ADAPTIVE OPTIMIZATION OF EDCA PARAMETERS FOR QOS IN MULTIMEDIA APPLICATIONS

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ABSTRACT

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The MAC layer of the IEEE 802.11 standard defines Distributed Channel Function (DCF), that provides simple and effective medium access mechanism. However with the increasing demand in multimedia applications and higher data rates, DCF can not provide QoS requirements since 802.11 standard does not meet the need of Quality of Service (QoS) in high data rate multimedia applications like VoIP and IPTV. For this reason, necessary extensions are made to improve data rates and MAC control schemes according to the dynamically changing needs. To provide QoS for the multimedia applications, IEEE defined a new MAC mechanism, 802.11e Enhanced Distributed Channel Access (EDCA). With new access categories that enable prioritization between different traffic classes. Such prioritization provides QoS for multimedia applications by increasing the probability of higher priority class packets accessing the medium.

The purpose of this thesis is to design an adaptive optimization system in EDCA mechanism in order to provide better QoS for multimedia applications from both delay and throughput perspectives. To achieve this goal, analytical model of the DCF mechanism, already defined in the literature, is improved for the EDCA mechanism. Based on the back-off procedure in EDCA, that is represented using a Markov model, the estimates of throughput and delay are obtained by determining the probabilities in the Markov model. In order to validate the estimated throughput and delay, real time simulations in OPNET are carried out and the correlation between analytical results and simulation results are included in the thesis. Based on the validated analytical model, a new algorithm is proposed in order to maximize throughput while decreasing delay and minimizing collision probability for a higher access category.

ÖZET

ÇOKLU ORTAM UYGULAMALARINDA SERVİS KALİTESİ İÇİN EDCA PARAMETRELERİNİN UYARLANIR OPTİMİZASYONU

IEEE 802.11 standardı MAC katmanında, basit ve etkili bir ortam erişim mekanizması olan DCF'i (Dağıtık Kanal Erişimi) tanımlamıştır, fakat günümüzde gittikçe artan çoklu ortam uygulamaları ve yüksek veri hızı talebi karşısında, DCF gerekli servis kalitesini sağlayamamaktadır. Buna sebep olarak, DCF mekanizmasının VoIP ve IPTV gibi yüksek veri hızlı, çoklu ortam uygulamalarının gerektirdiği servis kalitesini sağlayacak bir yapıda tasarlanmamış olması gösterilebilir. Bu sebepten dolayı, veri hızları ve ortam kontrolü açısından gerekli geliştirmeler IEEE 802.11e standardı ile tanımlanmıştır. IEEE 802.11 standardının tanımladığı ortam erişim kontrol mekanizması EDCA (Geliştirilmiş Dağıtık Kanal Erişimi), farklı trafik türleri arasında öncelik sıralaması yaparak erişim kategorileri tanımlamış ve böylece çoklu ortam uygulamaları için gerekli servis kalitesini sağlamıştır.

Bu tezin amacı, EDCA mekanizmasının daha verimli kullanılabilmesi ve çoklu ortam uygulamaları icin gerekli servis kalitesinin gecikme süresi ve çıkan iş oranı açısından sağlanması için uyarlanır optimizasyon sistemi tasarlanmasıdır. Bu amaç doğrultusunda, DCF mekanizmasının literatürde bulunan analitik modeli, EDCA mekanizması icin uyarlanmıştır. EDCA mekanızmasındaki geri çekme prosedürü bir Markov modeli olarak tasarlanmış ve bu model yardımıyla gerekli olasılıklar ve kestirimler elde edilip, gecikme süresi ve iş çıkarma oranı açısından sistemin kestirimi yapılmıştır. Bu yapılmış olan kestirimin doğruluğunun sınanması için OPNET programı kullanılarak sistemin davranışı ile bizim yaptığımız analizin ilişkisi incelenmiş ve elde edilen doğrulanmış analitik model kullanılarak çoklu ortam uygulamaları için gecikme süresini kısaltırken, iş çıkarma oranını artıran uyarlanır optimizasyon algoritması tasarlanmıştır.

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$b_{i,k}$	The probability of being in state (i,k) as time goes to infinity in Bianchi's model
$b_{i,j,k}$	The state of the back-off timer in steady state for priority i after j collisions
$B_{drop,i}$	The average number of back-off slot times for a dropped packet without considering the other transmissions causing this back-off counter to pause for i.th class dropped packets
$B_{success,i}$	The average number of back-off slot times for a successful packet without considering the other transmissions causing this back-off counter to pause for i.th class dropped packets
$CW_{max,(i,j)}$	Maximum contention window size for the i.th priority after j collisions
D_i	The average delay for the i.th class traffic
$D_{dropped,i}$	The delay caused from dropped packets for i.th class
$D_{success,i}$	The delay caused from successful packets for i.th class
$E[P]$	The time needed for transmission of payload in Bianchi's model
$f_{pause,i}$	Number of slot times have to pass for a transmission to happen from any station for i.th class
H	The time elapsed for transmission of MAC and Physical Header in Bianchi's model
i	The priority class
j	The number of collision suffered
k	The back-off counter value
$L_{Retry,i}$	Number of retransmissions allowed for i.th class
$N_{back-off}$	Number of back-off slots
$N_{drop,i}$	Number of times that the back-off counter has to be paused for i.th class dropped packets
N_{idle}	The average number of idle slot times in Bianchi's model
$N_{success,i}$	Number of times that the back-off counter has to be paused for i.th class successful packets

$N_{Retry,i}$	The number of retransmissions before transmitting successfully for i.th class
p	The collision probability according to the Bianchi's model
$P_{tr,i}$	The probability that a station in i.th class transmits in a given slot time
P_i	The probability that a station in i.th class senses the medium busy in a slot time
P_{busy}	The probability that the medium is busy in a given slot time
P_{coll}	The probability of a collision in a slot time
$P_{drop,i}$	The probability of dropping a frame for i.th class
$P_{success}$	The probability of a successful transmission in a slot time
$P_{success,i}$	The probability of a successful transmission in a slot time for i.th class
$P_{transmitted,i}$	The probability of transmitting a frame for i.th class
P_s	The probability that a successful transmission occurs in a selected slot time in Bianchi's model
P_{tr}	The probability that there is at least one transmission on the channel at selected slot time in Bianchi's model
S	The throughput estimation in Bianchi's model
T_{ACK}	Time required for transmission ACK packet
$T_{back-off}$	The average back-off timer for a successful packet while accessing the medium
T_c	Time needed for unsuccessful transmission in Bianchi's model
T_{coll}	The time elapses for an unsuccessful transmission
T_{MAC}	Time required for transmission of MAC header
$T_{Payload}$	Time required for transmission of payload
T_{PHY}	Time required for transmission of physical header
T_s	Time needed for successful transmission in Bianchi's model
T_{slot}	The length of a slot time
$T_{success}$	The time elapses for a successful transmission

δ	The propagation delay
σ	Length of a slot time in Bianchi's model
τ	The probability that a station transmits in a given slot time in Bianchi's model
τ_i	The probability that a station transmits in a given slot time in EDCA model
τ_{opt}	Optimum transmission probability that provides maximum throughput
AC	Access Category
$AIFS$	Arbitrary Interframe Space
$CSMA/CA$	The Carrier Sense Multiple Access with Collision Avoidance
CW	Contention Window
DCF	Distributed Coordination Function
$DIFS$	DCF Interframe Space
$EDCA$	Enhanced Distributed Channel Access
MAC	Medium Access Control
$PIFS$	PCF Interframe Space
QoS	Quality of Service
$SIFS$	Short Interframe Space

1. INTRODUCTION

The purpose of this thesis is to develop up a model in order to estimate the throughput and delay for a given wireless LAN system and to optimize the overall system from both throughput and delay perspectives for wireless video transmission.

Considering the increase in the usage of wireless communication in every part of the life, the end-user solutions became very important in recent years. During the last few years, IEEE 802.11 wireless LAN has become a dominant technology for the indoor broadband wireless networking. IEEE 802.11 has been deployed extensively in many different environments. Especially by supporting high-speed multimedia services and providing high data rates, IEEE 802.11 standard has found widespread deployment.

The growth of the multimedia applications increased the requirement of Quality of Service support from delay, jitter and bandwidth perspectives. However, the IEEE 802.11 standard was designed for data transmission and the MAC layer of the 802.11 systems are defined for lower data rates and do not provide QoS for multimedia applications. 802.11 does not consider the need of Quality of Service (QoS) in high data rate multimedia applications like VoIP and IPTV. For this reason, necessary extensions are made to improve data rates and MAC control schemes according to the dynamically changing needs.

To provide the required QoS for multimedia applications, IEEE defined a new standard 802.11e. This standard defines new access mechanisms which enable prioritization between different traffic classes. This prioritization provides QoS for multimedia applications by increasing the probability of higher priority class packets accessing the medium. Therefore the packets with higher priority have lower delay and higher throughput compared to other traffic classes.

1.1. IEEE 802.11 DCF

In a medium without a control mechanism and with multiple stations communicating, there is a high probability that transmissions from these stations will collide. In order to avoid possible collisions, the 802.11 standard defines access mechanisms. The fundamental access mechanism in IEEE 802.11 is Distributed Coordination Function (DCF), which is also called carrier sense multiple access with collision avoidance (CSMA/CA). In this mechanism, each station implements DCF in order to access medium. The operation of Distributed Coordination Function is given in Figure 1.1.

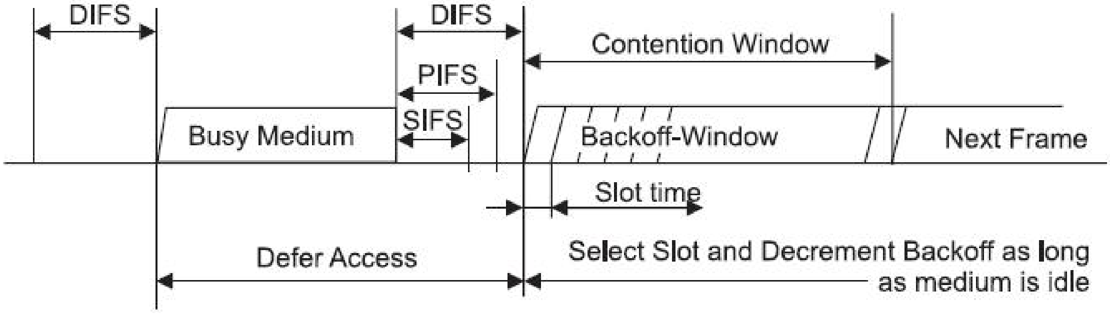

Figure 1.1. IEEE 802.11 DCF Mechanism [1]

In DCF, each station senses the medium in order to determine whether there is a transmission on the medium or not. If the medium is sensed busy, the station waits until the channel is idle. After finding the channel idle, all stations waits for a predefined interval called DCF interframe space (DIFS). Considering that multiple stations trying to access the medium, a random time interval is selected by each station. This random interval is called back-off and the station having the smallest back-off timer value will access the medium. On the other hand, when the station with the smallest back-off time value accesses the medium, other stations sense the medium and pause their back-off timers and store the value of the current back-off timer value in order to use in the next transmission attempt.

The random back-off timer values have to be selected from an interval which is defined. At the first transmission attempt, the station selects its back-off timer value from the interval 0 to CW_{min}. If the transmission attempt is unsuccessful due to a collision, the contention window (CW) size is multiplied by 2 in order to decrease the collision probability. Therefore the collision probability for the next transmission attempt will be lower. However this increase in the CW sizes is limited by a maximum value (CW_{max}). After reaching CW_{max}, the CW size is not increased anymore. The exponential increase in CW sizes in DCF mechanism is given in Figure 1.2

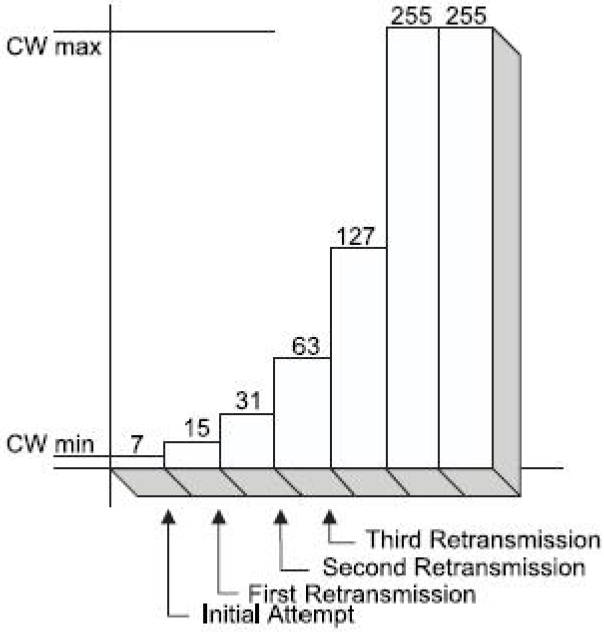

Figure 1.2. The exponential increase in CW sizes in DCF mechanism

DCF mechanism is quite effective in avoiding the collisions in the medium, but on the other hand there is no prioritization defined in this standard. For this reason, QoS required multimedia applications and less important traffic packets access the medium with the same parameters. Considering the delay and throughput requirements, DCF is insufficient while supporting multimedia applications such as VoIP and IPTV. For this reason, a new standard 802.11e is defined in order to provide QoS for multimedia applications.

1.2. IEEE 802.11e EDCA

In order to provide differentiated and distributed channel accesss, a new mechanism called Enhanced Distributed Channel Access (EDCA) is defined in the 802.11e standard. In EDCA, 8 different frame priorities and 4 access categories are defined. The priorities and access categories are given in Table 1.1

Table 1.1. Priorities and Access Categories Defined in EDCA

Priority	User Priority	Access Category (AC)	Description
Lowest	1	AC_{BK}	Background
-	2	AC_{BK}	Background
-	0	AC_{BE}	Best Effort
-	3	AC_{VI}	Video
-	4	AC_{VI}	Video
-	5	AC_{VI}	Video
-	6	AC_{VO}	Voice
Highest	7	AC_{VO}	Voice

As we have stated in the previous section, the CW sizes are very important parameters while accessing the medium. For this reason, when defining the priorities, different contention window sizes and interframe spaces are used. By defining the parameters given in Table 1.2, it is possible to prioritize a traffic class and satisfy the QoS requirements for that traffic class. As can be seen from the Table 1.2, the expected back-off sizes for multimedia traffics are defined lower, therefore the chance of video and voice frames accessing the medium is higher which results higher throughput and lower delay.

Table 1.2. Default Access Parameters in EDCA

AC	CW_{min}	CW_{max}	AIFS
AC_{BK}	aCW_{min}	aCW_{max}	7
AC_{BE}	aCW_{min}	aCW_{max}	3
AC_{VI}	$(aCW_{min}+1)/2-1$	aCW_{min}	2
AC_{VO}	$(aCW_{min}+1)/4-1$	$(aCW_{min}+1)/2-1$	2

When a packet arrives to the MAC layer, it carries a specific priority value and by using this priority value, the packet is enqueued in the specified queue with defined access parameters. The back-off procedure is used according to the default EDCA parameters or the parameters which are transmitted in the beacon frame from access point.

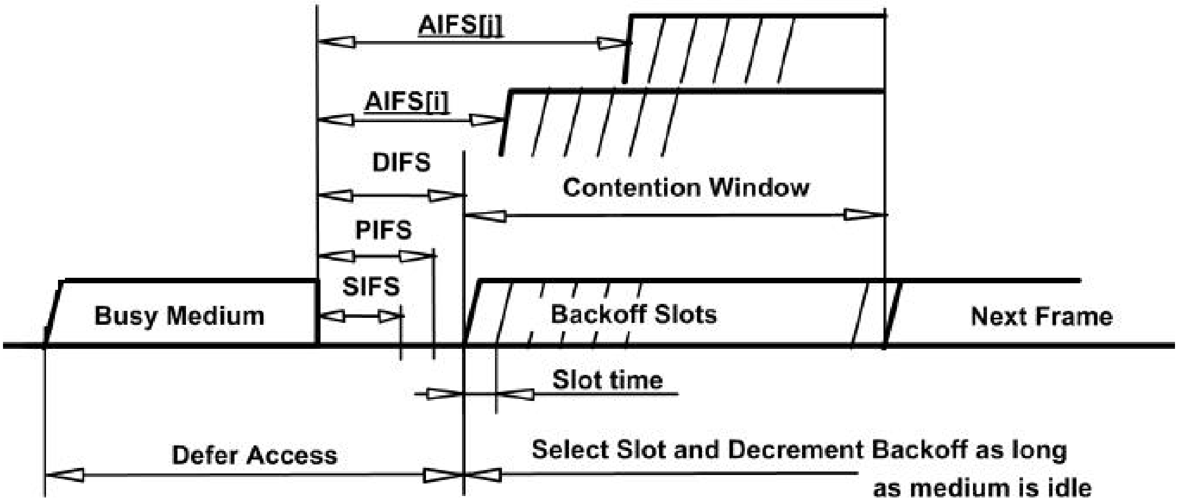

Figure 1.3. IFS defined in EDCA mechanism [2]

The time interval between frames is called the IFS and different IFSs are defined to provide priority levels for access to the wireless medium. A new parameter defined in EDCA mechanism is Arbitrary Interframe Space (AIFS). As we have explained in the DCF mechanism, the station starts back-off procedure when the medium is idle for DIFS duration. In order to prioritize the access categories, different length of AIFSs are defined as seen from in Table 1.2. The video and voice frames gain higher priority by using lower AIFSs compared to background or best effort traffic.

1.3. Literature Review

While there are several studies on the MAC layer of 802.11 and the most celebrated study about analyzing the performance of IEEE 802.11 DCF is made by Bianchi [3], [4]. In these studies, the back-off procedure is modeled as a Markov model assuming that the probability of a collision is independent of the collisions encountered. By modeling the mechanism as a Markov chain, the probabilities of successful and collided transmissions are obtained in order to obtain a throughput estimate. Because of forming a basis to this thesis, Bianchi's articles are investigated in a chapter in this thesis.

In [3],Bianchi analyzes MAC performance, but he does not take into account the retry limit or the delay performance of the DCF. Moreover according to the model in [3], the station in back-off does not pause its back-off timer until transmission is over.

Ziouva's study [5] investigates the theoretical performance of DCF by improving Bianchi's model by adding the case of pausing the back-off timer when a transmission from another station begins. This improvement in the DCF model requires a new Markov model including the probability of a pause during back-off. On the other hand, the retry limit is ignored in this study.

Chatzimisios et.al.[6], [7] extended Bianchi's model in order to find a delay estimate using the probabilities obtained from [3]. Chatzimisios et.al.[7] calculate the average number of back-off slot times and also take into account the number of times the back-off timer is paused due to transmissions from other stations. On the other hand, the pause in the back-off timer is not considered in the throughput analysis. Furthermore the authors of [6] and [7] do not consider retry limit for a packet enqueued.

When a packet encounters multiple collisions the packet is dropped after reaching a predetermined number of collisions. This mechanism is included in the analysis of the studies of Chatzimisios et.al.[8] and Wu et.al.[9]. This improvement of the model required recalculating the probabilities and extending delay analysis. The throughput and delay for the mechanism for different retry limit values are investigated in these studies.

The data rate and the throughput do not have a linear relationship because the back-off time before accessing the medium and DIFS length are independent of the data rate. For this reason, the average time before accessing the medium is constant regardless of the data rate. The effect of this constant time before accessing the medium increases as we increase our data rate. This relationship is investigated in [10] from both throughput and delay limits perspectives.

With the increasing demand for high data rate multimedia applications, the need for QoS arose. In order to provide the required QoS for the multimedia traffic, prioritization while accessing the medium is needed. To this end Xiao [11] and Deng et.al. [12] purposed priority schemes by modifying retry limits, minimum and maximum contention window sizes. With these updates they were able to show that by modifying these parameters, providing QoS for important traffic streams were possible. In addition they built up a basis for the upcoming EDCA mechanism.

After the introduction of the IEEE 802.11e, Mangold et.al.[13] and Choi et.al.[14] compared the legacy DCF and EDCA from supporting QoS point of view and also the performance evaluation according to the CW sizes and AIFS sizes. According to the priority classes defined in EDCA, the Markov models for DCF are improved for the EDCA standard in [15], [16], and new performance analysis for throughput and delay are made in these studies. These studies are successful in improving the DCF model for the EDCA while calculating the throughput, back-off time elapsed for a collided transmission and the delay resulting from the dropped packets are ignored in these studies.

In order to develop an adaptive system for throughput optimization, in [17] an algorithm is proposed. In this algorithm, the best effort data traffic parameters are dynamically decreased in order to optimize multimedia traffic. The chance of best effort traffic is decreased continuously in order to provide the required throughput for multimedia. But in this study optimization of overall traffic for a given scenario is not considered. The effect of controlling collision probability in order to increase total throughput is also ignored.

Fallah et.al. [19] and Ksentini et.al.[18] examined the video transmission performance of the contention based solutions for 802.11e. The effect of prioritization both among different traffic classes and among video packets with different priorities are investigated in these studies.

While there is a vast amount of literature on performance analysis of 802.11 medium access mechanisms, in this thesis analysis of the performance of the EDCA mechanism from the throughput, delay and collision probability point of view is made and adaptive mechanism according to the results of the model developed is included in order to provide a better QoS for multimedia applications.

The organization of this thesis is as follows. In chapter 2, the article of Bianchi [3] is investigated and based on the model in [3], a new model for EDCA mechanism is made in chapter 3. In chapter 3, the throughput and delay estimations is made according to the model developed. In chapter 4, the EDCA model estimates is compared with the real time simulation results in order to validate the model. Finally in chapter 5, the optimization algorithm that provides better QoS for multimedia applications is proposed.

2. PERFORMANCE ANALYSIS OF IEEE 802.11 DCF

The purpose of this thesis is to analyze the performance of the EDCA mechanism from the throughput, delay and collision probability point of view and to develop an adaptive mechanism according to the results of the model developed. In order to form a basis for our EDCA model, we shall first discuss Giuseppe Bianchi's article [3].

In [3] Bianchi studies the behavior of a single station with a Markov model in order to obtain the probability that a station transmits in a given time. By using the probability that a station transmits in a given slot time (τ), the probability of collided and successful transmission occurring in a slot time are calculated and throughput analysis is made by using these probabilities.

Let us consider n stations that are trying to access the channel where all of them have packets to transmit and these stations have to use back-off procedure to access the medium. The key assumption of Bianchi's study is that, at each transmission, the collision probability for a transmitted packet is constant and is independent of the number of the collision that has already been encountered. By using this fact, it is possible to model the back-off procedure of a station as a Markov chain. This assumption becomes more valid as the number of the stations in the medium and the contention window sizes increase. The Markov model of Bianchi's DCF analysis is given in Figure 2.1.

The i state in the equations (2.1)-(2.20) represents the number of collision encountered. When a station encounters a collision, the new CW interval is calculated by multiplying the maximum window size of that state (W_i) by 2 until it reaches a number of CW increase limit (m).

for $i \leq m$;

$$W_i = 2^i W_0 \tag{2.1}$$

for $i > m$;

$$W_i = 2^m W_0 \tag{2.2}$$

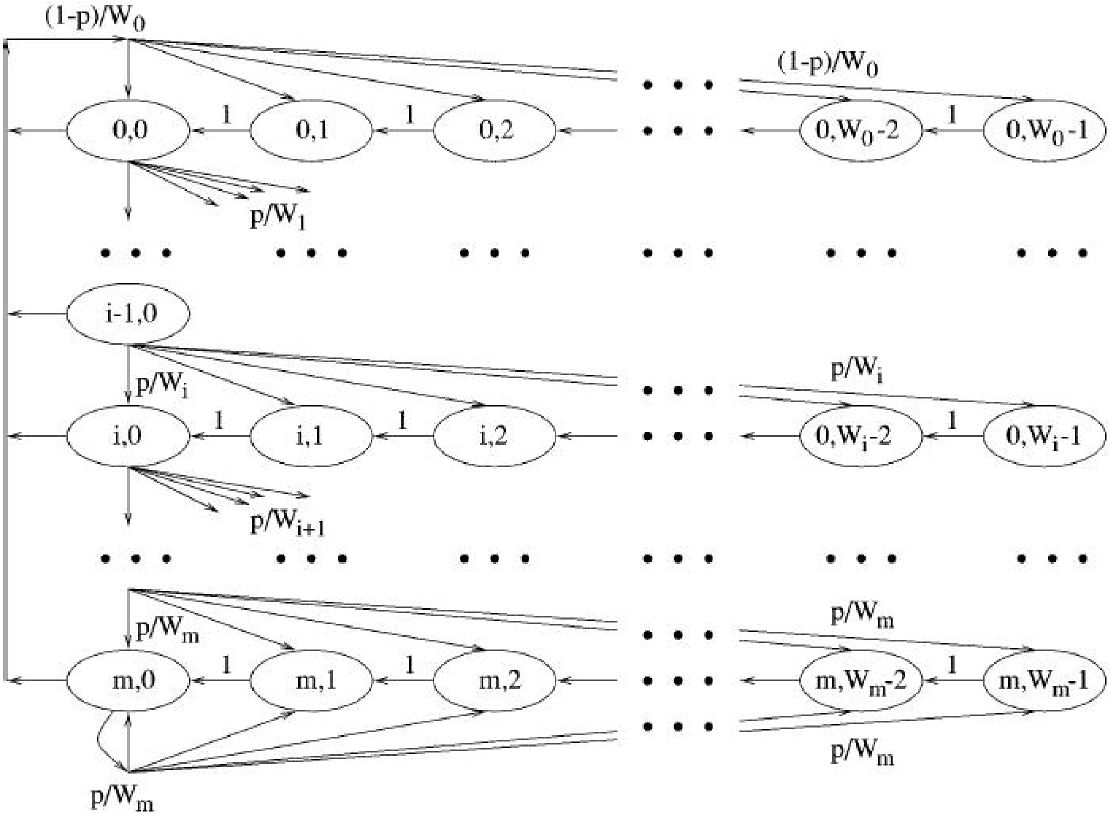

Figure 2.1. Bianchi's Markov model for analytical analysis of DCF

The k in the equations equations (2.1)-(2.20) represents the back-off timer value at the given time slot. The slot time decreases until it reaches 0 and after that the packet is transmitted into the medium. In the classic operation of the DCF mechanism, when a transmission from another station is sensed, then the back-off timer is paused until that transmission is over. But in Bianchi's study, this fact is ignored. The transition probability for a decreasing back-off timer for each slot time is always 1.

Let $P(i,k|i,k+1)$ represents transition from (i,k+1) to (i,k) and $k\epsilon(0,W_i-2)$, $i\epsilon(0,m)$;

$$P(i,k|i,k+1) = 1 \tag{2.3}$$

When the back-off timer reaches 0, the station immediately starts transmission. If there are no stations whose back-off timer reaches 0 at the same slot time, then the packet will be successfully transmitted to the receiver and the contention window sizes will be initialized for the new packet to be transmitted.

for $k\epsilon(0,W_0-1)$ $i\epsilon(0,m)$;

$$P(0,k|i,0) = \frac{(1-p)}{W_0} \tag{2.4}$$

On the other hand, in case more than one station's back-off counter reaches to 0, then there will be a collision resulting a transition from state i to state $i+1$ which will cause an increase in the maximum contention window size for next transmission attempt until the number of this transition reaches to m.

for $k\epsilon(0,W_i-1)$ and $i\epsilon(1,m)$;

$$P(i,k|i-1,0) = \frac{p}{W_i} \tag{2.5}$$

If the number of collisions reaches a predefined value of m, then the W_i size is no more increased and the new transmission attempts are made with the back-off counter,which is selected from the interval between 0 and $2^m W_0$. These retransmission attempts are made without a limit, but there is a limit defined in the standard DCF operation for a packet retransmission and if the packet reaches this limit, the packet is dropped. In Bianchi's model, this fact is ignored. In the (2.6) it is stated that, back-off counter values are selected from the interval $(0, W_m - 1)$ in case of a collision.

Given that $k \epsilon (0, W_m - 1)$;

$$P(m, k|m, 0) = \frac{p}{W_m} \tag{2.6}$$

2.1. Steady State Probabilities of DCF Model

After determining the state transition probabilities, we are able to determine the steady state probabilities. By using these steady state probabilities, we will be able to determine the behavior of the system. The $b_{i,k}$ represents the probability of being in state (i,k) as time goes to infinity.

If we define $b_{i,k} = lim_{t\rightarrow\infty} P(i, k)$ for $(0 \leq i \leq m)$ and $(0 \leq k \leq W_i - 1)$, we obtain the following equations for steady state:

$$b_{i-1,0}p = b_{i,0} = p^i b_{0,0} \tag{2.7}$$

$$b_{m-1,0}p = (1 - p)b_{m,0} \Rightarrow b_{m,0} = \frac{p^m}{1 - p} b_{0,0} \tag{2.8}$$

According to the chain regularities for $(0 \leq k \leq W_i - 1)$;

for i=0;

$$b_{i,k} = \frac{W_i - k}{W_i}(1-p)\sum_{j=0}^{m} b_{j,0} \tag{2.9}$$

for $0 < i < m$;

$$b_{i,k} = \frac{W_i - k}{W_i} p b_{i-1,0} \tag{2.10}$$

for i=m;

$$b_{i,k} = \frac{W_i - k}{W_i} p(b_{m-1,0} + b - m, 0) \tag{2.11}$$

Using the fact that $\sum_{i=0}^{m} b_{i,0} = b_{0,0}\frac{1}{1-p}$ and Equation(2.8), we can express $b_{i,k}$ as;

$$b_{i,k} = \frac{W_i - k}{W_i} b_{i,0} \tag{2.12}$$

As can be seen from the equations so far, it is possible to express all $b_{i,k}$ probabilities as functions of $b_{0,0}$ and p. Also considering that the sum of the all state probabilities is 1, we can obtain (2.13).

$$1 = \sum_{i=0}^{m}\sum_{k=0}^{W_i-1} b_{i,k} = \sum_{i=0}^{m} b_{i,0} \sum_{k=0}^{W_i-1} \frac{W_i - k}{W_i} = \sum_{i=0}^{m} b_{i,0}\frac{W_i+1}{2} = \sum_{i=0}^{m} b_{i,0}\frac{2^i W_0 + 1}{2} \tag{2.13}$$

$$1 = \sum_{i=0}^{m-1} p^i b_{0,0}\frac{2^i W_i + 1}{2} + \frac{p^m}{1-p} b_{0,0}\frac{W_m + 1}{2} \tag{2.14}$$

After expressing $b_{0,0}$ as a function of p,m and W_0, we can now derive the probability that a station transmits in a given slot time (τ). A station starts a transmission when its back-off counter reaches to 0, for this reason, τ is equal to the sum of all $b_{i,0}$ probabilities:

$$\tau = \sum_{i=0}^{m} b_{i,0} = \frac{p^m}{1-p} b_{0,0} + \sum_{i=0}^{m-1} p^i b_{0,0} \tag{2.15}$$

In (2.15), it is stated that the transmission probability of a station in a random slot time (τ) depends on the collision probability (p), which is still unknown. In order to find p, we should consider that a station encounters a collision when another station in the medium starts transmitting at the same time slot.

Assume that we have n stations in the medium and if we find the probability of (n-1) stations not transmitting in a slot time and subtract that probability from 1, then we will obtain the collision probability as can be seen in the (2.16). Using Equations (2.15) and (2.16), it is possible to find a unique solution for both τ and p:

$$p = 1 - (1-\tau)^{n-1} \tag{2.16}$$

2.2. Throughput Estimation for DCF Model

Throughput is defined as the fraction of time channel is used to successfully transmit payload bits. In order to compute the throughput, we should analyze what can happen in a slot time.

P_{tr} is defined as the probability that at least one station is transmitting in a selected slot time in a medium with n stations. If we compute the probability of no stations transmitting in that slot time and subtract this probability from 1, then we can obtain the probability that at least one station is transmitting.

$$P_{tr} = 1 - (1 - \tau)^n \tag{2.17}$$

After calculating the probability of a transmission into the medium in a slot time, we need to define new probabilities for successful and collided transmissions. A transmission can be successful if any other station does not transmit on the channel. We find the probability of successful transmission for a station in a selected slot, by multiplying the probability of a station transmits in a slot time (τ), with the probability of no other stations from $n-1$ stations in the medium do not transmit in the slot time ($(1-\tau)^{n-1}$) conditioned on the fact that at least one station transmits (P_{tr}).

Considering that there are n stations in the medium, we can find the total probability of having a successful transmission in a selected slot time as in (2.18).

$$P_s = \frac{n\tau(1-\tau)^{n-1}}{P_{tr}} \tag{2.18}$$

We have mentioned that the throughput is the fraction of the time channel is used in order to transmit payload. By using the probabilities we have found so far, we are now able to obtain an expression for our throughput estimation as:

$$S = \frac{P_s P_{tr} E[P]}{(1-P_{tr})\sigma + P_{tr} P_s T_s + P_{tr}(1-P_s)T_c} \tag{2.19}$$

In (2.19), E[P] represents the average length of the payload and $P_s P_{tr} E[P]$ is the fraction of the time used for transmitting the payload. $(1 - P_{tr})\sigma$ in the equation is the idle slot time considering that with the probability $(1-P_{tr})$ the slot time is idle. On the other hand, if there is a transmission, it can be a successful one with the probability P_s or it can encounter a collision with probability $(1 - P_s)$. Therefore the expected time elapsed during transmission, whether it is successful or not, is $P_{tr} P_s T_s + P_{tr}(1 - P_s)T_c$ where T_s and T_c are time elapsed during a successful transmission and unsuccessful transmission, respectively. The definition of T_s and T_c is given below:

$$T_s = H + E[P] + SIFS + \delta + ACK + DIFS + \delta \tag{2.20}$$

$$T_c = H + E[P] + DIFS + \delta \tag{2.21}$$

In (2.20) and (2.21), H denotes the time needed for transmission of MAC and Physical headers and δ is the propagation delay.

After determining the probabilities and throughput estimate, now we are able to see how throughput is affected from the change in transmission probability (τ), initial contention window sizes (W_0) and number of terminals (n).

2.3. Graphical Results of Throughput Estimation

As can be seen from (2.18) and (2.17), P_{tr} and P_s depend on τ. Considering that P_s and P_{tr} are forming the saturation throughput, the relationship between τ and throughput is investigated in Figure 2.2.

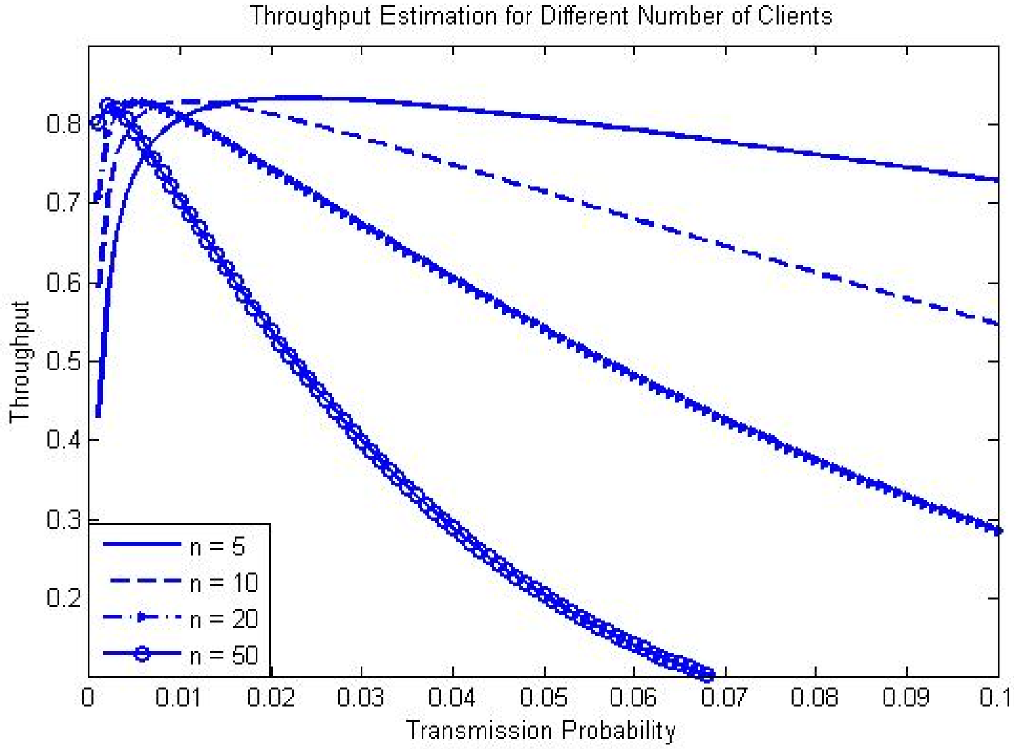

Figure 2.2. The throughput versus transmission probability τ

Figure 2.2 shows that there is an optimum τ for different number of clients. As the number of the stations in the medium increases, the optimum τ value providing maximum throughput is becoming smaller. As we have found in (2.15), the τ value depends on the initial window size and number of transition between states.

As the number of stations increases, the probability of collision increases, therefore it will be a suitable approach to enlarge the contention window size interval in order to avoid collisions. In order to observe the optimum initial window size, the relationship between throughput and initial contention window size is investigated in Figure 2.3.

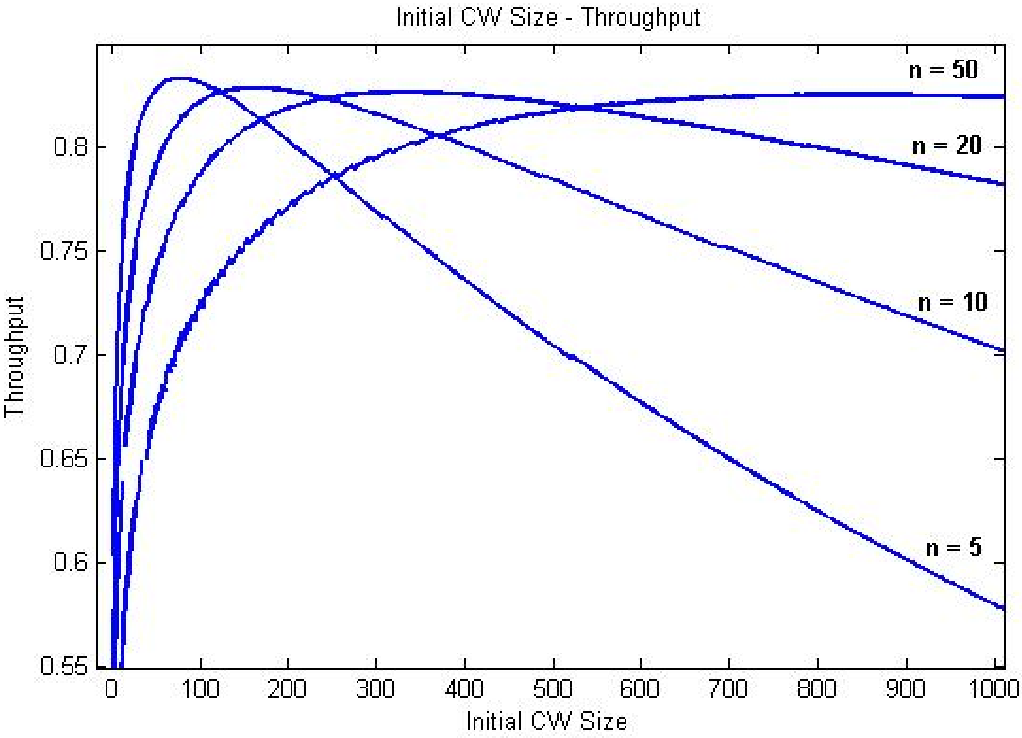

Figure 2.3. The throughput versus initial window size

Figure 2.3 shows that for each number of stations in the medium, there is a different optimum window size. The collision probability decreases with the increasing back-off time interval but on the other hand, as the back-off window sizes increase, the idle time will increase in the medium as well, which causes the throughput to decrease. So there is a need to re-define the contention window size according to the change in the number of the stations.

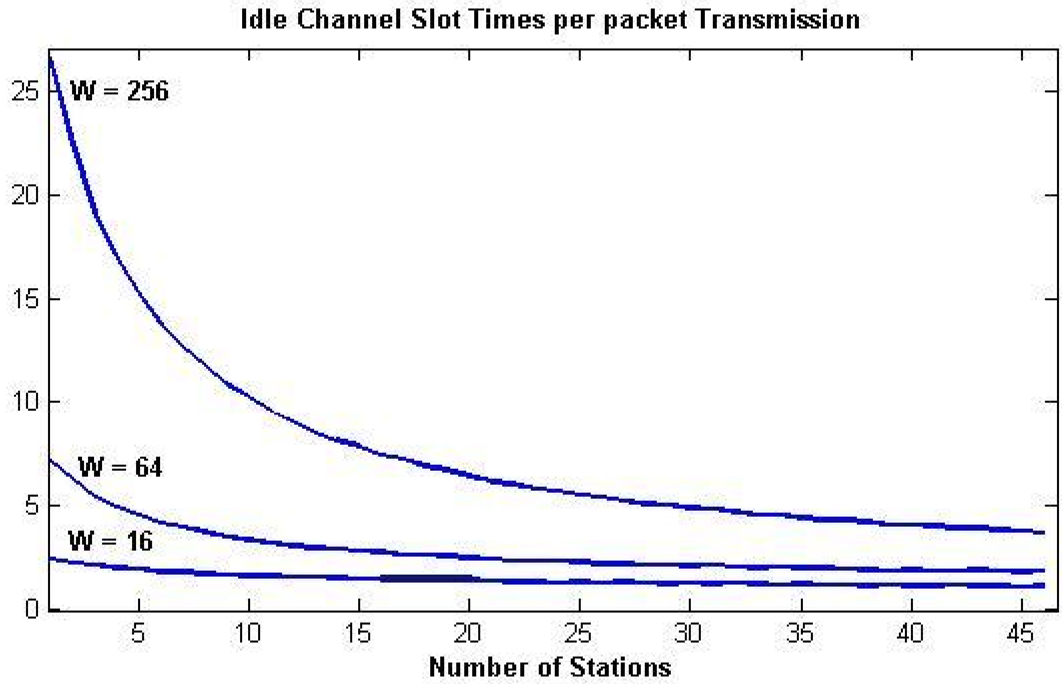

Figure 2.4. The average number of idle slot times per successful transmission

$$N_{idle} = \frac{1 - P_{tr}}{P_{tr} P_s} \tag{2.22}$$

The number of slot times spent on the channel in order to have a successful transmission is $1/(P_{tr}P_s)$ and a fraction of $(1 - P_{tr})$ is idle. The average number of idle slot times can be found from (2.22) and as a graphical representation, the number of idle slot times per successful transmission is depicted in Figure 2.4 which shows that as the size of the initial contention window size increases, the idle slot time increases as expected.

Using the transition probabilities in a back-off counter of a station, it has been possible to obtain a throughput estimate in Bianchi's article [3]. This study shows that an optimization of CW sizes is needed according to the changing number of stations in the medium. As the number of stations in the medium increases, the probability of collision increases which will cause inefficient usage of the channel and increased delay. On the other hand, large CW sizes have the drawback of increasing idle slot times because of the larger back-off timer values. For this reason, an optimization in CW sizes is needed. The system should adapt itself according to the changing medium.

In this thesis, our aim is to design an adaptive system for video streams to provide QoS. For this purpose, the EDCA mechanism, which provides QoS is analyzed in the following chapter, by extending Bianchi's DCF model for the EDCA mechanism.

3. ANALYTICAL ANALYSIS MODEL FOR EDCA

3.1. Markov Chain Model for EDCA

According to the Bianchi's model, it is possible to obtain the collision probability and throughput with a given initial back-off timer window size and the number of the terminals accessing the channel. Analysis of the DCF mechanism showed that, the efficient usage of the medium depends on the CW sizes according to the changing number of the stations in the medium. As the number of the stations in the medium changes, the optimum CW size providing maximum possible throughput also changes. Therefore changing the CW sizes according to the increased/decreased number of stations, is a requirement. On the other hand, by differentiating CW sizes for different traffic classes in EDCA mechanism, it is possible to provide priority among different traffic classes. Our approach in this thesis will be to find the optimum contention window sizes to provide an efficient usage of the medium and then differentiate the access parameters which provides satisfactory QoS for a given video stream.

The multimedia applications require QoS from the throughput and delay point of view, however the DCF mechanism does not provide differentiated service for different traffic classes. Therefore, the EDCA mechanism, defined in the IEEE 802.11e, have to be modeled according to the requirements. For this purpose, we need to re-model the Bianchi's model in order to obtain throughput and delay estimates for the EDCA mechanism. Bianchi's model is not sufficient for our purpose because;

i.) The DCF mechanism, investigated in the Bianchi's model, does not allow prioritization for different classes of traffic. The model should be re-designed for the EDCA mechanism where each access category (AC) has a different back-off window size and AIFS. So the probabilities and throughput calculations will be different for different ACs.

ii.) In the standard operation of DCF and EDCA, when a station starts transmission, all other stations pause their back-off timers until the channel becomes idle again. In Bianchi's model, this fact is not being taken into account. The model has to be re-designed that the probability of decreasing the back-off timer probability is not equal to 1.

iii.) The delay will be a measure in our adaptive QoS algorithm, since delay is the most important constraint in multimedia applications. For this purpose, a delay model will be created for delay of each AC.

iv.) In Bianchi's model, no retry limit was considered. For this reason, in Bianchi's model a packet can be retransmitted infinite times if the transmission is not successful and this situation can cause large delays. This is a very important factor for the multimedia transmission because if a packet has exceeded it is delay limit, that packet is not valuable in the receiver side anymore. For this reason, retry limits will be added into Bianchi's model.

In the given model in Figure 3.1, each different traffic class is investigated in a different model because of having different access parameters so they have different probabilities of accessing the medium.

As can be seen from the model, there are 3 variables in each state. The first variable i represents the priority of the transmitted packet. For different priorities, the CW_{min}, CW_{max} and AIFS are different, which differentiates the probabilities. For this reason, for each priority, probabilities are calculated separately.

It is assumed that, for a given priority, the probability of a collision to occur is constant and independent from the number of collisions encountered. P_i represents the probability that a station in the i-th priority senses the channel busy. So we will calculate P_i probabilities for each priority class and estimate a throughput for each of them.

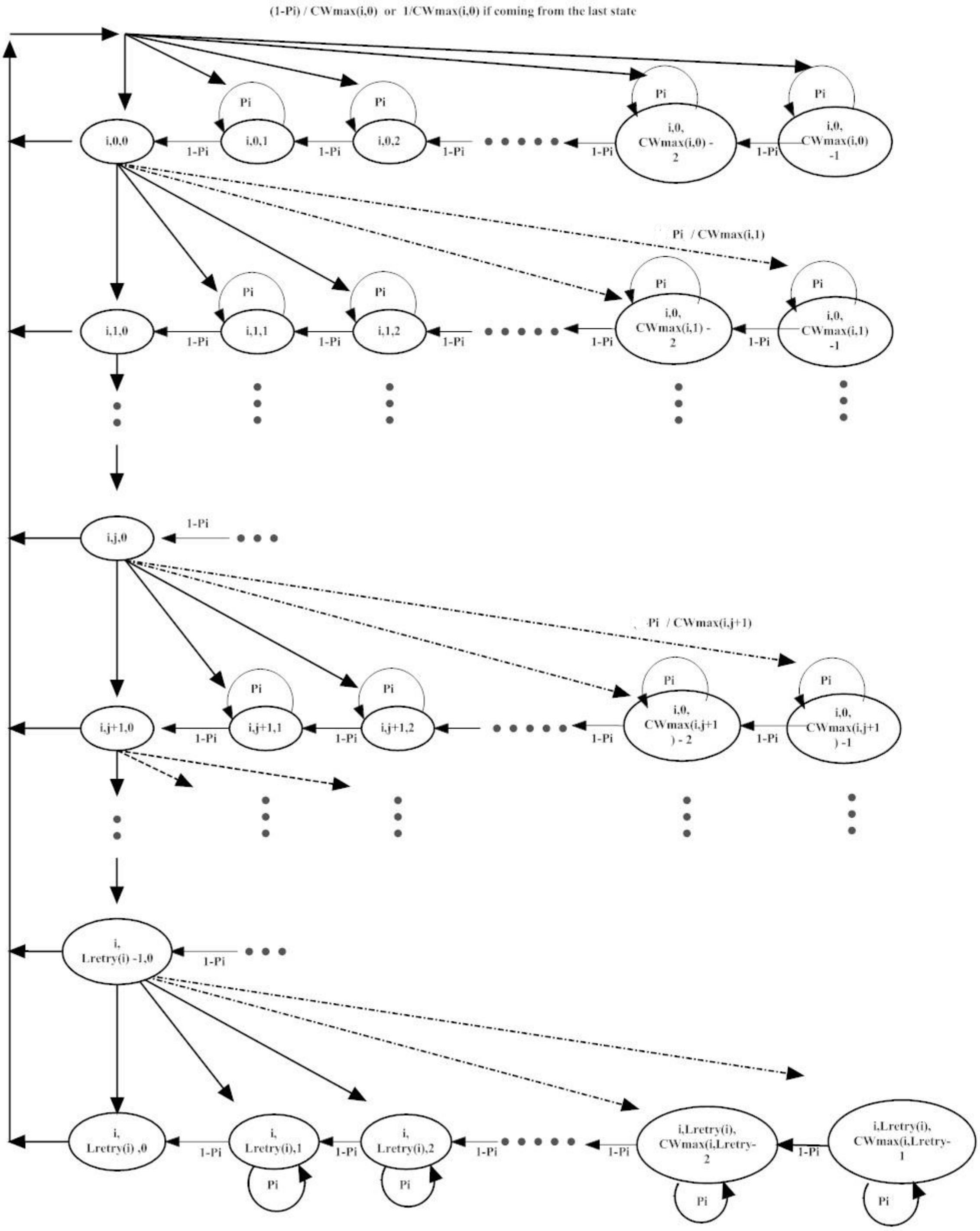

Figure 3.1. Proposed Markov model for analytical analysis of EDCA

The variable j in the model represents the number of collisions occurred when trying to transmit a packet. As the packet encounters a collision, the new contention window sizes are calculated. But different from the Bianchi's DCF model, j can be increased until it reaches $L_{Retry,i}$.

The last variable k is representing the back-off counter of the station. Also different from Bianchi's model, the probability of decrementing the back-off counter value is not equal to 1. Instead, back-off time decreasing probability of a station is $1 - P_i$, keeping in mind that the station pauses the back-off counter when another transmission captures the medium.

3.2. Analysis of EDCA Markov Model

In order to obtain a throughput estimate, we have to investigate the probabilities for each priority of accessing medium, probabilities of successful transmission, collision probability and the probability of having the channel busy in a given slot time. For this purpose, the probabilities are obtained from the model.

The simulation results obtained from Matlab are given as a graphical representation. For these simulations in Matlab, two different traffic classes are investigated. The first priority represents a higher priority like video traffic while the second priority represents lower priority like data traffic. The variables used during this simulation are given in Tables 3.1 and 3.2.

Table 3.1. The Variables Used During Matlab Simulation for EDCA Model

Variable	Priority#1	Priority#2
Number of Stations	5	$[1 \rightarrow 25]$
L_{Retry}	4	8
CW_{init}	16	32
CW_{max}	32	1024
AIFS	43 μs	61 μs

As can be seen from the Table 3.1, the initial contention window size for priority 1 is smaller than the priority 2, which increases the chance of accessing the medium for priority 1 stations. On the other hand, the retry limit defined for the priority 1 is smaller than priority 2 because, the video signal is delay sensitive, such that a received packet with a delay higher than a predefined value, has no meaning at the receiver side.

Table 3.2. Parameters Used for Both Priorities

Variable	Value
Payload	1024 Bytes
MAC Header	38 Bytes
PHY Header	24 Bytes
T_{slot}	9 μs
SIFS	16 μs
DIFS	34 μs
ACK Length	14 Bytes+MAC Header
$ACK_{Timeout}$	DIFS+T_{ACK}

3.2.1. Transmission Probabilities of the Model

In our study, our aim is to investigate the behavior of the system in steady state, so we will calculate the probabilities of each state in the model in steady state. For this purpose, we will firstly calculate the transition probabilities of the Markov model and then determine the steady state probabilities.

Different from the transition probability of the back-off counter in Bianchi's model, the back-off counter decreases if there is no other station transmitting into the medium. P_i is the probability that a station senses the medium busy in a slot time so (3.1) represents that, the counter decreases in a slot time if no transmission has started at that slot time from other stations.

During the back-off timer period, the process of decreasing the back-off timer value is paused if any transmission occurs from other stations in the medium. The back-off counter continues to decrease when the transmission on the medium is completed. (3.1) shows the transition from the (i,j,k+1) state to (i,j,k) state in case of no station transmitting in the medium. On the other hand (3.2) states that, this transition does not occur in case of a transmission in the medium.

$$P[(i,j,k)|(i,j,k+1)] = 1 - P_i \tag{3.1}$$

$$P[(i,j,k)|(i,j,k)] = P_i \tag{3.2}$$

If a collision occurs while transmitting into the medium, then the next random back-off timer value will be taken from the possible values of the next vertical state (j). The probability of the back-off timer horizontal state, representing the back-off timer value, is uniformly distributed between 0 and $CW_{max(i,j)} - 1$. But we should keep in mind that this vertical state transition in the Markov model is possible, if the retransmission number has not reached retry limit defined for that priority ($L_{Retry,i}$). (3.3) states the state transition probability in the model in case of a collision encountered.

for $j < L_{Retry,i}$;

$$P[(i,j+1,k)|(i,j,0)] = \frac{P_i}{CW_{max(i,j+1)}} \tag{3.3}$$

For the case of reaching retry limit defined for that priority ($L_{Retry,i}$), we have to drop that packet and repeat this process again for the new packet to be transmitted. Therefore, vertical state (j), representing the retransmission number for the packet, will be 0. In (3.4), the state transition probability of a station after reaching maximum number of collisions is given.

$$P[(i,0,k)|(i,L_{Retry,i},0)] = \frac{1}{CW_{max(i,0)}} \tag{3.4}$$

In case of a successful transmission, the process starts again for a new packet to be transmitted. This means transition in the model from state $(i,j,0)$ to $(i,0,k)$. In (3.5) the transition to the first state in case of a successful transmission is given.

$$P[(i,0,k)|(i,j,0)] = \frac{1-P_i}{CW_{max(i,0)}} \tag{3.5}$$

After obtaining the transition probabilities in the Markov model, we are able to determine the steady state probabilities. We will represent the steady state probabilities as $b_{i,j,k}$.

$$b_{i,j,k} = lim_{t\to\infty}P(i,j,k) \tag{3.6}$$

$b_{i,j,k}$ represents that our packet with priority i has encountered j collisions and it's back-off counter is in $k-th$ state in steady state.

$$b_{i,j,0} = P_i^j b_{i,0,0} \tag{3.7}$$

$$b_{i,j,k} = \frac{CW_{max(i,j)}-k}{CW_{max(i,j)}}\frac{1}{1-P_i}b_{i,j,0} \tag{3.8}$$

Using the fact that the sum of probabilities for each state is equal to 1, we can find the probability of $b_{i,0,0}$ from (3.9).

$$\sum_{j=0}^{L_{Retry,i}} \sum_{k=0}^{CW_{max(i,j)}} b_{i,j,k} = 1 \tag{3.9}$$

Using the equations (3.7) and (3.8), we can solve $b_{i,0,0}$ as in (3.10).

$$b_{i,0,0} = \frac{1}{\sum_{j=0}^{L_{Retry,i}} \sum_{k=0}^{CW_{max(i,j)}} \frac{CW_{max(i,j)}-k}{CW_{max(i,j)}} \frac{P_i^j}{1-P_i}} \tag{3.10}$$

A station begins transmission when its back-off timer reaches zero. For this reason, if we find the sum of state's probabilities where the back-off counter is zero, then we will be able to find the probability that a station in probability i class transmits in a given slot time. In (3.11), the probability that a station transmits in slot time is calculated by summing the steady state probabilities of states where the back-off counter is equal to 0.

$$\tau_i = \sum_{j=0}^{L_{Retry,i}} b_{i,j,0} = \sum_{j=0}^{L_{Retry,i}} P_i^j b_{i,0,0} \tag{3.11}$$

As expected, because of having a smaller initial contention window, our expectation was to have a higher τ_i for priority 1. The result obtained from the Matlab simulation for the τ_i for both priorities is given in Figure 3.2.

After obtaining the probabilities of back-off timer states and the probability that a station in probability i class transmits in a given slot time according to the P_i, we need to find P_i. P_i is the probability that a station senses the medium busy in a slot time.

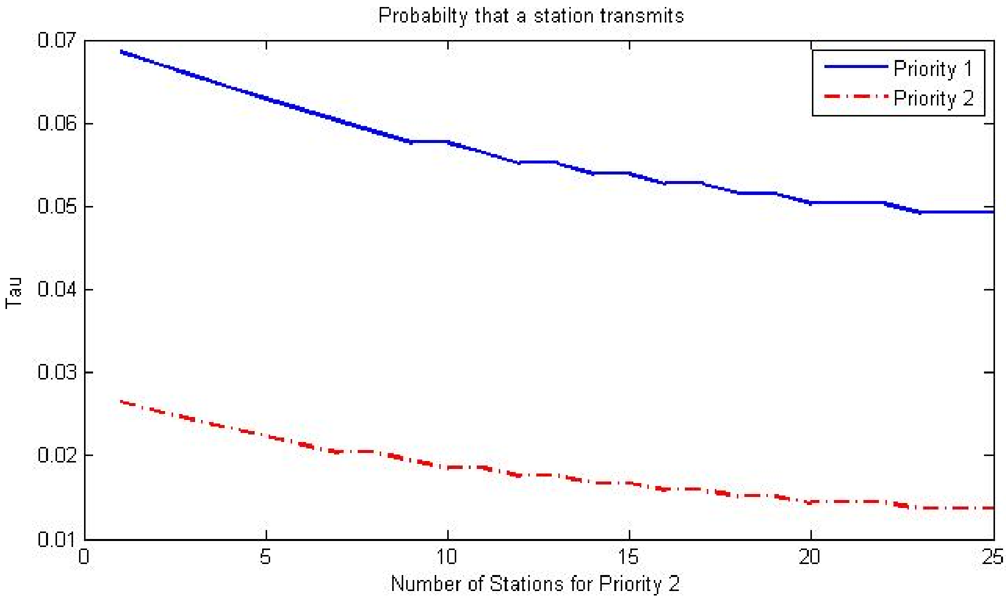

Figure 3.2. Probability that a station transmits in a given slot time

In order to find the probability of sensing the medium busy, we need to find the probability that at least one station is transmitting in given slot time. If we subtract the probability of no station transmitting into the medium from 1, then we will find the probability that at least one station transmits.

$$P_i = 1 - (1 - \tau_i)^{n_i - 1} \prod_{l=0, l \neq i}^{N} (1 - \tau_l)^{n_l} \tag{3.12}$$

In (3.12) the τ_l is representing the probability that a station with priority class l transmits in the given slot time, N represents number of priorities and n_l represents the number of terminals with priority class l. The graphical representation of the probability that a station in priority i senses the medium is given in Figure 3.3

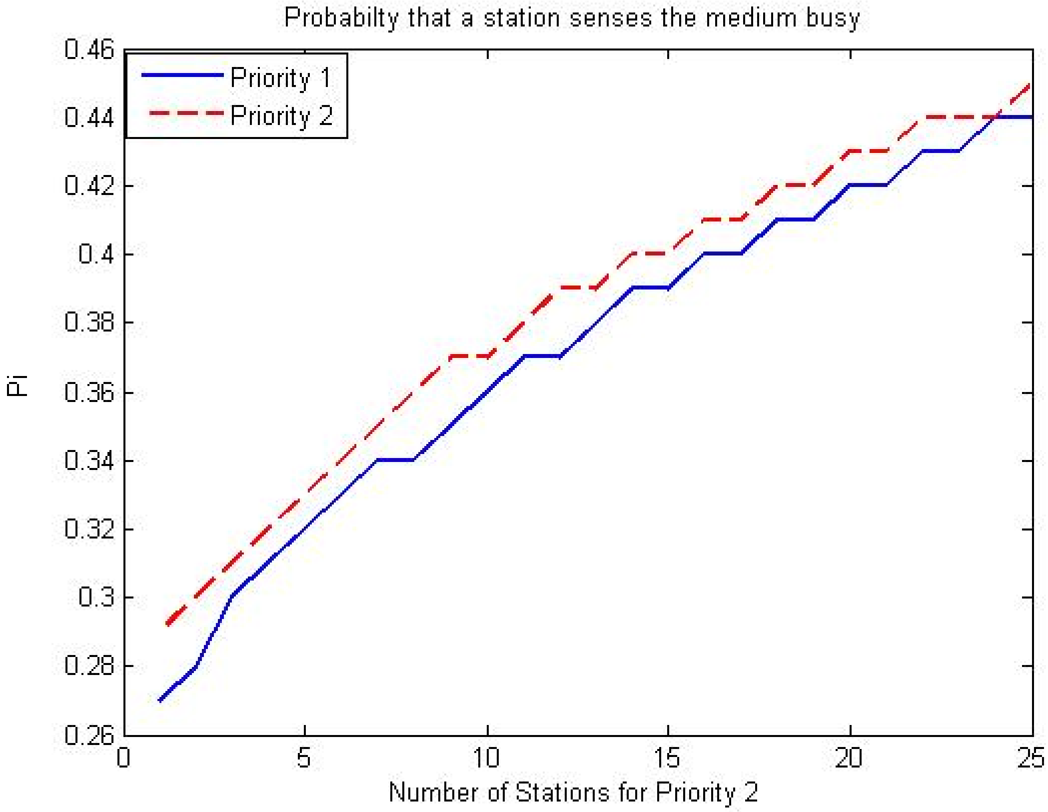

Figure 3.3. Probability that a station senses the medium busy

It is possible to solve (3.12) and (3.11), in order to find P_i and τ_i values for the given number of terminals and CW sizes.

3.2.2. Probabilities of Successful and Collided Transmission

Using the same approach followed in deriving (3.12), it is also possible to calculate the probability that the medium is busy in a given slot time. The graphical representation of the P_{busy}, whose expression is given in (3.13), is shown in Figure 3.4.

$$P_{busy} = 1 - \prod_{l=0}^{N} (1 - \tau_l)^{n_l} \tag{3.13}$$

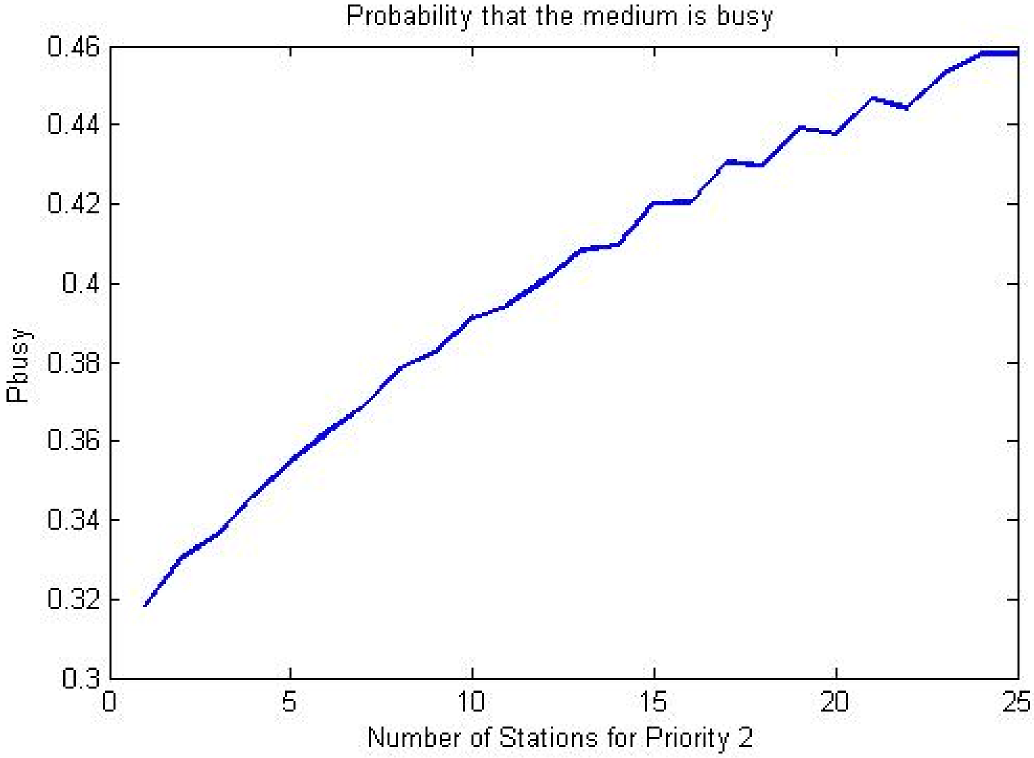

Figure 3.4. Probability that the medium is busy

In order to compute the probability that a successful transmission occurs in a slot time, it is required that there is no other stations transmitting while a station in $i-th$ priority class is transmitting. So the probability that a station in $i-th$ priority class is transmitting a packet without collision can be calculated as:

$$P_{success,i} = n_i\tau_i(1-\tau_i)^{n_i-1}\prod_{l=0,l\neq i}^{N}(1-\tau_l)^{n_l} \tag{3.14}$$

$$P_{success} = \sum_{i=0}^{N-1} P_{success,i} = \sum_{i=0}^{N-1} n_i\tau_i(1-\tau_i)^{n_i-1}\prod_{l=0,l\neq i}^{N}(1-\tau_l)^{n_l} \tag{3.15}$$

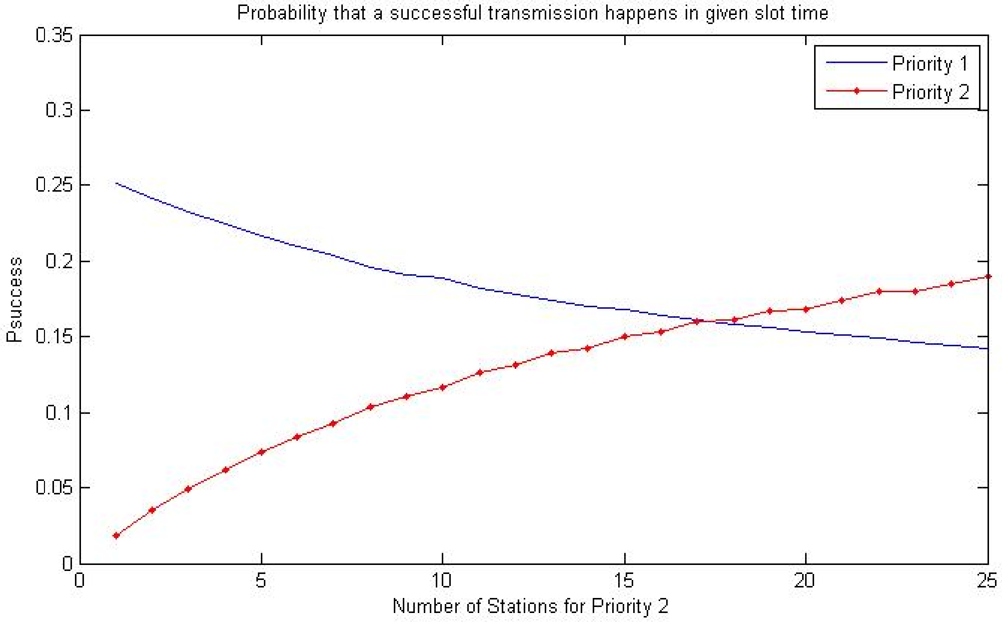

Figure 3.5. Probability that successful transmission occurs in given slot time

As can be seen from Figure 3.5, while the number of class 1 stations is constant and the number of the station for the priority 2 increases in the medium, the successful transmission probability of the priority 1 decreases because of decreased chance of accessing the channel. This is a good example for reasoning the need for an adaptive system. The differentiation in contention window sizes was a good solution for QoS at the beginning of the simulation but as can be seen, as the number of the stations increases, the initial contention window sizes become insufficient.

If the medium is busy but a successful transmission does not occur in a slot time, that means there is a collision in the medium. For this reason, we can calculate the probability of a collision in a slot time with a much simpler approach.

$$P_{coll} = P_{busy} - P_{success} \tag{3.16}$$

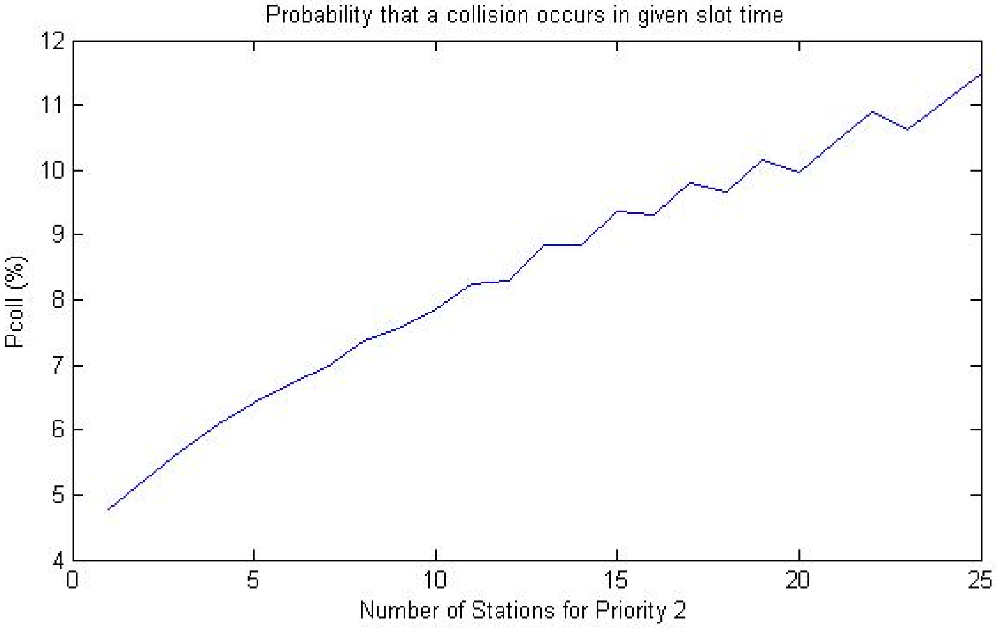

Figure 3.6. Probability that collision occurs in given slot time

The resulting probability of a collision in a slot time, as the number of the priority 2 stations increases is given in Figure 3.6. The resulting graph shows that the collision probability increases as the number of stations in the medium increases. This increase in the collision probability not only decreases throughput, but also increases delay, where low delay is an important requirement in multimedia applications such as video or voice streaming. Our purpose in designing an adaptive mechanism for contention window sizes is to decrease this collision probability according to the dynamic changing nature of the medium, which will increase throughput while decreasing delay.

In order to evaluate the effect of increasing CW sizes for both priorities, a new simulation in Matlab is carried out. In this simulation, the initial CW size for priority 1 is increased from 2 to 256 while the initial CW size for priority 2 is increased from 4 to 512. Therefore the CW size ratio between priorities remains constant. Our expectation was to observe that the probability of collision will decrease as we have larger window sizes. In Figure 3.7, it can be seen that the collision probability is decreasing as we increase the CW sizes.

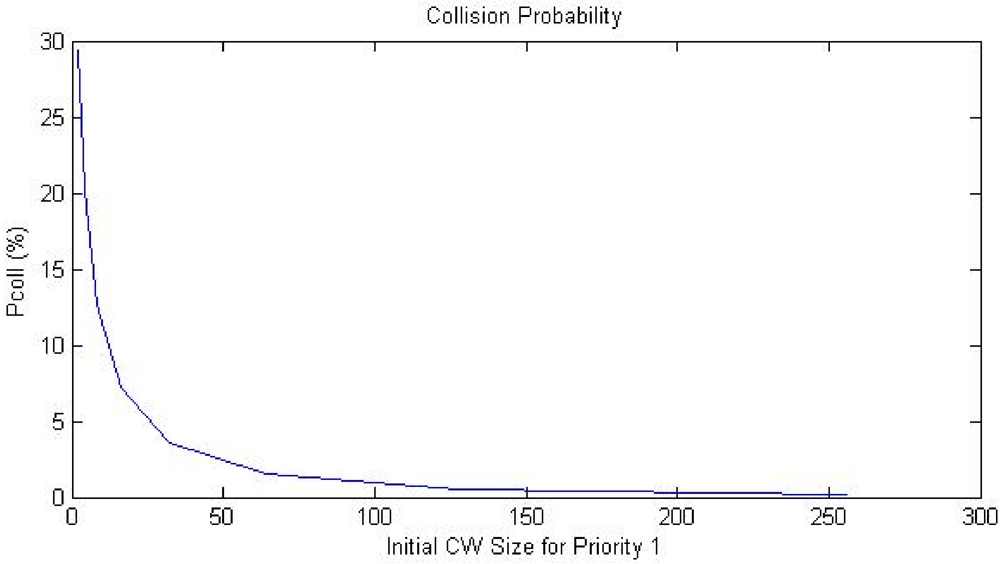

Figure 3.7. P_{coll} versus initial window size

3.3. Throughput Analysis of EDCA

In the previous section, we have determined the probability for a station to sense the medium busy (P_i), the probability that the medium is busy (P_{busy}), probability that a successful transmission occurs in a slot time ($P_{success}$) and the probability that a collided transmission occurs in a given slot time (P_{coll}). Now we are in a position to calculate the throughput for each priority.

In order to calculate the throughput of a priority, we should determine the length of the time elapsed during a successful $T_{success}$ and unsuccessful transmission T_{coll}. In the following equations (3.17) and (3.18) T_{MAC}, T_{PHY}, $T_{Payload}$ and T_{ACK} represent time required for transmission of MAC header, physical header, ACK packet and payload respectively.

$$T_{success} = AIFS + T_{back-off} + T_{MAC} + T_{PHY} + T_{Payload} + SIFS + T_{ACK} \tag{3.17}$$

$$T_{coll} = AIFS + T_{back-off} + T_{MAC} + T_{PHY} + T_{Payload} \tag{3.18}$$

$T_{back-off}$ represents the average back-off timer for a successful packet while accessing the medium. Among all the stations trying to access the channel, the station with the minimum back-off timer length will obtain the opportunity for transmission. For this reason, as the number of the terminals increases, the probability of a station having smaller back-off timer length increases. So by using P_{busy}, we can find the number of the idle slot times before accessing the medium. By multiplying length of a slot time with number of idle time slots, we are able to find $T_{back-off}$:

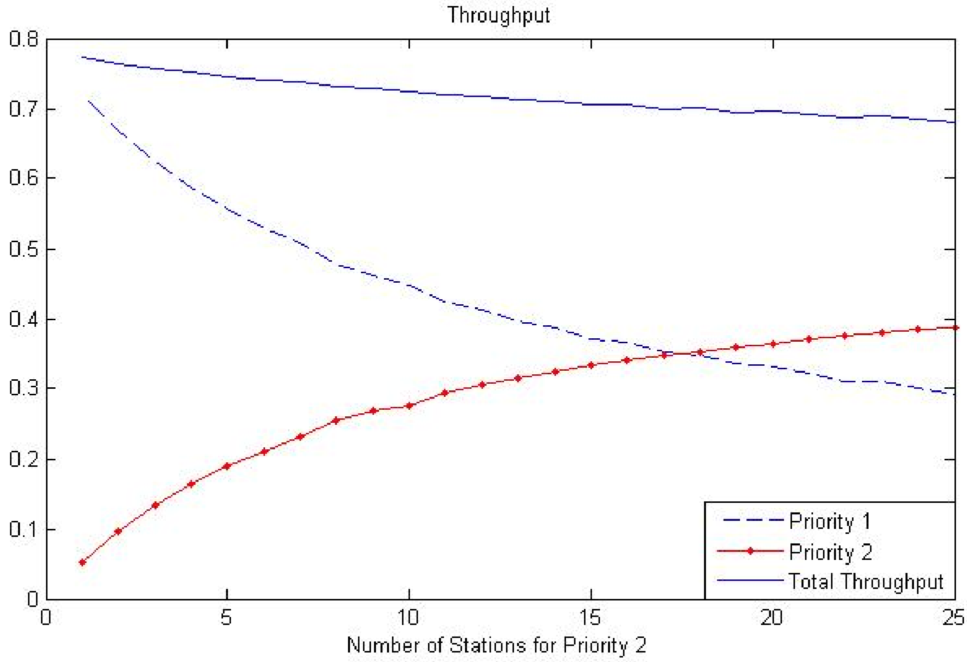

Figure 3.8. Throughput versus number of data stations

$$N_{back-off} = \frac{1}{P_{busy}} - 1 \tag{3.19}$$

$$T_{back-off} = T_{slot} N_{back-off} = T_{slot}(\frac{1}{P_{busy}} - 1) \tag{3.20}$$

Finally, we can calculate the throughput by using the $T_{success}$, T_{coll}, $P_{success}$ and P_{coll}

$$Throughput_i = \frac{T_{Payload} P_{success,i}}{T_{success} P_{success} + T_{coll} P_{coll}} \tag{3.21}$$

Figure 3.8 shows that as the number of the stations increases, the collision probability increases and the total throughput decreases because of the collisions. Also the number of the terminals with video traffic is constant where the number of data stations are increasing. So the lower priority traffic gains more opportunity of accessing the channel as the higher priority traffic gets a lower throughput.

We have observed that, the transmission probability of a station depends on the number of the stations in the medium and the contention window sizes. As we can not control the number of the stations in order to adjust the transmission probability, we can optimize our contention window sizes according to the transmission probability that provides maximum throughput. In the following section, we will find the optimum transmission probability that provides maximum throughput.

3.4. Optimum Transmission Probability that Provides Maximum Throughput

According to the throughput equation we have obtained (3.21), it is possible to find the maximum achievable throughput for a given priority. The probability of a successful or collided transmission depends on the number of the stations and the transmission probability (τ). For this reason the optimum transmission probability that provides the maximum achievable throughput is different for different number of the stations in the medium.

The maximum throughput for a given priority is achieved when the stations in the medium belongs to that given priority, i.e.,when $P_{success,i}$ is equal to the $P_{success}$.

After re-arranging the (3.21), throughput can be expressed as:

$$Throughput_i = \frac{T_{Payload}}{T_{success} + T_{coll}\frac{P_{coll}}{P_{success}}} \tag{3.22}$$

Assuming that the time elapses for a successful transmission and a collided transmission are approximately equal, then (3.22) becomes:

$$Throughput_i = \frac{T_{Payload}}{T_{success}(1 + \frac{P_{busy} - P_{success}}{P_{success}})} = \frac{T_{Payload}}{T_{success}\frac{P_{busy}}{P_{success}}} \tag{3.23}$$

As can be seen in (3.17) and (3.18), the time needed for a successful or a collided transmission includes constant and variable portions. The variable part ,$T_{back-off}$, represents the average back-off time slots needed before accessing into the medium. We can represent $T_{success}$ as summation of these constant (N_{const}) and variable ($N_{back-off}$) portions.

After defining the constant portion of the summation as given below, it is possible to define $T_{success}$ again.

$$N_{const} = AIFS + T_{MAC} + T_{PHY} + T_{Payload} + SIFS + T_{ACK} \tag{3.24}$$

$$T_{success} = T_{slot}(N_{const} + N_{back-off}) = T_{slot}(N_{const} + \frac{1 - P_{busy}}{P_{busy}}) \tag{3.25}$$

After replacing $T_{success}$ in (3.23), the throughput is expressed as:

$$Throughput_i = \frac{T_{Payload}P_{success}}{T_{slot}[1 + (N_{const} - 1)P_{busy}]} \tag{3.26}$$

Recalling that $T_{payload}$ and T_{slot} are constant, in order to find maximum achievable throughput, we can neglect these and obtain a simpler form to optimize (3.27). By taking the derivative of (3.27) with respect to τ and determining where the derivative is equal to 0, it is possible to obtain the optimum transmission probability (τ) which provides maximum achievable throughput.

$$\frac{P_{success}}{1+P_{busy}(N_{const}-1)} = \frac{n\tau(1-\tau)^{n-1}}{1+(N_{const}-1)(1-(1-\tau)^n)} \tag{3.27}$$

Taking the derivative of (3.27) with respect to τ and making the simplifications, we obtained the following equation:

$$\frac{\partial \frac{P_{success}}{1+P_{busy}(N_{const}-1)}}{\partial \tau} = -n(\frac{-(N+1)(1-\tau)^n + N(1-\tau)^{2n} + (N+1)\tau(1-\tau)^n n)}{(-1+\tau)^2(-(N+1)+N(1-\tau)^n)^2} \tag{3.28}$$

In the (3.28), $N_{const} - 1$, which is a constant scalar, is represented as N for simplification.

In order to find the optimum τ value, that provides the maximum throughput, we should equate the numerator of (3.28) to 0, i.e.,

$$(-(N+1)(1-\tau)^n + N(1-\tau)^{2n} + (N+1)\tau(1-\tau)^n n) = 0 \Rightarrow (N+1)(n\tau-1) = -N(1-\tau)^n \tag{3.29}$$

Under the condition $\tau \ll 1$;

$$(1-\tau)^n = 1 - n\tau + \tau\frac{n(n-1)}{2} \Rightarrow \frac{Nn(n-1)}{2}\tau^2 + \tau n - 1 = 0 \tag{3.30}$$

Finally by solving (3.30), we are able to determine the optimum transmission probability that provides the maximum achievable throughput.

$$\tau_{opt} = \frac{\sqrt{n(2nN + n - 2N)} - n}{(n^2 - n)N} \tag{3.31}$$

3.5. Comparison Between DCF and EDCA

The multimedia applications require QoS and the DCF mechanism is insufficient while providing this QoS because of not having a service differentiation. Therefore the channel resources will be shared among stations without having any priority while using the DCF mechanism. On the other hand, in EDCA mechanism, we are able to prioritize different traffic classes by differentiating the CW sizes belonging to these traffic classes. So the higher priority traffic will have higher change of capturing the channel.

According to the Bianchi's model and our proposed model, it is possible to observe how the channel resources are shared among stations with data traffic and video traffic. For this purpose, Matlab simulations are carried out using both DCF and EDCA models.

Table 3.3. Parameters Used for DCF-EDCA Comparison

Variable	EDCA Video	EDCA Data	DCF Video	DCF Data
Number of Stations	5	$[1 \rightarrow 25]$	5	$[1 \rightarrow 25]$
CW_{init}	16	32	16	16
CW_{max}	32	1024	1024	1024

In order to observe the service differentiation better, a scenario, where the total throughput for both models are approximately equal, is considered. The CW sizes are selected accordingly. In both simulation scenarios, there are 5 stations with video traffic and the number of the stations with data traffic are increasing from 1 to 25.

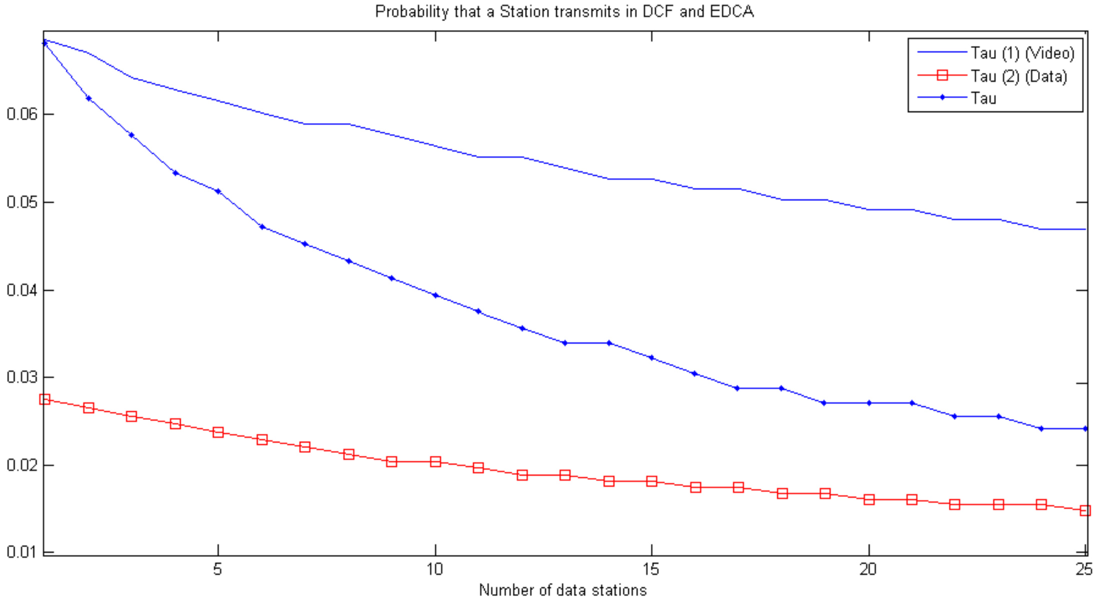

Figure 3.9. The probability that a station transmits

Firstly the probabilities that a station transmits (τ in Bianchi's model, τ_i in our EDCA model) are investigated. In Figure 3.9, the probability of a DCF station to transmit (τ) is equal to the probability of a video station in EDCA to transmit in a slot time (τ_1) because of having the same CW_{init}. Also because of the prioritization in the access parameters, in EDCA mechanism, the data stations have lower probability of accessing the channel compared to the video stations. On the other hand, as the number of the data stations in the medium increases, the video stations in the DCF mechanism lose the control in the medium and shares the medium with the data stations equally. So the probability that a video station in DCF mechanism transmits in a slot time decreases as the number of the stations in the medium increases. On the other hand video stations in the EDCA mechanism protect their prioritization and the decrease in τ_1 is lower than the decrease in τ.

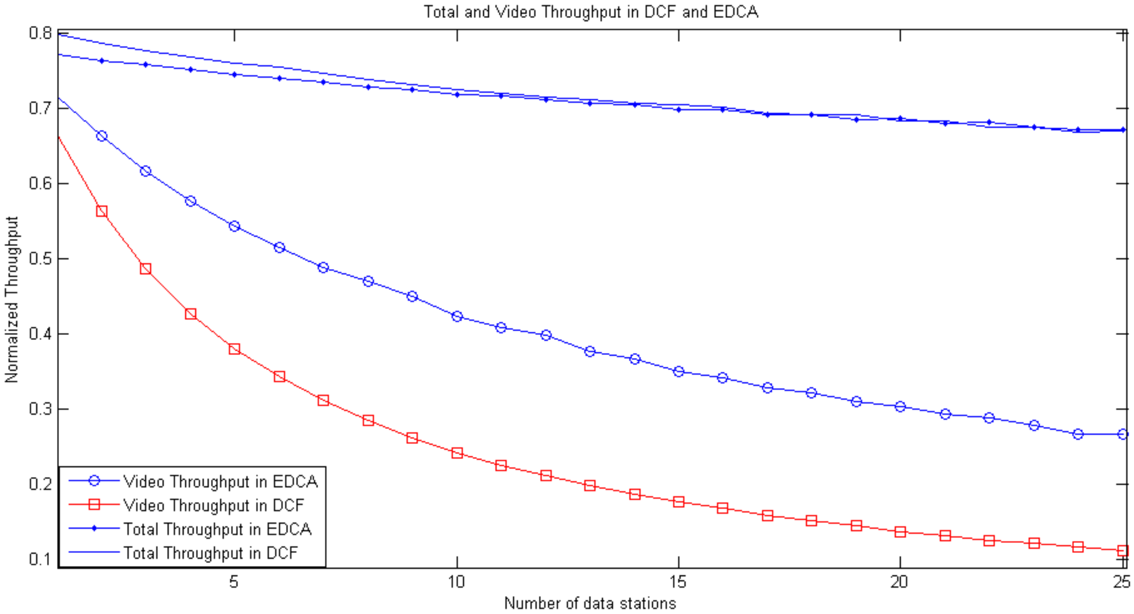

Figure 3.10. Video and total throughput in DCF and EDCA mechanisms

After investigating the probabilities of accessing the channel, the video throughput in both mechanisms are investigated. As mentioned above, in order to observe the service differentiation better, the CW sizes, which provides that the total throughput for both models are approximately equal, are selected. As can be seen from Figure 3.10, there is a high correlation between probability of a station to transmit (τ or τ_i) and throughput. At the beginning, the video throughput in both mechanism are close but as the data station in the medium increases, the throughput of video stations in DCF mechanism decreases much faster than the ones in EDCA mechanism. This shows that the service differentiation in EDCA provides priority to video traffic.

The throughput of the video traffic in EDCA mechanism is better compared to the DCF mechanism however, the video throughput decrease in the EDCA is an indication of a need for an adaptive system, which redefines the access parameters according to the dynamically changing medium.

3.6. Drop Probability Analysis of EDCA

After a packet with the access category i is being tried to be transmitted, it will have the opportunity of being retransmitted $L_{Retry,i}$ times. If this packet encounters collision more than $L_{Retry,i}$, then we can see that this packet will be dropped. It is a suitable approach especially for multimedia streams, because of having small delay constraints. If a packet has accessed higher delay, at the receiver side, the received packet is meaningless.

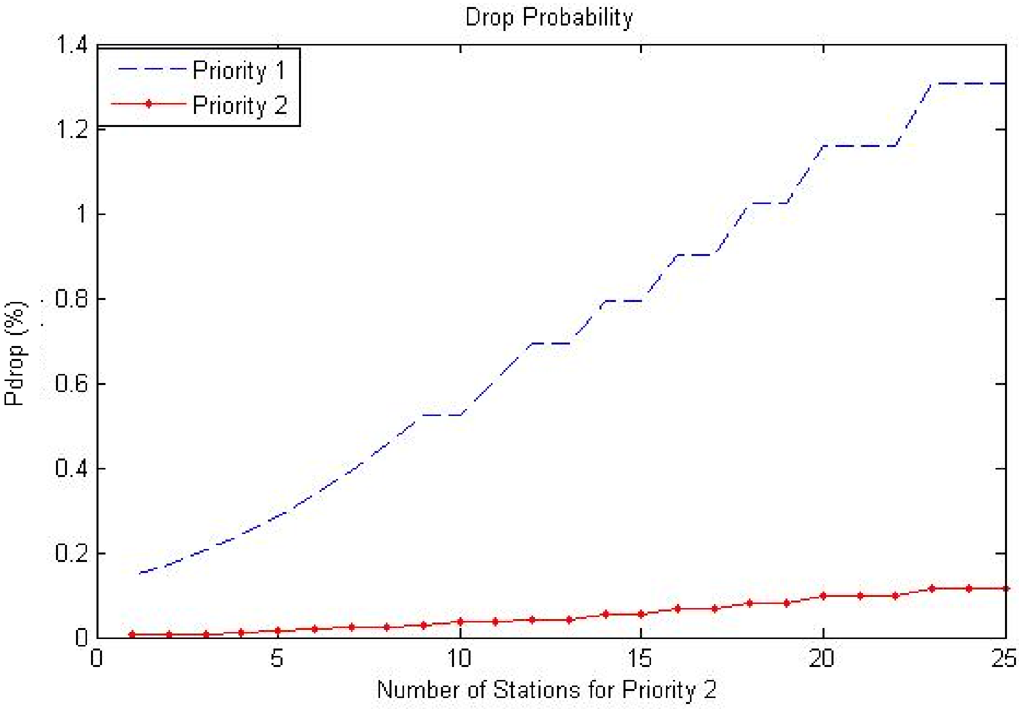

Figure 3.11. Drop probability

Keeping in mind that the probability P_i is the probability that a packet of the access category i encounters a collision in a slot time, we can find the probability of dropping a packet, which also means of encountering collision $L_{Retry,i} + 1$ times. In our implementation of the model, the $L_{Retry,i}$ for priority 1 was selected lower than priority 2 in order to have lower delay for priority 1 traffic. For this reason, the drop probability of priority 1 traffic is higher than priority 2 as can be seen in Figure 3.11.

$$P_{drop,i} = P_i^{L_{retry}(i)+1} \tag{3.32}$$

Using (3.32), it is also possible to obtain the probability of a packet to be transmitted without being dropped can be computed as follows:

$$P_{transmitted,i} = 1 - P_i^{L_{retry}(i)+1} \tag{3.33}$$

3.7. Delay Analysis of EDCA

While we investigate the saturation delay of the system,we have to take into account, the expectation of back-off timer for priority i, the transmissions from other stations which causes back-off timer to pause, the delay caused from the dropped packets, and retransmission count because of the collisions suffered.

3.7.1. Delay Analysis for Dropped Packets

For this purpose, we should first calculate the delay resulting from dropped packets. A dropped packet means that the packet had encountered $L_{Retry,i} + 1$ times collision. The expected number of back-off slot times that a dropped packet waits can be found as follows:

$$B_{drop,i} = \sum_{j=0}^{L_{Retry,i}} \frac{CW_{max(i,j)}}{2} \tag{3.34}$$

$B_{drop,i}$ is the average number of back-off slot times for a dropped packet without considering the other transmissions causing this back-off counter to pause until their transmissions are over. By using the P_i it is possible to find out how many slot times have to pass for a transmission to happen from any station.

By using the data of how frequently a transmission from another station happens, we can find out how many times that the back-off counter has to be paused in $B_{drop,i}$ slot times.

$$f_{pause,i} = \frac{1}{P_i} \tag{3.35}$$

$$N_{drop,i} = \frac{B_{drop,i}}{f_{pause,i}} \tag{3.36}$$

Finally, we are able to calculate the delay resulting from the dropped packets;

$$D_{dropped,i} = B_{drop,i}T_{slot}+N_{drop,i}[\frac{P_{success}}{P_{busy}}T_{success}+\frac{P_{coll}}{P_{busy}}T_{coll}]+(L_{Retry,i}+1)(T_{coll}+T_{Timeout}) \tag{3.37}$$

As can be seen from the equation above, the delay resulting from the dropped packets is the sum of the time elapsed during back-off timer, the other station's transmissions during back-off timer, and the retransmission of the station itself until it reaches the retry limit.

3.7.2. Delay Analysis for Successfully Transmitted Packets

After calculating the delay caused by the dropped packets, we will calculate the delay caused by successfully transmitted packets. Firstly, we have to find the expected number of slot times before transmitting a packet successfully. This depends on the contention window length of that priority and the probability of a station in priority i sensing the channel busy (P_i).

Multiplication of the probability of a station to transmit successfully in j-th try and the expected number of the back-off timer until j-th state gives us the number of expected back-off slot if the packet is transmitted in j-th try. If we sum the expected number of slot times for all state's probabilities, we will be able to obtain the average expected number of slot time without considering the slot times paused because of other station's transmissions.The expected number of slot times for priority i is calculated as follows:

$$B_{success,i} = \sum_{j=0}^{L_{Retry,i}} P_i^j (1 - P_i) \sum_{s=0}^{j} \frac{CW_{max,i,s}}{2} \tag{3.38}$$

$B_{success,i}$ is the average number of back-off slot times for a dropped packet without considering the other transmissions causing this back-off counter to pause until their transmissions are over. By calculating how often another station can start transmission during a back-off interval, we can find out how many times back-off timer will be paused in a given back-off length:

$$N_{success,i} = \frac{B_{success,i}}{f_{pause,i}} \tag{3.39}$$

When calculating the delay for the dropped packets, we are sure that these packets were retransmitted $L_{Retry,i}$ times, but for successfully transmitted packets, we must estimate the number of retransmissions before transmitting successfully.

$$N_{Retry,i} = \sum_{j=0}^{L_{Retry,i}} j P_i^j (1 - P_j) \tag{3.40}$$

Finally, we are able to calculate the delay for the given priority for successfully transmitted packets;

$$\begin{aligned} D_{success,i} &= B_{success,i}T_{slot} + N_{success,i}[\frac{P_{success}}{P_{busy}}T_{success} + \frac{P_{coll}}{P_{busy}}T_{coll}] \\ &\quad + N_{Retry,i}(T_{coll} + T_{Timeout}) + T_{success} \end{aligned} \tag{3.41}$$

3.7.3. Average Delay

The final estimate of delay should include the estimation for dropped and transmitted packets according to their probabilities of $P_{transmitted,i}$ and $P_{drop,i}$ which were calculated in equations (3.32) and (3.33).

$$D_i = P_{drop,i}D_{dropped,i} + P_{transmitted,i}D_{success,i} \tag{3.42}$$

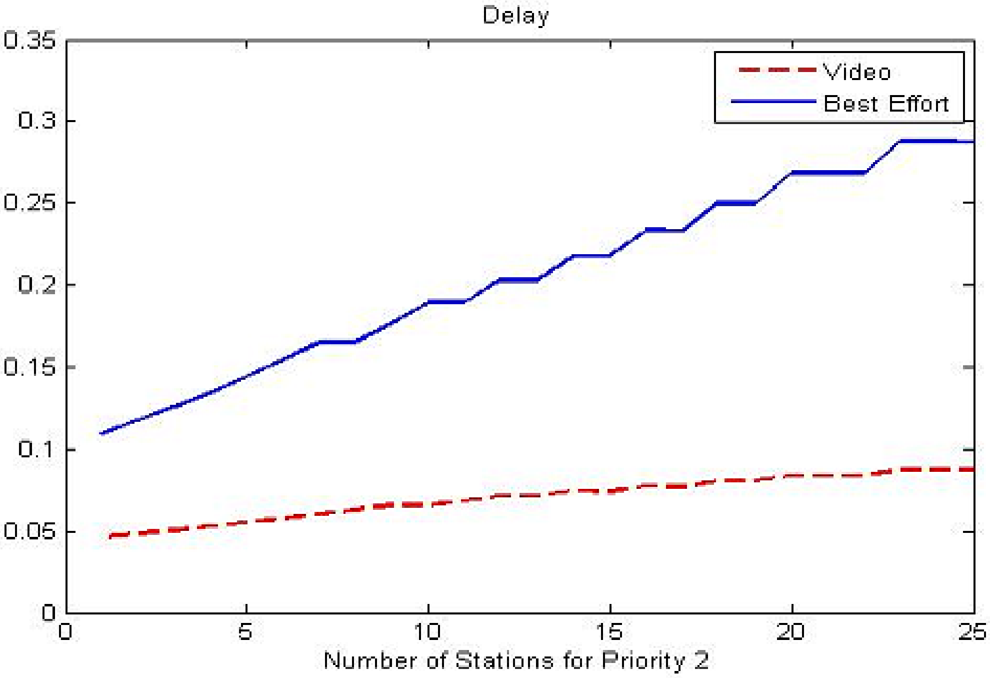

Figure 3.12. Average delay

In Figure 3.12, it is obvious that the delay for both priorities increases as we increase the number of stations in the medium. This increase in the delay results from both increased number of collisions and also increased contention window sizes.

As we have investigated, as the number of the collision increases, the expected number of retransmissions and so the state of the stations representing the retransmission number converges to higher states resulting the increase in delay. During the simulation, the number of increase for CW size for the priority 1 traffic was defined as 1 and defined as 5 for the priority 2. Therefore when the collisions in the medium increases, the contention window sizes for the priority 2 becomes much larger than priority 1. So the delay of priority 2 increases much faster than priority 1 as in Figure 3.12.

Our expectation was to observe that the average delay would increase as we have larger window sizes. As can be seen from Figure 3.13, the delay is increased as the CW sizes for both priorities are increased. On the other hand, the collision probability decreases (Figure 3.7) as we increase the CW sizes for the stations trying to access the medium. For this reason, an adaptive algorithm that decreases the collision probability, while keeping the delay below a certain threshold value is developed in this study.

During the analysis of the IEEE 802.11e EDCA mechanism, the required modifications are added to the model of former medium access mechanism model, IEEE 802.11 DCF. In order to allow prioritization, for each access category, a new Markov model is developed and the state transition probabilities of this model are analyzed separately. Furthermore the probability of sensing the channel busy and pausing the back-off timer is taken into account. On the other hand, the retry limit, which is defined in the standard, is also considered. Finally, delay analysis is also made in order to obtain the relationship between access parameters and delay.

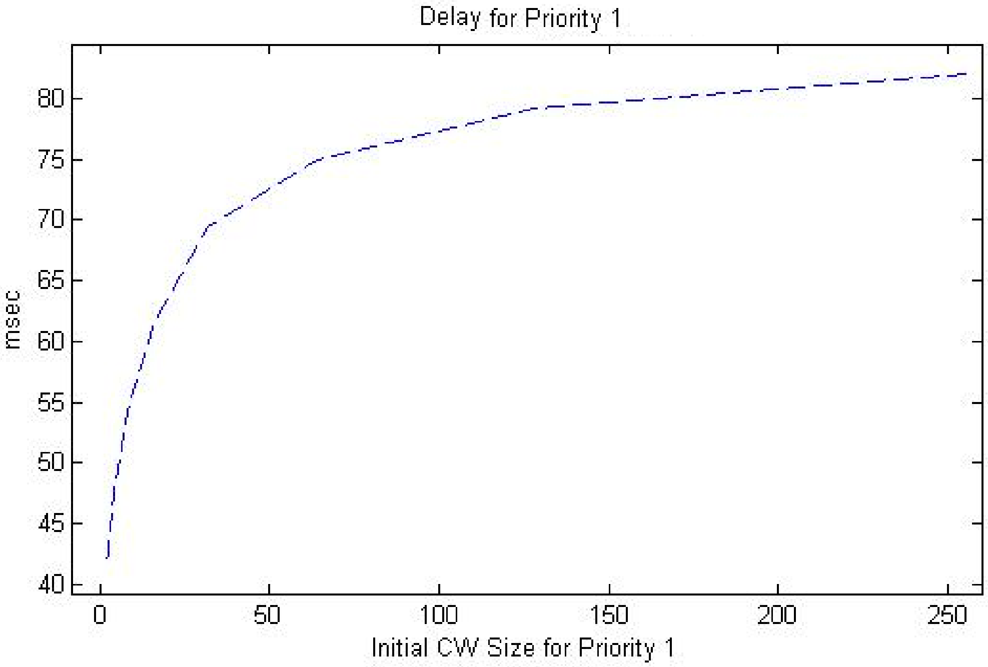

Figure 3.13. Average delay versus initial CW sizes

After finding the probabilities of the model and analyzing the throughput, the performance of the DCF and EDCA mechanisms are compared and we have found out that the service differentiation in EDCA provides priority to video traffic compared to DCF. However, the video throughput decreases and the increase of the delay for the video packets due to the increased number of stations in the medium pointed out a need of redefining the access parameters according to the dynamically changing medium.

In order to validate the EDCA model we built up, the analysis results will be compared with the real time simulation results. The throughput and delay are the most important factors in multimedia applications while providing QoS. For this reason, this validation will be discussed in the following chapter from the delay and throughput perspectives.

4. MODEL VALIDATION

In the previous chapter, we have developed a model for defining the probabilities that allowed us to estimate throughput and delay. In order to validate our model for EDCA, simulations are carried out by using the network simulation tool OPNET Modeler 14.5.

In OPNET, it is possible to simulate IEEE 802.11 network scenarios by modifying the parameters like application type, application definitions and wireless LAN parameters including the MAC parameters, which will be used in our network simulation.

In the simulations, two types of traffic will be defined as applications. First priority will be video traffic where the second priority is defined as best effort data traffic. In order to validate the proposed model, the same simulation is done in both OPNET and Matlab with the same parameters. The parameters used during this validation process are given in Table 4.1

Table 4.1. The Parameters Used During Validation Simulations in Matlab and OPNET

Variable	Video	Best Effort
Number of Stations	2	$[1 \rightarrow 12]$
L_{Retry}	4	4
CW_{init}	16	32
CW_{max}	32	1024
$DataRate$	1 Mb/s	1 Mb/s

We have already observed from Matlab simulations of the model in Chapter 3, the chance for video stations accessing the channel decreases as the number of the stations with best effort traffic increases. On the other hand, because of the increased unsuccessful transmission, the delay for both priorities increases.

In the OPNET Simulation, 2 video stations and 12 best effort data stations are defined. At the beginning of the simulation, two video stations and one station with data traffic are transmitting and after every 30 second, one of the best effort stations starts transmission. Since we are interested in the normalized throughput in both simulations, the data rate is set to 1 Mb/s not to encounter with the decrease in the utilization as the data rate increases. The network built up for this purpose is shown in Figure 4.1

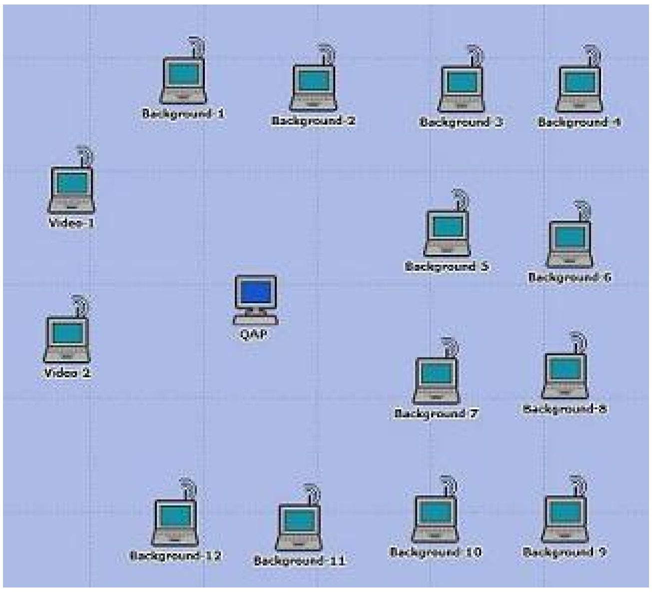

Figure 4.1. The network used in OPNET simulation

The throughput results for the OPNET and Matlab simulations are shown in Figure 4.2 and Figure 4.3 respectively. Considering that a data station starts transmission into the medium every 30 seconds, both figures show that the throughput of the video and best effort data traffic become equal when the number of data stations becomes 5. Considering the tendency of decreasing the throughput for video traffic in case of increased data traffic capturing the channel, we can say there is high correlation between our model and the behavior of the EDCA mechanism.

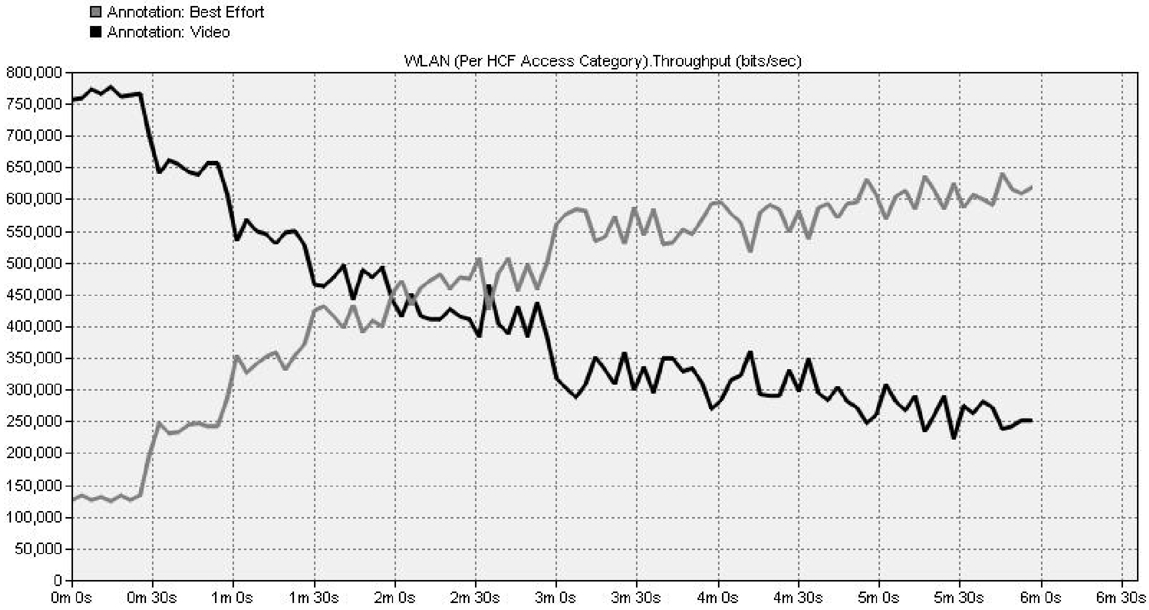

Figure 4.2. The throughput results for OPNET simulation

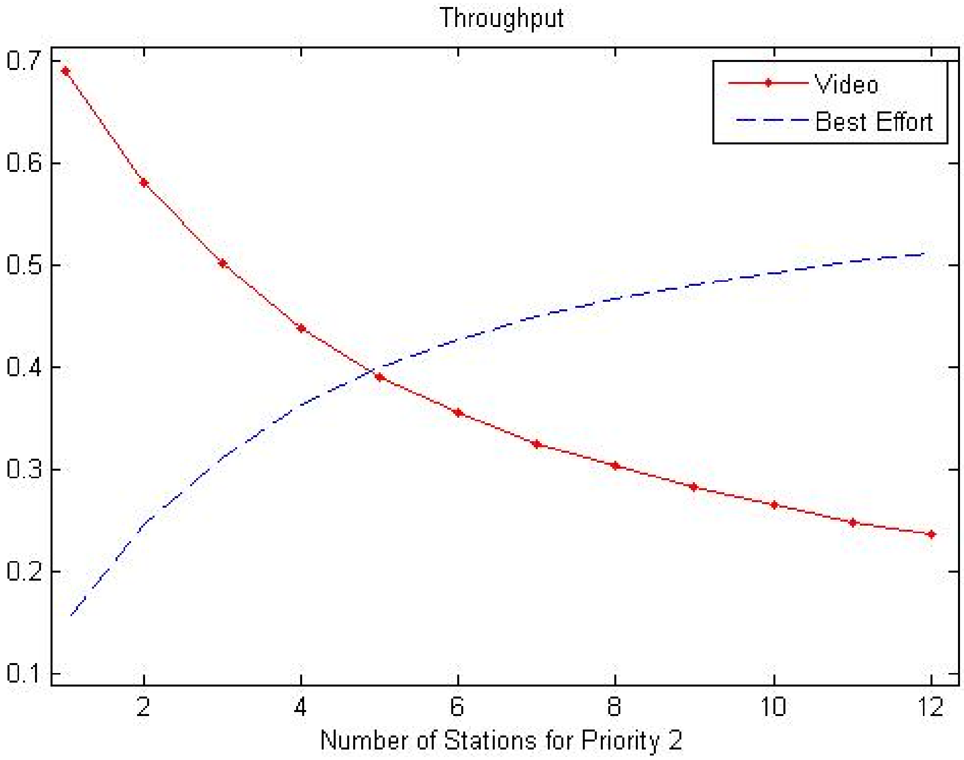

Figure 4.3. The throughput results for Matlab simulation

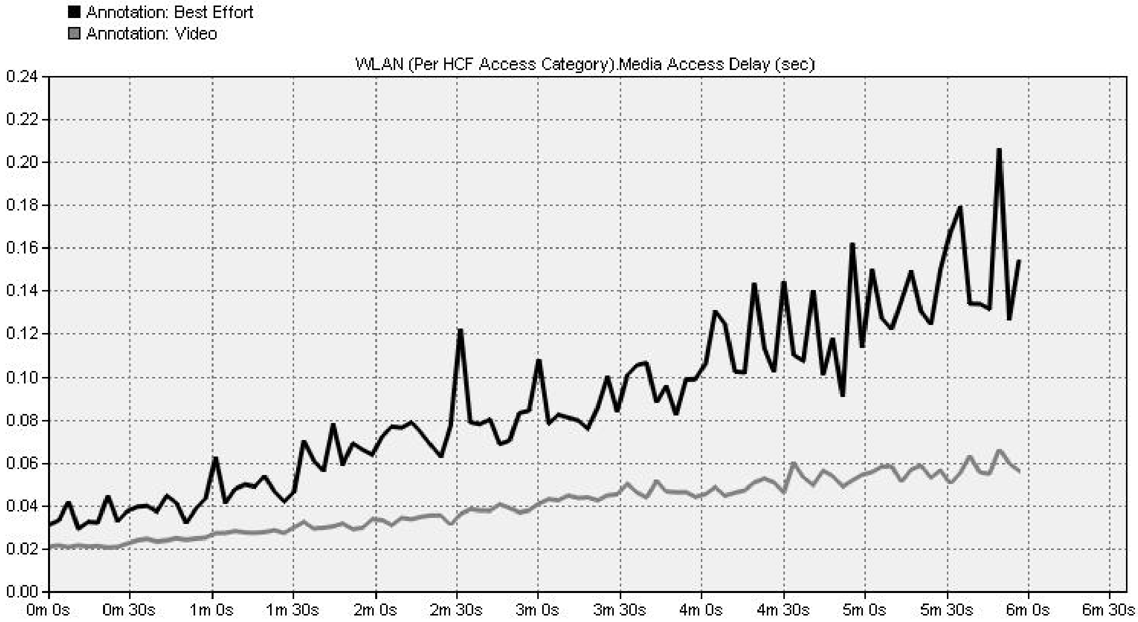

Figure 4.4. The delay results for OPNET simulation

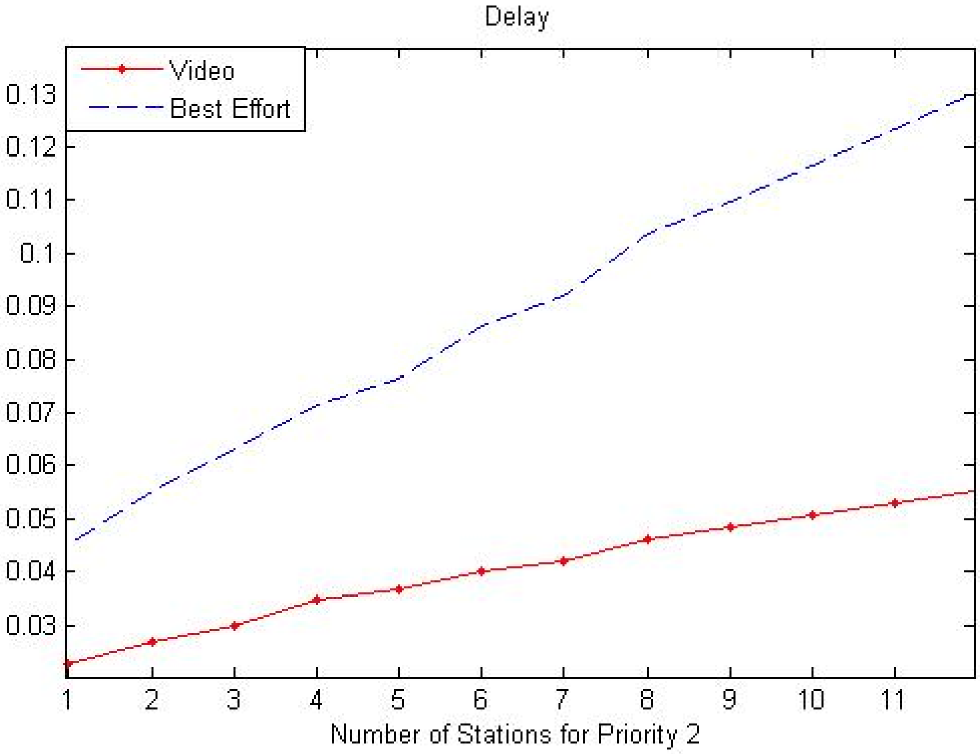

Figure 4.5. The delay results for Matlab simulation

After investigating the correlation between our model and the real time behavior of our model from the throughput point of view, we validate the model in terms of delay. The delay results for both video and best effort data traffic are given in Figure 4.4 and Figure 4.5. Delay for the best-effort traffic is increasing between 40 msec and 130 msec in both cases. Also the delay for the video traffic is increasing from 20 msec to 55 msec for both our model and real-time simulations. As can be seen from the both figures there is a good match between delay results for both simulations.

In this chapter, our objective has been to validate the EDCA model by investigating the correlation between our model and the real time simulations in OPNET. Based on the simulation results, it is possible to say that our model is validated from both throughput and delay point of view.

When the best-effort station number increases, the throughput of the video stations also decreases. After the fifth best-effort station starts transmission, the throughput of the best-effort traffic and video traffic became equal in both cases. From then on, the video throughput continues to decrease, which is an indication of a need for an adaptive system, which redefines the access parameters according to the dynamically changing medium.

5. OPTIMIZATION OF THE CW SIZES FOR VIDEO TRANSMISSION

In the previous chapters, we have developed a model for EDCA mechanism in order to estimate the throughput, delay and collision probabilities for different priorities. After computing the probabilities according to this model and estimating these desired outcomes ($Throughput_i, D_i, P_{coll}$), we have validated these results with the real time behavior of the EDCA mechanism.

In both the validation process and the analytical calculation of the throughput of the video traffic, we have observed that the throughput for video traffic is quite satisfactory compared to data traffic in case of having the same number of stations for both traffic classes. But as the number of the best-effort data traffic station increases, the probability of these stations capturing the channel increases which causes a decrease in the video throughput. Furthermore the increase in the collision probability (P_{coll}) also decreases the throughput for both priorities. This shows that the EDCA mechanism can provide QoS for video transmission from the throughput point of view until the traffic from other stations start to capture the channel and the number of collisions increases.

By increasing the CW sizes for both priorities, it is possible to decrease the collision probability. As can be seen in Figure 5.1, decrease the collision probability, increases throughput for both priorities until throughput reaches optimum value.

The algorithm used in determining the CW sizes is given in Figure 5.2. As the number of the stations trying to access the medium dynamically changes, the optimum CW sizes that maximize throughput are also changing. The increase in the station number in the medium requires higher CW sizes in order to avoid collision but this increase should not exceed optimum size and cause decrease in throughput because of higher $T_{back-off}$.

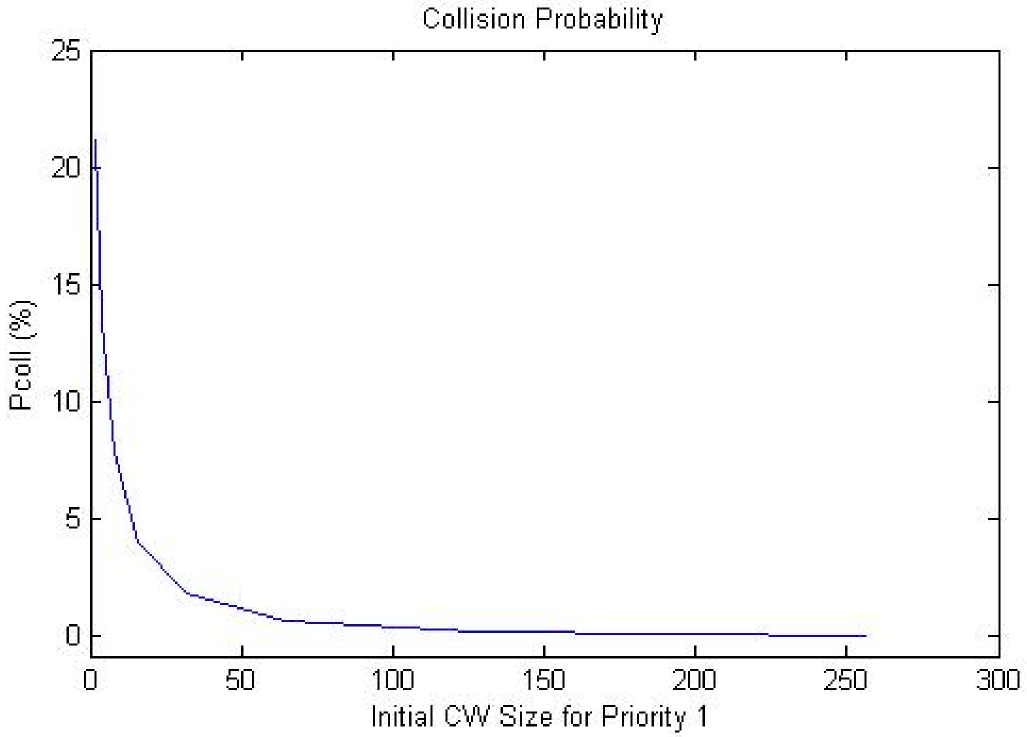

Figure 5.1. Collision probability versus initial CW size for video

5.1. Proposed Adaptive Control

As can be seen from the algorithm in Figure 5.2, the main purpose is to optimize the overall system by decreasing the collision probability under a predetermined value. If P_{coll} is higher than that predetermined value, it means that we are not using the channel with optimum parameters; therefore our first move in the algorithm is to decrease P_{coll} by increasing the initial CW sizes for both priorities. Increase in the CW sizes should not cause a decrease in the total throughput of the system because of increased $T_{back-off}$. For this reason, the initial CW sizes are increased until the throughput reaches its optimum value. On the other hand, this proposed solution of increasing CW sizes for both priorities has a drawback of increasing delay because of the increased average back-off time ($T_{back-off}$).

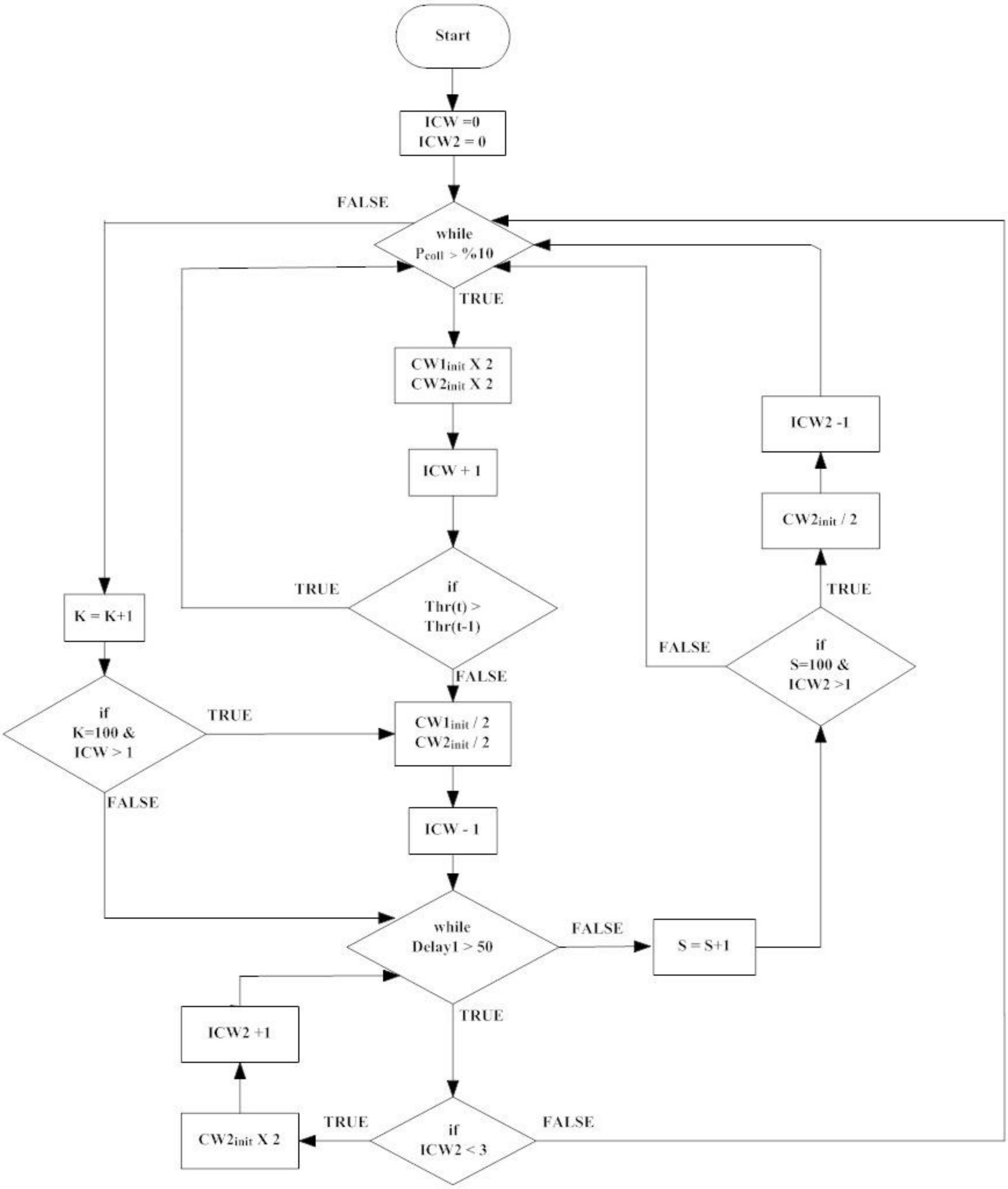

Figure 5.2. Algorithm for optimization of CW sizes

In the algorithm, the upper limit for P_{coll} is defined as 10 percent. When the collision probability is higher than this pre-defined value, the algorithm tries to optimize the CW sizes for both priorities. The collision probability depends on the number of the stations in the medium. If we have a high P_{coll}, this means we need higher CW sizes to avoid collision so as can be seen from the algorithm, initial CW sizes for both priorities are multiplied by 2. After each increase in both initial CW sizes, the ICW counter is increased.

Increase in the CW sizes is limited by the optimum CW sizes for maximum throughput because this approach works well in an interval because as we increase the initial CW sizes, we also increase $T_{back-off}$ time which has a decreasing effect on throughput. If we increase the back-off time length more than its optimum value for the given channel conditions, then the throughput will decrease. So if the CW sizes provide optimum throughput or the P_{coll} is lower than 10 percent, the algorithm will switch to the optimizing delay phase.

After optimizing the overall system for optimizing the total throughput, we are interested in the video traffic in the optimizing delay phase. The cause of an increase in delay or decrease in throughput for video traffic can be due to increased number of data stations in the medium. By the increasing number of data stations in the medium, the performance of the video traffic will be degraded, for this reason we have to increase the priority of video traffic by increasing the CW sizes for lower priority class traffic. By this approach, the video packets will have higher chance of accessing the medium resulting in decreased delay for video traffic. On the other hand, the delay for data traffic will increase while throughput decreases. After each increase in CW sizes for lower priority traffics, the ICW2 counter is increased. In our algorithm, this increase of CW sizes for lower priority traffics is limited to a pre-defined number.

5.2. Performance Evaluation Proposed Algorithm

In this section, we evaluate the performance of the proposed algorithm in a dynamical changing environment. For this purpose, the same simulation parameters defined in Table 3.2 and Table 3.1 are used, but in this simulation the number of best effort data stations are increased from 1 to 100 in order to observe the behavior of the system with increased number of best effort data stations.

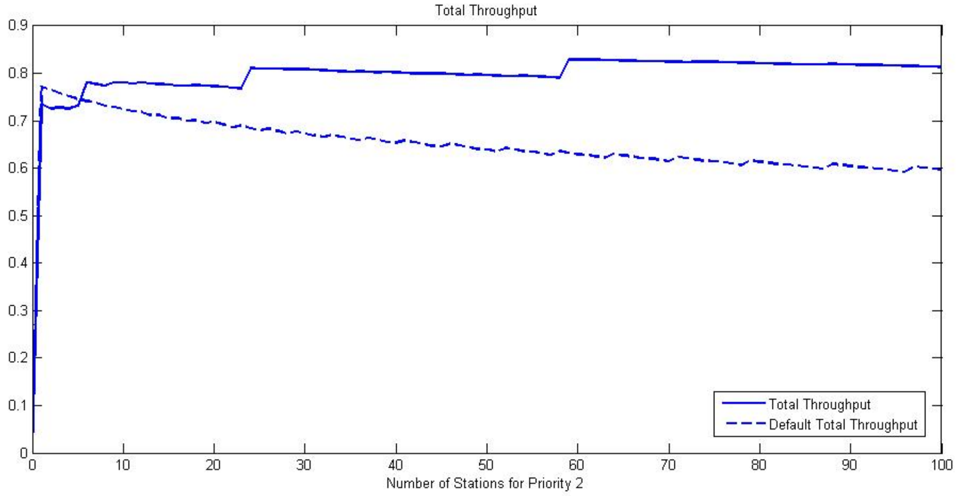

Figure 5.3. Total throughput for default and adaptive CW sizes

In the simulations, the total throughput is calculated for both default and optimized CW sizes. The resulting total throughput for default and adaptive CW sizes are shown in Figure 5.3. We observe a better performance from the total throughput point of view for the optimized CW sizes since our adaptive algorithm adjusts CW sizes according to the collision probability especially when the number of the stations in the medium increases.

In our algorithm, P_{coll} is used in order to optimize the overall system by increasing the total throughput. P_{coll} for both default and optimized cases are shown in Figure 5.4 demonstrating that our algorithm is successful while decreasing the collision probability even with the large number of stations in the medium.

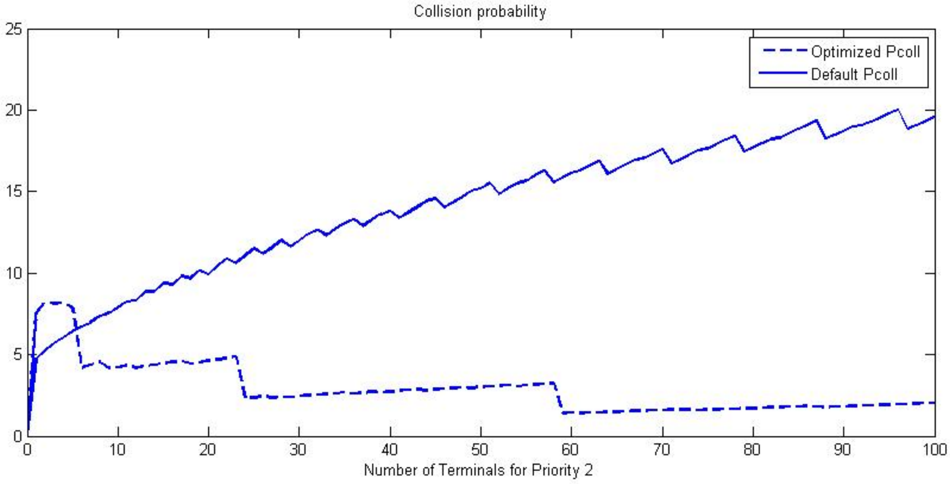

Figure 5.4. Collision probability for default and adaptive CW sizes

In addition to an increase in the total throughput compared to the throughput obtained from the default CW sizes, we also need to determine the effect of this adaptive system on video throughput. For this purpose, we investigate the throughput for both priorities. The resulting graph is given in Figure 5.5 and this graph shows that, using the adaptive system, the video stations can perform better in spite of the large number of data stations in the medium. In addition to improvement in the total throughput, also by increasing the video traffic priority, the video QoS from the throughput perspective becomes quite satisfactory.

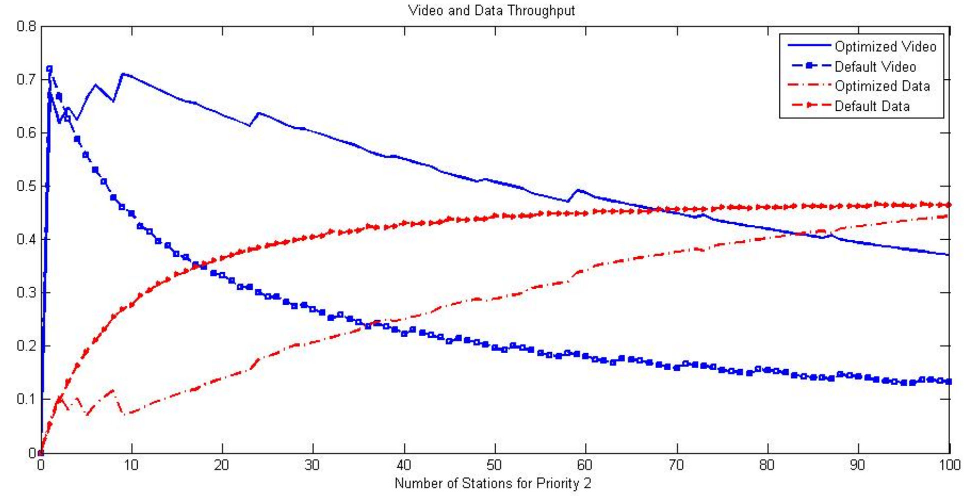

Figure 5.5. Video and data throughput for default and adaptive CW sizes

Finally, in order to see the effect of the adaptive algorithm from the delay perspective, delay analysis is made for both data and video packets. In Figure 5.6 and Figure 5.7 the results of the delay analysis are given.

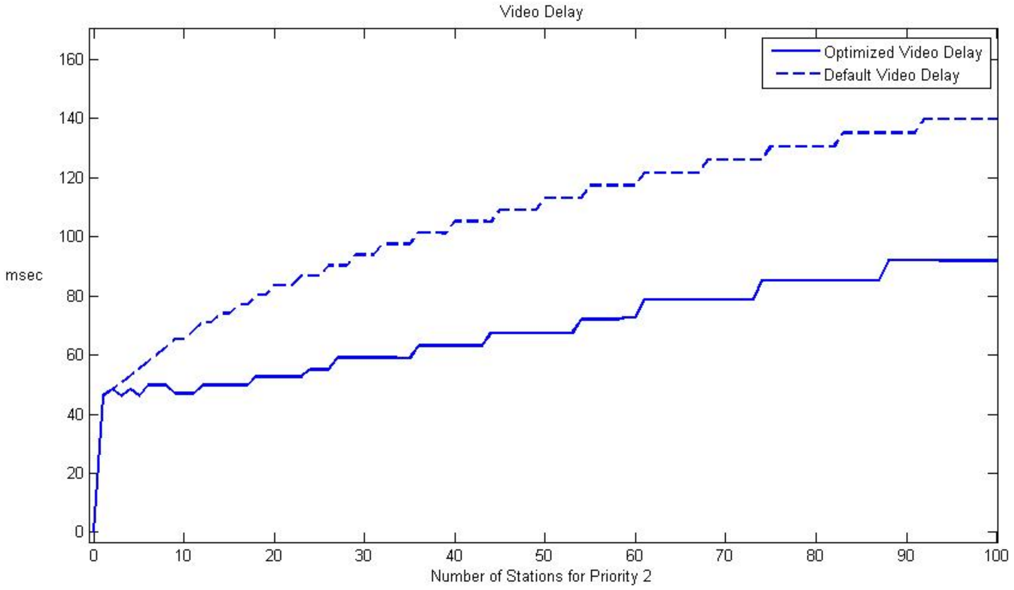

Figure 5.6. Delay for video packets for default and adaptive CW sizes

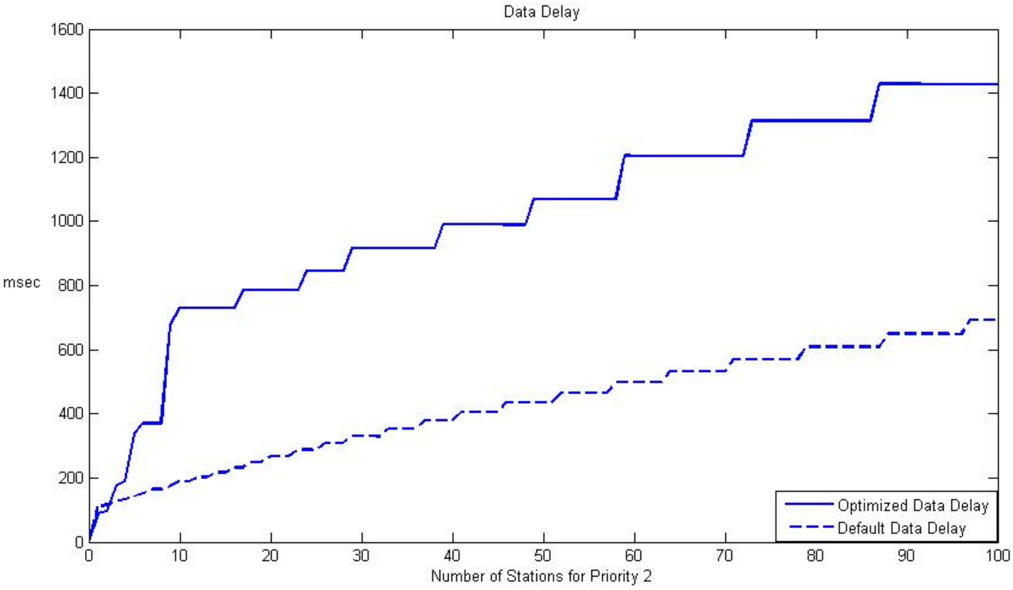

Figure 5.7. Delay for data packets for default and adaptive CW sizes

After optimizing the total system according to the number of the stations in the medium our proposed algorithm decreases delay for video traffic by increasing the priority of video traffic in the delay optimization phase. For this reason, the data packets have to wait more while the video packets have higher chance accessing the channel. As can be seen from the Figure 5.6 and Figure 5.7, the delay for video traffic is decreased while making the data packets wait longer when accessing the channel. This shows that our algorithm is successful in providing QoS from the delay perspective.

6. CONCLUSIONS

In this thesis, an adaptive algorithm for optimizing EDCA parameters according to the QoS requirements for multimedia traffic is proposed. For this purpose delay and throughput estimates are obtained using the analytical model developed for the EDCA back-off procedure. The model takes the traffic prioritization, retry limit and pause intervals during back-off into account. Furthermore the effect of dropped packets on average delay and the idle time for collided transmission are considered in the analytical model.

From the results of the analytical model, it has been observed that a new approach is needed to provide QoS for multimedia packets in a dynamically changing medium with changing number of stations. As the number of stations is increased, the video throughput decreases due to other stations capturing the medium. Moreover also with the increased number of retransmissions and longer back-off times, the delay for multimedia traffic increases.

Rather than increasing the CW sizes for lower priority traffic, the overall system is optimized by decreasing the collision probability. Thus it has been possible to obtain an optimum total throughput. In case of still being under the required QoS limits, the priority of lower class traffic is further decreased. Hence with the use of this adaptive algorithm, it has been possible to achieve much higher throughput and lower delay compared to the default EDCA mechanism.

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