Markov Chain Model and Performance Enhancement for EDCA Protocol

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Abstract —The real-time application is the main challenge that faces the wireless network. It must achieve the quality of service (QOS) requirements to success. IEEE802.11 standards use Enhanced Distributed Channel Access (EDCA) protocol and Distributed Coordinate Function (DCF) protocol in its Media Access Control (MAC) layer. EDCA protocol is used to overcome the drawback of supporting QOS in DCF protocol. The mechanism of EDCA protocol is based on dividing the data to four queues; voice, video, best effort and background. The queues have different priorities to access the medium. The voice queue has the highest priority and its data will serve firstly. Using the default parameter values for EDCA protocol will lead to increasing the collisions in the wireless network and decreasing the capacity. In this paper, the limitation of supporting QOS in EDCA protocol was presented by using OPNET simulation and mathematical model. The new mathematical model was designed by using Markov chain mechanism to evaluate the performance of the EDCA protocol under saturation and non-saturation conditions, aimed to separate between uplink and downlink throughput with different data types. Moreover, the new algorithm was proposed to enhance the capacity of the EDCA protocol and increase the number of the active voice users. Through our algorithm, the throughput was increased by 38.3% and the EDCA can support 14 voice users rather than 11 in the default EDCA protocol. The evaluation of our new algorithm was verified by using OPNET simulation and mathematical model.

Index Terms—DCF, EDCA, QoS, markov chain

I. INTRODUCTION

IEEE 802.11 standard is considered one of the most popular techniques which are used in the wireless network. The simplicity of configuration, ease to expand and low costs are its main properties [1]. Recently, there is huge usage of the Internet in different fields of life, and there is a need of supporting the Internet in public locations such as parks, restaurants, bus stations and airports. These locations do not have a fixed number of users, and can be increased dramatically. Therefore, a wireless network is suitable in these locations as it can accept more users with low cost [2]. Real-time applications such as the voice over internet protocol (VOIP) and video conference are considered the main challenges that face wireless network [3]. The real-time applications succeed within specific conditions in the delay time and packet loss percentage. For example, the VOIP calling will be accepted when the end-to-end delay is less than 150ms and the packet loss percentage does not exceed than 1% [4]. Thus, the real-time applications must achieve quality of service (QOS) requirement to succeed. The first protocol which is used in IEEE802.11 standard is the Distributed Coordination Function (DCF) protocol. The mechanism of the DCF protocol is based on using one queue, where all data enter the queue according to the arriving time without any advantages for real-time data [5]. Hence, the DCF protocol does not support QOS. On the other hand, the Enhanced Distributed Channel Access (EDCA) protocol is used to overcome this problem, as the EDCA mechanism depends on distributing the data traffic between four queues; voice, video, best effort and background. Each queue receives its data type. The voice queue has the highest priority, followed by video, best effort and background respectively. Serving the voice queue first will lead to decreasing the delay time and the packet loss percentage. Through this mechanism, the EDCA protocol supports QOS [6]. However, there is a limitation in supporting QOS in the EDCA protocol. Increasing the load of real time data traffic will increase the collision in the network. Thus, delay time and packet loss percentage will increase more than the requirements of QOS [7]. Now, researchers concern to analyze the EDCA protocol through simulators and mathematical model. The Markov chain mechanism is used to evaluate the performance of the EDCA protocol mathematically. Furthermore, the authors in [8]–[11] evaluate the performance of the EDCA protocol under the saturation condition. However, the network reaches saturation after it goes through the non-saturation condition. Hence, there is a need to analyze the EDCA protocol under non saturation. Besides that, the researchers in [12], [13] created the Markov chain models to evaluate the EDCA protocol under non saturation, but all these models calculate the total throughput without any difference between uplink and downlink traffic. Thus, it is important to separate between uplink and downlink traffic to find the bottleneck of the network, and to indicate who prevents the wireless network to accept more real-time application users.

In this paper, the Markov chain model is proposed to evaluate the EDCA protocol under saturation and non-saturation conditions with separation between uplink and downlink throughput. Also, the limitation of supporting QOS in the EDCA protocol is presented by using

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mathematical model and OPNET simulation. In addition, a new algorithm is proposed to enhance the capacity of the EDCA protocol to support more voice users by adjusting contention window. As well as, the results are verified by using the mathematical model and OPNET simulation.

II. EDCA PROTOCOL

The EDCA protocol is used to overcome the drawback of the DCF protocol, which does not support QOS [14]. In EDCA, there are four queues as compared to one queue in DCF. These queues are specified to four types of traffic; voice, video, best effort and background. The queues have different priorities to access the medium. The default EDCA protocol gives the voice queue the highest priority, therefore the voice data will be delivered faster than other types and will achieve QOS requirements [15]. Besides that, each queue has specific parameters which are used to define the priorities. CWmin, CWmax, Arbitration Inter Frame Space (AIFS) and Transmit Opportunity Length limit (TXOPlimt) are four parameters that affect the priorities between the queues [16].

CWmin is the first value which is used to calculate the backoff time. Using a small value of CWmin will lead to get a small backoff time counter value, and the queue will access the medium rabidly. In addition, a small value of CWmax will decrease the backoff time counter value. Therefore, the voice queue has the smallest value of CWmin and CWmax [17].

The station must wait a period of time before sensing the medium, as well as the backoff time counter will start decreasing when the medium is free. The period of waiting time is determined by using AIFS parameter. Furthermore, setting small value of AIFS will help to start counting down the backoff time counter faster, and increase the priority of the queue. AIFS value can be calculated by using (1).

$$\text{AIFS[AC]=SIFS+AIFSN[AC]}\times\text{Slot time} \tag{1}$$

where, AIFSN is an integer number which is used to define the number of waiting slot time. Therefore, the voice and video queues have smaller values of AIFSN rather than other queues, in order to increase the priorities of real time data traffic [18].

TABLE I: DEFAULT VALUES OF EDCA PARAMETERS

AC	CWmin	CWmax	AIFSN	TXOPlimit
AC_VO	7	15	2	3.264ms
AC_VI	15	31	2	6.016ms
AC_BE	31	1023	3	0
AC_BK	31	1023	7	0

Meanwhile, the TXOPlimt parameter is used to define how many frames can be sent during reserving the medium. If TXOPlimt value is equal to zero, the queue can send only one frame every time it accesses the medium. However, if the value is more than zero, it will represent the period of time for reserving the medium, and the queue can send more than one frame during this time [19]. Hence, the voice and video queues have values of TXOPlimt more than zero to increase the number of frames when accessing the medium. Table I shows the default values of EDCA protocol parameters [20].

III. MARKOV CHAIN MODEL

The Markov chain is a technique which is used to evaluate the performance of network mathematically. Fig. 1 shows the probabilities of movement state for our Markov chain model. Each state has three variables; *i*, *j* and *k*. Variable *i* is used to present the class priority. In the EDCA protocol, there are four priorities, hence the value of variable *i* is close between zero to three. Apart from that, variable *j* represents the number of retransmission attempts in the network. If Access Category (AC) fails to send the data, the state will move to the next level by increasing the value of variable *j* whereas the maximum number of retransmission attempts is *Lrety*. Variable *k* is used to describe the backoff counter, thus the state moves in the same level by decreasing the variable *k* by one. Suppose *pi* is the probability of sensing the medium busy, the state will not move if the medium is busy. However, the state will change in the same *j* level when the channel is free with (1-*pi*) probability.

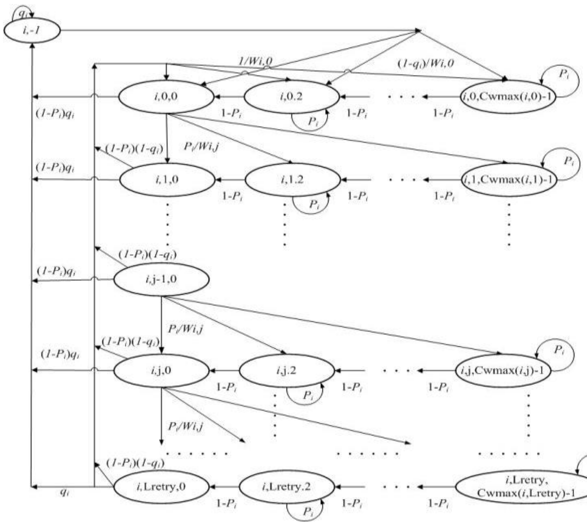

Fig. 1. Markov chain model for EDCA

Two types of traffic are used in the Markov chain model; uplink and downlink traffics. The traffic that moves from station to AP called uplink traffic, but the downlink traffic moves from AP to station. Therefore, $ACup[i]$ and $ACdw[i]$ are the access category in station and AP respectively with priority *i*. The movement state probabilities can be concluded as:

$$P[(i,j,k)|(i,j,k+1)] = 1 - P_i \tag{2}$$

$$P[(i,j,k)|(i,j,k)] = P_i \tag{3}$$

$$P[(i,j+1,k)|(i,j,0)] = \frac{P_i}{CW(i,j+1)} \tag{4}$$

$$P[(i,0,k)|(i,j,0)] = \frac{(1-P_i)(1-qi)}{CW(i,0)} \tag{5}$$

$$P[(i,0,k)|(i,Lrety,0)] = \frac{(1-qi)}{CW(i,0)} \tag{6}$$

$$P[(i,-1)|(i,j,0)] = (1-P_i)qi \tag{7}$$

$$P[(i,-1)|(i,-1)] = qi \tag{8}$$

$$P[(i,0,k)|(i,-1)] = \frac{(1-qi)}{CW(i,0)} \tag{9}$$

Supposed that $b(i,j,k)$ is the steady state for the Markov chain. By using the Markov chain properties that the summation of all state probabilities equals to one, $b(i,0,0)$ can be derived as:

$$b(i,j,0) = P_i^j b(i,0,0) \tag{10}$$

$$b(i,j,0) = b(i,j-1,0)P_i \tag{11}$$

$$bi,-1 = \frac{qi}{(1-qi)} bi,0,0 \tag{12}$$

$$b(i,j,k) = \frac{CW(i,j)-k}{CW(i,j)} \quad \frac{b(i,j,0)}{1-P_i} \tag{13}$$

$$b(i,0,0) = \frac{1}{[\frac{qi}{(1-qi)} + \sum_{j=0}^{Lretry}[1+\frac{1}{1-P_i}\sum_{k=1}^{CW(i,j)-1}\frac{CW(i,j)-k}{CW(i,j)}]P_i^j]} \tag{14}$$

The probabilities of transmission for the uplink (τup_i) and downlink (τdw_i) traffics can be calculated by summing all state probabilities when k equals zero.

$$\tau up_i = \sum_{j=0}^{Lretry} b(i,j,0), \text{in station} = b(i,0,0)\frac{1-pup_i^{Lretry+1}}{1-pup_i} \tag{15}$$

$$\tau dw_i = \sum_{j=0}^{Lretry} b(i,j,0), in\ AP = b(i,0,0)\frac{1-pdw_i^{Lretry+1}}{1-pdw_i} \tag{16}$$

Pup_i and Pdw_i are the probabilities of sensing that the medium is busy in station and AP are respectively. The medium is sensed to be free by the AC in the station when the rest of the ACs in all stations and AP do not send the data. On the other hand, the medium is sensed to be free by AC in the AP, when all ACs in the stations and other ACs in AP do not send the data. Thus, Pup_i and Pdw_i can be calculated as:

$$Pup_i = 1-(1-\tau up_i)^{n_i-1}(1-\tau dw_i)\prod_{l=0\ l\neq i}^{N-1}(1-\tau up_l)^{n_l}(1-\tau dw_l) \tag{17}$$

$$Pdw_i = 1-(1-\tau up_i)^{n_i}\prod_{l=0\ l\neq i}^{N-1}(1-\tau up_l)^{n_l}(1-\tau dw_l) \tag{18}$$

where N is the number of priorities and ni is the number of ACs that have data with priority i. $Psup_i$ and $Psdw_i$ represent the probability of successful transmission in the station and AP, respectively. Their equations can be derived as:

$$Psup_i = n_i\,\tau up_i(1-\tau up_i)^{n_i-1}(1-\tau dw_i)\prod_{l=0\ l\neq i}^{N-1}(1-\tau up_l)^{n_l}(1-\tau dw_l) \tag{19}$$

$$Psdw_i = \tau dw_i\,(1-\tau up_i)^{n_i}\prod_{l=0\ l\neq i}^{N-1}(1-\tau up_l)^{n_l}(1-\tau dw_l) \tag{20}$$

The probability of success for all priorities in the station ($Psuccess\ up$) is calculated by summing all $Psup_i$ at different priorities, but the summation of all $Psdw_i$ will lead to get the probability of success for all priorities in AP ($Psuccess\ dw$).

$$Psuccess\ up = \sum_{i=0}^{N-1} n_i\,\tau up_i(1-\tau up_i)^{n_i-1}(1-\tau dw_i)\prod_{l=0\ l\neq i}^{N-1}(1-\tau up_l)^{n_l}(1-\tau dw_l) = (1-Pbusy)\sum_{i=0}^{N-1}\frac{n_i\,\tau up_i}{1-\tau up_i} \tag{21}$$

$$Psuccess\ dw = \sum_{i=0}^{N-1}\tau dw_i\,(1-\tau up_i)^{n_i}\prod_{l=0\ l\neq i}^{N-1}(1-\tau up_l)^{n_l}(1-\tau dw_l) = (1-Pbusy)\sum_{i=0}^{N-1}\frac{\tau dw_i}{1-\tau dw_i} \tag{22}$$

where, Pbusy is the probability when the whole system becomes busy, so:

$$Pbusy = 1-\prod_{i=0}^{N-1}(1-\tau up_i)^{n_i}(1-\tau dw_i) \tag{23}$$

Throughput of the network can be calculated by getting the ratio between the time of successful payload transmission and the time between two successful transmissions. Let say s_i is the throughput for $AC[i]$:

$$S_i = \frac{E(payload\ successful\ transmission\ time\ for\ class\ i)}{E(total\ \ time\ betwen\ two\ successive\ transmission)} \tag{24}$$

The total time between two successful transmissions has three parts: the first part represents the ideal time slots when the backoff counter is decreased by one. The second part is the time of successful transmission. The last part calculates the collision time. In our work, the throughput is divided into uplink (sup_i) and downlink (sdw_i). Let, ∂, $TE(L)$, T_s, T_c, and $P_{successful\ all}$ denote slot time period, time of successful transmission for payload, period of time of successful transmission, wasted collision time, and the total probabilities for $Psuccess\ up$ and $Psuccess\ dw$, respectively:

$$Sup_i = \frac{Psup_i T_E(L)}{(1-Pbusy)\partial + P_{success\ up}T_s + [Pbusy - P_{successful\ all}]T_c} \tag{25}$$

$$Sdw_i = \frac{Psdw_i T_E(L)}{(1-Pbusy)\partial + P_{success\ dw}T_s + [Pbusy - P_{successful\ all}]T_c} \tag{26}$$

T_s and T_c values depend on the method of transmission used. In our work, we use the basic method of transmission. According to the timing diagram for the basic method, T_s and T_c can be calculated as:

$$T_s = AIFS[i] + \left(\frac{H_p}{speed_p} + \frac{H_m + E(L) + FCS}{speed_m}\right) + SIFS + \left(\frac{H_p}{speed_p} + \frac{ACK}{speed_m}\right) \tag{27}$$

$$T_c = AIFS[i] + \left(\frac{H_p}{speed_p} + \frac{H_m + E(L) + FCS}{speed_m}\right) \tag{28}$$

where, H_p, H_m, $speed_m$, $speed_p$, and FCS are the size of the header for the physical layer, the size of header for the MAC layer, the speed of MAC layer, speed of physical layer, and frame check sequence for error detection, respectively. Table II presents the IEEE802.11b parameter values.

TABLE II: IEEE802.11B PARAMETERS VALUES

Parameters	Values
MAC speed($speed_m$)	11Mbps
Physical speed ($speed_p$)	1Mbps
Physical header (H_p)	192bit
MAC header(H_m)+FCS	288bit
ACK frame(ACK)	112bit
Slot time (∂)	20μs
SIFS	10μs
DIFS	2×∂+SIFS
AIFS[AC]	SIFS+AIFSN[AC]×∂

IV. ESTIMATE THE TRAFFIC LOAD(qi)

The state (i, -1) is used to define the non-saturation condition. There is a need to control the traffic load and increase it gradually until the system reaches the saturation. Moreover, the qi variable represents the probability that there is no ready data to send it after the successful of transmission. To control the traffic in the system, the qi must be estimated. Suppose the traffic is generated based on Poisson distributed technique, λ is the data rate of traffic (packet/second). Let b (t) is the probability density function (PDF) of time slot.

$$b(t) = p_{success}\delta(t - T_s) + \left(1 - b_{busy}\right)\delta(t - \partial) + \left(p_{busy} - p_{success}\right)\delta(t - T_c) \tag{29}$$

Thus, qi can be calculated according to Poisson distribution.

$$qi = \int_0^\infty e^{\lambda_i t}\, b(t)dt = p_{success}e^{-\lambda_i T_s} + \left(1 - b_{busy}\right)e^{-\lambda_i \partial} + \left(p_{busy} - p_{success}\right)e^{-\lambda_i T_c} \tag{30}$$

V. CHALLENGES THAT FACE VOIP IN THE WIRELESS NETWORK

The VOIP is the real-time application. It works under restricted delay time and packet loss percentage. The VOIP calling succeeds when it achieves the requirement of QOS. The limitation of supporting QOS for end-to-end delay and packet loss percentage are 150 ms and 1% respectively. That means, the data rate, which is created in the source must reach the destination in the same rate without any decreasing in the packets more than 1%. Several codec standards are used in the VOIP such as G.711, G.729 and G.723 [21]. By using G.711 codec standard, two ways of traffic load are created (uplink and downlink). The data rate is 80kbps in each way. Moreover, the data rate value depends on the value of voice frame which contains 160B payload traffic. In addition, 40B is added through the network layers. If the capsulation time is 20ms, then the data rate is 200B×8/20ms =80kbps [22].

Usually, the VOIP application is applied in the common wireless network scenario containing one access point and several wireless workstations. The access point connects to the internet by the wire. Uplink and down link traffic loads are created when the session of the VOIP starts. Suppose 802.11b standard is used with G.711 (20ms) voice codec, the data rate of 802.11b standard is 11Mbps [23]. Besides that, the capacity of the voice user can be calculated simply by dividing the wireless 802.11b data rate over two ways G.711 data rates (11Mbps/2×80kbps), equivalent to 68 voice users. However, in actual fact, the wireless network can cover much less than 68 voice users. Various factors affect on the capacity of the wireless network, and the collision is the main one. All active voice users need to send their data to access point. In contrast, the access point needs to ensure download data traffic for all voice stations. This means that there is high load of real time data traffic at the access point. Thus, the collision will increase in the access point, causing decline in the download throughput. To achieve QOS for N voice G.711 users, the uplink and downlink throughput must be equal to N×80kbps. The difficulty is to ensure (N×80kbps) download traffic from the access point to the voice stations. Because there is high load voice traffic data in access point, and the CW range for voice is small, which will lead to increase the collision in the network. In addition, the percentage of the drop packets will rise. Therefore, the access point is the bottleneck of the network which prevents increasing the capacity of the wireless network.

VI. Limitation of Supporting QOS in EDCA Protocol

In this section, two scenarios were designed to present the limitation of supporting QOS in the default EDCA protocol. The OPNET simulation is used to design wireless network and analysis the result.

A. Scenario 1

In this scenario, a wireless network was created containing 11 wireless workstations, one access point and one server. All the wireless stations accessed the Internet through the access point. The EDCA protocol was used with all wireless stations and access point. In addition, the default EDCA parameter values were set in all stations. Three applications were defined; voice, best effort and background. All these three applications were working at the same time. The voice application used G.711 codec standard with 20ms capsulation time. The voice application sent 50packet/second in both direction uplink and downlink. In this scenario, QOS parameters were measured to check if they achieved the requirement or not.

The end-to-end delay is the total time taken when the sound was created from the source until it reached the destination, which must be less than 150ms to support QOS. Fig. 2 presents the end-to-end delay in scenario 1 and the average of end-to end-delay is about 80ms. This indicates that with 11 voice stations, the end-to-end delay still supports QOS.

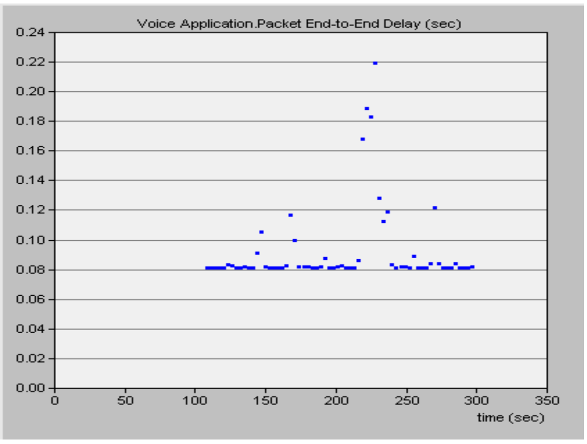

Fig. 2. The voice end-to-end delay in scenario 1

Through a transmission operation, some packets dropped and did not reach the destination. To support QOS, the percentage of the packet loss must be less than 1%. Fig. 3 presents the data rates for the voice application which were sent from the source. On the other hand, Fig. 4 shows the received data traffics in the destination. By doing a comparison between the results in Fig. 3 and Fig. 4, the packet loss percentage is less than 1%.

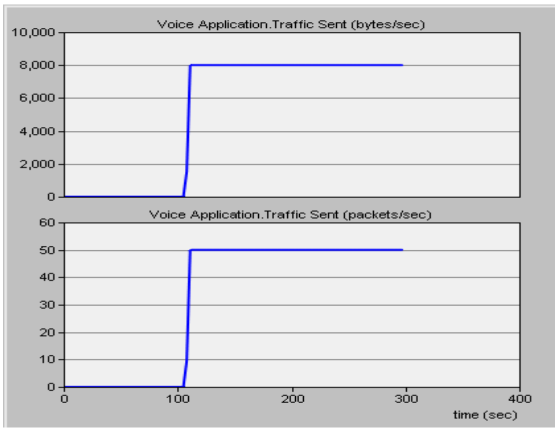

Fig. 3. The rate of sending voice data traffic in scenario 1

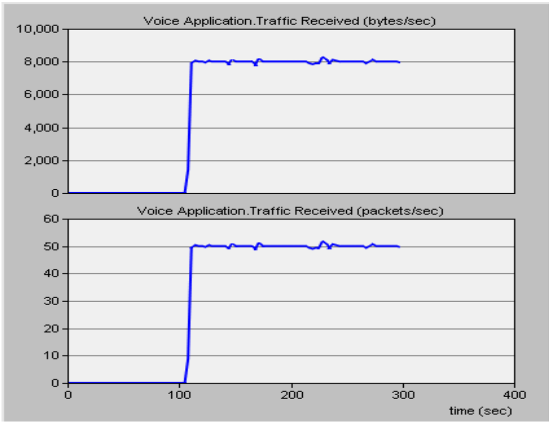

Fig. 4. The rate of receiving voice data traffic scinario1

B. Scenario 2

In this scenario, the wireless network was designed. It contained 12 wireless workstations, in addition to one access point and one server. Three applications were defined as in scenario1; voice, best effort and background. All of the workstations used EDCA protocol in its MAC layer. The only difference between scinario1 and scenario2 is the number of workstations in scenario 2 became 12 rather than 11. That means, the aim of this scenario was to increase the load in the wireless network and to check if the EDCA protocol still supports QOS or otherwise.

The end-to-end delay was measured when increasing the voice load traffic. Fig. 5 presents the end-to-end delay for voice traffic in scenario 2. When the end-to-end delay value was about 380ms, it did not achieve the requirement of the QOS. Moreover, Fig. 6 shows the received voice traffics data. The average of the packet loss percentage is 23.75%, showing that it did not achieve the QOS requirements. Thus, the EDCA protocol has limitations of supporting QOS when the load of high priority data traffic is increased. This is in regards to increasing the collisions in the network. The EDCA protocol can support QOS until 11 voice users. More than this, the end-to-end delay and packet loss percentage have not achieved the requirements of QOS.

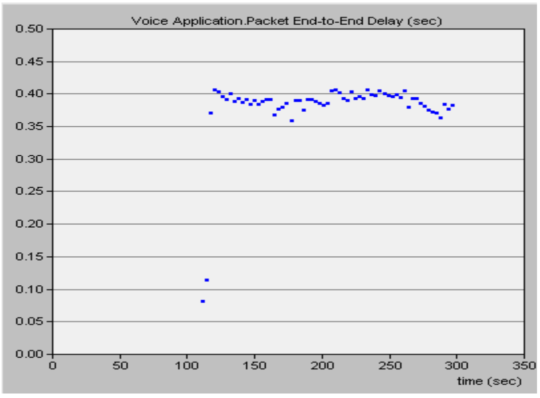

Fig. 5. The voice end-to-end delay in scenario 2

Fig. 6. The rate of receiving voice data traffic scenario 2

VII. PROPOSED NEW ALGORITHM TO ENHANCE THE PERFORMANCE OF EDCA PROTOCOL

In this section, a new algorithm was proposed to enhance the capacity of wireless networks and increase the number of voice users. Our algorithm aimed to decrease the collision in the network and increase the throughput. Using default parameter values in the EDCA protocol leads to increasing the collision in the networks. Thus, two types of collision can happen in the network; internal collision and external collision. For instance, the EDCA protocol has four queues in its MAC layers. Each station represents four stations in actual. Thus, the percentage of collision will rise and affect the capacity. In addition, the new algorithm was created based on the adaption of contention window for the voice queue to decline the collision. It creates different ranges for contention window between access point and stations. In addition, it helps the access point to increase the downloaded throughput to accept more voice users. The process started with calculating the collision ratio (C_{Ratio}) by dividing the number of the collision over the number of successful frames as in (33). In our work, the collision was measured during 100 successful frames.

$$C_{Ratio}=\frac{Number\ of\ the\ collision(NC)}{Number\ of\ success\ sending\ frames(NSF)} \qquad (31)$$

When the C_{Ratio} is more than threshold value (Th_{val}), two procedures were taken on the access point and station separately. In the access point, contention window minimum for the voice queue was changed to become one rather than seven. In addition, the AIFSN [voice] in the access point will be set to one rather than two. While, in the stations, contention window maximum of the voice queue was increased to become 63 rather than 15 in default value. Moreover, the technique of calculating new contention window was changed at the station by multiplying the old contention window by seven rather than two. Fig. 7 describes the flowchart of our algorithm. Through our algorithm, different ranges for backoff time were created between the access point and stations, which led to enhancing the downlink throughput and increasing the capacity of the network.

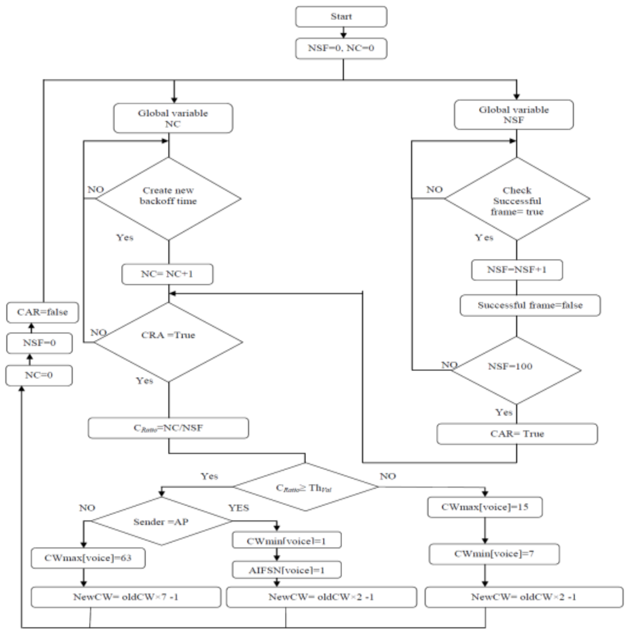

Fig. 7. Proposed algorithm to enhance the capacity of EDCA protocol

VIII. THE RESULTS OF SIMULATION AND MATHEMATICAL MODEL

A. *The Results of Default EDCA Protocol*

Our mathematical model was applied in the common scenario network, which included several wireless workstations and one access point. IEEE802.11b standard was used with 11Mbps data rate. Three types of traffic were generated, namely voice, best effort and background in which all of these traffics worked at the same time. Our mathematical model was used to analyze EDCA protocol under saturation and non-saturation conditions. Both calculate the throughput of the downlink and uplink traffic separately for different data types. The separation between uplink and down link traffic helps to find the bottleneck of the network, which prevents network to accept more voice users. In our work, we are concerned with finding the capacity of the voice users in wireless network. First, our mathematical model was applied without estimate qi. Fig. 8 shows the throughput of the voice data for the uplink traffic at the station and downlink traffic at the access point. The best effort and background throughput are shown in Fig. 9. The default EDCA protocol gave the voice traffic the highest priority. Thus, the voice recorded the highest throughput.

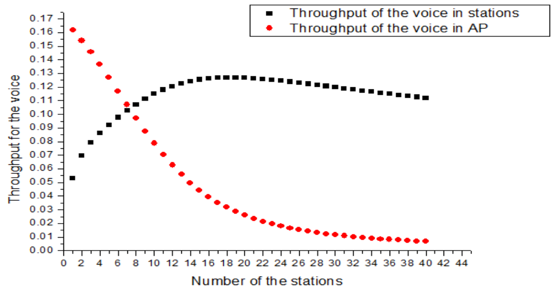

Fig. 8. Throughput of the voice traffic in stations and AP

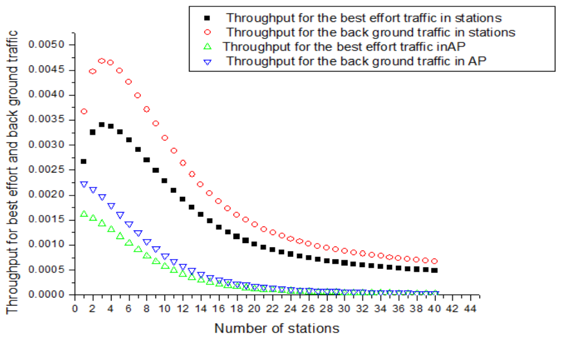

Fig. 9 Throughput of best effort and background in the stations and AP.

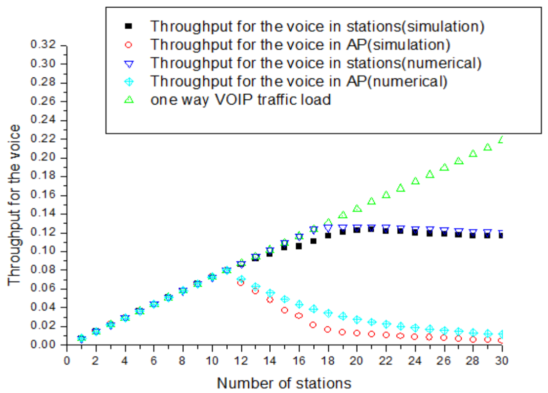

Fig. 10. Voice capacity for EDCA protocol

However, the curves for the uplink and downlink voice throughput changed when the number of the voice users increased. For instance, the downlink throughput at the access point deceased when the voice load went up. To achieve the QOS, the network must ensure that there are two ways 80kbps data rates. According to the Fig. 8, the throughput of the downlink voice traffic did not achieve the requirements after eleven nodes. Thus, this shows that the wireless network can support until eleven voice users only. In Fig. 10, the mathematical model is used with estimate *qi* for station and access point. The value of λ is 50 packet/second when G.711 codec (20ms) was used. Besides that, the OPNET simulation was used to validate the result that we got it from the mathematical model. The wireless network was designed by using OPNET simulation and default EDCA parameters and the number of stations was increased gradually. In addition, three types of data traffic were defined as in the mathematical model. Thus, the simulation results support the result we obtained from the mathematical model. Both results emphasis that the access point is the bottleneck of the wireless network, and the network can support 11 voice users with achieving QOS.

B. The Results after Applying Our Algorithm which Enhance the Capacity of EDCA Protocol

In this section, our algorithm and default EDCA protocol are discussed. The OPNET simulation was used to evaluate the performance of our algorithm. The wireless network contained fourteen wireless stations and one access point. As the previous scenarios, three types of traffic were defined; voice, best effort and background. The codec of the voice is G.711with 20ms capsulation time. In this experiment, our algorithm was used to enhance the capacity of the wireless network to tolerate more voice users. As the default EDCA protocol can support until eleven voice users, our algorithm was evaluated in terms of end-to-end delay, the packet loss percentage, throughput and the retransmission attempts.

Fig. 11 shows the end-to-end delay values for the default EDCA protocol and our algorithm. The end-to-end delay in the default EDCA protocol is 580ms for 14 voice users. This shows us that the calling failed because it did not achieve the QOS requirement. However, after applying our algorithm, the end-to-end delay decreased and became 80ms, which supported QOS.

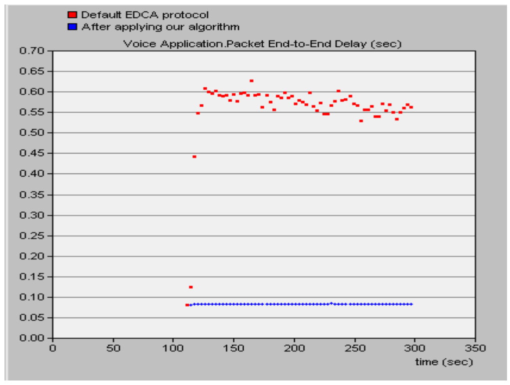

Fig. 11. End-to-end delay values for default EDCA protocol and our algorithm

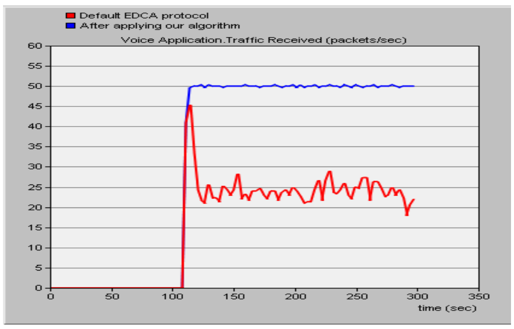

Fig. 12 packet received values for default EDCA protocol and our algorithm

The voice users in the scenario sent 50packets/second. To support QOS, the percentage of the packet loss must be less than 1%. Fig. 12 shows the received packet for the voice in the default EDCA protocol and our algorithm. According to Fig. 12, the percentage of the packet loss of our algorithm is less than 1%. On the other hand, the packet loss for the default EDCA protocol is 54%. Thus, our algorithm helps the wireless network to achieve the QOS.

Enhancing the end-to-end delay and the packet loss percentage will lead to increasing the throughput. Fig. 13 presents how our algorithm enhances the throughput. For instance, the throughput was increased by 38.3% compared with the default EDCA protocol. Our algorithm decreased the overlapping between the values of backoff time. Therefore, the collision and the retransmission attempts decreased. Fig. 14 shows the comparison between our algorithm and the default EDCA protocol in terms of the retransmission attempts.

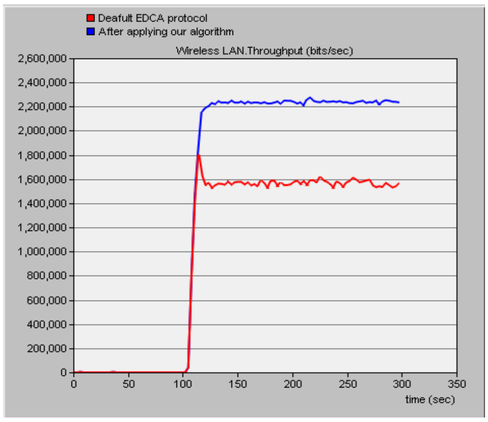

Fig. 13. Throughput of network values for the default EDCA protocol and our algorithm

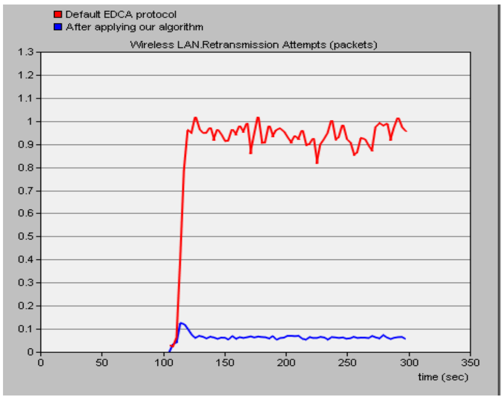

Fig. 14. Retransmissions values for the default EDCA value and our algorithm

All wireless stations in the network need access point to send and receive the data. Therefore, there was high traffic load in the access point and our algorithm gave the access point advantage to send its data faster. Setting CWmin [voice] in the access point to one rather than seven helped to create small values of backoff time counter for downlink voice traffic. In addition to that, increasing CWmax [voice] in the station to become 63 rather than 15 led to creating different ranges between stations and access point. Moreover, increasing the value of the contention window for the voice station was not enough to enhance the capacity, as it needs to increase the ranges when the random value is selected in the station. In addition multiplying the old contention window by seven rather than two led to increasing the range of selecting a random value for calculating new backoff time. Thus, the collision was avoided and the throughput of the network increased. Our algorithm was also applied in our mathematical model to verify the result that we got from OPNET simulation. Fig. 15 evaluates our algorithm mathematically and simulation. We found that the capacity of the voice user increases when applying our algorithm and the network can support fourteen voice users rather than eleven in the default EDCA protocol.

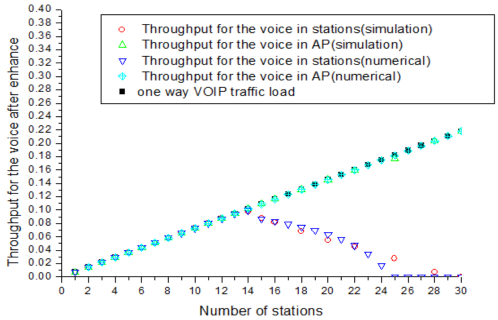

Fig. 15. The capacity after applying our algorithm

IX. Conclusion

In this paper, the capacity of wireless networks using the EDCA protocol has been presented. The default EDCA protocol has a limitation in supporting QOS for the voice users. We found that the network can serve just only 11 voice users. Increasing the number of the voice users will lead to increasing the end-to-end delay and packet loss percentage more than the QOS requirements. Besides that, using the default values of EDCA protocol contributes to increasing the collision for high priority data. In addition, the Markov chain mechanism was used to create a new mathematical model. Also, our mathematical model was used to evaluate the performance of EDCA protocol under saturation and non-saturation conditions. As well as, the new mathematical model separates between uplink and downlink throughput for different data types. On the other hand, the models in [12] and [13] calculated the throughput for all wireless networks without any difference between uplink and downlink traffics. Therefore, the separation between the uplink and downlink traffic helps to determine the bottleneck of the wireless network, and allows the researcher define different parameters between workstations and access point. Through our analysis, the access point is considered the bottleneck of the network that prevents the increase of the capacity of the wireless network, where the downlink throughput decreased when the load of the voice increased. Also, a new algorithm was proposed to enhance the capacity of EDCA protocol for supporting more voice users. The algorithm was used to create a different contention window range between access point and stations, and it increased the range by selecting random values in the station rapidly. By applying our algorithm, the throughput of the network increased by 38.3%, showing that the capacity of the network can cover 14 voice stations rather than 11 when using the default EDCA protocol. The mathematical model and the OPNET simulation were used to evaluate our new algorithm and verify the results.

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