

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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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 دراسته عوامل كفاءة الخدمة : كميته البيانات المستقبلية بنجاح ، الإختلاف في زمن وصول حزم البيانات و زمن التأخير لمعيار المنقول بواسطه بروتوكول الانترنت الاصدار الرابع و السادس . يهدف هذا البحث الى دراسته تحليل ومحاكاة نقل الصوت عبر بروتوكول الانترنت في الشبكات اللاسلكيه وفقا لعوامل كفاءة الخدمة الاتيه : كمية البيانات المستقبلية بنجاح ، الإختلاف في زمن وصول حزم البيانات و زمن التأخير.بعد الحصول على النتائج ومنقشتها 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 تاخير صغير والاختلاف في زمن وصول البيانات .