at the cost of wasting several idle time slots which in turn led to high values of delay. The sharp transition from a large *CW* size to $CW_{min}$ in case of IEEE 802.11 DCF and to half of the current *CW* size of the *EIED* scheme after successful transmission increased the amount of jitter. Thereafter, the average delay started to decrease since the number of sources decreased by one every 5 seconds.

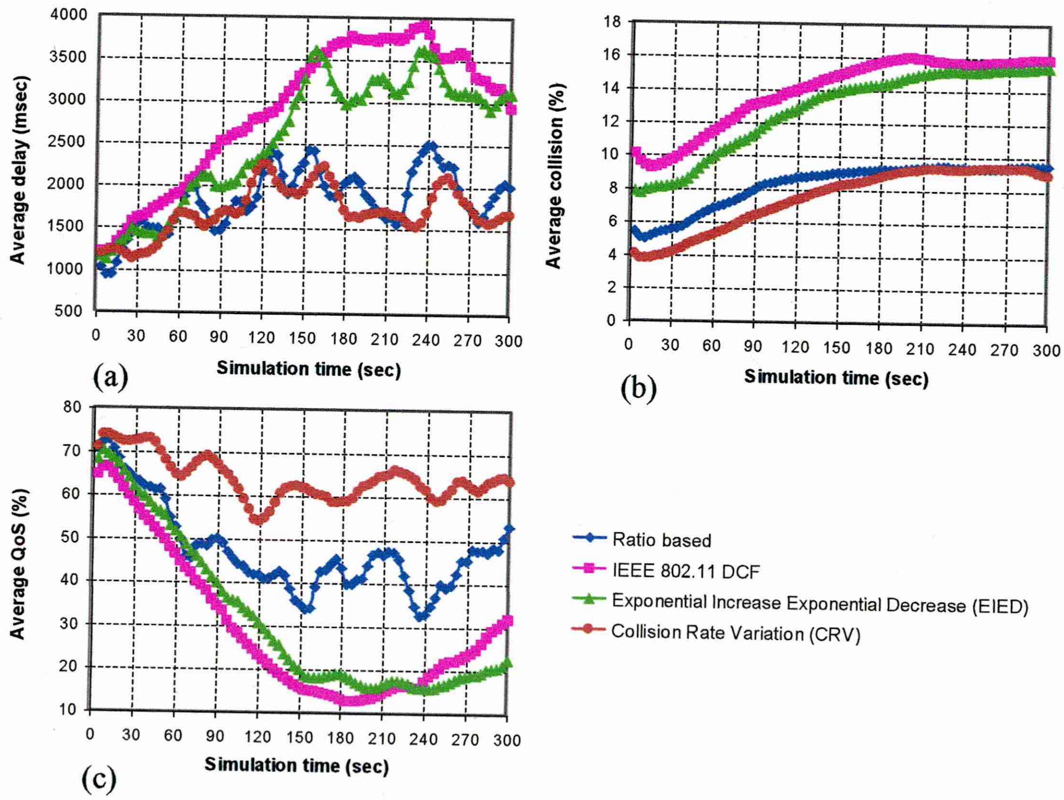



**Figure 7.11:** QoS parameters and the assessed QoS when the number of connections was changed over time, (a) average delay (msec), (b) average collision rate (%), and (c) average QoS (%).

The Ratio based and *CRV* schemes maintained approximately similar values of delay as the other two schemes up to 100$^{th}$ second, since the network was still lightly loaded and the number of contending stations was less than 20 connections. After the 100$^{th}$ second, the network became busier and the load heavier, therefore, the Ratio based and *CRV* schemes performed better and maintained lower values of average delay and average jitter. This was due to the capability of the Ratio based and *CRV* of adaptively selecting the *CW* size after successful and unsuccessful transmission in a way that achieved a tradeoff between collisions decrease and idle time slots increase.

Similarly, the average throughput was 16% and 11% higher than the IEEE 802.11 DCF and *EIED* schemes, respectively when the Ratio based scheme was used and 11.6% and

6% higher when the *CRV* scheme was employed. The reduction in the average throughput and the increase in the number of packet loss when the IEEE 802.11 DCF and *EIED* schemes were used were due to the following; in the IEEE 802.11 DCF scheme, as the station resets its *CW* to $CW_{min}$ after successful transmission or decreases it to the half of its current value in *EIED* scheme, the station forgets about the collision history. In this case when all stations kept transmitting with the same data rate; it is likely that the new transmission noticed contention and collisions as before. This in turn increased the collision rate especially during a high contention period as shown in Figures 7.11b. This was mitigated by keeping some history of the observed successful and collisions packets. In this case instead of resetting the *CW* to $CW_{min}$ after successful transmission or doubling it after collisions, the *CW* size was changed adaptively based on the history of collision rate.

The behaviour of the Ratio based and *CRV* schemes was apparent on both the achieved MAC efficiency and the collision rate parameters. For instance, the Ratio based scheme had a 90% average MAC efficiency and 8.3% average collision rate; whereas 81% average MAC efficiency and 14% average collision rate were observed for the IEEE 802.11 DCF scheme. The *CRV* scheme also achieved higher performance than the IEEE 802.11 DCF and *EIED* schemes. It had 64% mean QoS, and this was 53% higher than the ones obtained using the IEEE 802.11 DCF and *EIED* schemes as shown in Figure 7.11c. In general, for the selected scenarios, the *CRV* scheme gave better performance than the Ratio based scheme as indicated in Table 7.6.

**Table 7.6:** Statistical results for four different schemes when the number of sources was increased.

| Parameter | Statistic measure | Ratio based scheme | IEEE 802.11 DCF scheme | EIED scheme | CRV scheme |
|---|---|---|---|---|---|
| Average delay (msec) | Mean | 1843.9 | 2904.2 | 2594.6 | 1698.8 |
| | Stdev | 361.4 | 828.9 | 759.3 | 295 |
| Average jitter (msec) | Mean | 37.2 | 72.1 | 61.7 | 26.7 |
| | Stdev | 9.8 | 25.3 | 22.8 | 7.3 |
| Average throughput (Kbps) | Mean | 998.6 | 834.4 | 888 | 944.4 |
| | Stdev | 204.8 | 157.5 | 183.8 | 204.6 |
| Average packet loss (%) | Mean | 7.7 | 19.7 | 12.9 | 7 |
| | Stdev | 2.9 | 7.8 | 5.9 | 2.4 |
| Average efficiency (%) | Mean | 89.9 | 80.6 | 82.3 | 89.7 |
| | Stdev | 1.7 | 3.9 | 3.6 | 2.7 |
| Average collision (%) | Mean | 8.3 | 14 | 12.9 | 7.5 |
| | Stdev | 1.4 | 2.2 | 2.6 | 1.9 |
| Average QoS (%) | Mean | 47.4 | 29.5 | 31.8 | 64.1 |
| | Stdev | 9.8 | 16.4 | 17.8 | 4.9 |
| | QoS level | Good | Poor | Poor | Good |

Another special case was deduced from this scenario, which was related to examining how the Ratio based scheme behaves when the number of stations varies sharply. The results showed that the Ratio based scheme performed better when the network experienced highly changing configurations with different traffic volume over time. The Ratio based scheme showed fewer fluctuations and faster response to the abrupt change in the network size and to the variation in the offered load. The results obtained also indicate that the Ratio based scheme was capable of providing better channel utilisation usage and shared the channel capacity more fairly among the contending stations. More information about this scenario and the main findings are provided in Appendix D.3.

To this point, different simulations have been carried out to study the performance of Ratio based and *CRV* schemes when all stations transmitted *CBR* traffic. In the following section, the performance of the proposed schemes will be studied when *VBR* traffic is considered.

### 7.5.3 Performance Evaluation of CW Adjustments Using VBR Traffic

The same scenarios discussed in sections (7.5.2.3.1, 7.5.2.3.2, and 7.5.2.3.3) were examined when the wireless sources transmitted heavy *VBR* traffic. The topology shown in Figure 4.2d (see Chapter 4) was also employed. Moreover, the same QoS metrics were considered for the performance evaluation of the four schemes.

As shown in Figure 7.12, the QoS values were presented for three different network sizes (i.e. small, medium and large). Generally, the trend of the QoS curves showed more fluctuations than the *CBR* traffic. This was due to the fact that *CBR* traffic keeps a continuous overall offered load on the network during the simulation time which kept the network under a heavy load situation during the period of simulation. Conversely, the *VBR* traffic has a variable frame length and a variable inter-frame interval. In this case the maximum offered load may not occur at the same time. This gives the network the ability to deal with the brief offered load bursts exhibited by different traffic sources at different time scales. When this occurs, the number of packets dropped will increase and this leads to high fluctuations especially in the packet loss parameter on all network sizes. These offered load bursts (i.e. maximum offered load cases) and the variations in the frame length also affected the other QoS parameters and the evaluated QoS for all network sizes. The quantitative results of these parameters are given in Table 7.7.

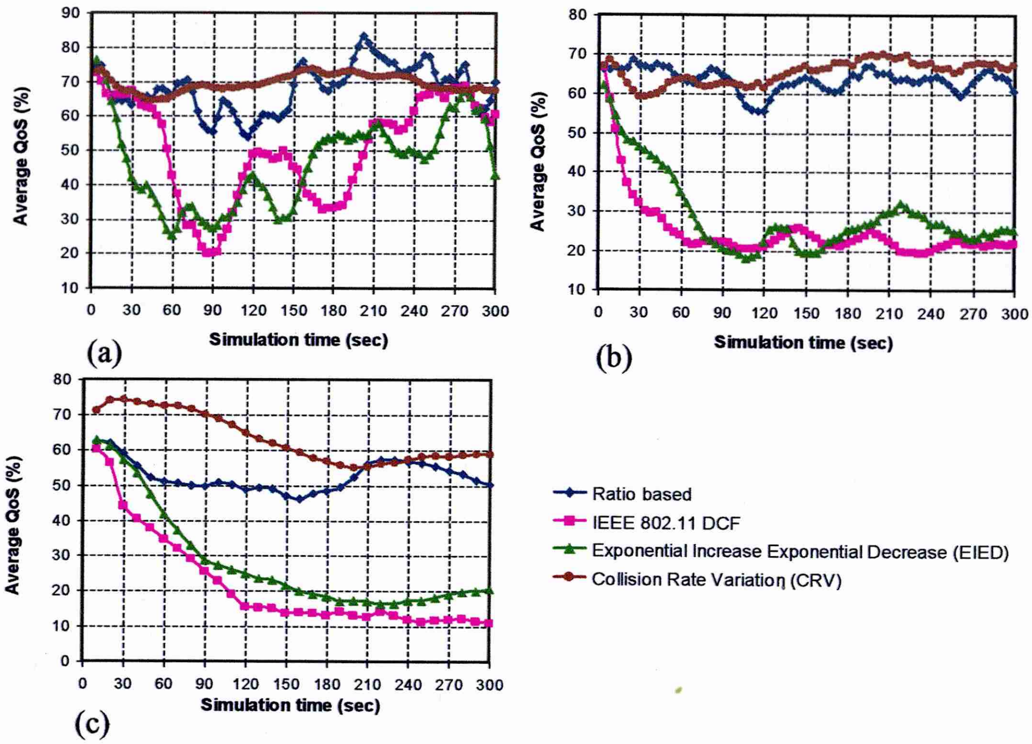



**Figure 7.12:** Assessed QoS for the heavy *VBR* traffic, (a) 5, (b) 10, and (c) 20 connections.

**Table 7.7:** Statistical results obtained for four different schemes at heavy load *VBR* traffic.

| Parameters | No. of connections | Ratio based scheme | IEEE 802.11 DCF scheme | EIED scheme | CRV scheme |
|---|---|---|---|---|---|
| Average delay (msec) | 5 | 241.3 | 430.0 | 469.0 | 434.5 |
| | 10 | 381.9 | 1499.3 | 1231.1 | 538.5 |
| | 20 | 526.5 | 2729.2 | 1958.5 | 908.1 |
| Average jitter (msec) | 5 | 11.3 | 14.6 | 14.9 | 10.8 |
| | 10 | 21.3 | 49.8 | 43.4 | 21.4 |
| | 20 | 35.5 | 97.7 | 88.7 | 34.5 |
| Average throughput (Kbps) | 5 | 1338.6 | 1307.4 | 1326.1 | 1262.8 |
| | 10 | 1304.1 | 1124.5 | 1196.7 | 1291.3 |
| | 20 | 1088.1 | 812.7 | 966.4 | 857 |
| Average packet loss (%) | 5 | 2.5 | 3.4 | 4.0 | 6.5 |
| | 10 | 2.0 | 13.2 | 7.5 | 3.4 |
| | 20 | 1.0 | 10.1 | 6.0 | 2.7 |
| Average collision rate (%) | 5 | 7.9 | 14.9 | 14.1 | 5.7 |
| | 10 | 9.5 | 21.2 | 16.4 | 9.8 |
| | 20 | 9.8 | 15.5 | 17.9 | 10.1 |
| Average QoS (%) | 5 | 68.1 | 50.4 | 46.3 | 69.5 |
| | 10 | 63.6 | 25.1 | 29.2 | 65.4 |
| | 20 | 52.7 | 21.8 | 28.2 | 63.3 |

As shown in Figures 7.12a, 7.12b and 7.12c, the average QoS values for the Ratio based and *CRV* schemes were excellent (i.e. 68%) for the small network size, degraded to good levels (i.e. 53% - 65%) for the medium and large networks. The standard IEEE 802.11 DCF and *EIED* schemes achieved good QoS levels (i.e. 46% - 50%) in the small network and then degraded to poor QoS levels (i.e. 15% - 28%) for the medium and large networks.

Figures 7.13a and 7.13b depict the average collision rate and average QoS as a function of network size for different schemes. In all network sizes, the ratio based and *CRV* schemes resulted in higher average QoS and lower collision rate compared to IEEE 802.11 DCF and *EIED* schemes.

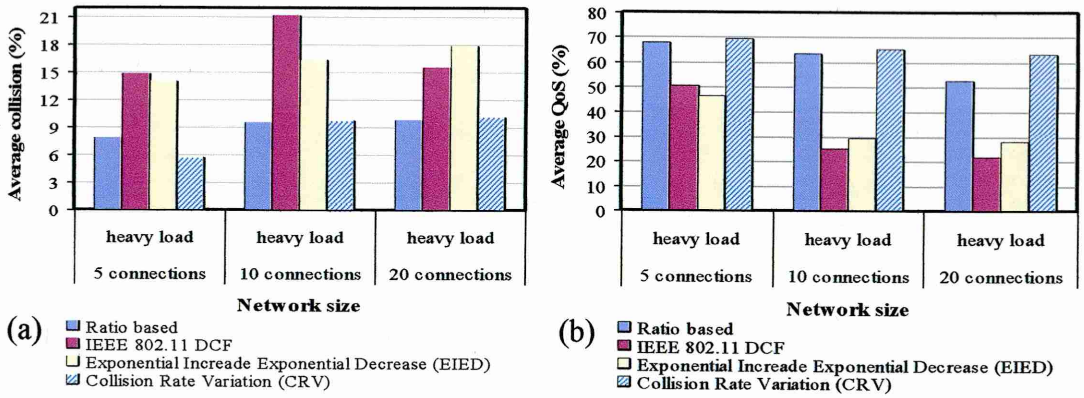



**Figure 7.13:** Collision rate and QoS parameters as a function of network size for the heavy load *VBR* traffic, (a) average collision rate, and (b) average QoS.

It can also be observed that increasing the network size had a negative impact on other QoS parameters. For instance, average delay increased by 54% and average throughput degraded by 23% when the number of *VBR* connections was increased from 5 to 20 connections in the case of the Ratio based scheme (see Table 7.7). This can be explained as follows: the *CW* size in the basic access mode of the IEEE 802.11 MAC protocol was network size and offered load dependent, thus it was significantly affected by the network size and the offered load. A small network size required smaller competition than a large network size, therefore the *CW* size in a small network should be very small in order to reduce the effect of wasted time slots and therefore provide better performance. In a large network, a large *CW* size was required in order to reduce the number of collisions and to improve the MAC efficiency parameter particularly for the case of a heavy load. Subsequently, using an adaptive technique to deal with these situations resulted in better performance for all network sizes compared with the IEEE 802.11 DCF and *EIED* schemes.

In case of *VBR* traffic, the Ratio based and *CRV* schemes showed high responsiveness to the variation in the traffic load during the simulation time. Additionally, they achieved better QoS values than the ones obtained when *CBR* traffic was considered as indicated in Table 7.8. The average QoS was 5% higher than the QoS obtained when the *CBR* traffic was transmitted in a small network, 9% higher in a medium network, and 29% higher in a large network when the Ratio based scheme was used. Significant improvements were also observed when the *CRV* scheme was employed. Furthermore, the IEEE 802.11 DCF and *EIED* schemes also showed noticeable improvements particularly for a small network.

**Table 7.8:** Average QoS in (%) for the heavy *CBR* and *VBR* traffic and using the four schemes.

| No. of connections | Traffic type | Statistic measure | Ratio based scheme | IEEE 802.11 DCF scheme | EIED scheme | CRV scheme |
|---|---|---|---|---|---|---|
| 5 | *CBR* | Mean | 64.7 | 10.8 | 22.2 | 69.2 |
| | | Stdev | 6.7 | 4.7 | 15.4 | 9.2 |
| | *VBR* | Mean | 68.1 | 50.4 | 46.3 | 69.5 |
| | | Stdev | 6.9 | 14.9 | 12.7 | 2.5 |
| 10 | *CBR* | Mean | 57.8 | 11.5 | 13.7 | 59.4 |
| | | Stdev | 4.2 | 6.4 | 5.8 | 3.4 |
| | *VBR* | Mean | 63.6 | 25.1 | 29.2 | 65.4 |
| | | Stdev | 2.9 | 8.4 | 10.1 | 3.0 |
| 20 | *CBR* | Mean | 37.4 | 18.6 | 19.8 | 42.9 |
| | | Stdev | 3.5 | 11.4 | 9.7 | 3.0 |
| | *VBR* | Mean | 52.7 | 21.8 | 28.2 | 63.3 |
| | | Stdev | 4.3 | 13.9 | 14.4 | 6.9 |

It can be concluded that the performance of the Ratio based and *CRV* schemes were better than the standard IEEE 802.11 DCF and *EIED* schemes when *CBR* and *VBR* traffic were transmitted regardless of the network size. The Ratio based and *CRV* schemes also showed a good response to the variations in the traffic rate (*VBR* traffic) especially when the maximum bit rate occurred at the same time.

Previous sections described the performance of Ratio based and the *CRV* schemes were evaluated and compared with the IEEE 802.11 DCF and *EIED* schemes when one type of traffic *CBR* or *VBR* traffic was transmitted over a single-hop network. For a further insight into the performance of the Ratio based and the *CRV* schemes for mixed traffic and for multi-hop networks Appendices D.4 and D.5 provide the required information.

### 7.5.4 Performance Evaluation of the Adaptive DIFS Scheme

To demonstrate the properties of *ADIFS* scheme, the topology shown in Figure 4.2d (see Chapter 4) and the scenarios discussed in sections (7.5.2.3.1, 7.5.2.3.2, 7.5.2.3.3, 7.5.2.3.4, and 7.5.3) were used.

Figures 7.14a, 7.14b and 7.14c show the average delay, average collision rate, and the evaluated QoS, respectively when 5 *CBR* connections were active. In Figure 7.14a the average delay remained fixed around 600 msec during the simulation with small fluctuations around its mean value. The mean delay value was 593.7 msec which resulted in a poor QoS level (i.e. average QoS = 25.3%).

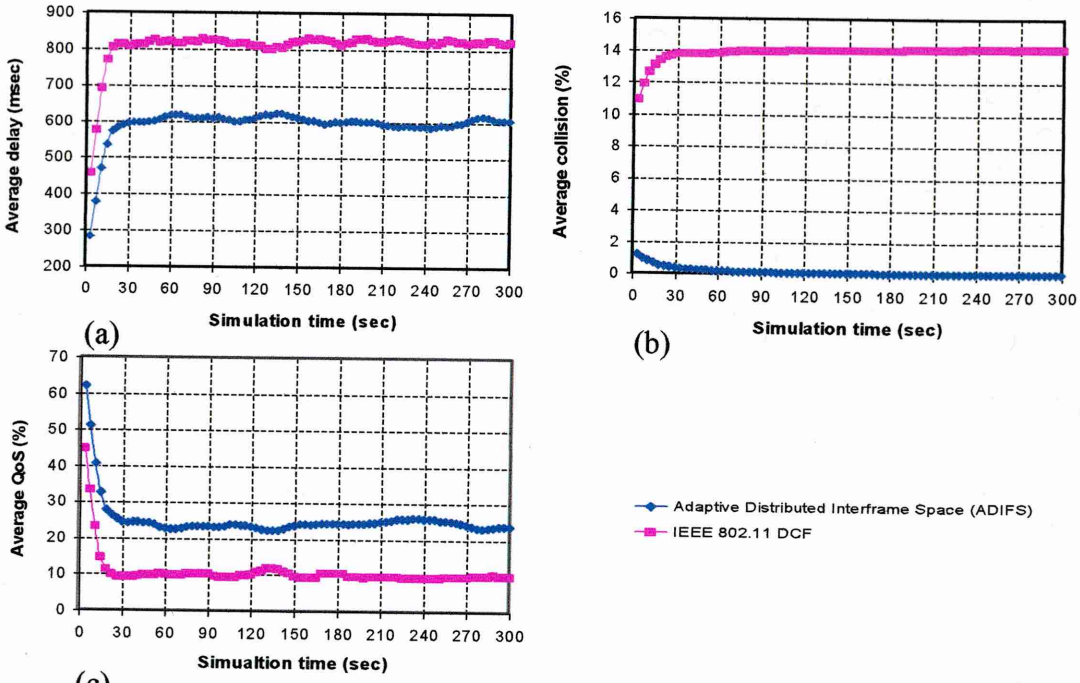



**Figure 7.14:** *ADIFS* vs. IEEE 802.11 DCF for 5 *CBR* connections using the heavy load case, (a) average delay (msec), (b) average collision rate (%), and (c) average QoS (%).

The *ADIFS* scheme performed well when the number of *CBR* connections was increased to 10. Compared to the IEEE 802.11 DCF scheme, a reduction of more than 50% in the average delay and more than 90% of collision rate accompanied with more than 70% improvement in the average QoS were observed as shown in Figures 7.15.

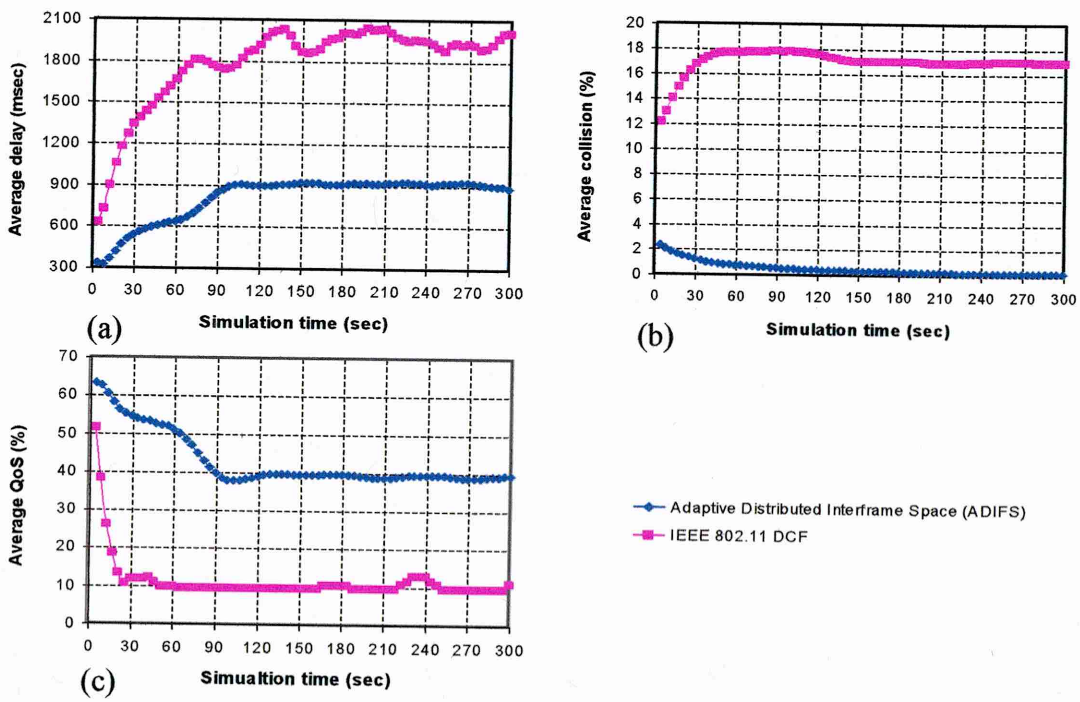



**Figure 7.15:** *ADIFS* vs. IEEE 802.11 DCF for 10 *CBR* connections using the heavy load, (a) average delay (msec), (b) average collision rate (%), and (c) average QoS (%).

When the number of connections was increased to 20, the *ADIFS* scheme showed a significant improvement compared with the IEEE 802.11 DCF scheme. For example, the average delay was 44% less and the average collision rate was 92% less than the IEEE 802.11 DCF scheme as shown in Figure 7.16a and Figure 7.16b, respectively. This reduction in average delay and collision rate resulted in a good QoS for the *ADIFS* scheme with a mean value equal to 44% as shown in Figure 7.16c.

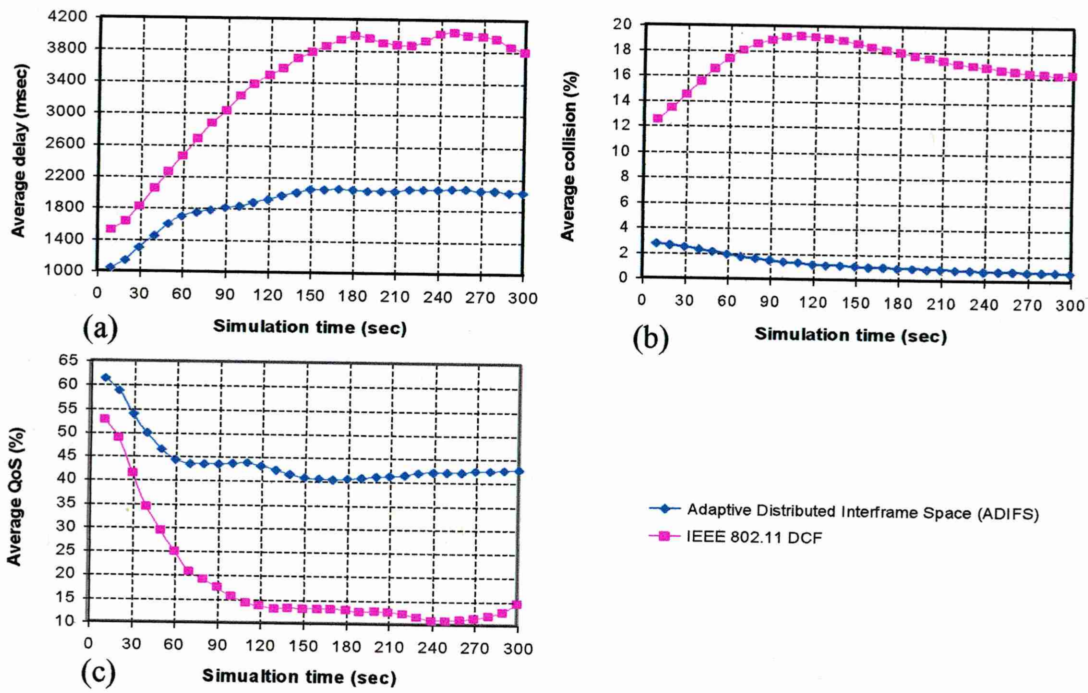



**Figure 7.16:** *ADIFS* vs. IEEE 802.11 DCF for 20 *CBR* connections using heavy load, (a) average delay (msec), (b) average collision rate (%), and (c) average QoS (%).

So far, the *CBR* traffic was considered as input traffic to demonstrate the performance of the *ADIFS* scheme and compare it with the IEEE 802.11 DCF scheme. The results indicated that the *ADIFS* scheme outperformed the IEEE 802.11 DCF in small, medium and large networks with fewer fluctuations. To study the performance of the *ADIFS* scheme for *VBR* traffic, the same scenarios discussed in an earlier part of this section were carried out.

Figures 7.17a, 7.17b and 7.17c show the average QoS for 5, 10 and 20 *VBR* connections, respectively. An excellent QoS level with mean value equal to 82% was obtained when 5 *VBR* connections accessed the medium. This value was 39% higher than the achieved QoS when the IEEE 802.11 DCF was employed. For 10 and 20 *VBR* connections, the *ADIFS* scheme resulted in a significant improvement in average QoS (i.e. more than 50%) compared to the standard IEEE 802.11 DCF as shown in Figures 7.17b and 7.17c, respectively.

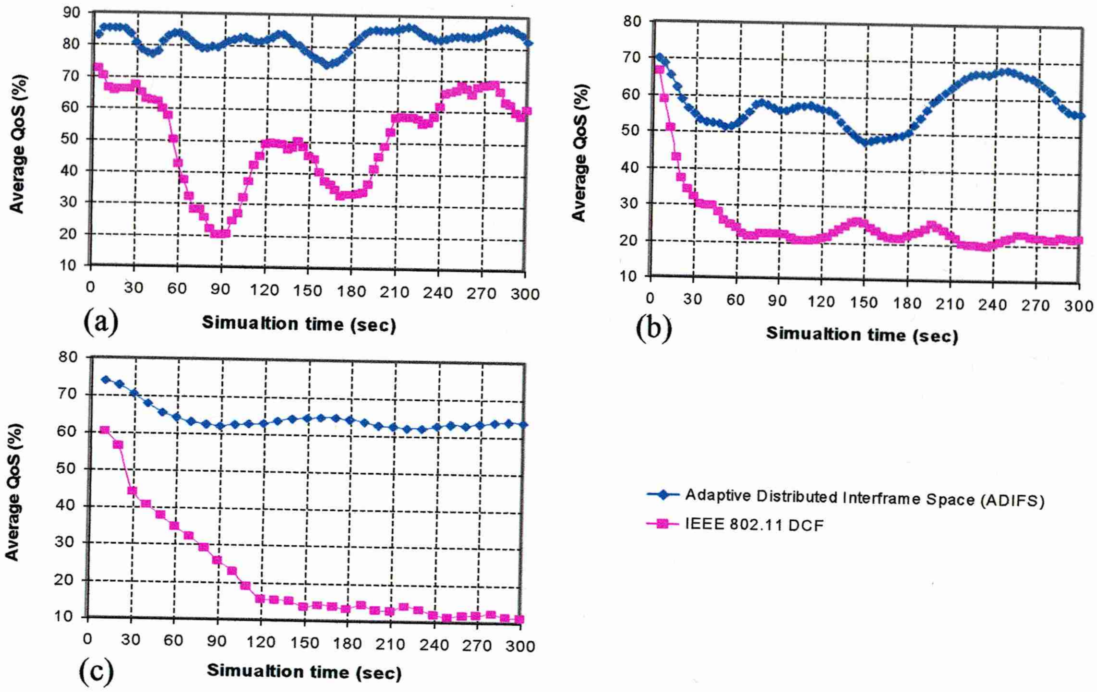


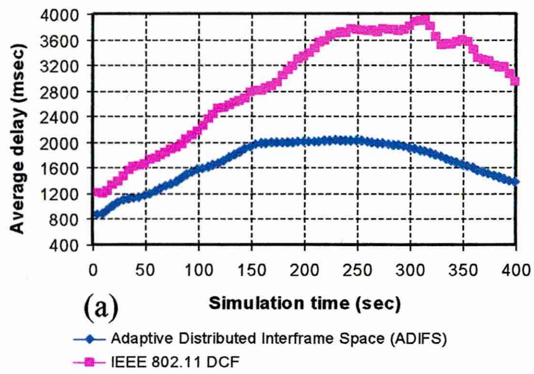

**Figure 7.17:** Average QoS of *ADIFS* vs. IEEE 802.11 DCF for 5, 10 and 20 *VBR* connections and using the heavy load, (a) 5 *VBR* connections, (b) 10 *VBR* connections, and (c) 20 *VBR* connections.

Using the scenarios discussed in section 7.5.2.3.4, the performance of the *ADIFS* scheme was investigated when the number of active stations varied over time. The results obtained were compared with the standard IEEE 802.11 DCF scheme.

As shown in Figure 7.18a, the average delay of the *ADIFS* and the IEEE 802.11 DCF schemes increased over time, since every 5 seconds a new station attempted to access the medium. The maximum average delay was observed during the 150$^{th}$ to 300$^{th}$ second time period of the simulation, as all 20 stations were active. After the 300$^{th}$ second, the average delay decreased, because the number of contending stations decreased by one station every 5 seconds until the end of the simulation at the 400$^{th}$ second. The adaptive adjustment of the *DIFS* value (*ADIFS*) resulted in a smaller average delay and a smaller variation from its mean value, while the standard IEEE 802.11 DCF scheme produced a higher average delay, by 42%. The reason was: in the IEEE 802.11 DCF, the *DIFS* value was kept fixed during the whole simulation period; whereas, the length of the *DIFS* in *ADIFS* scheme was dynamically and locally adjusted in each individual station. Therefore, a variable length *DIFS* resulted in less time slots being wasted by reducing the amount of waiting time and reducing the probability of collisions. This impact can be seen on the QoS curve of the *ADIFS* scheme as shown in Figure 7.18b. During the period 0 - 150 seconds of simulation, average QoS for *ADIFS* scheme deteriorated from an excellent level to a good level as a result of increasing the number of active stations. During 150 - 300 second of simulation, *ADIFS* sustained constant good QoS levels with a mean value equal to 53.3%. Although, the IEEE 802.11 DCF scheme showed the same QoS trend as *ADIFS* scheme, its average QoS degraded from an excellent to a poor QoS level. A performance comparison of the proposed schemes and the IEEE 802.11 DCF and *EIED* schemes is given in Table 7.9.

**Table 7.9:** QoS parameters and the assessed QoS values for 5 different schemes.

| No. of connections /traffic type | Parameter | Schemes | | | | |
|---|---|---|---|---|---|---|
| | | ADIFS | IEEE 802.11 DCF | EIED | CRV | Ratio based |
| 5 / CBR | Average delay (msec) | 593.7 | 809 | 691.1 | 362.3 | 351.1 |
| | Average jitter (msec) | 7.6 | 12.7 | 11.6 | 7.5 | 10.1 |
| | Average throughput (Kbps) | 1384.4 | 1224.8 | 1248.1 | 1229.9 | 1293.9 |
| | Average loss (%) | 10.3 | 21.1 | 16.1 | 8.9 | 8 |
| | Average MAC efficiency (%) | 99.9 | 85.6 | 86.8 | 95.4 | 92.8 |
| | Average collision ratio (%) | 0.1 | 13.9 | 11.8 | 4.8 | 7 |
| | Average QoS (%) | 25.2 | 10.9 | 23.3 | 69.2 | 64.6 |
| 10 / CBR | Average delay (msec) | 814.7 | 1788 | 1520.3 | 688.3 | 621.6 |
| | Average jitter (msec) | 14.6 | 40.1 | 36.3 | 16.5 | 21.3 |
| | Average throughput (Kbps) | 1322 | 1025.5 | 1089.1 | 1160.3 | 1240.7 |
| | Average loss (%) | 8.3 | 27.1 | 19.5 | 9.7 | 8 |
| | Average MAC efficiency (%) | 99.8 | 81.1 | 83.8 | 92.7 | 91.4 |
| | Average collision ratio (%) | 0.5 | 16.9 | 14.9 | 6.3 | 8.6 |
| | Average QoS (%) | 42.9 | 11.4 | 13.7 | 59.4 | 57.8 |
| 20 / CBR | Average delay (msec) | 1863.8 | 3348.6 | 3635.1 | 1180.5 | 1465.6 |
| | Average jitter (msec) | 29 | 95.1 | 92.7 | 34 | 48.2 |
| | Average throughput (Kbps) | 1185.3 | 931.6 | 976.3 | 974.4 | 1147.9 |
| | Average loss (%) | 8.8 | 24.7 | 23.9 | 8 | 11.8 |
| | Average MAC efficiency (%) | 99.4 | 78.6 | 80.4 | 88.1 | 88.7 |
| | Average collision ratio (%) | 1.2 | 17.1 | 16.9 | 11.2 | 11.3 |
| | Average QoS (%) | 44.1 | 18.5 | 19.8 | 42.9 | 37.4 |
| 5 / VBR | Average delay (msec) | 70.7 | 429.9 | 469 | 434.6 | 241.3 |
| | Average jitter (msec) | 7.4 | 14.6 | 14.9 | 10.8 | 11.3 |
| | Average throughput (Kbps) | 1384.8 | 1307.4 | 1326.1 | 1262.8 | 1293.9 |
| | Average loss (%) | 0.3 | 3.3 | 4 | 6.5 | 2.5 |
| | Average MAC efficiency (%) | 99.9 | 84.1 | 84.8 | 94.7 | 92.1 |
| | Average collision ratio (%) | 0.3 | 14.8 | 14.1 | 5.7 | 7.9 |
| | Average QoS (%) | 82.3 | 50.4 | 46.3 | 69.5 | 68.1 |
| 10 / VBR | Average delay (msec) | 531.8 | 1499.3 | 1231.1 | 538.5 | 381.9 |
| | Average jitter (msec) | 19.2 | 49.8 | 43.4 | 21.4 | 21.3 |
| | Average throughput (Kbps) | 1361.4 | 1124.5 | 1196.7 | 1291.3 | 1304.1 |
| | Average loss (%) | 2 | 13.1 | 7.5 | 3.4 | 2 |
| | Average MAC efficiency (%) | 99.6 | 76.8 | 81 | 89.5 | 90.6 |
| | Average collision ratio (%) | 1 | 21.2 | 16.4 | 9.8 | 9.5 |
| | Average QoS (%) | 57.9 | 25.1 | 29.2 | 65.4 | 63.6 |
| 20 / VBR | Average delay (msec) | 345.8 | 2729.2 | 1958.5 | 908.1 | 526.5 |
| | Average jitter (msec) | 26.8 | 97.7 | 88.8 | 34.5 | 35.5 |
| | Average throughput (Kbps) | 1121.1 | 812.7 | 966.4 | 857.1 | 1062.2 |
| | Average loss (%) | 0.9 | 10.1 | 6 | 2.7 | 1 |
| | Average MAC efficiency (%) | 98.7 | 76.8 | 76.4 | 87.6 | 88.7 |
| | Average collision ratio (%) | 2.1 | 15.4 | 17.9 | 10.2 | 9.8 |
| | Average QoS (%) | 64.4 | 21.7 | 28.2 | 63.3 | 52.7 |



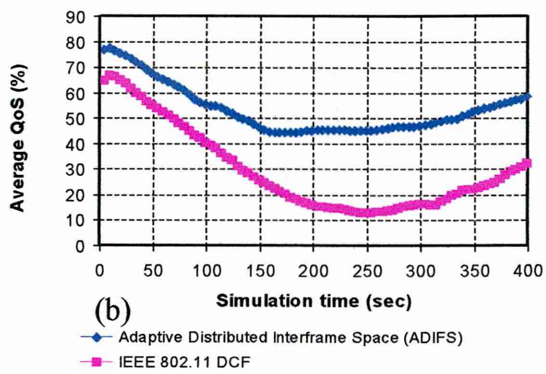



**Figure 7.18:** Average QoS for *ADIFS* vs. IEEE 802.11 DCF when the number of *CBR* connections was increased over time, (a) average delay (msec) and (b) average QoS (%).

In this section, a detailed description of the *ADIFS* scheme for improving the network performance was presented. The simulation results confirmed that the *ADIFS* scheme provided better performance than the IEEE 802.11 DCF and *EIED* schemes. It also resulted in an equivalent performance to the Ratio based and *CRV* schemes. The dynamic adjustment of the *DIFS* parameter resulted in a significant reduction in the probability of collisions. This led to a small number of packets being retransmitted and hence fewer variations in the *CW* size which gained *ADIFS* a better stability.

### 7.5.5 Ratio based with the Presence of RTS/CTS Access Mode

The *RTC/CTS* access mode is optional in the IEEE 802.11 standard (IEEE, 1999). It is used to alleviate the influence of the hidden terminal problems in multi-hop networks through the exchange of *RTS* and *CTS* control frames prior to packet transmission. The Ratio based and *CRV* schemes were used with the basic access mode of the IEEE 802.11 DCF scheme, since collisions occurred in the data packets rather than in the control frames. To relax this assumption, the performance of the Ratio based scheme was also evaluated and compared with the IEEE 802.11 DCF and *EIED* schemes when *RTS/CTS* access mode was used. In order to use the Ratio based and *CRV* schemes with the *RTS/CTS* access mode, Equations 1.1 to 1.10 and the procedures described in Figures 7.1 to 7.3 were used. The only difference was, in the case of the *RTS/CTS* access mode, the collision ratio was computed by considering the *RTS* and *CTS* control messages and the successfully received acknowledgements; while only data packets were considered when the basic access mode was used. To demonstrate the performance of the Ratio based approach when using the *RTS/CTS* access mode, the scenario discussed in section 7.5.2.3.4 was used. According to Figures 7.19a and 7.19b, the use of *RTS/CTS* access mode did not significantly affect the performance of the Ratio based scheme. The Ratio based scheme still provided a smaller average delay and a larger average QoS than the ones obtained for the standard IEEE 802.11 DCF and *EIED* schemes. The average delay was 13% and 16% less than the average delay obtained using the other two schemes. The average QoS obtained using the Ratio based scheme was 37.5% and 29.1% higher than the ones obtained for the IEEE 802.11 DCF and *EIED* schemes, respectively.

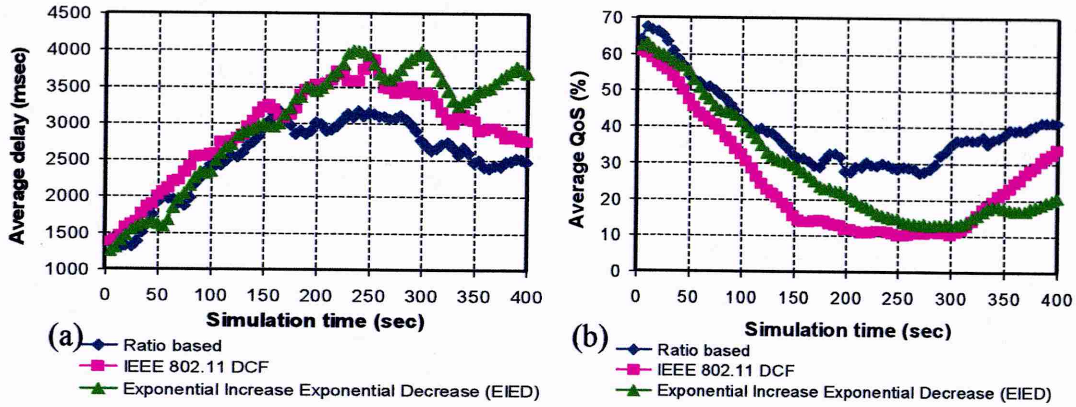



**Figure 7.19:** Performance evaluation of Ratio based, IEEE 802.11 DCF, and *EIED* schemes when *RTS/CTS* access mode was used, (a) average delay and (b) average QoS.

## 7.6 Summary

The main objective of this chapter is to describe the developed adaptive techniques namely Ratio based and *CRV* scheme that were used to enhance the performance and to improve the QoS of the IEEE 802.11 DCF scheme. In this respect, the chapter first reviewed the state of the art in section 7.2. A detailed description of the Ratio based and *CRV* schemes was provided in section 7.3. The simulation model was highlighted in section 7.4. A full description of the main findings was given in section 7.5.

The Ratio based and *CRV* schemes extended the legacy IEEE 802.11 DCF mechanism. The aim of developing these approaches was to reduce the probability of collisions in an attempt to improve QoS in IEEE 802.11 DCF protocol. The Ratio based and *CRV* schemes are easy to implement since they do not require major modifications to the IEEE 802.11 DCF frames format. The simulation results indicated that the Ratio based and *CRV* schemes showed better performance than the other two schemes regardless of the network size, traffic type, and the access mechanism used. The results indicated that the Ratio based scheme performed as the IEEE 802.11 DCF scheme in multi-hop networks with minor improvements. In the next chapter the Ratio based and *CRV* schemes including traffic type (i.e. time sensitive and time insensitive) will be used to improve the network performance and to provide service differentiation in single and multi-hop networks.

# *CHAPTER 8*

# Adaptive Service Differentiation in the IEEE 802.11 MAC Protocol

## 8.1 Introduction

In this chapter an extension to the IEEE 802.11 DCF scheme to support QoS and to provide service differentiation has been proposed. The service differentiation schemes were based on the Ratio based and *CRV* schemes, (see Chapter 7), that were used to adaptively adjust the *CW* and *DIFS* at runtime. The adjusted *CW* and *DIFS* parameters including the application type (i.e. time-sensitive and time-insensitive) have been employed for providing service differentiation at MAC layer in single-hop networks.

Time-sensitive applications were assigned as high priority traffic while time-insensitive applications have been assigned as low priority traffic[5]. Traffic differentiation can be achieved by accessing the Type of Service (*ToS*) field in the Internet Protocol (*IP*) header. In the *ToS* field up to six classes can be handled where class zero is assigned for high priority (Muller, 2003). The variation of *CW* and *DIFS* values for time-sensitive applications has been implemented in an adaptive manner that attempts to meet the QoS requirements of those applications. For time-insensitive applications the variation of *CW* and *DIFS* parameters is related to achieving high throughput regardless of the amount of delay they might experience taking into account less packet drops at the buffer.

In the standard IEEE 802.11 MAC protocol one single queue is employed with only best-effort service (Garroppo et *al.*, 2006) and (IEEE, 1999). In multi-hop networks any station can be an intermediate station; therefore its InterFace Queue (*IFQ*) can be occupied by different traffic types which are serviced in First In First Out (*FIFO*)[6] manner. Subsequently, this best-effort service is insufficient to meet the QoS requirements for time-sensitive applications. Consequently, a queue status monitoring scheme with packet drop mechanism in multi-hop networks was proposed. The queue status monitoring scheme was used by following two strategies: (i) adaptively adjust the transmission rate of low priority stations by sending a feedback control message to low

[5] High and low priority traffic, high and low priority packets, high and low priority stations, and high and low priority connections were used interchangeably, since in the proposed approaches were based on per station differentiation.
[6] In IEEE 802.11 MAC protocol there is only one single queue called *IFQ*. It is an interface queue that queues all arriving packets to be served by MAC protocol in a *FIFO* manner.

priority sources to adjust their transmission and (ii) discarding low priority packets when the queue-status exceeded certain limits known as queue-status thresholds. However, by combining the queue-status monitoring scheme with the adaptive service differentiation scheme, QoS differentiation and increased network performance in multi-hop networks can be achieved.

This chapter is organised as follows: The current state of the art of providing service differentiation in IEEE 802.11 DCF scheme is described in the following section. Section 8.3 introduces a detailed description of the adaptive service differentiation schemes. The simulation model is presented in section 8.4. The results obtained are analysed and discussed in section 8.5.

## 8.2 Previous Studies for Providing Service Differentiation

In section 2.12.5 (see Chapter 2) a taxonomy of QoS and the state of the art of the recent studies in providing QoS over the IEEE 802.11 DCF scheme were discussed in detail, with the most relevant studies described. Most of the proposed priority-based approaches were aimed to support service differentiation by providing different MAC parameters values that enabled high priority classes to access the medium faster than low priority classes. For instance, faster access could be provided by assigning a smaller *CW* causing a smaller Backoff Interval (*BI*) as reported in (Veres et *al.*, 2001), (Kim et *al.*, 2001), (Barry et *al.*, 2001), (Gannoune, 2006) or by assigning smaller Inter Frame Space (*IFS*) as reported in (Deng and Chang, 1999), (Aad and Castelluccia, 2001), and (Ksentini et *al.*, 2004).

Service differentiation through allocating different *CW* values was based on two techniques: (i) *CW* differentiation (*CWD*) and (ii) *CW* separation (*CWS*). These were discussed in details in section 2.12.5 (see Chapter 2). The work proposed in (Barry et *al.*, 2001), (Ayyagari et *al.*, 2000), (Chen et *al.*, 2002), and (Gannoune, 2006) was based on *CWD* scheme. For instance, in (Ayyagari et *al.*, 2000), the small values of $CW_{min}$ and $CW_{\max}$ were assigned to high priority traffic where the $CW_{min}$ of high priority class was less than the $CW_{min}$ of low priority class and the $CW_{\max}$ of high priority class was less than the $CW_{\max}$ of low priority class. Another example that used the *CWD* scheme to provide service differentiation was proposed in (Chen et *al.*, 2002). In this approach the *CW* range was dynamically adjusted with respect to the variation in the number of active stations. A priority reference value called priority limit that was piggy-backed with the transmitted packets to help each individual station to compute its *CW* size. In this *CWS* scheme, the $CW_{min}$ and $CW_{max}$ values of high priority traffic are completely separated from the $CW_{min}$ and $CW_{max}$ of low priority traffic. The scheme in (Deng *et al.*, 1999) is an example of the *CWS*. Two different *CW* values for high and low priorities were specified.

Using *IFS* is another technique for providing service differentiation in IEEE 802.11 MAC protocol, which is based on: (i) using the existing *IFS* values defined by the standard such as *SIFS*, *PIFS*, and *DIFS* and (ii) using new *IFS* values. Different schemes were proposed based on the already available *IFS* values. For example, the proposed approaches in (Deng et *al.*, 1999), (Shue and Shue, 2001), and (Banchs et *al.*, 2001) used *PIFS* and *DIFS* values to differentiate between time-sensitive and time-insensitive applications. Some other approaches used new *IFS* values to differentiate between high and low priority traffic. These new *IFS* values were based on allocating the low priority traffic longer *IFS* value than the *IFS* value of high priority traffic. For instance, in (Aad and Castelluccia, 2001) different schemes to provide service differentiation were proposed: different *BI*, different $CW_{min}$, different *IFS*, and different frame length. The $CW_{min}$ and *IFS* values in their scheme were statically assigned.

In (Kanodia et *al.,* 2001), relative priorities for delay and throughput in multi-hop networks were proposed. This approach aimed to send back the scheduling information into an *RTS/DATA* packet and then used this information to modify the backoff times. This approach required all stations to monitor all transmitted packets in order to obtain the scheduling information which in turn increased the overhead of the network.

Other studies such as (Barry et *al.*, 2001), (Ayyagari et *al.* 2000) and (Imad and Castelluccia, 2000) were proposed to provide service differentiation based on the distributed function of the standard. These schemes were based on modifying the backoff time of IEEE 802.11 MAC protocol. Although significant research efforts have been carried out on supporting service differentiation in IEEE 802.11 DCF by adopting the priority-based scheme, several issues have still not been considered which were described in section 2.13 (see Chapter 2). In this chapter the following points were considered for providing service differentiation in the basic IEEE 802.11 DCF scheme: (i) MAC protocol parameters (such as *CW* and *DIFS*) were dynamically adjusted, (ii) different QoS metrics such as delay, jitter, throughput, packet loss, MAC efficiency, collision rate, and cumulative distribution of delay and QoS were used, (iii) an adaptive

*CW* differentiation scheme was developed, (iv) an adaptive *DIFS* differentiation scheme was devised, (v) a hybrid adaptive service differentiation scheme was developed in order to avoid the drawbacks of *CWD*, *CWS* and *IFS* differentiation, and (vi) a feedback control system was developed based on monitoring the queue of the intermediate station to avoid congestion and to provide service differentiation in multi-hop networks.

## 8.3 Description of the Approach

This chapter presents a detailed description of the proposed service differentiation schemes. As depicted in Figure 8.1, an adaptive service differentiation scheme composed of four main parts is presented. The first part was classification of traffic into high and low priorities. This was achieved by accessing the Type of Service (*ToS*) bit pattern in the *IP* header. The second part was the recording part. Each station recorded the required information such as the number of generated packets, sent packets, successfully acknowledged packets, collided packets, and the number of packets residing in the *IFQ*. This recorded information was used as input to the third part (i.e. calculation). In the calculation part, *CR*, *CRV*, packet loss rate, and queue status thresholds values were computed and fed to the final part (i.e. adjustment). The final part made the decision on choosing a appropriate parameter values.

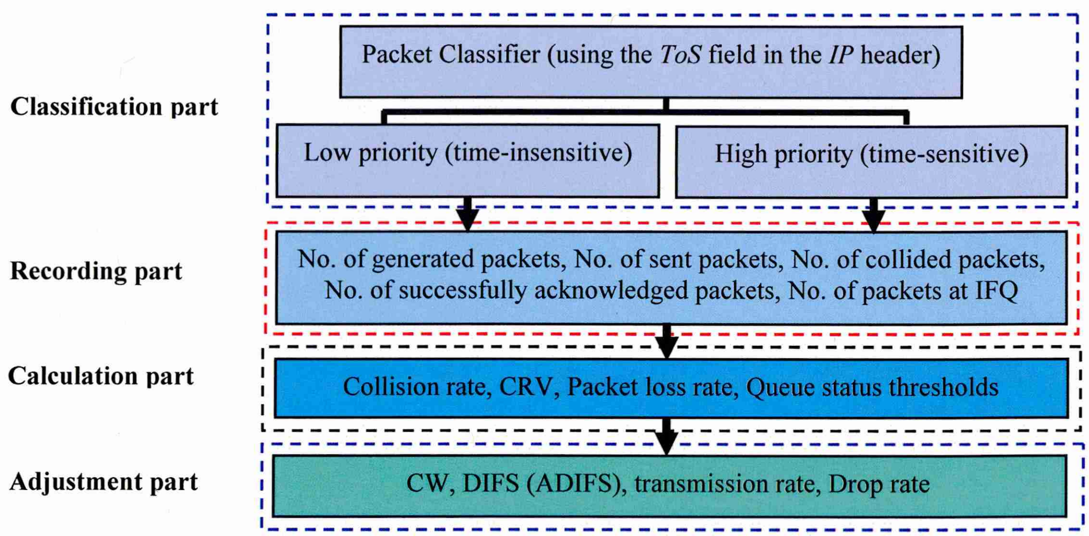



**Figure 8.1:** Adaptive service differentiation scheme.

In Chapter 7, the Ratio based and *CRV* schemes were used to adjust *CW* and *DIFS* values without considering the application type. In this chapter, these parameters were adjusted when the application type was considered in order to provide service differentiation in single and multi-hop networks. The same algorithms shown in Figures 7.1, 7.2 and 7.3 (see Chapter 7) were used for service differentiation with some modifications based on the application type (i.e. traffic priority). These modifications were presented for each individual parameter in the next sections.

### 8.3.1 Contention Window Adjustment for Service Differentiation

In order to provide service differentiation and to improve protocol performance, the proposed scheme used two different cases to compute the *CW* size. The first one was used for high priority traffic and the second one was used for low priority traffic.

#### 8.3.1.1 Contention Window Adjustment after Successful Transmission

For *CW* differentiation, packet loss rate and collision ratio values were required by each station to adjust the *CW* value. Collision ratio and $R_{average}^{wi}$ [7] values were computed using Equations 7.1 and 7.2 (see section 7.3.1.1 in Chapter 7). Packet loss rate ($l[N_i]$) was calculated using Equation 8.1. This was computed based on the number of successfully received acknowledgments and the number of generated packets at the sender. Packet loss rate was updated at a constant period of time. The update period was chosen to be sufficiently long in order to get good reactivity (i.e. provided the loss rate value when required) and not to be too short in order to avoid the complexity (i.e. less computation).

$$l[N_i] = 1 - \frac{Num(success\_Ack[N_i])}{Num(gen\_packets[N_i])} \qquad (8.1)$$

Where $Num(success\_Ack[N_i])$ is the number of successfully received acknowledgement for a station $N$, $Num(gen\_packets[N_i])$ is the number of generated packets by the station, $i$ stands for high priority class, and $l[N_i]$ is the packet loss rate of high priority station $N_i$.

The packet loss rate was chosen since it provided a relevant indication of the application perceived QoS. Therefore, when the packet loss rate ($l[N_i]$) of a given high priority station exceeds a certain threshold ($l\_ths[N_i]$) (the value of this threshold was chosen to be 5% to meet the QoS for time-sensitive applications), the *CW* of the high priority station is rapidly decreased as depicted in Equation 8.2 in order to reduce the delay and to avoid excessive packet loss. As a result of the sharp decrease in the *CW* size of the high priority station, the high priority station becomes more aggressive to access the medium which in turn increases the probability of collisions over the medium.

[7] This value was obtained when part of previous collisions was considered with the current collision ratio.

$$CW_{new}[N_i] = CW_{new-1}[N_i] - \left( \frac{CW_{new-1}[N_i] * R^{wi}_{average}[N_i]}{f-1} \right) \tag{8.2}$$

The increase in the number of collisions leads to an increase in the *CRV* value of low priority traffic (i.e. $CRV[N_j] > 0$), this forces the low priority station to gently increase its *CW* size as described in Equation 8.3.

$$CW_{new}[N_j] = CW_{new-1}[N_j] + \left( \frac{CW_{new-1}[N_j] * R^{wi}_{average}[N_j]}{f} \right) \tag{8.3}$$

Where $CW_{new}[N_i]$ is the new *CW* for high priority traffic and $CW_{new}[N_j]$ is the new *CW* for low priority traffic, $CW_{new-1}[N_i]$ and $CW_{new-1}[N_j]$ are the previous *CW* for high and low priorities, respectively, $f$ is a scaling factor which was chosen based on extensive simulations for several scenarios as discussed in section 7.5.1 (see Chapter 7), $R^{wi}_{average}[N_i]$ and $R^{wi}_{average}[N_j]$ represent the average collision ratio of high and low priority traffic, respectively.

When the packet loss rate of high priority traffic goes below the $l\_ths[N_i]$ threshold, the *CW* of the high priority station is gently decreased as depicted in Equation 8.4.

$$CW_{new}[N_i] = CW_{new-1}[N_i] - \left( \frac{CW_{new-1}[N_i] * R^{wi}_{average}[N_i]}{f} \right) \tag{8.4}$$

The gradual decrease in the *CW* size of the high priority station leads to a reduction in the *CRV* value of low priority traffic, once the *CRV* value of low priority traffic becomes below zero (i.e. $CRV[N_j] < 0$), a low priority station gradually decreases its *CW* as described in Equation 8.5. The gradual decrease in the *CW* size of the low priority station provides more access opportunities. This in turn improves the whole network performance.

$$CW_{new}[N_j] = CW_{new-1}[N_j] - \left( \frac{CW_{new-1}[N_j] * R^{wi}_{average}[N_i]}{f+1} \right) \tag{8.5}$$

Note that, the *CW* size of low priority traffic is gradually changed according to the computed *CRV* value ($CRV[N_j]$). So, if $CRV[N_j] > 0$ (i.e. positive), the low priority station assumes high contention over the network and gradually increases its *CW* using Equation 8.3 in order to reduce the probability of collisions. When $CRV[N_j] < 0$ (i.e. negative), a low priority station assumes less contention and therefore decreases its *CW* using Equation 8.4 to reduce the idle time slots.

#### 8.3.1.2 Contention Window Adjustment after Unsuccessful Transmission

As in the case of successful transmission, the collision ratio ($CR$) and the $R^{wi}_{average}$ values were obtained using Equations 7.1 and 7.2 in section 7.3.1.1 (see Chapter 7). Large values of $R^{wi}_{average}$ indicate that many stations contend to access the medium, whereas, small values of $R^{wi}_{average}$ indicate fewer stations contend to gain access to the medium. Packet loss rate ($l[N_i]$) in Equation 8.1 and packet loss threshold ($l\_ths[i]$) were also used to maintain a high level of QoS for high priority traffic and to provide service differentiation. For low priority traffic, the obtained $CR$ and $R^{wi}_{average}[N_j]$ values were used in order to limit the access of low priority traffic particularly in overloaded networks. To achieve this goal, each high and low priority station computes its $CR$ and $R^{wi}_{average}$ values, as the collision ratio value is an indication of the number of active stations. Thus, low priority stations use the $CRV$ value to update their $CW$ size as discussed in section 7.3 (see Chapter 7). The $CRV$ value determines whether the current collision ratio is smaller or larger than the previous one. High priority stations use the packet loss rate ($l[N_i]$) and packet loss rate threshold ($l\_ths[N_i]$) to update their $CW$ size in order to maintain high throughput and less delay.

For high priority traffic and after a collision, the packet loss rate value is examined. If the packet loss rate ($l[N_i]$) is greater than the packet loss rate threshold ($l\_ths[N_i]$), the $CW$ size is slightly increased in order to minimise the wasted time slots as described in Equation 8.6.

$$CW_{new}[N_i] = CW_{new-1}[N_i] + \left( \frac{CW_{new-1}[N_i] * R^{wi}_{average}[N_i]}{f} \right) \tag{8.6}$$

If the packet loss rate ($l[N_i]$) is smaller than the packet loss rate threshold ($l\_ths[N_i]$), the $CW$ size is rapidly increased in order to reduce the probability of collision and to reduce the retransmission of the collided packets as described in Equation 8.7.

$$CW_{new}[N_i] = CW_{new-1}[N_i](1 + R^{wi}_{average}[N_i] * f) \tag{8.7}$$

For low priority traffic and after unsuccessful transmission, the $CRV[N_j]$ value is examined. If the $CRV[N_j]$ is greater than zero (i.e. a positive value), the value of $CW$ is sharply increased by a multiplication factor ($f + 2$) in order to reduce the probability of collisions and to protect the high priority traffic from degradation as in Equation 8.8. The selection of the scaling factor ($f$) was discussed in section 7.5.1 (see Chapter 7).

$$CW_{new}[N_j] = CW_{new-1}[N_j](1 + R^{wi}_{average}[N_j]*(f+2)) \tag{8.8}$$

If the $CRV[N_j]$ value is less than zero, the $CW$ size of low priority traffic is also increased by a smaller multiplication factor as shown in Equation 8.9 taking into account the current and the previous collision ratios.

$$CW_{new}[N_j] = CW_{new-1}[N_j](1 + R^{wi}_{average}[N_j]*(f+1)) \tag{8.9}$$

In order to ensure that the $CW$ sizes of the two classes do not go below the $CW_{min}$ or do not exceed the $CW_{max}$, the following conditions are set.

If ($CW_{new}[N_i]$ or $CW_{new}[N_j]) < CW_{min}$ then ($CW_{new}[N_i]$ and $CW_{new}[N_j]) = CW_{min}$.

If ($CW_{new}[N_i]$ or $CW_{new}[N_j]) > CW_{max}$ then ($CW_{new}[N_i]$ and $CW_{new}[N_j]) = CW_{max}$.

### 8.3.2 Adaptive Distributed Inter Frame Space for Service Differentiation

In order to set a proper *DIFS* length for each class, the Collision Ratio (*CR*), *CRV*, and packet loss rate values have to be computed at runtime. Based on these computed values, the proposed *ADIFS* approach can determine how much the *DIFS* length should be increased or decreased as discussed in section 7.3 (see Chapter 7).

In order to ensure that service differentiation among different priorities was fulfilled, the maximum *DIFS* length of high priority traffic (i.e. $ADIFS_{new}[N_i]$) was limited to initial *DIFS* ($DIFS_{init}$), where $DIFS_{init}$ is equal to 50 $\mu s$ as defined by the standard; whereas the minimum length of ($ADIFS_{new}[N_i]$) was limited to one slot time (one slot time equal to 20 $\mu s$ as defined by the standard). The minimum value of *DIFS* of high priority traffic ($ADIFS_{new}[N_i]$) was chosen to be longer than the Short Inter Frame Space (*SIFS*) that was assigned for the *ACK* frame as defined by the standard, while the maximum value of *DIFS* of high priority traffic ($ADIFS_{new}[N_i]$) was chosen in order to avoid excessive waiting times (i.e. reduce the delay and packet drops of high priority).

Regarding low priority traffic, the minimum *DIFS* length of low priority traffic (i.e. $ADIFS_{new}[N_j]$) was bounded to $DIFS_{init}$ while the maximum length was bounded to seven time slots (i.e. 140 $\mu s$). The minimum value of $ADIFS_{new}[N_j]$ was selected in order to alleviate the overlap with the *DIFS* of high priority traffic, while the maximum value of $ADIFS_{new}[N_j]$ was chosen in order to reduce the probability of collision on heavily loaded networks.

For high priority traffic, when the Collision Rate Variation value of a high priority station ($CRV[N_i]$) is greater than zero, the proposed scheme examines the packet loss rate ($l[N_i]$), if the packet loss rate is below the packet loss rate threshold (i.e. $l[N_i] < l\_ths[N_i]$), the $DIFS$ length of high priority ($ADIFS_{new}[N_i]$) is set equal to $DIFS_{init}$ in order to give low priority traffic a greater chance to access the channel. If the packet loss rate exceeds the packet loss rate threshold (i.e. $l[N_i] > l\_ths[N_i]$), the $DIFS$ of high priority traffic ($ADIFS_{new}[N_i]$) is reduced by one slot time to reduce the delay and to prevent excessive packet loss for high priority packets.

When the $CRV[N_i]$ of high priority traffic is less than zero, the adaptive approach examines the packet loss rate $l[N_i]$ value, if this value is below the packet loss rate threshold ($l\_ths[N_i]$), the $DIFS$ of high priority ($ADIFS_{new}[N_i]$) is set equal to $DIFS_{init}$, while if the packet loss rate value $l[N_i]$ is above the packet loss rate threshold ($l\_ths[N_i]$), the $DIFS$ value of high priority packets ($ADIFS_{new}[N_i]$) is updated as given in Equation 8.10.

$$ADIFS_{new}[N_i] = DIFS_{init}(1 + CRV[N_i]) \tag{8.10}$$

For low priority traffic when the $CRV[N_j]$ value is larger than zero, this implies that the number of contending stations is increased and as a result, the probability of collisions is also increased since the current collision ratio is larger than the previous one. Therefore the $ADIFS_{new}[N_j]$ length is increased and updated using Equation 8.11.

$$ADIFS_{new}[N_j] = DIFS_{init} + (f * CRV[N_j] * ADIFS_{new-1}[N_j]) \tag{8.11}$$

If the $CRV[N_j]$ value is less than zero, this means that the current probability of collisions is smaller than the previous one, and as a result, the proposed approach decreases the $ADIFS_{new}[N_j]$ by one slot time as represented in Equation 8.12.

$$ADIFS_{new}[N_j] = ADIFS_{new-1}[N_j] - (one\_slot\_time) \tag{8.12}$$

To ensure that the lengths of $ADIFS_{new}[N_i]$ and $ADIFS_{new}[N_j]$ are within the specified ranges the following conditions are applied:

In Equation 8.10, if $ADIFS_{new}[N_i] <$ one time-slot then $ADIFS_{new}[N_i]$ = one slot.

In Equation 8.11, if $ADIFS_{new}[N_j] >$ seven slots then $ADIFS_{new}[N_j]$ = seven slots.

In Equation 8.12, if $ADIFS_{new}[N_j] < DIFS_{init}$ then $ADIFS_{new}[N_j] = DIFS_{init}$.

### 8.3.3 Adaptive Differentiation Scheme

In this scheme the *CW* and *DIFS* parameters were considered in the adjustment process. The adaptive service differentiation approach employed the same metrics discussed in sections 8.3.1 and 8.3.2 to adjust the *CW* and *DIFS* for high and low priority traffic.

In the adaptive service differentiation approach, the two parameters were updated simultaneously according to the collision ratio, *CRV*, and packet loss rate values for each class. Following Equations 8.1 to 8.12 that were discussed in sections 8.3.1 and 8.3.2, the high priority traffic gain smaller values of *CW* and *DIFS* than low priority ones. This enables a high priority station to gain more access to the wireless medium which results in a smaller delay and a smaller packet loss rate. At the same time, low priority is not ignored. The values of *CW* and *DIFS* for both priorities are varied in an adaptive manner in which service differentiation is provided. High priority traffic has small values of delay and packet loss rate and low priority traffic has high throughput and fewer drops at the buffer.

To avoid starvation for low priority traffic, after each update of the *CW* and *DIFS*, the adaptive service differentiation approach examines the values of these parameters. If either *CW* or *DIFS* experience high values (these values were determined by extensive simulations), the proposed scheme sets these parameters as shown in Equations 8.13, 8.14 and 8.15. A full description of the adaptive service differentiation schemes is provided in Figure 8.2.

$$if\,(priority = high) \quad then \quad CW_{new}[N_i] = CW_{\min} \tag{8.13}$$

$$if\,(priority = low)\ then\ CW_{new}[N_j] = CW_{new-1}[N_j] - \left(\frac{CW_{new-1}[N_j] * R^{wi}_{average}[N_j]}{f+1}\right) \tag{8.14}$$

$$if\,(priority = low) \quad then \quad ADIFS_{new}[N_j] = ADIFS_{new-1}[N_j] - (one\_slot\_time) \tag{8.15}$$

Following Equations 8.1 to 8.15 and Figure 8.2, the adaptive service differentiation approach tries to narrow the gap between the two priorities in normal operating conditions (i.e. light load conditions) and extends this gap when the network becomes overloaded. This can be explained as follows: small values of packet loss rate and collision ratio indicate that the network operates in normal conditions. As a result, high priority stations increase slightly their *CW* and *DIFS* in order to give low priority stations more opportunities to access the channel. Simultaneously, due to the increase in *CW* and *DIFS* of high priority stations, the *CRV* values of low priority stations decrease and go below zero (i.e. become negative). Because the current collision ratio is less than the previous collision ratio and, low priority stations decrease their *CW* and *DIFS* values this improves their access to the medium and finally achieve better overall performance. This in turn reduces the differentiation between low and high priorities. For a heavily loaded network, when the packet loss rate of high priority stations exceeds the packet loss rate threshold (i.e. $l[N_i] > l\_ths[N_i]$), high priority stations decrease their *CW* and *DIFS* values as they attempt to reduce their delay and to avoid excessive packet losses. Hence after, the reduction in the *CW* and *DIFS* of high priority stations, results in more collisions. Subsequently, this increases the *CRV* values of low priority stations which in turn make low priority stations increase their *CW* and *DIFS* values. This increases the service differentiation gap between high and low priorities in order to protect high priority traffic from the impact of low priority traffic at overloaded conditions.

An overview of the adaptive differentiation operation is provided in Figure 8.3. It is assumed that there are two stations, one is a high priority station and the other one is a low priority station. Following the previous transmission, the high priority station is competing with smaller *CW* and smaller *DIFS*, while the low priority station is competing with larger *CW* and larger *DIFS*. Therefore, the high priority station accesses the medium first. After successful or unsuccessful transmission, both stations update their *CW* and *DIFS* parameters and follow the adaptive differentiation scheme operation discussed in the previous sections (see sections 8.3.1, 8.3.2 and 8.3.3).

The proposed adaptive service differentiation scheme in the MAC layer discussed so far for the situation when the contending stations are located in the same Independent Basic Service Set (*IBSS*). In multi-hop networks, in order to improve the network performance and to provide service differentiation, the same adaptive differentiation scheme explained in Figure 8.2 is combined with the queue status monitoring approach. The full description of the queue status monitoring scheme is presented in the next section.

***Adaptive Contention Window (ACW) differentiation scheme***
***(ACW) After successful transmission***

*If (priority is high) {*
*If (packet loss rate of high priority < packet loss threshold) {*
*Update CW using Equation 8.4*
*} Else if (packet loss rate of high priority > packet loss threshold) {*
*Update CW using Equation 8.2*
*}*
*}*
*If (priority is low) {*
*If (Collision Rate Variation (CRV) of low priority > 0) {*
*Update CW using Equation 8.3*
*} Else if (Collision Rate Variation (CRV) of low priority < 0){*
*Update CW using Equation 8.5*
*}*
*}*

***(ACW) After unsuccessful transmission***
*If (priority is high) {*
*If (packet loss rate of high priority < packet loss threshold) {*
*Update CW using Equation 8.7*
*} Else if (packet loss rate of high priority > packet loss threshold) {*
*Update CW using Equation 8.6*
*}*
*}*

*If (priority is low) {*
*If (Collision Rate Variation (CRV) of low priority > 0) {*
*Update CW using Equation 8.8*
*} Else if (Collision Rate Variation (CRV) of low priority < 0){*
*Update CW using Equation 8.9*
*}*
*}*

***Adaptive Distributed Inter Frame Space (ADIFS) differentiation scheme***

*If (priority is high) and (Collision Rate Variation of high priority > 0) {*
*If (packet loss rate of high priority < packet loss threshold) {*
*DIFS of high priority = DIFS (i.e. 50 $\mu s$ as defined by the standard)*
*} Else if (packet loss rate of high priority > packet loss threshold) {*
*DIFS of high priority = DIFS - one slot time (i.e. 20 $\mu s$ as defined by the standard)*
*}*
*}*
*If (priority is high) and (Collision Rate Variation of high priority < 0) {*
*If (packet loss rate of high priority < packet loss threshold) {*
*DIFS of high priority = DIFS (defined by the standard)*
*} Else if (packet loss rate of high priority > packet loss threshold) {*
*Update DIFS of high priority using Equation 8.10*
*}*
*}*
*If (priority is low) {*
*If (Collision Rate Variation of low priority > 0) {*
*Update DIFS of low priority using Equation 8.11*
*} Else if (Collision Rate Variation of low priority < 0) {*
*Update DIFS of low priority using Equation 8.12*
*}*
*}*

**Figure 8.2:** Adaptive differentiation scheme.

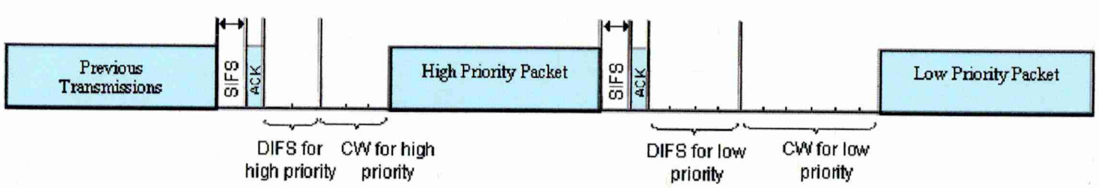



**Figure 8.3:** Adaptive differentiation scheme operation.

### 8.3.4 Queue Status Monitoring Scheme

In wireless networks congestion occurs when the total amount of data delivered to the network exceeds the available channel capacity at any point in a network. During congestion the interface queue of the intermediate station in multi-hop networks grows quickly which causes a longer delay and higher packet drops at the buffer for packets traversing through this intermediate station. Ultimately, this degrades the overall channel quality. Therefore, congestion is undesirable in wireless networks as they have limited resources and dynamic nature, particularly in multi-hop networks where data packets go across a large number of radio hops.

Unlike the IEEE 802.11e MAC protocol, the standard IEEE 802.11 MAC protocol has only one single queue that is used in the original best-effort service without having the capability neither to support QoS nor to provide service differentiation (Garroppo et *al.*, 2006). Therefore, using a monitoring approach supported by a proper feedback control message may provide service differentiation and lead to better network performance.

Congestion feedback can be implemented in different ways. In the queue status monitoring scheme proposed here, when congestion takes place, the intermediate station either sends a feedback control message to the data source to slow down its transmission or initiates a dropping mechanism when the arriving packets can not be accommodated in its buffer. The feedback is determined according to three parameters: queue status ratio ($\rho$), minimum queue threshold ($Q_{\min\_ths}$) and maximum queue threshold ($Q_{\max\_ths}$).The queue status ratio ($\rho$) is obtained using Equation 8.16. $Q_{\min\_ths}$ and $Q_{\max\_ths}$ are set to 25% and 75% of the maximum queue length, respectively. The minimum and the maximum queue thresholds are chosen to have an average queue occupancy close to 50% of the maximum queue limit.

$$\rho = \frac{Num(packets\ currently\ in\ the\ queue)}{Maximum\ queue\ length} \tag{8.16}$$

When the queue status ratio goes below the minimum queue threshold (i.e. $\rho < Q_{\min\_ths}$), no feedback message is activated. The feedback control message includes the queue status value, the source address, destination address, intermediate station address, and frame control. When the queue status ratio goes above the maximum queue threshold (i.e. $\rho > Q_{\max\_ths}$), the feedback control message and the dropping scheme are stimulated. The intermediate station searches for low priority packets at the buffer and drops them to accommodate the arriving high priority packets, since high priority packets are considered more important than low priority packets. Once the computed queue status ratio locates a value between the minimum and the maximum queue status thresholds (i.e. $Q_{\min\_ths} < \rho < Q_{\max\_ths}$), the intermediate station sends a feedback control message to low priority stations to reduce their transmission rate. The intended source, upon receiving the control message extracts the queue status value and reduces its transmission rate by modifying the inter-packet interval as in Equations 8.17.

$$T_{new}[N_j] = T_{new-1}[N_j] + (T_{new-1}[N_j] * \rho) \tag{8.17}$$

Where $T_{new}[N_j]$ and $T_{new-1}[N_j]$ are the new and previous inter-packet intervals (i.e. time interval between two consecutive packets) for low priority traffic, respectively.

The station transmission rate is a function of its inter-packet interval; therefore, the intended station updates its transmission rate according to the new inter-packet interval using Equation 8.18.

$$Gen[N_j] = \frac{PK[N_j]}{T[N_j]} \tag{8.18}$$

Where $Gen[N_j]$ is the generation rate of a low priority station, $PK[N_j]$ is the packet size generated by the low priority station ($N$), and $T[N_j]$ represents the inter-packet interval (i.e. time interval between two consecutive low priority packets).

From Equations 8.17 and 8.18, it can be observed that the generation rate is a function of packet size and inter-packet interval. However, an increase in the inter-packet interval leads to a reduction in the generation rate. Therefore, once the low priority station receives the feedback control message it computes the inter-packet interval and then updates its generation rate (i.e. reducing the number of generated packets per unit of time). The reduction in the number of generated low priority packets reduces the chances of congestion which in turn ease traverse of the high priority traffic to their corresponding destination.

If the low priority station does not receive the feedback control message, it continues its transmission according to the previous inter-packet interval. This ultimately causes a fast growth in the *IFQ* of the intermediate station. As a result, the intermediate station, in order to avoid the congestion checks the timeout of the previous feedback control message and then sends another feedback control message with the new value of the queue status ratio. Note that, the feedback control message does not impose extra overhead, since it is only transmitted when congestion occurs.

The operation of the proposed queue status monitoring scheme when it is combined with the adaptive differentiation schemes (see sections 8.3.1, 8.3.2 and 8.3.3) is illustrated in Figure 8.4. Using these schemes, congestion at the intermediate station can be avoided, which in turn improves the network performance and provides service differentiation in multi-hop networks.

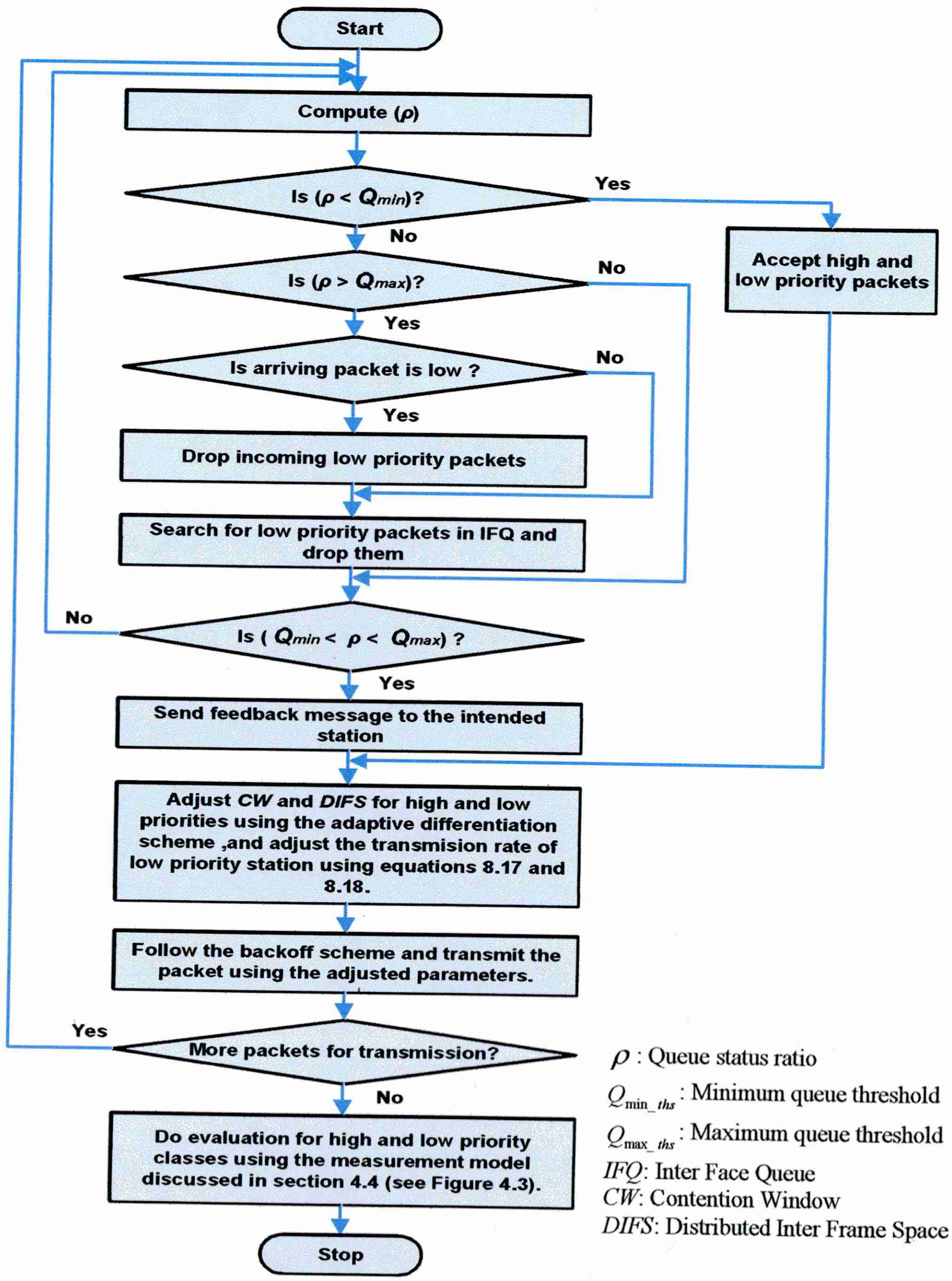



**Figure 8.4:** Flow chart for the Queue status monitoring approach.

## 8.4 Simulation Model

To evaluate the validity of the proposed service differentiation and queue status monitoring schemes, and compare their functionalities with the standard IEEE 802.11 DCF scheme, two network models with different scenarios have been proposed for the simulations. In the first model, the *IBSS* topology depicted in Figure 4.2d (see Chapter 4) was used. The stations in this model were classified into high priority stations that transmitted high priority *CBR* traffic and low priority stations that transmitted low priority *CBR* traffic. The second network was based on a random multi-hop topology as shown in Figure 4.2c (see Chapter 4). The packet sizes used for *CBR* traffic were, 512 and 800 bytes for high and low priority traffic, respectively. Each high priority station generated 192 Kbps. Each low priority station generated 480 Kbps and 160 Kbps as high and low bit rate, respectively. The simulation for each model was carried out 10 times to reduce the bias of random number generation. The simulation time lasted 300 seconds in order to obtain an accurate and consistent result in a steady state condition.

The simulations were performed for two sets. (i) single-hop networks and (ii) multi-hop networks. The first set included the following scenarios:

(i). This scenario consisted of five connections, two high priority connections and three low priority connections. Each high priority station transmitted at 192 Kbps while each low priority station transmitted at 480 Kbps. This scenario was carried out for *CW* differentiation, *DIFS* differentiation, and adaptive differentiation schemes.

(ii). This scenario involved 10 connections, five high priority connections and five low priority connections. Each high priority station transmitted 192 Kbps and each low priority station transmitted 160 Kbps. *CW* differentiation, *DIFS* differentiation, and adaptive differentiation schemes were considered in this scenario.

In the first set of scenarios, the total offered load in each scenario was more than 110% of the effective channel capacity (i.e. it is considered 1.6 Mbps without considering the protocol overhead) and more than 90% of the total channel capacity (i.e. 2 Mbps, with considering the impact of protocol overhead).

In the multi-hop network, two high priority and three low priority connections were setup. Each station transmitted 192 Kbps to its correspondent destination traversing more than two radio hops as depicted in Figure 4.2c (see Chapter 4).

## 8.5 Results and Discussion

In order to demonstrate the feasibility and the functionality of the proposed service differentiation and the queue status monitoring schemes, this section is divided into two subsections. The first demonstrates the results of single-hop networks and the second one presents the results of multi-hop networks. In single-hop networks, the results of *CW*-based differentiation, *DIFS*-based differentiation, and adaptive service differentiation schemes are evaluated.

### 8.5.1 QoS Differentiation in Single-hop Networks

In this section, the results of each individual differentiation scheme are discussed and compared with the basic IEEE 802.11 DCF scheme.

#### 8.5.1.1 Adaptive Contention Window for QoS Differentiation

In this section, the two scenarios discussed in section 8.4 were carried out to demonstrate the capability of using the adaptive *CW* differentiation scheme in providing service differentiation in single-hop networks.

In the first scenario, the transmission rate for each high priority station was 192 Kbps while each low priority station transmitted at 480 Kbps. Figure 8.5a plots the cumulative distribution of packets that have delay below certain values. For time-sensitive application (i.e. high priority packets), these values should not exceed 400 msec to meet the minimum QoS requirements ($ITU_{(a)}$, 2001). The distribution of delay for high priority packets was clearly better than the distribution of delay for low priority packets. More than 80% of high priority packets had values of delay less than 400 msec, while less than 70% of low priority packets had a delay of less than 400 msec. Thus, the adaptive *CW*-based differentiation scheme reduced the delay for high priority traffic and hence provided service differentiation between high and low priority packets as depicted in the QoS curves (see Figure 8.5b). In Figure 8.5b, the QoS of low priority traffic was also improved. The three low priority connections had good QoS levels with mean values equal to 43.9%, 53.2%, and 44.6% for the first, second, and the third connections, respectively. At the same time, high priority stations maintained excellent QoS levels with means of 73% and 71.6% for the first and the second high priority connections, respectively. It can be observed that the average QoS of high priority connections was 25.1% higher than the average QoS of low priority connections. This confirmed the capability of the *CW* differentiation scheme in providing service differentiation as shown in Figure 8.5c. The mean values of delay, jitter, throughput, MAC efficiency and the assessed QoS of each connection are shown in Table 8.1.

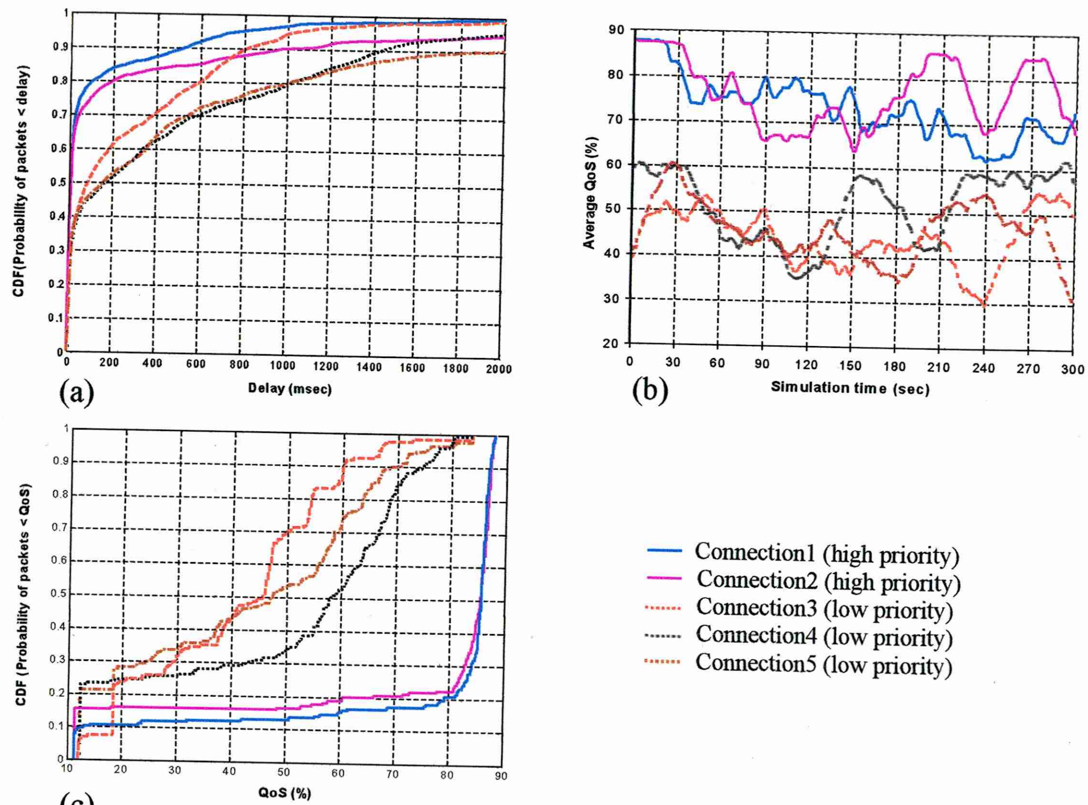



**Figure 8.5:** Adaptive *CW* differentiation for 5 connections (high and low priority stations), (a) cumulative distribution of delay, (b) average QoS, and (c) cumulative distribution of QoS.

**Table 8.1:** QoS parameters values using adaptive *CW* differentiation scheme.

| Bit rate / connection | Connection / priority | Average delay (msec) | Average jitter (msec) | Average throughput (Kbps) | Average MAC efficiency (%) | Average QoS (%) |
|---|---|---|---|---|---|---|
| 192 Kbps | **Connection1** / high | 213.7 | 9.7 | 190.1 | 92.5 | 73 |
| | **Connection2** / high | 218.2 | 10.2 | 176.9 | 93.5 | 71.6 |
| 480 Kbps | **Connection3** / low | 381.4 | 9.3 | 302.4 | 94 | 43.9 |
| | **Connection4** / low | 549 | 11.6 | 314.9 | 96.6 | 53.2 |
| | **Connection5** / low | 412.2 | 10 | 338 | 94.4 | 44.6 |

In the second scenario, each high priority station transmitted 192 Kbps, while each low priority station transmitted 160 Kbps. In this scenario, the *CW*-based differentiation scheme improved the performance when the number of high and low priority connections was increased to 10. For instance, Figure 8.6a shows that more than 80% of high priority packets met the QoS requirements for the time-sensitive applications. Figures 8.6b and 8.6c indicate that more than 70% of high priority packets maintained an excellent QoS with an average of 72.7%. This was at the cost of average QoS of low priority stations which had an average QoS equal to 36.7%. The reason for this is that: in an overloaded condition, it is critical for high priority traffic to get faster access to the medium than low priority traffic. The *CW*-based differentiation scheme assigns a larger *CW* size (i.e. a longer Backoff Interval (*BI*) on average) for low priority traffic and a smaller *CW* size for high priority traffic (i.e. a shorter *BI* on average). This enables high priority traffic to gain an earlier access to the medium before low priority traffic and therefore achieves higher QoS and smaller delays (see Table 8.2).

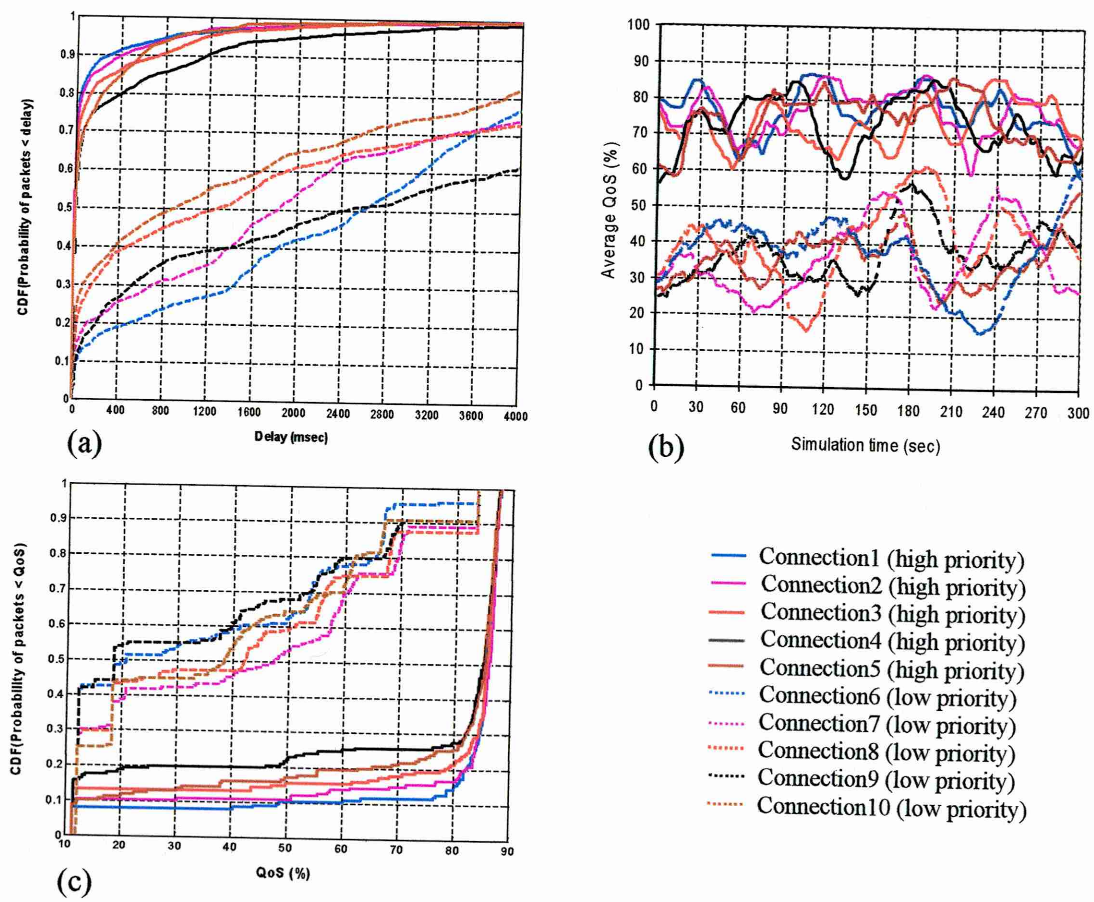



**Figure 8.6:** Adaptive *CW* differentiation for 10 connections (5 high priority stations transmitted 192 kbps and 5 low priority stations transmitted 160 kbps), (a) cumulative distribution of delay, (b) average QoS, and (c) cumulative distribution of QoS.

**Table 8.2:** QoS parameters values using the adaptive *CW* differentiation scheme for 10 connections.

| Bit rate / connection | Connection / priority | Average delay (msec) | Average jitter (msec) | Average throughput (Kbps) | Average MAC efficiency (%) | Average QoS (%) |
|---|---|---|---|---|---|---|
| 192 Kbps | Connection1 / high | 210.7 | 10.5 | 183.3 | 91.6 | 74.7 |
| | Connection 2 / high | 258.2 | 11.1 | 174.5 | 92.3 | 73.4 |
| | Connection 3/ high | 300.2 | 12.2 | 165.6 | 92.7 | 71.4 |
| | Connection 4 / high | 300.1 | 12.5 | 168.6 | 91.6 | 70.4 |
| | Connection 5 / high | 229.7 | 11.5 | 169.1 | 91.6 | 73.6 |
| 160 Kbps | Connection 6 / low | 2188.1 | 46.5 | 69 | 93.1 | 37 |
| | Connection 7/ low | 2242.4 | 51.2 | 88.1 | 93.8 | 39.5 |
| | Connection 8 / low | 3193.2 | 65.5 | 69.5 | 93.7 | 35.1 |
| | Connection 9 / low | 2299.4 | 50.4 | 94.8 | 93.1 | 36 |
| | Connection 10 / low | 2222.1 | 49.9 | 90.9 | 93 | 35.7 |

Although, the *CW*-based differentiation scheme revealed its ability for providing service differentiation, the trend for the delay and QoS curves of the two priorities showed some fluctuations and sometimes a narrow gap of differentiation. This was due to the overlap in the *CW* sizes among low and high priority classes. This in turn enabled low priority traffic to occasionally access the channel earlier than high priority traffic. This could be considered a drawback of *CW*-based differentiation scheme since the quality of differentiation depended on the amount of overlap between the *CW* values of the two classes. The next section presents another scheme which is based on the *DIFS* length to provide service differentiation in IEEE 802.11 DCF scheme.

#### 8.5.1.2 Adaptive Distributed Inter Frame Space for QoS Differentiation

In this section, another solution for providing service differentiation in IEEE 802.11 DCF protocol namely the *ADIFS* differentiation scheme is proposed.

In the *ADIFS* scheme, the *CW* size is updated according to the Binary Exponential Backoff (*BEB*) procedure as defined by IEEE 802.11 DCF. In IEEE 802.11 DCF the *ACK* frame is assigned a higher priority over data packets by having a shorter *IFS* known as short Inter Frame Space (*SIFS*), while data packets have a longer *IFS* known as *DIFS* (i.e. $SIFS < DIFS$). The same concept is applied for the *ADIFS* scheme where the *DIFS* length is dynamically adjusted for each priority based on the packet loss rate and *CRV* values as discussed in section 8.3.2.

The scenarios discussed in section 8.4 were also used with *ADIFS*-based differentiation scheme. In the first scenario, the two high priority stations transmitted at 192 Kbps each and the three low priority stations transmitted at high rate of 480 Kbps each.

As indicated in Table 8.3, the average delay for the high priority connections was less than 13 msec. This resulted in an excellent QoS for high priority connections with a mean value equal to 86%. This significant improvement in the QoS of high priority traffic was at the cost of low priority traffic. For instance, the third low priority connection had a poor QoS with an average of 22%. This was due to the longer waiting time prior to the transmission which led to high packet drops at the buffer. Although, the first and the second low priority connections had a good QoS, their average QoS was degraded by 14% compared to the average QoS obtained for the same connections when the *CW*-based differentiation scheme was used.

**Table 8.3:** QoS parameters values obtained using the adaptive *ADIFS* differentiation scheme.

| Bit rate / connection | Connection / priority | Average delay (msec) | Average jitter (msec) | Average throughput (Kbps) | Average MAC efficiency (%) | Average QoS (%) |
|---|---|---|---|---|---|---|
| 192 Kbps | Connection1 / high | 12.7 | 6.1 | 189.3 | 99.9 | 86.8 |
| | Connection2 / high | 10 | 6 | 188.5 | 99.9 | 87 |
| 480 Kbps | Connection3 / low | 1471.5 | 22.3 | 281.7 | 99.9 | 39.3 |
| | Connection4 / low | 709.6 | 9.9 | 304.9 | 99.9 | 44.3 |
| | Connection5 / low | 955.4 | 11.4 | 284.8 | 99.8 | 22 |

The *ADIFS*-based differentiation scheme was also evaluated when the number of high and low priority stations was increased. In this scenario, five high priority and five low priority stations contended to access the channel. Each high priority station transmitted 192 Kbps and each low priority station transmitted at 160 Kbps. As discussed in section 7.3 (see Chapter 7), the *ADIFS* scheme performed well when the number of contending stations was increased. For instance, in this scenario, the average delay of high priority stations was less than 28 msec in which QoS requirements in terms of delay for the time-sensitive applications could be met. When the *ADIFS* scheme was applied for service differentiation, a high priority station was required to wait for a shorter period so it could get access the channel earlier than a low priority station. At the time when a low priority station tried to access the medium, it found the channel busy and had to wait until the transmission of high priority packet was complete. Once the channel became idle, all stations commenced their backoff duration. Due to shorter lengths of *DIFS* for high priority stations, they waited for shorter time periods and started to decrease their backoff time earlier than low priority stations. This behaviour led to better performance for the QoS parameters and the average QoS of all connections as depicted in Figures 8.7a, 8.7b and 8.7c and shown in Table 8.4. The average QoS for the high priority traffic was 83.4% with fewer fluctuations and the average QoS for the low priority traffic was 37%.

The results obtained indicated that the *ADIFS*-based differentiation scheme provided a better service differentiation than the *CW*-based differentiation scheme. This was due to the impact of overlap that might occur in *CW* values between the two classes. *ADIFS*-based differentiation scheme also resulted in more stable QoS particularly for the high priority connections. However, the *ADIFS*-based differentiation scheme led to QoS degradation for some low priority connections as discussed earlier in this section. Therefore, to mitigate the drawbacks of the *CW*-based and *ADIFS*-based differentiation schemes when they are used individually, the next section presents the results obtained when the two parameters are combined together.

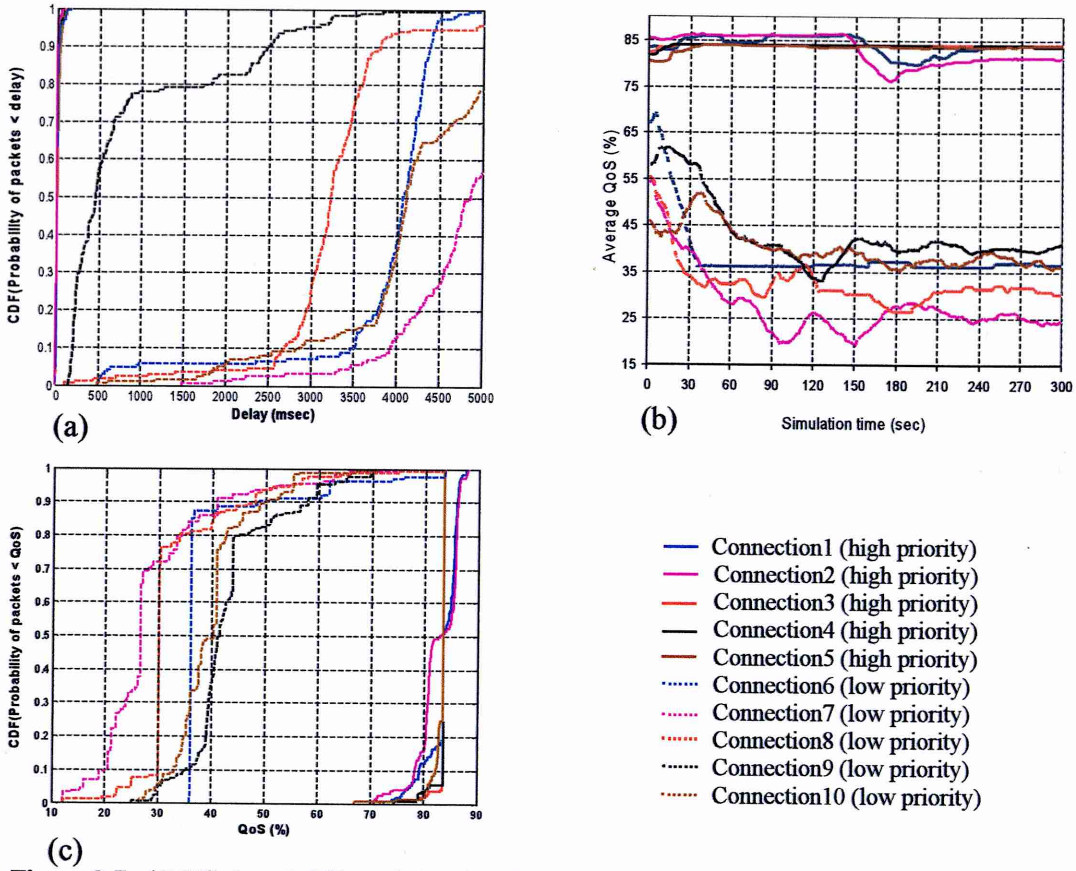



**Figure 8.7:** *ADIFS* –based differentiation for 10 connections (5 high and 5 low priority stations), (a) cumulative distribution of delay, (b) average QoS, and (c) cumulative distribution of QoS.

**Table 8.4:** QoS parameters values obtained using the *ADIFS* differentiation scheme for 10 connections.

| Bit rate / connection | Application type / priority | Average delay (msec) | Average jitter (msec) | Average throughput (Kbps) | Average MAC efficiency (%) | Average QoS (%) |
|---|---|---|---|---|---|---|
| 192 Kbps | connection 1 / high | 24.3 | 7.03 | 193.8 | 99.7 | 83.9 |
| | connection 2 / high | 26.7 | 6.5 | 188.7 | 99.8 | 82.9 |
| | connection 3 / high | 25.4 | 8.2 | 174 | 99.9 | 83.6 |
| | connection 4 / high | 18.3 | 7.2 | 181.6 | 99.8 | 83.6 |
| | connection 5 / high | 27.8 | 7.3 | 178.5 | 99.7 | 83.3 |
| 160 Kbps | connection 6 / low | 3859.4 | 61.4 | 86.9 | 99.6 | 39.5 |
| | connection 7 / low | 5009.8 | 60.1 | 71.4 | 99.9 | 29.1 |
| | connection 8 / low | 3263.9 | 46.5 | 80.4 | 99.7 | 33.4 |
| | connection 9 / low | 860 | 25.2 | 123 | 99.5 | 43.3 |
| | connection 10 / low | 4145.4 | 60.1 | 97 | 99.5 | 40 |

#### 8.5.1.3 Adaptive Scheme for Quality of Service Differentiation

In this section, the adjusted *CW* and *DIFS* parameters were combined in order to provide improved service differentiations. Three scenarios were discussed in this section. In the first scenario, two high priority and three low priority connections transmitted *CBR* traffic at 192 Kbps and 480 Kbps for each high and low priority connection, respectively. In the second scenario, 10 connections (5 high priorities and 5 low priorities) were considered. Each high priority connection transmitted at 192 Kbps and each low priority connection transmitted at 160 Kbps. The average QoS and QoS parameters were discussed as a function of offered load in the third scenario when 10 connections were active in the network (5 high and 5 low priorities).

Figures 8.8a and 8.8b show the average delay and average QoS for both classes. The average delay of high priority traffic was less than 20 msec which led to an excellent QoS with a mean value equal to 86% for high priority connections. These values (i.e. delay and QoS) remained stable along the simulation time with small values of standard deviation summarised in Table 8.5. The distribution of delay and QoS of high and low priority classes are shown in Figure 8.8c and 8.8d. The two classes were clearly differentiated. Up to 99% of high priority packets had a delay of less than 200 msec and more than 85% of high priority packets had an excellent QoS level. Low priority packets also maintained a good QoS level for all connections with an average of 42%.

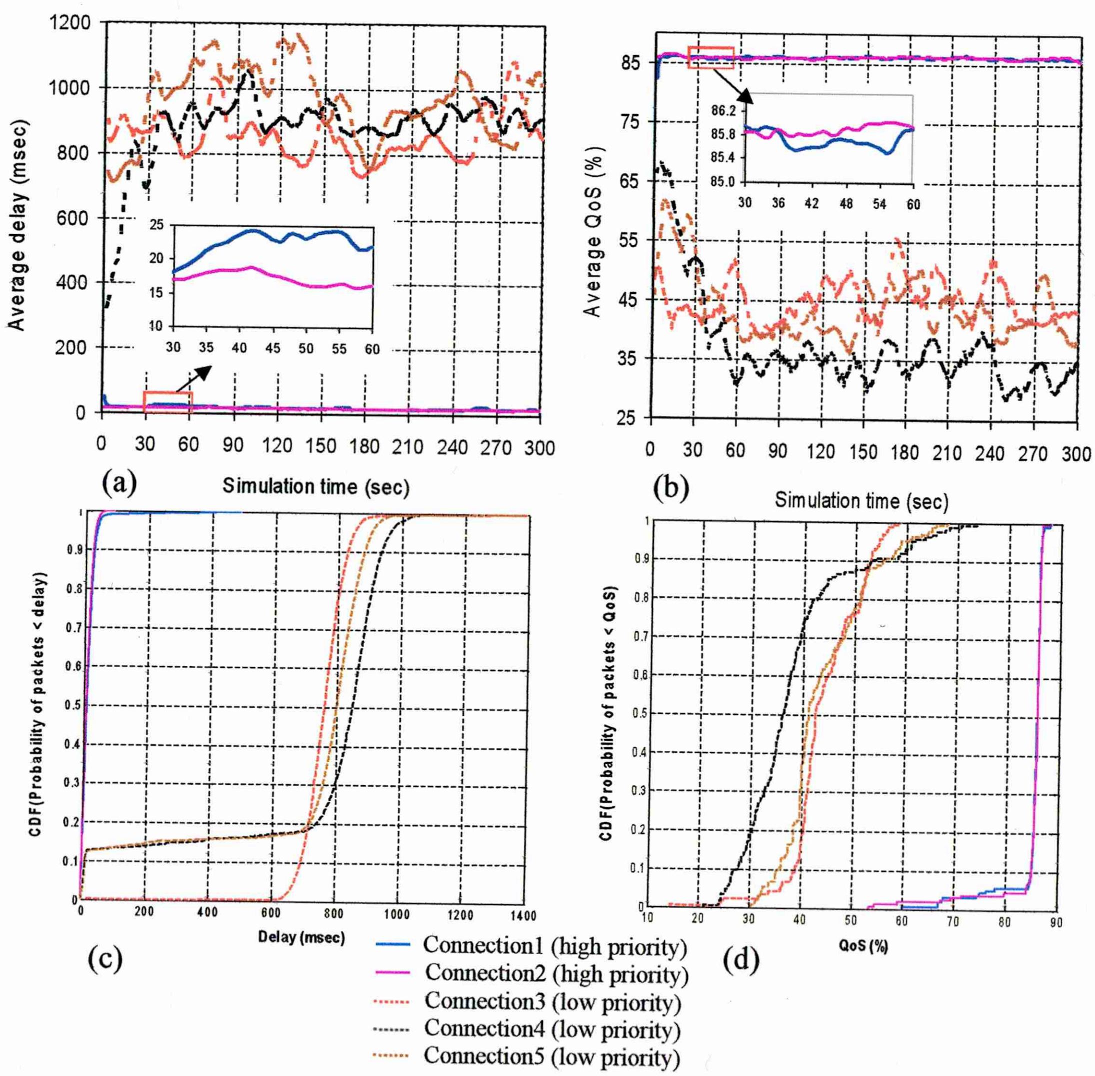



**Figure 8.8:** Adaptive differentiation scheme for 5 connections (2 high and 3 low priority stations), (a) average delay, (b) average QoS, (d) cumulative distribution of delay (c) cumulative distribution of QoS.

**Table 8.5:** QoS parameters values obtained using the adaptive differentiation scheme.

| Bit rate / application | Application type / priority | Statistic measure | Average delay (msec) | Average jitter (msec) | Average throughput (Kbps) | Average MAC efficiency (%) | Average QoS (%) |
|---|---|---|---|---|---|---|---|
| 192 Kbps | connection1 / high | Mean | 18.3 | 7.7 | 192 | 99.4 | 86.0 |
| | | Stdev | 3.7 | 1.2 | 1.0 | 1.7 | 0.34 |
| | connection 2 / high | Mean | 15.8 | 7.7 | 191.8 | 99.3 | 86.0 |
| | | Stdev | 1.1 | 0.8 | 2.3 | 2.3 | 0.2 |
| 480 Kbps | connection / low | Mean | 853.2 | 11.9 | 348.3 | 99.3 | 45.0 |
| | | Stdev | 70.5 | 6.1 | 38.1 | 1.01 | 3.5 |
| | connection / low | Mean | 881.1 | 11.8 | 302.0 | 98.8 | 37.1 |
| | | Stdev | 111.2 | 7.4 | 8.2 | 1.4 | 8.3 |
| | connection / low | Mean | 959.3 | 11.5 | 343.4 | 99.7 | 43.6 |
| | | Stdev | 113.7 | 6.0 | 5.3 | 1.2 | 5.4 |

The performance of the adaptive differentiation scheme for the second scenario is represented by Figures 8.9a, 8.9b, 8.9c and 8.9d. Figure 8.9a shows that average delay of high priority packets was relatively small compared to average delay of low priority packets. These small values of delay with other QoS parameters for time-sensitive applications resulted in 84% average QoS for high priority connections as shown in Figure 8.9b and provided in Table 8.6.

Figures 8.9c and 8.9d confirmed that high and low priorities were clearly differentiated without harming each other. High priority traffic was completely protected from the impact of low priority traffic and it achieved 84% as an average QoS. Low priority traffic also had a good QoS level with an average of 53%. In the adaptive differentiation scheme, when packet loss rate of high priority traffic exceeded the packet loss rate threshold (i.e. $l[N_i] > l\_ths[N_i]$), the high priority stations reduced their *CW* and *DIFS* values in order to maintain their packet loss rates below the specified threshold ($l\_ths[N_i]$). The small values of *CW* and *DIFS* of high priority stations resulted in more collisions, and subsequently the values of *CRV* of low priority stations increased. In order to eliminate the probability of collision the *CW* and *DIFS* values of low priority stations were increased. In a heavily loaded network, this behaviour might lead to a starvation for the low priority traffic, therefore the adaptive differentiation scheme examines whether the *CW* and *DIFS* of low priority traffic experience large values after collision. If so, the adaptive differentiation scheme activates the starvation routine (see section 8.3.3) in order to give low priority stations more opportunities to access the medium.

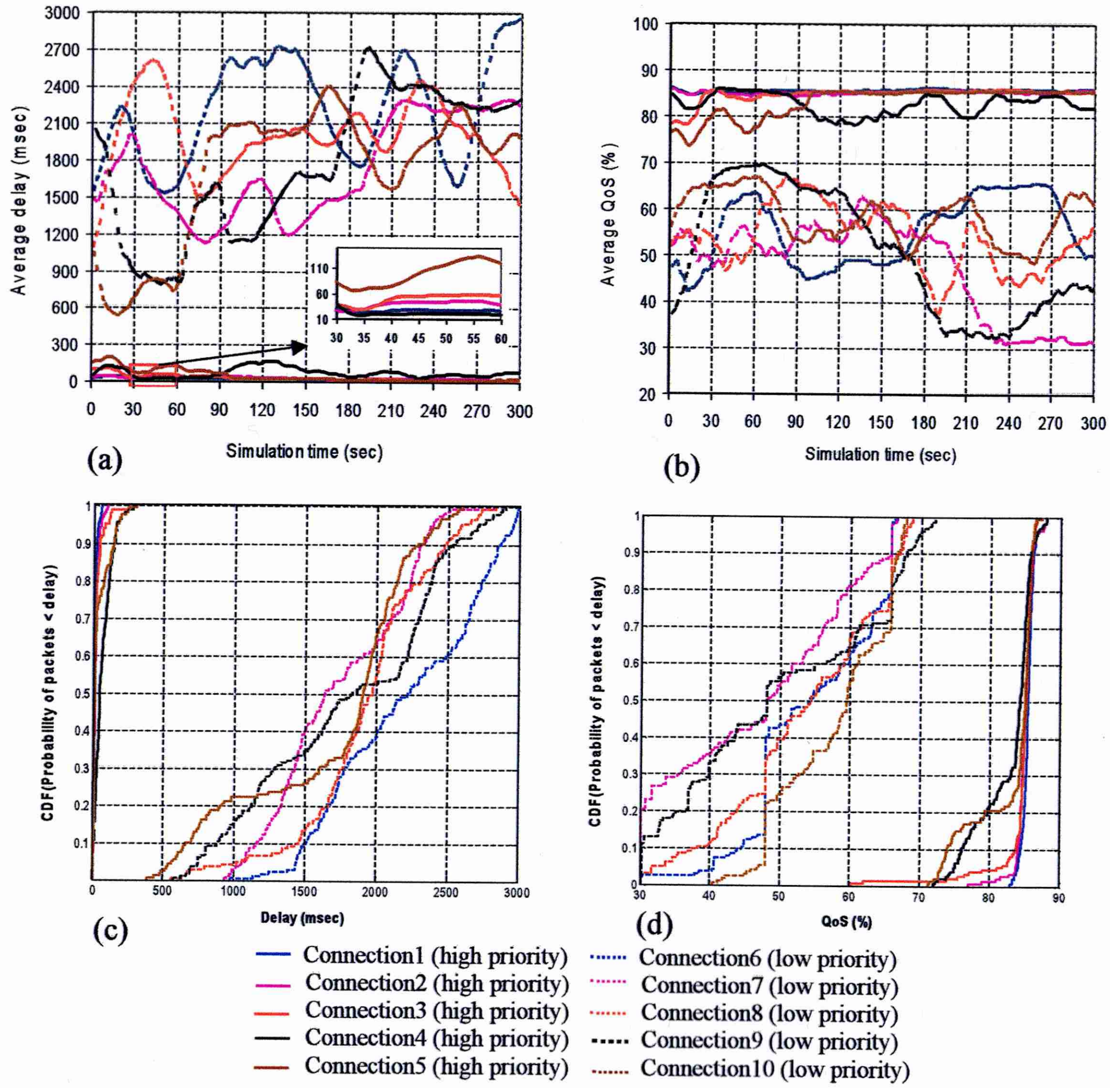



**Figure 8.9:** Adaptive differentiation scheme for 10 connections (5 high 5 low priority stations, (a) average delay, (b) average QoS, (d) cumulative distribution of delay (c) cumulative distribution of QoS.

**Table 8.6:** QoS parameters values obtained using the adaptive differentiation scheme for 10 connections.

| Bit rate / connection | Connection / priority | Average delay (msec) | Average jitter (msec) | Average throughput (Kbps) | Average MAC efficiency (%) | Average QoS (%) |
|---|---|---|---|---|---|---|
| 192 Kbps | **connection 1 / high** | 22.6 | 8.1 | 192.0 | 99.3 | 85.6 |
| | **connection 2 / high** | 28.6 | 8.4 | 191.0 | 99.6 | 85.2 |
| | **connection 3/ high** | 32.9 | 8.6 | 187.2 | 99.4 | 84.8 |
| | **connection 4 / high** | 73.6 | 8.9 | 184.6 | 99.7 | 82.7 |
| | **connection 5 / high** | 49.4 | 8.8 | 180.2 | 99.6 | 83.2 |
| 160 Kbps | **connection 6 / low** | 2209.6 | 38.3 | 107.5 | 98.7 | 55.1 |
| | **connection 7 / low** | 1733.8 | 32.2 | 82.7 | 97.8 | 47.3 |
| | **connection 8 / low** | 1991.5 | 39.6 | 95.9 | 97.4 | 53.7 |
| | **connection 9 / low** | 1761.5 | 32.8 | 91.4 | 99.04 | 50.6 |
| | **connection 10/ low** | 1706.4 | 30.6 | 109.8 | 98.9 | 58.3 |

The results indicate that the adaptive differentiation scheme (i.e. combining *CW* and *DIFS*) could provide service differentiation. High priority traffic had small values of delay and jitter, absolute throughput and an excellent QoS level. Low priority traffic also had a good QoS level and could avoid starvation.

The performance of the adaptive differentiation scheme was also evaluated using scenario3. Here, the offered load was increased from 200 Kbps to 2 Mbps (i.e. from light load conditions to heavily loaded conditions) and the average delay and average QoS were measured for each value.

The average delay and average QoS as a function of offered load are shown in Figures 8.10a and 8.10b, respectively. When the offered load was equal to 1.4 Mbps, the adaptive differentiation scheme maintained small values of delay and excellent QoS levels for all connections with a minor differentiation between the two classes. Hereafter, the average delay of the low priority traffic started to increase sharply as the offered load increased, while the average delay of high priority traffic slightly increased and remained within the QoS requirement for time-sensitive applications. Average QoS was also affected by the increase of the offered load. When the offered load was increased above 1.4 Mbps, the average QoS of low priority packets started to degrade sharply from an excellent level at 1.4 Mbps to a good level at 1.8 Mbps and then to a poor level at 2.0 Mbps as shown in Figure 8.10b. High priority connections maintained excellent QoS levels. However, minor degradation was observed at 1.8 and 2.0 Mbps of offered load as shown in Figure 8.10b.

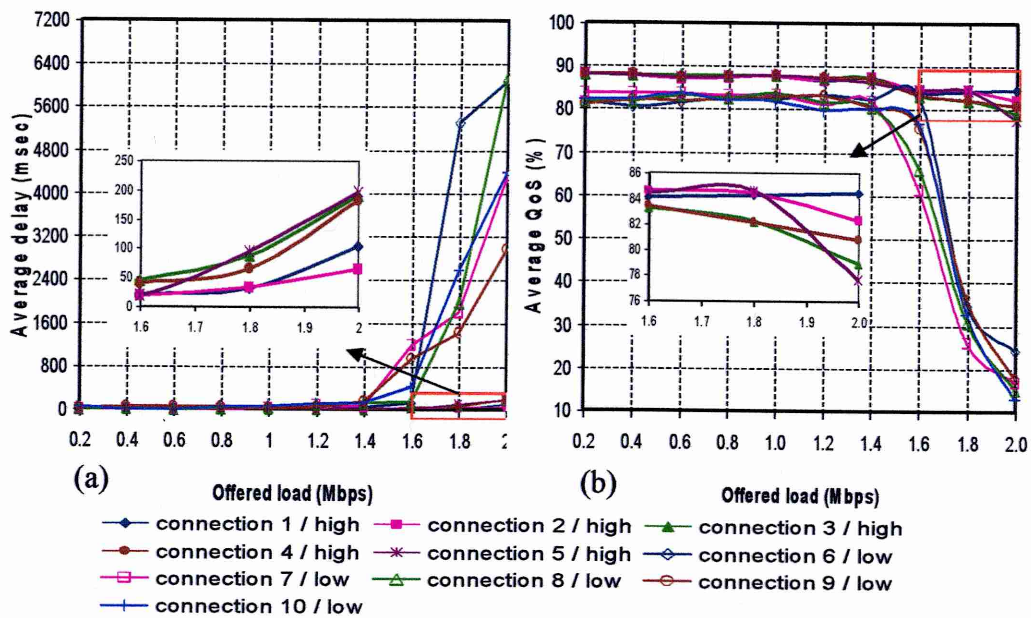



**Figure 8.10:** Performance of adaptive differentiations scheme as a function of offered load. (a) average delay, (b) average QoS.

It can be seen that the adaptive differentiation scheme was capable of preserving a narrow gap of differentiation between the two classes at light and medium traffic loads. For light and medium load conditions, packet loss rate and *CRV* values were very small indicating that the network was working in normal conditions. Consequently, high priority stations slightly increased their *CW* and *DIFS* values in order to offer low priority stations higher accessibility to the medium. This can be achieved as follows: when high priority stations increase their *CW* and *DIFS* values, the probability of collisions starts to decrease and then the *CRV* values of low priority stations also decrease which force low priority stations to decrease their *CW* and *DIFS* values.

For the heavily loaded case, the adaptive differentiation scheme was also able to widen the differentiation gap between traffic classes. This can be explained as follows: when the network becomes overloaded, the packet loss rate of high priority traffic exceeds the packet loss rate threshold ($l\_ths[N_i]$). Thus, high priority stations avoid excessive packet loss rate by decreasing their *CW* and *DIFS* values. This certainly increases the probability of collisions. Concurrently, low priority stations observe the increase in the number of collisions through their *CRV* values, and then increase their *CW* and *ADIFS* values to reduce the probability of collisions.

After evaluating each individual differentiation scheme (*CW* differentiation, *ADIFS* differentiation and the adaptive differentiation), their performances are compared with the performance of the IEEE 802.11 DCF scheme. In the standard IEEE 802.11 DCF, the default settings that were defined by the standard were used throughout the scenarios discussed in section 8.4 (see Table 4.1 in Chapter 4).

It is well known that the standard IEEE 802.11 DCF scheme only supports best-effort service without any service differentiation. This was obvious through the results shown in Figure E.1 (see Appendix E). It can be seen that both high and low priority traffic had large values of delay, poor QoS, and no service differentiation for 5 and 10 connections.

The performance of the IEEE 802.11 DCF scheme was also investigated as a function of offered load. High priority stations had small values of average delay and an excellent QoS level up to 1.2 Mbps of offered load. After this, a considerable increase in average delay for high priority traffic was observed as the offered load was increased as shown in Figure 8.11a. Low priority stations also had small values of average delay at offered load equal to 1.2 Mbps and then a dramatic increase in the average delay was observed when the offered load was increased above 1.2 Mbps. Low priority stations had an excellent QoS level at light load conditions and then started to degrade to a good

QoS and ultimately to a poor QoS level at heavily loaded conditions as shown in Figure 8.11b.

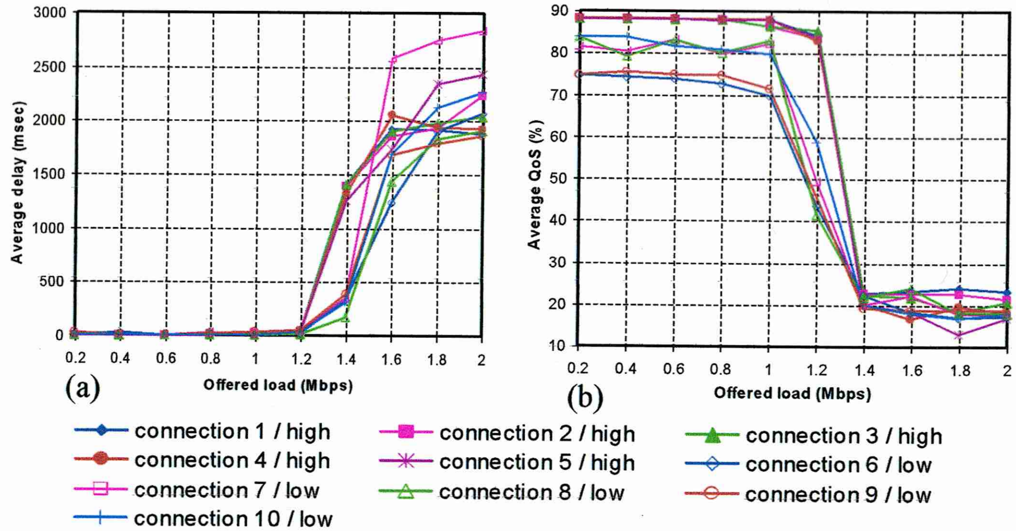



**Figure 8.11:** Performance of IEEE 802.11 DCF as a function of offered load, (a) average delay, and (b) average QoS.

The results indicate that the adaptive differentiation schemes provided a better performance and a clearer differentiation as compared to the IEEE 802.11 DCF scheme.

Having discussed the performance of *CW* differentiation, *ADIFS* differentiation, and adaptive differentiation schemes for single-hop networks, it is essential to evaluate the performance of the adaptive differentiation scheme when it is combined with the queue status monitoring scheme for multi-hop networks.

### 8.5.2 Service Differentiation in Multi-hop Networks

The adaptive differentiation schemes confirmed their capability of providing service differentiation and improving the network performance in single-hop networks. This section highlights the main findings of the queue status monitoring approach combined with the discussed adaptive differentiation scheme to provide QoS in multi-hop networks. A comparison with the standard IEEE 802.11 DCF scheme is also considered.

This section discusses the second set of scenarios demonstrated in section 8.4 using the network topology shown in Figure 4.2c (see Chapter 4). In this scenario the queue size was set to 50 packets and afterwards it was varied in the range of (10 to 50 packets). The variation of the queue size was considered in order to evaluate the performance of the adaptive differentiation and queue status monitoring scheme for different queue sizes and also to demonstrate the impact of queue size on the network performance.

Figures 8.12a and 8.12b show the distribution of delay for the adaptive and IEEE 802.11 DCF schemes in multi-hop network, respectively. In Figure 8.12a, high priority connections of the adaptive system had an average delay less than 400 msec; whereas low priority packets had delays higher than 400 msec. With the IEEE 802.11 DCF scheme, more than 80% of high and low priority packets had an average delay greater than 1000 msec and no service differentiation was observed as shown in Figure 8.12b.

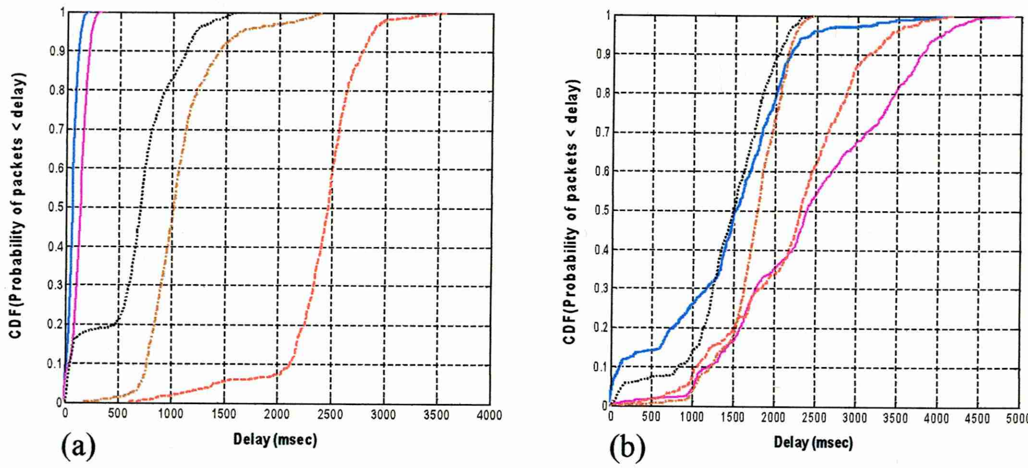



**Figure 8.12:** Cumulative distribution of delay in multi-hop networks (queue size equal 50 packets), (a) adaptive differentiation and queue status monitoring scheme, (b) IEEE 802.11 DCF scheme.

Average QoS for the high priority traffic in both schemes (i.e. adaptive differentiation and IEEE 802.11 DCF schemes in multi-hop networks) is shown in Figure 8.13.

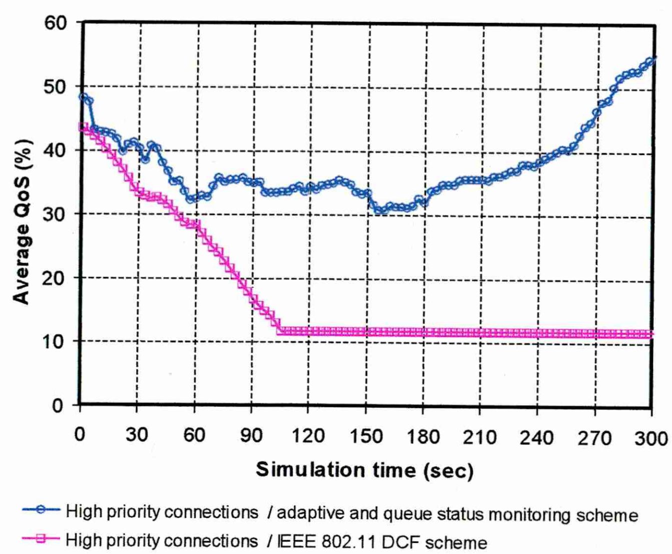



**Figure 8.13:** Average QoS for high priority traffic in the adaptive differentiation and queue status monitoring scheme and in IEEE 802.11 DCF scheme (queue size equals 50 packets).

IEEE 802.11 DCF scheme showed a poor average QoS. Initially, IEEE 802.11 DCF scheme resulted in a good QoS until 30 seconds of the simulation had elapsed. Afterwards a considerable degradation was observed to the end of simulation. This was due to the lack of an adaptive technique in the IEEE 802.11 DCF scheme to deal with variation in network conditions such as congestion and collisions when packets traverse over multiple hops. In the adaptive differentiation and queue status monitoring scheme, a good QoS level was maintained throughout the simulation time with an average QoS equal to 38%. This value was 53% higher than the average QoS that was obtained when the standard IEEE 802.11 DCF scheme was used (i.e. average QoS of the IEEE 802.11 DCF scheme equal to 17.6%).

In the next experiment, the queue size was varied in the range of (10 to 50 packets). The performance of the adaptive differentiation and queue status monitoring scheme in multi-hop networks was evaluated and was also compared with IEEE 802.11 DCF. It can be observed that the average delay of high priority packets of the adaptive scheme was kept less than 400 msec for all queue sizes as shown in Figure 8.14a. These small values of average delay resulted in a good QoS for time-sensitive applications in multi-hop networks. Conversely, average delay for the low priority packets in the adaptive scheme was increased with an increase in the queue size (i.e. low priority packets experienced long defers at the *IFQ*) and it had its maximum value at queue size equal to 50. It was also observed that average delay of high priority packets was not significantly affected by the variation of the queue size, since the adaptive scheme provided them with better treatment than low priority packets. In contrast, IEEE 802.11 DCF neither provided service differentiation nor met the QoS requirements for high priority packets. Further, its average delay values were significantly affected by the variation of the queue size. It can also be seen that average delay values of low priority packets of the adaptive scheme were smaller than the average delay of high priority packets when IEEE 802.11 DCF was used. This confirmed that the adaptive scheme was also capable of improving the performance of low priority traffic besides its ability of providing service differentiation.

In Figure 8.14b, the number of successfully received packets as a function of queue size is plotted. It can be observed that the adaptive scheme successfully received more packets than the IEEE 802.11 DCF scheme. For instance, at queue size equal to 50 packets, more than 77.4% of high and low priority packets were successfully received when the adaptive scheme was used. When the IEEE 802.11 DCF was employed only 44.9% of high and low priority packets were successfully received. Thus, with respect to the number of successfully received packets, the adaptive scheme outperformed the

IEEE 802.11 DCF scheme by 39.4%. This was due to the large number of packet drops at the buffer and due to the large number of collisions when the IEEE 802.11 DCF scheme was used. More than 48.4% of high and low priority packets were dropped at the buffer and more than 14.6% of collisions were observed when IEEE 802.11 DCF was used as shown in Figures 8.14c and 8.14d.

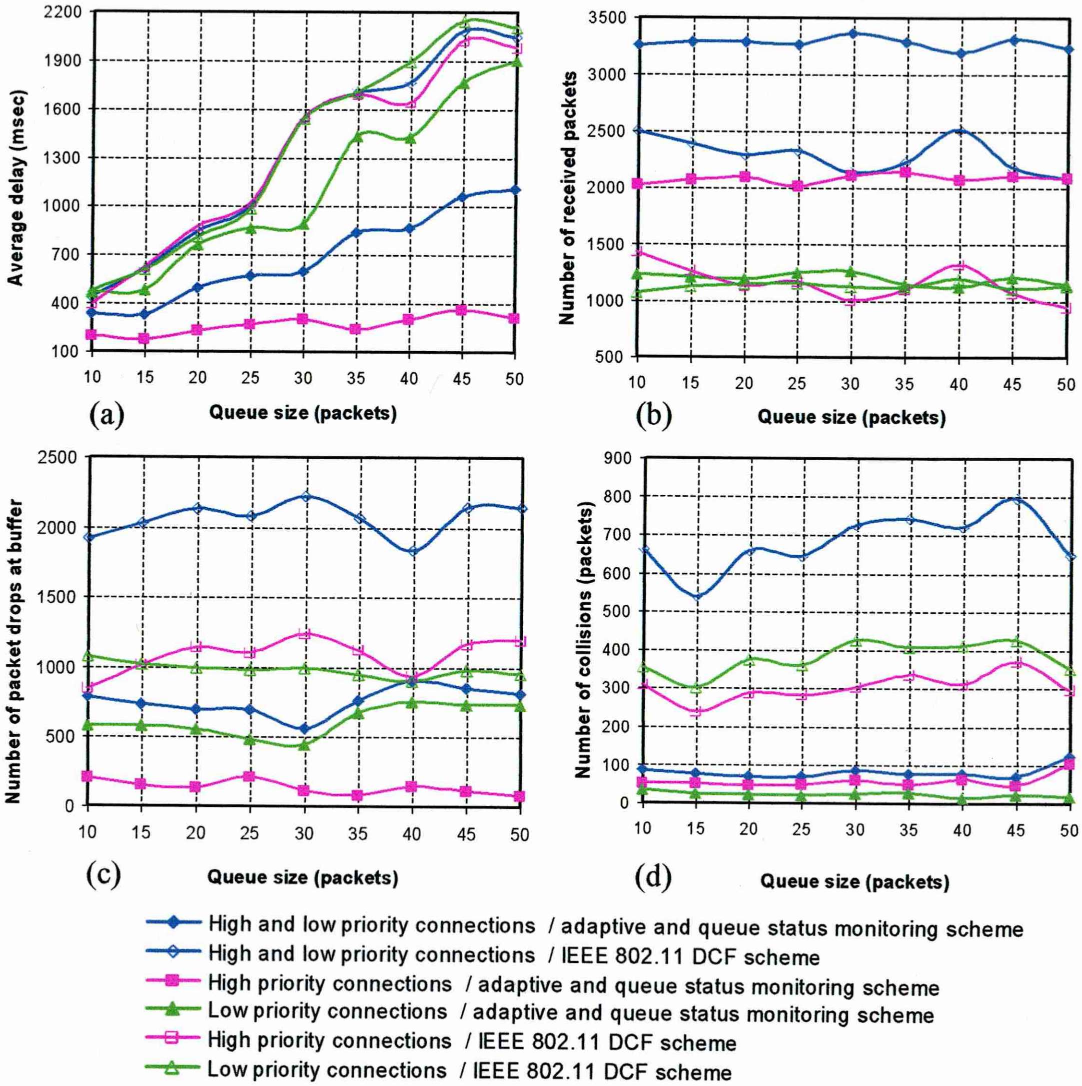



**Figure 8.14:** Performance evaluation of the adaptive differentiation and queue status monitoring scheme vs. the standard IEEE 802.11 DCF as a function of queue size, (a) average delay, (b) number of successfully received packets, (c) number of packets drop at the buffer, and (d) number of collisions.

The cause of high packet drops at the buffer and the increase in the number of collisions in the standard IEEE 802.11 DCF was due to the lack of an adaptive scheme. In the standard IEEE 802.11 DCF scheme, the intermediate station was not capable of differentiating between traffic classes, and it was not able to adjust its *CW* and *DIFS* parameters to adapt the network variation. Basically, at heavily loaded conditions, the intermediate station drops all arriving packets regardless of their priorities. Furthermore, the *DIFS* length was kept fixed for all stations and the *CW* size was doubled after unsuccessful transmission and reset to $CW_{min}$ after successful transmission regardless of the traffic class and the past network conditions.

When the adaptive scheme was used, smaller number of packets was dropped at the buffer (only 700 packets were dropped, 100 high priority packets and 600 low priority packets) and a smaller number of collisions were observed (i.e. 90 collisions in both classes) as shown in Figures 8.14c and 8.14d, respectively. However, each intermediate and data source station in the adaptive scheme was capable of dynamically adjusting its *CW* and *DIFS* values for each traffic class according to the collision rate (*CR*), *CRV*, and packet loss rate values. Furthermore, the intermediate station in the adaptive scheme was able to execute two polices either drops low priority packets or sends a feedback control message to the intended source according to the queue status ratio as discussed earlier in section 8.3.4. So, each low priority station in the adaptive scheme adjusted its transmission rate upon receiving a feedback control from an intermediate station. Therefore, the behaviour of the adaptive scheme resulted in a good QoS for high priority traffic which was not achieved when the standard IEEE 802.11 DCF scheme was employed as shown in Figure 8.15 and as summarised in Table 8.7.

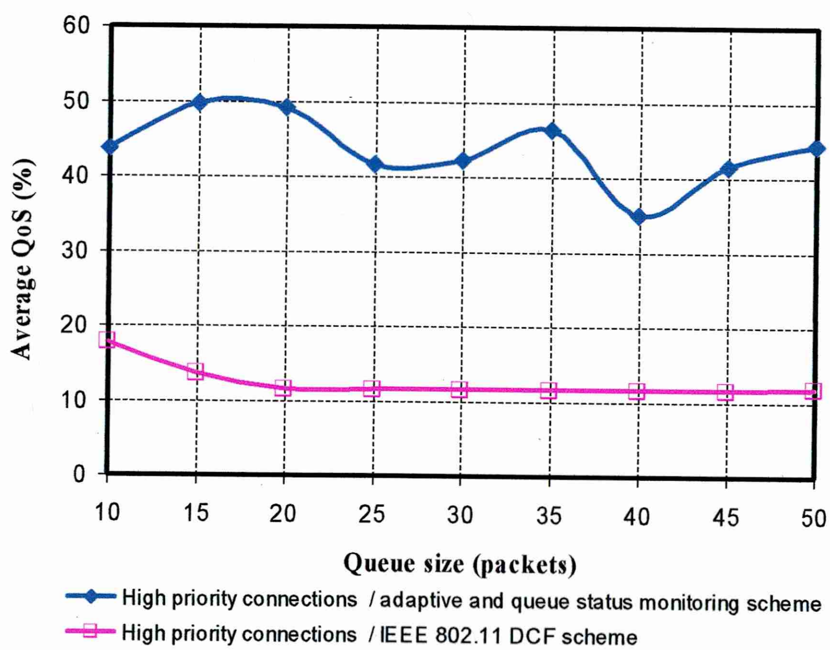



**Figure 8.15:** Average QoS for the high priority traffic as a function of queue size.

**Table 8.7:** Adaptive differentiation and queue status monitoring scheme vs. IEEE 802.11 DCF scheme.

| Parameter | Queue Size | Adaptive differentiation scheme | | | IEEE 802.11 DCF scheme | | |
|---|---|---|---|---|---|---|---|
| | | High and Low Priority | High Priority | Low priority | High and Low Priority | High Priority | Low Priority |
| Average Delay (msec) | 10 | 332.6 | 199.9 | 465.3 | 437.6 | 396.7 | 478.5 |
| | 15 | 330.7 | 179 | 482.3 | 615.2 | 624.3 | 606 |
| | 20 | 494.9 | 228.7 | 761 | 844.8 | 877 | 812.5 |
| | 25 | 569.1 | 271.9 | 866.3 | 1007.6 | 1026.7 | 988.5 |
| | 30 | 598.9 | 303.2 | 894.7 | 1544.4 | 1547.2 | 1541.5 |
| | 35 | 837.8 | 242.6 | 1433 | 1701.5 | 1693.9 | 1709 |
| | 40 | 866.2 | 304.4 | 1428 | 1768.6 | 1641.8 | 1895.5 |
| | 45 | 1064.7 | 360 | 1769.3 | 2085.6 | 2026.7 | 2144.5 |
| | 50 | 1109.5 | 311.7 | 1907.3 | 2045.7 | 1982.7 | 2108.7 |
| No. of received packets | 10 | 3259 | 2023 | 1236 | 2502 | 1433 | 1069 |
| | 15 | 3286 | 2073 | 1213 | 2387 | 1264 | 1124 |
| | 20 | 3290 | 2094 | 1196 | 2287 | 1135 | 1152 |
| | 25 | 3264 | 2016 | 1248 | 2325 | 1170 | 1156 |
| | 30 | 3371 | 2106 | 1266 | 2133 | 1011 | 1122 |
| | 35 | 3292 | 2141 | 1150 | 2226 | 1098 | 1128 |
| | 40 | 3194 | 2068 | 1126 | 2518 | 1320 | 1198 |
| | 45 | 3311 | 2098 | 1213 | 2177 | 1070 | 1107 |
| | 50 | 3233 | 2087 | 1146 | 2076 | 945 | 1131 |
| No. of packet drops at buffer (packet) | 10 | 785 | 203 | 583 | 1920 | 843 | 1078 |
| | 15 | 732 | 155 | 578 | 2031 | 1011 | 1020 |
| | 20 | 693 | 138 | 555 | 2133 | 1139 | 995 |
| | 25 | 695 | 214 | 481 | 2081 | 1105 | 977 |
| | 30 | 558 | 114 | 445 | 2226 | 1234 | 992 |
| | 35 | 761 | 86 | 676 | 2068 | 1120 | 948 |
| | 40 | 904 | 146 | 757 | 1836 | 935 | 901 |
| | 45 | 851 | 113 | 737 | 2145 | 1164 | 982 |
| | 50 | 814 | 81 | 733 | 2146 | 1192 | 954 |
| No. of collisions (packet) | 10 | 86 | 52 | 34 | 661 | 307 | 354 |
| | 15 | 76 | 50 | 26 | 540 | 239 | 301 |
| | 20 | 69 | 46 | 23 | 659 | 286 | 374 |
| | 25 | 69 | 48 | 20 | 645 | 283 | 362 |
| | 30 | 85 | 59 | 26 | 727 | 302 | 425 |
| | 35 | 76 | 49 | 27 | 742 | 335 | 407 |
| | 40 | 77 | 62 | 15 | 722 | 311 | 412 |
| | 45 | 69 | 47 | 22 | 797 | 371 | 426 |
| | 50 | 123 | 104 | 18 | 648 | 297 | 351 |

## 8.6 Summary

The main objective described in this chapter is to develop new Quality of Service (QoS) adaptation schemes for single and multi-hop networks. These are: adaptive Contention Window (*CW*) differentiation, Adaptive Distributed Inter Frame Space (*ADIFS*) differentiation, and adaptive service differentiation scheme (combined *CW* differentiation and *DIFS* differentiation schemes), and adaptive service differentiation with queue status monitoring schemes. In this respect, this chapter first reviewed the previous efforts in section 8.2. A full description of the adaptive service differentiation schemes is outlined in section 8.3. In section 8.4, the simulation model and the selected scenarios to validate the performance of the adaptive QoS differentiation schemes and to compare them with the standard IEEE 802.11 DCF were outlined. A detailed description of the main findings was given in section 8.5.

The *CW*-based differentiation scheme depended on dynamic adjustment of the *CW* size after successful and unsuccessful transmission for each traffic class, taking into account the current and past network information. The *DIFS*-based differentiation scheme depended on dynamic adjustment of *DIFS* length of each traffic class by providing a high priority class with shorter lengths of *DIFS*. The adaptive service differentiation scheme was constructed from a combination of *CW* differentiation and *DIFS* differentiation schemes.

In this chapter, the queue status ratio was considered as a performance metric in the queue status monitoring technique in order to adjust the transmission rate of low priority stations at congestion conditions. Combining the queue status monitoring scheme was the adaptive service differentiation scheme, QoS differentiation in multi-hop networks was provided. The results revealed that the adaptive service differentiation scheme were capable of providing service differentiation and improving the network performance in single and multi-hop networks.

The ratio based and *CRV* scheme discussed in Chapter 7 and the adaptive QoS differentiation schemes outlined in Chapter 8 were based on dynamic adjustment of MAC protocol transmission parameters according to current and previous network conditions. Incorporating the future state of the network through a prediction technique is a valuable tool. In the following Chapter, the use of an autoregressive model as a prediction mechanism to improve the network performance and to provide service differentiation will be explored.

# *CHAPTER 9*

# QoS Provision Using Autoregressive Modelling

## 9.1 Introduction

In IEEE 802.11 DCF MAC protocol, there are many examples where knowledge of the future values can be useful in improving the network performance (IEEE, 1999). For instance, if the Collision Rate (*CR*) and Collision Rate Variation (*CRV*) values can be predicted in advance, protection from excessive collisions can be provided by adjusting some of the MAC protocol transmission parameters such as the *CW* and *DIFS*. As a result, significant improvements in the network performance can be realised.

In multi-hop networks, part of existing stations operates as routers that forward data packets to their corresponding destinations. Congestion in these intermediate stations causes performance degradation particularly for time-sensitive applications. Therefore, monitoring the queue status of these intermediate stations and determining the queue status ratio reduces the congestion. This can be achieved by either dropping low priority packets or by adjusting the transmission rate of low priority packets as discussed in chapter 8 (see section 8.3.4).

If the queue status ratio can be predicted in advance, precautions such as adaptive adjustments of the transmission rate or discarding arriving data packets based on the predicted value can provide significant performance improvements. Several studies have addressed the use of prediction in the wireless domain. Most of the studies have focused in the prediction of the route of mobile nodes such as the work presented in (Lee et *al.*, 1999), (Hongxia and Hughes, 2003) and (Doss et *al.*, 2004). Other studies have focused on prediction of channel fading in wireless networks. These can be found in (Ekman, 2001) and (Svantesson and Swindlehurst, 2006). Linear and nonlinear models were used for power prediction such as (Gao et *al.*, 1996) and (Choe et *al.*, 1999). Other studies investigated the prediction of node location such as (Shen et *al.*, 2005). In (Kim and Noble, 2001), the stability of Exponentially Weighted Moving Average (*EWMA*) models for estimation of available channel capacity in wireless networks was studied.

To our knowledge no study has been carried out to use regression models to forecast the *CR*, *CRV* and queue status ratio; and subsequently adjusting the *CW*, *DIFS*, and transmission rate of the participant stations in wireless networks. This chapter shows that the developed prediction models are able to improve the network performance and to provide service differentiation by using probabilistic descriptions of the medium characteristics, such as collisions, collision variation, packet loss, and queue occupancy.

The main goal of this chapter is to develop online prediction models for the IEEE 802.11 DCF scheme to appropriately adjust the protocol transmission parameters such as *CW*, *DIFS* and transmission rate according to an accurate prediction of the future values of collisions and queue status occupancy. The methods used to develop the prediction models for the IEEE 802.11 DCF scheme can be applied to any version of the IEEE 802.11 wireless protocol. Such models are useful in improving the protocol performance by predicting some of the metrics that significantly affect the performance of the protocol such as collisions and network congestion. In this chapter, the impact of collisions and congestion will be statistically analysed and used to build regression models to forecast their future values in order to adjust the main MAC protocol transmission parameters and eventually providing QoS over the IEEE 802.11 DCF.

This chapter is organised as follows: sections 9.2 and 9.3 provide an overview about linear regression and its assumptions. The implementation of autoregressive models are presented in section 9.4. Section 9.5 outlines the methodology of the proposed models. Section 9.6 introduces the simulation model. Simulation results are demonstrated in section 9.7. The final section introduces the summary of this chapter.

## 9.2 Linear Regression

Linear regression tries to model the relationship between two variables by fitting a linear line to the observed data. One variable is considered to be explanatory or independent variable, and the other is considered to be a response or a dependent variable. In order to fit the linear model to the observed data, there should be a relationship between the variables of interest. Otherwise, fitting the linear regression model to the data will not provide any valuable model (Stat, 2006).

## 9.3 Linear Regression Assumptions

The models illustrated in this chapter were developed using linear regression. This type of regression is commonly used in data analysis. A detailed explanation of it can be found in most texts on statistics or regression analysis including (Chatterjee and Price, 1991), (Jain, 1991), (Allen, 1990), and (Draper and Smith, 1981). The use of linear regression imposes some assumptions which must be met in order to have a valid regression model (Jain, 1991) and (Baldwin et *al.*, 1999). These assumptions are summarised as follows: (i) the relationship between the dependent (i.e., response) variable such as collision ratio and the independent variables (i.e., predictor or explanatory) variables such as the previous values of collision ratio are linear, (ii) the independent variables are not stochastic and are specified without error, (iii) model errors are statistically independent, and (iv) errors are normally distributed.

The regression models presented in this chapter were examined with respect to the aforementioned assumptions. In some cases, the relationship between the dependent variable and the independent variables was nonlinear. In these cases, an appropriate transformation of the independent variable was performed in order to make the response as linear as possible.

To ensure that model errors are statistically independent, a scatter plot of the predicted variable versus the residuals (i.e., the difference between the actual and the predicted values or errors) should not have noticeable trends. Figures 9.1a and 9.1b show how the proposed *AR* models meet the requirements of assumptions 3 and 4, respectively. Figure 9.1a shows the scatter plot for the residuals as a function of the predicted collision ratio. As shown in Figure 9.1a there are no visible trends between the residuals and the predicted collision ratio, this confirms that the third assumption is met. To satisfy that errors are normally distributed, Figure 9.1b shows the frequency distribution of the residual. The distribution of the residuals is approximately normal which implies that the forth assumption is also met.

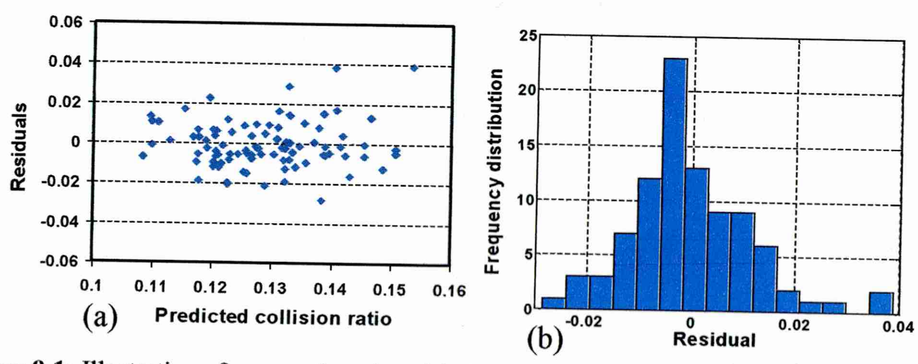



**Figure 9.1:** Illustration of assumptions 3 and 4, (a) scatter plot of the residual, and (b) frequency distribution of the residual.

## 9.4 Linear Regression Implementation

Collision ratio, *CRV*, *CW*, *DIFS*, and queue status ratio are network parameters that vary over time. So, the variation in the values of these parameters represents a time series signal. The prediction of the time series is mainly based on the current and past values of *CR*, *CRV*, *CW*, and queue status values. Consequently, it is essential to obtain a data record of these parameters. The data records (observations) should have an informative and a sufficient number of records for achieving prediction. Consequently, the future values of these parameters can be predicted as a function of their past values.

Different methods are used for prediction. Basically, they can be classified into linear and nonlinear models. For instance, linear models include AutoRegressive (*AR*) (Box and Jenkins, 1976) and (Harvey, 1981), Moving Average (*MA*) (Vandaele, 1983), mixed of *AR* and *MA* (*ARMA*) (Box and Jenkins, 1976), Integrated *ARMA* model (*ARIMA*) (Ljung, 1999) and Seasonal *ARMA* models (*SARMA*) (Vandaele, 1983). The nonlinear models include Volterra Series Expansions (Priestley, 1988), Wiener and Hammerstein models (Ljung, 1999), Bilinear model (Priestley, 1988), Threshold Autoregressive model (*TAR*), and Exponential Autoregressive model (*EAR*) (Priestley, 1988). A detailed explanation about linear and nonlinear time series models can be found in (Box and Jenkins, 1976), (Vandaele, 1983), (Ljung, 1999), (Chatterjee and Price, 1991), and (Priestley, 1988).

In this chapter, the linear *AR* model is used. Both, the simple and multiple linear regression techniques are considered for prediction purposes. An explanation of the use of the two methods in the IEEE 802.11 MAC protocol is provided in this section.

Regression analysis can be defined as the analysis of relationship among variable (Chatterjee and Price, 1991). It is the most widely employed statistical method due to its simplicity for creating a functional relationship among variables (Chatterjee and Price, 1991). The simple methods for prediction are preferred for practical reasons. Simple models are easier to test against for cross-validation studies. They are less costly to put into practice in predicting and controlling the output in the future. They are also easier to understand, and therefore have been used in this study. The relationship is shown in a formula that indicates the dependent variable ($y$) in terms of one or more independent variables, ($x_1, x_2, \ldots, x_p$). The regression equation takes the following form (Chatterjee and Price, 1991).

$$y=b_0+b_1x_1+b_2x_2+\ldots+b_px_p \qquad (9.1)$$

Where $b_0$, $b_1$, $b_2$, ..., $b_p$ are called regression coefficients that are determined from the data. If the regression equation has one independent variable it is called a simple regression equation. An equation that has more than two independent variables is pertained to as a multiple regression equation (Chatterjee and Price, 1991). Due to the relationship between the dependent variable ($y$) and the independent variable ($x$), the regression equation can be used to predict values of ($y$) for a given set of ($x$) values.

Initially, a simple linear regression model was considered. Hereafter, a multiple linear regression was employed. Generally, the *CR*, *CRV*, *CW*, and the queue status values were considered as the observations unit. These observations consist of a dependent variable $y$ (i.e. *CR*, *CRV*, *CW*, and queue status ratio) and an independent variable $x_i$ (i.e. the past values of *CR*, *CRV*, *CW*, and queue status ratio). So, the relationship between $y$ and $x_i$ is considered as a simple linear model as indicated in Equation 9.2 (Chatterjee and Price, 1991):

$$y_i = \beta_0 + \beta_1 x_{1i} + u_i, \qquad i = 1,2,\ldots,n \tag{9.2}$$

Where $\beta_0$ and $\beta_1$ are constants called regression coefficients, and $u_i$ is a random disturbance which measures the discrepancy between the actual value and the predicted one. The coefficient parameters $\beta_0$ and $\beta_1$, are estimated by using the Least Square Estimation (*LSE*). *LSE* is widely used for fitting a regression line for the observed data (Stat, 2006). It minimises the sum of the squares of the deviation of each data point above or below to the regression line ($S(\beta_0, \beta_1)$). The positive and negative values are considered since the deviations from the regression line are first squared then summed as indicated in Equation 9.3 (Chatterjee and Price, 1991):

$$S(\beta_0, \beta_1) = \sum_{i=1}^{n} u_i^2 = \sum_{i=1}^{n} (y_i - \beta_0 - \beta_1 x_{1i})^2 \tag{9.3}$$

The values of $\beta_0$ and $\beta_1$ that reduce the error $S(\beta_0, \beta_1)$; $b_0$ and $b_1$ are given by Equation 9.4 (Chatterjee and Price, 1991):

$$b_1 = \frac{\sum (y - \bar{y})(x_{1i} - \bar{x}_1)}{\sum (x_{1i} - \bar{x}_1)^2}, \quad \text{and } b_0 = \bar{y} - b_1 \bar{x}_1 \tag{9.4}$$

$$\text{Where } \bar{y} = \frac{\sum y_i}{n} \text{ and } \bar{x}_1 = \frac{\sum x_{1i}}{n} \tag{9.5}$$

According to the number of observations ($i^{th}$ observations), the predicted value by the simple regression model is given as in Equation 9.6 (Chatterjee and Price, 1991):

$$\hat{y}_i = b_0 + b_1 x_{1i} \tag{9.6}$$

So, the residual corresponding to the $i^{th}$ observations is given by:

$$e_i = y_i - \hat{y}_i \tag{9.7}$$

Where $y_i$ and $\hat{y}_i$ are the actual and the predicted values, respectively.

Simple linear regression has been briefly discussed when the model order was one. When the model order is increased, a multiple regression model is essential, since it provides a quantitative relationship between a group of independent variables that compose the matrix (**X**) and a dependent variable that composes the vector (**Y**) (this will be explained later in this section). In the multiple regression model, the data consist of ($n$) observations, one dependent variable ($y$) and ($p$) independent variables ($x_1$, $x_2$,..., $x_p$). The relationship between the response and the explanatory variables is still called a linear model and is given in Equation 9.8 (Chatterjee and Price, 1991):

$$y_i = \beta_0 + \beta_1 x_{1i} + \beta_2 x_{2i} + ... + \beta_p x_{pi} + u_i \tag{9.8}$$

Where $\beta_0$, $\beta_1$, $\beta_2$, ..., $\beta_p$ are constant represent the regression coefficients, and $u_i$ is a random disturbance. $\beta$'s values are estimated by minimising the sum of the squared errors. Following the procedure of the simple linear regression and after estimating $b_0$, $b_1$, $b_2$ and $b_p$, the predicted values can be given as in Equation 9.9 (Chatterjee and Price, 1991).

$$\hat{y}_i = b_0 + b_1 x_{1i} + b_2 x_{2i} + ... + b_p x_{pi} \tag{9.9}$$

In this chapter, the matrix notation of simple and multiple linear regression is used for implementing the prediction model. The value of **Y** is represented by the *CR*, *CRV*, *CW*, and queue status ratio. The independent variable **X** is represented by the previous values of *CR*, *CRV*, *CW*, and queue status ratio over time. The parameters $b_0$, $b_1$, $b_2$ and $b_p$ in Equation 9.9 are calculated for each parameter (e.g., *CR*) using the least squares technique. So, in matrix form the model can be written as in Equation 9.10.

$$\mathbf{Y} = \mathbf{XB} + \mathbf{u} \tag{9.10}$$

**Y** is the vector containing all the output samples obtained by computing the *CR*, *CRV*, and the queue status ratio values (see section 7.3 and 8.3).

$$\mathbf{Y} = \begin{bmatrix} y_1 \\ y_2 \\ \cdots \\ \cdots \\ y_n \end{bmatrix}, \mathbf{B} = \begin{bmatrix} \beta_0 \\ \beta_1 \\ \cdots \\ \cdots \\ \beta_p \end{bmatrix}, \mathbf{u} = \begin{bmatrix} u_0 \\ u_1 \\ \cdots \\ \cdots \\ u_n \end{bmatrix}, \text{ and } \mathbf{X} = \begin{bmatrix} -x_{p1} & \cdots\cdots & -x_{11} & x_{01} \\ -x_{p2} & \cdots\cdots & -x_{12} & x_{02} \\ \cdots & \cdots\cdots & \cdots & \cdots \\ \cdots & \cdots\cdots & \cdots & \cdots \\ -x_{pn} & \cdots\cdots & -x_{1n} & x_{0n} \end{bmatrix}$$

Note that **Y** is a ($n$ x $1$) vector, **X** is a ($n$ x $p$) matrix, $\beta$ is a ($p$ x $1$) vector, and **u** is a ($p$ x $1$) vector, where ($n$) is the number of observation and ($p$) is the model order. In the **X** matrix $x_{0i} = 1$ for all $i$, where $i = 1, 2, 3, \ldots, n$. The assumptions made about **u** for least squares estimation are **E(u) = 0**, that means the values of **u** have zero mean and constant variance. This implies that **E(Y) = XB**. As mentioned in an earlier part of this section, the least squares estimator **b** of $\beta$ is obtained by minimising the sum squared deviations of the observations from their expected value. This leads to the system of equations (Chatterjee and Price, 1991).

If $(X'X)\,\mathbf{b} = X'Y$ and assuming that $(X'X)$ has an inverse, then **b** can be rewritten as ($\mathbf{b} = (X'X)^{-1}X'Y$). The vector of predicted values $\hat{Y}$ corresponding to the observed **Y** is:

$$\hat{Y} = X\mathbf{b} = PY \tag{9.11}$$

Where $P = X(X'X)^{-1}X'$. Based on that, the vector of residual is given by

$$\mathbf{e} = Y - \hat{Y} = Y - X\mathbf{b} = (I - P)Y \tag{9.12}$$

Where ($I$) is the identity matrix.

A simple and multiple linear regression technique are presented as a prediction system. According to the nature of wireless networks such as the limited network resources, shared channel and risk of collision, the prediction system has to compromise between the complexity of the model and its prediction accuracy. The following section presents the method used for measuring the prediction model accuracy.

### 9.4.1 Measuring Prediction Accuracy

To assess the accuracy of the prediction model, the Mean Square Error (*MSE*), Root Mean Square Error (*RMSE*), Mean Absolute Error (*MAE*), Mean Relative Error (*MRE*), and the correlation coefficient were computed.

#### 9.4.1.1 Mean Absolute Deviation (MAD)

The *MAD* measures the prediction accuracy by computing the absolute error for each record in the data set. Then the average of the magnitudes of errors is calculated to obtain the *MAD*, which is given as (Farnum and Stanton, 1989).

$$MAD = \frac{1}{N}\sum|y - \hat{y}| \tag{9.13}$$

#### 9.4.1.2 Mean Square Error (MSE) and Root Mean Square Error (RMSE)

The *MSE* is obtained by averaging the prediction errors (i.e. $Y - \hat{Y}$). *MSE* uses the square values of the prediction errors instead of using the absolute values. So, using the square value gives more weight to large prediction error than the absolute error.

Equations 9.14 and 9.15 represent the *MSE* and *RMSE*, respectively (Farnum and Stanton, 1989).

$$MSE = \frac{1}{N}\sum(y-\hat{y})^2 = \frac{1}{N}\sum e^2 \tag{9.14}$$

$$RMSE = \sqrt{MSE} \tag{9.15}$$

#### 9.4.1.3 Mean Relative Error (MRE)

The *MRE* is evaluated by expressing each prediction error according to the actual value of the time series. *MRE* is given in Equation 9.16 (Farnum and Stanton, 1989).

$$MRE = \frac{1}{N}\sum\left|\frac{e}{y}\right| \tag{9.16}$$

#### 9.4.1.4 Correlation Coefficient (*R*)

The correlation coefficient (*R*) is widely used to evaluate the prediction accuracy and goodness. Also, it shows the level of model fit in Equation 9.2 to the observed data. The correlation coefficient is evaluated by using Equation 9.17 (Chatterjee and Price, 1991). The correlation coefficient (*R*) takes numerical values between 1 and -1. The value of 1 represents a positive linear relationship and the value of -1 represents a negative linear relationship between the actual and the predicted values. To obtain an easier assessment of the prediction accuracy or goodness, the coefficient of determination ( $R^2$ ) is calculated as seen in Equation 9.18. The coefficient of determination takes values between 0 and 1. The best value is the closest to 1.

$$R = \frac{\sum(y_i-\bar{y})(\hat{y}_i-\bar{\hat{y}})}{\left[\sum(y_i-\bar{y})^2\sum(\hat{y}_i-\hat{y})^2\right]^{1/2}} \tag{9.17}$$

$$R^2 = 1-\frac{\sum(y_i-\hat{y}_i)^2}{\sum(y_i-\bar{y})^2} \tag{9.18}$$

Where ( $y$ ) is the actual value, ( $\bar{y}$ ) is the mean of the actual values, ( $\hat{y}$ ) is the predicted value and ( $\bar{\hat{y}}$ ) is the average of the predicted $\hat{y}$ 's.

There are several ways for validating the prediction system: For instance, the prediction model can be validated by evaluating the prediction error over a new data set that is completely different than the observed data. Another useful technique for model validation is the residual analysis (see Equation 9.7).

As a summary, the main steps to construct the *AR* prediction model in IEEE 802.11 DCF scheme are summarised as follows:

- Model Identification: this is required to examine the data and identify the important characteristics of the selected parameter i.e. whether it is predictable or not (i.e. it should not be random and should have a trend). In this chapter *CR*, *CRV*, *CW*, and queue status ratio represent the time series processes, since they agreed with model requirements.
- Estimation: to determine the coefficient parameters.
- Forecasting: perform one-step-ahead prediction of the selected parameters (e.g. collision ratio).
- Model validation: to do residual analysis between the fitted model and the actual data and evaluate if the error inferred by the model fulfils the requirements. Otherwise, the model is invalid and needs to be modified until a satisfactory model is reached.

## 9.5 Description of the Approach

In this chapter the following steps were performed in designing the proposed modelling and prediction system: (i) performing computation of the *CR*, *CRV*, and queue status ratio; (ii) collecting the computed parameters in a matrix notation ; (iii) analysing and modelling these measured parameters using an *AR* statistical model; (iv) obtaining the model parameters and evaluating the goodness of the model based on the analysis, e.g. considering the *MSE* prediction errors and the correlation coefficient factor (*R*); (v) performing online prediction based on the model parameters obtained; and (vi) adjusting the MAC protocol transmission parameters such as *CW*, *DIFS*, and transmission rate based on the predicted values of *CR*, *CRV*, and queue status ratio.

Multimedia transmissions over the IEEE 802.11 DCF scheme impose strict QoS requirements in terms of delay and jitter. Therefore, the use of other modelling schemes such as Moving Average (*MA*) and AutoRegressive Moving Average (*ARMA*) models are more difficult and complex proposals for designing a prediction system. Modelling time series values by *MA* and *ARMA* models take a non-deterministic amount of time, since fitting a *MA* or *ARMA* model leads to a quadratic system. Therefore, *AR* models in such applications are commonly used since they can be fit to data in a deterministic amount of time (Cheng and Marsic, 2003). The results presented in section 9.7 confirm that *AR* model is adequately accurate for modelling the network parameters such as *CR*, *CRV*, and queue status ratio in IEEE 802.11 wireless networks.

### 9.5.1 Prediction of Collision Ratio

The values of collision ratio and *CRV* were obtained using Equations 7.2 and 7.7 (see section 7.3 Chapter 7), respectively. Equation 8.20 (see section 8.3.4 Chapter 8) was employed to compute the queue status ratio. The values obtained (i.e., observations) were represented in a matrix notation in order to be used for prediction purposes. For example, in a collision ratio prediction model, the collision ratio ($CR_t$) in the current time ($t$) and the collision ratios ($CR_{t-1}$, $CR_{t-2}$, $CR_{t-3}$, ...) in previous times were employed to predict one step ahead of the collision ratio. The purpose is to obtain a prediction model which is such that the mean square of the deviations between the actual and the predicted values (i.e. $CR_{t+1} - CR_t$) is as small as possible. Similarly, the prediction of the *CRV*, *CW*, and queue status ratio values followed the same procedure.

The observed data or the output samples obtained by computing the *CR*, *CRV*, queue status ratio is stored in vector **Y**. These samples were arranged in the **X** matrix as previous values that were used to predict the future value of the same parameters (i.e. the predicted *CR* is based on the previous values of *CR*). In this chapter, the number of samples or observations used was 50 and the model order was 4. These were experimentally selected as discussed later in this chapter (see section 9.7.1). So, these observations were arranged in **Y** vector and the **X** matrix as follows:

$$\mathbf{Y}=\begin{bmatrix} CR_1 \\ CR_2 \\ \ldots \\ \ldots \\ CR_{50} \end{bmatrix}, \text{ and } \mathbf{X}=\begin{bmatrix} 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 0 \\ -CR_4 & -CR_3 & -CR_2 & -CR_1 & 1 \\ -CR_5 & -CR_4 & -CR_3 & -CR_2 & 1 \\ \ldots & \ldots & \ldots & \ldots & \ldots \\ -CR_{49} & -CR_{48} & -CR_{47} & -CR_{46} & 1 \end{bmatrix}$$

By finding the model coefficient parameters using $P = X(X'X)^{-1}X'$, the forecasted value of *CR* at $t+1$ can be obtained from the previous value of *CR* at ($t-3$, $t-2$, $t-1$, $t$, and 1) as follows ($CR_{t+1} = [-CR_{t-3} \;\; -CR_{t-2} \;\; -CR_{t-2} \;\; -CR_t \;\; 1]\, P$). Note that by multiplying the matrix (*1* x *5*) (i.e. $[-CR_{t-3} \;\; -CR_{t-2} \;\; -CR_{t-2} \;\; -CR_t \;\; 1]$ ) by the vector (*5* x*1*) (i.e. $P$ model coefficient parameters) the predicted *CR* value at $t+1$ is equal the vector (*1*x*1*) (i.e. the predicted *CR*). The same procedure is also applied for predicting the *CRV*, *CW*, and queue status ratio.

After appropriate model parameters have been determined, the forecasting procedure follows immediately. Before using the predicted value for adjusting the MAC protocol transmission parameters, the *MSE* is computed as the deviations between the actual and the predicted values. If the *MSE* is less than a certain threshold (0.005), the obtained model parameters are used for forecasting the future values of *CR*, *CRV*, and queue status ratio. Otherwise, the model parameters will be updated based on a new observation data set. The *MSE* threshold is selected to enable the system for providing an accurate estimation of the network parameters in order to achieve appropriate adjustments of MAC protocol transmission parameters. Afterwards, the predicted value is used to adjust the MAC protocol transmission parameters according to the Ratio based, *CRV*, and queue status monitoring schemes that were discussed in Chapter 7 (see section 7.3) and Chapter 8 (see section 8.3) .

When the *AR* prediction model is used to provide service differentiation, the predicted values of *CR*, *CRV*, and the queue status ratio are used to adjust the *CW*, *DIFS*, and transmission rate by following the adaptive QoS differentiation and queue status monitoring schemes that were discussed in section 8.3 (see Chapter 8) .

The model order value determines the number of previous values that need to be used to predict the future value. In this chapter, the model order is selected based on a trade off value between the prediction errors (*MSE*) and the correlation coefficient factor (*R*). For instance, four previous *CR* values are required to predict the future value of *CR* for a model order of 4. The full description of the proposed *AR* model for predicting the network parameters is provided in Figure 9.2.

### 9.5.2 Prediction of the Queue Status Ratio

In multi-hop networks, the buffers of the intermediate stations have the capability of accommodating different traffic types. However, this feature may cause serious congestion problems that may result in considerable buffer overflow and serious degradation of the QoS. The prediction of the queue status ratio ($\rho$) and comparing it with certain thresholds $Q_{\min\_ths}$ and $Q_{\max\_ths}$ as discussed in section 8.3.4 can be used as a measure of congestion in the future. In this case, the predicted queue status ratio can be employed as an effective congestion control, and traffic management schemes that either drop the arriving low priority data packets or reduce the transmission rate can be used.

The queue status ratio ($\rho$) is considered as an indication of traffic congestion in the buffer. In this situation and as an example, the congestion prediction of the network is obtained as follows: $\rho_{t+1} = [-\rho_{t-3} \ -\rho_{t-2} \ -\rho_{t-2} \ -\rho_t \quad 1]\, P$, where $\rho_{t+1}$ is the predicted queue status ratio and $\rho_{t-3}$, $\rho_{t-2}$, $\rho_{t-2}$, $\rho_t$ are the previous values, 1 is the offset, and $P$ is the model coefficient parameters.

According to the predicted value of $\rho$, the feedback control scheme is activated (see section 8.3.4). If the predicted $\rho$ has a value greater than the current minimum queue status threshold and less than the maximum queue status threshold (i.e. $Q_{\min\_ths} < \rho_{t+1} < Q_{\max\_ths}$), the main control action is to reduce the transmission rate of low priority traffic sources. Another control action is to drop the arriving low priority packets when the predicted $\rho$ is bigger than the maximum queue status threshold.

Thus the prediction of the future value of $\rho$ can help the MAC protocol take control action on time in order to decrease the potential of congestion. This improves the QoS for the high priority traffic and reduces the number of packet drops at the buffer for the low priority traffic in which more data sources can be served.

### 9.5.3 Prediction of the CW Size

The autoregressive model was also applied to predict the *CW* size. In Chapter 7, the Ratio based and *CRV* schemes were used to adjust *CW* size. In this chapter, the *CW* size was again computed according to a Ratio based scheme. When the number of observations of *CW* size is fulfilled, the *AR* model is performed to predict the future value of *CW*. Hereafter; the predicted value is used by the backoff algorithm. After successful transmission the *CW* size is updated according to Equation 9.19; while Equation 9.20 is used after each collision.

$$CW_{new}[N] = \max(CW_{\min}[N], CW_{predicted}[N]) \tag{9.19}$$

$$CW_{new}[N] = \min(CW_{\max}[N], CW_{predicted}[N]) \tag{9.20}$$

Where $CW_{new}[N]$ is the new *CW* for station $N$ ; $CW_{\min}[N]$ and $CW_{\max}[N]$ are the minimum and the maximum *CW*, respectively; and $CW_{predicted}[N]$ is the predicted *CW* for the station $N$ using the proposed *AR* model.

Equation 9.19 guarantees that the $CW_{new}[N]$ is always greater than or equal to the minimum $CW$ size. Similarly, Equation 9.20 ensures that $CW_{new}[N]$ is always less than the maximum $CW$ size.

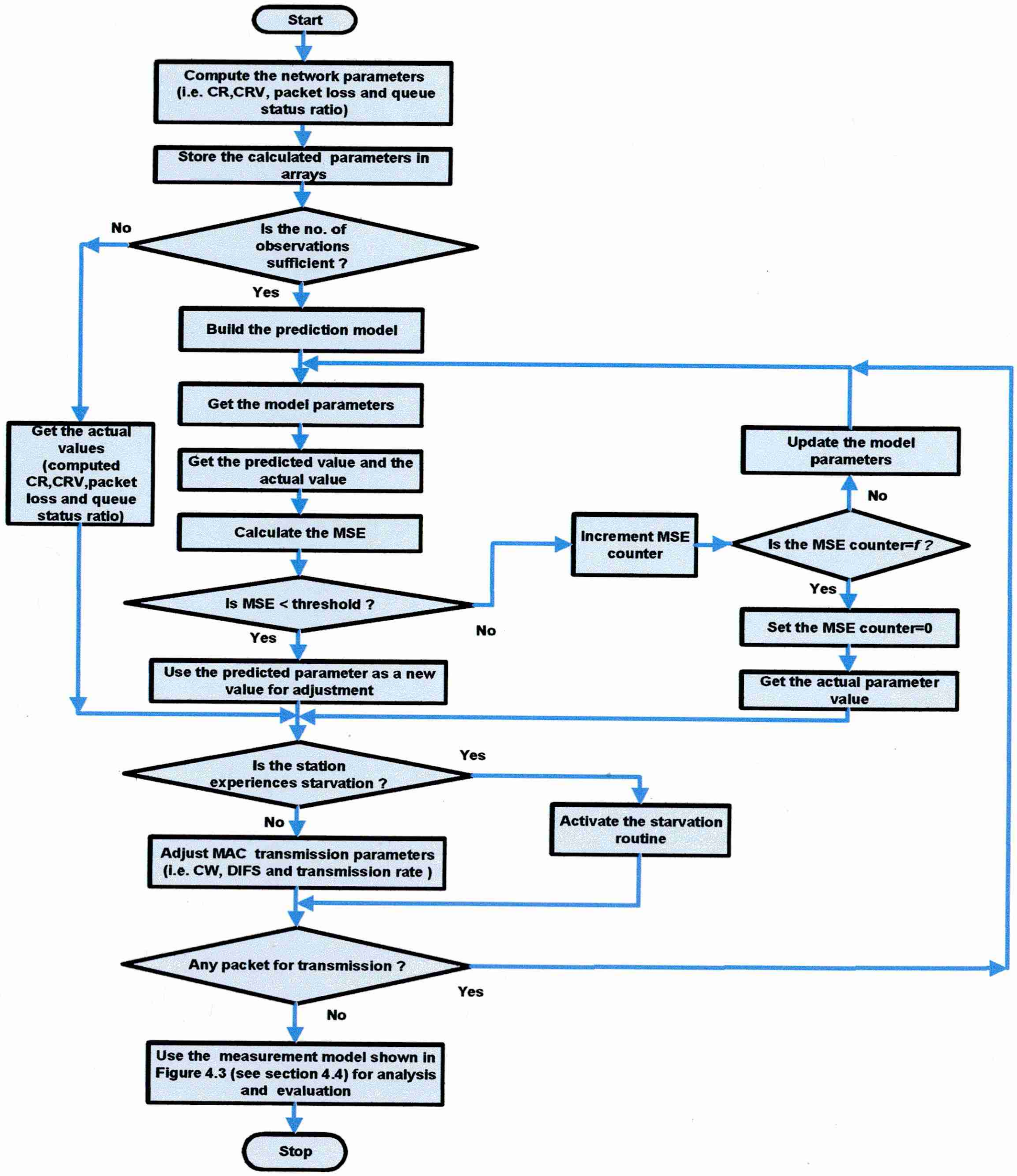



**Figure 9.2:** Autoregressive model flow chart.

## 9.6 Simulation Model

To evaluate the performance of the proposed $AR$ models for predicting $CR$, $CRV$, $CW$, and queue status ratio two network topologies were used. The network shown in Figure 4.2d (see Chapter 4) was used when all stations were able to hear the transmission of each other. Each station was capable of transmitting either $CBR$ or $VBR$ traffic. The second model represented a multi-hop topology as shown in Figure 4.2c (see Chapter 4).

The packet sizes for *CBR* traffic were 512 and 800 bytes for the high and low priority traffic, respectively. Each high priority station generated 192 Kbps. Low priority stations were able to generate either 480 Kbps or 160 Kbps as high and low bit rates, respectively. The *VBR* traffic had a variable frame size with a mean value equal to 800 bytes and a variable transmission interval. The simulation time was 300 seconds. In some cases the simulation time was varied and this will be highlighted for each scenario. Other simulation parameters are summarised in Table 4.1 (see Chapter 4).

The simulations were performed for several scenarios in order to evaluate the performance of the proposed schemes by means of comparison with the basic IEEE 802.11 DCF and the Exponential Increase Exponential Decrease (*EIED*) schemes. These scenarios include varying the network type (single-hop and multi-hop), changing network size (small, medium, and large network sizes), heavy load traffic, various traffic types (*CBR* and *VBR*), and when the number of active stations varies over time. A complete description of each scenario will be discussed in section 9.7.

## 9.7 Results and Discussion

In this section the performance of the basic IEEE 802.11 DCF, *EIED*, and the prediction schemes for several network configurations are analysed. There are four subsections. Section 9.7.1 presents the criteria of selection the model order and measuring the prediction model accuracy. Performance evaluation of the proposed *AR* model compared with the IEEE 802.11 DCF and *EIED* schemes is discussed in section 9.7.2 for different network sizes and various traffic types using heavy load traffic. The performance is discussed in context of the QoS parameters such as delay, jitter, throughput, packet loss, MAC protocol efficiency, and collision rate and in terms of the measured QoS. Section 9.7.2 includes other simulations conducted to investigate particular aspects of IEEE 802.11, *EIED*, and the proposed schemes such as varying the number of active stations over time. Section 9.7.3 discusses the capability of the prediction models in providing service differentiation in single-hop networks. Section 9.7.4 presents the use of the regression models for providing service differentiation in multi-hop networks.

### 9.7.1 Selection of Model Order and Measuring Prediction Accuracy

The model order was experimentally selected by plotting the *MSE*, *MAD*, *MRE*, and the correlation coefficient ($R$) as a function of the model order. For prediction, large model order results in a small estimated white noise variance (Hao et *al.*, 2005). Although, decreasing the model order reduces the system complexity, it may result in large *MSE* values and bad representation of the observed data.

Figures 9.3a to 9.3g show the relationship between the actual and the predicted *CR* values for different model orders. As shown in Figures 9.3a to 9.3g, as the model order increases the estimation generally improves. For instance, model order of 1 shows least estimation of the actual value as shown in Figure 9.3a. This is because using a model order equal to 1, *AR* analysis only gives one coefficient parameter, and one term in the series is insufficient to estimate the next term in most cases. On the contrary, a larger model order, for instance model order 7, the actual *CR* and the predicted *CR* values are closely correlated with $R$ = 82%. So, the larger the model order is, the superior the estimation is, the less the prediction errors are and the higher the complexity of the prediction system is. However, too high model order may cause the system to not work properly. This is because the transmission of time-sensitive applications over the IEEE 802.11 MAC protocol requires less delay and less computation time, therefore, the model order has to be critically selected.

In this chapter and for the selected scenarios, the model order is selected to have a trade off value between goodness and the validity of the model on one side (i.e. reasonable range of errors and an acceptable value of the correlation coefficient factor) and less complexity and computation costs on the other side. Note that the actual *CR* values (solid line) are dissimilar for all graphs in Figure 9.3. This is explained as follows: in the proposed *AR* model, the predicted *CR* value is used to adjust the *CW* size. The adjusted *CW* size varies with the variation in the model order, thus different *CW* size results in dissimilar computed *CR* over time (i.e. different actual values). For more explanation, the prediction errors and the correlation coefficient factor are plotted as a function of the model order as shown in Figures 9.4a, 9.4b, and 9.4c.

To evaluate the performance of the prediction model, the *MSE, MAD, MRE*, and the correlation coefficient factors were computed. A perfect prediction model has *MSE, MDA*, and *MRE* prediction errors equal to 0 and 1 or -1 correlation coefficient factor. To better appreciate the performance of the forecasting models, the *MSE*, *MAD*, and *MRE* prediction errors and the correlation factor need to be considered. In this chapter, the prediction errors and the correlation factor are plotted as a function of model order. In this discussion, the *CR* value is considered as an example.

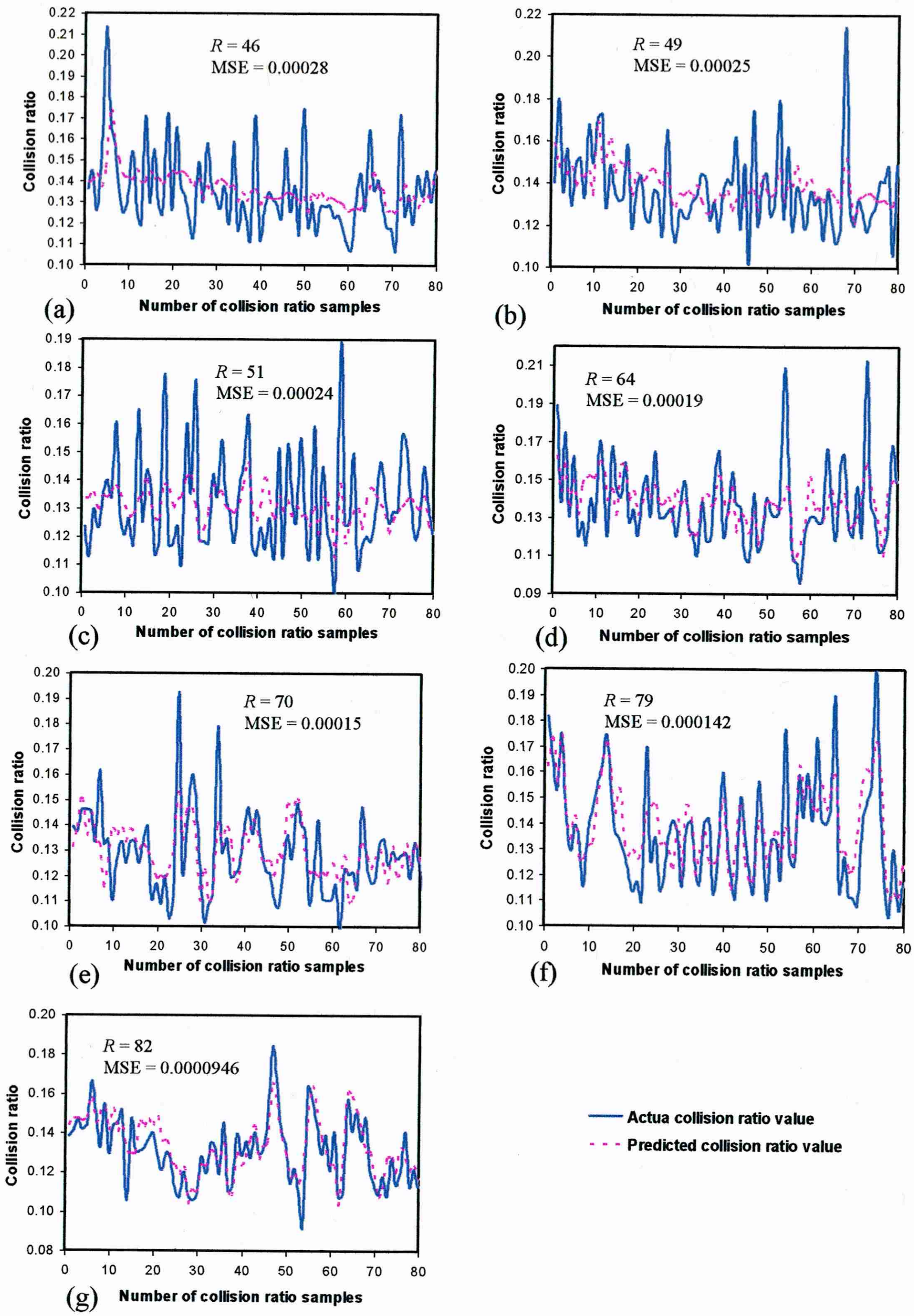



**Figure 9.3:** The actual and predicted collision ratio values using *AR* model for different model orders, (a) first, (b) second, (c) third, (d) fourth, (e) fifth, (f) sixth, and (g) seventh.

The *MSE*, *MAD*, and *MRE* prediction errors decreased as the model order increased. The smallest prediction errors were obtained when the model order was 7. The best correlation coefficient ($R$ = 82%) was also achieved at model order equal to 7 as shown in Figures 9.4a to 9.4c. Hence, there was no straightforward way to determine the correct model order. Therefore, in this chapter and in order to avoid the complexity and the overhead that might result from using a high model order such as 7, the model order of 4 was selected. Model order 4 provided a trade off value between the prediction errors and the correlation factor from one side (see the cross points in Figures 9.4a to 9.4c) and less complexity and computational time on the other side.

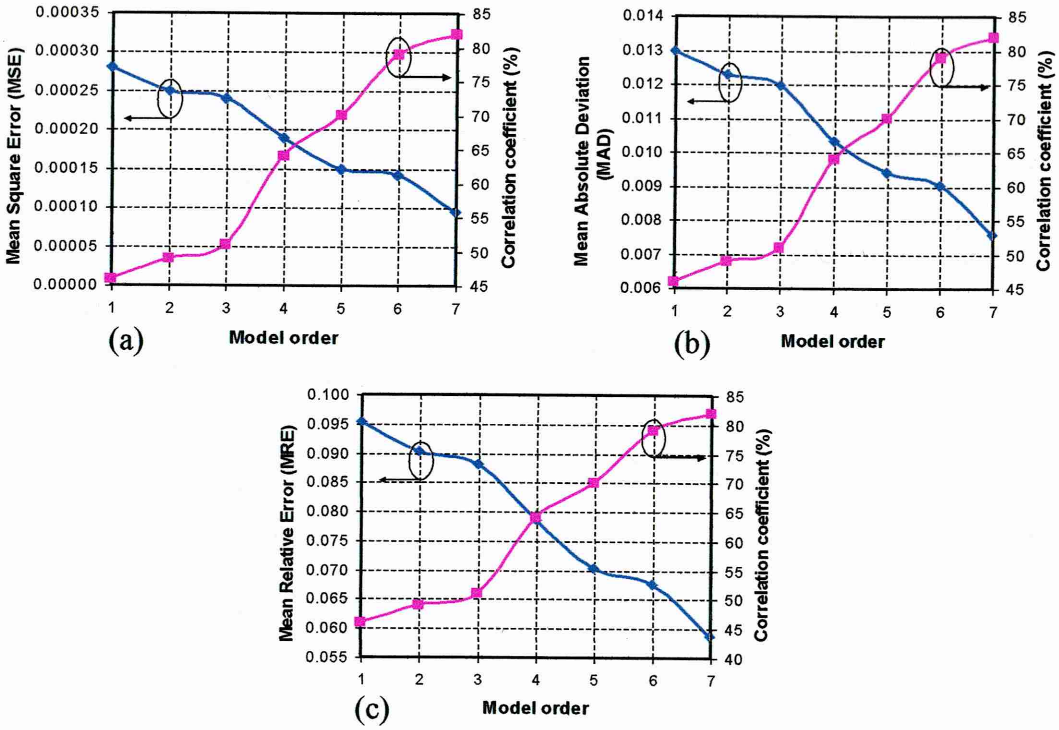



**Figure 9.4:** The prediction errors and the correlation coefficient factor as a function of model order, (a) *MSE*, (b) *MAD*, and (c) *MRE*.

### 9.7.2 Performance Evaluation of Autoregressive Models

This section is divided into two main parts. In the first part, the results of *CW* adjustment for transmitting *CBR* and *VBR* traffic and for varying number of active stations over time is presented. The second part demonstrates the results of adjusting the *CW* and *DIFS* combined for *CBR* and *VBR* traffic. The two parts are discussed for different network sizes (i.e. small, medium, and large networks).

#### 9.7.2.1 Autoregressive model for Contention Window Adjustments

Several scenarios are discussed in this section in order to critically investigate the performance of the proposed regression models, and then compare them with the standard IEEE 802.11 DCF and *EIED* schemes.

All the selected scenarios in this section used the topology shown in Figure 4.2d (see Chapter 4). The offered load that was delivered into the medium by the active senders represented around 80% of the channel capacity (i.e. more than 1600 Kbps). The transmission rate of each *CBR* source was 320 Kbps, 160 Kbps, and 80 Kbps for 5, 10, and 20 connections, respectively (i.e. small, medium, and large networks). The *VBR* traffic was obtained from a *VBR* trace file (see section 4.2.6 in Chapter 4) (TKN, 2005). Part of the trace file was used in this scenario with an 800 bytes mean frame length and 289 bytes standard deviation. The average values of QoS and QoS parameters were considered. Every point on the graph represented the average for 5, 10 or 20 connections based on the network size selected.

#### *9.7.2.1.1 Autoregressive Model for CW Adjustment with CBR Traffic*

Three cases were used to demonstrate the performance of the proposed *AR* model when *CBR* traffic was transmitted. These corresponded to small, medium, and large networks.

***Small network case***: only 5 connections transmitted at heavy load to 5 corresponding destinations. Each source transmitted 320 Kbps *CBR* traffic.

Figure 9.5a shows the average delay for the five schemes. The prediction of *CR*, *CRV*, and *CW* schemes were able to maintain lower values of average delay. For instance, the average delay was 58.5% and 51.4% less than that for the standard IEEE 802.11 DCF and the *EIED* schemes, respectively when the *AR* models was used to predict the *CR* value. Similarly, the prediction of *CRV* and *CW* showed smaller average delay than IEEE 802.11 DCF and *EIED* schemes. Indeed, the *AR* models tried to keep a lower delay less than (400 msec) for most of the transmitted packets in order to meet the minimum QoS requirements for time-sensitive applications.

The prediction schemes showed more fluctuations in their curves compared to the standard IEEE 802.11 DCF and *EIED* schemes. This is explained as follows: the prediction and adjustment processes were performed locally for each station. In the Ratio based, *CRV*, and the prediction schemes, when the number of contending stations was small, fewer collisions took place. Therefore, each station had smaller *CW* size to get more channel access. This caused several adjustments of the *CW* size for each station in order to keep the MAC protocol efficiency as high as possible, to reduce the collision rate to a minimum, and to lessen the number of wasted time slots.

IEEE 802.11 DCF and *EIED* collision rates were 50% and 40% higher than the collision rates of the *AR* prediction models as shown in Figures 9.5b. The prediction of *CR*, *CRV*, and *CW* schemes showed their capability of reducing the probability of collisions over time. The prediction schemes enhanced the collision avoidance mechanism of the IEEE 802.11 DCF by reducing the network collisions.

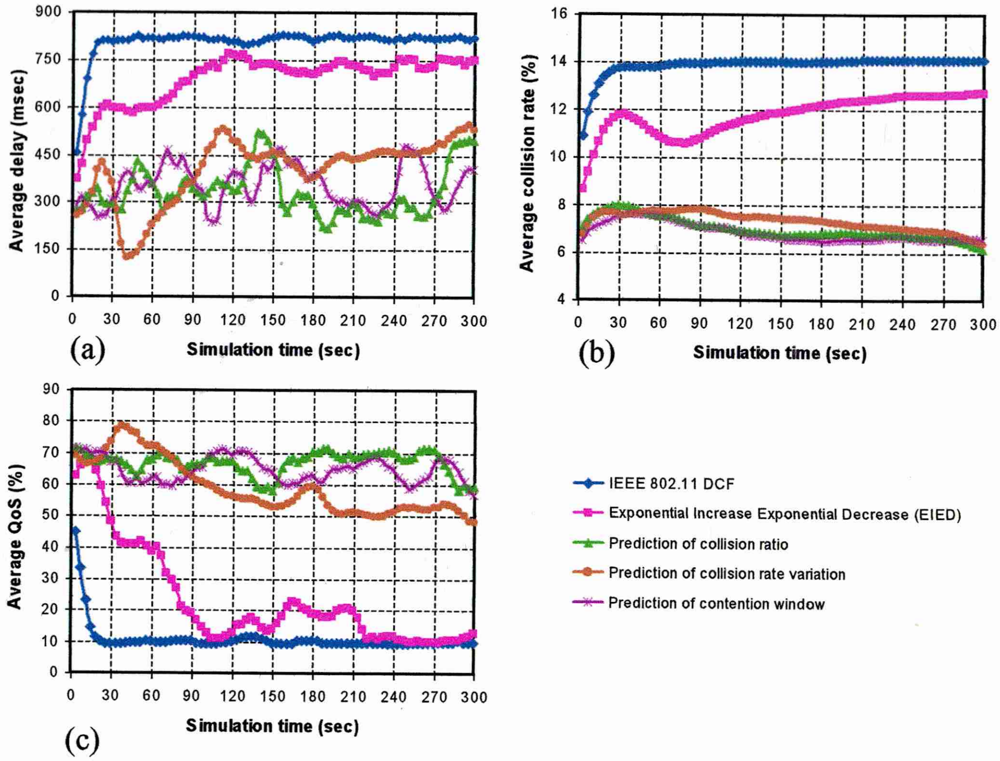



**Figure 9.5:** *AR* model for *CW* adjustment with *CBR* traffic for the small network, (a) average delay, (b) average collision, and (c) average QoS.

The simulation results for the QoS obtained are shown in Figure 9.5c. The figure indicated that IEEE 802.11 DCF and *EIED* schemes were not able to achieve the minimum QoS requirements for time-sensitive applications even for small network sizes and heavy traffic loads. The *AR* models performed dramatically better than the IEEE 802.11 DCF and *EIED* schemes. They had 67%, 58.9%, and 64.8% average QoS for *CR*, *CRV*, and *CW* prediction models, respectively. In contrast, IEEE 802.11 DCF and *EIED* schemes had a poor QoS with an average equal to 11% and 23.3%, respectively.

***Medium network case*:** the offered load was 80% of the channel capacity which was equally distributed between 10 connections. Each source transmitted 160 Kbps to its corresponding destination.

The average delay of the five schemes is shown in Figure 9.6a. For the medium network size, IEEE 802.11 DCF and *EIED* average delay values were considerably higher than the average delay of the prediction schemes. They had an average delay equal to 1788 msec and 1520 msec, respectively. Due to these long waiting times, buffer overflow in the IEEE 802.11 DCF and *EIED* schemes was increased, which degraded the QoS as shown in Figure 9.6c. In contrast, the *AR* prediction models had smaller values for average delay. They had mean values equal to 632.1 msec, 672.1 msec, and 604.4 msec for predicting *CR*, *CRV*, and *CW*, respectively.

According to Figure 9.6b the collision rate values during the initial 30 seconds of the simulation were different to those when the network was settled. This was due to the impact of routing information exchange during the initial period of the simulation which, once established, this effect became less. The probability of collisions reduced by more than 40% compared to the IEEE 802.11 DCF and *EIED* schemes when using collision ratio prediction model. As a result, good QoS levels were obtained with mean values equal to 57% when the *AR* regression models were employed as shown in Figure 9.6c. Conversely, IEEE 802.11 DCF and *EIED* had poor QoS levels.

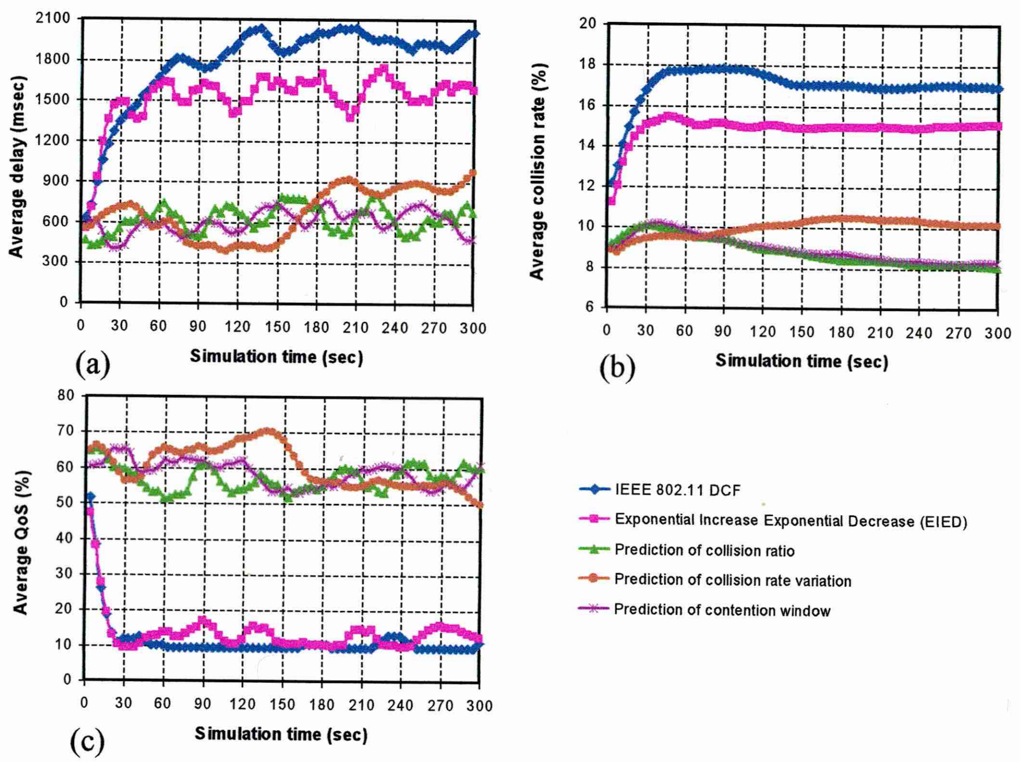



**Figure 9.6:** *AR* model for *CW* adjustment and using *CBR* traffic in the medium network, (a) average delay, (b) average collision, and (c) average QoS.

***Large network case*:** The performance of the *AR* models, IEEE 802.11 DCF, and *EIED* schemes was significantly affected when the network size switched from a small network size (i.e., 5 connections) to a medium network size (i.e., 10 connections). However, this impact was significant as the network size became large. In this section, the performance of these schemes was evaluated when the number of active stations was increased to 20 connections (i.e. large network). The volume of *CBR* traffic was 80% of channel capacity and each source transmitted 80 Kbps.

Figures 9.7a, 9.7b and 9.7c show that the performance of the five schemes was degraded in a large network. However, the prediction of *CR*, *CRV*, and *CW* schemes performed better than the IEEE 802.11 DCF and *EIED* schemes. It can be seen that the mean delay was reduced by 55% compared with the delay obtained for the IEEE 802.11 DCF and *EIED* schemes. The mean jitter value for the *AR* schemes was 48% smaller than the values obtained for the IEEE 802.11 DCF and *EIED* schemes. Throughput was improved by 18.8% and 14.9% compared to the IEEE 802.11 DCF and *EIED* schemes, respectively. This was due to the reduction in the number of packets dropped. Less than 10% of data packets were lost when the prediction schemes were used and more than 24% of the transmitted packets were lost for the IEEE 802.11DCF and *EIED* schemes.

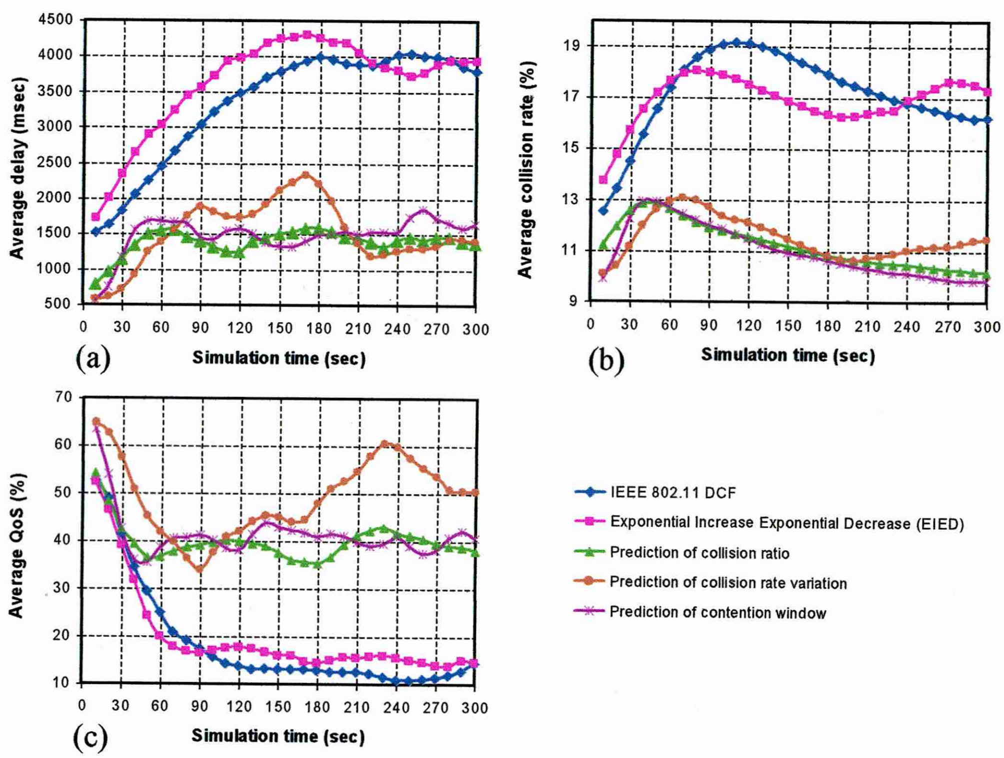



**Figure 9.7:** *AR* model for *CW* adjustment with *CBR* traffic in the large network, (a) average delay, (b) average collision, and (c) average QoS.

The prediction of *CR*, *CRV*, and *CW* schemes were able to maintain a lower collision rate than the IEEE 802.11 DCF and *EIED* schemes as shown in Figure 9.7b. This behaviour can be explained by the fact that the prediction of the future values of collision enables the adaptive scheme to be aware of the network conditions in the future. If the predicted *CR* or *CRV* values are very high the adaptive systems discussed in section 7.3 (see Chapter 7) have to increase the *CW* sizes to alleviate more excessive collisions in the future. Otherwise, this reduces the *CW* size to lessen the number of wasted time slots. As a result, a considerable reduction in the collision rate value is obtained causing an improvement in the performance of the network. This can be observed in the QoS curve shown in Figure 9.7c. The *AR* prediction of *CR*, *CRV*, and *CW* schemes achieved good QoS levels with mean values of 40%, 49%, and 41.4%, respectively; whereas, the IEEE 802.11 DCF and *EIED* schemes had poor QoS levels with mean values of 19% and 20%, respectively. Statistically, the prediction of *CR*, *CRV*, and *CW* schemes performed better than IEEE 802.11 DCF and *EIED* schemes. A summary of the performance of the *AR* models, IEEE 802.11 DCF, and *EIED* schemes for small, medium, and large network sizes is given in Table F.1 (see Appendix F).

So far, different simulations have been carried out to study the performance of QoS parameters in different schemes. In the previous sections, this performance has been studied when all stations transmitted *CBR* traffic. In the following section, the performance of the *AR* models will be studied when *VBR* traffic was considered.

##### *9.7.2.1.2 Autoregressive Model for CW Adjustment with VBR Traffic*

In this section three cases were also employed to demonstrate the performance of the proposed *AR* models when *VBR* traffic was transmitted. These are small, medium, and large networks. The active stations in each case transmitted heavy *VBR* traffic.

As shown in Figures 9.8a, 9.8b and 9.8c, the QoS values were presented for a small network, a medium network, and a large network. Generally, the trend of the QoS curves for *VBR* traffic showed more fluctuations than the *CBR* traffic for the reason discussed in section 7.5.3 (see Chapter 7).

As shown in Figures 9.8a, 9.8b and 9.8c, the average QoS values for the prediction models were in the excellent level with mean QoS equal to 72% for a small network and then degraded to a good level with mean QoS equal to 64% for a medium network. It remained in the good QoS range with a mean value of 54% in a large network. The IEEE 802.11 DCF and *EIED* schemes achieved a good QoS level with mean values equal to 50.4% and 46.3%, respectively in a small network. Hereafter, their QoS degraded to a poor level (i.e. less than 33%) for medium and large networks. A summary of the performance of the *AR* models, IEEE 802.11 DCF, and *EIED* schemes for small, medium, and large network sizes is given in Table F.2 (see Appendix F).

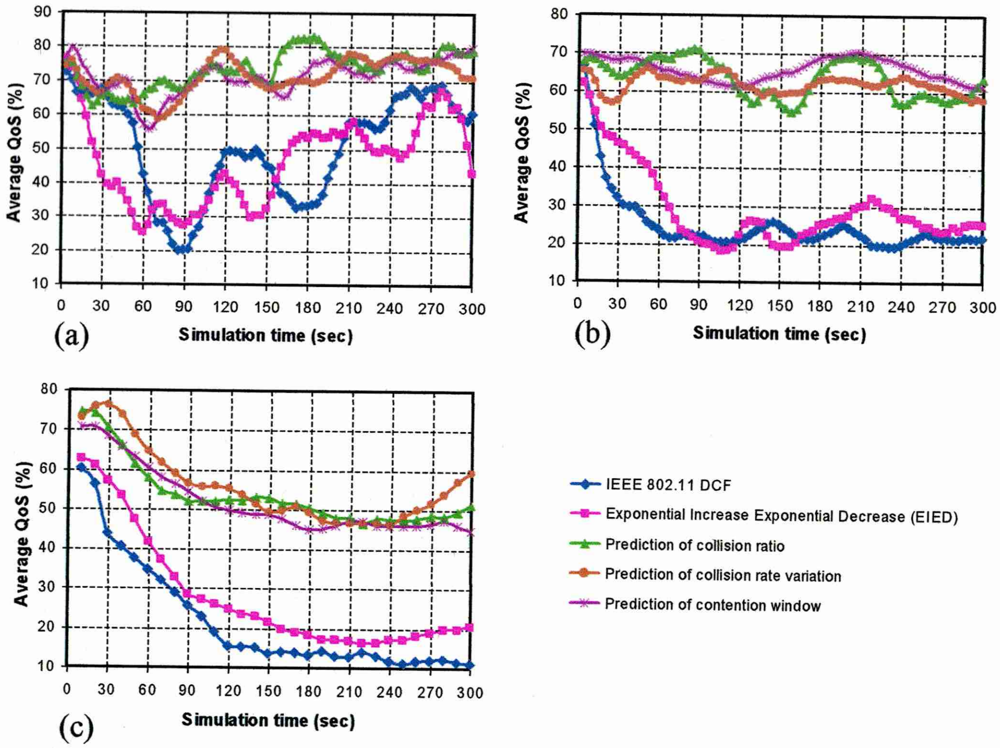



**Figure 9.8:** Average QoS when *AR* model was used for *CW* adjustment using *VBR* traffic, (a) small network, (b) medium network, and (c) large network.

It can be concluded that the *AR* models for predicting the *CR*, *CRV*, and *CW* values showed better performance than the standard IEEE 802.11 DCF and *EIED* schemes when *CBR* and *VBR* traffic were transmitted regardless of the network size.

#### *9.7.2.1.3 Autoregressive Model for CW Adjustment with Varying Number of Stations*

The performance of the *AR* model for predicting the collision rate (*CR*) was evaluated when the number of stations varied over time and the network faced highly changing configurations (i.e., different *CBR* traffic volume). This imposed an overhead on the *AR* model every time a new station joined the network. These changes in the network conditions pushed the *AR* model to update its coefficient parameters in order to keep the *MSE* as minimum as possible. The predicted *CR* value was then used to adjust the *CW* size which was appropriately chosen to avoid network performance degradation. For this cause, the scenario discussed in section 7.5.2.3.4 (see Chapter 7) was used to validate the performance of the regression model for forecasting the *CR* value.

The average delay of the standard IEEE 802.11 DCF, *EIED*, and *AR* model is shown in Figure 9.9. Average delay increased with the increase in the simulation time, since every 5 seconds a new station joined the network and shared the channel with other stations. However, the *AR* model reduced the value of average delay by 50% compared with the IEEE 802.11 DCF and *EIED* schemes. The maximum magnitude of delay was seen to be between the 200 and 300 second time period of simulation, because all stations were active during this period. Thereafter, the average delay decreased since the number of sources commenced to diminish by one every 5 seconds.

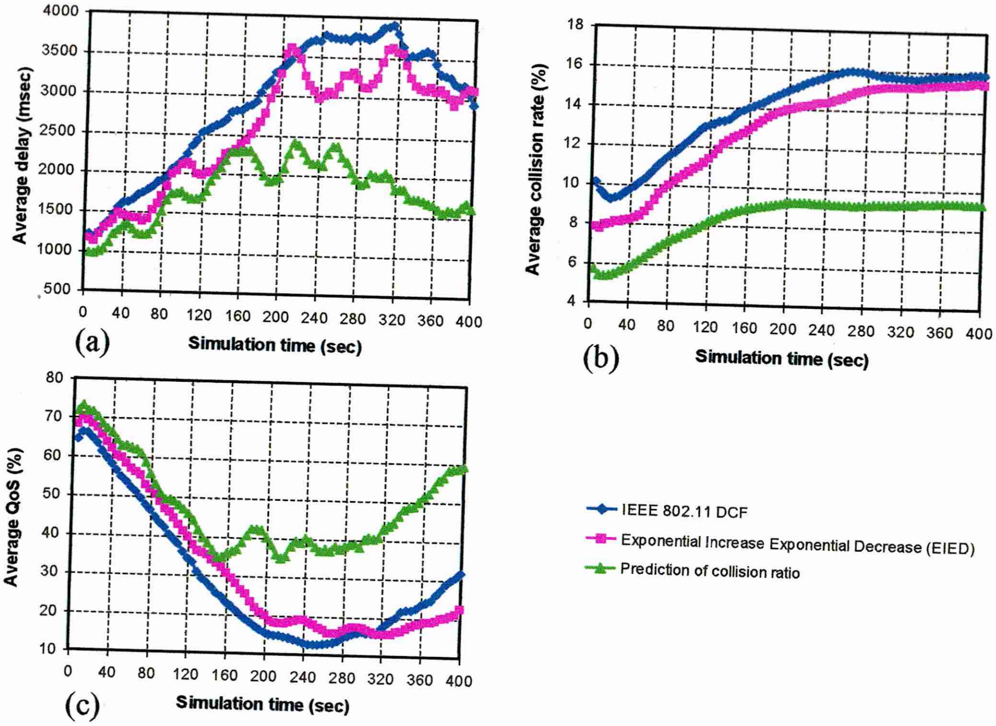



**Figure 9.9:** *AR* model for *CW* adjustment when the number of stations was increased with time, (a) average delay, (b) average collision, and (c) average QoS.

The prediction model closely maintained small values of delay as the IEEE 802.11 DCF and *EIED* schemes up to 100 seconds where there was small number of contending stations with the light load traffic. Afterwards the network became busier and the traffic status of the network was heavier. Once adequate observations were available, the *AR* model for each station performed prediction of the future values of *CR*. The predicted values were used to adjust the *CW* size after successful and unsuccessful transmissions in order to be familiar with future network conditions. Subsequently, better performance and lower values of average delay were obtained.

The action of the model was to adjust the *CW* size based on the predicted *CR*. The adjusted *CW* size significantly reduced the probability of collision by 40% compared to IEEE 802.11 DCF and *EIED* schemes. For instance, the average collision rate for *AR* model was 8.4%; whereas 14% average collision rate was observed for the IEEE 802.11 DCF scheme as shown in Figure 9.9b. The autoregressive model also achieved a higher average QoS than the IEEE 802.11 DCF and *EIED* schemes. The prediction model had

48% mean QoS which was 37% higher than that obtained for the standard IEEE 802.11 DCF and *EIED* schemes as shown in Figure 9.9c.

According to Figures 9.10a and 9.10b, the prediction model showed its capability to react to the variation in network conditions. Moreover, it revealed high correlation with actual *CR* value. The correlation coefficient factor (*R*) was 87.5% and the *MSE* was very small (0.0002). The value of *MSE* prediction error was acceptable since the minimum QoS requirements were maintained and most the *MSE* values obtained were less than the threshold value that was used to update the model parameters during the online prediction (i.e. threshold equal 0.005) as shown in Figure 9.10b. This value also agreed with the *MSE* value shown in Figure 9.4.

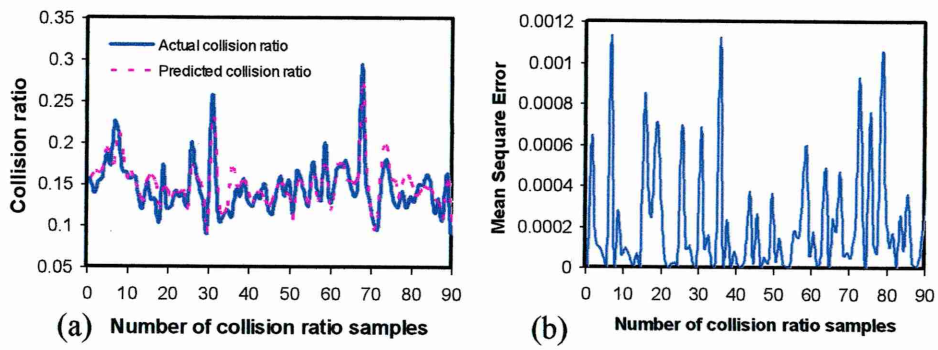



**Figure 9.10:** *AR* model validation, (a) correlation between the actual and the predicted values, and (b) calculated *MSE*.

In the previous sections, the performance of the proposed *AR* models for prediction the *CR*, *CRV*, and *CW* values were evaluated and compared with the IEEE 802.11 DCF and *EIED* schemes when the *CW* size was adaptively adjusted. The results obtained indicated that the *AR* model outperformed the standard IEEE 802.11 DCF and *EIED* schemes in all the discussed scenarios. The prediction model of *CR* value revealed slight improvements compared to the prediction models of *CRV* and *CW* values.

#### 9.7.2.2 Autoregressive Models for Adjusting *CW* and *DIFS* Parameters

In this section, the performance of predicting the *CR* and *CRV* values to simultaneously adjust both the *CW* and *DIFS* parameters is discussed in two scenarios. The results were compared with the IEEE 802.11 DCF and *EIED* schemes. In the first scenario, *CBR* traffic was transmitted and in the second scenario, *VBR* traffic was considered. The two scenarios used the network topology shown in Figure 4.2d (see Chapter 4) when 10 connections were considered. The transmitted load represented 80% of the channel capacity (i.e. more than 1600 Kbps). Each *CBR* source transmitted 160 Kbps. The *VBR* traffic had a variable packet size and a variable inter-packet interval. The average values of QoS and QoS parameters were considered in the evaluation process. Note that the performance comparison included the *AR* prediction model of *CR* value when it was used to adjust the *CW* parameter only.

***Constant Bit Rate Traffic Case*:** The offered load was 80% of the channel capacity which was equally distributed among 10 connections. The transmission rate for each source was 160 Kbps and the traffic type was *CBR*.

The average delay of IEEE 802.11 DCF, *EIED*, and *AR* schemes is shown in Figure 9.11a. The figure indicated that the *AR* model for adjusting the *CW* and *DIFS* values resulted in smaller values of delay. Average delay reduced by 17.5% compared to the value obtained when the prediction of *CR* was used to adjust the *CW* size only. It was reduced by 70.8% and 65.7% compared with the IEEE 802.11 DCF and *EIED* schemes, respectively.

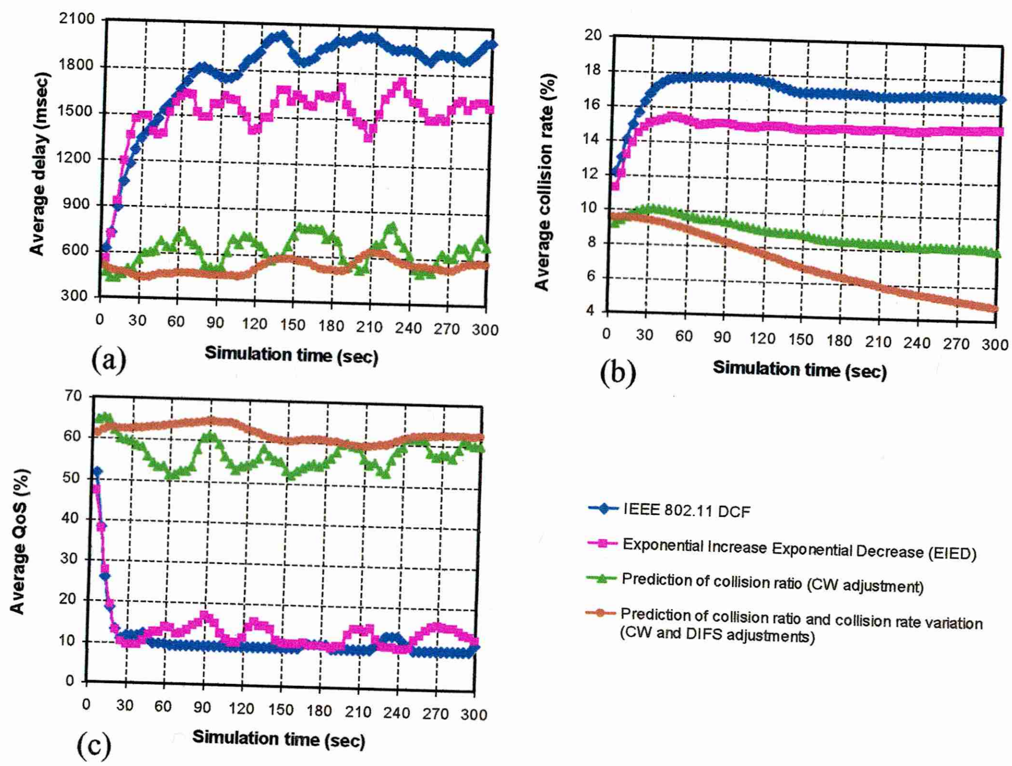



**Figure 9.11:** *AR* model for *CW* and *DIFS* adjustments with *CBR* traffic for the medium network, (a) average delay, (b) average collision, and (c) average QoS.

A significant reduction was also observed in the collision rate value when the *AR* model was used to predict the *CR* and *CRV* values as shown in Figure 9.11b. The collision rate was minimised by 20.5%, 58.5%, and 52.5% compared to the *AR* model for predicting the *CR*, IEEE 802.11 DCF, and *EIED* schemes, respectively. This reduction positively influenced the QoS obtained using the *AR* model for adjusting the *CW* and *DIFS* values.

A good QoS level was obtained with mean value of 62%. This was 7.6% higher than the QoS obtained when the *AR* model was used to adjust the *CW* size only.

***Variable Bit Rate (VBR) Traffic Case*:** The performance of *AR* prediction model to adjust *CW* and *DIFS* was validated when data sources transmitted *VBR* traffic in a medium size network.

A summary of the performance of *AR* prediction model to adjust *CW* and *DIFS* values compared with IEEE 802.11 DCF, *EIED*, and *AR* model for adjusting *CW* is given in Figures 9.12a, 9.12b and 9.12c. The prediction model for adjusting *CW* and *DIFS* resulted in delay values of less than 200 msec on average. This corresponded to an excellent QoS level with mean value equal to 74%. The average QoS achieved was 66%, 60.5%, and 13.8% higher than the IEEE 802.11 DCF, *EIED*, and the *AR* prediction model for adjusting only the *CW* value, respectively.

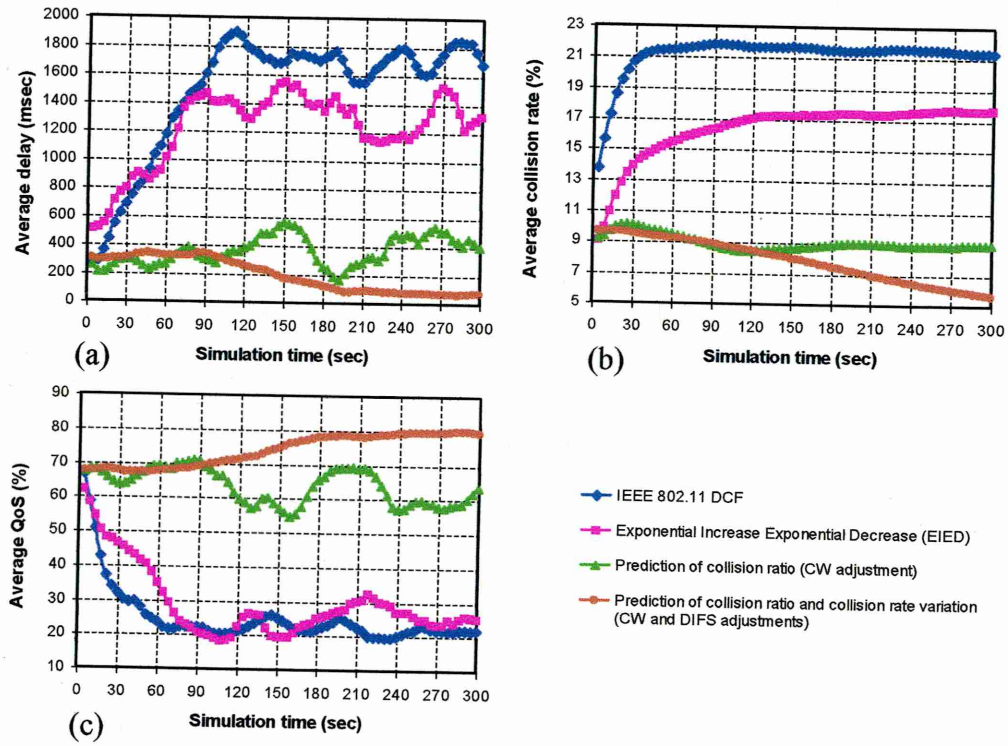



**Figure 9.12:** *AR* model for *CW* and *DIFS* adjustments with *VBR* traffic using the medium network, (a) average delay, (b) average collision, and (c) average QoS.

The proposed *AR* model for adjusting the *CW* and *DIFS* according to the predicted *CR* and *CRV* values resulted in fewer fluctuations as shown in Figures 9.11 and 9.12. As discussed in section 9.7.2.1, the predicted *CR* was used as an alert signal to the adaptive system to adjust the *CW* size according to the future network conditions which resulted in higher QoS and lower collision rate values than the standard IEEE 802.11 DCF and

*EIED* schemes. In the IEEE 802.11 DCF and *EIED* schemes, the *CW* size was adjusted according to the current network conditions without considering the past and future information, and the *DIFS* value had a fixed length. In Chapter 7 (see section 7.5.3) the dynamic adjustment of *DIFS* led to a considerable reduction in the probability of collisions. Therefore, adjusting both, the *CW* and *DIFS* according to the predicted *CR* and *CRV* values provided the adaptive system (see section 8.3.3) with a sufficient knowledge about the network conditions to appropriately adjust the *CW* and *DIFS* values. The predicted *CR* was used as an indication to adjust the *CW* size; while the predicted *CRV* values was employed as a guide for adjusting the *DIFS* value. A summary of the performance of the *AR* models, IEEE 802.11 DCF, and *EIED* schemes is given in Table 9.1.

**Table 9.1:** Statistical summary of the performance of *AR* models, IEEE 802.11 DCF, and *EIED* schemes.

| No. of connections | Parameters | Schemes | | | |
|---|---|---|---|---|---|
| | | IEEE 802.11DCF | *EIED* | Collision rate prediction | Collision rate and *CRV* prediction |
| 10 / *CBR* | Average delay (msec) | 1788 | 1520.3 | 632.1 | 521.3 |
| | Standard deviation (msec) | 308 | 197 | 96.1 | 53.9 |
| | Average jitter (msec) | 40.1 | 36.3 | 21.3 | 16 |
| | Standard deviation (msec) | 5.1 | 3.5 | 2.2 | 0.6 |
| | Average throughput (Kbps) | 1027.3 | 1089.1 | 1234.8 | 1281 |
| | Standard deviation (Kbps) | 40.3 | 67.4 | 106.1 | 75.9 |
| | Average packet loss (%) | 27.1 | 19.5 | 8 | 4.3 |
| | Standard deviation (%) | 5.5 | 3.6 | 1.7 | 0.5 |
| | Average MAC efficiency (%) | 81.1 | 83.7 | 91.1 | 95.6 |
| | Standard deviation (%) | 1.6 | 1.2 | 1 | 2.2 |
| | Average Collision rate (%) | 17 | 14.9 | 8.9 | 7.1 |
| | Standard deviation (%) | 1 | 0.6 | 0.6 | 1.6 |
| | Average QoS (%) | 11.5 | 13.7 | 57.3 | 62 |
| | Standard deviation (%) | 6.4 | 5.8 | 3.4 | 1.5 |
| 10 / *VBR* | Average delay (msec) | 1497.4 | 1231.1 | 366.9 | 178.1 |
| | Standard deviation (msec) | 415.1 | 265.9 | 103.7 | 109 |
| | Average jitter (msec) | 49.8 | 43.4 | 20.9 | 15.8 |
| | Standard deviation (msec) | 4.6 | 5.7 | 2.3 | 3 |
| | Average throughput (Kbps) | 1130.1 | 1196.7 | 1322.1 | 1352.4 |
| | Standard deviation (Kbps) | 95.8 | 103.8 | 130.3 | 82.3 |
| | Average packet loss (%) | 13 | 7.5 | 1.7 | 0.6 |
| | Standard deviation (%) | 3.3 | 2.4 | 0.9 | 0.6 |
| | Average MAC efficiency (%) | 76.9 | 81 | 90.8 | 95.6 |
| | Standard deviation (%) | 1.4 | 1.6 | 1 | 2.9 |
| | Average Collision rate (%) | 21.3 | 16.4 | 9.1 | 7.5 |
| | Standard deviation (%) | 1.3 | 1.9 | 0.4 | 1.5 |
| | Average QoS (%) | 25.1 | 29.2 | 63.9 | 74.7 |
| | Standard deviation (%) | 8.3 | 10.1 | 4.8 | 5 |

Up to this point, the *AR* prediction models combined with the adaptive *CW* and *DIFS* systems were used to adjust the *CW* and/or *DIFS* based on the predicted *CR*, *CRV*, and

*CW* values. In the next section, the *AR* prediction model is introduced to provide service differentiation in single and multi-hop networks.

### 9.7.3 Quality of Service Differentiation Using Autoregressive Models

In chapter 8, three schemes (*CW* differentiation, *ADIFS* differentiation, and hybrid of *CW* and *DIFS* differentiation) were proposed to provide service differentiation in single-hop networks and one scheme (queue status monitoring scheme) was also suggested to provide service differentiation in multi-hop networks. These four schemes were based on current and previous network conditions. In this section service differentiation is studied when the previous, current and future values of *CR*, *CRV*, and queue status ratio were considered.

#### 9.7.3.1 QoS Provision in Single hop Networks using Autoregressive Modelling

In this section, the adaptive differentiation scheme outlined in section 8.3.3 combined with the proposed *AR* model for predicting *CW* and *DIFS* is presented.

Two scenarios were discussed in this section. In the first scenario, two high and three low priority connections transmitted *CBR* traffic. Each high priority source transmitted 192 Kbps with a packet size equal to 512 bytes. Each low priority source transmitted 480 Kbps with packet size equal to 800 bytes. In the second scenario, 10 connections (5 high priorities and 5 low priorities) were considered. Each high priority source transmitted using 192 Kbps and each low priority source transmitted using 160 Kbps.

Figures 9.13a, 9.13b and 9.13c show the performance of *AR* prediction model for providing service differentiation for 5 connections in single-hop networks. Average delay of high priority traffic was less than 10 msec. This value of average delay was 48% less than the average delay obtained when the adaptive differentiation scheme was used without the prediction system (i.e. average delay equal 17.05 msec). Simultaneously, low priority traffic also had a smaller value for delay with a mean value equal to 656.6 msec. This was 26.9% less than the average delay obtained when the adaptive differentiation scheme was used without the prediction system. These reductions led to noticeable improvements in the QoS for high and low priority traffic.

The distribution of delay and QoS shown in Figures 9.13a and 9.13b indicated that the *AR* model was capable to maintain the average delay for the time-sensitive application as small as possible (i.e. all packets had delay less than 10 msec) and to sustain an excellent QoS level along with the simulation time. In addition to that, the *AR* model also improved the QoS for low priority traffic. As depicted in Figure 9.13c, the average QoS for low priority traffic improved by 22.8% in comparison with the average QoS obtained when the adaptive service differentiation scheme was used without the prediction model.

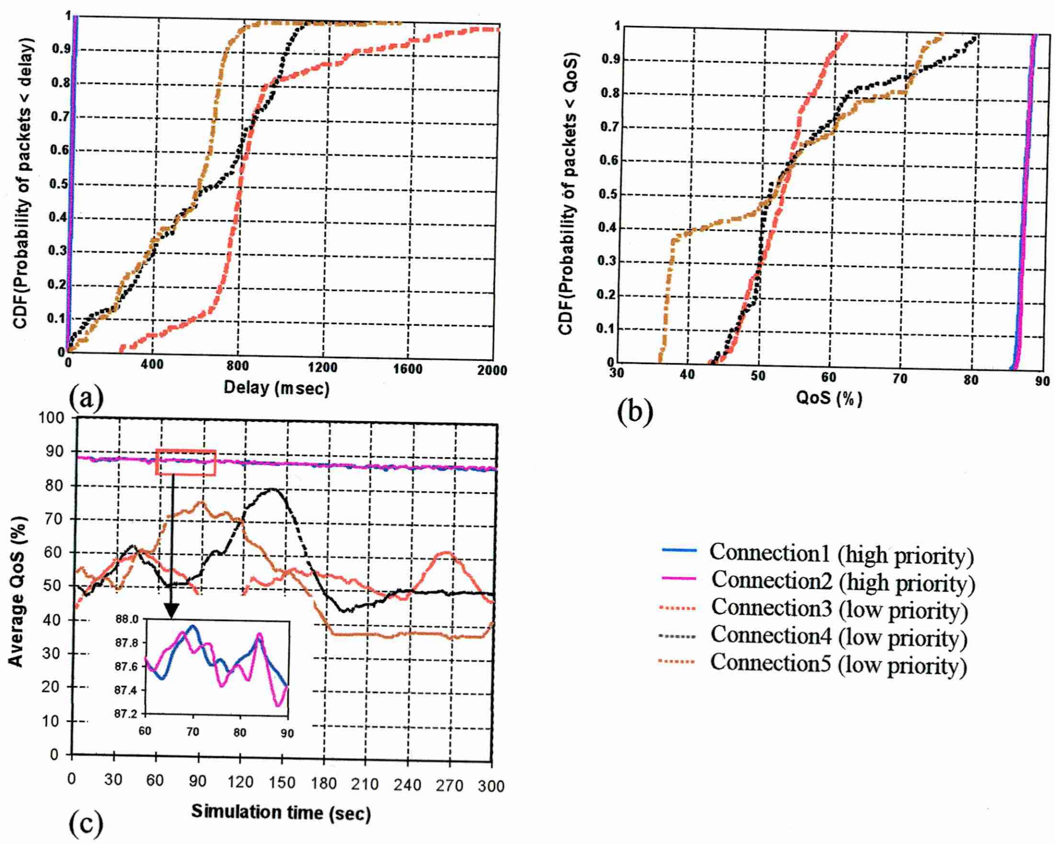



**Figure 9.13:** *AR* model for providing service differentiation in a single-hop network (*CW* and *DIFS* adjustments) with *CBR* traffic in a five connections network, (a) distribution of delay, (b) distribution of QoS, and (c) average QoS.

When 10 connections were considered in the single-hop network, the *AR* prediction model proved its potential by providing service differentiation and improving the whole network performance. As shown in Figure 9.14c the values of QoS for the high and low priority connections during the first 60 seconds of the simulation were different to those when the network was stabilised. This was because during the initial period, more control, routing, and management frames were exchanged between stations in the same *IBSS*. Another cause was that the *AR* model in a large network requires sufficient observations to build up the model and to derive its coefficient parameters. However, initially, these observations were insufficient. Once completed the *AR* model showed stable performance especially for high priority traffic.

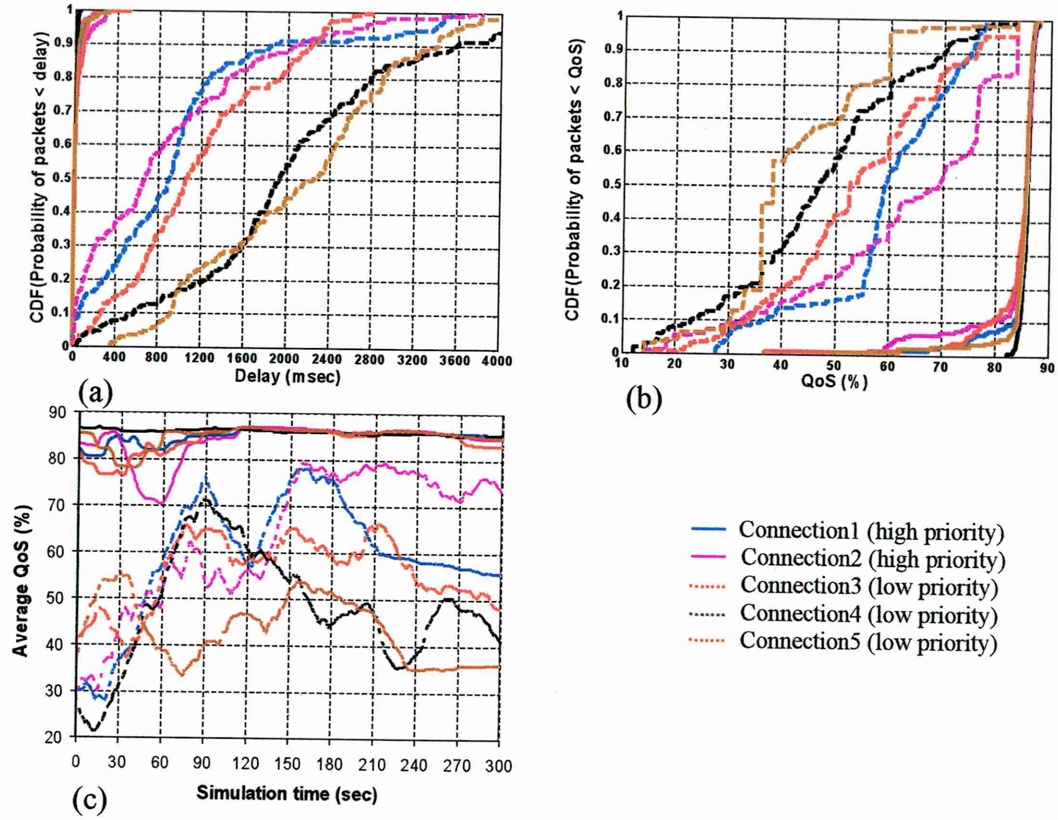



**Figure 9.14:** *AR* model for providing service differentiation in a single-hop network (*CW* and *DIFS* adjustments) with *CBR* traffic in ten connections network, (a) distribution of delay, (b) distribution of QoS, and (c) average QoS.

**Table 9.2**: A summary of quantitative results of the *AR* model for providing service differentiation.

| No. of connections | Application type / priority | Average delay (msec) | Average jitter (msec) | Average throughput (Kbps) | Average MAC efficiency (%) | Average QoS (%) |
|---|---|---|---|---|---|---|
| 5 | connection1 / high | 9.2 | 5.5 | 194.6 | 98.5 | 87.2 |
| | connection 2 / high | 8.8 | 5.4 | 188.6 | 98.8 | 87.3 |
| | connection 3 / low | 843.7 | 10.4 | 275.2 | 99.2 | 52.8 |
| | connection 4 / low | 612.6 | 8.9 | 387.1 | 98.7 | 55.6 |
| | connection 5 / low | 521.1 | 8.4 | 355.1 | 98.8 | 51.1 |
| 10 | connection 1 / high | 30.3 | 8 | 194.7 | 98 | 84.9 |
| | connection 2 / high | 43.3 | 7.7 | 191 | 96.8 | 83.9 |
| | connection 3/ high | 40.8 | 8 | 187.8 | 97.5 | 84.1 |
| | connection 4 / high | 18 | 7.9 | 147.6 | 96.5 | 85.9 |
| | connection 5 / high | 29.3 | 8 | 182.5 | 97.4 | 85 |
| | connection 6 / low | 985.3 | 26.3 | 102.7 | 97.5 | 59.3 |
| | connection 7 / low | 882 | 23 | 118.9 | 97.1 | 63.5 |
| | connection 8 / low | 1082 | 23.8 | 102.2 | 95.9 | 55.8 |
| | connection 9 / low | 2031.3 | 40.6 | 89.4 | 96.3 | 48.2 |
| | connection 10/ low | 2090.2 | 48 | 62.8 | 96.3 | 43.3 |

Figure 9.14a, 9.14b and 9.14c plot the distribution of delay and QoS and the measured QoS for high and low priority classes. As shown in these graphs, the two classes were obviously differentiated. More than 99% of high priority packets had delays less than 200 msec and more than 95% of high priority packets had excellent QoS levels. Low priority packets also maintained a good QoS level for all connections with an average

QoS equal to 53%. Compared with the adaptive service differentiation scheme discussed in Chapter 8, the *AR* model for service differentiation provided noticeable improvements in average QoS for the high and low priority traffic. A summary of the performance of the *AR* model for providing service differentiation in single-hop network for 5 and 10 connections is given in Table 9.2.

#### 9.7.3.2 QoS Provision in Multi-hop Networks Using Autoregressive Modelling

The *AR* model combined with the adaptive differentiation schemes confirmed their capabilities of providing service differentiation and improving the network performance in single-hop networks. This section discusses the ability of providing service differentiation in multi-hop networks when prediction is employed. It considers two cases: (i) using the adaptive QoS differentiation scheme to adjust *CW* and *DIFS* based on the predicted *CR* and *CRV* values. At the same time, the queue status monitoring scheme (see section 8.3.4) was used to adjust the transmission rate in relation with the actual queue status ratio and (ii) employing the adaptive QoS differentiation scheme to adjust *CW* and *DIFS* according to the actual *CR* and *CRV* values. Concurrently, the queue status monitoring scheme was used to adjust the transmission rate based on the predicted queue status ratio. The topology shown in Figure 4.2c (see Chapter 4) was considered, where two high priority and three low priority connections were active. Each station transmitted using 192 Kbps to its correspondent destination passing through more than two hops.

***Case (1)***: The simulation results in Figures 9.15a and 9.15b show the average delay and average QoS for high and low priority traffic. The hybrid system (i.e. a combination of prediction, adaptive service differentiation, and queue status monitoring schemes) clarifies the differentiation between the high and low priority traffic. For instance, high priority traffic had small values of delay less than 400 msec on average. This resulted in a good QoS level with mean value of 35.2%. In contrast, high values of delay and poor QoS were observed for low priority traffic.

In IEEE 802.11 DCF scheme, both high and low priority traffic had the same performance and no service differentiation was observed. Low priority traffic in the hybrid scheme had a similar performance to the high priority traffic when the standard IEEE 802.11 DCF scheme was employed. Therefore, the standard IEEE 802.11 DCF scheme was incapable of providing service differentiation and it had a poor performance compared with the hybrid schemes as shown in Figures 9.16a and 9.16b.

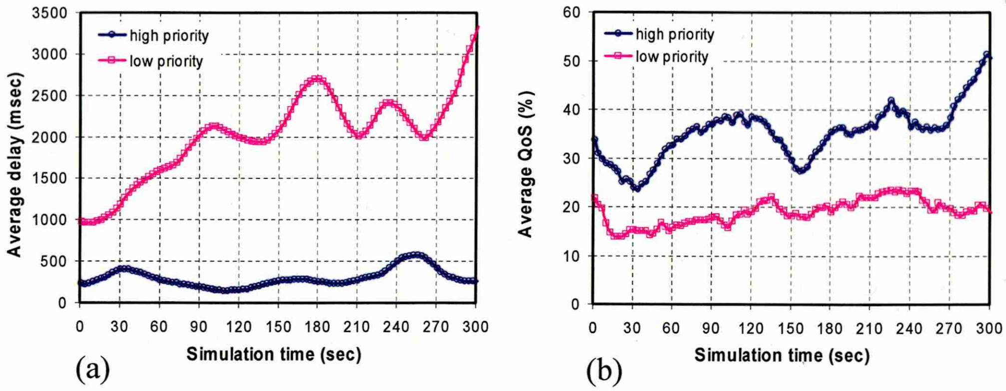



**Figure 9.15:** *AR* model for providing service differentiation in multi-hop network based on the actual queue status ratio and the predicted *CR* and *CRV* values, (a) average delay, and (b) average QoS.

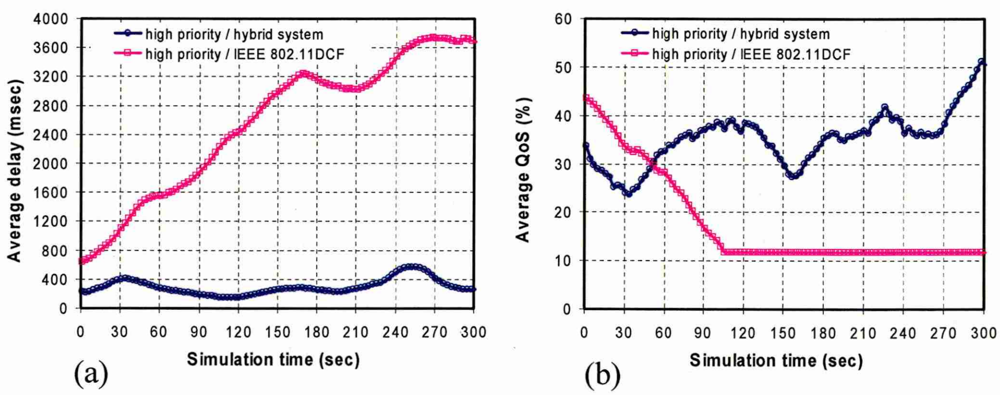



**Figure 9.16:** *AR* model versus IEEE 802.11 DCF scheme for providing service differentiation in multi-hop network. Service differentiation was based on the actual queue status ratio and the predicted *CR* and *CRV* values, (a) average delay, and (b) average QoS.

***Case (2)***: The queue status ratio ( $\rho$ ) (see section 8.3.4) was considered as an indication of traffic congestion in the buffer. Thus, the prediction of the future value of $\rho$ can reduce the potential congestion before it takes place in the future. Generally, this early information improves the network performance. According to Figures 9.17a and 9.17b, the prediction of $\rho$ reduced the average delay by 30.7% and improved the average QoS for high priority traffic by 8.3% compared to the achieved QoS when the transmission rate was adjusted based on the actual or computed queue status ratio ( $\rho$ ). This confirmed that the prediction of the future value of $\rho$ enabled the feedback control scheme to perform an early action before congestion took place, which improved the network performance.

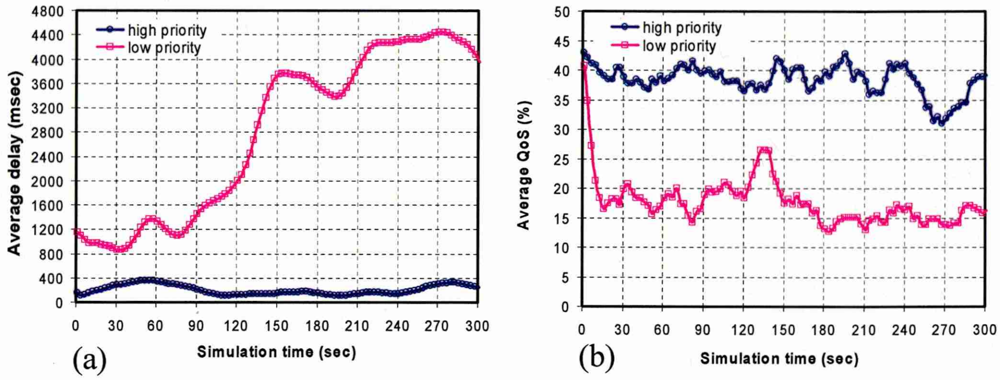



**Figure 9.17:** *AR* model for providing service differentiation in a multi-hop network based on the predicted queue status ratio and the actual *CR* and *CRV* values, (a) average delay, and (b) average QoS.

A summary of the performance of *AR* prediction model is given in Figure 9.18. It can be observed that the *AR* prediction model was capable of providing better performance than the standard IEEE 802.11 DCF scheme. For instance, 90% of high priority packets were successfully received when the prediction scheme was employed. On the other hand, 42% were successfully received when the standard IEEE 802.11 DCF scheme was used. This was due to the ability of the prediction and the control mechanisms to reduce the number of packets dropped at the buffer. Around 2.2% of high priority packets were dropped at the buffer, while 52.4% were dropped when the standard IEEE 802.11 DCF scheme was employed.

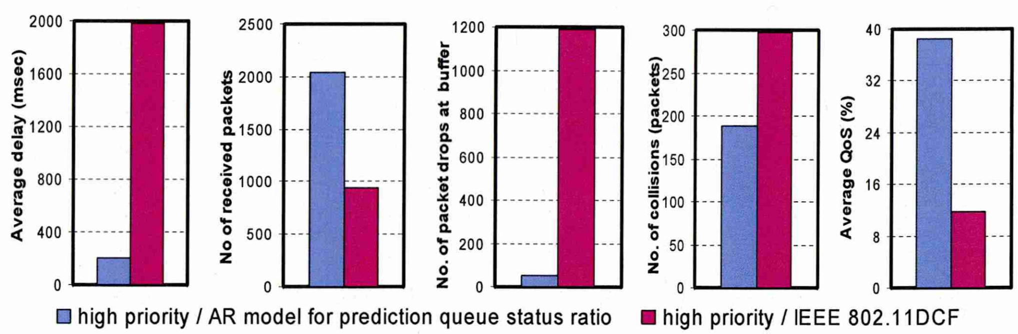



**Figure 9.18:** Performance comparison between *AR* model and IEEE 802.11 DCF scheme for providing service differentiation in a multi-hop network. Service differentiation was based on the predicted queue status ratio and the actual *CR* and *CRV* values.

## 9.8 Summary

In this chapter, an efficient and accurate online *AR* model is proposed to predict the network parameters such as collision rate, collision rate variation, *CW*, and queue status ratio. In this respect, this chapter first presented an introduction to linear regression. In section 9.3 the main linear regression assumptions was outlined. This is then followed by describing the implementation of linear regression in section 9.4. A detailed description of the proposed *AR* prediction models was given in section 9.5. The simulation model for different scenarios to validate the performance of the proposed *AR* models and to compare them with the IEEE 802.11 DCF and the *EIED* schemes was presented in section 9.6. A full description of the main findings is given in section 9.7.

The proposed *AR* model has the potential to improve the network performance and congestion control in IEEE 802.11 DCF scheme. According to the predicted *CR*, *CRV*, and queue status ratio, MAC protocol transmission parameters can be appropriately adjusted to provide service differentiation in single and multi-hop networks. The results also indicated that the developed regression model was capable of providing an online prediction of *CR*, *CRV*, and queue status ratio values with reasonable accuracy and relatively less complexity. Therefore, autoregressive models are easy to implement since they do not require major modifications to the IEEE 802.11 DCF frames format and do not impose extra overhead on the network.

# *CHAPTER 10*

# Discussion and Analysis

## 10.1 Introduction

This chapter presents an overall discussion and analysis of the main findings of this thesis.

The transmission of packets in wireless networks is controlled by IEEE 802.11 MAC protocol which does not consider the QoS for applications (IEEE, 1999). The transmission conditions of wireless networks can change due to variations in the channel conditions. Issues such as limited channel capacity, noise interference, risk of collision and contention between transmitting stations can significantly affect QoS, especially for multimedia applications. Therefore, this study has been concerned with improving the QoS by developing novel MAC mechanisms. This represents an important issue and forms the basis of this study.

## 10.2 Overall Discussion

This section provides an overall discussion of the main findings of this study. This includes the evaluation of the proposed schemes.

### 10.2.1 IEEE 802.11 DCF Performance Evaluations

A detailed evaluation of the limitations of the standard IEEE 802.11 DCF scheme such as the unfairness, hidden terminal, and transmission over multi-hop networks was discussed in Chapter 5. The unfairness of different traffic sources is a significant problem which occurs due to the impact of random backoff in the IEEE 802.11 DCF scheme when one connection can interfere with other connections. As a result, some connections can occupy almost the whole channel bandwidth. This results in a significant degradation in performance of other connections. Changing the transmission rate of the sender demonstrated the impact of this problem on the QoS parameters. The sender with high transmission rate captured the channel and achieved better throughput than the sender with low transmission rate.

The presence of the hidden terminal problem significantly affected the QoS when transmitting time-sensitive applications. Video, audio and data transmissions experienced high degradations. E.g., high values of delay for video transmission were observed, and a high packet loss rate that exceeded 32% for data transmission was obtained. These values prevented the desired QoS to be met for these applications. Increasing the number of hidden terminals made the situation worse and degraded the whole network performance.

The intermediate station in the IEEE 802.11 DCF scheme was incapable of forwarding the received data packets with high QoS in heavily loaded networks. As a result, a large reduction in average throughput, an increase in average delay and a packet loss rate were observed. This confirmed that the backoff algorithm of the IEEE 802.11 scheme performs inadequately in multi-hop networks.

The performance of the IEEE 802.11 DCF scheme was also affected by increasing the number of active stations, varying the data rate, and changing the protocol operation mode. Increasing the number of active stations increased the competition among stations which ultimately boosted the number of collisions and degraded the average throughput. This process depended on the type of access mechanism used. For instance, the basic access mechanism was significantly affected by an increase in the number of sources compared to the *RTS/CTS* access mechanism. This is because in the basic access mechanism there is an increase in collisions for data packets as the number of sources increases.

Increasing the data rate from 2 Mbps to 11 Mbps provided a larger channel bandwidth. This in turn improved the average throughput of the transmitted applications but the channel utilisation degraded. High data rates resulted in smaller values of average delay and jitter for transmitting *CBR* and *VBR* traffic and this allowed the desired QoS to be achieved. At high data rates, the average delay reduced by 69% and 66% for *CBR* and *VBR* traffic respectively, when the basic access mechanism was used and they were reduced by 58% and 63% when the *RTS/CTS* access mechanism was used.

All the aforementioned limitations are related to the operation of the IEEE 802.11 MAC protocol. Therefore, improving the protocol performance through developing novel MAC mechanisms is an important objective.

### 10.2.2 Impact of MAC Protocol Transmission Parameters

Initially, an investigation was performed to study the impact of varying the MAC protocol transmission parameters such as $CW_{min}$ and *DIFS* on the network performance. This provided a general overview concerning the relationship between $CW_{min}$ and *DIFS* parameters and the network performance under different network configurations.

The investigations showed that the variation in the $CW_{min}$ value had a significant impact on the network performance. High values of $CW_{min}$ caused long delays for data packets, where small values caused a large number of collisions. Thus, an inappropriate selection of the $CW_{min}$ size can lead to a large number of collisions and a large packet drop rate at the buffer. The results also showed that the optimal $CW_{min}$ size differed depending on the network conditions. For instance, a very small $CW_{min}$ size (less than 31) was not effective for large networks and large packet sizes due to the increased number of collisions. Conversely, a large $CW_{min}$ size (greater than 127) was inappropriate for small networks and small packet sizes due to many idle slots.

The impact of the *DIFS* parameter was also examined. Small values of *DIFS* reduced the values of delay and jitter; however, they still resulted in a high number of collisions particularly in large networks (i.e. 20 connections). Large values of *DIFS* (e.g., 5 slots) reduced the probability of collisions, causing high drops at the buffer, increased the total packet loss rate, degraded throughput, and increased the values of delay (e.g., more than 57%). Consequently, small and large values of *DIFS* provide undesirable performance which lead to QoS degradation. In order to alleviate these limitations and to maintain a satisfactory performance, the *CW* size and *DIFS* values have to be dynamically adjusted to adapt the network condition variations and to meet the application QoS requirements.

### 10.2.3 Developing New MAC Mechanisms for QoS Provisioning

Fuzzy logic, genetic algorithm, Ratio based, *CRV*, queue status monitoring scheme, adaptive service differentiation, and prediction approaches were proposed to develop new MAC mechanisms to provide QoS. The results of the study based on these proposals are discussed below.

#### 10.2.3.1 Evaluation of the AI Techniques

In Chapter 6, the application of *AI* techniques such as fuzzy logic and genetic algorithms for assessing QoS and adjusting MAC protocol transmission parameters when transmitting various applications was discussed. The fuzzy assessment approach was capable of combining multiple QoS parameters such as delay, jitter, and packet loss to provide QoS. The accuracy of the *FIS* approach was tested using typical values of audio, video, and data QoS parameters as indicated in Table 6.1 (see Chapter 6). For each case the *FIS* system accurately determined the percentage QoS as well as the corresponding QoS level. Additionally, the *FIS* system was also able to adjust the $CW_{min}$ value according to the assessed QoS, collisions, and the previous $CW_{min}$ size parameters. The *FIS* adjustment system showed its effectiveness in adjusting the $CW_{min}$ size and in providing satisfactory QoS for the transmitted applications. For instance, when multiple video and multiple audio were transmitted, the QoS for the applications improved and a fair access to the medium was achieved. Average QoS for video connections using fuzzy logic was 31.5% higher than the QoS achieved when the standard IEEE 802.11 DCF scheme was used. Similarly, high average QoS values were obtained for audio connections. A 59.5% improvement of average QoS for audio connections was observed using the *FIS* approach compared to the IEEE 802.11 DCF. Therefore, the *FIS* adjustment system had the potential to maintain good QoS levels (i.e. between 34% and 66%) for video connections, and excellent QoS levels (i.e. between 67% and 100%) for audio connections for most of the examined configurations (i.e., small and large number of connections).

A genetic algorithm was used to adjust the $CW_{min}$ and the *DIFS* values using different network configurations (i.e. small, medium and large networks). The hybrid genetic-fuzzy approach provided improvements in the network performance. Up to 50% improvement in the measured QoS was obtained compared with the one that obtained when the standard IEEE 802.11 DCF scheme was used. It was also observed that the hybrid genetic-fuzzy approach reduced the probability of collisions and fairly distributed the channel among the contending stations. The key issue of these improvements is that the range of the $CW_{min}$ and *DIFS* values and the fitness function are selected appropriately. The fitness function is chosen to avoid starvation and to maintain a satisfactory QoS level for all connections.

In Chapter 6, the linear adjustment of the $CW_{min}$ was also investigated in order to examine the possibility of using simple linear schemes to improve the protocol performance. Using a simple linear scheme, noticeable and comparable enhancements to the average QoS for video and audio traffic were observed compared with the *AI* techniques. Therefore, the simple linear mechanism combined with the *FIS* assessment mechanism were used for developing the Ratio based, Collision Rate Variation (*CRV*), *CW*-based differentiation, *DIFS*-based differentiation, adaptive service differentiation, queue status monitoring and prediction approaches. This simple linear mechanism enables dynamic and adaptive adjustment with less computational power when compared with *AI* techniques.

#### 10.2.3.2 Evaluation of Ratio Based and CRV Schemes

Ratio based and *CRV* schemes were proposed to reduce the probability of collisions by dynamically adjusting the $CW_{min}$ and *DIFS* values according to the current and past network condition variations. Extensive simulations of different configurations indicated that the two schemes outperformed the IEEE 802.11 DCF and the *EIED* schemes. For instance, in a large and heavily loaded network, the average QoS improved by 54.4% compared to the standard IEEE 802.11 DCF and *EIED* schemes. These improvements were due to significant reductions in delay, jitter, packet loss, and collision rate values of the transmitted applications. Ratio based and *CRV* schemes were proposed for the basic access mode since data packets were involved in collisions. When they performed for the *RTS/CTS* access mode, no significant impact on their performance was observed. Using the proposed scheme with the *RTS/CTS* access mode improved the QoS by 37.5% and 29.1% compared to the IEEE 802.11 DCF and *EIED* schemes, respectively.

The ratio based scheme was also examined for multi-hop networks. Only minor improvements were observed (less than 10%) using this approach. This was due to the incapability of the intermediary stations to forward the received packets from different sources. As a consequence, a large number of packets were dropped at the queues of both the data source and the intermediate stations. Consequently, the Ratio based and *CRV* schemes were slightly modified and were combined with the queue status monitoring scheme to further improve the protocol performance and to provide service differentiation in a single-hop and multi-hop networks.

#### 10.2.3.3 Quality of Service Provisioning in IEEE 802.11 DCF Scheme

The proposed service differentiation framework was based on four schemes: *CW*-based differentiation, *ADIFS*-based differentiation, adaptive service differentiation, and queue status monitoring schemes. These schemes were validated for different scenarios. In *CW*-based differentiation scheme, high priority stations achieved excellent QoS levels with mean QoS of 75%. The *CW*-based differentiation scheme also maintained a good QoS for low priority connections with an average of 36.6% in a large and heavily loaded network. Although, the *CW*-based differentiation scheme provided service differentiation, it resulted in noticeable fluctuations in the QoS curves of the two priorities. This was due to the overlap in *CW* values between the high and low priorities. The *ADIFS*-based differentiation scheme provided an excellent QoS for the high priority connections in small and large networks. It showed fewer fluctuations in the QoS curves compared to the *CW*-based differentiation scheme. *ADIFS*-based differentiation scheme resulted in a poor QoS for some low priority connections. Therefore, to overcome these minor drawbacks for each scheme they were combined producing a hybrid differentiation scheme. The hybrid differentiation scheme was also validated for different scenarios. For small networks, high priority connections achieved an excellent QoS with mean value of 86%, and low priority connections also maintained a good QoS with mean value of 42%. For a large network, the adaptive scheme provided mean QoS equal to 84.3% and 53% for high and low priority connections, respectively. When the network operated at light load conditions, the adaptive service differentiation scheme narrowed the gap of differentiation between high and low classes to utilise all the available channel bandwidth. However, it showed a wide gap of differentiation between classes when the network was heavily loaded to protect the high priority and to offer it more advantages of accessing the channel. Compared to IEEE 802.11 DCF scheme, the adaptive differentiation scheme in single-hop networks outperformed the IEEE 802.11 DCF scheme and it also showed that the standard IEEE 802.11 DCF scheme was incapable of providing service differentiation.

The proposed queue status monitoring scheme was combined with the adaptive service differentiation (i.e. new hybrid differentiation scheme) to improve the network performance and to provide service differentiation in multi-hop networks. The new hybrid scheme provided significant improvements for high and low priority connections. For instance, average QoS for high priority connections improved by 53% compared to the IEEE 802.11 DCF scheme. Moreover, using multi-hop networks and the standard IEEE 802.11 DCF scheme, the intermediate stations were incapable of accommodating the received packets in its single queue. Therefore, the new hybrid scheme was examined as a function of queue size. The simulation results showed that the new hybrid scheme improved the average QoS by 71.2% compared with the standard IEEE 802.11 DCF scheme.

#### 10.2.3.4 Evaluation of AR Online Models

Autoregressive models were employed to predict collision rate, collision rate variation, contention window, and queue status ratio. The predicted values were used to adjust *CW*, *DIFS*, and the transmission rate. The advantage of the prediction models was that they provided future knowledge about the network conditions. This knowledge enables the protocol to appropriately adjust its transmission parameters. The results obtained showed that the *AR* models provided better performance than the Ratio based and *CRV* schemes. For instance, in a medium network, the average QoS using prediction was 81.5%, 78%, 7%, and 4.2% higher than the average QoS obtained when the IEEE 802.11 DCF, *EIED*, Ratio based, and *CRV* schemes were used, respectively. Furthermore, the *AR* prediction models were employed for providing service differentiation in single and multi-hop networks. In single hop networks, high priority connections preserved an excellent QoS with mean values greater than 85% in small and large networks. For multi-hop networks, the prediction models resulted in remarkable improvements compared to the standard IEEE 802.11 DCF scheme. It was also shown that the developed regression models were accurate in predicting the behaviour of network conditions for various configurations.

T-test was carried out to perform statistical comparison of the average QoS obtained by the new methods (Ratio based, *CRV*, and *AR* prediction schemes), the basic IEEE 802.11 DCF, and *EIED* schemes (t-test, 2006). Using the t-test, a significant statistical difference between the considered methods was observed. More information is provided in Appendix G.

# *CHAPTER 11*

# Conclusions and Future Work

## 11.1 Conclusions

In this thesis extensive simulations were carried out using *NS-2* simulation software to investigate the limitations and the performance of the IEEE 802.11 MAC protocol. The performance of the developed new MAC mechanisms for QoS provision and differentiation was also validated and compared with the IEEE 802.11 DCF and *EIED* schemes for multiple QoS parameters (delay, jitter, throughput, packet loss, collision).

The results confirmed that the unfairness, hidden terminals and the transmission over multi-hop networks significantly affected the performance of the IEEE 802.11 DCF scheme. This confirmed that the random backoff algorithm poorly operates in such environments. The presence of these problems led to a performance degradation and a significant starvation. Applying a simple priority scheme according to $CW_{min}$ provided improved access among the active connections. In multi-hop networks, the intermediary station caused a bottleneck in the network. It was incapable of dealing with the data packets from different sources. This increased data packet drops at the buffer. Therefore, a large reduction in average throughput, an increase in average delay, and a high packet loss rate were produced and prohibited QoS to be achieved for the transmitted applications. Additionally, the results revealed that the stations' capability to send their data packets was affected by the amount of competition they experienced and the interference from other stations. This was because that the IEEE 802.11 MAC protocol allocated the channel bandwidth unequally between the competing stations for both the basic access and *RTS/CTS* access mechanisms. However, the performance of the basic access mechanism was affected by the increase in the number of active stations. A robust performance was obvious for the *RTS/CTS* access mechanism.

Varying the values of MAC protocol transmission parameters such $CW_{min}$ and *DIFS* affected the network performance. Small values of $CW_{min}$ improved the performance in small network size; however, they were still ineffective for large networks. Conversely, large $CW_{min}$ size was valuable for large networks; yet, they were undesirable for small networks. Similarly, small values of *DIFS* provided small values of delay and jitter; however, they led to high probability of collisions. Therefore, slight improvements were observed for small networks. Although, large values of *DIFS* reduced the collision, they increased the number of idle slots which degraded throughput and increased delay. Therefore, there were optimal $CW_{min}$ and *DIFS* values in which any deviation above or below caused degradation in the network performance. Therefore, the findings revealed that the transmission parameters of the standard IEEE 802.11 MAC protocol required dynamic adaptation to improve its performance. For this purpose new MAC mechanisms were developed.

***Fuzzy logic mechanisms***: Fuzzy Inference System (*FIS*) for assessing QoS and *FIS* for adjusting the $CW_{min}$ were developed. The developed *FIS* assessment system provided an effective mechanism for assessing QoS for multimedia applications such as audio, video, and data. The QoS assessment was based on combining QoS parameters (delay, jitter, and packet loss) using fuzzy inference system. The assessed QoS was a good indication of the network conditions and the resource availability. It was also used as a main metric for the performance evaluation process. Further, the developed *FIS* assessment method showed that the current IEEE 802.11 DCF scheme was incapable of providing the minimum QoS requirements for multimedia transmission. The study demonstrated that the application of the *FIS* system to adjust $CW_{min}$ size significantly improved the QoS for audio, video, and data applications. Average QoS for video and audio applications improved by 34% and 59.6%, respectively. The results also indicated that the improvement in QoS was achieved by enhancing channel utilisation. A proposal for implementing the *FIS* system in real networks has been provided. Using a systematic sampling method showed that there was no statistical significant difference between the actual data and the sampled version.

***A hybrid Genetic-Fuzzy mechanism***: The hybrid genetic-fuzzy system was also used to optimise the $CW_{min}$ and *DIFS* values. The hybrid genetic-fuzzy approach was capable of achieving closer solutions of $CW_{min}$ and *DIFS* values. These optimal values resulted in a significant improvement in the QoS for the multimedia applications. For large networks, the average QoS improved by 44.9%, 69.2%, and 55.6% for video, audio and data traffic, respectively.

***Ratio based and Collision Rate Variation (CRV) mechanisms***: The Ratio based and *CRV* schemes extended the legacy IEEE 802.11 DCF mechanism by dynamically adjusting the *CW* and *DIFS* values for each station according to the current and past history of successful and unsuccessful packet transmissions. The simulations indicated that the Ratio based and *CRV* schemes significantly reduced the collision rate and average delay values and improved the QoS for a large and heavily loaded network. The average delay reduced by 59% and 56% as compared with the standard IEEE 802.11 DCF and *EIED* schemes, respectively. The *CRV* scheme performed better than the Ratio based, the IEEE 802.11 DCF, and the *EIED* schemes in several scenarios. For instance, average QoS was 14% higher than the Ratio based scheme, 56% higher than the IEEE 802.11 DCF scheme, and 53% higher than the *EIED* scheme.

***Service Differentiation mechanisms***: The results indicated that using *CW* for service differentiation improves the performance of high and low priority classes compared to the standard IEEE 802.11 DCF scheme. However, the high priority traffic experienced performance degradation from burst transmission of low priority traffic. This occurred due to the contention windows overlap among high and low priority classes. Using *ADIFS*-based differentiation scheme can also improve the network performance and provide an effective service differentiation in which high priority can get strict QoS. The *ADIFS*-based differentiation scheme showed more priority effect and gave a more stable system than *CW*-based differentiation scheme. However, the *ADIFS* scheme sometimes led to performance degradation of low priority classes in which starvation might occur. Therefore, the combined adaptive service differentiation scheme demonstrated the drawbacks of *CW* and *ADIFS* differentiation schemes. The results confirmed that the adaptive service differentiation scheme was capable of providing service differentiation and improving the network performance. The adaptive differentiation scheme was also capable of providing improved service differentiation when the network operated in normal condition (i.e. light load) and a considerable differentiation when the network became heavily loaded. Using the queue status monitoring technique protected high priority traffic from the impact of low priority traffic. It also improved the network performance, reduced the packet drops at the buffer and improved the average QoS for high priority traffic by 53% compared to the standard IEEE 802.11 DCF scheme. The results revealed that the standard IEEE 802.11 DCF scheme was unable to meet the QoS requirements for time-sensitive applications and to provide QoS differentiation in single and multi-hop networks.

***Autoregressive prediction models***: The proposed *AR* models were sufficient for modelling and prediction of the time sequence values of collision ratio, *CRV*, *CW*, and queue status ratio in MAC sub-layer for the IEEE 802.11 protocol. The *CR*, *CRV*, *CW*, and queue status ratio were consistently predictable because of their related correlation between the past, present and future values. The *AR* technique was capable of modelling and predicting the future values of these parameters that were used to adjust the MAC protocol transmission parameters such as *CW*, *DIFS*, and transmission rate. The proposed method resulted in better network performance and less congestion particularly at heavily loaded network conditions. Average QoS improved by more than 60 %, average delay and packet loss reduced by 58%, and collision decreased by more than 50% compared to the standard IEEE 802.11 DCF and *EIED* schemes. The results indicated that using the prediction system to adjust both *CW* and *DIFS* values improved the performance more than only adjusting one MAC transmission parameter. The *AR* model was capable of improving the performance and providing service differentiation. Using prediction, the average QoS improved by 22.8% compared to the average QoS obtained when service differentiation relied on the computed *CW* and *DIFS* values. The effectiveness of the prediction model in providing accurate estimation of the queue status ratio resulted in an improvement equal to 8.3% in the average QoS for high priority traffic compared to the actual queue status ratio.

The main characteristics of the proposed schemes can be summarised as follows:

- **Multiple QoS Parameters**: Multiple QoS metrics (delay, jitter, throughput, packet loss and collision) were considered according to the application type. They were combined through using the *FIS* system to provide one output (assessed QoS).
- **Independent Operation**: All schemes are implemented in a distributed way without requiring any information from other stations. This implies that each station adjusts its parameters independently to achieve the required QoS according to the past, current and future network condition variations.
- **Simplicity and Less Overhead**: The proposed schemes do not impose major changes to the structure of the standard. This enables ease of implementation on the top of the existing standard. Additionally, no extra fields in the frame headers are required. This ultimately reduces any additional overhead.
- **Robustness and Scalability**: The simulation of different scenarios revealed the robustness, scalability, fairness and the satisfactory operation of the proposed schemes. Scalability was examined in terms of loads and the number of

connections. Robustness was investigated with respect to types of traffic (*CBR* and *VBR*) and protocol access mechanism (basic access and *RTS/CTS* access).

In summary, the IEEE 802.11 DCF protocol had limitations when transmitting various applications due to the limitations inherent in its operation. Furthermore, the study confirmed that the application of artificial intelligence techniques for assessing and improving the performance of the IEEE 802.11 MAC protocol when transmitting various applications was effective. The appropriate adjustments of MAC protocol parameters such as *CW* and *DIFS* using the developed adaptive schemes showed significant improvements and service differentiation for single-hop networks. When the queue status monitoring technique was used for the intermediate stations a significant improvement and QoS differentiation at MAC layer was obtained for multi-hop networks. Furthermore, an effective online *AR* models have been proposed. They had the capability of providing an accurate prediction of the network parameters for adjusting the MAC protocol transmission parameters and for efficient congestion control. The new schemes were capable of providing a comprehensive solution to overcome the limitations and the inadequacy of the standard IEEE 802.11 DCF scheme in providing QoS for multimedia transmission. Moreover, they were easy to calculate in real time and simple to implement in wireless stations. Through the use of these approaches, the findings of this study provide a framework that contributes to knowledge concerning the QoS over the IEEE 802.11 MAC protocol.

## 11.2 Future Work

This research effort has extended the operation and the boundaries of knowledge within the IEEE 802.11 DCF scheme areas. The enhancements and expansions of the developed schemes are worth mentioning, extensions of this work may provide more advantages. It is recommended that the following research areas to be carried out.

- ***Implementation of the proposed approaches in physical networks***: The next step to validate the performance of the proposed approach beyond simulation is its implementation in real networks (i.e., physical network). Positive results from real experiments will confirm the effectiveness and the robustness of the proposed schemes.
- ***Implementation of the proposed approaches in the enhanced version of the standard IEEE 802.11 MAC protocol:*** The enhanced version of the standard IEEE 802.11 MAC protocol (i.e. IEEE 802.11e *EDCF*) is not finalised yet.

Therefore, implementing the proposed schemes for the IEEE 802.11e *EDCF* version can provide a guaranteed QoS for multimedia transmission and a relative QoS for time-insensitive applications. Moreover, using the *FIS* system as a scheduling mechanism to control access between the Access Categories (*AC*) and stations can provide an indication of the QoS, per flow differentiation and per station differentiation.

- ***Implementation of the proposed FIS approaches in a Network Interface Card (NIC)***: Another area of future work is to investigate how fuzzy inference system approach can be used to improve the performance when multiple MAC protocol transmission parameters were considered. Furthermore, the proposed *FIS* can be implemented in hardware as a System-on-Chip (*SoC*) which would add a new feature to the wireless card manufactures to include both QoS and a control mechanism in MAC layer.
- ***Extended the domain of genetic algorithm***: The hybrid genetic-fuzzy approach can be conducted for many configurations, and hence all the optimal parameters can be stored in a database. The selected configurations and the optimal parameters can be trained using neural networks and hence can be used for online MAC protocol optimisation parameters.
- ***Incorporating Call Admission Control (CAC) mechanism***: In order to provide strict and absolute QoS for time-sensitive applications and to mitigate starvation for time-insensitive applications a Call Admission Control (*CAC*) mechanism combined with the adaptive service differentiation and queue status monitoring schemes discussed in this study can be introduced as future work. Time-sensitive applications can reserve the required channel bandwidth and determine the delay limit, while time-insensitive applications will be controlled by *CAC* mechanism with the outstanding network resources. Some schemes can reserve large part of the channel capacity for time-sensitive applications, however too many unsuccessful data transmission can degrade the transmission of time-sensitive applications. In this case, these time-sensitive applications become vulnerable to the impact of data traffic. This implies that priority based schemes can provide relative QoS and sometimes absolute QoS with some limitations. Accordingly, a control mechanism for data transmission is required to protect multimedia flows. To control data transmission, the most effective method is to reduce the probability of collisions caused by data transmission (Xiao, 2006). Because it is very difficult and unrealistic to get an accurate number of active

stations for data transmission a *CAC* scheme can be proposed and implemented. The interaction of this mechanism with the discussed schemes (adaptive service differentiation and queue status monitoring scheme), strict QoS for time-sensitive applications can cause further improvements in single and multi-hop networks and fairness can be provided for time-insensitive applications.

- ***Bandwidth allocation mechanisms:*** In order to provide QoS guarantee for multimedia transmissions and a satisfactory performance for low priority traffic, different bandwidth allocation schemes need to be designed. This will protect the high priority traffic (i.e. video and audio) from the impact of low priority traffic (i.e. data traffic), while simultaneously providing fair performance for the low priority traffic.

The study contributed significantly toward better understanding and assessing of QoS in wireless ad-hoc networks. The developed methods have provided significant improvements to the existing standards for medium access control. The field of study is expanding rapidly and there is a significant scope for a variety of challenging research studies.

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# APPENDICES

# Appendix A

## A.1 Network Simulator 2 (NS-2)

*NS-2* provides a framework for building a network model, specifying data input and analysing output data. It is a discrete event simulator targeted at networking research (NS, 2006). It is a widely used simulation tool for simulating inter-network topologies to test and evaluate various networking protocols. Since *NS-2* is an open source and freely available simulation tool that runs on different platforms such as Linux and windows. In addition, *NS-2* is also a useful tool because it supports a large number of network components and it can be extended either by modifying the *OTcl* or *C++* code. In order to add a new component in *NS-2*, modification on *C++* code is the most efficient manner to do it, since *C++* code represent the core of the simulator.

As stated, *NS-2* is written in an object oriented language such as *C++*, with an Object Tool Command Language (*OTcl*) interpreter. The simulator supports a class hierarchy in *C++*, and a similar class hierarchy within the *OTcl* interpreter. The two hierarchies are closely related to each other; from the user's view, there is a one-to-one correspondence between a class in the interpreted hierarchy and one in the compiled hierarchy as presented in Figure A.1.

In Figure A.1, the user generates the Tool Command Language (*TCL*) script with wireless nodes, traffic applications, communication pair, and all the required settings. The second step is the selection of the required parameters that are going to be traced during the simulation. During the compilation of the *TCL* script, all the simulation settings are handled by the *C++* libraries and the *OTcl* interpreter as the main step. Finally, the simulation results appear in the trace file which can be parsed and analysed for a variety of parameters that need to be measured.

Users can define arbitrary network topologies composed of nodes, routers, links and shared media. A rich set of protocol objects can then be attached to nodes such as Transmission Control Protocol (*TCP*) and User Datagram Protocol (*UDP*).

In *NS-2*, the physical layer characteristics such as data rate, delay, antenna and the wireless physical interface parameters can be defined. It also offers noticeable support for simulating wireless networks and interconnecting wired and wireless networks. In addition, it supports mobile networking models developed by CMU/Monarch group that allows simulations of multi-hop ad-hoc networks and *WLAN*s (Monarch, 2005).

The simulation tool supports trace file that used to trace and analyse the packets for both wireless and wired networks. Furthermore, the simulator supports a graphical tool for visualization of simulation results called Network Animator (*NAM*) to assist the users get more insights about their simulation (ETSI, 2000).

IEEE 802.11 DCF function is implemented within the simulation tool (Nee, 1999). The MAC sub-layer handles fragmentation, collision, acknowledgements and detection errors. The two MAC protocol mechanisms, the basic access mechanism and the *RTS/CTS* access mechanism are also implemented and supported by the *NS-2* simulation tool. Full details about the general architecture of the network components in *NS-2* can be found in the documentations supported by the *NS-2* group (NS, 2006).

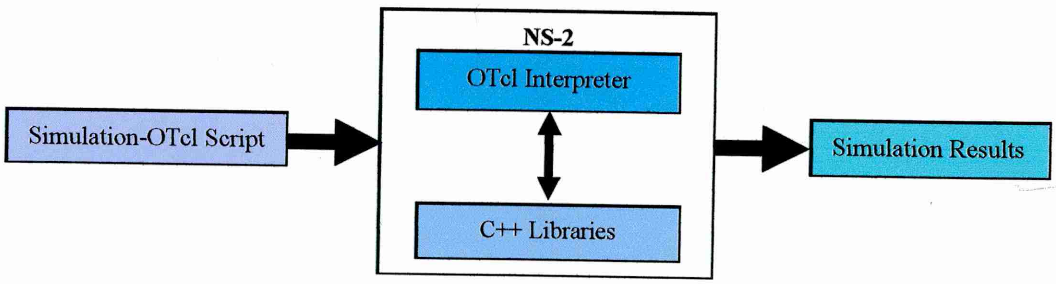


**Figure A.1:** Simulation process

# Appendix B

## B.1 Unfairness Problem with *VBR* Traffic

The simulation was carried out when transmitting *VBR* traffic. The network of Figure 5.1 (see Chapter 5) was employed. Two *UDP* source-destination connections were setup. These were labelled as 0 and 1 for the first connection, and 3 and 2 for the second connection. The sources were associated with *VBR* traffic. The sources were out of the transmission range of each other. The destinations were located within the transmission range of each other. The sources generated same rates and they offered the network with 50% of the channel capacity (i.e. each source transmitted 500 Kbps without the control frames overhead). The simulation time was 300 seconds and other simulation settings are as provided in Table 4.1 (see Chapter 4).

In this experiment, the first connection started its transmission during the first second of the simulation. The second connection commenced its transmission during the third second of the simulation in order to reduce the impact of the management and routing frames at the beginning of the simulation. When the basic access mechanism was used, the first connection achieved 0.42 Mbps of the average throughput. This is 5% larger than the average throughput achieved by the second connection as shown in Figure B.1a. For the *RTS/CTS* access, average throughput of the first connection was 11% larger than the average throughput achieved by the second connection as shown in Figure B.1a.

With regard to average delay and jitter, the second connection was deferred for a long period by the transmission of the first connection since the latter captured the channel for continuous transmissions. This resulted in a high average delay and jitter for the second connection in both MAC protocol access mechanisms. The average delay of the second connection was 53% larger than average delay of the first connection when the basic access mechanism was used and 8% larger when the *RTS/CTS* access mechanism was employed as shown in Figure B.1b.

Average jitter of the second connection was also higher than that of the first connection by 21% and 4% for the basic and the *RTS/CTS* access mechanisms, respectively as shown in Figure B.1c. The high value of average jitter for the second connection was due to collisions. For the basic access, collisions occurred for data packets. The MAC protocol as a result of lack of acknowledgement retransmitted these collided packets.

The retransmission of these collided packets caused a high variation in time of the successfully received packets at the destinations which increased the values of jitter. This variation depended on the number of packet retransmissions. When the *RTS/CTS* access mechanism was used, collisions occurred in *RTS* and *CTS* control frames, and there were no drops in data packets. As a result, most of the received packets at the destination had less time variation resulting in a smaller jitter as shown in Figure B.1c.

Packet loss occurs due to many reasons including transmission errors, route failure (no route to the destination), broken links, congestion and collisions. Congestion in a network occurs whenever the demands or the traffic exceed the channel capacity. Collision in a wireless networks occurs whenever two or more stations start transmission simultaneously. Data packets drops were mainly due to collisions, buffer overflow, and exceeding number of retries by the MAC protocol. More than 22% of data packets were dropped in the second connection when the basic access mechanism was used. Less packet drops in the second connection (i.e., only 9%) were observed when the *RTS/CTS* access mechanism was employed. This was due to long defer (unfairness MAC protocol) by the transmission of the first connection. The simultaneous transmission of the two connections caused large drops of data packets due to collisions in both connections. For the *RTS/CTS* access mechanism, the exchange of the *RTS* and *CTS* control frames resulted in large drops of data packet at the buffer.

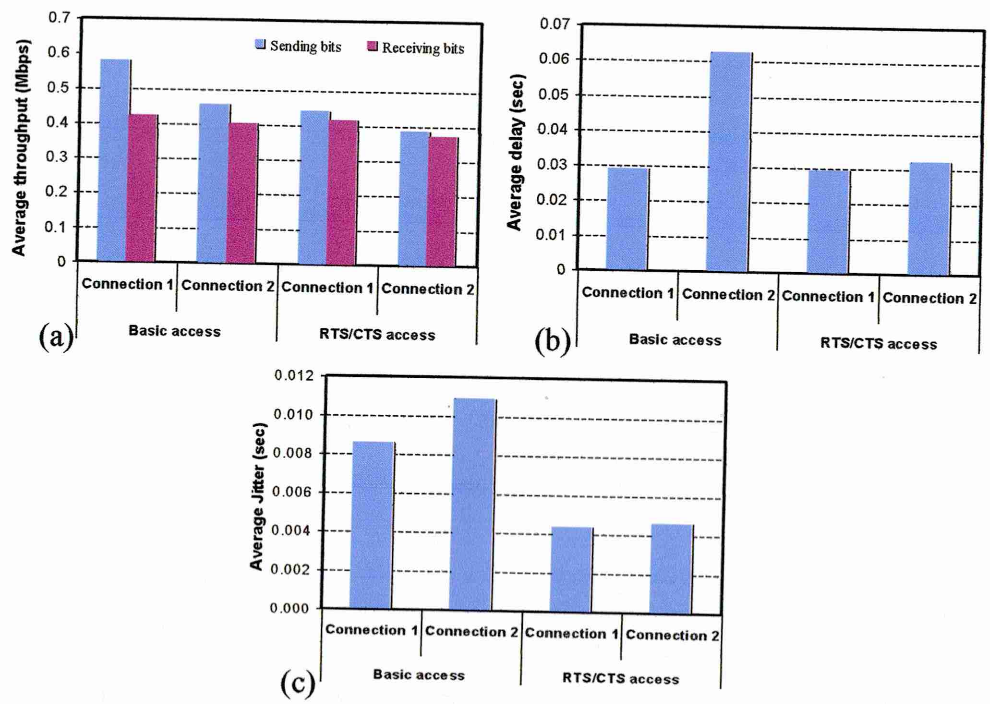


**Figure B.1:** Unfairness problem with *VBR* traffic for the basic and *RTS/CTS* access mechanisms, (a) average throughput for sending and receiving stations, (b) average delay, and (c) average jitter.

## B.2 Impact of $CW_{min}$ on the QoS Parameters for *VBR* traffic

**Table B.1:** Impact of $CW_{min}$ on the QoS parameters for four connections at different packet sizes.

| $CW_{min}$ | Average throughput at 500 bytes (Mbps) | Average throughput at 900 bytes (Mbps) | Average throughput at 1400 bytes (Mbps) | Average delay at 500 bytes (sec) | Average delay at 900 bytes (sec) |
|---|---|---|---|---|---|
| 16 | 1.2 | 1.4 | 1.4 | 0.67 | 0.87 |
| 32 | 1.3 | 1.4 | 1.5 | 0.55 | 0.83 |
| 48 | 1.3 | 1.5 | 1.5 | 0.56 | 0.86 |
| 64 | 1.3 | 1.5 | 1.5 | 0.56 | 0.85 |
| 80 | 1.3 | 1.5 | 1.5 | 0.55 | 0.80 |
| 112 | 1.3 | 1.4 | 1.5 | 0.56 | 0.83 |
| 128 | 1.2 | 1.4 | 1.5 | 0.57 | 0.83 |
| 191 | 1.2 | 1.4 | 1.5 | 0.60 | 0.87 |
| 256 | 1.1 | 1.4 | 1.5 | 0.63 | 0.89 |
| 511 | 0.9 | 1.2 | 1.4 | 0.9 | 1.02 |

| $CW_{min}$ | Average delay at 1400 bytes (sec) | Average jitter at 500 bytes (sec) | Average jitter at 900 bytes (sec) | Average jitter at 1400 bytes (sec) | Collision rate at 500 bytes (%) |
|---|---|---|---|---|---|
| 16 | 1.2 | 0.0108 | 0.0165 | 0.0221 | 27.8 |
| 32 | 1.2 | 0.0088 | 0.0125 | 0.0179 | 16.1 |
| 48 | 1.1 | 0.0084 | 0.0122 | 0.0177 | 12.7 |
| 64 | 1.1 | 0.0077 | 0.0115 | 0.0144 | 10.3 |
| 80 | 1.1 | 0.0074 | 0.0114 | 0.0147 | 7.1 |
| 112 | 1.1 | 0.0076 | 0.0105 | 0.0141 | 5.6 |
| 128 | 1.1 | 0.0073 | 0.0106 | 0.0147 | 4.8 |
| 191 | 1.1 | 0.0077 | 0.0109 | 0.0151 | 3.6 |
| 256 | 1.2 | 0.0079 | 0.0109 | 0.0146 | 2.4 |
| 511 | 1.3 | 0.0093 | 0.0127 | 0.01675 | 1.7 |

| $CW_{min}$ | Collision rate at 900 bytes (%) | Collision rate at1400 bytes (%) | Buffer drops at 500 byte (%) | Buffer drops at 900 byte (%) | Buffer drops at 1400 byte (%) |
|---|---|---|---|---|---|
| 16 | 28.1 | 29.8 | 40.1 | 25.9 | 20.4 |
| 32 | 17.2 | 19.2 | 32.5 | 22.2 | 16.6 |
| 48 | 14.7 | 16.9 | 33.1 | 22.5 | 16.1 |
| 64 | 10 | 10.4 | 32.3 | 21.5 | 14.3 |
| 80 | 8.2 | 8.3 | 32.2 | 21.2 | 14.4 |
| 112 | 5.3 | 7.5 | 33.9 | 21 | 14.2 |
| 128 | 4.9 | 5.6 | 33.98 | 21.6 | 14.7 |
| 191 | 3.4 | 4.2 | 36.7 | 21.8 | 16.3 |
| 256 | 2.7 | 2.9 | 39.4 | 24.1 | 17.3 |
| 511 | 1.6 | 1.8 | 42.5 | 25.6 | 23.3 |

## B.3 Impact of Varying the Number of Retry Limits on the QoS parameters

The IEEE 802.11 standard defines two retry counters, Station Short Retry Count (*SSRC*) and Station Long Retry count (*SLRC*) (IEEE, 1999). These counters are defined as the maximum number of retransmission of a data packet. The values of *SSRC* and *SLRC* are initialised by 0 and increased by 1 every time a packet experiences a failure transmission. A packet is discarded if the retry count exceeds the maximum retry limit. The values are reset to 0 when transmission succeeds. Short retry count is used when the packet size is shorter than the *RTSThreshold*. Thus, it is used with the basic access mechanism. Long retry count is used when the packet size is longer than the

*RTSThreshold*[8]. Accordingly, it is used with the *RTS/CTS* access mechanism. The short count is reset to 0 when:

- A *CTS* is received in response to a transmitted *RTS*.
- An *ACK* is received after a successful transmission.
- A broadcast or multicast packet is received.

The long retry count is reset to 0 when:

- An *ACK* is received for a packet longer than *RTS* threshold.
- A broadcast or multicast packet is received.

This section discusses the impact of varying the number of retry limits on the QoS parameters. Three scenarios were considered when the network topology shown in Figure 4.2d (see Chapter 4) is employed. In the first scenario, 5 sources were considered when they directly transmitted *CBR* traffic to 5 destinations. Each source transmitted 320Kbps. In the second and third scenarios, 10 and 20 connections were considered when each source transmitted 160Kbps and 80Kbps *CBR* traffic, respectively. The packet size was 512 bytes for all scenarios. The simulation time was 300 seconds and it was performed 10 times in order to avoid the bias of random number generation. Other MAC and *PHY* parameters are given in Table 4.1 (see Chapter 4).

Figure B.2a shows average delays for three network sizes as a function of the number of retry limits. Average delay increased as the number of retries increased. Since a larger retry limit indicated a larger backoff window size and longer delay to access the channel. Average delay was saturated for 5 and 10 connections after the $6^{th}$ retry count, while it continuously increased for 20 connections until the $9^{th}$ retry limit. Figure B.2b shows that average jitter was affected less by the alteration of the number of retry limits particularly for 5 and 10 connections. For a large network size (i.e. 20 connections) significant values of average jitter were observed for small number of retries. The reason was that a smaller retry limit determined a smaller backoff window size and a higher number of collisions especially when the number of contending stations was high. Therefore, a smaller backoff window and a large number of collisions caused high variation in the packet arrival time which increased the average jitter.

Figure B.2c shows average throughput as a function of the retry limit. As the retry limit increased, the average throughput increased since the probability of collisions decreased

[8] RTSThreshold is a value that determines whether RTS-CTS exchange is used or not.

particularly in 10 and 20 connections networks. It can be also observed that the average throughput decreased when the retry limit was increased in the 5-connection network. The reason was that in the 5-connection network, the competition between stations was very small; therefore, increasing the retry limit increased the backoff window and decreased the number of packet received per unit of time, which in turn reduced the average throughput.

Figure B.2d shows packet loss rate as a function of retry limit. Packet loss rate decreased as the number of retry limits was increased in 10 and 20 connections networks. The main participant in the reduction of the packet loss rate was the reduction in the probability of collisions as the retry limit was increased. In 5-connections network, the reduction in the probability of collision did not provide any significant help on the reduction of packet loss rate since the number of contending stations was small. On the contrary, increasing the retry limit led to a larger backoff window which caused buffer drops and eventually increased packet loss rate and throttled the average throughput. These findings agreed with the results obtained in (Zhu et al., 2004).

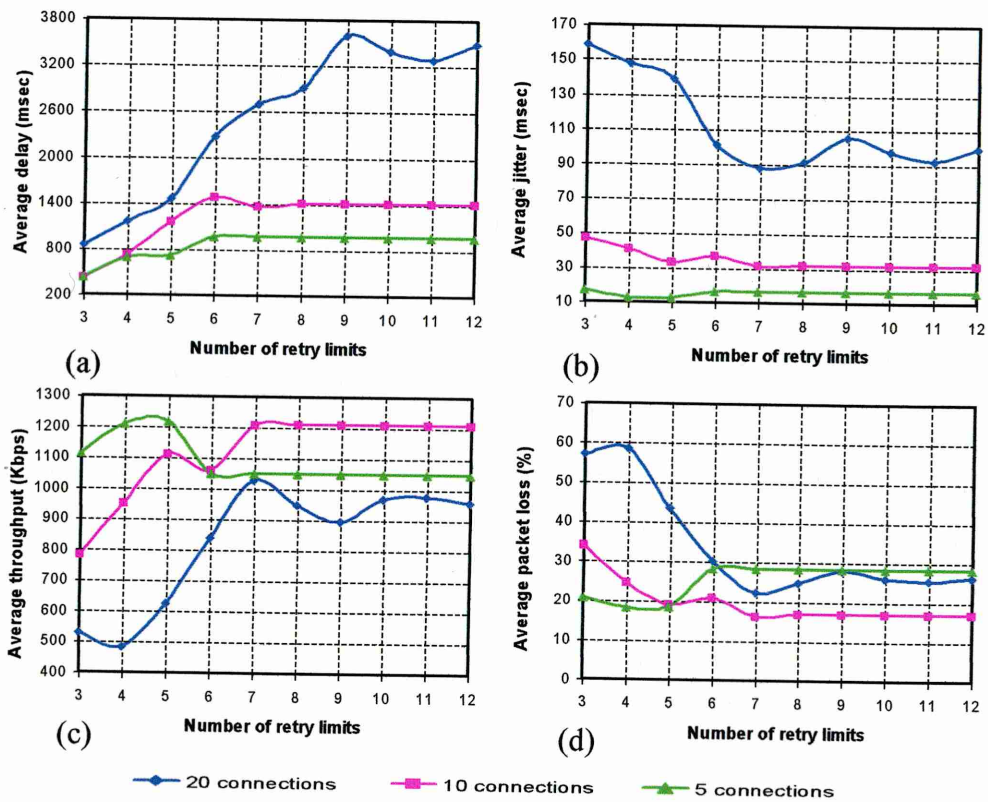


**Figure B.2:** QoS parameters versus number of retry limits, (a) average delay, (b) average jitter, (c) average throughput, and (d) average loss.

# Appendix C

## C.1 Fuzzy Inference System Rules for *CW* Adjustments

'*IF* (*CWprev* is **Medium**) and (*QoS* is **Poor**) and (*collision* is **ELow**) *THEN* (*CWmin_new* is **Low**)'.
'*IF* (*CWprev* is **Medium**) and (*QoS* is **Poor**) and (*collision* is **VLow**) *THEN* (*CWmin_new* is **Low**)'.
'*IF* (*CWprev* is **Medium**) and (*QoS* is **Poor**) and (*collision* is **Low**) *THEN* (*CWmin_new* is **Low**)'.
'*IF* (*CWprev* is **Medium**) and (*QoS* is **Poor**) and (*collision* is **Medium**) *THEN* (*CWmin_new* is **High**)'.
'*IF* (*CWprev* is **Medium**) and (*QoS* is **Poor**) and (*collision* is **High**) *THEN* (*CWmin_new* is **High**)'.
'*IF* (*CWprev* is **Medium**) and (*QoS* is **Poor**) and (*collision* is **VHigh**) *THEN* (*CWmin_new* is **High**)'.
'*IF* (*CWprev* is **Medium**) and (*QoS* is **Poor**) and (*collision* is **EHigh**) *THEN* (*CWmin_new* is **High**)'.
'*IF* (*CWprev* is **High**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **ELow**) *THEN* (*CWmin_new* is **Medium**)'.
'*IF* (*CWprev* is **High**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **VLow**) *THEN* (*CWmin_new* is **Medium**)'.
'*IF* (*CWprev* is **High**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **Low**) *THEN* (*CWmin_new* is **Medium**)'.
'*IF* (*CWprev* is **High**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **Medium**) *THEN* (*CWmin_new* is **VHigh**)'.
'*IF* (*CWprev* is **High**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **High**) *THEN* (*CWmin_new* is **VHigh**)'.
'*IF* (*CWprev* is **High**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **VHigh**) *THEN* (*CWmin_new* is **VHigh**)'.
'*IF* (*CWprev* is **High**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **EHigh**) *THEN* (*CWmin_new* is **VHigh**)'.
'*IF* (*CWprev* is **High**) and (*QoS* is **Poor**) and (*QoSdiff* is **Negative**) *THEN* (*CWmin_new* is **Medium**)'.
'*IF* (*CWprev* is **VHigh**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **ELow**) *THEN* (*CWmin_new* is **High**)'.
'*IF* (*CWprev* is **VHigh**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **VLow**) *THEN* (*CWmin_new* is **High**)'.
'*IF* (*CWprev* is **VHigh**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **Low**) *THEN* (*CWmin_new* is **High**)'.
'*IF* (*CWprev* is **VHigh**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **Medium**) *THEN* (*CWmin_new* is **EHigh**)'.
'*IF* (*CWprev* is **VHigh**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **High**) *THEN* (*CWmin_new* is **EHigh**)'.
'*IF* (*CWprev* is **VHigh**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **VHigh**) *THEN* (*CWmin_new* is **EHigh**)'.
'*IF* (*CWprev* is **VHigh**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **EHigh**) *THEN* (*CWmin_new* is **EHigh**)'.
'*IF* (*CWprev* is **VHigh**) and (*QoS* is **Poor**) and (*QoSdiff* is **Negative**) *THEN* (*CWmin_new* is **High**)'.
'*IF* (*CWprev* is **EHigh**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **ELow**) *THEN* (*CWmin_new* is **VHigh**)'.
'*IF* (*CWprev* is **EHigh**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **VLow**) *THEN* (*CWmin_new* is **VHigh**)'.
'*IF* (*CWprev* is **EHigh**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **Low**) *THEN* (*CWmin_new* is **VHigh**)'.
'*IF* (*CWprev* is **EHigh**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **Medium**) *THEN* (*CWmin_new* is **EHigh**)'.
'*IF* (*CWprev* is **EHigh**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **High**) *THEN* (*CWmin_new* is **EHigh**)'.
'*IF* (*CWprev* is **EHigh**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **VHigh**) *THEN* (*CWmin_new* is **EHigh**)'.
'*IF* (*CWprev* is **EHigh**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **EHigh**) *THEN* (*CWmin_new* is **EHigh**)'.
'*IF* (*CWprev* is **EHigh**) and (*QoS* is **Poor**) and (*QoSdiff* is **Negative**) *THEN* (*CWmin_new* is **VHigh**)'.
'*IF* (*CWprev* is **Low**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **ELow**) *THEN* (*CWmin_new* is **VLow**)'.
'*IF* (*CWprev* is **Low**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **VLow**) *THEN* (*CWmin_new* is **VLow**)'.
'*IF* (*CWprev* is **Low**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **Low**) *THEN* (*CWmin_new* is **VLow**)'.
'*IF* (*CWprev* is **Low**) and (*QoS* is **Poor**) and (*QoSdiff* is **Negative**) and (*collision* is **Medium**) *THEN* (*CWmin_new* is **Medium**)'.
'*IF* (*CWprev* is **Low**) and (*QoS* is **Poor**) and (*QoSdiff* is **Negative**) and (*collision* is **High**) *THEN* (*CWmin_new* is **Medium**)'.
'*IF* (*CWprev* is **Low**) and (*QoS* is **Poor**) and (*QoSdiff* is **Negative**) and (*collision* is **VHigh**) *THEN* (*CWmin_new* is **Medium**)'.
'*IF* (*CWprev* is **Low**) and (*QoS* is **Poor**) and (*QoSdiff* is **Negative**) and (*collision* is **EHigh**) *THEN* (*CWmin_new* is **Medium**)'.
'*IF* (*CWprev* is **VLow**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **ELow**) *THEN* (*CWmin_new* is **ELow**)'.
'*IF* (*CWprev* is **VLow**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **VLow**) *THEN* (*CWmin_new* is **ELow**)'.
'*IF* (*CWprev* is **VLow**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **Low**) *THEN* (*CWmin_new* is **ELow**)'.
'*IF* (*CWprev* is **VLow**) and (*QoS* is **Poor**) and (*QoSdiff* is **Negative**) and (*collision* is **Medium**) *THEN* (*CWmin_new* is **Low**)'.
'*IF* (*CWprev* is **VLow**) and (*QoS* is **Poor**) and (*QoSdiff* is **Negative**) and (*collision* is **High**) *THEN* (*CWmin_new* is **Low**)'.
'*IF* (*CWprev* is **VLow**) and (*QoS* is **Poor**) and (*QoSdiff* is **Negative**) and (*collision* is **VHigh**) *THEN* (*CWmin_new* is **Low**)'.
'*IF* (*CWprev* is **VLow**) and (*QoS* is **Poor**) and (*QoSdiff* is **Negative**) and (*collision* is **EHigh**) *THEN* (*CWmin_new* is **Low**)'.
'*IF* (*CWprev* is **ELow**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **ELow**) *THEN* (*CWmin_new* is **ELow**)'.
'*IF* (*CWprev* is **ELow**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **VLow**) *THEN* (*CWmin_new* is **ELow**)'.
'*IF* (*CWprev* is **ELow**) and (*QoS* is **Poor**) and (*QoSdiff* is **Positive**) and (*collision* is **Low**) *THEN* (*CWmin_new* is **ELow**)'.
'*IF* (*CWprev* is **ELow**) and (*QoS* is **Poor**) and (*QoSdiff* is **Negative**) and (*collision* is **Medium**) *THEN* (*CWmin_new* is **VLow**)'.
'*IF* (*CWprev* is **ELow**) and (*QoS* is **Poor**) and (*QoSdiff* is **Negative**) and (*collision* is **High**) *THEN* (*CWmin_new* is **VLow**)'.
'*IF* (*CWprev* is **ELow**) and (*QoS* is **Poor**) and (*QoSdiff* is **Negative**) and (*collision* is **VHigh**) *THEN* (*CWmin_new* is **VLow**)'.
'*IF* (*CWprev* is **ELow**) and (*QoS* is **Poor**) and (*QoSdiff* is **Negative**) and (*collision* is **EHigh**) *THEN* (*CWmin_new* is **VLow**)'.
'*IF* (*CWprev* is **Medium**) and (*QoS* is **Good**) and (*collision* is **ELow**) *THEN* (*CWmin_new* is **Low**)'.
'*IF* (*CWprev* is **Medium**) and (*QoS* is **Good**) and (*collision* is **VLow**) *THEN* (*CWmin_new* is **Low**)'.
'*IF* (*CWprev* is **Medium**) and (*QoS* is **Good**) and (*collision* is **Low**) *THEN* (*CWmin_new* is **Low**)'.
'*IF* (*CWprev* is **Medium**) and (*QoS* is **Good**) and (*collision* is **Medium**) *THEN* (*CWmin_new* is **Low**)'.
'*IF* (*CWprev* is **Medium**) and (*QoS* is **Good**) and (*collision* is **High**) *THEN* (*CWmin_new* is **High**)'.
'*IF* (*CWprev* is **Medium**) and (*QoS* is **Good**) and (*collision* is **VHigh**) *THEN* (*CWmin_new* is **High**)'.
'*IF* (*CWprev* is **Medium**) and (*QoS* is **Good**) and (*collision* is **EHigh**) *THEN* (*CWmin_new* is **High**)'.
'*IF* (*CWprev* is **High**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) *THEN* (*CWmin_new* is **VHigh**)'.
'*IF* (*CWprev* is **High**) and (*QoS* is **Good**) and (*QoSdiff* is **Negative**) *THEN* (*CWmin_new* is **Medium**)'.
'*IF* (*CWprev* is **VHigh**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) *THEN* (*CWmin_new* is **EHigh**)'.
'*IF* (*CWprev* is **VHigh**) and (*QoS* is **Good**) and (*QoSdiff* is **Negative**) *THEN* (*CWmin_new* is **High**)'.
'*IF* (*CWprev* is **EHigh**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) *THEN* (*CWmin_new* is **EHigh**)'.
'*IF* (*CWprev* is **EHigh**) and (*QoS* is **Good**) and (*QoSdiff* is **Negative**) *THEN* (*CWmin_new* is **VHigh**)'.
'*IF* (*CWprev* is **Low**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **ELow**) *THEN* (*CWmin_new* is **VLow**)'.
'*IF* (*CWprev* is **Low**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **VLow**) *THEN* (*CWmin_new* is **VLow**)'.
'*IF* (*CWprev* is **Low**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **Low**) *THEN* (*CWmin_new* is **VLow**)'.
'*IF* (*CWprev* is **Low**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **Medium**) *THEN* (*CWmin_new* is **VLow**)'.
'*IF* (*CWprev* is **Low**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **High**) *THEN* (*CWmin_new* is **Medium**)'.
........................................................................ *Continued*

'*IF* (*CWprev* is **Low**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **VHigh**) *THEN* (*CWmin_new* is **Medium**)'.
'*IF* (*CWprev* is **Low**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **EHigh**) *THEN* (*CWmin_new* is **Medium**)'.
'*IF* (*CWprev* is **Low**) and (*QoS* is **Good**) and (*QoSdiff* is **Negative**) *THEN* (*CWmin_new* is **Medium**)'.
'*IF* (*CWprev* is **VLow**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **ELow**) *THEN* (*CWmin_new* is **ELow**)'.
'*IF* (*CWprev* is **VLow**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **VLow**) *THEN* (*CWmin_new* is **ELow**)'.
'*IF* (*CWprev* is **VLow**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **Low**) *THEN* (*CWmin_new* is **ELow**)'.
'*IF* (*CWprev* is **VLow**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **Medium**) *THEN* (*CWmin_new* is **Low**)'.
'*IF* (*CWprev* is **VLow**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **High**) *THEN* (*CWmin_new* is **Low**)'.
'*IF* (*CWprev* is **VLow**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **VHigh**) *THEN* (*CWmin_new* is **Low**)'.
'*IF* (*CWprev* is **VLow**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **EHigh**) *THEN* (*CWmin_new* is **Low**)'.
'*IF* (*CWprev* is **VLow**) and (*QoS* is **Good**) and (*QoSdiff* is **Negative**) *THEN* (*CWmin_new* is **Low**)'.
'*IF* (*CWprev* is **ELow**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **ELow**) *THEN* (*CWmin_new* is **ELow**)'.
'*IF* (*CWprev* is **ELow**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **VLow**) *THEN* (*CWmin_new* is **ELow**)'.
'*IF* (*CWprev* is **ELow**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **Low**) *THEN* (*CWmin_new* is **ELow**)'.
'*IF* (*CWprev* is **ELow**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **Medium**) *THEN* (*CWmin_new* is **VLow**)'.
'*IF* (*CWprev* is **ELow**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **High**) *THEN* (*CWmin_new* is **VLow**)'.
'*IF* (*CWprev* is **ELow**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **VHigh**) *THEN* (*CWmin_new* is **VLow**)'.
'*IF* (*CWprev* is **ELow**) and (*QoS* is **Good**) and (*QoSdiff* is **Positive**) and (*collision* is **EHigh**) *THEN* (*CWmin_new* is **VLow**)'.
'*IF* (*CWprev* is **ELow**) and (*QoS* is **Good**) and (*QoSdiff* is **Negative**) *THEN* (*CWmin_new* is **VLow**)'.
'*IF* (*CWprev* is **ELow**) and (*QoS* is **Excellent**) *THEN* (*CWmin_new* is **ELow**)'.
'*IF* (*CWprev* is **VLow**) and (*QoS* is **Excellent**) *THEN* (*CWmin_new* is **VLow**)'.
'*IF* (*CWprev* is **Low**) and (*QoS* is **Excellent**) *THEN* (*CWmin_new* is **Low**)'.
'*IF* (*CWprev* is **Medium**) and (*QoS* is **Excellent**) *THEN* (*CWmin_new* is **Medium**)'.
'*IF* (*CWprev* is **High**) and (*QoS* is **Excellent**) *THEN* (*CWmin_new* is **High**)'.
'*IF* (*CWprev* is **VHigh**) and (*QoS* is **Excellent**) *THEN* (*CWmin_new* is **VHigh**)'.
'*IF* (*CWprev* is **EHigh**) and (*QoS* is **Excellent**) *THEN* (*CWmin_new* is **EHigh**)'.

**Figure C.1:** A complete set of rules that were used with the *FIS* adjustment mechanism.

## C.2 Implication of the Developed AI Approaches in Real System

Figure C.2 depicts the measurement method that can be used to carry out the proposed methods in Chapter 6 in physical system.

For many applications, recorded data have to be transmitted to collection points or exchanged between the pair of communication to be analysed. The massive exchange of data has a negative impact on the network performance of the associated stations. Moreover, the transmission of measured or recorded data between stations or to the collection points can consume significant amounts of network resources which in turn degrades the performance of the whole network. To overcome these shortcomings a form of sampling such as systematic sampling can be employed. The use of sampling can offer information about a specific characteristic of the parent population (Zseby and Scheiner, 2002).

The network parameters of each application were calculated for each connection. For instance the average delay was calculated based on the difference between the values of the timestamps of arrival times for two monitoring points of the sampled packet. In order to ensure the correlation between the two timestamps of the same packet, packet ID has to be the same at the monitoring points (sending point and receiving point). The count-based trigger frequency was set to 10 packets. Thus, the selection of 10 for this aim was sufficient. The same remarks were considered for the rest of network parameters, i.e., throughput, packet loss and jitter.

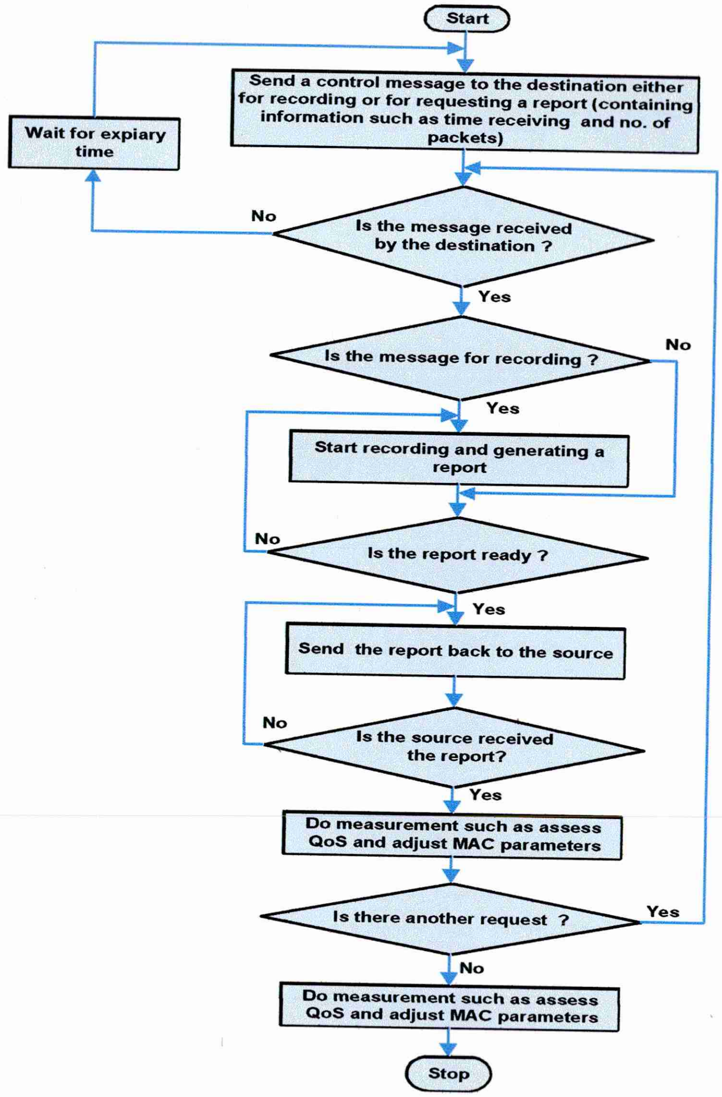


**Figure C.2:** Proposed method for measuring the QoS parameters and adjusting MAC protocol transmission parameters such as $CW_{min}$ and *DIFS* values in physical networks.

After generating the sampled population of the network parameters using systematic sampling, the discrepancy between the actual data and the sampled one was statistically analysed for the linear increase and fuzzy logic approaches. This included the mean value, the standard deviation, and Standard Error (SE) of difference. Some of these statistic values which were carried out using the t-test are summarised in Table C.1 and Figures C.3a and C.3b (Graph, 2005).

The standard error of difference was calculated according to Equation C.1, where *SD* is the standard deviation and *n* is the number of samples (t-test, 2006):

$$SE = \frac{SD^2}{n} \qquad \text{(C.1)}$$

The standard error of difference between the actual data and the sampled data is given by Equation C.2, where $SD_p$ and $SD_s$ are the standard deviations of the original data and the sampled version, respectively, and $n_p$ are the number of original packets and the sampled versions sizes, respectively (t-test, 2006).

$$SE(diff) = \sqrt{\left(\frac{SD_p}{n_p} + \frac{SD_s}{n_s}\right)} \quad \text{(C.2)}$$

**Table C.1:** The means and the standard deviations of the QoS obtained for the population and the sampled version for, (a) simple linear adjustment mechanism, and (b) fuzzy logic adjustment technique.

| Approach | Statistic measure | Basic access mechanism | | | RTS/CTS access mechanism | | |
|---|---|---|---|---|---|---|---|
| | | QoS for video (%) | QoS for audio (%) | QoS for data (%) | QoS for video (%) | QoS for audio (%) | QoS for data (%) |
| Linear increase | **Mean** / sampling | 82 | 89 | 86.8 | 80.7 | 89.5 | 89.5 |
| | **STD** / sampling | 15 | 1.7 | 2.8 | 17.3 | 0.1 | 0.5 |
| | **Mean** / population | 82 | 89.5 | 85.2 | 80.7 | 89.5 | 89.7 |
| | **STD** / population | 14.8 | 0.2 | 8 | 17.3 | 0 | 0 |
| Fuzzy logic | **Mean** / sampling | 76.3 | 89.2 | 77 | 74.2 | 76.2 | 89.7 |
| | **STD** / sampling | 31.5 | 0.6 | 16.5 | 31.7 | 31.7 | 0 |
| | **Mean**/ population | 75.3 | 89.3 | 83.3 | 83.3 | 81 | 89.7 |
| | **STD** / population | 23.7 | 0.5 | 14.8 | 12.2 | 25.5 | 0 |

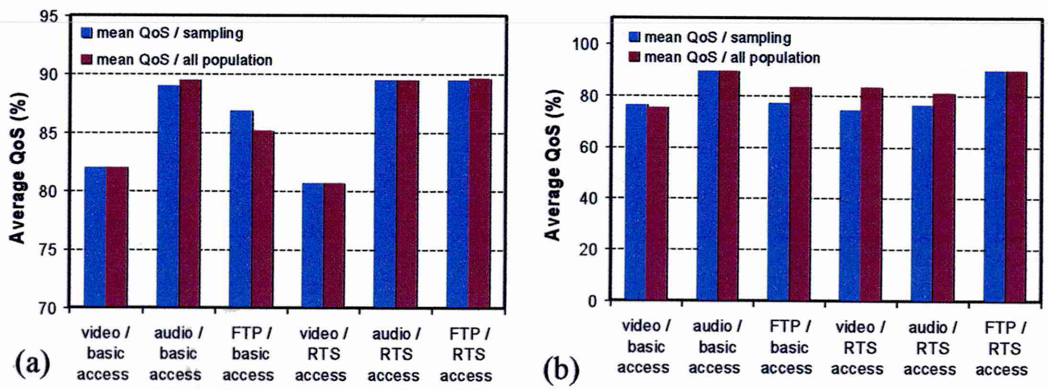


**Figure C.3:** The means of the QoS obtained for the population and the sampled version for, (a) simple linear adjustment mechanism, and (b) fuzzy logic adjustment technique.

The degree of significance was performed to check if the discrepancy between the parent population and the sampled version was statistically significance. This was recognised depending on the $P$ threshold value which was set to 0.05 (a value that has been widely adopted). If the $P$ value was smaller than the threshold value, the difference was statistically significant. Otherwise, the difference was not statistically significant. T -test was carried out for the linear increase scheme and the fuzzy logic approach. The $P$ values were 0.472 for linear increase and 0.192 for fuzzy logic approach. This implied that the systematic sampling method can be used to represent the whole population since there was no statistical difference between the parent population and the sample version in the selected scenarios.

# Appendix D

## D.1 Light Load Traffic

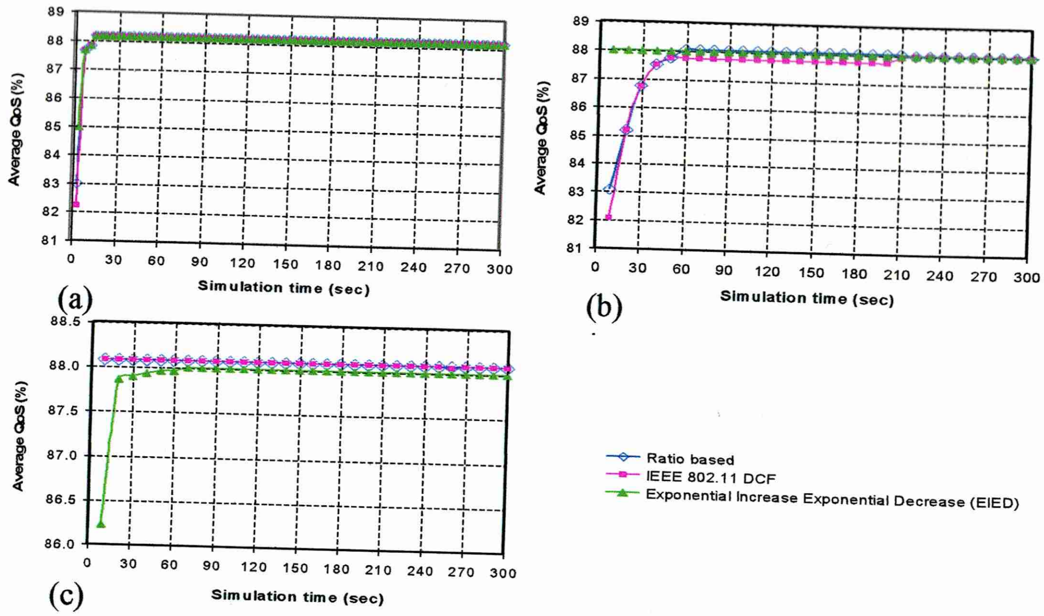


**Figure D.1:** Measured QoS for four schemes for the light load *CBR* traffic, (a) 5, (b) 10, and (c) 20 connections.

## D.2 Medium Load Traffic

Approximately 60% of the channel capacity was offered into the network (i.e. 1.2 Mbps). Each connection transmitted 240 Kbps, 120 Kbps and 60 Kbps corresponding to 5, 10 and 20 connections, respectively. The results obtained for QoS parameters and the assessed QoS for all schemes were similar when 5 connections were transmitting. There was a minor difference between the proposed schemes and the other two schemes in case of 10 and 20 connections (see Figure D.2). The average QoS achieved for 10 connections was 88% (i.e. excellent QoS level) for both the legacy IEEE 802.11 DCF and the *EIED* schemes, whereas, 87% (an excellent QoS level) was achieved when Ratio based and *CRV* schemes were used. In case of 20 connections, the achieved QoS by the proposed schemes was 2% less than the achieved QoS for the standard. This minor reduction in the average QoS of the proposed schemes in the medium load case was due to the impact of the minor increase in the values of delay and jitter. In contrast, the Ratio based and the *EIED* schemes outperformed the IEEE 802.11 DCF scheme in terms of average throughput by 5.7% and 10% for 10 and 20 connections, respectively. This minor difference was insignificant because the QoS parameters obtained were within the QoS requirements for video application as defined by (ITU, 1996) and ($ITU_{(a)}$, 2001). The measured QoS values for all schemes were excellent (i.e. more than

86%). The trends of QoS parameters and the measured QoS for 5, 10 and 20 connections in case of light and medium cases were as shown in Figures D.2.

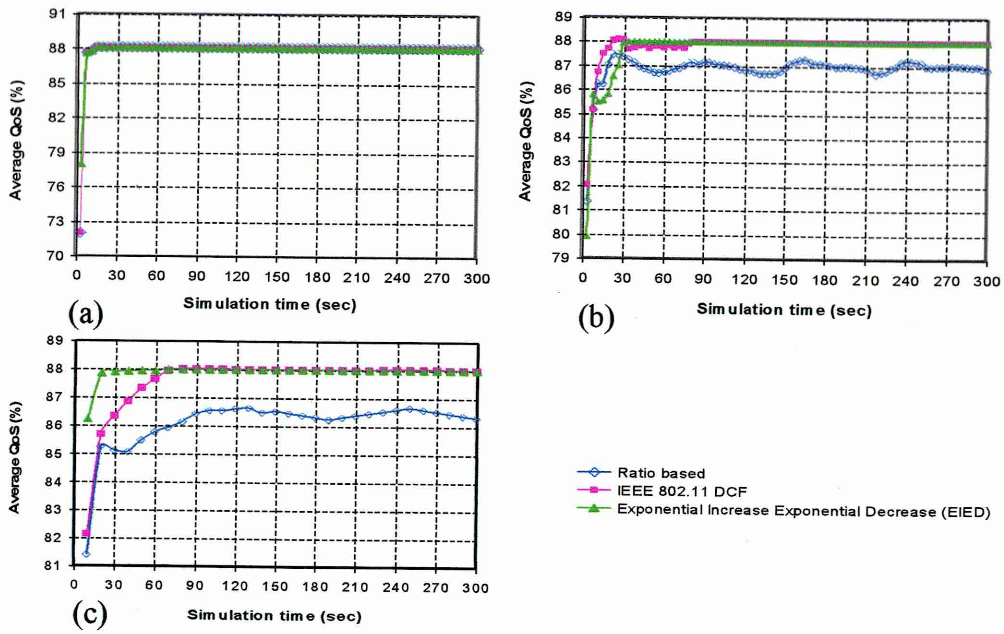


**Figure D.2:** Measured QoS for four schemes for the medium load *CBR* traffic, (a) 5 connections, (b) 10 connections, and (c) 20 connections.

## D.3 Ratio base Scheme with the variation of the Number of Source

In this section, the performance of Ratio based scheme is evaluated when the number of data sources changes sharply. A simulation was carried out using the network shown in Figure 4.2d (see Chapter 4). During the period 1 to 50 seconds of the simulation only one station was transmitting. Hereafter, every 2 seconds a further station was started. There were 20 stations transmitting to different 20 destinations when simulation time reached 86 seconds. The sources offered the network with 1.6 Mbps. At the 150$^{th}$ second, all stations (only one) stopped their transmission to the end of the simulation at 300$^{th}$ second. As shown in Figure D.3a, it can be observed that the Ratio based scheme achieved higher value of throughput with mean value equal to 1354 Kbps while the standard IEEE 802.11 DCF achieved 1273 Kbps during the 70 - 143 seconds. Moreover, the ratio based scheme showed less variation than the IEEE 802.11 DCF scheme. This implied that the Ratio based scheme was capable of appropriately responding to the abrupt changes in the network configurations.

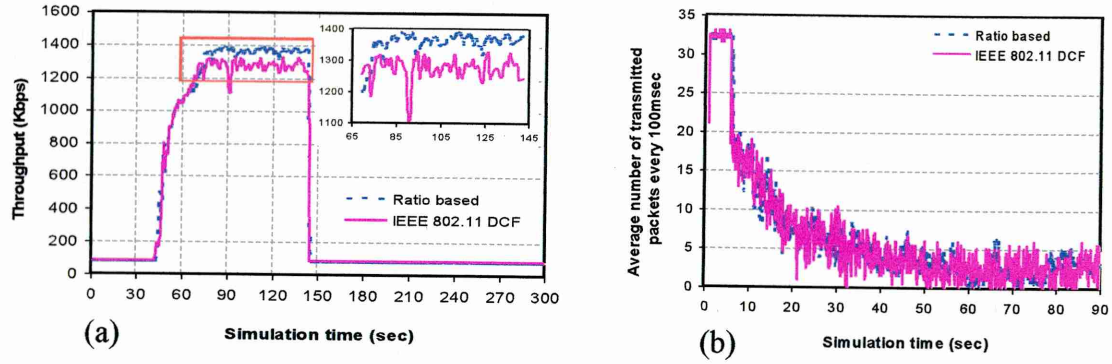


**Figure D.3:** (a) Behaviour of the Ratio based and the IEEE 802.11 DCF schemes when the number of source was changed sharply and (b) throughput of station 1 for the Ratio based and IEEE 802.11 DCF schemes as the number of stations increases over the time.

The fluctuations shown in the throughput curves for the Ratio based and the IEEE 802.11 DCF can be explained as follows: Referring to the main scenario discussed in section 7.5.2.3.4, assume that a station starts its transmission at the beginning of the simulation. As another station commences contending for every 5 seconds, the achieved throughput of station 1 starts to decrease since it has to share the channel with the new sources as shown in Figure D.3b. The throughput pattern of the IEEE 802.11 DCF has greater fluctuations compared to the Ratio based as shown in Figure D.3b. This implies that sometimes station 1 transmits very often and, at other times, only few transmissions are made. This can be explained as follows: In the case of the IEEE 802.11 DCF scheme when the station's packet collides with other stations, the *CW* value is exponentially increased (i.e. increase the backoff duration), and as a result, only few transmissions have taken place for a fixed period of time. When the station transmits without collisions while other stations collide frequently, the *CW* for these stations becomes large (i.e. increase idle time slots) which causes a long defer time due to the large backoff durations. As a result, the station can have more opportunities to send packets.

Conversely, using the Ratio based scheme shows fewer fluctuations since it aims to reduce the number of collisions the station experiences. As a result, the variation in the value of *CW* will be less, which will share the channel in a way that achieves fair transmission opportunities. Because in the Ratio based schemes as discussed earlier, if the station experiences a consecutive defer it resets its *CW* to $CW_{min}$ in order to avoid starvation. This also gives all stations the opportunity to occupy the channel fairly.

## D.4 Mixed Traffic Scenario

In order to evaluate the performance of the Ratio based and *CRV* schemes when different traffic types were considered, a simulation with the topology shown in Figure 4.2d (see Chapter 4) has been carried out. Nine connections were considered, 8 *CBR* connections and one *VBR*. The *CBR* connections were modelled as one video, five audio and two data sources. Audio sources were represented by *CBR* traffic with 160 bytes packet size and 20 msec inter-packet interval between two consecutive voice[9] packets. This model was similar to G.711 voice encoding scheme that generated 160 byte at 20msec inter-packet interval in order to generate audio traffic with 64 Kbps generation rate (Markopoulou et *al.*, 2003) and (Tobagi et *al.*, 2001). Similar to audio traffic, there were several video codecs that could be used as a compression algorithm for video. In this scenario, the H.263 video encoding scheme that generated 512 bytes packet size and 15 msec inter-packet interval was also used in order to generate video traffic with 384 Kbps generation rate (TKN, 2005). The data sources were modelled as *CBR* traffic with 200 bytes packet size and 12.5 msec inter-packet interval in order to generate 128 Kbps traffic rate as described in section 7.4. The *CBR* and *VBR* sources offered the network with a heavy load up to 1.4 Mbps (without the impact of the protocol overhead) throughout the lifetime of the simulation which was 300 seconds.

Figure D.4a show that the Ratio based scheme achieved an excellent QoS for all applications with an overall average equal to 77%. This was 27% and 21% higher than the overall QoS achieved for the IEEE 802.11 DCF and *EIED* schemes.

A considerable reduction was observed in the achieved QoS for audio and data applications when the IEEE 802.11 DCF was used compared with the Ratio based and *CRV* schemes as shown in Figure D.4a. The mean QoS was 57% (i.e. good level of QoS) and 13.69% standard deviation. The *EIED* scheme showed a similar trend of the IEEE 802.11 DCF scheme. It also resulted in a reduction in the achieved QoS for audio applications compared with the achieved QoS when the Ratio based and *CRV* schemes were employed. This was due to the reliance of these schemes on the current network conditions for adjusting the *CW* size without considering the history of each station.

An excellent level with 82% an overall average QoS was observed when the *CRV* scheme was used as shown in Figure D.4b. It performed better the Ratio based scheme

[9] Audio and voice are employed interchangeably throughout this study unless specified.

by 6% but with a higher standard deviation (standard deviation = 9.17). A performance comparison between the four schemes is provided in Table D.1.

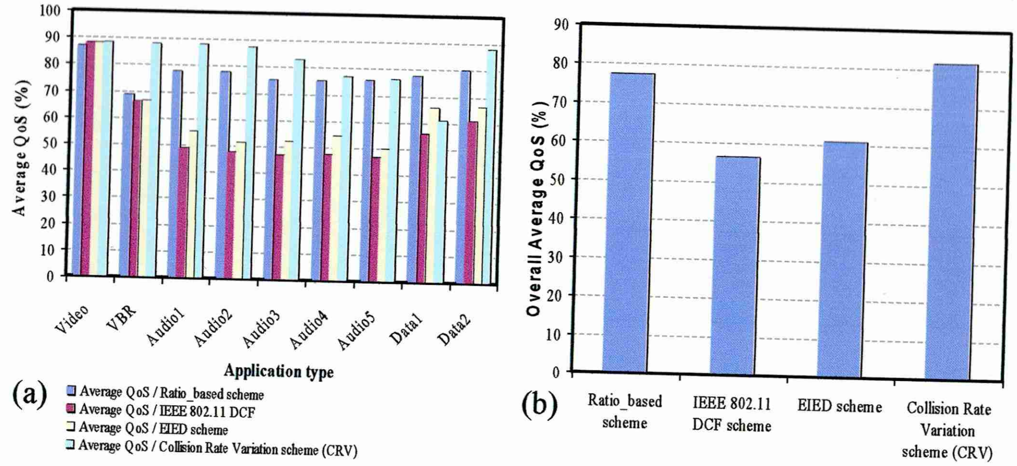


**Figure D.4:** Achieved QoS for mixed traffic for four schemes, (a) average QoS for each application, and (b) overall QoS achieved by each scheme.

**Table D.1:** Average QoS in (%) achieved for mixed traffic scenarios.

| Application | Statistic measure | Ratio based scheme | IEEE 802.11 DCF scheme | EIED scheme | (CRV) scheme |
|---|---|---|---|---|---|
| **Video** | Mean | 86.8 | 88 | 88 | 88.1 |
| **VBR** | Mean | 68.8 | 66.4 | 66.6 | 88.1 |
| **Audio1** | Mean | 78.1 | 48.9 | 55.3 | 88.1 |
| **Audio2** | Mean | 78.2 | 47.9 | 51.3 | 87.1 |
| **Audio3** | Mean | 75.2 | 47.3 | 52.2 | 83.1 |
| **Audio4** | Mean | 75.2 | 47.6 | 54.4 | 76.9 |
| **Audio5** | Mean | 75.7 | 46.6 | 50 | 76.2 |
| **Data1** | Mean | 77.7 | 56.1 | 66 | 61.1 |
| **Data2** | Mean | 80.3 | 60.9 | 66.5 | 88.1 |
| | | | | | |
| | Overall QoS | **77.3** | **56.6** | **61.1** | **81.8** |
| | Stdev | **4.8** | **13.7** | **12.2** | **9.2** |
| | QoS level | **Excellent** | **Good** | **Good** | **Excellent** |

The results indicated that the Ratio based and *CRV* schemes showed better performance than the standard IEEE 802.11 DCF and *EIED* schemes when the network had different traffic types. The *CRV* scheme performed better than the Ratio based scheme by achieving higher value of an overall QoS. However, the Ratio based showed fewer fluctuations and was able to maintain fair share of the channel capacity among the contending stations.

## D.5 Ratio Based in Multi-hop Scenario

In this section, the performance of the Ratio based scheme in comparison with the standard IEEE 802.11 DCF scheme was discussed. The network topology shown in Figure 4.2b (see Chapter 4) was used. In this scenario, the number of sources was varied from one to five and the number of hops was varied from one to three. The performance comparison only included the Ratio based and the standard IEEE 802.11 DCF schemes, and two types of traffic *CBR* and *VBR* traffic.

As shown in Figures D.5a and D.5b, the achieved QoS values for both the Ratio based and the IEEE 802.11 DCF schemes were considerably degraded as the number of hops and sources was increased. In case of one-hop, the two schemes achieved an excellent level of QoS with 88% mean value regardless of the number of sources (i.e. 5 sources in this scenario) for both the *CBR* and *VBR* traffic. When the number of hops was increased to two-hops, only two sources were capable of achieving an excellent QoS level; whereas, good QoS levels were maintained when the number of sources was increased to 5 sources in both schemes. It can be observed that the Ratio based scheme has an average QoS 6% higher than the IEEE 802.11 DCF in case of two-hops when *CBR* traffic was considered and 3% less when *VBR* traffic was considered. Only one source was able to achieve an excellent QoS when the number of hops was increased to three. A good QoS level was maintained for up to three sources in case of *CBR* and up to four sources in case of *VBR*. Thereafter, a poor QoS level was obtained when the number of sources was further increased. In case of three-hops, the average QoS for the Ratio based scheme was 3% higher than the average QoS for the IEEE 802.11 DCF scheme for both *CBR* and *VBR* traffic.

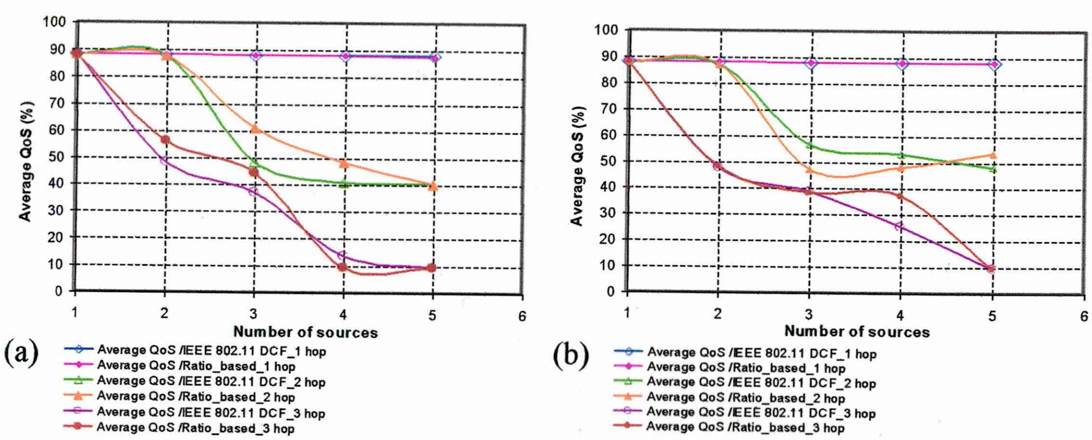


**Figure D.5:** Performance of the Ratio based and the IEEE 802.11 DCF schemes in multi-hop networks, (a) *CBR* traffic and (b) *VBR* traffic.

The two schemes were performed similarly in multi-hop networks as indicated in Table D.2. This was due to the impact of the hidden terminal problem (based on the selected grid topology), since it caused several problems for the Ratio based and the IEEE 802.11 DCF schemes. For instance, the stations in the transmission range of a transmitting station sense the medium as busy and then freeze their backoff timers, while the stations out of the range sense the medium as idle and activate their backoff timers. Subsequently, these stations will adjust their *CW* size based on two different network conditions. As a result, the impact of the hidden terminal problem will be dominant even if the adjustments are occurred adaptively as the case of the Ratio based scheme.

Another cause for the network degradation in multi-hop networks was due to inability of the intermediate stations to deal with the amount and the type of traffic that were forwarded to them. This led to a large number of packets drop at the queues of both the source and the intermediate station. Therefore, in Chapter 8, the Ratio based and *CRV* schemes combined with the application type (i.e. time-sensitive or time-insensitive applications) and the queue status monitoring approach, the performance of multi-hop networks can be improved and service differentiation can be provided.

**Table D.2:** Average QoS in (%) for multi-hop networks for two schemes and two traffic types.

| Number of hops | Number of sources | Traffic type | | | |
|---|---|---|---|---|---|
| | | CBR | | VBR | |
| | | IEEE 802.11 DCF scheme | Ratio based scheme | IEEE 802.11 DCF scheme | Ratio based scheme |
| 1 hop | 1 | 88.2 | 88.2 | 88.2 | 88.2 |
| | 2 | 88.2 | 88.2 | 88.2 | 88.2 |
| | 3 | 88.1 | 88.1 | 88 | 87.9 |
| | 4 | 88 | 87.9 | 87.9 | 88 |
| | 5 | 87.9 | 87.5 | 87.9 | 87.8 |
| 2 hop | 1 | 88.2 | 88.2 | 88 | 88 |
| | 2 | 87.7 | 87.7 | 87.1 | 87.2 |
| | 3 | 48.5 | 60.9 | 56.7 | 47.3 |
| | 4 | 40.9 | 48.5 | 53.1 | 48.1 |
| | 5 | 40.5 | 40.5 | 48.1 | 53.4 |
| 3 hop | 1 | 88.2 | 88 | 88 | 87.8 |
| | 2 | 48.3 | 56.1 | 47.7 | 48 |
| | 3 | 37.2 | 44.5 | 38.6 | 38.3 |
| | 4 | 13.8 | 9.5 | 25.2 | 37.3 |
| | 5 | 9.5 | 9.5 | 9.6 | 9.5 |

# Appendix E

## E.1 Service Differentiation in IEEE 802.11 DCF scheme

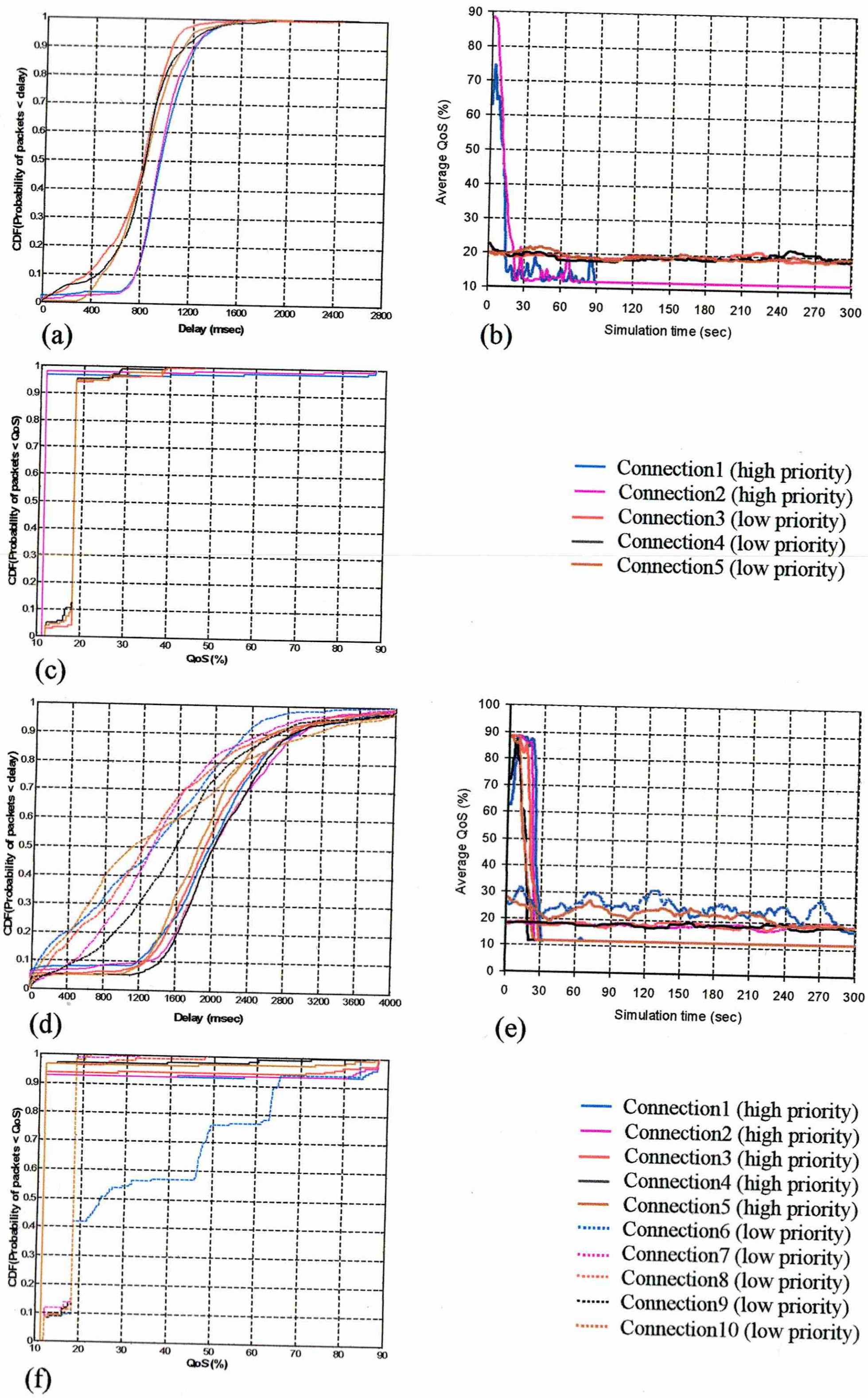


**Figure E.1:** Service differentiation for IEEE 802.11 DCF, (a) Cumulative distribution of delay (5 connections) (b) Average QoS (5 connections), (c) Cumulative distribution of QoS (5 connections), (d) Cumulative distribution of delay (10 connections), (e) Average QoS (10 connections), and (f) Cumulative distribution of QoS (10 connections).

# Appendix F

**Table F.1:** A summary of the QoS parameters and the measured QoS for *CBR* traffic in small, medium, and large networks for *AR*, IEEE 802.11 DCF, and *EIED* schemes.

| No. of connections | Parameters | Schemes | | | | |
|---|---|---|---|---|---|---|
| | | IEEE 802.11DCF | *EIED* | Collision rate prediction | *CRV* prediction | *CW* prediction |
| 5 / *CBR* | Average delay (msec) | 809 | 691.2 | 335.6 | 403 | 354.4 |
| | Standard deviation (msec) | 51 | 78.9 | 75.4 | 102.7 | 64.2 |
| | Average jitter (msec) | 12.8 | 11.6 | 10.1 | 8.3 | 10.1 |
| | Standard deviation (msec) | 0.4 | 0.9 | 1.4 | 1.3 | 1.3 |
| | Average throughput (Kbps) | 1224.8 | 1248.1 | 1298 | 1292.7 | 1292.7 |
| | Standard deviation (Kbps) | 32.9 | 78.4 | 57.4 | 64.3 | 59.2 |
| | Average packet loss (%) | 21.1 | 16.1 | 7 | 11.5 | 7.7 |
| | Standard deviation (%) | 1.8 | 2.4 | 1.7 | 4.2 | 1.8 |
| | Average MAC efficiency (%) | 85.7 | 86.8 | 93 | 93.3 | 93.1 |
| | Standard deviation (%) | 0.4 | 1.5 | 0.6 | 1.2 | 0.6 |
| | Average collision rate (%) | 13.9 | 11.8 | 7 | 7.3 | 6.8 |
| | Standard deviation (%) | 0.5 | 0.8 | 0.4 | 0.4 | 0.4 |
| | Average QoS (%) | 10.9 | 23.3 | 67.2 | 58.9 | 64.8 |
| | Standard deviation (%) | 4.9 | 16 | 3.6 | 8.5 | 3.9 |
| 10 / *CBR* | Average delay (msec) | 1788 | 1520.3 | 632.1 | 672.1 | 604.4 |
| | Standard deviation (msec) | 308 | 197 | 96.1 | 187.1 | 85.1 |
| | Average jitter (msec) | 40.1 | 36.3 | 21.3 | 17.7 | 20.6 |
| | Standard deviation (msec) | 5.1 | 3.5 | 2.2 | 2.3 | 1.7 |
| | Average throughput (Kbps) | 1027.3 | 1089.1 | 1234.8 | 1197.7 | 1220.1 |
| | Standard deviation (Kbps) | 40.3 | 67.4 | 106.1 | 134.4 | 117.1 |
| | Average packet loss (%) | 27.1 | 19.5 | 8 | 8.2 | 7.2 |
| | Standard deviation (%) | 5.5 | 3.6 | 1.7 | 2.5 | 1.3 |
| | Average MAC efficiency (%) | 81.1 | 83.7 | 91.1 | 89.5 | 91.1 |
| | Standard deviation (%) | 1.6 | 1.2 | 1 | 0.8 | 0.8 |
| | Average Collision rate (%) | 17 | 14.9 | 8.9 | 10 | 8.9 |
| | Standard deviation (%) | 1 | 0.6 | 0.6 | 0.4 | 0.6 |
| | Average QoS (%) | 11.5 | 13.7 | 57.3 | 60.2 | 58.5 |
| | Standard deviation (%) | 6.4 | 5.8 | 3.4 | 5.6 | 3.4 |
| 20 / *CBR* | Average delay (msec) | 3348.7 | 3635 | 1391.4 | 1509.7 | 1479.5 |
| | Standard deviation (msec) | 803.8 | 685.9 | 173 | 456 | 263.4 |
| | Average jitter (msec) | 95.2 | 92.7 | 46.8 | 46.3 | 48.6 |
| | Standard deviation (msec) | 22.5 | 16.7 | 3.4 | 10 | 5.4 |
| | Average throughput (Kbps) | 931.6 | 976.3 | 1147.3 | 1063 | 1139.9 |
| | Standard deviation (Kbps) | 99.5 | 86 | 132.2 | 168.1 | 142.8 |
| | Average packet loss (%) | 24.7 | 23.9 | 10.1 | 9.2 | 8.8 |
| | Standard deviation (%) | 8.8 | 6.3 | 1.6 | 3.3 | 2.1 |
| | Average MAC efficiency (%) | 78.7 | 80.4 | 89 | 87.1 | 89.2 |
| | Standard deviation (%) | 3.4 | 2.1 | 1.7 | 1.7 | 2.1 |
| | Average collision rate (%) | 17.2 | 16.9 | 11.3 | 11.5 | 11 |
| | Standard deviation (%) | 1.6 | 1 | 0.9 | 0.8 | 1 |
| | Average QoS (%) | 18.6 | 19.8 | 39.9 | 49.3 | 41.4 |
| | Standard deviation (%) | 11.4 | 9.7 | 3.8 | 8.1 | 5.2 |

**Table F.2:** A summary of the QoS parameters and the measured QoS for *VBR* traffic in small, medium, and large networks for *AR*, IEEE 802.11 DCF, and *EIED* schemes.

| No. of connections | Parameters | Schemes | | | | |
|---|---|---|---|---|---|---|
| | | IEEE 802.11DCF | *EIED* | Collision rate prediction | *CRV* prediction | *CW* prediction |
| **5 / *VBR*** | Average delay (msec) | 430 | 469 | 199.4 | 225.5 | 224.8 |
| | Standard deviation (msec) | 181.1 | 143.2 | 75.9 | 75.2 | 69.4 |
| | Average jitter (msec) | 14.6 | 14.9 | 10.6 | 9.9 | 10.8 |
| | Standard deviation (msec) | 1.4 | 1.1 | 1.3 | 1.5 | 1.1 |
| | Average throughput (Kbps) | 1307.4 | 1326.1 | 1337.1 | 1334.1 | 1333 |
| | Standard deviation (Kbps) | 110.7 | 103.8 | 118.3 | 112.3 | 118.9 |
| | Average packet loss (%) | 3.4 | 4 | 2.2 | 2.7 | 2.3 |
| | Standard deviation (%) | 2.3 | 2 | 1.5 | 1.5 | 1.4 |
| | Average MAC efficiency (%) | 84 | 84.8 | 92.8 | 92.2 | 92.4 |
| | Standard deviation (%) | 1.3 | 0.9 | 0.9 | 1.6 | 0.6 |
| | Average collision rate (%) | 14.9 | 14.1 | 7.6 | 8.4 | 7.5 |
| | Standard deviation (%) | 1.4 | 1.4 | 0.4 | 0.7 | 0.5 |
| | Average QoS (%) | 50.4 | 46.3 | 73.6 | 71.3 | 71.3 |
| | Standard deviation (%) | 14.9 | 12.7 | 5.5 | 5.1 | 5.4 |
| **10 / *VBR*** | Average delay (msec) | 1497.4 | 1231.1 | 366.9 | 552.9 | 317 |
| | Standard deviation (msec) | 415.1 | 265.9 | 103.7 | 105.7 | 69.2 |
| | Average jitter (msec) | 49.8 | 43.4 | 20.9 | 21.4 | 19.9 |
| | Standard deviation (msec) | 4.6 | 5.7 | 2.3 | 1.5 | 1.7 |
| | Average throughput (Kbps) | 1130.1 | 1196.7 | 1322.1 | 1291.2 | 1346.4 |
| | Standard deviation (Kbps) | 95.8 | 103.8 | 130.3 | 124.1 | 73.2 |
| | Average packet loss (%) | 13 | 7.5 | 1.7 | 3.4 | 1.4 |
| | Standard deviation (%) | 3.3 | 2.4 | 0.9 | 0.9 | 0.5 |
| | Average MAC efficiency (%) | 76.9 | 81 | 90.8 | 89.5 | 91.3 |
| | Standard deviation (%) | 1.4 | 1.6 | 1 | 1.3 | 0.6 |
| | Average Collision rate (%) | 21.3 | 16.4 | 9.1 | 9.8 | 9 |
| | Standard deviation (%) | 1.3 | 1.9 | 0.4 | 0.5 | 0.2 |
| | Average QoS (%) | 25.1 | 29.2 | 63.9 | 61.9 | 65.8 |
| | Standard deviation (%) | 8.3 | 10.1 | 4.8 | 2.3 | 2.9 |
| **20 / *VBR*** | Average delay (msec) | 2729.2 | 1958.5 | 637.2 | 669.7 | 705.9 |
| | Standard deviation (msec) | 1201.5 | 716.1 | 197.5 | 278.3 | 279.3 |
| | Average jitter (msec) | 97.7 | 88.7 | 38.3 | 38.6 | 40.8 |
| | Standard deviation (msec) | 27.8 | 23.1 | 7.4 | 9 | 8.2 |
| | Average throughput (Kbps) | 812.7 | 966.4 | 1078 | 1057.3 | 1107.1 |
| | Standard deviation (Kbps) | 163.4 | 163.4 | 194.8 | 210.8 | 180.3 |
| | Average packet loss (%) | 10.1 | 6 | 1.5 | 1.6 | 1.5 |
| | Standard deviation (%) | 3.8 | 2.5 | 0.7 | 0.8 | 1 |
| | Average MAC efficiency (%) | 76.8 | 76.4 | 89.3 | 88.9 | 88.5 |
| | Standard deviation (%) | 6.1 | 4 | 1 | 1.6 | 1.1 |
| | Average collision rate (%) | 15.5 | 17.9 | 9.4 | 9.4 | 10.2 |
| | Standard deviation (%) | 3.2 | 3.2 | 1 | 1.2 | 1.1 |
| | Average QoS (%) | 21.8 | 28.2 | 53.8 | 56.1 | 52 |
| | Standard deviation (%) | 13.9 | 14.4 | 7.8 | 9.4 | 8.2 |

# Appendix G

T-test is calculated in accordance with standard statistical procedure to do comparisons between the new methods (Ratio based, *CRV*, and prediction), the standard IEEE 802.11 DCF, and *EIED* schemes (t-test, 2006). This analysis is required in order to provide a method for comparing two different set of results for different mechanisms to specify whether the discrepancy between the mechanisms is statistically significant. The analysis was conducted by comparing the *P-value*, Degree of Freedom, and *95%* confidence interval. The *P-value* represents the probability of observing a difference between the two groups, which ranges from 0 to 1. Small value of *P* indicates that the two methods have different means (i.e., there is a statistical significant difference), while large value of *P* indicates that there is no statistical significant difference between the two methods. Degree of freedom (df) is the number of samples minus one. The confidence interval was chosen to be 95%. It represents the difference between the mean of the two groups in which there is 95% chance that the interval includes the true difference between the two group means (t-test, 2006). As indicated in Table G.1, the t-test analysis is useful for statistically comparing the average QoS obtained by the new methods, the basic IEEE 802.11 DCF and *EIED* schemes. Using t-test, there was a significant statistical difference between the considered methods (see Table G.1).

**Table G.1:** Statistical t-test analysis comparing the new MAC mechanisms and the standard IEEE 802.11 DCF and *EIED* schemes.

| Schemes | No. of connections) | P-value | Degree of freedom (df) | 95% confidence interval | Comments of difference |
|---|---|---|---|---|---|
| **Ratio based** vs. **IEEE 802.11 DCF** | **20** | less than 0.0001 | 79 | 16.04 to 19.83 | Extremely statistically significant |
| **Ratio based** vs. ***EIED*** | **20** | less than 0.0001 | 79 | 13.32 to 17.97 | Extremely statistically significant |
| ***CRV*** vs. **IEEE 802.11 DCF** | **20** | less than 0.0001 | 79 | 31.76 to 37.44 | Extremely statistically significant |
| ***CRV*** vs. ***EIED*** | **20** | less than 0.0001 | 79 | 29.1 to 35.52 | Extremely statistically significant |
| **Prediction** vs. **IEEE 802.11 DCF** | **20** | less than 0.0001 | 79 | 16.62 to 20.22 | Extremely statistically significant |
| **Prediction** vs. ***EIED*** | **20** | less than 0.0001 | 79 | 13.54 to 18.72 | Extremely statistically significant |
| **Ratio based** vs. **IEEE 802.11 DCF** | **5** | less than 0.0001 | 79 | 52.12 to 55.19 | Extremely statistically significant |
| **Ratio based** vs. ***EIED*** | **5** | less than 0.0001 | 79 | 37.68 to 44.87 | Extremely statistically significant |
| ***CRV*** vs. **IEEE 802.11 DCF** | **5** | less than 0.0001 | 79 | 55.98 to 60.68 | Extremely statistically significant |
| ***CRV*** vs. ***EIED*** | **5** | less than 0.0001 | 79 | 41.8 to 50.11 | Extremely statistically significant |
| **Prediction** vs. **IEEE 802.11 DCF** | **5** | less than 0.0001 | 79 | 55.01 to 57.51 | Extremely statistically significant |
| **Prediction** vs. ***EIED*** | **5** | less than 0.0001 | 79 | 40.37 to 47.40 | Extremely statistically significant |

The t-test analysis was also used with the *AR* prediction models to statistically determine whether the difference between the actual and the predicted values is statistically significant. As indicated in Table G.2, using t-test, there was no significant statistical difference between these values which confirmed the accuracy of the *AR* prediction models.

**Table G.2:** Statistical t-test to determine the accuracy of the AR prediction models.

| Model order | No. of connections | P-value | Degree of freedom (df) | 95% confidence interval | Comments of difference |
|---|---|---|---|---|---|
| 1st (actual vs. predicted) | 10 | 0.3706 | 99 | (0.00158 to 0.0019125) | the difference is considered to be not statistically significant |
| 2nd (actual vs. predicted) | 10 | 0.2479 | 99 | (0.004955 to 0.0012945) | the difference is considered to be not statistically significant |
| 3rd (actual vs. predicted) | 10 | 0.611 | 99 | (0.002565 to 0.004336) | the difference is considered to be not statistically significant |
| 4th (actual vs. predicted) | 10 | 0.6764 | 99 | (0.004505 to 0.00293862) | the difference is considered to be not statistically significant |
| 5th (actual vs. predicted) | 10 | 0.6519 | 99 | (0.0030186 to 0.001898) | the difference is considered to be not statistically significant |
| 6th (actual vs. predicted) | 10 | 0.5174 | 99 | (0.0030296 to 0.00153426) | the difference is considered to be not statistically significant |
| 7th (actual vs. predicted) | 10 | 0.1024 | 99 | (0.0036137 to 0.0003342) | the difference is considered to be not statistically significant |