Quality of Service Evaluation of VoIP over Wireless Networks

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Abstract- **The Voice over IP (VoIP) service demands high priority over other services and applications. Some constrains are associated with this real-time service, such as delay and throughput which need to be addressed before delivering to the customer. The mobility in IP networks is a demand that facilitates IP applications and services, especially in wireless networks. This paper demonstrates the performance of Voice over IP (VoIP) in 802.11 wireless networks and elaborates on the evaluation of voice packet end-to-end delay and throughput. Employing literature reviews and an experimental model created on OPNET that is simulated to assess the quality of service (QoS) of VoIP in 802.11g legacy and 802.11e wireless network; shows the enhancement of 802.11 reflects as enhancement in the quality of the VoIP service. The simulation results have indicated that the quality of VoIP service is influenced by the quality of the carrier which is IEEE 802.11 network. Therefore, the voice service over wireless network can be improved significantly by developing a quality of service policy that prioritizes the packet transmission based on the controlled access mechanisms. Eventually, the number of VoIP calls could be increased using the enhanced 802.11e standard rather than 802.11 standard.**

Keywords: IEEE 802.11; IEEE 802.11e; QoS; VoIP; Wireless Network

I. INTRODUCTION

Voice over Internet Protocol (VoIP) transforms analog audio signals, such as a phone call into digital data that is then transferred over any computer network based on IP such as Internet.

VoIP technology turns analog voice into digital data packets that can be stored, searched, manipulated, copied, combined with other data, and distributed to virtually any device that connects to the IP network. This capability virtually made it possible to achieve maximum flexibility in the transport or transmission of voice that has been transformed into data. Making use of the internet infrastructure or the World Wide Web in transmitting digitized voice the same way data packets are transmitted across great distances have virtually made VoIP the answer to the availability of infrastructure. Nowadays, massive movement of voice communication from legacy PSTN networks to IP networks. Some main factors allow PSTN to maintain the Quality of voice service such as voice compression, delay, echo, compliance with American standard TR-57 and others.

Convergence in the use and application of technology is inevitable because infrastructure and its raw materials are simply expensive, not only in terms of installation cost but also in terms of its maintenance. The onus therefore for telecommunication service provider companies is to maximize its usage thereby maximizing the revenue that can be derived from a single infrastructure. Convergence spawned Voice over IP and the demand for information about the new technology has never been the same. The IP networks' convergence aggregates all data services in one backbone, these services are data, voice and video. The quality of these services does not face any constrains in high bandwidth networks. On other hand, limited bandwidth networks such as wireless networks; are designed in a manner to handle priority services such as real-time voice.

The increase in the use of real-time applications has increased the demand for networks Quality of service (QoS). QoS is a measure of how well a network provides consistent network data delivery [1]. It can be measured in terms of packet loss, delay, jitter, and data rate. For the mobile users to be connected to the network or the Internet, a Wireless Local Area Network (WLAN) is needed. WLAN is a network that uses the air as the medium of information transmission rather than copper or fiber optic cables used in wired networks [2].

The rest of the paper is organised as follows. Section II is a literature review and related work on Voice over IP technology, and wireless standard IEEE 802.11 currently in use. Section III describes and explains in detail the network design and implementation model. Section IV presents the summary of the research results and analysis. Finally, the conclusions of this work with future directions are discussed in Section V.

II. RELATED WORK

A comparative study is presented in [3] which show a new technique called Modified Dual Queue (MDQ) scheme. This scheme was implemented in parallel with enhanced distributed channel access (EDCA) which is used as quality of service mechanism in 802.11e. The flexible parameter of channel access control in EDCA has the superiority over the MDQ.

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As a recommendation, the distance between the VoIP client and the access point (AP) should be 65 meter according to the study [4] which is done using different standards 802.11 b/g/n.

A real testbed model is conducted as shown in [5] which follow best practices from Cisco on 802.11n standard. As the client goes further away from the AP, the quality of the VoIP call can be affected significantly.

Network coding technique is used to enhance the throughput of the wireless network as illustrated in [6]. The COPE procedure is used in highly congested wireless network with a VoIP codec that requires higher traffic loads per second. The results show that network coding can enhance the VoIP performance in wireless mesh networks (WMN).

Packet End-to-End Delay, Jitter, Load, Throughput and codec are characteristics and factors that play a role in enhancing the quality of voice service in any network [7].

Delay: The ITU-T G.114 reports that the one-way transmission time and delay in echo-free network which is acceptable is less than 150 msec [8]. The three main components are highlighted in [9] that cause the End-to-End Delay of VoIP packets. First, the network status and congestion, which infers the ability to send signals and avoid collisions. Second, the encoding used by the sender and decoding used by the receiver, which mainly depends on the codec used by the voice system. Third, the packetization (i.e. compression of the packets) of the voice frames within the IP packet. Thus, most successful voice services have less delay time; with the consequence that more VoIP calls are allowed on the network.

Jitter: The delay variation between two packets (sent and received) is known as jitter. Jitter constrains the VoIP delivery to arrive at a constant rate and free of echo. A Jitter buffer is used to overcome the delay variation on a VoIP handset. The admitted jitter of successful VoIP calls is between 0 msec and 50 msec [10].

Throughput: [11] Defines the throughput as total received packets measured in bits per seconds. The packet loss and the throughput are inversely proportional; a robust network has a lower degree of packet loss.

A. *Audio Codec*

Voice over IP performance is based partially on the audio codec which is used. The audio codecs are divided into three types which are; Parametric codecs, Waveform codecs and Hybrid codecs [12]. ITU-T Recommends [13] a G.711 codec which is a non-uniform Pulse Code Modulation (PCM). The two main algorithms used in G.711 are; A-low and μ-low. The voice quality and performance are similar in both algorithms due to low distortion comparison. G.711 codecs do not support compression, which makes it the most popular in PSTN rather than IP networks. The G.729 codec uses Conjugate-Structure Algebraic-Code-Excited Linear-Prediction (CS-ACELP) [14]. It provides voice calls at low bit rate (i.e. 8 Kbit/s) according to ITU-T Recommendation. The G.729 consumes a high number of CPU resources which limits the number of VoIP calls.

B. *IEEE 802.11 ENHANCEMENT FOR VoIP QOS*

The IEEE 802.11 MAC protocol supports two access mechanisms, which are; Distributed Coordination Function (DCF) and Point Coordination Function (PCF). PCF is optional whereas DCF is mandatory based on CSMA/CA. The limitation of IEEE 802.11 is that DCF mechanism does not guarantee any type of quality of service. Instead, DCF provides best effort service with FIFO transmission technique. This limitation implies poor VoIP service which is a delay sensitive service. The newer standard IEEE 802.11e overcomes these limitations in wireless network. The major enhancement is the Hybrid Coordination Function (HCF) that consists of Enhanced Distributed Coordination Function (EDCF) and HCF controlled channel access. Each station that uses 802.11e can access the channel based on Access Category (AC), to grant the priority. Other techniques could be used to prioritize the voice traffic such as using smaller Contention Window (CW) and back-off size. These techniques depend on the condition of the wireless network, the running service and the quality of service policy [15].

C. *VoIP Quality of Service*

In wireless networks, bandwidth is shared between all converged services which demand different sizes of bandwidth. The prioritization of packets sorts the real-time sensitive data by controlling the channel access and allocating access time slots for each packet. The Mean Opinion Score (MOS) methodology is used to measure the quality of voice call based on a scale from 5 to 1. The Bad quality is represented with 1 whereas Excellent quality is represented as 5 [16].

Voice over IP (VoIP) can run on any IP network and can be utilized as hardware or software. VoIP is cost effective compared to analogue voice systems. It can be transferred to clients regardless of their location. In addition, a single network is used to accommodate the data and voice packets resulting in lower management and operation costs.

VoIP is a real-time service that tolerates no latency and needs a stable IP network. Therefore, VoIP networks are needed to maintain a quality of service to guarantee voice packet delivery. The voice packets are prioritized using mechanisms of control and prioritization flow.

Measuring the quality of voice service could be accomplished by monitoring some quality parameters such as packet end-to-end delay, or throughput and delay variation (jitter). From a network perspective, there are three types of QoS which are; best effort, integrated services and differentiated services. The prioritization of packets is categorized in three mechanisms which are; priority queuing (PQ), weighted fair queuing (WFQ) and custom queuing (CQ) [17]. VoIP over wireless is a trade-off between quality and simplicity. Integration between the voice and the wireless provides clients with simplicity and mobility. Associated concerns are the degree network coverage and security issues of wireless. Voice over wireless (VoWLAN) depends on wireless network quality and security. Enhancing the quality of voice service (voice QoS) is basically based on enhancement of the wireless network. The channel access control mechanisms in wireless networks can sort and minimize the VoIP packet end-to-end delay. It allocates sufficient bandwidth based on the used voice codec. Eventually, an excellent VoIP service could be achieved by implementing suitable codec and QoS policy on the wireless medium.

In this paper, the testbed model is used in a specific case study which is the conference events. This testbed elaborates on the most used standard which is 802.11g in comparison with 802.11e.

III. Network Model and Implementation

The following figure presents the network model. The model comprises a wireless network between two buildings. Each building consists of offices and a meeting room which is used for video conferencing to other building. The voice subnet (VLAN) is separated from video subnet (VLAN) and there is no data traffic except the normal management data between devices.

It emulates the distance-learning environment in a medium size campus as shown in Fig. 1. It concentrates on the voice service and videoconferencing service. The data service is assumed to be carried on a wired medium and isolated wireless channels, which do not overlap with voice wireless channels.

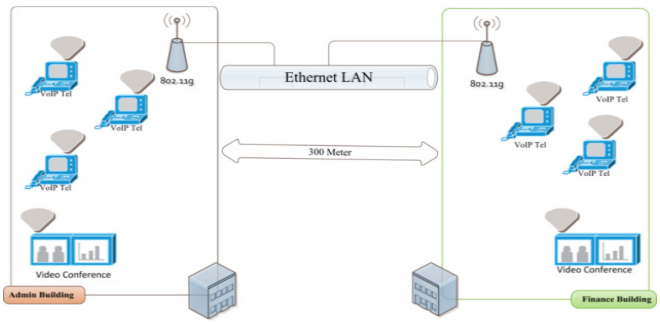

Fig. 1. Network Model

The network is modeled to evaluate various scenarios that are going to experiment the quality of service of voice traffic in different setups. The configuration of the simulation tool is based on four different scenarios, which will enable assessment of the performance of VoIP traffic over a wireless network as illustrated in Table I and Fig. 2, 3, 4 and 5. The enhancement of the wireless network, forces the traffic to follow a QoS policy. The QoS of VoIP depends on the quality of configuration in wireless network. At the beginning, legacy IEEE 802.11 standard is used to evaluate the quality of service parameters. Afterwards the enhanced standard IEEE 802.11e is used with specific amendments in the configurations. The amendments are expected to improve the quality of the voice service over the wireless network.

TABLE I. EXPERIMENTAL SCENARIOS

	Single client talks to two clients		**Multi clients send to Multi clients**		**Video application in parallel with VoIP**		**Existence of Jammer (2.4GHz)**	
Wireless	802.11g		802.11g		802.11g		802.11g	
Codec	G .711	G .729	G .711	G .729	G .711	G .729	G .711	G .729
Frames per packet	1, 3, 5	1, 3, 5	1, 3, 5	1, 3, 5	1, 3, 5	1, 3, 5	1, 3, 5	1, 3, 5
Application	voice		voice		Voice and video		voice	
Number of client	3 voice		15 voice		3 voice / 2 video		3 voice	
Simulation Time	10 Minutes (600 seconds)		10 Minutes (600 seconds)		10 Minutes (600 seconds)		10 Minutes (600 seconds)	

In all scenarios, the simulation has been run three times in order to validate the output data which was found to be identical at the end of all three simulations.

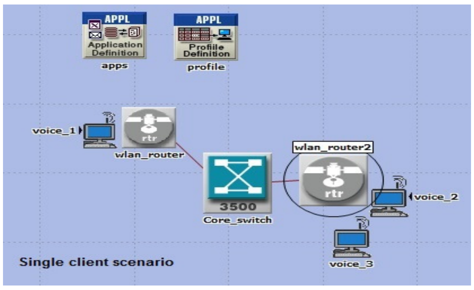

Fig. 2. Single Client Scenario

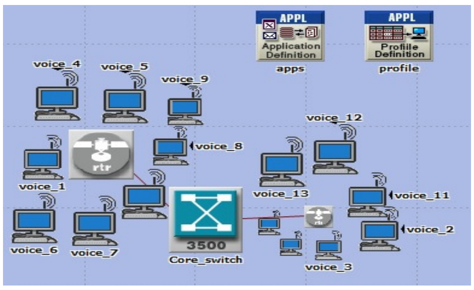

Fig. 3. Multiple Client Scenario

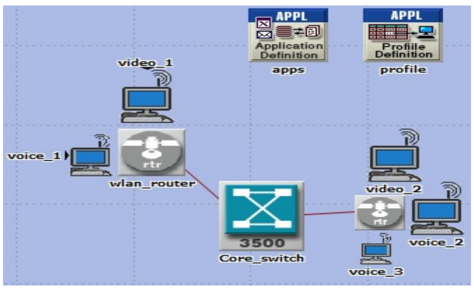

Fig. 4. Voice and Video Clients Scenario

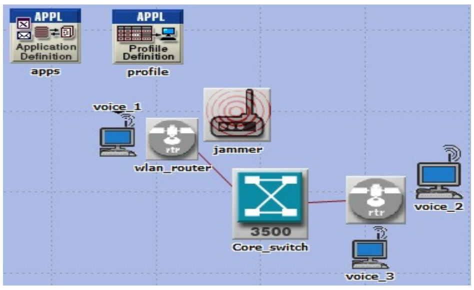

Fig. 5. Jammer Scenario

The new settings for QoS are applied to enhance the network performance, and by association improving VoIP performance. The main change is in enabling the QoS of wireless and Hybrid Coordination Function (HCF) in all wireless devices. The wireless parameters of the voice node, video node and router will use new settings of 802.11e as shown in Fig. 6.

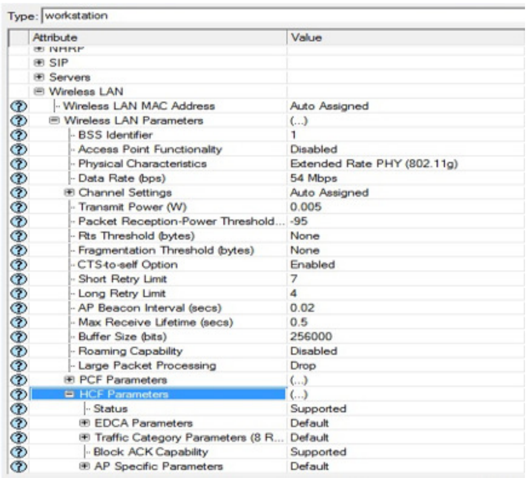

Fig. 6. Wireless Router, Voice Node and Video Node Attributes

IV. ANALYSIS OF RESULTS

The simulation has been run for various scenarios with different parameters that address the objectives of the research. The main affective metrics are Packet End-to-End Delay, Jitter and Throughput that evaluates the performance of the VoIP traffic in a wireless network.

A. *Bandwidth requirements*

The VoIP packet consists of a voice payload and the IP header. The IP header adds 20 bytes of IP, 8 bytes of User Datagram Protocol (UDP) and 12 byte of Real-Time Protocol (RTP) [18]. The Ethernet overhead is 38 bytes. The required bandwidth per VoIP packet is calculated follow:

The bandwidth for VoIP packet = (voice payload + header + Ethernet overhead) x packets per second x 8 bits

The G.711 codec sampling time is 20 msec (i.e. 50 packets per 1 second) and the size of the frame is 64 Kbit. For example, Payload of for 1 frame of G.711 = 64000/50 = 1280 bit = 160 byte. Hence, Bandwidth for G.711 at 1 frame per packet = (160+40+38) x 50 x 8 = 95.2 kbps.

The G.729 codec sampling time is 10 msec (i.e. 100 packets per 1 second) and the size of the frame is 8 Kbit. For example, Payload of for 1 frame of G.729 = 8000/100 = 80 bit = 10 byte. Accordingly, Bandwidth for G.729 at 1 frame per packet = (10+40+38) x 100 x 8 = 70.4 kbps.

TABLE II. THEORETICAL BANDWIDTH REQUIREMENTS FOR G.711 AND G.729 CODECS OVER ETHERNET WLAN

Codec	Bandwidth	Sample period	Frame size	Frame per Packet	Bandwidth required (Ethernet)
G.711	64 kbps	20 ms	160 byte	1	95.2 kbps
				3	223.2 kbps
				5	351.2 kbps
G.729	8 kbps	10 ms	10 byte	1	70.4 kbps
				3	86.4 kbps
				5	102.4kbps

The admitted and acceptable VoIP calls are calculated based on the simulation time and Packet End-to-End Delay curve. VoIP calls are acceptable if the delay does not exceed the delay tolerance of < 150 msec. The simulation time is 600 sec (10 minutes) and the VoIP calls start at 40 sec. All scenarios have 3 voice clients talk to each other, except in the multi call scenario which has 15 voice clients. Therefore, each voice client starts the first call at 40 sec, and after 10 sec another VoIP call is initiated till the end of simulation. Hence, 3 VoIP calls initiated at the beginning and another 3 VoIP calls are added every 10 sec. For example at Time 306 sec; the Delay is <150 msec. Hence, acceptable VoIP calls using G.711 codec with 5 frames per packet = 3+ {(306-40)/10} x3 = 82 VoIP calls. The following equation is used to calculate the VoIP calls in each scenario as follow:

Number of acceptable VoIP calls = $3_{initial\ calls}$ + {(Simulation Time$_{<150msec}$ – Simulation Time$_{start}$) / Time interval between calls} x $3_{calls\ every\ 10\ sec}$.

B. *Number of VoIP calls using G.711 Codec over Legacy 802.11g Wireless Network*

Table III shows the number of acceptable VoIP calls using G.711 codec over legacy 802.11g wireless network. The higher number of VoIP calls are reached using 5 frames per packet regardless the scenario. The single station scenario and jammer scenario have approximately same number of VoIP calls due to the access point functionality of interference resistance feature.

TABLE III. ACCEPTABLE VOIP CALLS USING G.711 CODEC

G.711	Frames per Packet	The Full Calculations	Acceptable calls (Delay ≤ 150ms)
Single client talks to two clients	1	3 + {(126- 40)/10} x 3	28
	3	3 + {(276- 40)/10} x 3	73
	5	3 + {(306- 40)/10} x 3	82

Multi clients send to Multi clients	1	15 + {(72- 40)/10} x 15	63
	3	15 + {(78- 40)/10} x 15	72
	5	15 + {(78- 40)/10} x 15	72
Video application in parallel with VoIP	1	3 + {(198- 40)/10} x 3	50
	3	3 + {(396- 40)/10} x 3	109
	5	3 + {(564- 40)/10} x 3	160
Existence of Jammer (2.4GHz)	1	3 + {(126- 40)/10} x 3	28
	3	3 + {(276- 40)/10} x 3	73
	5	3 + {(306- 40)/10} x 3	82

C. *Number of VoIP calls using G.729 codec over legacy 802.11g wireless networks*

The acceptable number of VoIP calls using G.729 codec over legacy 802.11g is demonstrated in Table IV. This table is calculated based on maximum number of VoIP calls at certain simulation time which corresponds to delay <150 ms of packet end-to-end delay. All scenarios show the advantage of using 5 frames per packet in G.729 codec; except the video scenario which has the superiority of 3 frames per packet.

TABLE IV. ACCEPTABLE VOIP CALLS USING G.729 CODEC

G.729	Frames per Packet	The Full Calculations	Acceptable calls (Delay ≤ 150ms)
Single client talks to two clients	1	3 + {(138- 40)/10} x 3	32
	3	3 + {(324- 40)/10} x 3	88
	5	3 + {(414- 40)/10} x 3	115
Multi clients send to Multi clients	1	15 + {(66- 40)/10} x 15	54
	3	15 + {(84- 40)/10} x 15	81
	5	15 + {(96- 40)/10} x 15	99
Video application in parallel with VoIP	1	3 + {(186- 40)/10} x 3	46
	3	3 + {(414- 40)/10} x 3	115
	5	3 + {(342- 40)/10} x 3	93
Existence of Jammer (2.4GHz)	1	3 + {(138- 40)/10} x 3	32
	3	3 + {(324- 40)/10} x 3	88
	5	3 + {(414- 40)/10} x 3	115

D. *Network Enhancement for VoIP QoS using 802.11e standard*

Table. V shows the number of acceptable VoIP calls in 802.11e.

TABLE V. NUMBER OF VOIP CALLS USING G.729 AND 802.11E

G.729 codec over 802.11e	Frames per Packet	The Full Calculations	Acceptable calls (Delay ≤ 150ms)
Single client talks to two clients	5	3 + {(648- 40)/10} x 3	185
Multi clients send to Multi clients	5	15 + {(144- 40)/10} x 15	171
Video application in parallel with VoIP	5	3 + {(840- 40)/10} x 3	243
Existence of Jammer (2.4GHz)	5	3 + {(636- 40)/10} x 3	181

The results and findings show the data outputs in OPNET starts in most scenarios after 30 sec, which is the nature of the OPNET simulation tool. The results present the effects of adding voice clients to the wireless network, and changing the sampling codec from G.711 and G.729. Further, the findings illustrate the outputs of QoS parameters that result from transferring voice traffic over legacy 802.11g and enhanced 802.11e wireless networks. The packet end-to-end delay has been reduced significantly when the voice traffic is transferred over 802.11e. This reduction of delay allows for the adding of more VoIP calls per scenario and per voice station. The delay variation (jitter) appears to be within ITU-T tolerance in all cases which indicates no echo on any VoIP calls. The throughput graphs illustrate high bit/s using 5 frames per packet in G.729 sampling, which aids adding more VoIP calls per voice station. The enhanced 802.11e HCF feature, gives priority to voice traffics over video and data traffics. This feature holds the video traffic to give way to voice traffic; therefore, bandwidth management should be included in a QoS policy that ensures fair distribution of bandwidth between all services.

In the Jammer scenarios, the jammer has a periodic pulse, which causes other stations to adjust their sending period to avoid interference. In addition, the access point is an intelligent device, which features with ability to change the channel to mitigate the interference.

Fig. 7. shows the superiority of using 5 frames per packet with G.729 codec over 802.11e which is reflected as higher number of VoIP calls compared to other settings over legacy 802.11g.

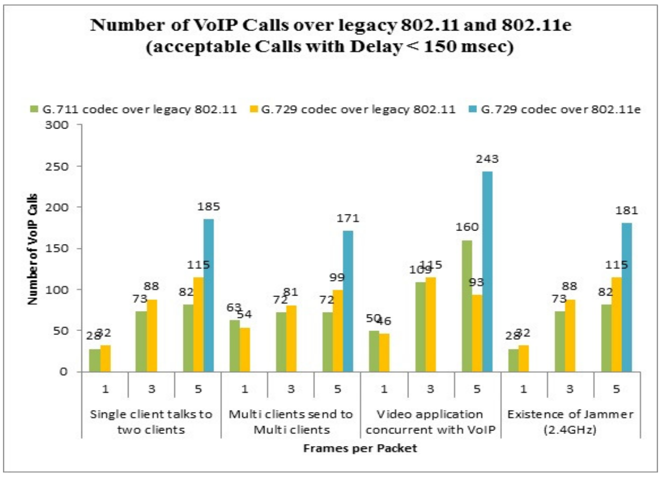

Fig. 7. Number of VoIP Calls over legacy 802.11 & 802.11e

V. CONCLUSIONS AND FUTURE DIRECTIONS

The voice service is an essential tool in the current era. The telephony systems are developed to meet the customers' requirements, which are mobility, security, performance and cost effectiveness. Packet switching networks are adopted to improve the efficiency of converged services such as voice calls. The transaction of voice calls over IP networks is constrained due to the competition for bandwidth between the converged services; data, voice and video. Nowadays, voice services are sent via Internet protocol (VoIP) and mobilized over wireless networks (VoWLAN). Operating wireless 802.11 networks as a medium is promising; although it has some limitations such as delay and bandwidth overload.

In this paper, the authors has evaluated the performance of Voice over IP (VoIP) in 802.11 wireless networks. The implementation was divided into two phases that cover all quality criteria in voice packets and wireless network mediums. The first phase was carried out by transferring voice packets over normal 802.11g wireless network. The second phase was run on selective scenarios over an enhanced 802.11e wireless network. The number of VoIP calls was calculated based on the voice packet end-to-end delay, which must not exceed 150 msec according to ITU-T Recommendation. Executing the VoIP service over 802.11e wireless network decreases the packet end-to-end delay as a result of the quality of service configuration in wireless access point devices. The throughput enhancement is a consequence of applying access control mechanisms in 802.11e access point devices. In addition, it has been concluded that the G.729 codec with 5 frames per packet is superior to G.711 codec in 802.11e networks. The number of frames per packet selection is a trade-off between the packet latency and packet overhead. The VoIP service over 802.11 wireless networks is valid with consideration of implementing quality of service policies, which guarantee a robust and stable unified communication system. The enhancement on 802.11e could have an undesirable implication on lower level traffic such as FTP and HTTP. Therefore, the QoS policy should consider all types of traffic in bandwidth allocations.

In the future, this paper could be extended and developed to cover the other wireless constrains that affect the quality of voice services. The wireless network is susceptible to eavesdropping on voice communications. The encryption of voice traffic could lead to higher packet delay due to encryption algorithms and the use of encapsulation methods such as IPsec tunnels. A comprehensive research would be carried out to develop a framework that mitigates the security bottlenecks on wireless network. The framework requires further evaluation, which will be applied on a robust test bed design.

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