Markov Chain Model for EDCA Protocol under Saturation and Non-Saturation Conditions

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Abstract—IEEE802.11 standards are the most popular techniques used in wireless network. Distributed Coordination Function (DCF) protocol and Enhanced Distributed Channel Access (EDCA) protocol are used in these standards. EDCA protocol is issued to get over the defect in DCF protocol as it does not support Quality of Service (QoS). The mechanism of EDCA protocol is based on dividing the data into four queues: video, voice, best-effort, and background. The priorities are distributed between queues by setting their parameters. In this paper, the Markov chain model is designed to evaluate the throughput of default EDCA protocol in saturation and non-saturation conditions. Our model is designed based on the difference between stations and Access Point (AP) to calculate the uplink and downlink throughput. In addition, the limitation of supporting voice users in EDCA protocol is presented. Moreover, the wireless networks are designed by using the Optimum Network performance (OPNET) simulation to verify the results that have been get from mathematical model. The analysis is also used to explain how the AP is considered as a bottleneck of the network and how it prevents the increase in the number of voice users.

Index Terms—DCF, EDCA, QoS, Markov chain

I. Introduction

Recently, there is an increase in the use of modern devices such as smart phones and Personal Digital Assistant (PDA). These devices include applications which need internet access to work. Therefore, it is important to provide public locations with internet. Wireless network is considered as the perfect choice to be used in public locations because it is easy to expand and covers large areas with low cost. The IEEE802.11 standards are used in the wireless network. These standards are very popular because they use unlicensed channels. In 1997, IEEE802.11a standard was issued with data rate 2Mb/s. Currently, the data rate has increased to 600Mb/s in IEEE802.11n standard [1]. IEEE802.11 (a, b, and g) standards use Distributed Coordination Function (DCF) protocol in their Medium Access Control (MAC) layer. DCF protocol has only one queue and all data enter the queue without any priority especially for data that have concentrations in delay time and packet loss percentage. Some applications such as Voice over Internet Protocol (VoIP) and video conference need to serve their data quickly. These types of data are called real time data. The success of real time data is restricted to achieve the requirements of delay time and packet loss [2]. For example, in voice data, the limitation of delay time and packet loss percentage are 150ms and 1%, respectively [3]. The DCF protocol does not provide any features for real time data, thus, it does not support Quality of Service (QoS). On the other hand, IEEE802.11e was issued to support QoS [4]. It uses Enhanced Distributed Channel Access (EDCA) protocol. EDCA protocol is based on creating different priorities between various data types. The data are divided between four queues: voice, video, best-effort, and background. The voice and video queues have more chances to access the medium than other queues [5]. This mechanism decreases delay time and dropped packets for real time data and leads to achieve QoS.

Nowadays, researchers are interested in analyzing the EDCA protocol mathematically using the Markov chain method. The researchers in [6]–[9] evaluated the throughput of EDCA protocol in saturation condition. However, the networks reach saturation after going through a non-saturation condition, because the load of data is growing up gradually. Thus, some researchers as in [10] and [11] analysed the EDCA protocol in a non-saturation condition. But all these analyses do not contain any discrimination between access point (AP) and stations and they do not calculate the uplink and downlink throughput of the network. Calculating the uplink and downlink throughput helps to determine the capacity of the network.

In this paper, a new Markov chain model is created to evaluate the throughput of EDCA protocol in saturation and non-saturation conditions. The throughput of AP and station will be calculated separately for different priorities. In addition, the limitation of supporting voice users will also be presented by using Markov chain model and OPNET simulation.

II. Distributed Coordination Function (DCF) Protocol

DCF protocol has two different methods for sending data through a medium. They are called Basic Access Method and Request To Send and Clear To Send RTS/CTS method [12]. Both methods use Carrier Sense Multiple Access/Collision Avoidance CSMA/CA mechanism. For Basic Method, the sender senses the

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medium during Distributed Inter Frame Space (DIFS) time. If the medium is still free after DIFS elapsed, the sender will transmit data through the channel, if not, the sender will postpone data transmission for a period of time, also known as backoff time. The backoff time works as a timer and the station decreases the timer until it reaches zero to try to retransmit the data again. If the backoff timer reaches zero and there is no collision, the sender will succeed in sending the data through the channel. When the data reach the destination, acknowledgement (ACK) will be sent to the sender after waiting for Short Inter Frame Space (SIFS) time. During transmission and until the ACK reaches the sender, the counter of the backoff time for other stations will be frozen. The backoff time counter will o

nly start counting when the medium becomes free. Fig. 1 describes an example of the Basic Access method [13].

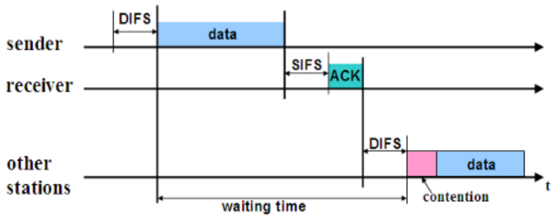

Fig. 1. Basic access method.

The backoff time counter value is calculated randomly by choosing an integer number between zero and the current Contention Window (CW). If the station fails to transmit data after backoff time has finished, the CW will increase as in (1) [14]. At the beginning of the transmission, the CW value will be set to CWmin. CWmax is the maximum value that can be reached by the CW during retransmission.

$$CW = 2 \times (\text{old } CW + 1) - 1 \quad (1)$$

However, the Basic Access Method fails to solve the hidden node problem. This problem occurs when the wireless network contains multiple overlapping ranges from different access points. Each access point uses different channel than others in the same wireless network, so the stations in different ranges cannot hear each other. Fig. 2 shows a simple example of the hidden node problem. It clearly shows how station C can reserve the medium to send data to Station B, while other stations in different ranges do not know that station B is busy at the moment [15].

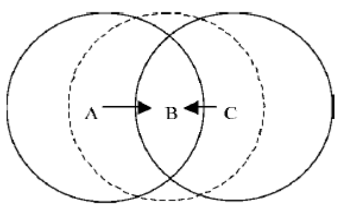

Fig. 2. Example of hidden node problem.

Because of that, RTS/CTS mechanism is issued to solve the hidden node problem [16]. In this mechanism, the sender sends RTS frame to check whether the status of the destination is ready to receive data or not. If the destination is free, it will reply with CTS frame as shown in Fig. 3. In the mean time, Network Allocation Vector (NAV) is set to inform all stations in the network about the duration time to complete the transmission. Thus, the hidden node problem is solved [17].

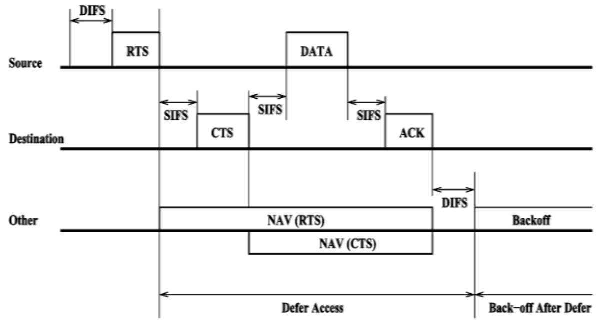

Fig. 3. RTS/CTS mechanism.

III. Enhanced Distributed Channel Access (EDCA) Protocol

The first usage of EDCA protocol was in IEEE802.11e standard to provide QoS. The EDCA mechanism is based on dividing the data that reached MAC layer into four queues. The queues are divided into voice, video, best-effort, and background traffic. Different priorities are determined for each queue. Voice queue reserves the highest priority, followed by other queues as shown in Fig. 4. The queues are separated and each of them has a special parameter, which is single DCF protocol. The priorities are defined by setting the parameter of each queue [18].

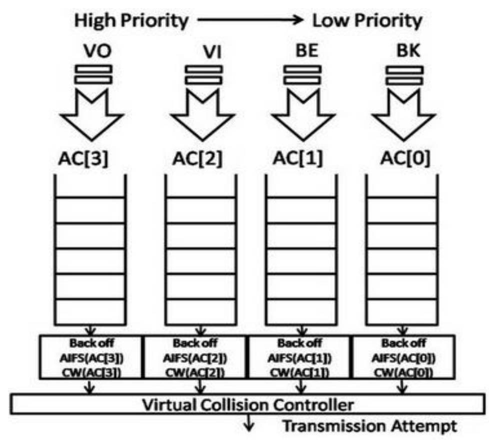

Fig. 4. EDCA access categories and priorities.

CWmin and CWmax are the two parameters used to set the priority of each queue. The result of backoff time calculation depends on the value of CW. Small values of CWmin and CWmax lead to a decrement in the value of backoff time and the queue will access the medium faster. Therefore, voice queue has the smallest values of CWmin and CWmax [19]. After the backoff timer reaches zero, the queue needs to wait for a period of time before accessing the medium. This period of time is determined by Arbitration Inter Frame Space (AIFS) parameter.

Adapting the AIFS parameter will affect the priorities. Higher priority queues must have small value of AIFS to reduce the waiting time to access the medium. AIFS time comes instead of SIFS time that is used in DCF protocol [20]. Equation 2 illustrates how to calculate AIFS.

$$AIFS[AC]=SIFS+AIFSN[AC]\times Slot \tag{2}$$

where the AIFSN value represents the number of slot time that the queue must wait. Thus, voice and video take the smallest value of AIFSN. Fig. 5 shows how AIFS parameter affects the priorities.

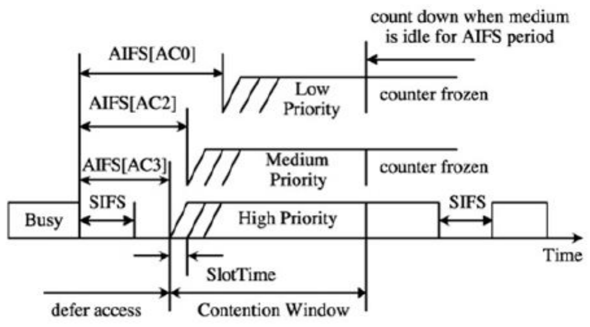

Fig. 5. How AIFS determines priorities.

The queue can reserve the medium to send one frame or more based on the value of Transmit Opportunity Length (TXOPlimt) parameter. If the value is zero, the queue can send one frame when accessing the medium, if not, this value represents the period of time for reserving the medium and the queue can send more than one frame during this time [21]. Therefore, queues that have real time data use TXOPlimt value of more than zero in order to increase the number of frames per access as shown in Table I.

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

IV. MARKOV CHAIN MODEL AND ANALYSIS

The Markov chain model is a technique used to evaluate the performance of EDCA protocol. Fig. 6 shows our version of the Markov chain model. Each state has three identifiers: *i*, *j*, and *k*. Variable *i* describes the number of access category priorities in stations and AP. The level of the state is defined using variable *j*. Increasing the value of *j* by one means that the state will move to the next level. Variable *k* is used to represent the backoff time counter. The state will move at the same level by decreasing the value of *k* by one. Each state senses the medium, the probability of sensing the medium to be busy is P_i. If the medium is busy, the state will not move and the *k* variable will be frozen. The medium will be free at ($1-P_i$) probability, thus, the *k* variable will decrease by one when the state moves to the other one at the same level. The access category tries to send data when the backoff time counter is equals to zero, meaning when *k* equals to zero. But if the medium is busy, the state will move to the next level and the CW will be duplicated. The maximum number of retransmission is *Lretry*, after that the data will be dropped.

State (*i*,-1) is used to describe a non-saturation condition. After a successful transmission, there is a probability that the queue does not have ready data to transmit: variable q_i represents this probability. Thus, q_i variable is used to control non-saturation traffic load.

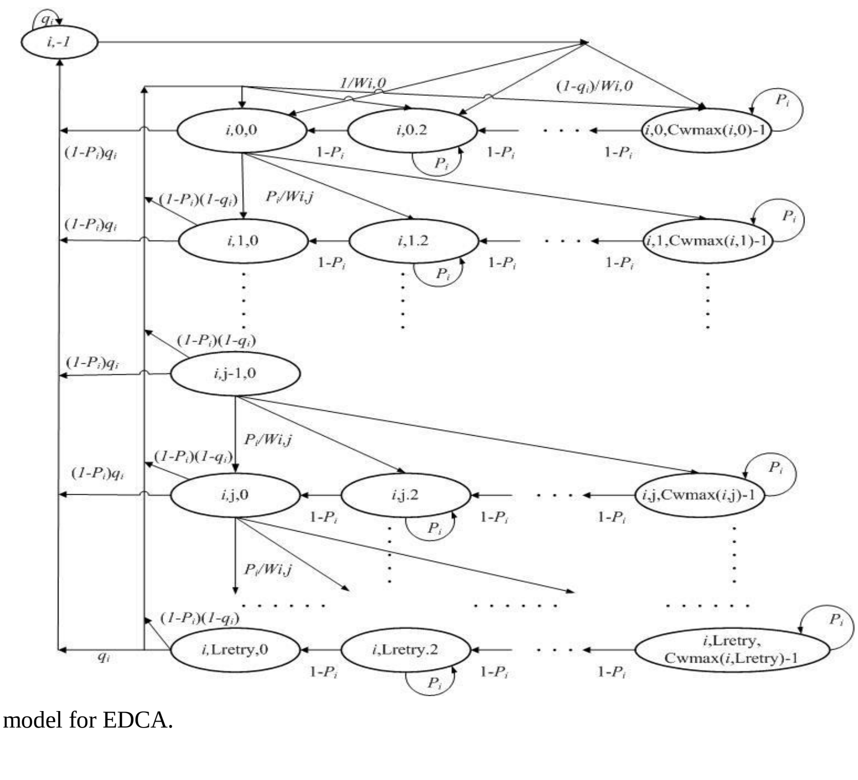

Fig. 6. Markov chain model for EDCA.

In our version of the Markov chain model, there are two types of traffic: uplink and downlink traffics. The uplink traffic flows from station to AP and the downlink traffic flows from AP to station. Therefore, $ACup[i]$ is the access category in station with priority i, and $ACdw[i]$ is the access category in the AP with priority i. The movement state probabilities can be concluded as:

$$P[(i,j,k)|(i,j,k+1)] = 1 - P_i \quad (3)$$

$$P[(i,j,k)|(i,j,k)] = P_i \quad (4)$$

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

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

$$P[(i,0,k)|(i,Lrety,0)] = \frac{(1-qi)}{CW(i,0)} \quad (7)$$

$$P[(i,-1)|(i,j,0)] = (1-P_i)qi \quad (8)$$

$$P[(i,-1)|(i,-1)] = qi \quad (9)$$

$$P[(i,0,k)|(i,-1)] = \frac{(1-qi)}{CW(i,0)} \quad (10)$$

Supposed that $b(i,j,k)$ is the steady state for the Markov chain and summation of all state probabilities equals to one, which is considered as one of the properties of the Markov chain method, thus, $b(i,0,0)$ can be derived as:

$$b(i,j,0) = P_i^j b(i,0,0) \quad (11)$$

$$b(i,j,0) = b(i,j-1,0)P_i \quad (12)$$

$$bi,-1 = \frac{qi}{(1-qi)} bi,0,0 \quad (13)$$

$$b(i,j,k) = \frac{CW(i,j)-k}{CW(i,j)} \quad \frac{b(i,j,0)}{1-P_i} \quad (14)$$

$$b(i,0,0) = \frac{1}{[\frac{qi}{(1-qi)} + \sum_{j=0}^{Lretry}[1+\frac{1}{1-P_i}\sum_{k=1}^{CW(i,j)-1}\frac{CW(i,j)-k}{CW(i,j)}]P_i^j]} \quad (15)$$

The summation for all state probabilities with k equals to zero will lead to the calculation of the probabilities of transmission for the uplink (τup_i) and downlink (τdw_i) traffics.

$$\tau up_i = \sum_{j=0}^{Lretry} b(i,j,0), in\ station$$

$$= b(i,0,0)\frac{1-pup_i^{Lretry+1}}{1-pup_i} \quad (16)$$

$$\tau dw_i = \sum_{j=0}^{Lretry} b(i,j,0), in\ AP$$

$$= b(i,0,0)\frac{1-pdw_i^{Lretry+1}}{1-pdw_i} \quad (17)$$

The probabilities of sensing that the medium is busy in station and AP are Pup_i and Pdw_i, 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:

$$Pup_i = 1 - (1-\tau up_i)^{n_i-1}(1-\tau dw_i)\times \prod_{l=0\ l\neq i}^{N-1}(1-\tau up_l)^{n_l}(1-\tau dw_l) \quad (18)$$

$$Pdw_i = 1 - (1-\tau up_i)^{n_i} \prod_{l=0\ l\neq i}^{N-1}(1-\tau up_l)^{n_l}(1-\tau dw_l) \quad (19)$$

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:

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

$$Psdw_i = \tau dw_i\,(1-\tau up_i)^{n_i}\times \prod_{l=0\ l\neq i}^{N-1}(1-\tau up_l)^{n_l}(1-\tau dw_l) \quad (21)$$

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 the calculation of the probability of success for all priorities in AP ($Psuccess\ dw$).

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

$$= (1-Pbusy)\sum_{i=0}^{N-1}\frac{n_i\,\tau up_i}{1-\tau up_i} \quad (22)$$

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

$$= (1-Pbusy)\sum_{i=0}^{N-1}\frac{\tau dw_i}{1-\tau dw_i} \quad (23)$$

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

$$Pbusy = 1 - \prod_{i=0}^{N-1}(1-\tau up_i)^{n_i}(1-\tau dw_i) \quad (24)$$

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 *si* is the throughput for AC[i]:

$$S_i = \frac{E(payload\ successful\ transmission\ time\ for\ class\ i)}{E(total\ \ time\ betwen\ two\ successive\ transmission)} \tag{25}$$

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 (*supi*) and downlink (*sdwi*). Let, δ, *TE*(*L*), *Ts*, *Tc*, 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:

$$Sup_i = \frac{Psup_i T_E(L)}{(1-Pbusy)\delta + P_{success\ up}T_s + [Pbusy - P_{successful\ all}]T_c} \tag{26}$$

$$Sdw_i = \frac{Psdw_i T_E(L)}{(1-Pbusy)\delta + P_{success\ dw}T_s + [Pbusy - P_{successful\ all}]T_c} \tag{27}$$

Ts and *Tc* values depend on the method of transmission used. In our work, we use the basic method of transmission. Fig. 7 shows the components of the frames in successful and collision states. According to the timing diagram for the basic method, *Ts* and *Tc* can be calculated as:

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

$$T_c = AIFS[i] + \left(\frac{H_p}{speed_p} + \frac{H_m + E(L) + FCS}{speed_m}\right) \tag{29}$$

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.

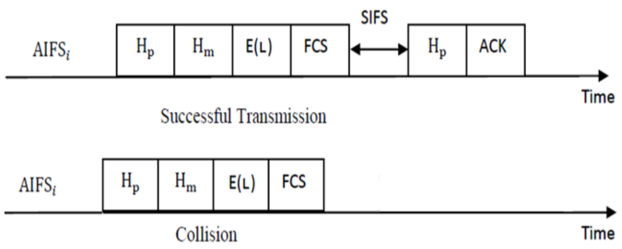

Fig. 7. Basic access method frames.

V. Result and Discussion

A common scenario is used to get the results for our version of the Markov chain model, which includes one AP and all wireless stations connected to it. Three types of data are used in all stations: voice, best-effort, and background. All of them try to send data at the same time. Our model is applied to this scenario to calculate the throughput for stations and AP for different access categories. The throughput of the stations and AP are calculated separately. Fig. 8 and Fig. 9 show the throughput of voice, best-effort, and background access categories for stations and AP. It is clear that voice has the highest throughput in stations and AP. This returns to the mechanism of EDCA protocol that gives voice the highest priority.

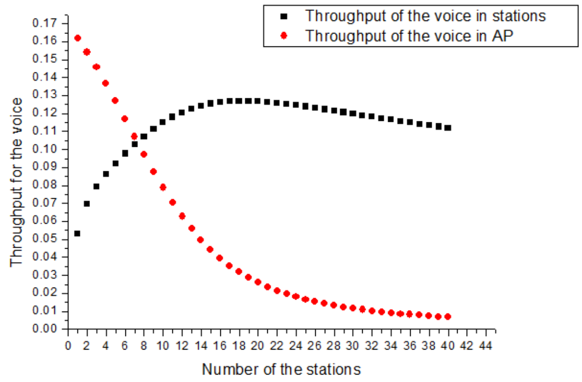

Fig. 8. Throughput of the voice traffic in stations and AP.

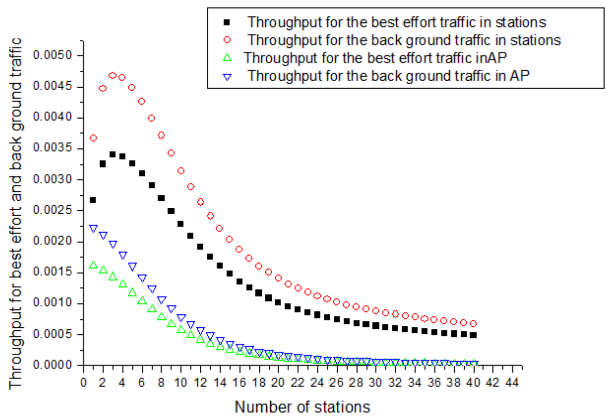

Fig. 9. Throughput of the best effort and background in the stations and AP.

However, the wireless network can cover a certain number of voice users with supporting QoS. There are constraints that prevent the wireless network from increasing its capacity. Supposed that we use 802.11b standard, which has data rate of 11Mb/s and G.711 voice codec, which has two ways 80kbps traffics [22]. The ideal capacity for the voice user can simply be calculated as (11Mbps/2 ×80kbps) and it will equal to 68 voice users. However, the capacity is much less than 68 users. Collisions which happen in the network affect on the capacity of the network. To be more accurate, the throughput of the AP is the main cause that prevents the wireless network from supporting more voice users. The AP must provide downlink traffic from AP to stations. If the wireless network has N number of users, the AP throughput must not be less than N×80kbps to support QoS.

The OPNET simulation is used to design the wireless networks. Three different traffics are defined; voice, best effort and background. All of them work at the same time. The voice traffic uses G.711 codec with capsulation time 20ms. The number of the stations is increased gradually until reach 30 stations. The uplink and downlink throughput are measured to verify the results that have been get from Markov chain model.

Fig. 10 shows the capacity of the network in the AP and stations with increasing load gradually by using mathematical model and OPNET simulation. The throughput of AP cannot provide the required download traffic after 11 voice users. Thus, the AP is considered as a bottleneck of the network and it has a negative effect on the capacity. This happens because all stations need the AP to access the network. Therefore, collisions will increase and will lead to throughput drop.

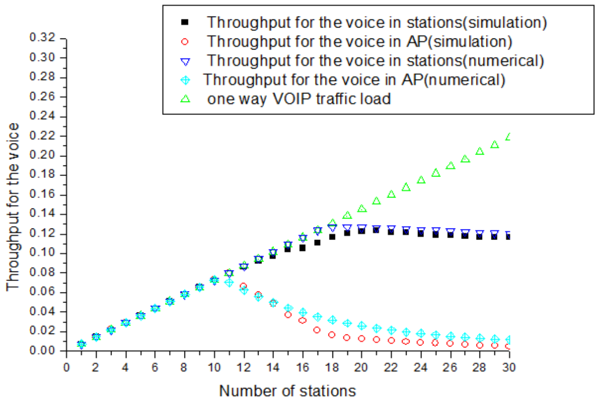

Fig. 10. Voice capacity for EDCA protocol.

VI. Conclusion

In this paper, the Markov chain model is used to evaluate the performance of the default EDCA protocol. Our method is based on describing the throughput of the network in the stations and AP separately at saturation and non-saturation conditions. Our version of the Markov chain model shows how the AP becomes the bottleneck of the network and how it prevents the increment in capacity. On the other hand, the researchers as in [10],[11] cover the non-saturation condition without calculating the uplink and downlink throughput of the network. Through our analysis, the limitation of default EDCA protocol is presented by two ways; mathematical model and OPNET simulation. The analysis describes the effect of the AP throughput on network capacity. The AP must provide the downlink traffic for the N stations in the network to achieve the QoS. That means its throughput should cover the summation of all downlink traffic for the stations in the network.

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