

QoS provisioning in future wireless local area networks Indira Paudel

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Ecole doctorale : Informatique, Télécommunications et Electronique de Paris

Spécialité : Informatique et Télécommunications

QoS PROVISIONING IN FUTURE WIRELESS LOCAL AREA NETWORKS

AMELIORATION DE LA QUALITE DE SERVICE

DANS LES FUTURS RESEAUX LOCAUX SANS FIL

Présentée le 15 janvier 2015 par

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Pour obtenir le grade de

DOCTEUR DE TELECOM SUDPARIS

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Abstract

Wireless Local Area Networks (WLANs) still remain the most popular access solutions at homes and offices. Although initially, WLANs were designed to carry data traffic, today there is a noticeable increase in throughput as can be seen in the latest IEEE 802.11n as well as with the upcoming IEEE 802.11ac standards.

WLANs are now considered as natural extensions of the wired and wireless ecosystems. In particular, these are naturally considered as complementary solutions to cellular and ADSL networks to provide additional resources, improve indoor and outdoor coverage and to offer ubiquitous access at homes and offices. This requires that WLANs provide similar QoS (Quality of Service) and QoE (Quality of Experience) levels as the above operated networks.

However, since WLANs standards are still based on the CSMA/CA technique, with competition among stations, providing QoS guarantees and differentiation is challenging, especially for voice and video applications, when the number of competing stations or flows increases.

Today, users expect to get similar performances over Wi-Fi, as experienced over cellular and ADSL access networks. In addition, network providers are more and more willing to rely on Wi-Fi hotspots in order to alleviate their network and extend their coverage (e.g. Wi-Fi offloading). This requires Wi-Fi technologies to provide equivalent QoS classes and guarantees as compared to cellular systems.

Although there has been a set of amendments to the WLAN standards, providing guaranteed QoS on Wi-Fi is still challenging. This is mainly due to the fact that the standard QoS solutions relying on centralized approaches (e.g. PCF, HCCA) are not widely used on terminals. The distributed approach, based on concurrent access, remains fundamental in WLANs.

In this regard, new enhancements are required in order to propose new QoS aware decentralized MAC solutions, while maintaining backward compatibility with existing Wi-Fi standards.

In this thesis, we propose new solutions to improve both the QoS and QoE of multimedia services over WLAN networks. If QoS measures the objective quality of the network transmissions, usually expressed through performance indicators like delay, jitter and throughput, QoE is a subjective measurement of users' experience towards a network service.

QoE best reflects the user's satisfaction, but both of them reveal the overall performances of the system.

In this thesis, we present contributions toward a QoS/QoE aware adaptive MAC. These enhancements allow, while keeping backward compatibility, to provide QoS differentiation among flows through a context aware tuning of some of the MAC parameters.

After the state of the art analysis, the first contribution starts with an evaluation of the latest techniques used in the current IEEE 802.11n standards such as frame aggregation and block acknowledgement. Based on this, a first MAC enhancement is proposed. It consists in an adaptive MAC level aggregation mechanism that relies on QoS differentiation by selecting the most suitable aggregation size for each of the service classes. We show through simulations that this enhancement leads to improved quality of service and better efficiency. We then evaluated the QoE of video services over IEEE 802.11n Wi-Fi networks for various radio, MAC, and load conditions.

Based on the above studies, a random neural network solution is then proposed to automate the video service QoE prediction from system parameters. This solution is then used to analyze more widely the effect of different MAC and network parameters on video QoE.

The second contribution provides a backward compatible modification of the distributed access mechanism used in Wi-Fi networks. The proposed approach allows QoS differentiation among concurrent flows and consists in two steps. First, we propose to select appropriate and specific Backoff values according to QoS requirements for each flow. Second, we propose a new flow prioritization based on AIFSN (Arbitration Inter-Frame Space Number) values, allocated according to traffic load and types.

Through analysis, we showed that the proposed solution can enhance QoS in different manners. It provides regular access and delays as required by delay sensitive flows and minimizes collision rates for better resource utilization and efficiency.

Keywords: Quality of Service, Quality of Experience, Medium Access Control, IEEE 802.11, video, AIFS differentiation, backoff differentiation, random neural network, learning algorithm.

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Table of Contents

Chapter	· 1	General Introduction	. 1
1.1	Intro	oduction	. 1
1.2	Mot	ivations	. 6
1.3	The	sis Contributions and outline	. 7
Chapter	· 2	State of the Art	. 9
2.1	Intro	oduction	. 9
Part I:	PH	Y and MAC evolution in IEEE 802.11 standards	
2.2	IEE	E 802.11: PHY evolutions	12
2.2.1	1	Frequency Hopping (FH) PHY	12
2.2.2	2	Direct Sequence PHYs: DSSS	13
2.2.3	3	Orthogonal Frequency Division Multiplexing (OFDM)	15
2.2.4	4	Modulation and Coding	17
2.2.5	5	Multiple Input Multiple Output (MIMO)	20
2.2.0	6	Channel Bonding	24
2.2.7	7	Summary	25
2.3	IEE	E 802.11: MAC evolutions	27
2.3.	1	IEEE 802.11 legacy MAC	28
2.3.2	2	MAC in IEEE 802.11e	34
2.3.3	3	MAC in IEEE 802.11n	37
2.3.4	4	MAC in IEEE 802.11ac	40
2.3.5	5	Summary	46
2.4	Upc	oming IEEE 802.11 standards	47
Part II	[: Qo	S provisioning in MAC over IEEE 802.11 standards	
2.5	QoS	provisioning	50
2.5.	1	QoS metrics	50
2.5.2	2	Main QoS provisioning methods based on MAC protocols	52
2.6	Con	clusions	59
Chapter	• 3	An adaptive aggregation scheme for QoS differentiation over IEEE 802.11n	51
3.1	Intro	oduction	51
3.2	Fran	ne Aggregation in IEEE 802.11n	52

2.0	a 1		(\mathbf{a})
3.2	2.1	A-MSDU aggregation	62
3.2	2.2	A-MPDU aggregation	63
3.2	2.3	Two Level Aggregation	65
3.2	2.4	Block Acknowledgement	65
3.2	2.5	Analytical model for aggregation	66
3.3	Rel	ated Work	68
3.4	Pro	posed Solution: QoS-HAN aggregation	68
3.5	Per	formance Evaluation	71
3.5	5.1	Simulation Scenarios	71
3.5	5.2	Performance Analysis	72
3.6	Cor	nclusions	78
Chapt	er 4	QoE Estimation for Video: WLAN MAC layer Perspective	81
4.1	Intr	oduction	81
4.2	Qua	ality of Experience: background and definitions	82
4.2	2.1	Subjective QoE test methods	83
4.3	Imp	pact of MAC level parameters on video QoE	84
4.3	3.1	Related work	85
4.3	3.2	Potential MAC-level parameters	86
4.3	3.3	An experimental platform to evaluate the impact of MAC parameters on QoE	87
4.3	3.4	QoE results and analysis	89
4.3	3.5	Conclusions	94
4.4	AR	andom Neural Network based QoE estimation and Learning	94
4.4	4.1	Random Neural networks	96
4.4	4.2	Proposed RRN based QoE estimation approach	99
4.4	4.3	QoE prediction	102
4.5	Cor	nelusions	105
Chapt WLAN	er 5 N	An enhanced channel access and differentiation scheme for IEEE 802.11	107
5.1	Intr	oduction	107
5.2	Bac	kground and related work	109
5.2	2.1	BackOff Algorithms	109

5.2.2	Related work	112
5.3 I-D	CF: an Improved Distributed Coordination Function for WLAN	114
5.3.1	BackOff-ID calculation	114
5.3.2	BackOff Calculation Algorithm	115
5.3.3	Performance Evaluation of I-DCF	116
5.4 An	enhanced traffic differentiation scheme	126
5.4.1	Related work	127
5.4.2	Performance Evaluation	131
5.5 Cor	nclusions	138
Chapter 6	Conclusions	139
REFERENC	ES	145

List of Figures

Figure 1.1 : Global devices and connection growth [2]	2
Figure 1.2: Global residential service adoption and growth [2]	2
Figure 1.3 : Monthly data consumption over cellular and Wi-Fi for video users	4
Figure 1.4 : Evolution of IEEE 802.11 Physical Layer Amendments [7]	6
Figure 2.1 : Frequency Hopping	12
Figure 2.2 : Orthogonal hopping sequences	13
Figure 2.3 : Basic DSSS technique	14
Figure 2.4 : Spreading of noise by correlation process	14
Figure 2.5: Channel layout in different standards	16
Figure 2.6: MIMO [15]	20
Figure 2.7: IEEE 802.11n Signal Processing Techniques that use MIMO Antennas [15]	21
Figure 2.8: A MIMO-OFDM transceiver	22
Figure 2.9: Single User MIMO versus Multi user MIMO	23
Figure 2.10: PHY level frame format	27
Figure 2.11 : Basic DCF operation	31
Figure 2.12: RTS/CTS	32
Figure 2.13 : CTS to self	33
Figure 2.14: PCF access method	34
Figure 2.15 : EDCA mechanism	36
Figure 2.16 : The CAP/CFP/CP periods	37
Figure 2.17 : (a) static bandwidth operation (b) dynamic bandwidth operation	40
Figure 2.18 (a) Static 80 MHz bandwidth operation and (b) Dynamic 20/40/80 MHz bandw	width
operation	43
Figure 2.19 : Successful channel acquisition	44
Figure 2.20 : Interference at channels	45
Figure 3.1 : A-MSDU aggregation	63
Figure 3.2 : A-MPDU aggregation	64
Figure 3.3 : Two level aggregation	65
Figure 3.4 : Block Ack Frame [19]	66
Figure 3.5 : Block Ack with aggregation	66
Figure 3.6 : QoS-HAN aggregation	70
Figure 3.7 : Total throughput comparison at low BER	73
Figure 3.8 : Total throughput comparison at high BER	73
Figure 3.9 : End to end delay at low BER	74
Figure 3.10 : End to end delay at high BER	75
Figure 3.11 : PDR at low BER	76
Figure 3.12 : PDR at high BER	76
Figure 3.13: Throughput of different Traffic flow	77

Figure	3.14 : Delay of different Traffic flow	78
Figure	4.1 : Subjective test Environment	89
Figure	4.2 : Frame delivery ratio vs. different MAC-level parameters	90
Figure	4.3 : QoS and QoE versus number of competing stations	90
Figure	4.4 : Impact of BER on video QoE	91
Figure	4.5 : Impact of number of competing stations on video QoE	92
Figure	4.6 : Impact of queue length on video QoE	93
Figure	4.7 : Impact of maximum retransmission limit on video QoE	94
Figure	4.8 : QoE estimation module	- 101
Figure	4.9 : Comparison between subjective and estimated QoE	- 101
Figure	4.10 : Probability distribution of the subjective and estimated MOS difference	- 102
Figure	4.11 : Variation of QoE in different load and different BER conditions	- 103
Figure	4.12 : Variation of QoE in different BER and aggregation limits (5 stations)	- 104
Figure	4.13 : Variation of QoE in different BER and aggregation limits (12 stations)	- 105
Figure	4.14 : Variation of QoE in different BER and aggregation limits (20 stations)	- 105
Figure	5.1 : Two dimensional Markov chain model	- 110
Figure	5.2 : Normalized Saturation Throughput Comparison with Channel Data rate =1 Mbp	DS
		- 120
Figure	5.3 : I-DCF/DCF Throughput vs. Number of active actions	- 122
Figure	5.4 : I-DCF/DCF delay vs. number of active stations	- 123
Figure	5.5 : I-DCF/DCF collisions vs. Number of active stations	- 123
Figure	5.6 : I-DCF/DCF PDR vs. number of active stations	- 124
Figure	5.7 : I-DCF/DCF each station throughput	- 125
Figure	5.8 : I-DCF/DCF each station delay	- 125
Figure	5.9 : I-DCF/DCF access delay of any nodes during a certain simulation time	- 126
Figure	5.10 : Schema showing the principle of proposed solution	- 130
Figure	5.11 : Throughput of each high priority flow vs. total number of flows (12 high priori	ty
flows)-		- 134
Figure	5.12 : Delay of each high priority flow vs. total number of flows (12 high priority flo	ws)
Figuro	5.13 Throughput of each high priority flow vs. total number of flows (4 high priorit	- 135 V
flows)-		y - 136
Figure	5.14 : Throughput of each low priority flow vs. total number of flows (4 high priority	1
flows)-		- 137
Figure	5.15 : Delay of each high priority flow vs. total number of flows (4 high priority flow	/s)
		- 137

List of Tables

Table 1.1 : Existing wireless technologies	4
Table 2.1: Carrier distribution in different standards	16
Table 2.2: Possible data rates in IEEE 802.11a	18
Table 2.3: Some IEEE 802.11n MCS values	19
Table 2.4: IEEE 802.11ac MCS value	19
Table 2.5 : Phy features in IEEE 802.11 standards	25
Table 2.6: Priority Mapping	35
Table 2.7 : Default EDCA AC Parameter Set	36
Table 2.8 : MAC features in IEEE 802.11 standards	46
Table 2.9 : Overview of MAC enhancement proposals	57
Table 3.1: Simulation Parameters	71
Table 4.1 MOS scores and its categories	83
Table 4.2: Video Characteristics	87
Table 4.3 : Simulation Parameters	88
Table 4.4 : MAC parameter values	88
Table 4.5 : Learning technique used for QoS/QoE	95
Table 4.6 : Video characteristics	99
Table 4.7 : MAC parameters values	99
Table 5.1 : List of parameters for mathematical analysis	120
Table 5.2 : BACKOFF-ID calculation	121
Table 5.3 : Simulation Parameters	121
Table 5.4 : Simulation Parameters	132
Table 5.5 : Priority flow parameters	132

Glossary

3GPP	3rd Generation Partnership Project
64-QAM	64-Quadrature Amplitude Modulation
ACK	Acknowledgement
ACs	Access Categories
ACU	Admission Control Unit
ADSL	Asymmetric digital subscriber line
AFR	Aggregation with Fragment Retransmission
AIFS	Arbitration Inter-Frame Space
AIFSN	Arbitration Inter-Frame Space Number
AMSDU	Aggregated MAC Service Data Unit
ANN	Artificial Neural Network
AP	Access Point
AQOS	Application Level QoS
ATM	Asynchronous Transfer Mode
BAR	Block Ack Request
BEB	Binary Exponential Backoff
BPSK	Binary Phase Shift Keying
CCA	Clear Channel Assessment
CDMA	Code Division Multiple Access
CFP	Contention Free Period
СР	Contention Period
CRC	Cyclic Redundancy Check
CSI	Channel state information
CSMA	Collision Sense Multiple Access
CSMA/CA	Collision Sense Multiple Access/Collision Avoidance
CTS	Clear To Send
CW	Collision Window
DARE	Destination-Assisted Routing Enhancement
DBPSK	Differential Binary Phase Shift Keying
DCF	Distributed Coordination Function

DHCP	Dynamic Host Configuration Protocol
DIDD	Double Increase Double Decrease
DIFS	DCF Interframe Space
DL	Down Link
DQPSK	Differential Quadrature Phase Shift Keying
DSCP	Differentiated Services Code Point
DSCQS	Double Stimulus Continuous Quality Scale
DSIS	Double Stimulus Impairment Scale
DSSS	Direct Sequence Spread Spectrum
DSSS/CCK	Complementary Code Keying (CCK)
EDCA	Enhanced Distributed Channel Access
EDCAF	Enhanced Distributed Channel Access Function
EIFS	Extended Inter Frame Space
ERP	Extended Rate Phy
FCS	Frame Check Sequence
FDMA	Frequency Division Multiple Access
FFT	Fast Fourier Transfer
FH	Frequency Hopping
FHSS	Frequency-Hopping Spread Spectrum
FIFO	First In First OUT
FSS	Frequency Space Spectrum
FTP	File Transfer Protocol
GFSK	Gaussian Frequency Shift Keying
GFSK HAN	Gaussian Frequency Shift Keying Home Automated Network/ Home Area Network
GFSK HAN HC	Gaussian Frequency Shift Keying Home Automated Network/ Home Area Network Hybrid Controller
GFSK HAN HC HCCA	Gaussian Frequency Shift Keying Home Automated Network/ Home Area Network Hybrid Controller HCF Controlled Channel Access
GFSK HAN HC HCCA HCF	Gaussian Frequency Shift Keying Home Automated Network/ Home Area Network Hybrid Controller HCF Controlled Channel Access Hybrid Contention Function
GFSK HAN HC HCCA HCF HR	Gaussian Frequency Shift Keying Home Automated Network/ Home Area Network Hybrid Controller HCF Controlled Channel Access Hybrid Contention Function High Rate
GFSK HAN HC HCCA HCF HR HT-LTF	Gaussian Frequency Shift Keying Home Automated Network/ Home Area Network Hybrid Controller HCF Controlled Channel Access Hybrid Contention Function High Rate High Throughput Long Training Field
GFSK HAN HC HCCA HCF HR HT-LTF HT-SIG	Gaussian Frequency Shift Keying Home Automated Network/ Home Area Network Hybrid Controller HCF Controlled Channel Access Hybrid Contention Function High Rate High Throughput Long Training Field High Throughput Signal Field
GFSK HAN HC HCCA HCF HR HT-LTF HT-SIG HT-STF	Gaussian Frequency Shift KeyingHome Automated Network/ Home Area NetworkHybrid ControllerHCF Controlled Channel AccessHybrid Contention FunctionHigh RateHigh Throughput Long Training FieldHigh Throughput Signal FieldHigh Throughput Short Training Field
GFSK HAN HC HCCA HCF HR HT-LTF HT-SIG HT-STF HTTP	Gaussian Frequency Shift KeyingHome Automated Network/ Home Area NetworkHybrid ControllerHCF Controlled Channel AccessHybrid Contention FunctionHigh RateHigh Throughput Long Training FieldHigh Throughput Signal FieldHigh Throughput Short Training FieldHypertext Transfer Protocol
GFSK HAN HC HCCA HCF HR HT-LTF HT-SIG HT-STF HTTP ICI	Gaussian Frequency Shift KeyingHome Automated Network/ Home Area NetworkHybrid ControllerHCF Controlled Channel AccessHybrid Contention FunctionHigh RateHigh Throughput Long Training FieldHigh Throughput Signal FieldHigh Throughput Short Training FieldHypertext Transfer ProtocolInter Carrier Interference

IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
IFFT	Inverse Fast Fourier Transfer
IPTV	Internet Protocol Television
ITU	International Telegraph Union
ITU-T	Telecommunication Standardization Sector
LAN	Local Area Network
LDPC	Low Density Parity check
LLC	Logical Link Control
L-LTF	Non-High Throughput Long Training Field
LPSC	Low Power Single Carrier
L-SIG	Non-High Throughput Signal Field
L-STF	Non-High Throughput Short Training Field
LTE	Long Term Evolution
LTF	Long Training Field
M2M	Machine to Machine
MAC	Medium Access Network
MCS	Maximal Ratio Combining
MDI	Media Delivery Ration
MILD	Multiplicative Increase and Linear Decrease
MIMO	Multiple Input Multiple Output
ML	Maximal Likelihood
MOS	Mean Opinion Score
MPDU	MAC Protocol Data Unit
MSDU	MAC Service Data Unit
MU-MIMO	Multiuser MIMO
NAV	Network Allocation Vector
NQoS	Network Level QoS
NTW	Network
OFDM	Orthogonal Frequency Division Multiplexing
OSI	Open Systems Interconnection
P2P	Peer to Peer
PCF	Point Coordination Function
PDR	Packet Delivery Network

PEVQ	Perceptual Evaluation of Video Quality
PF	Persistence Factor
РНҮ	Physical
PIFS	PCF Interframe Space
PLCP	Physical Layer Convergence Protocol
PN	Pseudo Random Noise
PSK	Phase Shift Keying
PSNR	Peak Signal-to-Noise Ratio
QAP	QoS Access Point
QoE	Quality of Experience
QoS	Quality of Service
QPSK	Quadrature Phase-Shift Keying
QSTA	QoS Station
RD	Reverse Direction
RF	Radio Frequency
RIFS	Reduce Interference Space
RNN	Random Neural Network
RTS	Request to Send
RX	Receiver
SAMVIQ	Subjective Assessment Methodology for VIdeo Quality
SDM	Space Division Multiplexing
SDSCE	Simultaneous Double Stimulus Continuous Evaluation
SIFS	Short Interframe Space
SIG	Signal
SM	Spatial Multiplexing
SM-STBC	Space-Time Block Coded Spatial Modulation
SM-STCC	Spatial Modulation With Space-time Complementary Coding
SNR	Signal to Noise Ratio
SSCQS	Single Stimulus Continuous Quality Scale
SSIM	Structural Similarity
STA	Station
STBC	Space Time Block Coding
STF	Short training field
TCF	Tournament Contention Function

TID	Traffic Identifier
TSPEC	Traffic Specification
TV	Television
TX	Transmitter
ТХОР	Transmission Opportunity
UL	Up Link
V-BLAST	Vertical BLAST (Bell Laboratories Layered Space-Time)
VBR	Variable Bit Rate
VHT	Very High Throughput
VOD	Video On Demand
VOIP	Voice Over IP
VQM	Video Quality Metric
Wi-Fi	Wireless Fidelity
WiMax	Worldwide Interoperability for Microwave Access
WLAN	Wireless Local Area Network
WMAN	Wireless Metropolitan Area Networks
WPAN	Wireless Personal Area Networks
WWAN	Wireless Wide Area Networks

Chapter 1 General Introduction

1.1 Introduction

Today, wireless technologies are used everywhere in our daily life, both for professional and personal services including telephony and entertainment services (e.g., music, TV, VoD, cooking, shopping). The concept of wireless might have initially given a layman a magical feeling of connection. However, due to the proliferation of wireless communication devices, today it has evolved as one of the eminent part of the Internet ecosystem and has augmented its potential to a greater extent.

In the recent years, there has been an explosion in the number of devices like smartphones, tablets, TVs, Blu-ray players, etc. According to researchers at Cisco, by 2018, there will be more than 8 billion personal devices that will be connecting to the mobile networks and also 2 billion machine to machine connections [1]. Apart from terminal devices, intermediate equipment like routers, home gateways, Internet Boxes, media servers, and Wi-Fi range-extenders are becoming the norm for wireless networks. Figure 1.1 shows the global devices and connection growth prediction from 2013 to 2018. For instance, we can see that M2M (Machine to Machine) devices are one of the fastest growing device categories that will increase its share from 18% to 35% over the considered period [2].

Due to the adoption of these personal and machine to machine communication devices, a wide range of new services and applications is expected to fill up the current networks. These include P2P (Peer to Peer)/M2M services, digital television, interactive gaming, video streaming, audio streaming, navigation services, emergency services, and health care applications. Most of these services generate time sensitive flows. For instance, Figure 1.2 shows the global residential service growth and adoption; we can observe that the video service is continuously growing in the coming years. Although the increase in devices and applications has offered mobile data capabilities, they have also led to an immense increase in data traffic.



The percentages in parentheses next to the legend denote the device share for the years 2013 and 2018, respectively.

Figure 1.1 : Global devices and connection growth [2]





In order to drive this evolution, there have been continuous amendments at the technological side. Today, WLANs (Wireless Local Area Networks) can be considered as one of the most popular wireless access networks, over which enormous amount of multimedia applications are exchanged, as likely as in the wired networks. The popularity of WLAN comes from their easy deployment, availability, and cost effectiveness. Moreover, the rapid maturity of this technology within a short span of time has also attracted the wireless and wire line operators to seek it as a supplement to expand their service and business coverage. Not only that Wi-Fi extends the

ADSL/Fiber access as the "last-mile access" at homes and offices, is also considered as one of the possible wireless access technologies within the 5G architecture.

The original design of WLANs was driven by the support of data services while the cellular networks targeted voice services [3]. However, during the continuous evolution process, both technologies are now targeting to cover all services. With the increasing number of applications over WLANs, there is an equal demand for new features and functions at the customer end.

Moreover, most of the modern user devices such as tablets and smart phones are equipped with a number of network interfaces, thus allowing them to use either of the many available access technologies. Currently there is a huge debate about which technology (Wi-Fi or cellular), should lead the future or should they complement each other for a converged network, for instance Wi-Fi competing with Femto and Pico cells. Nevertheless, it can be said that the network today is moving towards heterogeneity [4]. In the current scenario, more and more service providers are investing on Wi-Fi as there is no license cost associated with it. Thus, on the one hand, we can see the cellular technology trying to benefit from the unlicensed band used by Wi-Fi, in order to gain more spectrums. On the other hand, Wi-Fi is trying to improve its performance in dense environments to achieve better quality of service. With the enormous amount of traffic demands, it is reported that the average amount of traffic per Smartphone has doubled in 2010 and out of these, 31% of the traffic was offloaded to WLAN hotspots [1]. Furthermore, it can be seen from Figure 1.3 that the monthly traffic consumption for services like video is proportionally more over Wi-Fi as compared to that over cellular networks [5]. This shows the increased capacity and interest in Wi-Fi networks.

With the possible higher data rate over Wi-Fi, at par with the wired Ethernet, new applications can now be supported. However, due to the random based underlying access technique, supporting QoS and QoE stringent applications, like IPTV, VOIP, is still challenging. This demands profound understanding and innovative access mechanisms that can provide QoS-guarantees. Researchers are on their way to further enhance the standards, so as to make the technology more reliable in terms of quality assurance.



Figure 1.3 : Monthly data consumption over cellular and Wi-Fi for video users

Local wireless networking has evolved drastically in merely a few decades. Different versions of this technology have appeared in order to integrate WLANs in the wireless technology ecosystem, to extend its range, coverage, and functionalities. For instance, WLANs are basically used in private homes, offices, and hotspots, like cafeterias and airports. They can provide last mile wireless access alternative to many other wired technology.

Table 1.1 lists the main wireless technologies of today. Each of these individual technologies has undergone successive standardization to improve their performance and widen the panorama of the wireless environment.

Network	Standard	Data rate	Frequency band
Cellular networks	UMTS, 3G	Up to 2 Mbps	1990-2025 MHz
	4G	100 Mbps (high speed)	
		1 Gbps (stationary conditions)	
WLAN	IEEE 802.11b	1-11 Mbps	2.4 GHz
	IEEE 802.11n	100-540 Mbps	2.4 and 5GHz
	IEEE 802.11ac	433 Mbps-6.77 Gbps	5Ghz
Wireless Personal	IEEE 802.15.3	11-55 Mbps	2.4 GHz
Networks (WPAN)			
Zigbee	IEEE 802.15.4	20-250 Mbps	868 MHz, 915
			MHz
Wireless Metropolitan	IEEE 802.16.a	75 Mbps	2-11 GHz
Area Network			

Table 1.1 : Existing wireless technologies

Network	Standard	Data rate	Frequency band
(WMAN)			
WiMAX	IEEE 802.16c	134 Mbps	10-66 GHz
Wireless Wide Area	IEEE 802.20	2.25-18Mbps	3.5 GHz
Networks (WWAN)		_	

The development of WLAN came around with a major innovation in the US government spectrum policy [6]. Until 1985, frequency spectrums were only given on a licensed basis to launch TV, radio, satellites, and back haul communication. However, the door for new series of innovation like Wi-Fi opened only when the unlicensed junk bands were made available to be shared. It was, however, required that they do not create any kind of interference to the licensed band users. Thus, the first generation of Wi-Fi started in 1997 with the introduction of first Wi-Fi enabled device. This led to the deployment of WLANs. People were now free to move about in the room without any loss of connection. Since then, Wi-Fi has become universal on laptops, smartphones, video games' consoles, and many others home appliances like camera and TV.

Over the years, there has been a progressive amendment to the initial WLAN IEEE 802.11 standard, in order to increase its capacity and support different services. Starting with IEEE 802.11b as the first generation of WLAN technology (released in 1999), there has been a continuous evolution in the standard. IEEE 802.11a is considered as the second generation, 802.11g represents the third and 802.11n the fourth generation. Recently proposed 802.11ac is regarded as the fifth generation of this wireless technology. With each generation, there has been a breakthrough in the physical data rate that these technologies can attain. The initial IEEE 802.11b started with a data rate of 11 Mbps in the 2.4 GHz band, however, now, the technology has permitted the use of new physical enhancements like MIMO and spatial multiplexing, such as in IEEE 802.11n, to attain a data rate of 300 Mbps(4X4 MIMO in a 40 MHz channel). The recent fifth generation (802.11ac Wave 2) is ready to boost the theoretical data rate to about 7 Gbps as shown in Figure 1.4. This is due to the enhanced modulation, channel bandwidth, and the increased number of spatial streams. A detailed account of this evolution is presented in Chapter 2.



Figure 1.4 : Evolution of IEEE 802.11 Physical Layer Amendments [7]

1.2 Motivations

QoS support for multimedia services is a crucial requirement for any kind of future WLAN expansion. This can further increase its success probability to be embedded as one of the key elements in the future wireless communication solution like 5G. All multimedia services provided by the wireless networks require stringent quality of service support. This can be related to guaranteed bandwidth, reduced error rate, delay, and jitter. Furthermore, the perceived QoS can vary from various perspectives and contexts. It may depend upon a number of parameters. For instance, for services like video, user satisfaction play a crucial role and the meaning of QoS may further move towards Quality of Experience (QoE). According to the ITU-T Focus Group on IPTV, Quality of Experience (QoE) refers to "the overall acceptability of an application or service, as perceived subjectively by the end-user" [8].

Providing QoS and QoE requirements are challenging tasks when it comes to IEEE 802.11 WLAN protocols and Medium Access Control (MAC). Indeed, the medium access technique directly determines the efficiency with which the data are transmitted. Therefore QoS/QoE-aware wireless networks require well designed MAC access protocol. Although some progress has been made for QoS support in WLANs, there are many challenges and issues that further

pose problems in QoS provisioning. Some of these challenges may be related to the inherent nature of wireless channel conditions, interference, issues related to channel access, power consumption, synchronization, etc.

Despite important PHY evolutions of WLANs (see chapter 2), there are still rooms for improving MAC to consider QoS issues over these networks. Providing better Quality of Service (QoS) support to the time sensitive applications still remains challenging. Many MAC mechanisms have been proposed in the literature that aims to improve the network performance. However, the efficiency improvement at the MAC layer faces many challenges that can directly or indirectly affect the quality of service that this technology can provide. Therefore, wireless networks require a precisely designed MAC protocol that can improve QoS at the network level and QoE at the user level to comply with the QoE and QoS perceived over cellular networks.

1.3 Thesis Contributions and outline

In this thesis, the main contributions consider improving the QoS and QoE (Quality of Experience) of multimedia services from a MAC level perspective. In this regard, two main contributions are presented in this thesis:

- QoS/QoE aware adaptive MAC
- Improved MAC with backoff and AIFS differentiation

- QoS/QoE aware adaptive MAC

After the state of the art analysis, our first contribution starts with an evaluation of the latest techniques used in the current 802.11n standards such as frame aggregation and block acknowledgement. Based on this, a first MAC enhancement is proposed. It consists in an adaptive MAC level aggregation mechanism that relies on QoS differentiation by selecting the most suitable aggregation size for each of the considered service classes. We show through simulations that this enhancement leads to improved quality of service and better efficiency.

We then analyzed and evaluated QoE for video applications over IEEE 802.11n WLAN networks, considering various radio and MAC conditions. Based on the above study, a

random neural network solution is then proposed to automate the video service QoE prediction from system parameters. This neural network estimation system can be used in the design of Soft MAC techniques that can adapt parameters according to PHY/MAC conditions in order to provide better QoE.

- Improved MAC with backoff and AIFS differentiation

The second contribution is related to the enhancement of the inherent distributed access mechanism used in WLAN networks. A new approach is proposed for QoS differentiation among concurrent flows. It consists in two enhancements. First, we propose to select appropriate and specific Backoff values according to QoS requirements. Second, we propose a new flow prioritization based on the allocation of specific AIFSN (Arbitration Inter-Frame Space Number) values, according to traffic load and traffic types.

Through analysis, we show that the proposed solutions can enhance user QoS/QoE as well as network efficiency through different means: provide regular access delays as required by delay sensitive flows and minimize collision rates for better resource utilization.

The remainder of this thesis is divided into five chapters. In chapter 2, a deeper insight on wireless networks' standards is provided. It describes the different PHY and MAC evolutions in the IEEE 802.11 family. In this chapter, we also present the main proposals for quality of service improvement in wireless networks from the literature. Chapter 3 presents QoS-HAN, a new proposal for frame aggregation in IEEE 802.11n. Performances of the proposed scheme are analyzed. Chapter 4 is divided into two parts. The first part is dedicated to the evaluation of the performances of video traffic over IEEE 802.11n. In the second part, a Random Neural network approach is proposed for video QoE estimation from MAC layer parameters. Chapter 5 is also divided into two parts. The first presents and evaluates I-DCF: an enhanced DCF mechanism designed to improve the channel access mechanism over WLANs. The second part presents a new flow prioritization technique using AIFS differentiation. Finally, Chapter 6, concludes this thesis underlining our main contributions and results, and highlighting future perspectives, applications, and research directions

Chapter 2 State of the Art

2.1 Introduction

The IEEE 802.11 refers to the family of the IEEE specifications for WLAN that started with the IEEE 802.11b standard (completed in 1997) and is still being amended in order to offer higher and higher capacities.

IEEE 802.11 based WLANs are one of the major wireless networks deployed in the world today. As a radio technology for last mile communication, they are likely to play an important role in the next-generation wireless networks. Today, they are present in many personal and professional application areas. There is a phenomenal increase of Wi-Fi enabled gadgets, smart phones, tablets, and many other smart devices enabling a Wi-Fi mass market.

IEEE WLANs standards provide specifications for the two lower layers of the OSI (Open Systems Interconnection) model. The physical layer (PHY) defines the means of transmission of bits over the physical medium (air interface) providing electrical, mechanical, and other procedural specifications. The data link layer is divided into two parts: the logical link sub-layer (LLC) and the Medium Access control sub-layer (MAC). The latter is responsible for managing the access to the medium and its sharing among stations.

Over the last decades, different amendments have been proposed to WLAN standards. The main objective is to increase its capacity from 11 Mbps associated with the first IEEE 802.11b standard to 7 Gbps with the upcoming IEEE 802.11ac, there is a phenomenal boost in the data rate. This evolution comes with enhancements at both PHY and MAC layers. However, QoS provisioning is still based on the optional IEEE 802.11e amendment.

In the following sections, we present the state of the art into two parts. First, we provide the PHY and MAC evolution in IEEE 802.11. In the second part, we present the main QoS provisioning techniques from a MAC layer perspective and the related proposals based on them.

State of the Art

Part I: PHY and MAC evolution

In IEEE 802.11 standards

2.2 IEEE 802.11: PHY evolutions

The capabilities of the WLAN devices are often based on the implemented PHY. In the following section, a detailed account of the main PHY evolution is presented along with the associated IEEE 802.11 standard that integrated them.

2.2.1 Frequency Hopping (FH) PHY

Frequency hopping was the first step in the evolution of complex data transmission techniques. In frequency hopping scheme, the system jumps from one frequency to another in a pseudorandom pattern and transmits a short burst at each sub channel. The timing for each hop must be synchronized so that the receiver can hear the transmitter. Both frequency and time are divided into slots and the hopping pattern governs the usage of the slots [9].

In Figure 2.1, the hopping pattern of (2, 4, 6 and 8) is shown. Frequency Hopping is similar to Frequency Division Multiple Access (FDMA). In FDMA each device is assigned a different and permanent frequency, whereas, in Frequency Hopping, the allocated frequency is changed periodically, after each *dwell time* interval. If two frequency hopping systems need to coexist, different hopping patterns can be assigned so that at each time, different system uses different frequency slots as shown in Figure 2.2. The non-overlapping hopping sequences are called *orthogonal* and their use in one single area can maximize the overall throughput[9].



Figure 2.1 : Frequency Hopping



Figure 2.2 : Orthogonal hopping sequences

The FHSS system is cheaper, less complicated and quite stable in handling interference, however, it necessitates precise timing.

FHSS is used by the earliest IEEE 802.11 standard. The whole 2.4 GHz ISM band frequency is separated into channels that are spaced 1 MHz For frequency hopping these frequencies are then picked up according to the hopping pattern. They are then modulated using two-level GFSK (Gaussian Frequency Shift Keying) for 1 Mbps and four-level GFSK modulation for 2 Mbps data rate [10].

2.2.2 Direct Sequence PHYs: DSSS

In Direct Sequence System, power is spread over a wide frequency band by making use of mathematical coding functions. Figure 2.3 shows a basic DSSS technique where a narrow band signal is processed by a *spreader*. The spreader applies a mathematical function to spread the signal amplitude to a wide frequency band. This is done by making use of a chipping sequence also known are *pseudo random noise codes* (PN codes) that must run at a rate much higher than the underlying data rate [9]. These chipping sequences consist of *chipping streams* containing a number of chips or bits that are used to carry a single data bit. The idea is to multiply the data being transmitted to a pseudo random binary sequence of a higher bit rate [10].



Figure 2.3 : Basic DSSS technique

The number of chips used for the transmission of a single data bit is known as *spreading ratio*. For real systems, this ratio should be as low as possible to avoid the wastage of bandwidth. Large values of spreading ratio need wider bandwidth and also increase the cost. On the other hand, for signal recovery, a *correlator* is used at the receiver. The correlator inverts the spreading process by looking at any change to the Radio Frequency (RF) signal occurring across the entire frequency band. This correlation also helps in noise cancellation because the noise is also spread out across the band as shown in Figure 2.4.



Figure 2.4 : Spreading of noise by correlation process

Higher throughput can be achieved with DSS rather than FSS. However, sophisticated signal processing is needed to extract information from the signal and, therefore, high power and specialized hardware are required.

DSSS was used by the earliest IEEE 802.11 specification. It divides the 2.4 GHz band into 11 overlapping with channels spacing of 5MHz. The IEEE 802.11 transmitter always sends symbols (1 or several chips) at the rate of 11Mbps, which requires 22 MHz bandwidth. The transmitter uses the Differential Binary Phase Shift Keying (DBPSK) and Differential Quadrature Phase Shift Keying (DQPSK) modulation to achieve a data transmission rates of 1 Mbps and 2 Mbps respectively [10].

It also finds application in IEEE 802.11b standard where both the Direct-Sequence Spread Spectrum (DSSS) and the Complementary Code Keying (CCK) are used to provide a data rate up to 11 Mbps [11]. It allows transmitting the header with a short preamble at 1 Mbps. The rest of the header is transmitted at 2 Mbps (using DSSS DQPSK) and the data payload is either transmitted at the same 2 Mbps or at 5.5 Mbps or 11 Mbps using CCK.

2.2.3 Orthogonal Frequency Division Multiplexing (OFDM)

In OFDM system, the channel is divided into multiple sub-channels and a single transmission is encoded into multiple subcarriers in parallel. OFDM takes the coded signal from each sub channel and uses the Inverse Fast Fourier transform (IFFT) to create a composite waveform from the strength of each sub channel. The OFDM receiver then applies FFT to this waveform to get back the amplitude of each subcarrier component. In OFDM, there are multiple subcarriers that are used at different frequencies so they may cause interference between carriers known as *Inter-Carrier Interference* (ICI). On the other hand, *Inter Symbol Interference* (ISI) occurs when there are large delay spreads between different paths that may result in a delayed copy of one transmitted bit to shift onto the earlier received copy. To combat these two problems, OFDM uses the *guard time* at the beginning portion of the symbol time and FFT is done only on the non-guarded time portion of the symbol time. In general, the guard time should equal to two to four times the average delay spread which turns out to be 800 ns. However, symbol duration should always be higher than the guard interval [6].

The introduction of multi-carrier transmission with OFDM is the reason for the increase in speed from IEEE 802.11b to IEEE 802.11a/g. It is also used with MIMO systems that enabled the transitions from IEEE 802.11a/g to IEEE 802.11n. Even with IEEE 802.11ac, the way of transmitting data over the air remains the same hence is yet based on OFDM transmission.
In each of these standards, OFDM distributes the incoming data bits among the subcarriers in the same manner. However, the difference is due to the increased number of available subcarriers used for independent transmission and the number of reserved pilot *carriers* that depends on the available channel width in that standard.



Figure 2.5: Channel layout in different standards

In Figure 2.5 [6], the channel layout in terms of their OFDM data and pilot carriers is shown. Each horizontal line represents the layout of OFDM subcarriers in one type of channel and a dip in the line represents the pilot carriers that do not carry any data. This first channel used with OFDM was the 20 MHz channel that has now extended up to the widest channel i.e. 160 MHz in IEEE 802.11ac.

Table	2.1:	Carrier	distribution	in	different standards
-------	------	---------	--------------	----	---------------------

PHY standard	Subcarrier range	Pilot Subcarrier	Subcarriers (total/data)	Capacity relative to 802.11a/g
802.11a/g	-26 to -1,	±7, ±21	52 total, 48 usable	x1
	+1 to +26		(8%pilots)	
802.11n/	-28 to -1,	±7, ±21	56 total, 52	x1.1
802.11ac 20Mhz	+1 to +28		usable(7% pilots)	
802.11n/802.11ac	-58 to -2,	±11, ±25, ±53	114 total, 108	x2.3
40Mhz	+2 to +58		usable (5% Pilots)	
802.11ac 80Mhz	-122 to -2	$\pm 11, \pm 39, \pm 74,$	242 total ,234	x4.9

PHY standard	Subcarrier range	Pilot Subcarrier	Subcarriers (total/data)	Capacity relative to 802.11a/g
	+2 to +122	±103	usable (3 % pilots)	
802.11ac	-250 to -130, -126	±25, ±53, ±89,	484 total,468	x9.75
160Mhz	to -6, +6 to +126,	$\pm 117, \pm 139, \pm 167,$	usable (3 %	
	+130 to +250	±203, ±231	Pilots)	

Table 2.1 [6] summarizes all the different OFDM carrier numbering and pilot channels used in different standards. It can be seen that the range of the subcarriers depend on the channel width. The data carrying capacity of each subcarrier is the same in all the standards, so more the subcarrier the better the capacity. As pilot carrier do not carry any data information and is used to measure the channel information, they are considered as protocol overhead. However, from the table it can be seen that as the channel width increases, the amount of channel devoted to pilot carriers decreases. This results in a more efficient system. The table also shows the relative throughput in different standards as compared to that in IEEE 802.11a/g.

2.2.4 Modulation and Coding

Modulation describes the number of bits that are contained within one transmission time increment [6]. Higher the modulations more data can be packed into the transmission, however, this requires higher signal-to-noise ratios. Most of the channels do not operate in error free conditions and, therefore, error correction codes are incorporated in conjunction with OFDM known are *coded OFDM* (COFDM). These coded OFDM uses *Forward Error Correction* (FEC) on each channel with the help of which the receiver can detect the corrupted bits and repair the transmission as long as the lost or corrupted bits are below a certain level [9].

In general, convolution codes are used as the FEC because the frames transmitted over wireless LANs can be of different sizes. The two parameters that govern convolution codes are *constraint length* and *coding rate*. The constraint length determines the duration of time over which the data bits should be averaged into successive transmissions to improve reliability [9]. Code rate (R) determines the number of redundant bits added and is given by the ratio of the number of data bits transmitted to the total number of coded bits. For instance, a code rate of ¹/₂ means one transmitted bit for every two coded bits [18].

Most of the OFDM PHYs in different standards (IEEE 802.11a/g/n/ac) makes use of the convolution codes to support errors-correcting. In IEEE 802.11ac, LDPC (Low Density Parity-Check) coding is supported as an optional feature [6].

In addition, a combination of different parameters defines the available data rate offered by that standard. For instance, in IEEE 802.11a/g, the combination of the modulation and coding determines the data rate. In IEEE 802.11n, the combination depends on a number of factors: number of spatial streams, type of RF modulation, coding rate, channel width, and the guard interval. This combination is represented by a single index called MCS (Modulation and coding scheme). In IEEE 802.11ac, MCS is no longer tied to the channel width as in IEEE 802.11n. It only depends on the modulation and the code rate and to determine the link speed, the knowledge of the MCS must be combined with the channel width to produce an overall data rate.

In the following tables we list the different data rates that are offered in different standards using different modulation and coding schemes:

Table 2.2 [9] lists the available data rate in the IEEE 802.11a standard
--

Speed (Mbps)	Modulation and	Coded bits	Coded bits	Data bits
	Coding rate (R)	per carrier	per symbol	per symbol
6	BPSK, $R=1/2$	1	48	24
9	BPSK, $R=3/4$	1	48	36
12	QPSK, $R=1/2$	2	96	48
18	QPSK, R=3/4	2	96	72
24	16-QAM, R=1/2	4	192	96
36	16-QAM, R=3/4	4	192	144
48	64-QAM, R=2/3	6	288	192
54	64-QAM, R=3/4	6	288	216

 Table 2.2: Possible data rates in IEEE 802.11a

Table 2.3 [12] gives the list of some of the supported MCS values used in IEEE 802.11n. The given rates in the table uses the same modulation in all the subcarrier, however, IEEE 802.11n also adds the ability to use different modulation methods for different spatial streams. This can dramatically increase the number of supported data rates however they are optional.

MCS Index	Туре	Coding Rate	Spatial Streams	Data Rate (Mbps) with 20 MHz CH		s) Data Rate (Mbp with 40 MHz CH	
				200	400ns	800	400ns
0	DDCV	1/	1	800 ns	(SGI)	800 ns	(SGI)
0	BPSK	1/2	1	0.5	1.2	13.5	15
1	QPSK	1/2	1	13	14.4	27	30
2	QPSK	3/4	1	19.5	21.7	40.5	45
3	16-QAM	1/2	1	26	28.9	54	60
4	16-QAM	3/4	1	39	43.3	81	90
5	64-QAM	1/2	1	52	57.8	108	120
6	64-QAM	3/4	1	58.5	65	121	135
7	64-QAM	5/6	1	65	72	135	150
8	BPSK	1/2	2	13	14.4	27	30
9	QPSK	1/2	2	26	28.9	54	60
10	QPSK	3/4	2	39	43.3	81	90
11	16-QAM	1/2	2	52	57.8	108	120
12	16-QAM	3/4	2	78	86.7	162	180
13	64-QAM	2/3	2	104	115.6	216	240
14	64-QAM	3/4	2	117	13000	243	270
15	64-QAM	5/6	2	130	144.4	270	300
16	BPSK	1/2	3	19.5	21.7	40.5	45
31	64-QAM	5/6	4	260	288.9	540	600

Table 2.5. Some IEEE 002.1111 MCS values	Table	2.3:	Some	IEEE	802.11n	MCS	values
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Table 2.4 [6] lists the IEEE 802.11ac MCS values. It continues to adopt some of the modulation and coding schemes (MCS) of IEEE 802.11n like 0 to 7 which is considered mandatory in IEEE 802.11ac with a regular guard interval of 800ns. However, the use of guard interval of 400ns is considered optional. Furthermore, two more MCS values of 8 and 9 are introduced that used 256 QAM with code rate of 3/4 and 5/6 to further enhance the data rate to 20% and 33% respectively as compared to the 64 QAM code rate of 5/6 and they are considered as optional [13].

MCS index value	Modulation	Code rate (R)
0	BPSK	1/2
1	QPSK	1/2
2	QPSK	3/4
3	16-QAM	1/2

Table 2.4: IEEE 802.11ac MCS value

MCS index value	Modulation	Code rate (R)
4	16-QAM	3/4
5	64-QAM	2/3
6	64-QAM	3/4
7	64-QAM	5/6
8	256-QAM	3/4
9	256-QAM	5/6

2.2.5 Multiple Input Multiple Output (MIMO)

MIMO is the technology that allows simultaneous data transmission through several antennas i.e. multiple antennas both at the transmitter and the receiver side. When only one antenna is used at the transmitter and one at the receiver, in an indoor environment, multipath interference occurs. In traditional WLAN technology, multipath was a severe problem because the receiver received multiple copies of the original signal and thus sorting out the original message was difficult. However, with MIMO, things got reversed and with the help of multiple receiving antennas, multipath became advantageous. This means that when MIMO is used, multipath messages can be sorted out more easily and can improve reliability.

Reliability is the key advantage of using multiple antennas that could be obtained through diversity. Furthermore, the transmitter can use transmit beamforming or adaptive antenna array to improve the quality of the link [14]. In addition, it can be used to achieve higher data rate through spatial multiplexing. These concepts are briefly discussed below:



Figure 2.6: MIMO [15]

Spatial Multiplexing(SM)

MIMO system can be used to transmit individual data streams on different antenna. This is done by splitting the data bits into multiple streams and sending them simultaneously over the antennas. Each transmission propagates along different direction and these spatial streams reach the receiver with different delay and strength. When multiple spatial streams are used along with MIMO, the technique is known as Space Division Multiplexing (SDM). Thus with MIMO/SDM system, the increase in data rate is directly proportional to the number of parallel streams that are used or multiplexed into a single channel.



Figure 2.7: IEEE 802.11n Signal Processing Techniques that use MIMO Antennas [15]

Space- Time Block Coding (STBC)

In this technique the signal stream is send redundantly using the spatial streams. They are coded differently and are transmitted with the help of different antenna. After the comparison of these spatial streams, the receiver accurately determines the original signal in the presence of distortion and interference. STBC reduces the error rate at a given value of SNR and, thereby, improves reliability. Thus, with STBC redundancy is added to the system for reliability and robustness.

Transmit Beam forming (TxBF)

In this technique, the transmitter uses extra antennas to steer the streams to the receiver by concentrating the Radio Frequency (RF) transmitted energy in one particular direction. To use this technique, the transmitter must know the relative location of the receiver or the channel between them. The channel information can be obtained either implicitly (assuming identical propagation

at both the ends) or explicitly (using some kind of receiver feedback). The limitation on the transmitter side can be overcome by making use of a technique known as *Maximal Ratio Combining* (MRC). Using this, the extra antennas at the receiver end can add to improve the reception by optimally combining the received signal.

When beam forming is not used, a lower MCS value is selected for all the spatial streams to meet the channel conditions. However, when beam forming is used with a subsequent separation between the spatial stream, different MCS can be applied to individual stream to improve the overall throughput [14].

2.2.5.1 MIMO OFDM

The new wireless system should be able to provide high data rate with high performance over challenging channel conditions (time-selective and frequency-selective channels). MIMO combined with OFDM has the potential to meet these strict requirements. On the one hand, MIMO can boost the capacity and the diversity, on the other hand, OFDM can mitigate the detrimental effects due to multipath fading [16].



Figure 2.8: A MIMO-OFDM transceiver

In Figure 2.8, a simple MIMO-OFDM system is shown with M_t transmit antenna and M_r receive antennas and N-tone OFDM. At first, with the use of some form of modulation (say QAM), the incoming bits are mapped to a number of data symbols. Then, the blocks of these data symbols S are coded with a code word matrix C (size NT x M_t). These symbols will then be sent using M_t antennas in *T* OFDM blocks where each block consists of *N* sub channels. From each transmitting antenna *j*, codes c_{j}^{1} , c_{j}^{2} ,..., c_{j}^{T} will be transmitted in OFDM blocks 1, 2 ...T; this is done after appending the cyclic prefix on each OFDM block. These blocks pass through the MIMO channels and are then received by the receiver. These signals are first sent to the reverse OFDM block for removing the cyclic prefix and then sent to the decoder. Maximal likelihood (ML) detection can be done if the channel state information (CSI) is available at the receiver [16].

2.2.5.2 MU-MIMO

Using MU-MIMO, terminals are allowed to transmit/receive signals to and from multiple users in the same frequency band simultaneously as opposed to the single user MIMO where transmissions can occur only between two users. Figure 2.9(a) presents the single-user MIMO where four simultaneous streams are used to boost data speed and the multiuser version uses a pair of 2x2 streams for two users as in Figure 2.9(b).



⁽a) Single User MIMO, 4 streams

Figure 2.9: Single User MIMO versus Multi user MIMO

MIMO is one of the significant technologies used in IEEE 802.11n. The IEEE 802.11n APs must use two spatial streams but the maximum allowed is four, whereas the IEEE 802.11n stations can use one spatial stream as the minimum. Each spatial stream can transmit up to 150 Mbps of PHY layer throughput. This allows for two streams to transmit 300 Mbps, three streams up to 450 Mbps, and four streams of 600 Mbps (which is the maximum theoretical allowed throughput in IEEE 802.11n).

Spatial multiplexing is mostly used in IEEE 802.11n, however, STBC and TX beamforming are optional features of IEEE 802.11n [15].

⁽b) Multi User MIMO, 2 users, 2 streams each

IEEE 802.11ac is also based on MIMO and makes only one spatial stream as mandatory. Although it can go up to eight-stream operation, however, the use of more than one spatial stream is regarded as optional. STBC is also considered as an optional feature in IEEE 802.11ac, while TX beamforming will be used in case of MU-MIMO. However, it is considered as an optional feature.

MIMO is combined with OFDM and is used as MIMO-OFDM in both IEEE 802.11n and IEEE 802.11ac.

2.2.6 Channel Bonding

It is a technique where two adjacent contiguous channels are combined to form a wider channel. In 2.4 GHz, due to the presence of only 3 non-overlapping channels, channel bonding is not recommended. However, in 5 GHz band, 24 non-overlapping channels are available that can permit 40 MHz operations [15].

When channel bonding is used, the gaps between the channels are also utilized resulting in a data rate that is more than double than the one available when a single channel is used. For instance in IEEE 802.11a, the data rate is 54 Mbps for a single transmitter on a 20 MHz channel. However, with 40 MHz bonded channel, for a single transmitter the data rate is 135 Mbps which is more than double as compared to 54 Mbps.

IEEE 802.11n uses channel bonding. It defines two modes of transmission for a station i.e, transmission over 20 MHz and 40 MHz bandwidth. The wider channel is formed by combining contiguous or non-contiguous channels. For instance, a 40 MHz channel is formed by two contiguous 20 MHz channel. IEEE 802.11n uses the concept of primary and secondary channels in this band formation. One of the 20 MHz band is considered as the primary channel and the rest are considered as the secondary channels. Due to the use of two different types of channel, the IEEE 802.11n devices should be able to detect transmission on both the channels. In IEEE 802.11n, two networks can be overlapped and deployed. However, they always needed to be assigned the same primary channel [17].

IEEE 802.11ac also uses channel bonding in the similar way as IEEE 802.11n. For instance a 40 MHz channel is formed by two contiguous 20 MHz channel whereas 80 MHz is formed by two contiguous 40 MHz channel. However, for 160 MHz band formation either contiguous or non-

contiguous lower and higher 80 MHz can be combined. IEEE 802.11ac uses the concept of primary and secondary channels in all these band formation. As in IEEE 802.11n, one of the 20 MHz band is considered as the primary channel and all the rest are considered as the secondary channels. This allows only specific sets of primary and secondary channels to be used to obtain a wider channel bandwidth. Due to the usage of two different types of channels, the IEEE 802.11ac due to the enhancement in secondary channel CCA capabilities, two networks can be deployed without overlap which was not the case in IEEE 802.11n.

The use of 20 MHz, 40 MHz and 80 MHz channels is mandatory in IEEE 802.11ac whereas 160 MHz and 80+80 MHz operation is optional and not supported by the IEEE 802.11ac first wave devices.

2.2.7 Summary

The following table summarizes the different Phy features and their adoption in different standards.

Standards	Modulations	Data rates In Mbps	Frequency band	Bandwidth	Advances antenna technologies
802.11	DSSS, FHSS	1,2	2.4 GHz	20 MHz	N/A
802.11b	DSSS/CCK	5.5, 11	2.4 GHz	20 MHz	N/A
802.11a	OFDM	6-54	5 GHz	20MHz	N/A
802.11g	OFDM, DSSS/CCK	1-54	2.4 GHz	25 MHz	N/A
802.11n	OFDM	6-600	2.4 GHz and 5 GHz	20 and 40 MHz	MIMO up to 4 spatial streams
802.11ac	OFDM	600-3470 for wave 1 867-6930 for wave 2	5 GHz only	20, 40,80 for wave 1 20,40,80,160 for wave 2	MIMO, MU- MIMO up to 8 spatial streams
802.11ad	Single carrier and OFDM, Low Power Single Carrier (LPSC)	7168	60 GHz	2.16 GHz	Beamforming

Table 2.5 : Phy features in IEEE 802.11 standards

One of the key functional requirements that come with any new amendment is backward compatibility. When a new PHY is incorporated in a standard, different framing are used and one of their key requirement is to remain compatible with the previous PHYs. Below we provide a detailed account of the physical level framing that is used to maintain the backward compatibility.

The IEEE 802.11g devices using OFDM were made to be backward compatible with the IEEE 802.11b that uses different modulation (CCK) by either allowing the IEEE 802.11g stations to send at lower rates or use some protections frames (RTS/CTS, CTS to self (see section Figure 2.12). These frames are also sent using the IEEE 802.11b rules (PSK (Phase Shift Keying) at 1 Mbps or 2 Mbps) or use CCK at 5.5 or 11 Mbps.

The new devices using IEEE 802.11n are made compatible with the legacy IEEE 802.11a/g by defining waveform that is backward compatible with IEEE 802.11a and OFDM modes of IEEE 802.11g. An additional legacy preamble is added in the transmission using the mixed frame format where the waveform begins with the IEEE 802.11a/g preamble. It has the original non-HT (High Throughput) format of an OFDM frame that contains the Non-HT Short Training Field (L-STF), Non-HT Long Training Field (L-LTF) and Non-HT Signal Field (L-SIG) of the IEEE 802.11a/g as seen in the Figure 2.10b. This allows even the legacy devices to understand the IEEE 802.11n mixed format packets and defer their transmission based on the length specified in the signal field; however, the whole IEEE 802.11n packet cannot be decoded. The rest of the IEEE 802.11n mixed frame format contains the HT short training field (HT-STF), additional HT long training fields (HT-LTFs) and HT signal field (HT-SIG) that is needed for MIMO training and signalization. For 40 MHz operation to be backward compatible with the 20 MHz operation, the preamble of 40 MHz waveform that is identical with the 20 MHz, is repeated on the two adjacent 20 MHz channel.

For IEEE 802.11ac, to meet the interoperability requirement, VHT (Very High Throughput) physical layer frame were introduced that is similar to the one used in IEEE 802.11n for mixed mode operation as seen in Figure 2.10c. It begins with the same field as in IEEE 802.11a frames. One more addition header is incorporated in the IEEE 802.11ac frame to support the MU-MIMO transmission because the one used in IEEE 802.11n HT-SIG was not extensible to the new channel width and no information could be provided about the large number of spatial streams.

Figure 2.10 below shows the various frame format used in different standard, however the detail about each field in the header can be found in [6].

From the figure, we can see that in order to maintain backward compatibility, all the follow on technology starts with the same fields as IEEE 802.11a frames so that when one type of device transmits all the others can know about it and avoid simultaneous transmission.



⁽c) VHT mode (802.11ac)

←	OFDM PHY modulation (identical for all users)					VHT mod (received p	ulation per-user)	
	8 µs	8µs	4µs	8µs	4μs	8µs	4µs	Variable
	L-STF	L-LTF	L-SIG	VHT-SIG-A	VHT- STF	VHT-LTF	VHT SIG-B	Data



2.3 IEEE 802.11: MAC evolutions

In the previous section, we glanced over the evolution that have taken at the PHY and the associated IEEE 802.11 standards that adopted them for performance upgradation. In the following section, we will present the evolution in the MAC pertaining to the channel access scheme and other enhancements that have been added in parallel to the PHY enhancements, in the various generation of IEEE 802.11 standard. Before describing this, the basic modes of operation available in IEEE 802.11 are explained.

Infrastructure mode

An infrastructure mode network requires an Access Point (AP) for any kind of communication between two devices. The AP serves the users within its coverage area, however, the transmission range is limited by the one hop connections [18].

Adhoc mode

An adhoc network allows different devices to connect to each other directly without the need of a central controlling device or AP.

Device to device communication mode

Although above two are the main modes of operation in WLAN, however, there are upcoming new architecture trends that try to incorporate adhoc mode in IEEE 802.11 in both APs and mobile nodes. This is done to extend the traditional networks to multi hops. One of such architecture is WIANI [18]. In addition, a more practical and recently developed technology is Wi-Fi-direct by the Wi-Fi alliance that allows device to device connection without the presence of an access point. The IEEE 802.11z [19] also allows for a kind of device to device communication provided that the stations are connected to the same AP. On the other hand, Wi-Fi-direct instead of using the basic adhoc mode build up an IEEE 802.11 infrastructure mode with the devices negotiating about the device that will act like an AP [20].

2.3.1 IEEE 802.11 legacy MAC

The legacy IEEE 802.11a/b/g MAC defines two types of channel access mechanisms: Distributed Coordination Function (DCF) and Point Coordination Function (PCF). The former is the fundamentally used access scheme and is based on CSMA/CA. The latter is based on polling and is optional in IEEE 802.11 wireless devices [21].

In basic CSMA, each station first senses the channel (busy or idle) for a certain time before transmission. In case the channel is idle, station transmits the data, however, if it is sensed busy it keeps on sensing the channel until it becomes idle. This can thus reduce the chance of collisions and improve the throughput [22].

To determine if the channel is available, carrier sensing mechanisms are used. Two different mechanisms are used for carrier sensing are:

Physical carrier sensing: In this type of sensing, a node that needs to transmit first detects the signal strength from other sources. If the energy or the signal strength is above a given threshold called the carrier sense threshold, the medium is considered to be busy. If not, the channel is considered idle and free for transmission

Virtual carrier sensing: In this type of sensing, the Network Allocation Vector (NAV) is used to determine if any transmission is going on in the channel. Any station that is willing to transmit sets the NAV, by updating the duration field in the header of the transmitted frame equal to the time period needed to complete the frame transmission. Other stations listening to the channel check this duration field to update their own NAV timer and starts counting down to 0 (which means end of current transmission). Thus, whenever the NAV has non-zero value the channel is considered to be busy. NAV may be also set by using special handshake signals like RTS/CTS that can help to reserve the channel (see section 2.3.1.1).

Different variants of CSMA exist depending on how the transmissions are delayed. Following are some of its type.

Non-persistent CSMA

In this type of CSMA, station starts the transmission as soon as it finds the channel to be free. However, when packets arrive and the channel is busy, instead of continuously sensing the channel it reschedules its transmission by a random distribution and again starts the sensing process[22]. The disadvantage of this approach is the delay incurred during the loaded conditions [23].

P-persistent

It is basically used in the slotted channel and attempts to reduce the transmission delays. In this type of CSMA, a station transmits the packet with probability 'p' or with probability '1-p' depending upon whether the medium is free or busy, respectively [22]. The value of p can be constant or dynamic. Smaller values of p will result in lesser collision among the pending

transmission of different station, however, in this way, the average transmission delay is increased [23].

1-persistent

In this type of CSMA, station always transmits the data whenever it finds the channel idle, (i.e., transmits with a probability of 1). In case the medium is busy, the station senses the medium, waits for it to be free and repeats the algorithm. When two of more stations transmit at the same time, collision occurs and in such case the station waits a random amount of time to repeat the same algorithm [22].

Collision Avoidance (CA)

CSMA/CA was designed with the aim of avoiding collisions at the start of transmission. Here, a station cannot start any transmission until the channel is free for a predestined time. Thus CSMA/CA prevents collision that may occur when new packets arrive at the starting of another station's transmission, hence, improving the performance [23].

2.3.1.1 Distributed Coordination Function (DCF)

Under the DCF operation, two techniques for packet transmission are defined. One is based on the basic access and the other is based on the optional four way handshaking known as RTS/CTS mechanism [24].

Before giving a detailed account of these mechanisms, it is necessary to know about the different interframe space (IFS) time period used in IEEE 802.11. The IFS is the shortest time duration that a station waits after the end of the frame and before starting any frame transmission [25]. Four different IFS intervals are defined in IEEE 802.11 to provide priority access to the medium. They are the following

- *Short IFS (SIFS):* It is the smallest time period used for high priority frames like RTS/CTS, acknowledgements.
- *Point Coordination Function IFS (PIFS):* It is the time period used for time-bounded service in contention-free period using PCF (see section 2.3.1.2).
- *DCF-IFS (DIFS):* It is basically used in the DCF scheme and is the minimum idle time used in contention based services.

- EIFS (Extended IFS): It is used only when error occurs in transmission.

```
SIFS<PIFS<DIFS<EIFS
```

Basic DCF

In the basic DCF access mechanism, a station can transmit a packet if the medium is found to be idle for more than DIFS period as shown in Figure 2.11. If the medium is idle for this duration, the station gets access to the channel and may start transmitting a new or a pending frame. However, if the medium is found to be busy, the station waits for a random time called the backoff time [11]. The random backoff value ranges between CWmin and CWmax, where CW is the contention window size and is decremented periodically for every slot sensed to be idle. In case the medium is sensed busy for that slot, the timer freezes. Furthermore, when a number of stations compete for the same channel, the station that has the shortest backoff will be first get access to the channel. After the frame is successfully received at the destination, the receiver sends an ACK to the source station after a SIFS period. When two or more stations decrement their timer to zero simultaneously or if no ACK is received after the packet transmission, collision is said to occur and the station re-enters the backoff process with a new backoff timer value.



Figure 2.11 : Basic DCF operation

Four-way Handshaking with RTS/CTS

In this protocol before the start of any data-packet transmission, a successful handshake between sender and receiver is done by means of the exchange of control packets. Even though this protocol does not totally reduce collision, it wastes very less throughput during collision. The sending station broadcasts an RTS packet if it finds the channel free and if no CTS is heard from nearby station. The RTS packet includes both the intended destination of the ensuing data packet and the amount of channel time required for its transmission prior to sending the real data. If the destination station successfully receives the RTS packet, it responds with a CTS packet, which is broadcast in nature and contains the channel time required for the transmission of new packet. The CTS packet is an indication to the sending station to start transmitting and it also inhibits the neighboring stations from interfering during that period. The destination station then completes the four way handshake by sending an ACK to the sending station after receiving the packet without any errors [23].



Timeline with RTS/CTS

Figure 2.12: RTS/CTS

CTS to Self

In this protocol, whenever a station has anything to transmit, it first sends a CTS frame with the destination address as its own MAC address. This will allow the other CTS to update the NAV of all the other station that are sensing the channel. With the help of this, all other stations will be aware about the channel occupancy for the transmission of the CTS, the data frame and ACK.

The CTS frame is sent using the modulation that can be understood by all the stations. Typically they are used as protection frames to allow coexistence between IEEE 802.11b and IEEE 802.11g devices.



Figure 2.13 : CTS to self

2.3.1.2 Point Coordination Function (PCF)

It is based on "poll-and response mechanism" and is basically used in the infrastructure mode. PCF requires an AP to act as a point coordinator (PC) as it uses centralized polling method. In a PCF enabled channel access scheme, time is divided into two periodic intervals, the Contention Free Period (CFP) and the Contention Period (CP). PCF enjoys a higher priority than DCF by using the PIFS interval that allows the PC to gain access to the medium earlier than the others. In addition, the PC can select the length of CFP and distribute this information within the beacon frames as seen in the Figure 2.14. This sets the NAV in all the other stations within the range of the AP. The PC coordinates or polls the stations according to some list to permit a contention free access to the medium. All the stations are allowed to transmit a maximum length of frame in CFP. Each successful transmission needs to be acknowledged and the retransmission of any unacknowledged frames is possible in the CFP after PIFS duration [26].



Figure 2.14: PCF access method

2.3.2 MAC in IEEE 802.11e

In the DCF scheme, all station has the same priority as they all use the same value of CWmin. This does not allow for QoS support as there is no mechanism to differentiate between stations [11]. With an aim to support QoS features and multimedia services, the IEEE 802.11e specification was added to the existing IEEE 802.11 standard. The new protocol is able to prioritize between different flow types and provide guaranteed resources for specific flows [27]. The IEEE 802.11e STA implements four access categories (ACs) and this prioritization is ingrained from IEEE 802.1d specifications [28].

The IEEE 802.11 WLAN introduces the IEEE 802.11e Hybrid contention function (HCF) to support QoS. The HCF consists of two types of access mechanism: contention based channel access and controlled channel access. The Enhanced Distributed Channel Access (EDCA) is contention based and HCF controlled channel access (HCCA) is controlled based. EDCA is an extended variant of DCF which provides priority based service differentiation whereas HCCA enhances the polling based PCF scheme to support guaranteed channel access to certain flows based on the QoS requirements. EDCA is based on random access and only provides service differentiation and hence cannot provide bandwidth guarantee or meet delay requirements. However HCCA can provide higher QoS assurances, however, needs to be centrally controlled.

In the super frame used in IEEE 802.11e, there is the contention free period (CFP) and the contention period (CP) that alternates periodically over time. EDCA only uses the CP while HCCA can be used in both the periods [27].

EDCA Mechanism

EDCA is designed to provide prioritized QoS in WLANs. It enhances the IEEE 802.11 DCF function and provides station with a distributed access to the wireless medium with up to 8 levels of priority corresponding to different traffic categories (TC). This prioritization comes from the 802.1d bridge specification. To realize QoS support, EDCA defines four different types of access categories (AC) each of which is assigned a first-in first out (FIFO) queue. Each packet data from the higher layers along with its TC are mapped to a corresponding AC as shown in Table 2.6 below.

Prioirity	User Prioirty (Same as 802.D User Prioirty)	802.1D Designation	Access Category	Designation (Informative)
lowest	1	BK	AC_BK	Background
	2	-	AC_BK	Background
	0	BE	AC_BE	Best Effort
	3	EE	AC_BE	Best Effort
	4	CL	AC_VI	Video
	5	VI	AC_VI	Video
•	6	VO	AC_VO	Voice
Highest	7	NC	AC_VO	Voice

 Table 2.6: Priority Mapping

Each AC is an enhanced variant of DCF called Enhanced Distributed Channel Access Functions (EDCAF) that contends for the transmission into the wireless channel. Each AC behaves like a virtual station and is assigned individual values for CWmin, CWmax. To further distinguish the service, different AC is provided with different IFS values called AIFS. Each AC starts its backoff period after sensing the medium idle for its corresponding AIFS period just as the DIFS period in DCF as shown in Figure 2.15.

The AC with the highest priority is assigned the lowest value of AIFS and the value of AIFS is calculated as

AIFS[AC]= SIFS+AIFSN[AC] x Slots

Where, AIFSN[AC] is a positive integer and slots is the time unit related to the physical layer [11].

The three EDCF parameters AIFS [AC], CWmin [AC], and CWmax [AC] are announced in the beacon frames by the AP. The use of different values of these parameters for different AC allows the AC with low priority to have a short waiting time to access the channel as compared to the high priority ones. When the backoff values of two or more AC reaches zero at the same time, an internal collision is said to occur. This is handled by a virtual collision handler within the station that allows the high priority AC to transmit frames while the other performs a backoff with a new value of CW.



Figure 2.15 : EDCA mechanism

In addition to the above mention three parameters, fourth parameter is also associated with each AC. It is called the TXOP and is defined as the time interval when a QoS station (QSTA) has the right to transmit frames into the channel. This is also known as contention frees bursting (CFB). During a TXOP limit, a station can send multiple MPDU (media access control protocol data unit) from the same AC with a SIFS time gap between an ACK and the following frame. A TXOP limit corresponding to a value 0 means that only one MPDU frame may be transmitted during the TXOP duration [29].

AC	CWmin	CWmax	AIFSN	TXOP Limit(ms)
(0)AC_BK	15	1023	7	0
(1)AC_BE	15	1023	3	0
(2)AC_VI	7	15	2	3.008
(3)AC VO	3	7	2	1.504

 Table 2.7 : Default EDCA AC Parameter Set

HCCA Mechanism

Although EDCA improves the differentiating mechanism, under high traffic load it is not sufficient to provide enough traffic protection. Thus HCCA extends the EDCA rule by allowing the highest priority to access the channel in both the CFP and the CP [11]. HCCA uses a QoS aware hybrid coordinator at the QoS Access Point (QAP) in infrastructure WLAN. The QAP uses the PIFS to gain access to the channel before other stations. It then coordinates QSTAs and allocates the TXOP to the stations (either HCCA TXOP or polled TXOP) in the Controlled Access Phase (CAP). In HCCA, the QAP can poll the station even in the CP, unlike in PCF. It then takes into account the stations flow requirement.

A station requiring a QoS guaranteed service queries the HC for a reservation. It sends this information to the HC in the *Traffic Specification* (TSPEC) and the HC determines if the QoS reservation request can be admitted or dropped by its admission control unit (ACU). For a reservation request that is admitted by the HC, a TXOP is allocated to the station depending upon the scheduling scheme [27].



Figure 2.16 : The CAP/CFP/CP periods

2.3.3 MAC in IEEE 802.11n

To break the 100 Mb/s throughput barriers, different MAC enhancements were added to the IEEE 802.11n MAC as the method of increasing the efficiency. However, the channel access mechanism is still based on CSMA/CA and only some small modification is incorporated when different channel sizes are used for channel access. In the following section a detail of these MAC enhancements and the updated channel access scheme is presented.

2.3.3.1 MAC enhancements in IEEE 802.11n

In IEEE 802.11n MAC layer, novel mechanisms like frame aggregation, block acknowledgement has allowed the transmission of several frame for each transmission opportunity. This leads to the reduction of MAC/PHY overhead and together can provide a bulk of throughput enhancement as compared to that achievable with the earlier standard IEEE 802.11a/b/g. Enhancement at the MAC layer also included the use of *Reverse Direction* (RD) protocol. A new interframe space called *RIFS* was used during the transmission of burst frame instead of SIFS that also reduced further overhead [14]. A detail of some of these enhancements is given below.

Aggregation

Aggregation is one of the key features that can improve the efficiency at the MAC layer. It reduces the transmission time of the headers and preamble. In addition, it also reduces the waiting overhead needed for each successive transmission of frame. A detailed explanation of the various aggregation scheme used in IEEE 802.11n is given in section 3.2.

Block Acknowledgement

Block Acknowledgement is used to replace the ACK frame that was used in the earlier standard of IEEE 802.11(b/g). It is explained in section 3.2.4.

Reverse Direction

It is an enhancement in IEEE 802.11n that allows the holder of the TXOP give a part of its TXOP to its peer station, creating a "bi-directional TXOP". This scheme can be used in two ways. First, it can be used when the transmitter creates a room for a response frame to be sent by its peer at the end of TXOP. Second, when the recipient gets its own TXOP after transmitting the response frames, and then hands it over to its peer. This mechanism is efficient to be used with applications like HTTP and FTP which are highly asymmetric (forward direction traffic is more than the reverse direction traffic). However, the transfer speed is dependent on the delay of transfer even in the reverse direction [30]. There is less signaling associated with it as the reverse data transmission request may be piggybacked in the CTS control frame and this can increase the TCP performance that involves the transmission of TCP ACK segments [39].

2.3.3.2 Channel access in IEEE 802.11n

Applications like voice video are perceived as the most promising applications for IEEE 802.11n. This has led to a functional requirement that mandates IEEE 802.11e features to be incorporated within IEEE 802.11n station [14]. Thus, IEEE 802.11n implements all the basic mechanism used in IEEE 802.11e. Furthermore, it includes two modes of transmission that may introduce some modification in the channel access nevertheless; the main mechanism of channel access is still based on CSMA/CA.

The following are the basic channel access rules that are followed in IEEE 802.11n. A frame can be transmitted only if the medium is idle. In IEEE 802.11n, to determine whether the medium is idle requires the knowledge of the channel width that is used for transmission. The sharing of wider channels depends on the ability of the IEEE 802.11n devices to detect devices in both the primary and secondary channels. In the secondary channel, there may be other IEEE 802.11n or IEEE 802.11b/g networks being deployed with different devices operation over that channel. Thus the following rules apply:

- To transmit a 20 MHz frame on 20 MHz channel, CCA is done only at the primary channel
- To transmit a 40 MHz frame on primary 40 MHz channel, CCA is done on the primary and CCA check should also pass for the secondary 20 MHz channel.

Thus, for the 20 MHz bandwidth operation, same DCF principle is followed that exists for the legacy devices where each station checks if the channel is free for DIFS period. However, for a 40 MHz transmission the following new rules are defined:

When an IEEE 802.11n station needs to use a 40 MHz bandwidth, it first checks if its primary channel has been idle for DIFS period plus the backoff time. Now, if the primary channel is found to be idle for DIFS, the station checks if the secondary channel had been idle for PIFS period of time just before the expiration of the backoff counter. In case both these conditions are met, the station can send a 40 MHZ signal [17]. However, if the secondary channel is found to be busy, there are two possible operation modes as shown in the Figure 2.17.



Figure 2.17 : (a) static bandwidth operation (b) dynamic bandwidth operation

Static 40 MHz operation: For this, the station should retry to access the 40 MHz channel by using a new random number from its current collision window. Only after the entire secondary channels are idle, the station can continue the channel access attempts [17].

Dynamic 20/40 MHz operation: For this the station may only utilize the primary channel to transmit a 20 MHz signal. The station may transmit data over 40 MHz depending upon the availability of the secondary channels as obtained through CCA. However it should be noticed that the second channel for one AP may be the primary channel for the other AP when they are deployed together within an area [17].

2.3.4 MAC in IEEE 802.11ac

With the most recent IEEE 802.11ac standard, network throughput is improved by allowing the simultaneous transmission of multiple downlink traffic to different users. At the MAC level, it promises a throughput of more than 500 Mbps for a single user and aggregated MAC throughput of more than 1 Gbps for multi-user scenario. Both these utilizes a maximum of 80 MHz channel bandwidth.

Most of the MAC features in IEEE 802.11ac are evolutionary and they are enhanced versions of the ones introduced in IEEE 802.11n, like aggregation, block acknowledgement, etc. There have not been significant changes in the way how stations access the medium. However, there have been changes in the rules to determine a clear channel due to the introduction new channel

bandwidth of different sizes, new mechanism to share the obtained TXOP resource. In addition, rules are also added to the RTS/CTS mechanism to allow devices to indicate and adapt to different available bandwidth.

2.3.4.1 IEEE 802.11ac MAC features

Some of the MAC features introduced in IEEE 802.11ac are as follows:

Enhanced Aggregation

The IEEE 802.11ac enhances the aggregation scheme of IEEE 802.11n to allow more packets to be aggregated. Here, the maximum aggregation limit has been increased to further improve the MAC efficiency with the increasing PHY data rate. For the A-MSDU aggregation, the maximum allowed size has increased from 7935 to 11426 bytes, and for A-MPDU, it has increased from 65535 to 1048579 bytes [3].

MU TXOP-sharing

One of the key new features introduced in IEEE 802.11ac is the MU-MIMO features. According to this, an AP can simultaneously transmit multiple frames to multiple STAs over the same frequency spectrum by using the beam-forming scheme that did not find much use in IEEE 802.11n. In multi user beam-forming, each station receives only the spatial streams directed to it. During aggregation as well only the packets directed to the same station are aggregated [32].

In order to use MU-MIMO, IEEE 802.11ac has enhanced the concept of TXOP sharing in the downlink by introducing MU-TXOP sharing. One of the main reasons for the introduction of this mechanism is the limitation of EDCA TXOP to allow frames belonging to different ACs to be simultaneously transmitted.

Using MU-TXOP or TXOP sharing, once the AP gains access to the medium, it can share the obtained TXOP and transmit frames to multiple users with the help of Group ID[3]. Each EDCF uses its own EDCA parameters to compete for a TXOP and after it gains access, it becomes the owner of this TXOP (primary AC) and hence can share its TXOP with multiple other ACs (secondary AC). This winning AC or the primary AC, decides about the proportion of time and bandwidth to be assigned to different secondary ACs based on its own TXOP limit.

We now present the channel access rules to support the PHY/MAC enhancement in IEEE 802.11ac.

2.3.4.2 Channel access mechanism in IEEE 802.11ac

IEEE 802.11ac is more efficient to make use of the available spectrum. This is because in IEEE 802.11ac, the channel bandwidth can change dynamically on a frame by frame basis. When a wide channel is available, high data rates can be used, however, when only a narrow channel is available, IEEE 802.11ac can fall back to lower rates.

The following are the basic channel access rules that are followed in IEEE 802.11ac. A frame can be transmitted only if the medium is idle as in the earlier standard, however, in IEEE 802.11ac to determine whether the medium is idle, depends on the channel width that is used for transmission [17].

- To transmit a 20 MHz frame on 20Mhz channel, CCA is done only at the primary channel
- To transmit a 40 MHz frame on primary 40Mhz channel, CCA is done on the primary and CCA check should also pass for the secondary 20 MHz channel
- To transmit a 80 MHz frame on its primary 80 MHz channel, both the primary 40 MHz and the secondary 40 MHz channel must be idle
- To transmit a 160 MHz frame on 160 MHz channel, both primary and secondary 80 MHz channel must be idle.

In general this is done by extending the rules described for the IEEE 802.11n channel operation. An IEEE 802.11ac station can transmit an 80 MHz signal when its primary channel is idle for DIFS along with the backoff timer and the rest of the three secondary channels are idle for PIFS period immediately preceding the expiration of the backoff counter. In case any one of the secondary channel are busy, two kinds of operation is possible

• *Static 80 MHz operation*: For this operation, the station should retry to access the channel with a new random number from its contention window size at that stage. In Figure 2.18(a), we can see that until and unless all the secondary channels are idle the station continues with its channel access attempts [17]. This may not be a feasible solution if

there are many stations operating in the secondary channels as this may rarely allow IEEE 802.11ac station to get a chance to acquire a free channel.

• *Dynamic 20/40/80 MHz operation*: For this operation, the station can transmit data over 20 or 40 MHz depending upon the availability of the secondary channels as obtained through CCA. We can see in Figure 2.18 (b) that when the secondary channels are busy a narrow bandwidth is used for the transmission [17]. The receiver should know the channel over which the transmission will take place, so 20 MHz primary channel is always included in the 20 MHz or 40 MHz bandwidth operation.





Enhanced RTS/CTS

To further improve bandwidth negotiation process and to reduce the hidden node problem, IEEE 802.11ac introduces enhanced RTS/CTS that add bandwidth signaling. In the normal RTS/CTS, the RTS/CTS frames are used to clear the channel on which they are transmitted. In the enhanced RTS/CTS schemes, the RTS/CTS frames are replicated in all adjacent 20 MHz channels to clear out multiple channels and attain a wider bandwidth. To avoid hidden nodes problem, these

RCS/CTS frame exchanges take place at IEEE 802.11a rate so that the legacy devices can also decode them. In addition, the channel bandwidth information present in the RTS/CTS frames can be decoded by the IEEE 802.11ac devices to know the number of available secondary channel. This allows the initiator to know the amount of clear channel bandwidth around the responder and thus make a very high speed transmission if multiple channels are found available. The duplicate frames are used for bandwidth signaling. Even in the case of a network occupying a 80 MHz spectrum, the beacon frames are always send on the primary and all the access control is done on the primary channel. Thus a station interacts with IEEE 802.11a station on its primary 20 MHz channel and with IEEE 802.11n station on its primary 40 MHz channel [6].

Figure 2.19 shows the operation of the new RTS/CTS mechanism [6]. The sender that has a frame to be transmitted over the full 80 MHz will duplicate the RTS frame across all the four channels of 20 MHz to acquire the whole channel. The receiver does the clear-channel assessment in order to check the availability of the entire 80 MHz and then sends CTS to indicate it. This results the NAV to be set in all the four channels so that other networks will have the knowledge of this transmission and defer their channel access.



Figure 2.19 : Successful channel acquisition

In Figure 2.20, we can see what happens if there is an interference at any of the channels that the sender is willing to make a transmission. In case of an interference at the receiver (say in two

channels), no CTS is sent by the receiver. Hence, the receiver sends CTS frames only on free channels thus indicating the sender to acquire only the 40 MHz of spectrum for its transmission. Thus at the end of this RTS/CTS exchange, the sender can transmit its frame using a 40 MHz transmission.



Figure 2.20 : Interference at channels

In addition to the above enhancements, new backoff and ACK mechanisms are also needed to support the simultaneous transmission of multiple AC in one TXOP. For this purpose, different rules must be incorporated to solve some of the issues as listed below.

Mechanisms to send the ACK or Block Acknowledgement to the uplink: As MU-MIMO is only supported at the downstream and the AP may not be able to receive different ACK using different spatial streams in the uplink (UL). In this case, STA needs to use appropriate timings to transmit their ACK so that they do not collide with each other transmission. Two basic solutions are mentioned in the proposals for this purpose: scheduled based and polled based [33].

In the schedules based approach, the AP should calculate the time and inform the station to send their ACK at their allocated times. This involves scheduling complexity at the end of AP. In the polled based approach, the AP can poll the station to get the ACK by sending Block Ack Request (BAR) frames, one at a time. This approach is simpler and reliable as compared to the scheduled based approach and hence, the IEEE 802.11ac TG has selected it as the default acknowledgement scheme for downlink (DL) MU-MIMO transmission [33].

New Backoff procedures: Due to the simultaneous transmission of multiple types of AC frames to different destination, new mechanism are needed to solve the issues that may arise due to the loss of any one of the UL BA. Some simple rules are proposed for invoking the backoff procedures by the IEEE 802.11ac TG in the event of success and failed transmission [33]. For instance, a transmission is considered successful only if all the MPDU transmitted by the holder of the TXOP was successful and the TXNAV timer has expired. On the other hand, a transmission is considered a failure if the expected response for the first frame of the TXOP of that AC is not received.

However, the above two rules may not be sufficient to incorporate all cases of success and failure that can be seen when the MU-MIMO is used. For instance, question about how the transmission of one AC should be treated when the cause for packet loss was another AC, sent simultaneously during one single TXOP, still needs investigations? Thus, the treatment of different AC in terms of backoff procedure may need further specification.

2.3.5 Summary

The following table summarizes the different MAC features and access techniques and the different standards that incorporate them.

Standard	Channel access	Enhancements	RTS/CTS	Optional feature
	scheme		support	
802.11a/b/g	CSMA/CA using	N/A	Yes	PCF, CTS to self
	DCF			
802.11e	CSMA/CA using	TXOP, Access	Yes	HCCA
	EDCF	categories, Block		
		Ack		
802.11n	CSMA/CA	Aggregation,	Yes	Reverse Direction,
	with static or	Block		Support for RIFS,
	dynamic bandwidth	Acknowledgement		CTS to self
	operation			
802.11ac	CSMA/CA	Enhanced	Enhanced	MU-TXOP

Table 2.8 : MAC features in IEEE 802.11 standards

Standard	Channel access	Enhancements	RTS/CTS	Optional feature
	scheme		support	
	with static or	Aggregation,	RTS/CTS	sharing
	dynamic bandwidth	Block	with	
	operation	Acknowledgement	bandwidth	
			signaling	
802.11ad	CSMA/CA with	VA-MSDU, IBSS	Yes	NA
	TDMA			
802.11aa	CSMA/CA using	SCS, GCR,	Yes	NA
	EDCA and two more	multicast support		
	queues			

2.4 Upcoming IEEE 802.11 standards

2.4.1.1 IEEE 802.11 ad

IEEE 802.11ad is foreseen as the technology to support wireless communication in the 60GHz spectrum. It uses similar technology as that of IEEE 802.11ac with a difference that it allows for wide band channel operation (60 GHz) and can offer a data rate of 7 Gbps. The main feature of this technology is to allow the interoperation and coexistence with the legacy IEEE 802.11 and also among heterogeneous system like 802.15c WPAN [3]. IEEE 802.11ad enabled devices can transparently switch between 60 GHz band and 2.4/5 GHz band.

A new architecture is proposed for this standard PBSS (Personal Basic Service Set) which is similar to IBSS (Independent Basic Service Set), however its operation is not based on a particular device (AP) and any station within the PBSS can act as the PBSS central point (PCP).

At the physical layer, it combines OFDM modulation with single carrier (SC) coding schemes. OFDM allows for high data transmission up to 7Gbps and SC can support data rate over 4.6 Gbps and uses low power transceivers.

At the MAC level, IEEE 802.11ad continues to use the basic MAC access mechanisms. However, it also proposes an enhanced MAC to include new features to attain a throughout of 1 Gbps. This is to done to support its coexistence with other 60 GHz technology and to support QoS improvements. CSMA/CA is not sufficient for 60 GHz operation where the stations changes their antenna patters dynamically. Moreover, it cannot support application with rigorous QoS because it involves random backoff. Hence to support directional transmission, IEEE 802.11ad enhances CSMA/CA that allows station to know the time period during which they can be awake and point to their respective antenna for their transmission. Furthermore, to incorporate better QoS support, it also introduces both TDMA and CSMA in its beacon period. At the MAC layer, new aggregation scheme is also introduced called Video Aggregation MSDU (VA-MSDU) to support video traffic [3].

2.4.1.2 IEEE 802.11 aa

This specification is designed to allow better video transmission in the wireless networks. It brings enhancements to the EDCA prioritized mechanism that does not consider unicast and multicast transmission. It offers two enhancements at the MAC: Stream Classification Service (SCS) and Group Cast with Retries (GCR) [3]. With the SCS, IEEE 802.11aa extends the EDCA by adding intra-access category prioritization. This means that two more queues are added to the EDCA access categories one for audio (AAC-VO) and the other for video (AAC-VI). Moreover, there is a provision for tagging the packets with the drop eligibility indicator (DEI). When this bit is set it indicates the maximum number of retries (long or short) allocated to that packet. The standard also specifies new acknowledgement policy needed for multicast audio/video streaming.

2.4.1.3 IEEE 802.11 ax

It is a follow on technology for IEEE 802.11ac. It will be based on 2.4 GHz and 5 GHz unlicensed band, however, additional bands between 1 GHz and 6 GHz may also be available. With the improvement in the IEEE 802.11ac technology, more high speed connection is possible, however, the single individual connection may still be peaking around just 300 Mbps [34]. Thus, IEEE 802.11ax aims to increase the actual connection speed of individual devices rather than increasing the overall network capacity. It aims to quadruple the speed of devices thereby reaching an individual device connection to gigabit range.

802.11ax will be adopting a new radio technology called the MIMO-OFDA (orthogonal frequency division access). This will include four MIMO spatial streams to send multiple streams of data to same or different users. In addition, each of these streams will be multiplexed with OFDA. OFDA is a variant of a well know OFDM technologies that is used in 4G and other older Wi-Fi standards. According to Huawei, OFDA may increase the spectral efficiency by 10 times, which means 10 times the max theoretical bandwidth [35].

State of the Art

Part II: QoS Provisioning in MAC Over IEEE 802.11 standards

2.5 QoS provisioning

QoS may be defined as a basic measure to determine how well the system operates to meet the user's requirements. In general, QoS includes a set of system parameters that can control the functionality of the system and can be tuned in order to meet the user's satisfaction. Moreover when it comes to multimedia system, QoS management may involve setting of appropriate parameters, reserve necessary resource, in order to optimize the overall system performance and to achieve required system functionality [36].

QoS and traffic management in Internet Protocol (IP) and Asynchronous Transfer Mode (ATM) networks are widely studied in the literature. Similarly, there are standardization activities going on in the 3rd Generation Partnership Project (3GPP) to take care of the QoS architecture in the cellular networks [37]. In the IP networks, Internet Engineering Task Force (IETF) proposed two main approaches for end to end QoS. These were the Integrated Service (IntServ) and the Differentiated service (DiffServ). IntServ uses specific algorithms and scheduling mechanism and is a per-flow approach to provide strict QoS guarantees to individual streams, while DiffServ is a class based mechanism that includes traffic classification to provide aggregate guarantee for a group of applications [38].

Similar to these two mechanisms of QoS guarantee, wireless networks also defines two types of approaches at the MAC layer: one is the parameterized and the other is prioritized. In parameterized approach, the MAC data services require to meet the QoS requirement values like data rate, jitter and delay. This may demand the need to assign deterministic capacities to be assigned to each flow. In the prioritized approach, some violation to meet the QoS requirement may be acceptable for a certain limited period of time. This can be made possible by using the internal buffers to counteract those short time QoS violations [26].

2.5.1 QoS metrics

The level of QoS provisioning is based on certain parameters often know as QoS metrics that indicates the overall success or quality of the system. In case of multimedia services, they signify their overall quality. The measurement of these metrics or parameters can be conducted at different layer of the OSI model and may be defined separately for each layer. For instance, the application layer QoS metrics may determine the performance associated to the application layer

and the presentation of a multimedia content. The network level QoS metrics can indicate the quality of the end to end path by using IP level information. Similarly, MAC level QoS metric can indicate the quality of the link. However, all these metrics at different layers are strongly interrelated and a variation of a metric in one layer can impact the higher layer metrics. Some of the most commonly used quality of service metrics used at the MAC layer are as follows:

Throughput

It gives the volume of information that a station can successfully transmit over the wireless channel per unit time. It can be measured by the total transmitted bits per unit time from the source to its destination. Throughput is measured in Kbps/Mbps.

Throughput= $\frac{No.of Received Packet x 8}{Total Time Taken}$

Frame Delay

It is defined as the time that a frame needs to be successfully delivered at the destination. This includes the delay time waiting in the queue, delay for contention, transmission delay and delay due to retransmission. It is measured in seconds/milliseconds.

Jitter

The maximum variation of the frame delay is given by jitter. It shows how the latency changes from frame to frame. Low value of jitter indicates a good and uninterruptable connection whereas high value of jitter is a sign of congestion in the network. It is measured in seconds/milliseconds.

Packet delivery ratio (PDR)

It is defined as the ratio of the number of delivered data packet to the destination.

PDR= $\frac{Total \ number \ of \ packets \ received}{Total \ number \ of \ packet \ send}$
Frame loss ratio

It may be defined as the amount of frames that are lost at the MAC layer. This may be due to collisions or when the maximum limit of retransmission is reached. Frame drops may also occur due to MAC queue overflow or when the frame reaches their timeout values and is mainly caused in the event of congestion. It is measured in percentage.

2.5.2 Main QoS provisioning methods based on MAC protocols

Out of the many standards related to the IEEE 802.11, 802.11e was introduced to provide the QoS over the WLAN interfaces. The basic concept used in the standard is to offer differentiated channel access mechanism to the different traffic types as explained in section 2.3.2. High priority frames or traffic gets better privilege treatment than the ones belonging to low priority. However, using EDCA and only differentiating the traffic flow may not be enough to provide sufficient quality of service in all scenarios [39]. This demands for different QoS-aware MAC solutions that can provide better QoS in different conditions (load, radio and traffic types). This becomes more challenging when it comes to distributed network and yet a contemporary research problem.

In the literature, there exist several QoS-aware MAC protocols which may employ one or more of the following mechanisms to provide QoS at the MAC level. A detailed explanation of these mechanisms and their associated related work is given below:

2.5.2.1 Backoff differentiation

Using backoff differentiation, different backoff values may be assigned to different traffic types or to different stations. In addition, changes can be applied to the CWmin and CWmax values, their range, and incremental trend to provide priority to certain traffic.

In the literature number of MAC protocols have been proposed that incorporates backoff differentiation in one form or the other. The standardized solution based on backoff differentiation is the EDCA function proposed for IEEE 802.11e protocol that assigns different backoff values range to different traffic types. However, several other QoS solution based on this idea can be found in the literature to improve the MAC layer performance (i.e., improve fairness, reduce collisions, provide traffic differentiation, etc).

Authors in [40] proposed a scheme in which the CW size is changed based on the utility that a station achieves over its randomly selected backoff values in the previous transmission. Authors in [41] proposed a solution where AP adjusts its CW and selects the backoff depending upon the ratio between the total data rate of downlink flows to the packet rate of an uplink flow. Although it does not need any modification at the wireless terminals, the system considers many parameters to be known to the AP that demands for additional AP functionalities. Authors in [29] studied the backoff-based priority schemes both for IEEE 802.11 and IEEE 802.11e standard and evaluated their performances by differentiating the minimum backoff window size, the backoff window-increasing factor and the retransmission limit. Similar proposal AQMP, based on adapting the persistence factor (PF) of the CW is proposed in [42], where the PF is a function of the loss rate. When the frame loss rate is higher than the previously calculated value, the PF of high priority traffic is decreased and the PF of the low priority traffic is increased. However, the solution is reversed, when the frame loss rate is less as compared to the previously calculated average values.

Tuning of window parameters is also proposed in [43], where a fuzzy logic approach is used to tune the parameters based on network observations, application types, and access categories. Similar work to tune the shape of contention slot was proposed in [44], where the authors proposed solution to adaptively minimize collision by maximizing the selection of relatively less collision prone slots over the contention window. In addition mechanism proposed in [45] defines different priority classes and uses different backoff range for each kind such that the backoff range of different categories do not overlap, however, the number of distinct backoff values for each priority class is allowed to be the same. Authors in [46] make differentiation between flow even if they are assigned the same priority and uses same queue. This differentiation is made possible by changing the CW according to the ratio between the number of successful frame transmission of the current flow and the successful frame transmission of the flow of the same priority that was best served.

2.5.2.2 Reservation

Using the reservation techniques different time slots or streams can be allocated to stations for transmission. This can help in reducing collision, however, may demand some kind of reservation table. There can also be stream reservation techniques that can use contention based

technique for non-real time flows and dedicated reservation period for real time flows. The reservation can be done by making the control messages available to all the neighboring nodes.

Reservation used in EDCA is proposed in [47] as EDCA/RR where the high priority traffic is allowed to reserve a TXOP. This is done by introducing a traffic specification (TSPEC) that provides information about the characteristic of the traffic and its QoS expectation thus providing applications with hard QoS requirements and the possibility to reserve transmission time for guaranteed medium access. Authors in [48] have proposed SR-MAC where they propose to use the reserved bits of control frame to perform stream reservation. Each node maintains a reservation table to make efficient use of the bandwidth. Authors state that the proposed scheme can provide guaranteed bounded delays for voice traffic. Furthermore, DARE is proposed in [49] that allows distributed end-to-end allocation of time slots for real-time traffic. The solution reserves repeating time slots for QoS-demanding applications while using the DCF for best effort applications. However, both these schemes rely on reservations that have to be allocated for each channel access.

In [50], authors have provided a fully distributed reservation scheme where a user receives a tacit reservation. A tacit reservation is defined as a relation that can identify an immediate predecessor. Rules are then used to enable stations to know when to transmit exclusively in a fully distributed manner. As this is done by enabling contention free transmission, the collision in the system is minimized and also there is no use of any explicit control messages.

For general contention based resource reservation scheme, basically the control messages such as RTS/CTS are used to distribute the resource information to the neighbors. For instance, the channel reservation scheme proposed in [51], utilizes the new control messages to reserve the channel for the next packet waiting in the transmission queue during the current transmission. A new contention-based MAC protocol, the Channel Reservation MAC (CR-MAC) protocol is proposed that uses these control messages to exchange channel reservation information with little extra overhead.

Some schemes are also based on reserving backoff slots, for instance the work proposed in [52] allows a node to advertise its next backoff values, however this may cause a waste of bandwidth. Similar work is proposed in [53], where a fixed backoff cycle is used where a certain number of

slots can be reserved for the nodes in the contending region. Enhanced RTS/CTS for bandwidth signaling can also be found in the new standards like IEEE 802.11ac [6].

2.5.2.3 Inter frame space differentiation

The interframe space (IFS) is the time that the station needs to determine if the channel is free to either start a transmission or to start counting down of the backoff timer. IFS differentiation can allow different station to earn the right to transmit based on different network and traffic type. It generally works in association with backoff differentiation; however, there are some works that are entirely based on IFS differentiation.

One of the common deployments of IFS differentiation is in the EDCA, where different priority traffic is assigned different AIFS. However, there are some works that enhances this differentiation mechanism to further improve the performance. In [54], authors have proposed a method to dynamically change the AIFS parameter of low priority based on EDCA. Authors in [55] have employed a different AIFS differentiation based on the frame loss rate. In this scheme, if the frame loss rate exceeds some predefined threshold, the AIFS value for high priority is decreased and the AIFSN value for low priority traffic in increased. Some solutions instead of using the combination of IFS and backoff differentiation make use of IFS times exclusively, and they are selected randomly. For instance, authors in [56] have made used of random selection of AIFN for traffic differentiation. In [57], IFS is calculated independently for each head-of-line frame. It is proportional to the bandwidth available to this traffic class at that time and their throughput requirement.

In [58], authors have included the concept of assigning unique AIFSN values in each used channel to avoid frequent collisions. In addition, to make more efficient use of the available AIFSN values, they are assigned dynamically so that the used AIFSN values are as small as possible and right prioritization can be made between streams. However, for this purpose they propose to negotiate AIFSN values among the stations using special management frames that required dedicated in-band signaling.

2.5.2.4 Frame aggregation

Using Frame aggregation burst of data can be transmitted at once to reduce the number of contention in the channel. This can increase the overall throughput of the system. However, it

needs to be used carefully inorder to maintain fairness in the system. Most of the work related to frame aggregation started with IEEE 802.11e concept of TXOP and is now widely found in IEEE 802.11n/ac.

Authors in [59], have given a detail account for various frame aggregation mechanisms that can help in reducing the overhead problem at the MAC layer. They investigated frame aggregation based on the latest IEEE 802.11n draft standard and proposed solution to achieve high throughput and high efficiency. Authors in [60], classified the frame aggregation mechanism according to the different network aspect like distributed vs. centralized, uplink vs. downlink, Phy level vs. MAC level etc. In [61], authors analyzed the efficiency of frame aggregation. They demonstrated that applying maximum aggregation and adapting the modulation and coding schemes instead of the frame sizes, the performance of MAC layer can be enhanced. In [62], authors proposed changes to physical parameters that control the MPDU size thereby having a kind of cross layer adaptability. The proposed solution could achieve higher throughput at low SNR without compromising system performance at high SNR. In [63], a frame aggregation scheduler that allows switching between two aggregation methods according to the Bit Error Rate (BER) level is proposed. Authors demonstrated that selecting the frame aggregation size can be useful in lightly loaded conditions as compared to the fixed size A-MPDU method. Authors in [64] have demonstrated that basic aggregation scheme is not sufficient to meet the QoS for high priority traffic in unsaturated conditions. In [65], authors have proposed AFR where multiple frames are aggregated and the possibility of retransmission of fragment is incorporated. They used zero waiting time for frame transmission after a station wins the channel so as to avoid aggregation delay.

2.5.2.5 Jamming

Jamming incorporates the transmission of some energy pulses over the channel. This technique has been used in the literature to provide differentiation between stations and traffic classes and also to provide some means to combat the hidden station problem. Some of the work related to jamming or methods that combine jamming with some other techniques like backoff differentiation may be found in the literature.

Authors in [66] proposed a scheme based on contention based MAC that makes use of pulses of random length to make the stations aware of others transmission. Pulses are sent on the control channel when the receiver knows that the transmission is for itself. The data transmission is made on the data channel simultaneously. Though it effectively detects collisions and improves the throughput of the system, however, it uses out of band channel for control messages which may need further modification to the original standard devices. Authors in [67] proposed a MAC protocol based on busy tone that uses two narrow band busy tone channel and one information channel. At first the station transmits a busy tone for duration equal to the backoff period in one of the transmitting. It supports both voice and data traffic and the voice frames are send immediately after the transmission of the busy tone.

Though many solutions are proposed that utilizes the jamming techniques, however the protocol that introduced jamming was Black Burst [68] and [69] in which the high priority traffic jams the wireless channel with pulses of energy. The duration of the BB was determined by the amount of time that the station had to stay waiting for the channel to get idle.

The work presented in [70] proposes a new MAC mechanism based on the tournament contention function(TCF) in which each station needs to activate a functionality to generate a key. The optimal choice of selecting the key can allow limiting the collision probability to a minimum value. The solution requires the exchange of a number of signals for the tournament purpose and uses a secondary channel for this purpose. To avoid the interruption with the legacy devices, it also proposes to activate a control signal or jamming all along the duration of the tournament. However, this proposal requires some hardware modifications and may involve synchronization issues between different players of the tournament.

In the following Table 2.9, we present a synthesis of the above mentioned proposals.

Item	Year	Туре	Traffic type	Comments	
[40]	2009	B.D	NA	Based on utility of satisfaction in previous	
				transmission	
[41]	2008	B.D	Uplink	Provides uplink and downlink fairness but needs	
			downlink	additional functionalities at the AP	
[29]	2005	B.D	2	Evaluated the performances by changing the	

 Table 2.9 : Overview of MAC enhancement proposals

Item	Year	Туре	Traffic type	Comments	
				minimum backoff window size, the backoff	
				window-increasing factor and the retransmission	
				limit	
[42]	2009	BD	>1	Station changes their PR depending upon the queue	
				loss	
[43]	2012	BD	4	Tune the CW parameters based on fuzzy logic	
				approach	
[44],	2012	BD	2	Minimizes collision by maximizing the selection of	
				relatively less collision prone slots	
[45]	2008	BD and	4	Each traffic priority has non-overlapping backoff	
		FA		range with other priority traffic	
[46]	2011	BD	3	Makes a differentiation between flow even they are	
				assigned the same priority class and changes their	
				CW accordingly	
[47]	2006	BD and	4	Combines EDCA with stream reservation	
		FR			
[49]	2006	R	2	Reserves repeating time slots for QoS-demanding	
				applications while using the DCF for best effort	
				applications.	
[50]	2012	R	NA	Uses Tacit reservation where a tacit is obtained after	
				satisfying certain network conditions	
[48]	2013	R	2	Uses the reserved bits of control frame to perform	
				stream reservation for voice traffic.	
[49]	2009	R	>2	Allows for the distributed end-to-end allocation of	
				time slots for real-time traffic.	
[51]	2008	R	NA	Utilizes new control messages to reserve the	
				channel for the next packet waiting in the	
				transmission queue during the current transmission	
[52]	2008	R	NA	Allows a node to advertise its next backoff values	
[53]	2010	R	2	Uses fixed backoff cycle where a certain number of	
				slots can be reserved for the nodes in the contending	
				region	
[54]	2006	IFS	3	Dynamically change the AIFS parameter of low	
				priority based on EDCA to prevent unfair	
				bandwidth sharing	
[55]	2012	IFS	2	AIFS is varied depending upon the frame loss rate.	
				If exceeds some threshold, the AIFS value for high	
				priority is decreased and the AIFNS value for low	
				priority traffic in increased	
[56]	2007	IFS	2	Random selection of AIFN for traffic differentiation	
[57]	2002	IFS	2	IFS is calculated independently for each head-of-	
				line frame based on its priority, waiting time, and a	
				random number	
[58]	2010	IFS	4	Unique AIFSN values in each used channel to avoid	
				trequent collisions.	
[59]	2008	FA	NA	Different aggregation solution to achieve high	
				throughput and high efficiency	
[60]	2005	FA	NA	Classified the frame aggregation mechanism	
				according to the different network aspect	

Item	Year	Туре	Traffic type	Comments	
[61]	2010	FA	NA	Suggested adapting the modulation and coding	
				schemes instead of the frame sizes during	
[()]	2007	ГА	NT A	aggregation	
[62]	2007	FA	INA	Proposed changing the physical parameters to control the MPDU size	
[63]	2010	FA	NA	Switching between two aggregation methods	
				according to the Bit Error Rate (BER)	
[65]	2009	FA	4	Aggregating multiple frames and possibility of	
				retransmission of fragments	
[66]	2006	J	NA	Makes use of pulses of random length to make the	
				stations aware of others transmission	
[67]	2008	J	2	New protocol based on busy tone that uses two	
				narrow band busy tone and channel and one	
				information channel	
[68]	1999	J	2	Real-time nodes contend for channel access with	
				pulses of energy—so called BB's and its duration is	
[69]	1996	J	2	a function of the delay that the nodes had to wait for	
				the channel to become idle	
[70]	2012	J	N/A	Based on a new Tournament contention function for	
				channel access to reduce the number of collisions in	
				the system	

B.D=Backoff differentiation, FA-frame aggregation, J- Jamming, IFS-interframe differentiation, R- Reservation, N/A- Not applicable

2.6 Conclusions

This chapter details the different MAC and PHY evolution in IEEE 802.11, listing the enhancement that has been adopted in this technology. It also provides different QoS provisioning techniques based on the MAC layer and the related work based on this approach. Globally, this chapter shows that the IEEE 802.11 has evolved drastically in few decades allowing high data rate and throughput both at the PHY and MAC level. However the standard still lacks better QoS provisioning methods to support multimedia services widely found in the current wireless networks. Additional efforts are required to improve the MAC access schemes that can provide guaranteed service and better QoS to the multimedia services that run over these networks.

Chapter 3 An adaptive aggregation scheme for QoS differentiation over IEEE 802.11n

3.1 Introduction

The recent and the upcoming IEEE 802.11n and IEEE 802.11ac WLAN standards, come with a set of new promising technologies such as MIMO and OFDM. These can provide higher physical data rates, from 54Mbps (in IEEE 802.11b/g) up to 600 Mbps (in IEEE 802.11n) and probably up to 7Gbps (in IEEE 802.11ac).

These evolutions are enabling the fast adoption of QoS demanding applications such as High Definition TV, VOD (Video on Demand) and video conferencing, over personal Wi-Fi access points and hotspot.

The IEEE 802.11 family, is therefore, still considered as the main radio access technologies at homes and offices. Moreover, 3GPP is now considering it as a candidate access technology to extend/offload cellular networks/traffic.

Initial experimentations show that despite the enhanced PHY efficiency, Wi-Fi networks still suffer high overhead and instability at high loaded conditions. For instance, even in ideal channel conditions, there is a reduction of MAC efficiency from 42% to 10% although the PHY rate increases from 54 Mbps to 432 Mbps [65]. Indeed, Wi-Fi MAC is still based on the CSMA/CA access technique that involves large overhead. For these reasons, recent IEEE standards introduced new MAC layer mechanisms like aggregation and Block Acknowledgements. These are made possible because of the new promising high PHY data rates. The main objective is to reduce the overhead associated with packet transmissions by increasing the payload size compared to headers.

Nevertheless, with traditional aggregation, performance limitations can still be observed. For instance, for real time voice services aggregation is either not applied or if applied may be allocated same aggregation parameters as other traffic types. When aggregation is not applied to these services, they may suffer due to CSMA protocol overhead. When aggregation is allowed

without traffic type consideration, there can be performance degradation. Indeed, mixing delay tolerant and delay sensitive flows (traffic mixing) with long aggregated frames can lead to QoS degradations and mainly those belonging to delay sensitive flows. Furthermore, when different types of traffic are bundled together in one frame, irrespective of their initial QoS requirements, QoS differentiation is further made difficult.

In this chapter, we propose QoS-HAN aggregation, a QoS aware aggregation scheme to achieve better performances and QoS differentiation. QoS-HAN allows combining MPDU frames based on traffic types and other contextual information, including radio conditions and uses controlled aggregation sizes to meet their QoS requirements.

3.2 Frame Aggregation in IEEE 802.11n

In the IEEE 802.11n standard, frame aggregation and block Acknowledgement are features that are introduced as MAC enhancements. The objective is to reduce MAC/PHY overhead. As a consequence, the effects of the inter-frame waiting times like SIFS and DIFS are reduced leading to a considerable increase in throughput.

The standards define three aggregation methods for IEEE 802.11n, namely *A-MSDU*, *A-MPDU*, and *Two-Level aggregation*.

3.2.1 A-MSDU aggregation

In this method, multiple MSDUs are aggregated and transmitted at once in a single MPDU. These MSDU should share the same destination and should have the same level of QoS (i.e. same TID field). This means that the sender and destination addresses should match in every subframe of the A-MSDU. Once the aggregated frame is constructed, a unique MAC header is used to reduce the total MAC overhead without the necessity to transmit the IEEE 802.11n MAC/PHY header for each subframe. This feature is more efficient for small MSDU like voice and TCP acknowledgement.

To form an A-MSDU, received MSDUs are first buffered and then aggregated when at least one of the two following conditions is met:

- a maximum defined aggregation threshold is reached or
- the oldest MSDU reaches its maximal delay limit.

Figure 3.1 shows the structure of an A-MSDU frame where each subframe consists of a subframe header, a payload from the LLC and up to 3 bytes of padding.

The padding bytes are inserted in order to ensure that the sizes of all sub-frames are multiple of four bytes, except the last frame. This allows the receiver to predict the beginning of the next frame.

According to the standards, the aggregation threshold that determines the maximal length of the A-MSDUs can vary between 3839 and 7935 bytes.

The major disadvantage of A-MSDU frame aggregation method can be seen in error prone channels. Indeed, as the frame contains a single frame check sequence (FCS) for all the aggregated sub-frames, and if transmission errors occur, the entire frame needs to be retransmitted. In these conditions, this method can then lead to an increase in retransmissions overhead and, therefore, degrade the overall performances.



Figure 3.1 : A-MSDU aggregation

3.2.2 A-MPDU aggregation

In this method, aggregation is done at the interface with the PHY layer. Multiple MPDU frames are aggregated into a larger A-MPDU frame with a single physical header. Consequently, there is no restriction that all MPDU should have the same TID, but these must be addressed to the same

destination. Moreover, there is no restriction on the holding time of the frame to form an A-MPDU and this only depends on the number of packets in the transmission queue [71].

In addition, all A-MPDU sub-frames keep their own IEEE 802.11n MAC header and FCS as shown in Figure 3.2. A delimiter, inserted before each A-MPDU subframe, contains the MPDU length and a Cyclic Redundancy Check (CRC). The MPDU delimiter identifies the valid reserved MPDU length field and the CRC protects it. This allows the receiver to determine whether the received sub frame is corrupted by channel errors, thereby, improving the robustness of transmission under error prone channels.



Figure 3.2 : A-MPDU aggregation

When an A-MPDU is received, the station first scans over the four bytes delimiter, then locates a valid delimiter based on the CRC bits and the delimiter signature. If a delimiter is received correctly, the whole subframe is checked for errors using FCS bits. A-MPDU has a FCS to check the correctness of each frame while A-MSDU only has a FCS for the whole aggregated frames. Hence A-MPDU is more resilient to channel errors though it involves a higher overhead. According to the standards, the maximum size of the A-MPDU frame is 65,535 bytes and the number of aggregated sub-frames is limited to 64. In this mechanism the size of MSDU should not exceed 4 KB. For the A-MPDU scheme, Block Ack is used to acknowledge the received frames (see section 3.2.4).

3.2.3 Two Level Aggregation

In this method, aggregation is done by combining the two above features namely A-MPDU and A-MSDU as shown in Figure 3.3. The objective is to reduce the IEEE 802.11n overhead. The MAC header along with the A-MSDU frame and the FCS field are considered as one MPDU frame. This MPDU along with MPDU delimiter and the padding bytes are then inserted into a single MPDU sub-frame. Finally, all the MPDU sub-frames are aggregated to form one A-MPDU frame. A number of A-MSDU for different destinations may be formed at the upper layer. Any A-MSDU or MSDU's having the same destination address can be aggregated in one MPDU frame and can be used as a normal MPDU sub-frame during A-MPDU aggregation [59]. When used over high speed networks, the combination of the two types of aggregation can significantly reduce both the MAC and the PHY overhead.



Figure 3.3 : Two level aggregation

3.2.4 Block Acknowledgement

This mechanism was first introduced in IEEE 802.11e and then modified in IEEE 802.11n in order to support the A-MPDU aggregation feature. Block ACK frames contain a bitmap field which can be set to 0 or 1 to selectively differentiate the corrupted and uncorrupted MPDUs respectively. The bitmap field is 128 bytes in length as seen in Figure 3.4 and each frame is mapped using two bytes (64*16). This is the reason why only 64 frames can be supported in A-MPDU aggregation.

A corrupted subframe is indicated with a bitmap value set to 0 as shown in Figure 3.5. When an A-MPDU with errors is detected, the receiver transmits a Block ACK to selectively acknowledge the uncorrupted MPDUs [72]. This reduces the probability of errors and collision as compared to individual ACK transmissions. The sender is then able to retransmit only the non-acknowledged MPDUs. This mechanism can only be used with the A-MPDU aggregation mode and is very efficient in error prone environment.



Figure 3.4 : Block Ack Frame [19]



Figure 3.5 : Block Ack with aggregation

3.2.5 Analytical model for aggregation

Network throughput depends upon a number of elements like number of aggregated frames in an A-MPDU, its sub-frame length, the amount of users and the current channel BER. In [73], authors extended the model presented in [24] to obtain an analytical model for A-MPDU aggregation. The objective of these studies is to identify the optimal fragment size that maximizes the network throughput in error prone channels. This model can be used to analyze the behavior of aggregation in different conditions.

Considering that the system time is divided into virtual time slots, the network's saturation throughput is given by

$$S = E[p]/E[t] \tag{1}$$

Where,

E[p] gives the number of payload bits that are transmitted successfully in a virtual slot E[t] gives the expected duration of the virtual time slot and is calculated as

$$E[t] = T_{idle}P_{idle} + T_eP_{tr} (1-P_s) + T_eP_{err} + T_{succ}P_{succ}$$
(2)

Where, T_{idle} , T_e and T_{succ} are the idle, corrupted and successful virtual time slot duration respectively. P_{idle} is the probability of the slot to be idle, P_{tr} is the probability of having at least one transmission in the slot, P_s is the probability of successful transmission on the channel without collision, P_{err} is the probability of transmission failure due to transmission error and not due to collisions, P_{succ} is the probability of successful transmission without collision and transmission errors.

Mathematically,

$$P_{err} = P_{tr} P_s p_e \tag{3}$$

$$P_{succ} = P_{tr} P_s (1 - p_e) \tag{4}$$

$$T_{succ} = T_{DATA} + T_{BACK} + SIFS + DIFS$$
$$T_e = T_{DATA} + EIFS$$

Where, T_{DATA} is the time required for the transmission of data and T_{BACK} represents the time required to transmit a BACK and p_e is given by

$$p_e = \prod_i (1 - (1 - P_b)^{Li + Lsubdr}) \tag{5}$$

where, L_i is the payload length of the data that is present in the ith sub-frame, L_{subhdr} is the length of the overhead in a MPDU, and P_b is the bit error rate (BER) of the channel. For the case of A- MPDU transmission,

$$E[p]_{MPDU} = P_{succ} \sum_{i} L_{i} (1 - P_{b})^{Li + Lsubhdr}$$
(6)

The sub-frame length and the number of aggregated frames need to be adjusted and adapted depending upon the channel conditions [73].

3.3 Related Work

In this section, we will look over some of the related works on MAC frame aggregation. In [59], authors provide a detail account of various frame aggregation mechanisms that can help in reducing the overhead problem at the MAC layer. They investigated frame aggregation based on the latest IEEE 802.11n draft standard and proposed a solution to achieve high throughput and efficiency using those schemes. The work presented in [61] analyzed the efficiency of frame aggregation. They demonstrated that applying maximum aggregation and adapting the modulation and coding schemes instead of the frame sizes, the performance of MAC layer can be enhanced. Authors in [62] proposed changes to physical parameters that control the MPDU size through cross layer adaptability. According to the authors, their proposed solution achieves higher throughput at low SNR without compromising system performance at high SNR. In [24], a frame aggregation scheduler that can switch between two aggregation methods according to the Bit Error Rate (BER) level is proposed. Authors demonstrated that selecting the frame aggregation size can be useful in lightly loaded conditions as compared to the fixed size A-MPDU method. However, these algorithms do not take into account traffic differentiation and make use of a single aggregation size for all traffics types. Authors in [74] have indicated that the aggregation size depends on both traffic arrival rates and frame sizes. They studied the effect of aggregation size on the channel utilization and the frame latency. They concluded that in case of small aggregation size, the overhead due to preamble header dominates and lower the channel utilization, whereas in case of larger aggregation size, the time spent in the queue to form an aggregated frame may affect the higher layer quality of service. Moreover, they have modeled the transmission queue of 802.11n station and designed an analytical model to study both A-MSDU and A-MPDU aggregation schemes.

3.4 Proposed Solution: QoS-HAN aggregation

HANs stands for Home Automated Networks. In this design specific environment, Wi-Fi access points are more and more used to deliver Internet access and to interconnect multiple devices. Similarly, multimedia services are playing an essential role within this environment. Indeed, the wireless access points are also used to distribute digital video contents (from ADSL and Fiber accesses or from the personal video libraries). In addition, real-time flows may carry signaling for home automation applications over this Wi-Fi access technology. With these considerations, guaranteeing QoS and differentiation over Wi-Fi is becoming insistent.

As discussed in the previous sections and chapters, despite the recent PHY/MAC advances with the recent IEEE WLAN standards, new enhancements are still required, in particular at the MAC level. Moreover, there is no specific scheduling mechanism for aggregation defined in the standard. After the analysis of the state of the art, it appears that frame aggregation should consider more contextual information than simple source and destination addresses. We propose here a new adaptive A-MPDU aggregation scheme to enhance the QoS differentiation and QoS guarantee over the home networks. The proposed solution considers traffic classes and radio conditions in addition to packet destinations, to dynamically select suitable aggregation sizes. The main idea is to combine MPDU frames with the same QoS needs and to assign them controlled aggregation sizes. This will allow QoS differentiation at the aggregated frame level.

For instance, frames belonging to the delay sensitive traffic class like VoIP should be aggregated separately than those belonging to a data service class with large packet sizes and assigned aggregation limit accordingly. The aggregation limit or threshold is configured individually for different traffic types in different radio conditions. These sizes may then by adapted at regular intervals to meet QoS- requirements (delay threshold).

Figure 3.6 presents QoS-HAN aggregation scheme. Different MPDU frames belonging to different traffic types are present in the queue (say two traffic types in this figure). When a station gets an access to the channel, the packet type that was at the head of the line of the queue will represent the type of frame that will be aggregated to form the A-MPDU at that instant. For this, the queue will be looked for all the packets that belong to this traffic type and accordingly based on the allocated aggregation limit, packets will be aggregated and sent to the destination. The aggregation size threshold is, however, configured depending upon the traffic types and the radio conditions. It is carefully selected and adapted at regular interval for each traffic type depending upon their QoS requirement

The algorithm for QoS-HAN aggregation is given below

For all the packets in the queue { 1. Get the packet type of the first packet in the queue say P 2. Map it to a group say G 3. Get its destination address say D 4. Set the aggregation limit to L defined for this group G 5. Go to the next packet 6. If next packet == same group G and destination address = Da. Add the packet in the A-MPDU frame Else, goto 5 7. If the maximum number of packets in the A-MPDU < 64 and maximum *aggregation limit* < *L* a. goto 5 else, transmit the frame goto 1}





The effectiveness of aggregation varies in different radio conditions. For instance, in high BER environment, the probability of having errors increases with the increasing frame size. As

discussed in the related work section, this will result in the retransmission of large frames thereby lowering the efficiency of the entire system.

In order to study the behavior of the proposed scheme in different radio conditions, we used the Block Acknowledgement (BAck) mechanism along with the proposed aggregation scheme.

3.5 Performance Evaluation

3.5.1 Simulation Scenarios

The proposed aggregation scheme was implemented using the NS2 simulator. We used the 802.11n module available for NS2 [119] and modified the Queue class to support QoS-HAN aggregation. We considered a network topology that resembles a typical Home Automated Network (HAN) environment with multiple VOIP, video and FTP sessions running at the same time. The scenario is composed of fixed stations generating voice, video and data applications. Some of these stations send single flows to its destination station and some sent more than one flows. The three traffic classes are differentiated using CBR traffic with different packet sizes. We also considered that all nodes are operating in the decentralized mode and are within the same transmission range. We evaluated the system in two radio conditions, scenario with high BER (0.000205) and with low BER (0.000008). Table 3.1 gives the list of the parameters and their values used during the simulation.

Parameters	Default Values
Simulation Area size	500X500m ²
Simulation Time	15s
Node mobility	Static
Number of nodes	8
Bandwidth	96 Mbps [120]
Mac Layer	IEEE802.11n
Queue limit	70 packets
Aggregation size limit 1 for video	8000 bytes
Aggregation size limit 2 for voice	5000 bytes
Aggregation size limit 3 for data	8000 bytes
Time Slot	20µs
SIFS	10µs
Packet size for voice, video, data	160 bytes, 512 bytes and 1000 bytes

 Table 3.1: Simulation Parameters

TXOP limit	3.264 ms
Low BER	0.000008
High BER	0.000205
NS2 Antenna Type	Omni Directional
NS2 Network Interface	Type Wireless/Physical/MIMO
NS2 Interface Queue type	Aggregation queuing
Number of Antennas	4
NS2 MIMO type	SM-STBC
Propagation model	Two ray ground

3.5.2 Performance Analysis

Performance results are obtained by varying different parameters such as the aggregation size, traffic load, BER, type of traffic, and the presence or absence of the BAck. The performance metrics that are used to evaluate the proposed scheme are throughput, end to end delay, and PDR (see Section 2.5.1).

In the first set of experiments, we vary the traffic load and BER conditions and look over the results with the proposed aggregation scheme as compared to the basic scheme without aggregation:

Total Throughput Comparison in different BER conditions:

Figure 3.7 and Figure 3.8 show the results for the total throughput in low and high BER, respectively. Results showed an increase in throughput with the proposed aggregation scheme. It is almost eight times higher when the BER is low (Figure 3.7) and almost four times higher when the BER is high (Figure 3.8) as compared to that without aggregation. This confirms the advantage of the proposed aggregation technique. In addition, when the BER is high (Figure 3.8), the throughput reaches higher values with BAck than without BAck. This also confirms the advantage of using Block Acknowledgement when radio conditions are not optimal.







Figure 3.8 : Total throughput comparison at high BER

End to end delay Comparison in different BER conditions:

Figure 3.9 and Figure 3.10 illustrate the effect of traffic load on delay in both low and high BER conditions, respectively. For the end-to-end delay evaluation, we considered the delay sensitive flows that have small packet size traffic. It is shown that when the proposed QoS-HAN aggregation scheme is used, the end-to-end delay is reduced in both high and low BER cases. This is because of the reduction in the overhead time related to channel access and transmission between packets.

Furthermore, Figure 3.9 also shows that in low BER conditions, the end-to-end delay is almost the same, regardless of the use of BAck. This is because of few numbers of retransmission in low BER. However, in Figure 3.10, we can see that the delay with BAck is less than that without BAck. This is because in high BER scenario, frames get corrupted, and with BAck mechanism, only the selective corrupted frames are retransmitted rather than the whole AMPDU frame as is the case in the absence of BAck.



Figure 3.9 : End to end delay at low BER



Figure 3.10 : End to end delay at high BER

PDR comparison in different BER conditions

Figure 3.11 and Figure 3.12 illustrate the impact of traffic load. The PDR is close to 100% in low traffic load and degrades significantly with increasing load. In low BER (Figure 3.11), as the traffic load increases, the PDR value decreases considerably but is still higher than without aggregation. This is due to the fact that in the absence of aggregation, for each packet, an access to the radio channel is required. This results in higher collision ratio. On the other hand, aggregated packets. In addition, in high BER (Figure 3.12), aggregation with BAck provides a higher PDR value than without BAck. This is because of the selective retransmission of the corrupted packets when BAck is used. However, the PDR value continues to decreases at high load irrespective of the use of BAck.



Figure 3.12 : PDR at high BER

In the second set of experiment, we analyzed the performance of different traffic types in terms of throughput and end to end delay using our proposed aggregation solution.

Throughput and delay comparison of different traffic flow in different BER conditions

From Figure 3.13, we can see that using the proposed scheme, in low BER, the throughput obtained with packets of large sizes is higher than the ones obtained with smaller packet sizes. In contrast to this, in high BER, this behavior is reversed and large packets results in lower throughput. This can be explained by the fact that in the error prone environments, large frames are more likely to have errors and cause more frequent retransmissions because of delivery failure.

From Figure 3.14, it can be seen that the delay of small packet size traffic is always less than the larger ones in both high and low BER. This is because, for the same aggregation size limit, more number of small packets can be bundled in one frame and sent at once resulting in less delay between packets. This means that the proposed scheme can improve the delay of delay sensitive traffic that is generally characterized by small packet size (e.g. VOIP).



Figure 3.13: Throughput of different Traffic flow



Figure 3.14 : Delay of different Traffic flow

3.6 Conclusions

In this chapter, we proposed a new aggregation scheme that takes into account flows' requirements to guarantee adequate QoS for different applications. Aggregation can improve the performance; however, its effectiveness may degrade in high channel error. Hence, we also analyzed the performance of the proposed schemes in both channel conditions: one without error and the other prone to channel error conditions. Performance results showed that there is a reduction in the overhead and improvement in the throughput of the system with the use of the proposed aggregation solution. The use of Block Ack has proved to be a robust method to improve the decreasing throughput and increasing delay in high BER environment. In addition, the proposed solution enables distributed QoS differentiation without incorporating the IEEE 802.11e QoS mechanisms.

The presented work in this chapter has allowed gaining an insight on the use and performance of aggregation in different channel conditions and traffic types. Some considerations regarding the robustness of the scheme need to be addressed further. The performance of aggregation

depending upon the parameters like queue size, retransmission limit, apart from the traffic type and channel condition can be conducted. Furthermore, the use of aggregation with real traffic traces like video can also be evaluated. Adding dynamicity to the proposal to make it adaptive to channel conditions, traffic type etc (by making use of some learning techniques) can add supplement value to the proposal. In the following chapter, we will propose a learning based approach that can take into account some of this issues.

Chapter 4 QoE Estimation for Video: WLAN MAC layer Perspective

4.1 Introduction

As described in Chapter 1, internet traffic is growing exponentially due to the proliferation of mobile applications and the advances in wireless technologies that can offer connectivity with higher bit rates. Video traffic is expected to reach 66% of the global mobile traffic by the year 2015. More than one million minutes of video content will cross the Internet every second [75]. In these conditions, networking technologies should be able to adapt accordingly.

User satisfaction, namely Quality of Experience (QoE), is a key issue for the success of video services. For multimedia services, quantitative parameters, usually used to describe the Quality of Service (QoS) for traditional applications (i.e. voice and data) are not sufficient to render user satisfaction. QoS indicators are generally used for determining the service quality from the underlying network layer perspective (i.e. PHY, MAC, and NTW). Traditional QoS parameters reflect network and service level performance; however, they do not address/reflect user's reaction/satisfaction to the service or application that can depend on the global user context including network conditions, terminal capabilities, and user behavior.

QoE has become an important indicator, useful for both network operators and service providers. It helps them to understand user acceptability towards a particular service or application, and allows them to adapt or tune their networks in order to enhance the QoE for each user.

The relationship between QoS and QoE is hard to identify and express. For sure, this relationship is not linear and higher QoS level does not always yield to higher QoE. Indeed, there are many parameters that can affect the quality of video and their combined effect is not well identified and understood.

The ability to identify the perceived degree of video impairment with respect to MAC-level parameters is a key point in the quality estimation of video traffic.

In this chapter, we focused on mapping WLAN MAC-level parameters to user perceived QoE in order to provide tools to help measure the impact of the MAC-level parameters on the video QoE. The effect of MAC-level parameters on video can range from distortion-less to intolerable distortion. In order to achieve this objective, we first performed an intensive analysis of the impact of different MAC-level parameters on video QoS and QoE through simulation and subjective tests respectively. We then proposed a Random Neural Network (RNN) based solution for predicting the QoE from these parameters. The obtained results can help in MAC-level parameters tuning in order to increase overall performance of video transmission.

4.2 Quality of Experience: background and definitions

According to the ITU-T Focus Group on IPTV [76], Quality of Experience (QoE) refers to "*the overall acceptability of an application or service, as perceived subjectively by the end- user*". Therefore, QoE is a subjective measure and can vary according to the user expectation and context. Moreover, it is an overall end to end system effect (client, terminal, network, services infrastructure, media encoding, etc.) and depends on a number of factors that cannot be simply measured. It requires tests with actual users in a controlled environment which is costly and time consuming.

In the literature, different estimation approaches for video services are proposed. These can be subjective or objective.

Some of the objective methods available for video quality estimation are PSNR, SSIM, VQM, PEVQ and MDI [77]. Most of them measure video quality based on pixel to pixel comparison.

On the contrary, subjective methods consist in rating video quality according to predefined quality scale. Rating depends on the personal perception of a users' panel after viewing a set of multimedia samples. Subjective video quality estimation methods require appropriate test environments. In [78], general environment conditions for subjective laboratory tests are defined. Authors state that each user should be screened for proper eye sight using specially selected charts (Ishihara, for instance). At least 15 participants are required in order to consider these tests statistically reasonable.

4.2.1 Subjective QoE test methods

The main subjective test methods available for video quality estimation are Mean Opinion Score (MOS), Double Stimulus Continuous Quality Scale (DSCQS), Double Stimulus Impairment Scale (DSIS), Single Stimulus Continuous Quality Scale (SSCQE), and Subjective Assessment Methodology for Video Quality (SAMVIQ).

- Mean Opinion Score (MOS)

It is one of the most common subjective QoE estimation methods for video services. With MOS, participants are asked to rate the overall quality of the multimedia service, however, they are not provided the original reference. The Absolute Category Rating (ACR) [10] is used for rating the quality. The final MOS score is defined as the average of the scores collected from the users' panel. MOS score values depending upon impairments are presented in Table 4.1.

MOS Score	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very Annoying

Table 4.1 MOS scores and its categories

- Double Stimulus Continuous Quality Scale (DSCQS)

With DSCQS, participants provide scores to multiple video sequence pairs consisting of original and test sequences. Video sequences of around 10 seconds are required. The pairs are shown twice in an alternating fashion and in a random order. Participants are unaware which sequence is an original and which one is a test sequence. They rate the quality on a 0 to 100 scale or scale from bad to excellent). The difference between these two scales is used to remove the uncertainties caused by the material content and/or viewer's experience [78].

- Double Stimulus Impairment Scale (DSIS)

In DSIS method [78], multiple video sequence pairs (original and test) are presented once to users. The original sequence is always shown before the test sequence. As in DSCQS, if longer test sequences (above 10 seconds) are shown, the time interval between the original sequence

and the test sequence is increased. In this case and according to [79], it may be difficult to rate the sequences accurately and a psychological recency effect can be noticed.

- Single Stimulus Continuous Quality Scale (SSCQE)

With SSCQE, a video sequence of around 5 minutes is shown to the participants. Each participant has to evaluate the video quality instantaneously by continuously adjusting the slider each 1 to 2 seconds. The DSCQS scale (from Excellent to Bad) is used in this case [78]. The reference video sequence is not provided to participants. Due to a longer video sequence, this method may create difficulty at the user end when comparing scores for different test sequences and deciding the overall quality rating for a particular test sequence. Also, the scores provided by end-users can be affected by a "recency or memory" effect [79].

- Simultaneous Double Stimulus Continuous Evaluation (SDSCE)

This method was developed to perform the subjective test in TV or similar environment. The original and the test video sequences are simultaneously shown to the participants. They check the difference between these two sequences. The quality rating is done by moving the slider of a handset-voting device using a [0-100] scale. If the users do not notice a difference between the two sequences, the slider should be placed at the top i.e. 100. However, when the user notices a significant difference in the quality, the slider should be placed at the bottom, i.e. 0. Participants are unaware about the identity of original and test sequence [78].

- Subjective Assessment Methodology for Video Quality (SAMVIQ)

SAMVIQ is used for video rating on PC and mobile environments. Participants are shown different versions of the same video sequence and when all the sequences are rated, the following sequence content can be then accessed [80]. Here, participants are allowed to access any version of the sequence, replay, start or stop, change or keep the current score according to their preference. Participants are also allowed to view one version at time and the DSCQS method is used for rating purpose.

4.3 Impact of MAC level parameters on video QoE

In this section, we focus on analyzing the impact of MAC-level parameters on user perceived quality (i.e. QoE). The objective is to understand how MAC-level parameters can affect the video quality and to explore if new MAC adaptation and tuning techniques are possible.

4.3.1 Related work

Most of the studies in the literature in the area of video QoE estimation consider a network layer perspective and measure the impact of network QoS parameters on the perceived quality for video traffic (i.e. QoE) [81], [82], [83], and [77]. Most of these studies are conducted over wired networks where network layer impairments are considered as the main possible causes for video degradation. In this case, network load and therefore the associated end-to-end delay and packet dropping probabilities are the main considered QoS parameters.

Today, since wireless technology is becoming dominant and its performance being fairly governed by MAC, analyzing the effect of MAC-level parameters on video QoE becomes important. Indeed, the wireless access networks are more and more responsible for the overall QoS.

In the literature few works has been done in this area. In [72], authors evaluated the performance of VOIP services in IEEE 802.11n WLAN environment. They considered IEEE 802.11n MAC parameters: aggregation, block acknowledgement, reverse direction mechanism and BER as QoS parameters for analyzing audio quality. They showed that the combination of aggregation and block ACK mechanism can significantly improve the performance of services with small packets in need of high throughput. Moreover, they showed that IEEE 802.11n indeed improves the channel efficiency and provides high quality WLAN networking support for VoIP service. Authors in [84] studied the impact of IEEE 802.11n frame aggregation mechanisms on video streaming. They showed the impact of the different MAC aggregation methods and frame sizes in the decoded video quality. Source video content were streamed to the destination varying different aggregation size and frame sizes and used Peak Signal-to-Noise Ratio (PSNR), which is an objective metric, to determine the quality of distorted video content. A suitable MAC layer and video encoding parameterization was the found to optimize the overall video quality. In [85], authors analyzed the impact of background traffic on video quality in IEEE 802.11b WLAN environment. In this study, PSNR is used together with Video Quality Metric (VQM) and Structural SIMilarity (SSIM) to assess video quality. In contrast to the studies presented in [84] and [85], authors estimated QoE of audio-video services from MAC-level QoS parameters over IEEE 802.11e EDCA WLAN environment. They examined MAC-level QoS by varying parameters like distance between the nodes and TXOP (transmission opportunity) limit. They considered wireless stations that transmit audio and video flows to an access point (AP). Subjective tests were performed by the method of successive categories, which is a psychometric method to analyze the effect of these parameters on video and audio quality. Authors showed a mapping between MAC-level and user-level with multiple regression analysis.

The above studies did not consider the combinational effect of several other MAC and PHY related parameters. Although, some parameters like aggregation, block Ack were considered in [72]. However, it was used to analyze the VOIP quality however in this work we are focused on video services. In [85] authors considered the video services, however they based their evaluation on psychometric method. In contrast, this work is based on using learning techniques and considers IEEE 802.11n network as compared to IEEE 802.11b network used in their work.

4.3.2 Potential MAC-level parameters

Different MAC-level parameters are useful to evaluate the effect of the medium access procedure on video QoE. Some of these parameters are either directly associated to MAC layer (e.g. queue size, aggregation, retransmission limit) or indirectly associated. In this case, they provide information about the lower layer performances (e.g. BER) and the global context of the mobile terminal (e.g. number of competing stations). In the following, all these parameters are denoted as MAC-level parameters.

- **Bit Error Rate (BER):** Wireless networks are subject to error prone wireless channels. This results in a continuous variation of the BER that can cause the MAC frame to be received with errors, require retransmissions and impact the overall performance of the system [72].
- Aggregation Size: Frame aggregation allows combining a number of MAC frames into one larger aggregated frame. Streaming applications can take benefit from frame aggregation and the mean aggregate size can be determined depending upon both the resolution and the wireless channel conditions. Limiting the frame aggregation can severely impact both the average delay and the quality of a video stream. However, in high BER, aggregation size should be limited to reduce larger frame drops [86].
- Number of competing stations: The performance of IEEE 802.11 based wireless network degrades with increasing number of users. Indeed, as the number of competing

stations increases in the system, stations have to struggle most of the time to find a free channel (due to high contention and collisions). This may lead to larger delays and packets dropping ratio.

- Queue length: At the expense of larger delays, a large queue can hold more packets to avoid higher packet dropping ratio. On the other hand, small queue length can increase packet dropping ratio, which is more likely to happen in saturated conditions.
- Retransmission Limit (Maximum number of retransmissions): Retransmissions at the MAC layer are a good policy to deal with corrupted packets. When channel conditions are not highly error prone, using different values for maximum retransmission limit in different channel conditions can enhance the system performance. However, this should comply with delay bounds defined for each traffic type (i.e. TTL: Time To Live).

4.3.3 An experimental platform to evaluate the impact of MAC parameters on QoE

In order to evaluate the impact of MAC-level parameters on video QoE, we set a dedicated experimental platform to help the extraction of user perceptions on a number of video sequences, when transmitted over a wireless network with variable MAC conditions and parameters.

The experiments are composed of two main steps:

- Generating video sequences with different MAC-level perturbations through simulations and
- QoE evaluation through real user scoring

Step 1: Video sequences generation

Video sequences are generated through the simulation of a realistic home wireless environment. Simulations are based on the NS2 simulator with an embedded IEEE 802.11n module. Different stations are considered. Each of them transmits a video traffic to the corresponding paired station. Three different types of video were considered: sports, movies, and animations. The characteristics of each of these video are shown in Table 4.2.

Table 4.2: Video Cl	naracteristics
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Туре	bit rates	Resolution	Frame per	Duration
	(kbps)	(pixels)	second	(sec)
Movie	2549	640 x 360	24	15
Animation	2508	640 x 360	24	15
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Sports	2686	640 x 360	24	15

At first, video flows based on H264/AVC real traces are fed to a simulator using the Evalvid tool [87]. During simulations, different MAC-level perturbations were introduced. Besides video traffic, CBR (Constant Bit Rate) data flows are also injected at different stations. General simulation parameters are presented in Table 4.3, while MAC related parameters are presented in Table 4.4.

Parameters	Values
Simulation Area	250X250m2
Simulation time	30 sec
Node mobility	Static
Bandwidth	96 Mbps
MAC layer	IEEE 802.11n
Video packet size	1052 bytes
CBR packet size	1000 bytes
Time slot	20µs
SIFS	10µ s
TXOP limit	3.264 ms
NS2 Antenna Type	Omni Directional
NS2 Network Interface Type	Wireless/Physical/MIMO
NS2 Interface Queue Type	Aggregation queue
Number of antennas	4
NS2 MIMO Type	SM-STBC
Propagation model	Two Ray Ground

Table 4.3 : Simulation Parameters

Table 4.4 : MAC parameter values

Parameters	Values
BER	0.00001,0.0001,0.00025,0.0003, 0.00035,
	0.0004, 0.00045, 0.00045, 0.0005, 0.0007
Aggregation size	4000,8000,16000, 32000, 64000 bytes
Number of competing nodes	4, 8,10,12,16,18,20,24,30,34,
Total traffic load in the system (total CBR	4,8,12,16,18,20,24,30,34,120 Mbps
rate excluding the video traffic rate)	
Queue length	50,120,150,200,250,300 packets
Maximum Retransmission limit	1,2,3,4,5,6,7

At the end of this step, a total of 200 samples of video clips were constructed in this manner.

Step 2: Subjective Tests

Initial video clips and distorted ones (obtained from step 1) were shown to a panel of real users, in a random order, in order to conduct subjective scoring.

A total of 40 users registered for the test so far which is considered reasonable for this kind of subjective tests [88]. Out of the 200 generated video clips, each participant rated 100 video clips on average. Participants rated each video clip according to MOS scoring technique (see section 4.2.1). After watching 20 video clips a pause of 5 minutes was allowed. Figure 4.1 illustrates the subjective tests experimental environment used in this step.

4.3.4 QoE results and analysis

The impact of MAC level parameters on different QoS metrics as well as on the perceived QoE as scored by real users is then analyzed. The used QoS metrics are the following:

- Number of MPDU drops
- Collision rate and
- Jitter



Figure 4.1 : Subjective test Environment

The first set of experimentations is dedicated to analyze the impact of different MAC-level parameters on video frame delivery ratio. As shown in Figure 4.2, frame delivery ratio is very sensitive to MAC-level parameters variation. Since video quality is extremely sensitive to packet delivery and dropping rates, it is expected that the QoE as scored by users also follow this trend.



Figure 4.2 : Frame delivery ratio vs. different MAC-level parameters

Figure 4.3 presents both QoS and QoE results when varying the number of competing stations. As shown, QoS indicators (i.e. MPDU drops, collision rate, and jitter) increase when the number of competing stations increases. In addition, the impact of the number of competing stations on the video QoE in presented in sub-Figure 4.3 d.



Figure 4.3 : QoS and QoE versus number of competing stations

These initial results show that the variation in MAC-level parameters has a subsequent effect over both video QoS and QoE.

Similar behavior is observed with other parameters, i.e. BER, aggregation size and load.

In the following results, the relationship of different MAC-level parameters and QoE are presented and discussed. We considered different video contents (i.e. sport, movies, and animation). In the figures, dotted curves represent the average MOS score given by the participants for different video clips and the line-curves represent the trend that these scores follow. The trend lines were drawn using either exponential or logarithmic fit, according to the pattern, the dotted points followed.

<u>Impact of Bit Error Rate on video QoE</u>

Figure 4.4 shows the impact of BER on the video QoE. As shown, MOS values seem directly correlated to BER values. An increase in BER will increase the number of corrupted video frames due to transmission errors, leading to dropping of video frames. From subjective test results, we see that for BER value less than 1×10^{-4} , almost no degradation is seen on video transmission and the perceived MOS score is 5. However, when BER value increases above 2×10^{-3} , video transmission error increases, causing high degradation in video quality, hence, reducing the perceived quality by users.



Figure 4.4 : Impact of BER on video QoE

• Impact of the number of competing stations on video QoE

With CSMA/CA, an increase in the number of stations in the system makes channel competition more vulnerable and station has to struggle to get a successful channel access.

Figure 4.5 shows that the number of competing stations also directly impacts QoE. As shown, when the number of competing stations increases in the system, the user perceived video QoE decreases. Subjective tests show that QoE of video transmissions does not degrade much as long as the number of competing stations is less than 10. However, as the number of competing stations in the system increases above this value, the perceived QoE degrades accordingly.

<u>Impact of Queue length on video QoE</u>

Station queues can get full when the available bandwidth in the system is less than the offered data rate (incoming traffic). This results in queue overflows and hence leads to frame dropping. If a large queue size is assigned to the station, it may result in outdated information as packets can be stored in the queue for a long time.

The effect of different queue lengths on video QoE with different numbers of competing stations can be analyzed through Figure 4.6. We can observe that, as the queue lengths decrease, the video QoE decreases accordingly. The effect of queue length seems more important when the number of competing traffic flows is high. For example, for a sport clip with queue length of 200 and 2 competing stations, the MOS value is almost 5. However, for the same queue length with 5 competing stations, MOS score is only 2.5.



Figure 4.5 : Impact of number of competing stations on video QoE



Figure 4.6 : Impact of queue length on video QoE

• Impact of Maximum Retransmission Limit on video QoE

Using different retransmission limit values in different channel conditions can increase the performance of video transmissions. To analyze this effect, different values of BER were introduced in the radio channel in order to cause retransmissions. From Figure 4.7, we can see that as the retransmission limit increases, the video QoE increases for both high and low BER cases. For low BER, the value saturates at MOS of 5 when retransmission limit reaches 7. However, after a certain value of BER is exceeded, only increasing the transmission limit may not improve the video QoE. For instance, we can see that at high BER, the MOS value saturates between 1 and 2. This is because in high BER, most of the frames get corrupted even the retransmitted ones.



Figure 4.7 : Impact of maximum retransmission limit on video QoE

4.3.5 Conclusions

In this section, the effect of a set of MAC-level parameters on video QoS and QoE is analyzed. The perceived video QoE (subjective measurements) is obtained for different video contents, over a wireless IEEE 802.11 access enabled home environment. The impact of MAC parameters is more noticeable on sport and animation contents than on movie content.

In the following section, we explore the combinational effect of the MAC parameters on QoE aspect and propose a random neural network based solution for QoE prediction.

4.4 A Random Neural Network based QoE estimation and Learning

In the above section, we analyzed how different MAC parameters can affect the quality of video services (QoS and QoE) over IEEE 802.11n wireless networks. Understanding and predicting the combinational effect of these parameters seem important to make network adaptations. As the relation between MAC parameters and QoE appears to be non-linear and difficult to predict easily, we propose here a neural network based approach that can be used to predict the QoE of the video in different network conditions.

In the literature, different learning techniques and algorithms have been used for QoS/QoE estimation. The comparative analysis provided in [89] is extended with some of the relevant

work in this regard and presented in Table 4.5. It can be seen that most of the existing works for QoE estimation uses the application level QoS parameters (AQoS) or Network level QoS (NQoS) parameters or both. However, very few work consider MAC level parameters (MQoS) to predict QoE as we intend to do in this work. In addition, we also provide a comparison based on the type of evaluation method (subjective or objective or both) used for obtaining the video quality metrics in these works including the learning technique used.

Aspects/ Models	AQoS Parameters	NQ0S Parameters	MQoS Parameters	Video coding Parameters	Subjective Metrics	Objective Metrics	Learning Technique
Menkovski et al [90]	×				×		Support Vector Machine, Decision Tree
Machado et al [91]		×			×	×	Artificial neural network
Du et al [92]		×			×	×	BP (Back-Propagation) Neural Network
Khan et al[93]	×	×				×	Adaptive Neural Fuzzy Inference System & non-linear regression analysis
Frank et al [94]	×	×		×	×		Artificial neural network
Calyam et al [95]	×	×		×	×	×	Neural network
Staelens et al [96]		×		×	×		Decision tree
Kang et al [97]	×	×		×		×	Neural network (RFB(Radial Basis Function))
Cherif et al [98]		×			×		Neural network (RNN)
Suzuki et al., 2008 [99]			×		×	×	Regression Analysis
Our Proposal			×		×	×	Random Neural network (RNN)

 Table 4.5 : Learning technique used for QoS/QoE

AQoS - Application level QoS, NQoS -Network level QoS, Subjective metric - MOS, Objective metric - PSNR, SSIM, VQM, etc.

In our work, Random Neural Networks (RNN) will be used for QoE prediction based on MAClevel indicators and parameters. The success of the use of RNN for learning is suggested in a number of works [100], [101], [102], [103], [104], [105] and [106].

RNN are a variant of Artificial Neural Network (ANN), well adapted for QoS/QoE learning [106]:

- ANN can be trained is a short time as compared to RNN, however with RNN the overall time of calculation is less. This makes RNN attractive to be used in systems with real time constraints or for any type of light weight applications.
- RNN can capture the mapping function that is used to map various parameters to the quality metric in a more robust way and with higher accuracy.
- RNN are less sensitive to the selection of hidden nodes as compared to ANN.
- RNN are effective in extrapolating and predicting the output in a coherent way, even for the values of some parameters that are not within the range provided during the training phase.

Before presenting the proposed solutions, the following section is dedicated to briefly describe RNN based learning.

4.4.1 Random Neural networks

A Random Neural Network (RNN) is a variant of Neural Networks that combines classical Artificial Neural Network (ANN) and queuing networks. As ANN, RNN is composed of a set of interconnected processing elements (i.e. neurons/nodes). These neurons cooperate together to solve a specific problem by instantaneously exchanging signals between each other and from/to the environment. The network can be divided into different levels (or layers), each containing a number of neurons. Each neuron is associated with a potential *q* represented by an integer (random) variable. At any time *t*, the potential of neuron *i*, is given by $q_i(t)$. When this potential value is positive, the neuron is considered to be excited. In this state, the neuron randomly sends either a positive or a negative signal to the other neurons or to the external environment. The sent signals follow a Poisson process with rate r_i . The probability that neuron *i* sends a positive (resp. negative) signal to neuron *j* is denoted by p^+_{ij} (p^-_{ij} respectively). The probability that a neuron *i* send a signal to the external environment is denoted by d_i .

For the total number of *N* neurons in the system, we have:

$$d_i + \sum_{j=1}^{N} (P_{i,j}^+ + P_{i,j}^-) = 1$$
(5)

For every positive signal that a neuron receives from either the environment or from any other neuron, its potential is increased by 1. If it receives a negative signal its potential value is decreased by 1 (whenever its former potential is positive). If its potential was 0, it remains unchanged. Similarly, for every transmission of a positive or negative signal, the neuron reduces its potential by 1.

The positive and negative signals arriving at a neuron from the environment follow a Poisson processes with rate λ_i^+ and λ_i^- .

Let Q_i denotes the probability that in equilibrium, a neuron *i* is excited. According to [106], for a network to be stable, all nodes should have $Q_i < 1$.

To use RNN as a learning tool, we can consider RNN as a black box that can have N inputs and M outputs. Only positive signal flows may be considered to arrive from outside (i.e. $\lambda_i^- = 0$). The output is be denoted by Q_i s. When a neuron does not receive any signals from outside the value of λ_i^+ will also be 0 and the output will consists of only a certain subset of Q_i s [106].

Once the number of neurons is fixed and the topology is selected for the network, neurons start exchanging signals. The learning phase has as objective to find values for the rate r_i and the branching probabilities $p_{i,i}^+$ and $p_{i,i}^-$, that are not fixed initially.

If we consider a system composed of K input-output pairs, denoted by

$$\{(\underset{x}{\rightarrow}^{(k)},\underset{y}{\rightarrow}^{(k)}), K = 1, \dots, K\}, \text{ where } \underset{x}{\rightarrow}^{(k)} = (x_1^{(k)}, \dots, x_N^{(k)}) \text{ and } \underset{y}{\rightarrow}^{(k)} = (y_1^{(k)}, \dots, y_N^{(k)}).$$

we should find the values of the rate and the branching probabilities in such a way that if in the RNN we set $\lambda_i^+ = x_1^{(k)}$ for all *i*, then the steady state occupation probability Q_i should be close to $y_1^{(k)}$ and this should be true for any value of $k \in \{1, ..., K\}$

To achieve this result and to find the values of rate r_i and the branching probability, a new variable is used. It is known as the weights and is given by

$$w_{i,j}^+ = r_i p_{i,j}^+$$
 and $w_{i,j}^- = r_i p_{i,j}^-$

Where $w_{i,j}^+$ (*resp.* $w_{i,j}^-$) denotes the throughput of the positive signal going form neuron *i* to neuron *j* in an equilibrium state [106].

At first, these weights can be initialized with arbitrary positive values. Then, they will be modified iteratively (K iterations). Let $w_{i,j}^{+(0)}$ and $w_{i,j}^{-(0)}$ denote the initial weights for the connection that exist between neurons *i* and *j*. Then, for $k = 1, \dots, K$, the weights at step *k* can be calculated from those of step K-1 using a learning method.

If a network $\mathcal{R}^{(k-1)}$ is assumed to be obtained after k-l step with weights defined to be $w_{i,j}^{+(k-1)}$ and $w_{i,j}^{-(k-1)}$ and the input rates to be $x_1^{(k)}$ s, a steady state occupation can be obtained to be $Q^{(k)}$ s (in a stable condition).

Following this, the weights can then be defined by the following equations as given in [106].

$$w_{i,j}^{+(k)} = w_{i,j}^{+(k-1)} - \eta \sum_{l=1}^{N} c_l \left(Q_l^{(k)} - y_l^{(k)} \right) \frac{\partial Q_l}{\partial w_{i,j}^+}$$
(6)

$$w_{i,j}^{-(k)} = w_{i,j}^{-(k-1)} - \eta \sum_{l=1}^{N} c_l \left(Q_l^{(k)} - y_l^{(k)} \right) \frac{\partial Q_l}{\partial w_{i,j}^-}$$
(7)

After using the *K* learning values, the whole process is repeated until some convergence condition is met. Then, for each given input $\xrightarrow{k} (k) (k=1...N)$, the output $\xrightarrow{k} (k) (k=1...N)$ can be obtained.

Neurons are trained in such a way that the actual output of the network matches the desired output with minimum error.

As ANN, most RNN used for learning purposes are composed of a 3-layer network structure. Neurons are divided into three subsets:

- input nodes,
- intermediate or hidden nodes and
- output nodes.

Input nodes receive the positive signals from the environment and do not send any signal to it (*i.e.* $\lambda_i^+ > 0$ and d = 0). Output nodes send signals to the outside environment without receiving from it (i.e. $\lambda_i^+ = 0$ and d > 0). The intermediate or hidden nodes are not directly connected to the environment, ($\lambda_i^+ = \lambda_i^- = d_i = 0$). In this case, the input neurons connect to

the hidden nodes and the hidden nodes connect to the output nodes. Below are the three equations that gives the values of output, hidden and output node respectively [106].

$$\boldsymbol{Q}_{i} = \frac{\lambda_{i}^{+}}{r_{i} + \lambda_{i}^{-}} \tag{8}$$

$$\boldsymbol{Q}_{h} = \frac{\sum_{i=1}^{I} Q_{i} \quad W_{i,h}^{+}}{r_{h} + \sum_{i=1}^{I} Q_{i} \quad W_{i,h}^{-}}$$
(9)

$$Q_{o} = \frac{\sum_{h=1}^{H} Q_{h} \ W_{ho}^{+}}{r_{o} + \sum_{h=1}^{H} Q_{h} \ W_{ho}^{-}}$$
(10)

4.4.2 Proposed RRN based QoE estimation approach

The proposed RNN based QoE estimation solution is described below.

In order to determine the subsets of input, hidden and the output nodes of the RNN, the different parameters to be considered should be identified. Since we considered four MAC parameters namely BER, aggregation size, number of competing station and traffic load, these correspond to input nodes of the system, while the output correspond to the QoE estimated values. The number of hidden nodes will be determined to meet the acceptable root mean square error.

We followed similar procedure and simulation environment as explained in section 4.3.3 to obtain the video sequences affected by the combination of different MAC-level parameters. The characteristics of the video sequences used for this experimentation is shown in

Table 4.6.

During simulations, we examined the performance of video under the combined effect of four different MAC-level parameters: BER, number of competing stations, load and aggregation. Used values are given in Table 4.7.

Туре	Bit rates (kbps)	Resolution (pixel)	Frame per second	Time (sec)
Video clip	2508	640 x 360	24	15

 Table 4.6 : Video characteristics

Table 4.7 : MAC parameters values

Parameters	Values
BER	0.00001,0.0001,0.00025,0.0003, 0.00035,
	0.0004, 0.00045; 0.00045, 0.0005, 0.0007
Aggregation size	4000,8000,16000, 32000, 64000 bytes
Number of competing nodes	4, 8, 10, 12, 16, 18, 20, 24, 30, 34,
Total traffic load in the system (total CBR	4,8,10,12,16,18,20,24,30,34,120 Mbps
rate excluding the video traffic)	

Although the retransmissions limit and queue sizes are also considered as potential MAC parameters and varied accordingly in our earlier analysis, these two parameters are constant in this experimentation as in real scenarios, these are not varied frequently. The queue size is fixed to 200 packets and the retransmission limit is set to be 7.

For this set of experiment, a total of 390 distorted video clips are obtained. These correspond to different MAC-level perturbations. For the subjective tests, we followed similar approach as the one used in the section 4.3.3. The different video sequences (with different levels of distortions) were shown to users for subjective quality evaluation. A total of 25 users registered for this test which is considered reasonable for this kind of subjective tests [88].

<u>RNN module for video QoE prediction</u>

After all the videos were rated and the subjective data (MOS score for each distorted video) were obtained, two data subsets are extracted:

- a training set : used to train the RNN and
- a test sets used to test the accuracy of the RNN

Figure 4.8 presents the QoE estimation environment with RNN. The RNN used for QoE estimation has 4 input nodes (variable MAC-level parameters), 5 hidden nodes (selected based on the best root mean square error), and 1 output node (video QoE). Input, hidden and output nodes are denoted by 'I', 'H', and 'O' respectively.

Once the RNN is trained, the system is used for real-time QoE estimation without any human participation.



Figure 4.8 : QoE estimation module

• Validating the proposed RNN

In order to validate the proposed RNN, test video data sets (obtained from users' subjective tests) and those obtained from the proposed RNN system are compared.

Figure 4.9 presents the obtained subjective QoE from users and estimated QoE with RNN for a set of video contents. Each point along the X-axis represents a particular video clip that was randomly selected. Each red point in the figure represents the estimated QoE (RNN) of a particular video clip and blue point represents the subjective MOS as scored by users. The Pearson correlation coefficient between subjective and objective QoE was found to be 0.89, which is considered to be high correlation.



Figure 4.9 : Comparison between subjective and estimated QoE

Figure 4.10 provides the probability distribution of the difference between the participants' subjective scores (MOS) and the estimated scores. As it can be seen, the score differences were less than 0.5 in around 58% of the tests. It reached 78% for score differences less than 1. These values are considered acceptable for QoE estimation systems and hence justify the competency of the proposed system in reflecting the user's perception. This estimation accuracy thus

emphasizes the ability of the proposed system to measure the impact of the MAC-level parameters on the user satisfaction.



Figure 4.10 : Probability distribution of the subjective and estimated MOS difference

4.4.3 **QoE prediction**

The trained random neural network, is used it to evaluate the QoE of video services with a large set of MAC-level parameter variations. For this purpose, trained system is fed with different combinations of MAC-level parameter and the predicted QoE values are reported.

The first prediction results (Figure 4.11) are related to both the number of competing stations and BER. A maximum aggregation size of 64000 bytes and 1 Mbps data flows are considered. Video traffic rates are those given in Table 4.6.



Figure 4.11 : Variation of QoE in different load and different BER conditions

Results show that when the BER is low (≤ 0.00001) and the number of competing stations is below 16, user can experience a good video QoE with MOS values above 4. However, when BER increases slightly (BER ≤ 0.0005), the same video quality can be experienced when there are few competing stations in the system (< 12 as compared to 16 in the earlier case).

In addition, in high channel error condition (BER ≥ 0.00095), the video QoE can be affected even when there are only two competing stations in the system. This is true because in high BER scenario most of the frames are lost and retransmission cannot help much. The number of competing stations has no effect in this case. On the other hand, in very good channel conditions, the increase in the number of competing stations above 20 may degrade the users experience and the MOS values reduce below 3. This is expected because of the increase in the collision rate during channel access. We then analyzed prediction results when varying the aggregation size and BER simultaneously. Results presented on Figure 4.12, Figure 4.13 and Figure 4.14 are obtained for different numbers of competing stations (i.e. 5, 12 and 20).



Figure 4.12 : Variation of QoE in different BER and aggregation limits (5 stations)

Figure 4.12, shows that when the BER is between the interval 0.00001 to 0.0001 and the aggregation size is between 44000 and 64000, users can experience high video QoE with MOS values between 4 and 5. However, for the same BER case, small aggregation values 3000 to 12000 may result in almost bad video experience (MOS value less than 3). This is as expected since in good channel conditions, it is generally better to aggregate larger frames to reduce waiting time in the queue.

From Figure 4.13, we can see that when the number of stations increases to 12, it is possible to achieve better QoE with an aggregation of 64000 below a BER value of 0.00005. However, we can see from Figure 4.14 that in the same condition (aggregation of 64000 and BER value of 0.00005), the video quality is limited to a MOS of less than 3.5. This is because of the increase in the number of stations (= 20). This is in accordance with the results presented in Figure 4.11.



Figure 4.13 : Variation of QoE in different BER and aggregation limits (12 stations)



Figure 4.14 : Variation of QoE in different BER and aggregation limits (20 stations)

4.5 Conclusions

In this chapter, we first analyzed the impact of different MAC-level parameters on the QoE of video traffic over IEEE 802.11n wireless networks. Considered parameters are BER, number of competing stations, queue length, and retransmission limit. Intensive subjective tests were performed to analyze the effect of these parameters on video QoE. The experimental results confirm a direct relationship between individual MAC-level parameters and video quality. Furthermore, it was observed that sport and animation contents are more sensitive than movie

content. Therefore, video service vendors/operator should be more cautious while providing QoS/QoE to different type of video contents. Our studies show that careful parameterization of MAC layer parameters can provide an improved QoE for video applications, when used over a wireless network.

The second contributions consist in the proposal of a Random Neural Network (RNN) based solution for QoE prediction. RNN is constructed and trained with a subjective QoE dataset. We then analyzed the combined impact of the different MAC-level parameters on video QoE. For this, the RNN is fed with different input sequences of variable MAC parameters and the predicted QoE is reported. Obtained results show that the proposed approach constitutes a good tool to predict QoE when varying the input parameters.

The proposed scheme can be extended for further studies by incorporating more parameters and use cases for its training and evaluation. It can be useful in the design of adaptive or Soft MAC techniques that can adapt parameters according to network conditions and user context, in order to provide better QoE to users in the system.

Through the results, it appears that even with the increasing data rates provided by recent IEEE 802.11n standards, increasing number of competing nodes can still impact the performance of video services severely. This is true even in good radio conditions (i.e. low BER cases). This is due to the increase in collision rates, caused by the CSMA/CA technique. Hence, enhanced (or alternative) channel access solutions for WLANs are required to meet QoS requirements for multimedia services in such scenarios.

Chapter 5 An enhanced channel access and differentiation scheme for IEEE 802.11 WLAN

5.1 Introduction

One of the important aspects of WLANs is wireless medium sharing. This is handled by the medium access control (MAC) protocol that uses the coordination mechanism to provide stations a fair channel access for successful transmission. DCF (Distributed Coordination Function) was defined as the primary medium access method based on Carrier Sense Multiple Access/Collision Avoidance (CSMA/CA). In chapter 2, we presented the evolution of IEEE 802.11 standards. We mainly described the enhancements proposed at the MAC layer, including aggregation and block acknowledgement in IEEE 802.11n, and the TXOP-sharing in IEEE 802.11ac.

Despite these enhancements, DCF remains the basic access technique at layer 2 for all these standards. It is based on the CSMA/CA technique that uses a Binary Exponential BackOff (BEB) algorithm for collision recovery. Though, BEB is a simple BackOff strategy, it comes along with the limitation of increased overhead due to wasted idle slots during collision resolution.

Large BackOff values can significantly limit the throughput of some backlogged flows. Furthermore, at high load, the random nature can accelerate the occurrence of collisions. In these circumstances, access to the channel cannot be guaranteed for QoS sensitive applications. Network efficiency is reduced in terms of throughput, delay, jitter, etc.

BackOff algorithms are initially designed to reduce collisions and to maintain fairness among stations in the system. They are not intended to provide regular access in order to support delay sensitive traffic like voice and video.

When we consider different evolutions in terms of usage and traffic types (see Chapter 1), WLAN access mechanisms have to evolve to provide QoS guarantees while maintaining backward compatibility. For WLAN environment, new Physical enhancements are proposed in the standard to promise data rate up to 7Gbps (IEEE 802.11ac). However, MAC enhancements

are not sufficient to meet the QoS. Indeed, with the available bandwidth, one can consider that enough resources are available to handle multimedia traffic. In the literature, not many proposed solutions can improve MAC performance for WLANs. They are still based on CSMA/CA techniques that suffer low efficiency.

When analyzing the behavior of Wi-Fi for QoS sensitive flows, we can easily observe performances degradations when the number of competing flows increases. In addition, DCF is not able to provide traffic differentiation needed to support a variety of traffic mix in the system.

In the IEEE 802.11 standards, the EDCA mechanism is proposed to handle QoS. However, this mechanism is not really efficient. On the one hand, when the load increases, the low priority flows may restrain the high priority flow from fulfilling their QoS requirements. On the other hand, when there are many concurrent flows belonging to the same access category, service differentiation may not be completely realized. For instance, in the typical current home environment, several multimedia streams can be seen running in parallel and each may be assigned the same Access Category (AC). In such cases, these flows competes with the same traffic parameters, and at a certain level of saturation, they start degrading the performance of each other, resulting in neither of the flow fulfilling its QoS requirements.

In this chapter, we propose new approaches to handle QoS in Wi-Fi networks. First, we propose I-DCF, a new BackOff scheme to reduce collisions in the system. I-DCF makes use of contextual parameters (from stations and traffic flows) to determine personalized BackOff values to use in channel contention. The basic idea behind I-DCF is to maintain a similar behavior as the original IEEE 802.11 DCF while providing TDMA-like performances in terms of channel access regularity.

We then propose a new AIFS differentiation algorithm to improve the efficiency of the system and to provide traffic differentiation between flows. The basic principle is to select AIFS values of the different traffic types based on queue information.

5.2 Background and related work

5.2.1 BackOff Algorithms

BackOff algorithms are used in order to avoid collisions by requiring the node to wait for a time called Backoff time before trying to access the channel. The main known BackOff algorithms are the following:

5.2.1.1 Binary Exponential BackOff (BEB)

It is one of the widely used algorithms for making the selection of the random wait time. When the medium is found busy, a station starts a BackOff timer using a random value within the contention window (CW). Contention window is a parameter that governs the range of random values at each state (collision state). The initial size of the contention window is set to be CWmin=32. This size gets doubled after every successive collision. Thus window size takes values from 32, 64, 128, and so on. For instance, a station encountering collision for the first time will select a random wait time ranging from 0-64. For a second collision for the same access, it will select a value from 0 to 128 and so on. This algorithm is called binary exponential BackOff because the value of CW doubles after every collision. After a certain number of retransmission attempts, packets are dropped. However, after a successful transmission the value of CW resets to the initial value of CW, we can reduce the chance of collision. This is because of the fact that it is quite unlikely for different station to select the same value of BackOff when the CW range is large [22].

<u>Bianchi model for BEB</u>

One of the oldest and important models used to analyze the behavior of BackOff algorithm using different stages is the two dimensional Markov chain model of [24]. A station that has some traffic to send will be in its back off period in three different cases: *initial transmission*, *transmission after collision*, and *transmission after success*.

The model allows calculating the saturation throughput and the probability of transmission failure due to collisions. It assumes an ideal channel conditions and the presence of saturated

traffic at each station. It uses m+1 BackOff stages where each stage represents the BackOff time counter of each node as shown in the Figure 5.1.

It assumes that in each transmission attempt, irrespective of the number of retransmission suffered, each packet collides with an independent and constant probability p. Hence, for each successful transmission, the transition takes places to the lowest stage (stage 0) and for each encountered collision, the transition takes place to a higher stage (say from stage *i* to stage *i*+1). At each stage, W_i represents the maximum window size possible at that stage and its value is equal to $2^i(CWmin+1)$. Thus, for a successful transmission at any stage the random BackOff values will be between 0 and W_o -1, with a probability of $\frac{1-p}{W0}$. These are the cases for stage (0, 0), (0, 1)... to (0, W_0-1).



Figure 5.1 : Two dimensional Markov chain model

In case of collision, the random values will be selected between 0 and W_i -1 with a probability of p/W_i . These are the states for (i, 0), (i, 1)... to (i, W_i-1).

Thus, p is the probability that in one of the slot time, atleast one of the *N-1* stations also starts its transmission. In this case, if the transmission probability of each station is given by r, then p can be expressed as

$$p = 1 - (1 - r)^{N-1}$$
(11)

Using this model, the channel access probability r of nodes is calculated as a function of the number of BackOff stage level m, the minimum contention window *Wmin*, and the collision probability p to be

$$\Gamma = \frac{2}{1 + \text{Wmin} + p\text{Wmin} \sum_{k=0}^{m-1} (2p)^k}$$
(12)

Equation (13) and (14) form two non-linear equation that can be solved to obtain unique solutions for p and r

5.2.1.2 Multiplicative increase and linear decrease (MILD)

According to this algorithm, whenever a station encounters a collision, its CW is increased by a certain multiplicative factor say 2. This algorithm sets the value of CW with an initial value, CWmin, say 5. This gives a value of CW to be 10 for the first encountered collision and 20 for the second successive collision and so on. On the other hand, for the successful transmission the CW is linearly decreased by 1(say) until it reaches the initial CW [22].

5.2.1.3 Double increase double decrease (DIDD)

This algorithm makes use of a 'smooth' decrease of the CW after a successful packet transmission. According to this algorithm, if a station meets collision, it operates exactly as BEB and doubles the CW in order to reduce the probability of collision. However, in the case of a successful packet transmission, it makes the CW half of its initial value (BEB reduces it to CWmin) in order to avoid the future potential packet collisions. In DIDD scheme, packets that

reach their maximum number of retransmission attempts are not discarded, as done in BEB [107].

5.2.2 Related work

The BackOff mechanism used in DCF protocols directly influences the performance of the wireless system. BackOff does not guarantee collisions avoidance at high loaded conditions, especially when the CW size is much less than the contending population size.

Additionally, when packets have large sizes, the throughput gets worse as the number of wasted time due to collision scales with the packet size [108]. This is mostly true for video flows.

To be efficient, traditional BackOff algorithms exponentially increase the BackOff interval to accommodate load. This causes an increase in the stations' waiting times and hence, reduces the throughput per station. In addition, the exponential BackOff approach cannot provide regular access times.

In Time Division Multiple Access (TDMA), the channel is shared by allocating conflict free transmission slots to different stations, such that stations within interference range transmit at different times. Though, TDMA can prevent collisions and provide regular access, it suffers overhead issues. This is because, it needs to maintain detailed topology information for slot assignment and to ensure synchronization [12].

In recent years, approaches based on the combination of both CSMA and TDMA techniques are gaining interest. It is expected that MAC access mechanism of the upcoming new standard IEEE 802.11ad will also include both TDMA and CSMA in its beacon period (see section 2.4.1.1).

In the literature, different solutions related to random access techniques are proposed in order to improve the efficiency of the MAC layer. Out of the many proposed solutions, collision avoidance mechanism using BackOff, collision resolution schemes, and other solutions resulting in reduced overhead can be found (see section 2.5.2.1). In [40], authors propose a scheme in which the contention window size is set based on the utility (satisfaction) that a station gets upon its randomly selected BackOff value on the previous transmission attempt. The scheme provides better performances in terms of throughput and delays, however, it cannot avoid cross collisions.

In [44], authors modify the collision avoidance part of the CSMA/CA mechanism, mainly the contention slot detection distribution over the contention window. An improved collision avoidance based on the prioritization of less collision prone slots and the splitting of the contending pool into smaller groups. However, in [109], it is shown that the system performance depends on mean BackOff value rather than its distribution.

In all of the above mentioned schemes, the basic principle is to adjust the range of the CW to reduce collisions. In [110], it was shown that disabling BEB and optimizing CWmin can improve short term fairness and stabilize the system even when the number of contending users increases. However, this cannot avoid cross collisions between stations at different stages of contention and still adhere to random BackOff allocation leading to irregular slot allocation. Some other solutions for improving the MAC efficiency based on sharing the characteristic of CSMA and TDMA are proposed in [111], [112], [113], and [114]. In [111], authors proposed L-BEB. It is a modified BEB with constant BackOff after successful transmissions, denoted as virtual frame size. The scheme allows station to learn from its success and collision and reduce further chances of collision. Nevertheless, the convergence of their system to collision free operation is susceptible to changing network conditions.

In [112], authors proposed an algorithm that uses the same BackOff value for all stations. Each station estimates the idle slots between two transmission attempts and uses this to compute its contention window. However, in some cases, idle slots may not be a suitable parameter to reflect load. For instance, when multiple simultaneous attempts are made in the same slot there will be collisions, however, other slots may still be idle. However, if the collision window will be decreased based on the number of these idle slots, further collisions can arise. Nevertheless, it is a new approach and is based on a different variable (idle slots) to determine the access unlike some of the well known works based on collisions like [44] and [24] and can prove efficient in some scenarios.

All these schemes are solely based on adjusting the initial contention window. This may take a long time before reaching a collision free state with increasing or variable network load. Authors in [113] utilize the concept of [112] for CW adjustment. They use a hashing based BackOff to come up with orthogonal BackOff values for the different nodes. The hashing BackOff avoids almost all collisions by taking advantage of orthogonality. However, the convergence to a

collision-free state may yet be questionable since the calculation of hashing BackOff is still based on some random behavior. In [114], authors proposed the use of learning to reduce the convergence time. The two proposed approaches (i.e. A-L-MAC and A-L-ZC MAC) show better convergence time. However, A-L-ZC MAC uses additional information to achieve better performance and needs some changes in the original DCF standard. In [70], authors propose a novel Wi-Fi access protocol that is compliant with the traditional access mechanism. It makes use a Tournament Contention Function (TCF) based on binary keys that allows stations to decide whether to transmit or to listen to the channel. Authors proved that this scheme can provide a reduction in collision rates. However, this is only possible at the expense of additional overhead, the use of a secondary channel as well as some additional hardware modifications.

5.3 I-DCF: an Improved Distributed Coordination Function for WLAN

In this section, we propose a new DCF enhancement scheme (I-DCF) that aims to guarantee more regular access. The basic principle of I-DCF operation is similar to that of IEEE 802.11 DCF but combines the features of CSMA and TDMA.

In I-DCF, BackOff values selection is customized for each station according to its context. The scheme also shortens BackOff periods in order to reduce the number of wasted idle slots. I-DCF reduces collision probability and increases success probability. Thereafter, in collision-free state, each node back offs for a constant time leading to a round-robin like functionality. I-DCF targets to achieve performances similar to TDMA schemes.

The proposed solution is composed of two steps:

- Computing unique BackOff-ID
- Adaptive BackOff based on BackOff-ID

5.3.1 BackOff-ID calculation

The station BACKOFF-ID is calculated based on the station ID and the traffic type. We propose to use the last octet of IP address for the station ID. Within a WLAN, this part of the IP address is unique. Most access point allows the use of Dynamic Host Configuration Protocol (DHCP) to obtain an IP address. This protocol supports three different types of IP allocation:

- Automatic allocation: the DHCP server allocates permanent IP address to a client.
- *Dynamic allocation*: the DHCP server assigns IP address to a client for certain duration of time.
- *Manual allocation*: IP addresses are allocated by the network administrator and the DHCP only performs the task to convey those addresses to the client.

For the traffic types, we propose to use the ToS (Type Of Service) field in the IP header. Packets may be prioritized based on the IP precedence bits or DSCP (Differentiated services code point) contained in the ToS byte of their header. Based on these ToS values, a packet would be placed in a prioritized outgoing queue. The 8 user priorities defined in the IEEE 802.1d standard map to 4 ACs (Access Category): Voice, Video, Best Effort, and Background [37].

We propose to combine the station ID and traffic type to obtain a unique value. This can be done using hashing functions. The obtained values represent unique IDs for each station denoted here as BACKOFF-ID.

BACKOFF- ID = f (Station ID, Traffic Type)

5.3.2 BackOff Calculation Algorithm

In the basic DCF, each station needs to select a BackOff value randomly for each transmission. To avoid collisions and provide regular access delays, we propose the I-DCF, which selects BackOff value in a deterministic manner by making use of unique ID denoted as BACKOFF-ID.

In IEEE 802.11, BackOff values are calculated for different situations. Here, we propose a new BackOff calculation strategy for I-DCF.

- *BackOff at initial transmission*: At initial transmissions, the BackOff value is calculated based on the stations BACKOFF-ID.
- *BackOff after collisions*: After a collision occurs, the BackOff value is calculated based on two parameters:
 - the station BACKOFF-ID and
 - the collision stage.

• **BackOff after successful transmissions:** after a successful transmission, the BackOff is determined based on the network load.

The detailed algorithm for I-DCF is presented in Algorithm 1, where:

- *n* : denotes the total number of nodes in the system (network load).
- R: denotes the retransmission limit after which the packets gets dropped (generally R=4)
- *stage* : denotes the collision stage of the station and can take the following values:
 - \circ -1 : when the station attempts to access the channel for the first time
 - \circ 0 : is the stage after a successful transmission
 - $i \in \{1, 2, .., R\}$: after the ith collision
 - \circ *R*+*1* : when the retransmission retry limit is reached

Algorithm 1: I-DCF algorithm

for all stations do get the value of stage if stage is -1 or 5 then Backoff = BACKOFF-ID else if stage is 0 then Backoff = 2* n else Backoff = BACKOFF-ID * stage end if end if end for

5.3.3 Performance Evaluation of I-DCF

Performance evaluations of the proposed solution are conducted with both mathematical analysis and simulations. The proposed solution was implemented in NS2 simulator. The package ns-allionone-2.33 was modified to incorporate I-DCF solution by making the modification to the files associated to the 802.11 mac (mac.cc, mac.h, mac-802_11.cc, mac-802.11.h, mac-timers.cc and mac-timers.h).

5.3.3.1 Mathematical Analysis

In order to carry out this analytical study, we make the following assumptions:

- Ideal channel conditions
- Fixed number of stations
- All stations have saturated traffic conditions such that a packet is available after every successful transmission

Each packet needs to wait for a certain random BackOff time before transmitting and a slot is chosen randomly with probability τ which is also the average attempt rate [109]. Each station selects a BackOff value before transmission and collides with an independent probability p to move to the next contention stage. Though the BackOff ID value is deterministic, the allocation of the IP address is random but unique.

We follow the generalized BackOff-behavior model of [109] for the analysis. The attempt rate for a given packet before for a successful transmission is given by

$$\boldsymbol{\tau} = \frac{\sum_{i=0}^{R} p^{k}}{\sum_{k=0}^{R} b_{k} p^{k}}$$
(13)

Where,

 b_k : is the mean BackOff duration (in slots) for a packet at the kth attempt, b_k depends on the range of the contention window and the probability distribution of backoff duration does not matter as according to [109].

p: is the collision probability

Now, for a system with n stations, the probability p that a transmitted packet encounters a collision is the probability that at least one of n-1 remaining stations transmit at the same time slot. This is given by

$$p = 1 - (1 - \tau)^{n-1}$$
(14)

Equations (15) and (16) define a non-linear system which can be solved numerically to obtain the value of p and τ . These can be further used to estimate the performance of the network.

This approach gives the simplified analysis of the Bianchi approximation model (see Section 5.2.1.1) to come to fixed point equations. It then leads to the calculation of fixed points (p and τ) that is unique and can characterize the operating points of the system. These points are then used to calculate the general throughput of the system as detailed below.

- Throughput estimation

For the throughput estimation, we rely on the equations proposed in [24]. The normalized system throughput S is defined as the portion of time the channel is busy successfully transmitting packets.

We first need to calculate the probability that there is at least one transmission in the considered time slot (P_{tr}) .

$$\boldsymbol{P_{tr}} = \boldsymbol{1} - (\boldsymbol{1} - \boldsymbol{\tau})^n \tag{15}$$

Now, the probability that the transmission is successful is given by the probability that only one station transmits on the channel and this is given by

$$\boldsymbol{P}_{s} = \frac{n\tau(1-\tau)^{n-1}}{P_{tr}} = \frac{n\tau(1-\tau)^{n-1}}{1-(1-\tau)^{n}}$$
(16)

The normalized throughput is then obtained as the following ratio:

$$S = \frac{E[payload information transmitted in a slottime]}{E[length of a slottime]}$$
(17)

Let E[P] be the average packet payload size. Since a transmission occurring in a slot is successful with probability $P_{tr}P_s$. Then, the numerator of the above equation can be expressed as:

$E[payload information transmitted in a slot time] = P_{tr}P_sE[P]$

Now, each time slot can be a successful transmission, a collision or an idle slot. The probability of a slot being empty is given by $I-P_{tr}$ and the probability for it to contain a collision is given by $P_{tr}(I-P_s)$.

Using all the above mentioned equations we can calculate the normalized saturation throughput as

$$\mathbf{S} = \frac{\mathbf{E}[\mathbf{P}]\mathbf{P}_{\mathbf{s}}\mathbf{P}_{\mathbf{tr}}}{(\mathbf{1} - \mathbf{P}_{\mathbf{tr}})\boldsymbol{\sigma} + \mathbf{P}_{\mathbf{tr}}\mathbf{P}_{\mathbf{s}}\mathbf{T}_{\mathbf{s}} + \mathbf{P}_{\mathbf{tr}}(\mathbf{1} - \mathbf{P}_{\mathbf{s}})\mathbf{T}_{\mathbf{c}}}$$
(18)

Where,

 T_s is the time required transmitting the payload successfully, T_c is the average collision time and σ is the empty slot time duration

- Delay estimation

To perform delay estimation, we use the formulations given in [116] where average delay is defined as the time duration between the arrivals of a packet at the head of the MAC queue for transmission until the time of its successful reception at the receiver. The dropped packet due to last retry limit failure is not included in the calculation of the average delay. Thus the average delay is given by

$$E[D] = E[X] \cdot E[slot] \tag{19}$$

Where E[X] gives the average number of slot time needed for successful transmission of a packet and is given by

$$E[X] = \sum_{i=0}^{K} \frac{(p^{i} - p^{K+1}) \cdot \frac{W_{i} + 1}{2}}{1 - p^{K+1}}$$
(20)

and *E*[*slot*] is the average length of a slot time.

$$\boldsymbol{E[slot]} = (1 - \boldsymbol{P}_{tr})\boldsymbol{\sigma} + \boldsymbol{P}_{tr}\boldsymbol{P}_{s}\boldsymbol{T}_{s} + \boldsymbol{P}_{tr}(1 - \boldsymbol{P}_{s})\boldsymbol{T}_{c}$$
(21)

Validation of the mathematical analysis

In order to validate the above model, we run a set of simulations of the proposed algorithm and compare them analytical and with simulation results. Comparison of the normalized saturation throughput of both simulation and analytical model were performed for both DCF scheme and the proposed I-DCF schemes. For this validation, we used data rate of 1 Mbps.

The mathematical model is solved using MATLAB whereas for simulations, we used the NS2 simulator. Table 5.1 lists the parameters used for this validation.

Parameters	Values
Slot Time	20µs
DIFS	50 µs
SIFS	10 µs
MAC Header	224 bits
PHY Header	192 bits
RTS packet	160bits+PHY Header
CTS packet	112 bits+PHY Header
ACK packet	112 bits+ PHY header
Channel data rate	1 Mbps,2 Mbps
Control rate	1 Mbps, 11 Mbps
Retransmission limit	4
Minimum CW	256
Maximum CW	1024

Table 5.1 : List of parameters for mathematical analysis

Validation results are presented on Figure 5.2. As depicted, the proposed analytical model provides a good approximation of the real behavior of the proposed approach. However, some difference can be observed between the mathematical and simulation models for I-DCF. This can be explained by the fact that in the proposed mathematical model, we used mean values for the *CW* range or backoff duration, while for simulations, real values are used.



Figure 5.2 : Normalized Saturation Throughput Comparison with Channel Data rate =1 Mbps

5.3.3.2 Simulation metrics and scenarios

The performances of the proposed scheme are evaluated using the NS2 simulator. We considered network topology consisting of a set of fixed radio stations within the same transmission range. Every station sends Constant Bit Rate (CBR) traffic to one of the other stations. All stations are considered saturated (i.e. there is always packets to send). The reason behind using CBR traffic is to offer more firm constraints on the network so that it sustains the offered load throughout the simulation. This will result in a MAC queue overflow in the long run which may not be possible with VBR traffic in which the bit rate of all traffic do not reach the peak simultaneously.

The BACKOFF-ID is calculated based on Table 5.2:

Table 5.2 : BACKOFF-ID calculation

Station ID	Traffic Priority	BACKOFF-ID
Ι	High	Ι
Ι	Low	n+I

Initial BACKOFF-ID values range between 0 to 2n. This allows avoiding collisions when new stations join the system even at the expense of some unallocated slots. Having idle slots in the system may be preferred to having collisions. For instance, with the new IEEE 802.11n standard, the time slots gets shorten due to high data rate while the packet sizes increase due to aggregation. In this case, the time wasted with idle slots gets scaled down to a much lesser values while longer time may be wasted due to collisions, due to large aggregated packets.

During simulations, the network load is increased by increasing the number of competing stations. This also leads to an increase in the number of traffic flows. Parameter settings used during the simulations are presented in Table 5.3.

Parameters	Values
Simulation Area size	250X250 m2
Simulation Time	600 ms
Node mobility	Static
Bandwidth	11 Mbps
IEEE Standard	802.11b
Control data rate	1 Mbps
Time Slot	20s

 Table 5.3 : Simulation Parameters

SIFS	10s
DIFS	50 s
CBR packet size	1000 bytes
CBR traffic data rate	1 Mbps
Retransmission limit	4

5.3.3.3 Performance results and analysis

The proposed I-DCF mechanism is compared with the basic DCF scheme in terms of throughput, delay, PDR, and number of collisions (see Section 2.5.1).



Figure 5.3 : I-DCF/DCF Throughput vs. Number of active actions

Figure 5.3 shows the throughput comparison of both schemes as the number of active stations increases in the system. With the traditional DCF scheme, the increase in traffic load results in the reduction of the overall system throughput. However, I-DCF manages to provide almost a constant system throughput even with the increasing traffic load. This is as expected due to the adaptive behavior of I-DCF regarding the selection of BackOff values. As the result, in saturated condition, there is about 34 % increase in throughput using I-DCF as compared to DCF.

Figure 5.4 shows the average end-to-end packet delay. Two main results are important. First, I-DCF provides lower delays as compared to the traditional DCF. This can be explained by the fact that I-DCF minimizes collisions and retransmissions and therefore the overall delay. Second,

the delay is less even with a slight increase in the number of idle slots. This shows that it is preferable to keep some idle slots than to take risks to increase collisions.







Figure 5.5 : I-DCF/DCF collisions vs. Number of active stations
Figure 5.5, illustrates that the number of collisions is drastically reduced with I-DCF, even at highly loaded conditions. This is due to the fact that I-DCF prevents the selection of identical contention slot as compared to the random nature of slot selection in DCF. Also for I-DCF, collision is basically seen at the initial phase of simulation when stations start generating the traffic and access the channel. After a certain activity period, the system converges to a collision-free like state, during which the number of collision is almost nullified (as observed during the simulation).



Figure 5.6 : I-DCF/DCF PDR vs. number of active stations

Figure 5.6 presents the packet delivery ratio within the system. This ratio is higher when I-DCF is used. This is a direct result of the reduction in the number of collisions in the system.

Per station throughput and delay is shown in Figure 5.7 and Figure 5.8, respectively. A scenario with 70 stations is considered, where each station sends CBR traffic to one of the other stations. From the figures we observe that, when using I-DCF, each station gets almost equal throughput and less varying delays.









In addition, both the values of delay and throughput are better than those obtained with DCF. This demonstrates the regularity that I-DCF can reach in terms of access delay. This is more visible in Figure 5.9 where the difference between consecutive accesses to the channel by a particular station is shown. It can be noticed that the station maintains a kind of regularity in accessing the channel. In contrast with DCF, we can observe significant variations of the access delays, caused by the random nature of BackOff selection.



Figure 5.9 : I-DCF/DCF access delay of any nodes during a certain simulation time

5.4 An enhanced traffic differentiation scheme

With the proposed I-DCF scheme (see section 5.3), BackOff differentiation is explored in order to improve the channel access scheme. However, I-DCF is effective in saturated traffic conditions and does not provide QoS differentiation for different flow types.

Different applications can have different quality-of- service (QoS) requirements. For instance, the QoS parameters of real time traffic are delay and jitter. On the other hand, throughput and fairness can indicate the QoS for data traffic. QoS provisioning at MAC layer should provide

priority access to real-time traffic in the presence of data traffic. Meanwhile it should also allow certain level of QoS for data. Moreover, MAC scheme must ensure efficient channel utilization [67].

In order to improve fairness and efficient channel utilization among stations, in this section, we propose an AIFS differentiation based solution.

The first standard dedicated to traffic differentiation was introduced in the IEEE 802.11e with the EDCA mechanism (see section 2.3.2). In the literature, some studies have shown that the performance of EDCA drops with increasing density of users. On the one hand, the increase in density of high priority traffic flows can increase collisions in the system, due to the small collision window range set for these traffic. On the other hand, EDCA may completely make the low priority traffic starve for bandwidth, even when the high priority flows are in limited number.

Nevertheless, with the new IEEE standards, like IEEE 802.11n/ac, the available bandwidth and data rates of the system has significantly increased. However, very little work has been done regarding QoS aspects. It becomes necessary to design new efficient mechanisms for QoS guarantee and differentiation while maintaining high resource usage efficiency.

5.4.1 Related work

5.4.1.1 Traffic Differentiation in IEEE 802.11e

In IEEE 802.11e (see section 2.3.2), both the EDCA (Enhanced Distributed Channel Access) and HCCA (HCF Controlled Channel Access) methods are dedicated to provide QoS differentiation to different flow types.

HCCA is a centralized scheme that provides strict deterministic QoS guarantees. Users specify their QoS requirements and are polled by the AP accordingly. On the contrary, EDCA can work in decentralized manner and therefore more adapted to adhoc scenarios. It is based on the IEEE 802.11 DCF function and provides QoS differentiation (among stations) by varying access parameters associated to each station such as CWmin, CWmax TXOP and AIFS.

• CWmin and CWmax

These are used to control the number of medium accesses made by competing classes and STAs. Higher values may be used to decrease the probability of collisions; however, this may increases the latency (i.e. access delay).

• AIFS

This parameter can help the high priority stations to exclusively gain access to the channel. There may not be an easy relationship between different flows requirements and the optimal AIFS values.

• TXOP (Transmission Opportunity)

This parameter can help to allow multiple packets to be transmitted at once, with a single access made to the channel. However, the frequency of access can be determined by adjusting the CW and AIFS parameters.

Manipulating these parameters can provide QoS differentiation. However, AIFS differentiation seems more promising since a single slot difference among AIFS values may result in a substantial difference in terms of performance [118].

5.4.1.2 Other proposals and approaches from the literature

In the literature, different AIFS based differentiation solutions are proposed. In [54], authors propose a method to dynamically change the AIFS parameter for low priority traffic based on EDCA. The main idea is to increase or decrease the AIFS depending on the ratio between the available and desired throughput in the system. However, this may demand the knowledge of the entire system performance level, which may not be easily available and may not be realistic in many cases. In [55], authors propose a new AIFS differentiation solution based on the measurement of the frame loss rate. When the frame loss rate exceeds a predefined threshold, the AIFS value for high priority is decreased and the AIFNS value for low priority traffic is increased. A random selection of AIFN for traffic differentiation is proposed in [56]. Another proposal regarding IFS calculation is proposed in [57], where IFS values are determined for each head-of-line frame based on the priority, the waiting time and random numbers.

5.4.1.3 Proposed Solution

In the basic EDCA, the AIFSN values are fixed for each AC and it is used to calculate the values of AIFS timer. This value is the smallest for the high priority flow and the largest for the low priority ones. Moreover, each type of AC is assigned a respective CW range which is the shortest for high priority flows and the largest for lower ones (see section 2.3.2). However, when the number of high priority traffic increases in the system they will compete within the small CW range and there can be an increase in the collision which can reduce their overall throughput. Moreover, with basic EDCA even when the number of high priority flows is fairly less in the system, none of the low priority traffic can gain access before the high priority ones because of the difference in their AIFS values. This can make them starve for bandwidth. This may not be efficient with system having high data rate and the system resource may be underutilized.

In the proposed scheme, our objective is to first reduce the number of simultaneous channel access attempt by similar traffic. It uses the queue information and delays the access to the channel until a certain threshold is reached unlike in the basic scheme where station make a channel access attempt even with a single packet is in the queue. Moreover, in the proposed scheme, aggregation can act as a counter attack to the delay caused by holding the packets to reach a certain queue threshold to start the contention. This is because all the delayed packets in the queue can be sent at once when the channel access is successful.

Second, we allow the low priority traffic to occasionally share a part of the contention zone of the high priority ones in order to avoid starvation. This can be done only when the service quality of the high priority flows can be maintained within the system. In addition, to maintain traffic differentiation between high and low priority flows, so that they do not completely use the contention zone of the highest priority flows, we allow a minimum of a slot difference between them. Hence, even when both the AC types reach their queue threshold simultaneously, they maintain differentiation with the highest priority flows.

Different values of queue threshold (Qth_i) is assigned to different AC (i = 1,2,3 and 4) where AC[1] is the lowest priority AC. The threshold for each type of AC can depend on the traffic type, traffic rate and the QoS requirement of the each traffic. It can also depend on the number of competing flows in the system. Thus, in the proposed solution, AIFS differentiation is based on queue status in addition to AC types.



Figure 5.10 : Schema showing the principle of proposed solution

To elaborate more on this, let us consider one of the cases shown in Figure 5.10 (say case 1). Three different AC are considered with different level of queue threshold (Qth_i) assigned to each queue. At any instant, depending upon the status of the queue (>Qth_i or < Qth_i), differen AC can take different value of AIFSN (AIFSN x, AIFSN y, AIFSN z) that is used to calculate the AIFS time (AIFS x, AIFS y and AIFS z). This allows the AIFSN values to adapt dynamically. The AIFS time is the amount of time any station waits for the medium to be free for any kind of action (transmission or counting down of the backoff timer).

Furthermore, in the basic EDCA, the relationship between AIFS and AIFSN for each AC is given by the following:

AIFS = SIFS + *AIFSN* * *SlotTime*

The standard does not allow the value of AIFS to be less than DIFS. It uses the values of 20, 10 and 50 μ s for slot time, SIFS and DIFS, respectively. This leads to the value of AIFSN to be greater than 2 so that AIFS is not less than DIFS.

We used the same principle for this proposal and select the AIFSN to be ≥ 2 as the minimum value of AIFSN for any AC at any queue threshold.

Each AC is assigned two values for AIFSN (*AIFSN_i_min* and *AIFSN_i_max*). These values can provide the necessary service differentiation. The value for AIFSN_i_min and AIFSN_i_max are predetermined integers specific to each AC. The algorithm can be described as follows:

```
For each AC i

If (queue_AC_i > Qth<sub>i</sub>)

\begin{cases}
AIFSN_AC_i = AIFSN_i\_min; \\
aifs_AC_i = SIFS+ (AIFSN\_i\_min * SlotTime()) \\
\}
else

<math display="block">\begin{cases}
AIFSN = AIFSN\_i\_max; \\
aifs\_AC\_i = SIFS+ (AIFSN\_i\_max * SlotTime()) \\
\end{cases}
```

5.4.2 Performance Evaluation

Performance evaluations of the proposed solution are conducted through simulations. We first incorporated the proposed solution in NS2 by modifying the 802.11n package [120]. Modification were made to the files mac-802_11n.cc, mac-802_11n.h and aggr.cc to incorporate the changing values of AIFS for different AC. Performance evaluation are then conducted for both the proposed scheme and the basic EDCA scheme. The considered network topology consists of wireless stations within the same transmission range. Two priority flows are considered. The network load is increased by increasing simultaneously the high priority and low priority flows. The different parameters used during the simulations are presented in

The value for the queue threshold for each kind of traffic should be selected carefully. For the high priority traffic, if the threshold is set to a very large value, it may result in long queuing delay that can affect both delays and jitter.

For the low priority traffic, if the threshold is set to a very small value, they may gain frequent access to the channel and occupy the channel for a long time. This may affect the delay of the high priority flows. In this case, we selected one of the pre-determined value obtained through a series of simulation analysis.

Table 5.4 and the associated values used for each priority flow are given in Table 5.5.

The value for the queue threshold for each kind of traffic should be selected carefully. For the high priority traffic, if the threshold is set to a very large value, it may result in long queuing delay that can affect both delays and jitter.

For the low priority traffic, if the threshold is set to a very small value, they may gain frequent access to the channel and occupy the channel for a long time. This may affect the delay of the high priority flows. In this case, we selected one of the pre-determined value obtained through a series of simulation analysis.

Parameters	Values		
Simulation Area	250X250m2		
Simulation time	50 sec		
Node mobility	Static		
Bandwidth	96 Mbps		
MAC layer	IEEE 802.11n		
High priority packet size	1000 bytes		
Low priority packet size	1000 bytes		
Time slot	20µs		
SIFS	10µs		
TXOP limit	3.264 ms		
NS2 Antenna Type	Omni Directional		
NS2 Network Interface Type	Wireless/Physical/MIMO		
NS2 Interface Queue Type	Aggregation queue		
Number of antennas	4		
NS2 MIMO Type	SM-STBC		
Aggregation Size	64000		

Table 5.4 : Simulation I	Parameters
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Table 5.5 : Priority flow parameters

Flow	Threshold	AIFSN min	AIFS max	Data rate
High	20%	2	7	5 Mbps
Low	100%	3	7	5 Mbps

Simulation Results

In this section simulation results are presented. The proposed traffic differentiation scheme is compared with the standard EDCA scheme. Comparisons are conducted in terms of throughput and delay of both high and low priority flows. Two cases are illustrated

Case 1: With many high priority flows in the system (12 high priority flows)

Figure 5.11shows the throughput and Figure 5.12 shows the delay of each high priority flow in the scenario where the total number of high priority flow in the system is 12. In the figures where the total number of flow is less than 12, all the flows belong to high priority.

From Figure 5.11, we can observe that the throughput of each high priority flow is better with the proposed scheme than that with the basic scheme when the total number of flows increases in the system. This is because, in the basic scheme each station attempts to access the channel even when there is a single packet in the queue, however, with the proposed scheme the transmission is delayed and so the number of contention at any instant is reduced. This allows only limited number of similar priority flows to access the channel at a given time. Moreover, the proposed scheme is able to provide traffic differentiation when the number of high priority flow increases in the system. In this case, the lower priority traffic is given least priority for channel access and hence their throughput is almost nullified.



Figure 5.11 : Throughput of each high priority flow vs. total number of flows (12 high priority flows)

The delay of each individual high priority for this case can be seen in Figure 5.12. The delay of high priority traffic is better with the proposed scheme as compared with the basic one. This is because delaying the transmission reduces the possibility of simultaneous contention and also delaying the transmission by few slots is better than to get caught up in collisions and retransmissions. This reduces the retransmission time and leads to an overall improvement in the delay performance.



Figure 5.12 : Delay of each high priority flow vs. total number of flows (12 high priority flows)

Case 1: With less high priority flows in the system (4 high priority flows)

Figure 5.13 and Figure 5.14 show the throughput of individual high and low priority traffic, respectively, and Figure 5.15 shows the delay of individual high priority flow in the scenario where there are a total of 4 video flows in the system. The total number of flows increases with the increase of the low priority flows, however, the high priority flows remains the same in this case.



Figure 5.13 : Throughput of each high priority flow vs. total number of flows (4 high priority flows) From Figure 5.13, it can be seen that the individual throughput of high priority flows is almost the same with both the basic and the proposed scheme as there is not much contention between similar high priority flows in this case. However, there is a gain in the throughput of low priority (see Figure 5.14) traffic with the proposed scheme as they are allowed to occasionally access the channel. This increases the overall efficiency of the system. For the delay, it can be seen from Figure 5.15 that when the total number of high priority flows is less in the system, delaying the transmissions of high priority flows may increase the delay by a small value. Nevertheless, since delay remains lower than 100ms, high priority flows will not suffer from this degradation and some bandwidth will be shared to the low priority. This leads to an improvement in the total efficiency of the system.



Figure 5.14 : Throughput of each low priority flow vs. total number of flows (4 high priority flows)



Figure 5.15 : Delay of each high priority flow vs. total number of flows (4 high priority flows)

5.5 Conclusions

In this chapter, two contributions are presented. We first proposed the I-DCF scheme that aims to reduce collisions and the number of wasted idle slots through adaptive BackOff differentiation. Through simulations, it was verified that I-DCF outperforms DCF and can offer better throughput, less delay, and reduced collisions. Unlike the basic DCF mechanism, I-DCF is able to provide guaranteed access to the channel. The analysis of the proposed BackOff algorithm at steady state shows improvements of the wireless transmission efficiency. However, the performance evaluation of the system in dynamic environment with changing number of stations, require further studies.

In the second contribution, we propose an AIFS differentiation based solution in order to improve the performances of both high and low priority flows in the system. The proposed solution utilizes contextual information (e.g. queue information, traffic type) to adapt AIFSN values. Simulations results show improvements in throughput and delay for both high and low priority traffic. The proposed scheme also improves the overall efficiency of the system without affecting the QoS requirement of the high priority flows. Although, the results presented uses a particular threshold values, the proposed solution was analyzed with different threshold values and the performance was, in most cases, better than the basic scheme. Further studies are required to dynamically tune the threshold values and the minimum and maximum value of AIFSN depending upon different condition (number of flows, traffic rate etc). Moreover, the two proposals can be combined to obtain a system that can offer regular access, minimize collisions, provide traffic differentiation and fairness, simultaneously.

Chapter 6 Conclusions

With the tremendous growth of multimedia services and their wide adoption for both personal and professional use, users are becoming more and more demanding in terms of quality of services support. The same quality of experience is expected over different network access technologies.

Similarly, operators and service providers are also equally interested to provide users with better service options and coverage. Recently operators from the cellular technology are increasingly attracted to invest over WLANs in order to increase their service coverage. In addition, one of the most popular services over internet i.e, video, is used more over Wi-Fi as compared to that over cellular networks.

However, WLAN technologies are still unreliable. The lack of quality of service support and network instability against load variations, are among the main reasons. Although some QoS solutions are proposed, most of these are based on the centralized scheme that are not widely implemented in the terminals and distributed scheme remains fundamental.

QoS aware solutions can be applied at different layers of the OSI stack. When it comes to home networks with single hop connection, MAC plays a crucial role as it directly determines the efficiency with which the data is transmitted. Hence, it is very important to focus on QoS aware MAC solutions that can play a vital role in improving the overall performances of the network. Furthermore, for some services like video, improving network QoS may not be enough to offer better service quality as perceived by users. In such cases, QoE that reflect user's satisfaction should also be considered.

In such scenario, adaptive MAC protocols that can offer QoS/QoE aware wireless services taking into account various application types is needed. It should adapt or tune specific parameters according to local context including traffic patterns, channel conditions and network topology to offer an inherent robust system to accommodate traffic load in a more efficient way.

In this thesis, we mainly focus on solutions that can improve the QoS and QoE over IEEE 802.11 local wireless networks.

In the first contribution, we proposed an adaptive aggregation scheme that takes into account traffic type requirement during aggregation to guarantee adequate QoS for different applications. It varies the maximum aggregation limit for different types of traffic and also adapts to the different channel conditions. Performance results show that there is a significant reduction in overhead and improvement in throughput of the system. The use of Block Ack proved to be a robust method to limit the decreasing throughput and increasing delays in high BER conditions. However, there are multiple parameters to consider that can help to define adaptive MAC solutions.

In the second contribution, we took into account multiple parameters in addition to aggregation. For this we first investigated the impact of different MAC-level parameters on one of the most popular application type: video. We analyzed the QoE (Quality of Experience) of video over IEEE 802.11n wireless networks. The considered MAC-level parameters includes aggregation, BER, number of competing stations, queue length and retransmission limits. Intensive subjective tests were performed to analyze the effect of these parameters on video QoE. Experimental results showed a possible mapping between individual MAC-level parameters and video quality. Furthermore, it was observed that the impact is more pronounced on sport and animation contents than on movie contents, signifying that video service vendors/operator should be more cautious while providing QoS/QoE to different type of videos. From our evaluation we were able to see the effect of different parameters, however since there is no simple relation between these effects we need some prediction techniques based on learning to map the combinational impact of these parameters.

As a second part of this contribution, we then proposed a solution based on Random Neural Network (RNN) for QoE estimation that can predict the QoE in different network and load conditions. RNN was trained with subjective datasets that were obtained from user's evaluation of different video sequences. The experimental results showed that the estimated QoE from RNN correlates with subjective QoE provided by users with reasonable accuracy. This can be used to provide adaptive MAC solutions for video.

From the analysis of the results, we pointed out that an increasing number of competing nodes can severely impact the performance of the video services even in low BER cases. This may be due to

the increase in collision in the system that corrupts the whole efficiency of the system. In order to overcome these limitations, better channel access solutions are needed to provide regular access for time sensitive applications and to reduce collisions in the system.

Hence, in the third contribution, we focused on issues related to channel access and proposed I-DCF that can provide an efficient adaptive backoff algorithm to avoid collisions. Simple rules were used to assign unique backoff values to the competing stations, to avoid the selection of identical contention slots to reduce collisions. The proposed backoff algorithm also ensures that the backoff value range is adapted with the increasing network load. This helps to reduce collisions and the number of wasted idle slots even in highly loaded conditions. Through simulations we verified that I-DCF outperforms DCF and can offer better QoS values in terms of increased throughput, less delay and reduced collision rates.

The analysis of the proposed backoff algorithm at steady state shows improvements on the wireless transmission efficiency. However, it may lack fairness when different traffic flows need prioritization in the system. In order to address this point, we then proposed a solution based on AIFS differentiation. The proposed solution takes into account the queue details along with traffic types to assign AIFS value that can allow traffic differentiation and simultaneously improve the efficiency of the whole system. Performance results show that the proposed solution can improve the performance of high priority traffic even when their number increases in the system. Moreover, when the number of high priority flows is low in the system, the bandwidth can be efficiently shared with the low priority flows, so that they are not starved of bandwidth according to the available resources in the system.

Future Work and Open Issues

The work presented in this dissertation has several contributions to provide solutions to improve both the QoS and QoE over local wireless networks. At the same time, it also unlocks some of the issues that need to be further addressed in the future work. In the following, we remark some of these issues that can be taken into account in the future continuation of this work.

The prediction solutions based on RNNs can be compared with other proposals and may also incorporate additional parameters from Network and Application levels. This kind of system can be used in the design on Soft MAC techniques that can adapt parameters according to the changing conditions, in order to provide better QoE to users.

In chapter 5, the proposed solution with I-DCF can also expose some open issues: What can be the best method to invigilate user density and what should be the frequency of this update? What can be the most effective IP addresses allocation technique that can maintain or improve the consistency of the proposed algorithm to reduce collisions?

Regarding the proposed scheme using AIFS differentiation, more than two traffic classes should be incorporated in the proposal instead of two traffic classes (high and low). Selection of an efficient threshold value for each traffic type should also be addressed in future studies. We also consider merging both the backoff differentiation and AIFS differentiation schemes.

Furthermore, we aim to combine these proposed adaptive schemes into one adaptive MAC solution that can provide a feasible operating point in varying load, traffic types and channel conditions to provide better efficiency and traffic differentiations.

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French Summary (Résumé)

1-Introduction

Les réseaux locaux sans fil (WLANs: Wireless Local Area Networks) constituent encore le moyen le plus populaire de connexion à domicile ou au bureau. Ils sont considérés comme une extension naturelle de l'accès haut débit filaire (e.g. ADSL). De plus, la technologie WiFi est de plus en plus présente dans les divers équipements et objets connectés, tels que les téléviseurs, les serveurs vidéo, l'électroménager, ... etc. Des applications médicales pour les patients suivis à domicile reposent également sur l'accès sans fil pour pouvoir communiquer les diagnostics et les mesures des différents capteurs vers l'hôpital.

Même si ces réseaux ont été initialement conçus pour le transfert de données, avec des débits relativement faibles (cas de la norme IEEE 802.11b), il y a eu ces dernières années de fortes évolutions technologiques avec de nouveaux standards et des débits allant jusqu'à plusieurs dizaines de Mb/s voire même plusieurs Gb/s (IEEE 802.11n, ac, ...).

Avec ces récentes évolutions, les réseaux de type WLAN sont maintenant de plus en plus considérés comme extensions naturelles pour des réseaux WAN cellulaire. Ils permettraient non seulement l'extension de la couverture de ces derniers, mais aussi de les décharger d'une partie du trafic à moindre coût (technique dite d'OffLoading).

Dans ce cas, la problématique de la gestion de la Qualité de Service (QoS) ressurgit de nouveau. En effet, malgré les évolutions géantes au niveau de la couche Physique (PHY), les standards IEEE pour les réseaux locaux se basent tous et toujours sur une technique d'accès aléatoire au niveau de la couche lien (LINK) et plus particulièrement au niveau de la sous couche MAC (Medium Access Control).

Cette technique d'accès, connue sous le nom de CSMA/CA (Carrier Sensing Multiple Access With Collision Avoidance), est elle-même une évolution de la fameuse technique ALOHA, basé sur l'accès aléatoire compétitif. Or ces techniques sont connues pour leur manque d'efficacité en cas de forte charge et pour le fait qu'ils ne gèrent et ne garantissent pas la QoS.

La gestion de la QoS sur les réseaux locaux sans fil basés sur ces techniques constitue donc une problématique et un défi majeur pour les prochaines années, surtout si l'on considère la volonté des opérateurs de faire transiter des flux tels que la voix ou la vidéo, ainsi que l'émergence d'application de domotique et d'e-santé, considérée comme temps réel.

La plupart des solutions existantes pour la gestion de la QoS sur les réseaux WiFi (standards, littérature) reposent sur des architecture en mode centralisées (e.g. PCF, HCCA). Or ces modes usage n'est pratiquement pas en usage en situation réelles. C'est l'approche distribuée qui reste la référence sur les réseaux WiFi et ce pour des raisons de flexibilité et d'efficacité.

Dans ce contexte de nouvelles améliorations sont aujourd'hui plus que nécessaires sur l'approche distribuée afin de prendre en compte la QoS et ce tout en restant compatible avec l'existant. C'est ce que nous nous proposons d'explorer dans cette thèse.

Après l'analyse de l'état de l'art, nous commençons notre première contribution avec l'évaluation des techniques les plus récentes utilisées dans le dernier standard IEEE 802.11n. Celles-ci comprennent l'agrégation de trames et l'acquittement par blocs. Grace à cette analyse, une première contribution d'amélioration de la couche MAC est proposée. Elle consiste en un mécanisme d'agrégation adaptatif qui permet une différentiation de la QoS en sélectionnant la taille d'agrégation la plus adaptée pour chaque classe de service. A travers les analyses de performance par simulation, nous montrons que cette proposition permet d'améliorer la QoS pour les différents services mais aussi d'améliorer l'efficacité globale du système.

Ensuite, nous avons considéré l'étude de la Qualité d'Expérience (QoE). Nous l'avons évalué pour le service vidéo pour différentes conditions radio et de charge pour les réseaux IEEE 802.11n.

A partir de là, nous avons proposé un système de prédiction de la QoE utilisant les systèmes de réseaux de neurones aléatoires (Random Neural Networks). Cette solution est ensuite utilisée pour l'analyse de l'impact des différents paramètres MAC sur la QoE pour la vidéo.

Nous proposons ensuite deux améliorations du mécanisme MAC existant qui reste compatible avec les réseaux WiFi actuels. Cette modification permet une différentiation de la QoS entre les différents types de flux. La première amélioration consiste à sélectionner des valeurs appropriées pour le Backoff, dépendants de la QoS exigée par chaque flux. La seconde amélioration permet de renforcer la propriétarisation des flux en agissant sur les valeurs du paramètre MAC AIFSN (Arbitration Inter-Frame Space Number). Les analyses de performances montrent que la solution proposée permet d'améliorer considérablement la QoS, particulièrement en permettant un accès assez régulier aux ressources pour les flux de type voix et vidéo. Mais aussi de minimiser les collisions et d'accroitre l'efficacité de l'usage des ressources radio disponibles.

2-Etat de l'art

Depuis l'avènement du premier standards IEEE 802.11b, différentes améliorations ont été proposées aux niveaux PHY et MAC via des amendements successifs. Nous en citons ici les principales.

a-IEEE 802.11n

Pour atteindre des débits de l'ordre de 100 Mb/s, le standard 802.11n apporte les améliorations suivantes:

• Agrégation:

L'agrégation est une des fonctions clef qui permet d'améliorer l'efficacité de la couche MAC. Elle permet de réduire l'overhead en réduisant les entêtes et les préambules. Elle permet également de réduire le temps moyen inter-trames. Le détail de cette procédure est décrit dans la section 3.2 du manuscrit.

Block Acknowledgement

Ce mécanisme permet de remplacer l'acquittement individuel des trames jusqu'ici utilisé pour les standards IEEE 802.11(b/g). Ce mécanisme est expliqué à la section 3.2.4.

• Reverse Direction

C'est une amélioration qui permet à la station ayant le TXOP de donner une portion de ce temps à ses stations paires.

• Accès au canal

Le mécanisme d'accès est toujours basé sur la technique CSMA/CA. Cependant la norme IEEE 802.11n considère certains mécanismes utilisés dans la norme IEE 802.11e.

L'utilisation de canaux de largeur multiples nécessite en effet certaines modifications du mécanisme d'accès. Les stations doivent en particulier reconnaitre les canaux primaires des canaux secondaires La coexistence possible avec des stations WiFi des standard précédents (b/g) impose le respect de certaines règles. Pour les canaux de 20 MHz, le même mécanisme DCF est utilisé. Par contre, pour des canaux de 40 MHz, de nouvelles règles de transmission sont définies.

b-IEEE 802.11ac

C'est le standard WiFi le plus récent. Il peut utiliser des canaux de 80 MHz et permet entre autre de transmettre différents flux DL en simultané. Il promet un débit allant jusqu'à 1 Gbps. Les principales nouvelles fonctionnalités introduites sont :

• Enhanced Aggregation

Les tailles maximales d'agrégation ont été étendues de 7935 à 11426 octets pour le cas A-MSDU et de 65535 à 1048579 octets pour l'option A-MPDU.

• MU TXOP-sharing

Une des principales évolutions de ce standard est l'introduction de la technique MU-MIMO, entre autres pour l'utilisation du beam-forming et la réutilisation spatiale.

De nouvelles règles sont également introduites pour adapter la méthode d'accès à ces évolutions.

- Pour transmettre une trame sur un canal de 20 MHz, le CCA est exécuté uniquement sur le canal primaire (de 20 MHz)
- Pour transmettre une trame sur un canal de 40 MHz frame, CCA est exécuté sur le canal primaire mais une vérification est nécessaire sur le canal secondaire de 20 MHz
- Pour transmettre une trame sur un canal de 80 MHz, les deux canaux primaire et secondaire de 40 MHz doivent être libres.
- Pour transmettre une trame sur un canal de 160 MHz, les deux canaux primaire et secondaire de 80 MHz doivent être libres.

• Enhanced RTS/CTS

Le standard IEEE 802.11ac introduit une nouvelle amélioration avec la duplication des trames RTS/CTS sur tous les canaux de 20 MHz adjacents, ce qui permet de s'assurer de la disponibilité de plus d'un canal radio et donc de connaitre la bande passante disponible à chaque instant.

Le tableau suivant résume les principales caractéristiques pour les standards IEEE 802.11a/b/g/e/n/ac.

Standard	Mode d'accès au canal	Améliorations	support du RTS/CTS	Fonctions Optionnelles
802.11a/b/g	CSMA/CA avec DCF	N/A	Oui	PCF , CTS to self
802.11e	CSMA/CA avec	TXOP, catégories	Oui	HCCA

	EDCF	d'accès, Block Ack		
802.11n	CSMA/CA	Agrégation, Block	Oui	Reverse
	avec static ou dynamic	Acknowledgement		Direction,
	bandwidth operation			Support du
				RIFS, CTS
				to self
802.11ac	CSMA/CA	Enhanced	Enhanced	
	Avec des opérations	Agrégation, Block	RTS/CTS	Partage
	statiques ou	Acknowledgement	avec	MU-TXOP
	dynamiques		Signalisation	

c-IEEE 802.11 ad

Le standard IEEE 802.11ad est destiné aux transmissions dans la bande des 60GHz. Technologiquement similaire à l'amendement IEEE 802.11ac, ce standard vise cependant des débits allant jusqu'à 7 Gbps. Il propose une nouvelle architecture PBSS (Personal Basic Service Set) permettant à chaque station d'agir en tant que point d'accès. Au niveau de la couche PHY, ce standard combine OFDM et SC (Single Carrier). Au niveau MAC, il introduit une combinaison de CSMA et de TDMA et introduit un nouveau mode d'agrégation (VA-MSDU) pour le support du trafic Vidéo.

d-IEEE 802.11aa

Ce standard apporte des améliorations dans le mécanisme de propriétarisation EDCA. Il introduit au niveau MAC:

- Le Stream Classification Service (SCS) et
- Group Cast with Retries (GCR).

Avec le service SCS, deux nouvelles files sont ajoutées, une pour l'audio (AAC-VO) et l'autre pour la vidéo (AAC-VI). Les paquets peuvent être marqués avec le drop eligibility indicator (DEI). Dans ce cas, ce marquage indique le nombre de retransmission (long or short) pour ce paquet. Ce standard introduit également une nouvelle politique d'acquittement pour les flux multicast audio/vidéo

e-IEEE 802.11 ax

Ce prochain standard est dérivé du IEEE 802.11ac. Il vise à accroître le débit individuel des stations plutôt que le débit global du système. Il va probablement se baser sur la technologie MIMO-OFDA (orthogonal fréquence division accès), avec la possibilité pour chaque station de gérer quatre stream radio. OFDA est une variante de la technique OFDM, qui pourrait multiplier la bande passante par 10.

f- Principales propositions concernant la gestion de la QoS dans la littérature

Dans la littérature, il existe plusieurs propositions pour la gestion de la QoS au niveau MAC pour les réseaux WiFi. Outre les techniques de réservations, les principales approches proposées se basent sur la différentiation des paramètres de Backoff ou d'agrégation. Certaines techniques proposent des mécanismes de Jamming permettant à certaines station de monopoliser le medium au dépend des autres stations. Une synthèse des principales techniques est donnée sur le tableau suivant :

Ref	Année	Туре	Type de	Description
			Trafic	
[40]	2009	B.D	NA	Basé sur le degré de satisfaction Durant les
				transmissions précédentes
[41]	2008	B.D	Uplink	Permet une équité UL et DL mais nécessite des
			downlink	fonctions supplémentaires sur l'AP
[29]	2005	B.D	2	Evaluation des performances en fonction du
				Backoff et du nombre maximum de retransmissions
[42]	2009	BD	>1	Les stations changent leurs PR en fonction du taux
				de perte de trames
[43]	2012	BD	4	Adapte les paramètres de CW en utilisant la logique
				floue
[44],	2012	BD	2	Minimise les collisions en sélectionnant les slots
				susceptibles de subir moins de collisions
[45]	2008	BD and	4	Les BackOff ne se superposent pas entre les flux
		FA		prioritaires
[46]	2011		3	Les CW sont différentes pour chaque flux
[47]	2006	BD and	4	Combinaison de EDCA avec un mécanisme de
		FR		réservation

 Tableau 1: Synthèse des principales propositions de gestion de QoS pour les norms IEEE 802.11

Ref	Année	Туре	Type de Trafic	Description
[49]	2006	R	2	Réserve des slots successifs pour les flux temps réel
[[]	2000		-	et utilise le mécanisme DCF pour les autres flux.
[50]	2012	R	NA	Utilise une réservation tacite .
[48]	2013	R	2	Utilise les bits réservés pour des réservations de
				trafic de type Voix.
[49]	2009	R	>2	Permet une allocation distribuée des time-slots pour
				le trafic temps réel.
[51]	2008	R	NA	Utilise un nouveau message de contrôle pour
				réserver le canal pour le paquet suivant en attente
				dans le buffer durant la transmission en cours.
[52]	2008	R		Permet aux stations de diffuser leurs valeurs de
				BackOff
[53]	2010	R	2	Utilise un cycle de fixe de BackOff pendant lequel
				un certain nombre de slots peuvent être réservés
5.5.43				pour les stations en compétition.
[54]	2006	IFS	3	Change dynamiquement la valeur de AIFS pour le
				trafic basse priorité, base sur EDCA, afin d'éviter
5553	2012	IEC	2	un desequilibre de bande passante entre utilisateurs.
[22]	2012	IFS	2	La valeur de AIFS est modifiée en fonction du taux
				de perte de trames. S'il depasse un certain seuil,
				I AIFS des flux prioritaires est diminue et celui des
[5(]	2007	IEC	2	Tiux moins prioritaires est augmente.
[30]	2007	162	2	Propose une differentiation basee sur la selection
[57]	2002	IEC	2	Le valeur de IES est esteviés neur cheque neuveeu
[37]	2002	115	2	La valeur de IFS est calculée pour chaque nouveau flux de trames, en considérant on niveau de priorité
				temps d'attente ainsi qu'un nombre généré
				aléatoirement
[58]	2010	IFS	4	Un AIFSN unique pour chaque canal pour
[50]	2010	11.5		minimiser les collisions
[59]	2008	FA	NA	Propose plusieurs approaches d'agrégation
[60]	2005	FA	NA	Analyse les mécanismes d'agrégation existant
[61]	2010	FA	NA	Propose d'adapter la modulation et du codage au
				lieu de la taille des trames
[62]	2007	FA	NA	Propose des modifications des paramètres PHY
				pour contrôler les tailles des MPDU
[63]	2010	FA	NA	Propose de switcher entre deux méthodes
				d'agrégation en fonction du BER.
[65]	2009	FA	4	Agrégation avec possibilité de retransmettre les
				fragments de trames
[66]	2006	J	.NA	Utilise des "pulsations" de longueur variables afin
				de notifier par dichotomie sa volonté de transmettre
[67]	2008	J	2	Propose un nouveau protocole basé sur la notion de
				"busy tone" sur deux canaux étroits et un canal de
				signalisation.
[68]	1999	J	2	Les flux temps réel entre en compétition avec des
F 667	100-	-	_	pulsations "BB"
[69]	1996	J	2	
Ref	Année	Туре	Type de Trafic	Description
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[70]	2012	J	N/A	Propose un mécanisme de tournoi entre les stations ayant des trames à transmettre.

B.D=Backoff differentiation, FA-frame aggregation, J- Jamming, IFS-interframe differentiation, R- Reservation, N/A- Not applicable

3- Différentiation de QoS sur les réseaux IEEE802.11n par un mécanisme d'agrégation adaptatif

Nous proposons dans ce chapitre une nouvelle approche d'agrégation qui considère les contraintes QoS de chaque type de flux. Les techniques d'agrégation classiques ne tiennent pas compte de l'état du canal, et particulièrement du taux d'erreurs binaires. Ce qui a pour effet de dégrader encore plus la qualité des transmissions à cause des retransmissions de trames agrégées. Nous proposons que la taille d'agrégation soit dépendante de plusieurs paramètres et indicateurs provenant aussi bien des caractéristiques du flux que des conditions du canal radio. Nous proposons également l'utilisation combinée de cette proposition avec le mécanisme d'acquittement par bloc (BlockAck).

Les résultats de performance montrent que nous pouvons réduire de façon sensible les niveaux d'overhead mais aussi d'augmenter le débit effectif et de diminuer les délais (c.f. Figures 1 et 2).









La solution proposée permet aussi une différentiation de la QoS de manière distribuée, sans avoir nécessairement besoin des mécanismes centralisés proposés par l'amendement IEEE 802.11.e relatifs à la gestion de la QoS.

4- Estimation de la QoE pour le service Vidéo, considération de la couche MAC

Nous avons ici analysé l'effet d'un ensemble de paramètres et d'indicateurs MAC sur aussi bien la qualité de service (QoS) que de la qualité d'expérience (QoE) pour les applications vidéo. La qualité d'expérience est une mesure subjective qui peut refléter de manière plus fidèle le ressenti des utilisateurs.

Cette étude a été réalisée pour différents types de vidéo, échangés sur des réseaux de type IEEE 802.11n. Nous avons en réalité considéré l'environnement domestique où des routeurs WiFi sont de plus en plus utilisés pour inter connecter l'ensemble des équipements connectés tel que la télé, les serveurs multimédia et les divers autres objets connectés (Smart Phones, PDA, Laptop, ...etc.).

L'environnement pour mesurer la Que est décrit sur la Figure 3.



Figure 3: Environnent de mesure de la QoE

Les résultants soulignent les dépendances qui existent entre le paramétrage et les indicateur MAC et la Que vidéo. De plus, l'impact de la couche MAC semble plus prononcé pour certains

types de vidéo, notamment quand il s'agit de sport. Nous avons ainsi analysé l'impact de différents paramètres MAC sur la QoE sur des réseaux de type IEEE 802.11n.

Les paramètres considérés sont essentiellement le BER, le nombre de stations en compétition sur le réseau, la taille des files d'attente, et la limite maximale du nombre de retransmissions.

Les résultats (Figures 4 à 6) confirment qu'il existe une relation directe entre les conditions MAC et la QoE.

Un paramétrage adéquat au niveau de cette couche semble donc nécessaire.

Or la compréhension de l'effet combiné de l'ensemble des paramètres et indicateurs MAC n'est pas triviale.



Figure 4: Impact du BER sur Que



Figure 5: Impact du nombre de stations sur la QoE





Nous avons alors mené dans un second lieu, une analyse combinant des tests subjectifs par un panel d'utilisateurs, alimentant une solution d'apprentissage et de prédiction par les réseaux de neurones aléatoires (RNN) (c.f. Figure 5). Une fois le RNN optimal déterminé et entrainé, nous l'avons utilisé pour prédire la QoE, avec des séquences de test déjà apprises, mais aussi et surtout avec des séquences de test inédites.



Figure 7: Module d'estimation de la QoE

Le réseau de neurone ainsi entrainé, nous permet de prédire la QoE pour différentes conditions MAC. Un exemple des ces résultats est donné sur les Figures 8 à 10.



Figure 8 : Variation de la QoE en fonction la charge et du BER



Figure 9: Variation de la QoE en fonction du BER et de la taille max d'agrégation (5 stations)



Figure 10: Variation de la QoE en fonction du BER et de la taille max d'agrégation (20 stations) Les résultats obtenus montrent que l'approche proposée permet de prédire convenablement la QoE perçue par l'utilisateur en fonction des conditions MAC.

5- Un mécanisme d'accès à différentiation de QoS pour les réseaux de la famille IEEE 802.11

a-I-DCF : Amélioration de la fonction de coordination distribuée pour les WLANs

Nous proposons ici la solution I-DCF qui vise à réduire le nombre de collisions et le nombre de slots inutilisés grâce à un mécanisme de BackOff adaptatif. Les résultats de performances montrent que la solution proposée est meilleure en termes de performances comparée au mécanisme standard. Elle offre un meilleur débit, moins de délais et de collisions. Elle permet surtout de garantir les délais d'accès.



Figure 11 : Modèle avec chaines de Markov à deux dimensions



Figure 12 : I-DCF/DCF debit vs. nombre de stations actives



Figure 13: I-DCF/DCF délai vs. nombre de stations actives



Figure 14 : I-DCF/DCF collisions vs. nombre de stations actives

b-Une nouvelle solution pour la différentiation de la QoS

Dans cette contribution, nous proposons une solution de différentiation basée sur le paramètre AIFSN permettant d'améliorer les performances MAC aussi bien pour les flux de haute priorité que de basse priorité.

La solution proposée utilise des informations contextuelles telles que le type de trafic, l'état des tampons mémoires afin de sélectionner la valeur de AIFSN la plus adaptée. Les résultats de performances montrent une bonne amélioration des indicateurs de QoS pour tous les types de trafic considérés. Nous remarquons également une amélioration des performances et de l'efficacité globale du système, sans affecter négativement les flux individuels.

6-Conclusions

Dans cette thèse, nous nous sommes intéressées à l'amélioration de la QoS et de la QoE sur les réseaux locaux sans fil de la famille IEEE 802.11.

Nous avons dans un premier temps proposé une solution d'agrégation qui tient compte des caractéristiques de chaque flux et de ses exigences en termes de QoS. Les résultats de performance montrent une nette amélioration des indicateurs de QoS.

La solution proposée a aussi pour avantage d'introduire intrinsèquement la notion de différenciation de QoS. L'utilisation conjointe du mécanisme d'acquittement par bloc (Block Ack) permet de renforcer la robustesse de notre mécanisme.

Nous avons également proposé une solution de prédiction basée sur l'apprentissage par les réseaux de neurones aléatoires. Pour construire cette méthode, la compréhension de l'impact des différents paramètres et indicateurs MAC était nécessaire. Un panel d'utilisateurs a permis de recueillir et d'utiliser des notations de QoE pour des flux vidéo transmis sur un réseau de type IEEE 802.11n.

Après la phase d'apprentissage, le réseau de neurone proposé a été validé en comparant ses prédictions aux notations de QoE donnés par le panel d'utilisateurs. Il a ensuite permis de prédire des valeurs de QoE pour des variations aléatoires de l'ensemble des paramètres.

Nous avons ensuite proposé la solution I-DCF, qui permet d'adapter la valeur du BackOff de chaque station en fonction des caractéristiques et exigences des flux manipulés.

La solution proposée permet de réduire le nombre de collision, d'augmenter les débits et de garantir des délais d'accès assez similaires aux systèmes de type TDMA.

Dans notre dernière contribution, nous avons proposé une solution de différentiation basé sur la sélection de valeurs adaptées au paramètre AIFS.

La solution proposée considère l'état des mémoires tampon (trafic en attente de transmission) ainsi que le type de trafic en question. Elle attribue des valeurs de AIFS qui permettent d'avantager les flux les plus prioritaires ou ayant attendu le plus longtemps. Les analyses de performances ont démontré que la solution proposée permet d'atteindre les objectifs de QoS même dans des conditions de charge importantes. En attribuant plus rapidement l'accès aux flux de type temps réel, la solution proposée permet malgré tout de garantir un accès minimum aux flux les moins prioritaires.

7-Perspectives

Dans cette thèse, nous avons présenté des contributions pour améliorer et la qualité de service et la qualité d'expérience sur les réseaux locaux sans fil. Plusieurs aspects méritent encore d'être approfondis, principalement concernant la robustesse des solutions proposées. Par exemple, les solutions de prévision basées sur les RNNs peuvent intégrer des paramètres supplémentaires pour prédire peut être de façon plus précise la QoE. Ceux-ci peuvent inclure des paramètres des niveaux réseau et application. Ce type de système peut être utilisé dans la conception de techniques MAC adaptatives en fonction des conditions radio et réseau changeantes, afin de fournir la meilleure QoE pour les utilisateurs en fonction du contexte.

En ce qui concerne la différenciation QoS basée sur le paramètre AIFS, nous avions considéré dans nos études que deux classes de trafic. Le choix du seuil approprié pour chaque type de trafic mérite d'être adressé dans les études futures.

Nous considérons également de combiner les solutions proposées dans le cadre d'une solution unique, dans le cadre des réseaux logiciel et plus particulièrement concernant la couche MAC (Software defined MAC).

List of Publications

- Indira Paudel and Badii Jouaber."I-DCF: Improved DCF for channel access in IEEE 802.11 wireless networks ", to be published in IEEE 79th. Vehicular Technology Conference (VTC Spring), 2014
- Jeevan Pokhrel, Indira Paudel, Bachar Wehbi, Ana Cavalli, and Badii Jouaber. "Performance evaluation of video transmission over 802.11 n wireless networks: A MAC layer perspective." In International Conference on Smart Communications in Network Technologies (SaCoNeT), 2014
- 3. Indira Paudel, Jeevan Pokhrel, Bachar Wehbi, Ana Cavalli, and Badii Jouaber. "Estimation of video QoE from MAC parameters in wireless network: A Random Neural Network approach" to be published In IEEE 14th International Symposium on Communications and Information Technologies (ISCIT), 2014
- 4. Indira Paudel, Fatma Outay, and Badii Jouaber. "QoS-HAN: QoS provisioning in Home Automated Networks over IEEE 802.11n." In IEEE Symposium on Computers and Communications (ISCC), 2012