

الآية

DEDICATION

TO

My mother and father

My sisters and brothers

My friends

My respectable teachers

My supervisor

To all who supported this thesis

Thanks ,,

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Table of contents

الآية.....	i
Dedication.....	ii
Acknowledgements.....	iii
Table of contents.....	v
List of Tables.....	viii
List of figures	ix
Abbreviations.....	xii
Abstract.....	xiii
المستخلص.....	Xiv
Chapter One: Introduction	
1.1Background.....	1
1.2 Problem Statement.....	2
1.3 Proposed Solution	2
1.4 Objectives	2
1.5 Methodology.....	2
1.6 Project Scope.....	3
1.7 Thesis Outline.....	3
1.8 summary	4
Chapter Two: Literature Review	
2.1 Introduction.....	5
2.2 Voice over IP (VoIP).....	5
2.2.1 VoIP Protocols.....	6
2.2.2 VoIP over Transmission Protocol.....	7
2.2.3 Over network layer	9

2.2.4 VoIP Network Components.....	15
2.2.5 VoIP Codecs	16
2.2.6 VoIP Challenges	18
2.3 Background of Wireless LAN.....	19
2.3.1 Ad Hoc mode.....	19
2.3.2 Infrastructure mode.....	19
2.4 Benefits of Wireless LAN.....	20
2.4.1 Mobility.....	20
2.4.2 Cost Stability.....	20
2.4.3 Easy to Install.....	20
2.5 Deficiency in Wireless LAN.....	21
2.5.1 Security.....	21
2.5.2 Interference.....	22
2.6 Architecture of Wireless LAN.....	22
2.6.1 Station.....	22
2.6.2 Basic Service Set.....	22
2.7 IEEE 802.11 Standards.....	22
2.7.1 IEEE 802.11a....	23
2.7.2 IEEE 802.11b	23
2.7.3 IEEE 802.11g.....	23
2.7.4 IEEE 802.11n.....	24
2.8 Quality of Service (QoS)	25
2.9 VoIP QoS Factor.....	26
2.9.1 Throughput.....	26
2.9.2 End To End Delay.....	27

2.9.3 Jitter	28
2.10 Related Work	29
2.11 Summary	30

Chapter Three: Methodology

3.1 Introduction	31
3.2 Simulation Scenario.....	32
3.2.1Network Designed Component.....	32
3.2.2 OPNET Implementation	35
3.2.3 Simulation to measure QoS.....	36
3.2.4 Simulation Parameters	41
3.3 Summary.....	43

Chapter Four: Results and Discussion

4.1 Introduction.....	44
4.2 Throughput	44
4.2.1 Throughput In Small Network.....	44
4.2.2 Throughput In large Network.....	46
4.3 End-To-End delay.....	48
4.3.1 End-To-End Delay In Small Network	48
4.3.2 End-To-End Delay In Large Network.....	50
4.4 Jitter.....	52
4.4.1 Jitter In Small Network.....	52
4.4.2 Jitter In Large Network.....	54
4.5 Summary.....	56

Chapter Five: Conclusions and Recommendations

5.1 Conclusions	58
5.2 Recommendation	58
References.....	59

List of Tables

Table No	Title	Page
(2.1)	bandwidth requirements of some common codec's	17
(2.2)	explain the main feature of each IEEE802.11 release	25
(3.1)	Simulation Environments	42
(4.1)	Summarization 802.11 a IPv4, 6	56
(4.2)	Summarization 802.11 a IPv4, 6	56
(4.3)	Summarization 802.11 n IPv4, 6	57
(4.4)	Summarization 802.11 n IPv4, 6	57

List of figures

Figure	Title	Page
(2.1)	UDP header	8
(2.2)	IPV4 header	9
(2.3)	IPV6 header	13
(2.4)	VoIP protocols over TCP/IP stack	14
(2.5)	VoWiFi component	15
(3.1)	IPV4 Network showing VoWiFi small network	32
(3.2)	IPV4 Network showing VoWiFi large network	33
(3.3)	IPV6 Network showing VoWiFi small network	34
(3.4)	IPV6 Network showing VoWiFi large network	34
(3.5)	Profile_Config attribute dialogue box	35
(3.6)	VoIP application configuration	36
(3.7)	flowchart Throughput vs. simulation time	38
(3.8)	flowchart End To End Delay	39
(3.9)	Flow chart jitter	40
(3.10)	Example for Wireless network parameters	41
(3.11)	Codec and transmission protocol configuration	42
(4.1)	Throughput_a_ipv4, 6	45
(4.2)	Throughput_n_ipv4, 6	46
(4.3)	Throughput_a_ipv4 , 6	47
(4.4)	Throughput_n_ipv4, 6	48
(4.5)	Delay_a_ipv4, 6	49
(4.6)	Delay_n_ipv4, 6	50
(4.7)	delay_a_ipv4, 6	51

(4.8)	delay_n_ipv4, 6	52
(4.9)	Jitter_a_ipv4, 6	53
(4.10)	Jitter_n_ipv4, 6	54
(4.11)	Jitter_a_ipv4, 6	55
(4.12)	Jitter_n_ipv4, 6	56

Abbreviations

AP	Access Point
BER	Bit Error Rate
BSS	Basic Service Set
BSS	Basic Service Set
CGI	Common Gateway Interface
Codecs	Coders/Decoders
ECN	Explicit Congestion Notification
ESS	Extended Service Set
FCC	Federal Communications Commission
GK	GateKeepers
GW	GateWay
IBSS	Independent Basic Service Set
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
IP	Internet Protocol
IPv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
ITU	International Tele- communication Union
MAC	Media Access Control
MCU	Multi point Control Units
NAT	Network Address Translation
NIC	Network Interface Card
OPNET	Optimized Network Engineering Tools
PCM	Pulse code modulation
PSTN	Public Switching Telephone Network

QoS	Quality of Service
RF	Radio Frequency
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol
RTP	Real-time Transport Protocol
SIP	Session Initiation Protocol
SOHO	Small Office Home Office
SSID	Service Set Identification
TKIP	Temporal Key Integrity Protocol
UDP	User Datagram Protocol
VAD	Voice Activity Detection
VoIP	Voice over Internet Protocol
VoWiFi	Voice over Wireless Fidelity
WEP	Wired Equivalent Privacy
Wi-Fi	Wireless Fidelity
WPA	Wi-Fi Protected Access

Abstract

In recent years, Voice over IP (VoIP) has gained a lot of popularity and become an industry favorite over Public Switching Telephone Networks (PSTN) with regards to voice communication. The performance of VOIP over Wi-Fi (**VoWIFI**) is sensitive to throughput, jitter and end to end delay .so, it is very important to analyze the performance of **VoWIFI** stander such as IEEE 802.11 a, n. This Research mainly aimed to study, analyze and simulate VoIP over WLAN, to explore the quality of service of VoIP over WLAN in term of (jitter, delay, and throughput). it has been concluded that WLAN 802.11a ipv4 Standards is considered the best choice for VoIP when our concern in throughput, ipv6 have better performance in delay than ipv4 and jitter. Also concluded that the ipv6 is better for real time applications when using 802.11 n more throughputs and the delay more stable, smaller jitter.

المستخلص

في السنين الحالية كسب تطبيق نقل الصوت عبر بروتوكول الانترنت انتشاراً واسعاً واصبح المفضل في الصناعة من شبكات تبديل الهواتف العامة مع الاخذ في الاعتبار إتصالات الصوتيه . اداء نقل الصوت عبر بروتوكول الانترنت في الشبكات اللاسلكيه حساس جدا لذا من المهم جدا دراسه عوامل كفاءة الخدمة : كمية البيانات المستقبلة بنجاح ، الإختلاف في زمن وصول حزم البيانات و زمن التأخير لمعيار المنقول بواسطه بروتوكول الانترنت الاصدار الرابع و السادس . يهدف هذا البحث الى دراسه تحليل ومحاکاه نقل الصوت عبر بروتوكول الانترنت في الشبكات اللاسلكيه وفقا لعوامل كفاءة الخدمة الآتيه : كمية البيانات المستقبلة بنجاح ، الإختلاف في زمن وصول حزم البيانات و زمن التأخير. بعد الحصول على النتائج و منقشتها وجد ان المرسل عبر بروتوكول الانترنت الاصدار الرابع هو افضل في نقل الصوت حيث يعطي انتاجيه اكثر . وعندما يرسل عبر بروتوكول الانترنت الاصدار السادس يكون احسن اداء في زمن التأخير . وايضا وجد ان بروتوكول الانترنت الاصدار السادس افضل في نقل تطبيقات الزمن الحقيقي عند استخدام حيث توجد انتاجيه اكثر ومستوى تأخير صغير والاختلاف في زمن وصول البيانات .