Performance of Reliable Transport Protocol over IEEE 802.11 Wireless LAN: Analysis and Enhancement*

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Abstract- **IEEE 802.11 Medium Access Control(MAC) is proposed to support asynchronous and time bounded delivery of radio data packets in infrastructure and ad hoc networks. The basis of the IEEE 802.11 WLAN MAC protocol is Distributed Coordination Function(DCF), which is a Carrier Sense Multiple Access with Collision Avoidance(CSMA/CA) with binary slotted exponential back-off scheme. Since IEEE 802.11 MAC has its own characteristics that are different from other wireless MAC protocols, the performance of reliable transport protocol over 802.11 needs further study.**

This paper proposes a scheme named DCF+, which is compatible with DCF, to enhance the performance of reliable transport protocol over WLAN. To analyze the performance of DCF and DCF+, this paper also introduces an analytical model to compute the saturated throughput of WLAN. Comparing with other models, this model is shown to be able to predict the behaviors of 802.11 more accurately. Moreover, DCF+ is able to improve the performance of TCP over WLAN, which is verified by modeling and elaborate simulation results.

I. INTRODUCTION

There is an increasing need towards portable and mobile computers or workstations with the development of wireless technology and Internet. Wireless networks need to provide communications between mobile terminals, in addition, access to high speed wired networks needs to be provided too. Wireless Local Area Networks[1-6](WLANs), which provides better flexibility and convenience than their wired counter part, are being developed to provide high bandwidth access for users in a limited geographical area. IEEE Project 802 recommends an international standard 802.11[1-3] for WLANs. The standards include detailed specifications both for Medium Access Control(MAC) Layer and Physical(PHY) Layer.

In WLANs, the physical media, which is shared by all stations and has limited connection range, has significant differences when compared to wired media. The design of WLAN MAC protocol is further complicated by the presence of hidden terminal and capture effects[7,8]. Currently, the IEEE 802.11 WLAN standards include a basic medium access protocol Distributed Coordination Function(DCF) and an optional Point Coordination Function(PCF).

In 802.11, the DCF is the fundamental access method used to support asynchronous data transfer on a best effort basis. As specified in the standards, the DCF must be supported by all the stations in a basic service set(BSS). The DCF protocol is based on Carrier Sense Multiple Access with Collision Avoidance(CSMA/CA). CSMA/CD is not used because a

station is unable to listen to the channel for collision while transmitting. In 802.11 CS is performed both at physical layer, which is also referred to as physical carrier sensing, and at the MAC layer, which is known as virtual carrier sensing. The PCF in the 802.11 is a polling-based protocol, which is designed to support collision free and real time services. This paper focuses on the performance analysis and modeling of DCF in 802.11 WLAN.

There are two techniques used for packet transmitting in DCF. The default one is a two-way handshaking mechanism, also known as basic access method. A positive MAC acknowledgement(ACK) is transmitted by the destination station to confirm the successful packet transmission. The other optional one is a four-way handshaking mechanism, which uses request-to-send/clear-to-send(RTS/CTS) technique to reserve the channel before data transmission. This technique has been introduced to reduce the performance degradation due to hidden terminals. However, the drawback of using the RTS/CTS mechanism is increased overhead for short data frames.

The modeling of 802.11 has been a research focus since the standards has been proposed. Paper [8] considers the effect of capture and hidden terminal and paper [9] gives the theoretical throughput limit of 802.11 based on a *p*-persistent variant. However, none of these captures the effect of the Contention Window(CW) and binary slotted exponential back-off procedure used by DCF in 802.11. Unlike those ones, Paper [10,11] use Markov process to analysze the saturated throughput of 802.11 and show that the Markov analysis works well. We believe that the Markov chain analysis method is fit for examining the performance of IEEE 802.11, which is based on binary slotted exponential backoff. This paper also uses Markov chain and considers the frame retry limits to analyze the saturated throughput, therefore, a more exact model is proposed in this paper.

On the other hand, with the prosperity of Internet, Transport Control Protocol(TCP), which is the widely used reliable protocol in the Internet, is supposed to work well in heterogeneous environment. Since the WLAN MAC has its own characteristics, such as MAC Layer ACK frame, MAC retransmissions, which are different from traditional wireless medium[13-15], the performance evaluation and enhancement will be somewhat different from the research before. The performance of TCP over WLAN is being studied recently[16-20], however, none of these give a TCP performance enhancement based on the WLAN MAC layer solutions.

In fact, when TCP runs over WLAN, where a shared channel is used for multiple access, the forward TCP data and

the backward TCP ACK will compete the channel, which may cause collisions and degrade the overall performance. Meanwhile, 802.11 has been standardized and any proposed enhancement scheme must keep backward compatibility with 802.11, i.e., it can work with 802.11 without introducing performance degradation. This paper introduces a DCF+ to enhance the performance of TCP, which satisfies all the requirements we mention. Our proposed scheme DCF+ is shown to be able to improve the performance of TCP over WLAN both by modeling and simulations.

This paper is organized as following. Section 2 briefly describes the DCF of IEEE 802.11 MAC protocols, which includes both basic access and RTS/CTS mechanism. A scheme named DCF+ is introduced in section 3, which is compatible with DCF and designed to enhance the TCP performance over WLAN. In section 4, an analytical model to compute the saturated throughput of DCF is proposed. Section 5 validates the accuracy of this model by simulations. DCF+ is analyzed in section 6. TCP performance over WLAN, both on DCF and DCF+ has been examined in section 7. Finally, section 8 concludes the paper.

II. DISTRIBUTED COORDINATION FUNCTION IN 802.11

The basic service set(BSS) is the fundamental building block of IEEE 802.11 architecture. The geographical area covered by the BSS is known as the basic service area(BSA), which is similar to a cell in a cellular network. IEEE 802.11 supports both the ad hoc network and infrastructure network architecture. This paper only give a brief introduction of 802.11 DCF, the readers are referred to [1-3] for detailed information about 802.11.

The DCF is based on CSMA/CA and it only provides asynchronous access for best effort data transmission. DCF consists of both a basic access method and an optional channel access method using RTS/CTS exchanges.

A. The basic access method

In 802.11, priority access to the wireless medium is controlled by the use of inter-frame space(IFS) time between the transmission of frames. Totally three IFS intervals have been specified by 802.11 standard: short IFS(SIFS), point coordination function IFS(PIFS), and DCF-IFS(DIFS). The SIFS is the smallest and the DIFS is the largest.

The station may proceed with its transmission if the medium is sensed to be idle for an interval larger than the Distributed Inter Frame Space(DIFS). If the medium is busy, the station defers until a DIFS is detected and then generate a random back-off period before transmitting. The back-off timer counter is decreased as long as the channel is sensed idle, frozen when the channel is sensed busy, and resumed when the channel is sensed idle again for more than a DIFS. A station can initiate a transmission when the back-off timer reaches zero. The back-off time is uniformly chosen in the range $(0, w-1)$. Also $(w-1)$ is known as Contention Window(CW), which is an integer with the range determined by the PHY characteristics CW_{min} and CW_{max}. After each unsuccessful transmission, w is doubled, up to a maximum value $2^{m'}W$, where W equals to $(CW_{min}+1)$ and $2^{m'}W$ equals to $(CW_{max}+1)$.
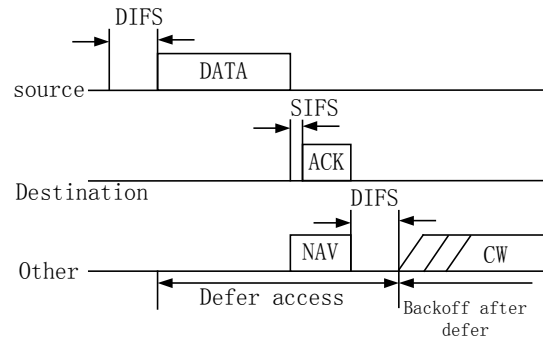

Fig.1 Basic access mechanism in DCF

Upon having received a packet correctly, the destination station waits for a SIFS interval immediately following the reception of the data frame and transmits a positive ACK back to the source station, indicating that the data packet has been received correctly(Fig.1). In case the source station does not receive an ACK, the data frame is assumed to be lost and the source station schedules the retransmission with the CW for back-off time doubled. When the data frame is transmitted, all the other stations hearing the data frame adjust their Network Allocation Vector(NAV), which is used for virtual CS at the MAC layer, based on the duration field value in the data frame received correctly, which includes the SIFS and the ACK frame transmission time following the data frame.

B. The RTS/CTS access method

In 802.11, DCF also provides an optional way of transmitting data frames that involve transmission of special short RTS and CTS frames prior to the transmission of actual data frame. As shown in Fig.2, an RTS frame is transmitted by a station, which needs to transmit a packet. When the destination receives the RTS frame, it will transmit a CTS frame after SIFS interval immediately following the reception of the RTS frame. The source station is allowed to transmit its packet only if it receives the CTS correctly. Note that all the other stations are capable of updating the NAVs based on the RTS from the source station and the CTS from the destination station, which helps to combat the *hidden terminal* problems. In fact, a station able to receive the CTS frames correctly, can avoid collisions even when it is unable to sense the data transmissions from the source station. If a collision occurs with two or more RTS frames, much less bandwidth is wasted when compared with the situations where larger data frames in collision.
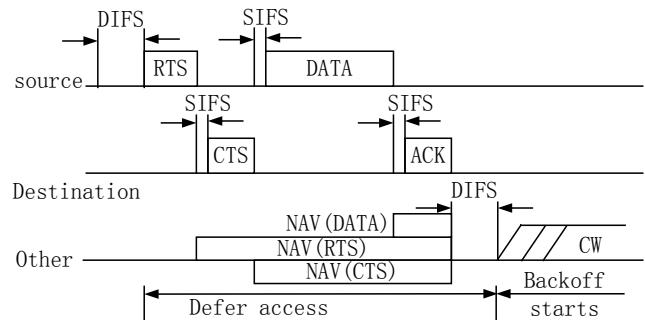

Fig.2 RTS/CTS access mechanism in DCF

III. ILLUSTRATIONS OF DCF+

This paper introduces a new scheme to improve the performance of reliable transport protocol over WLAN, such as TCP, which needs to receive the Transport layer acknowledgement (ACK) on the backward direction. In the scenario of TCP over WLAN where a shared channel is used for multiple access, the forward TCP data and the backward TCP ACK will compete the channel, which may cause collisions and degrade the overall performance. Our proposed scheme is shown to be able to improve the performance of TCP over WLAN both by modeling analysis and simulations.

Since our scheme is based on DCF and can be regarded as an enhancement for reliable transfer or two-way traffic over shared media wireless channel, we name this scheme DCF+. Note that DCF+ is fully compatible with DCF, i.e., in a wireless LAN, if some stations support DCF+ while others not, they can coexist and transfer data traffic to each other. The access method in DCF+ can be considered as a data exchange on the backward direction after the original data exchange on the forward direction, which may use either basic access method or optional RTS/CTS exchange method.
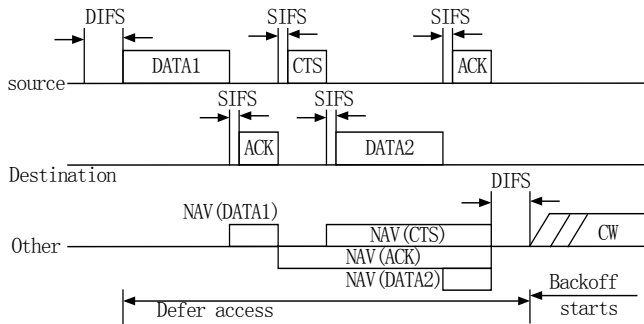

Fig.3 DCF+ starts with basic access mechanism

DCF+ works as following: suppose that the source station starts with basic access method to compete the channel(Fig.3), when the data packet(DATA1 in Fig.3) arrives at the destination station and currently the destination has a packet(DATA2 in Fig.3) to the source which sends DATA1, it needs to send an ACK frame to the original source station. In DCF+, the duration field in the MAC header is also used to set the NAV value as that in DCF, so the destination station needs to set the NAV of other stations by setting the duration field on the ACK field. When such an ACK arrives at the source, the source will reply with a CTS, which is used to set the NAV in the receiving range of next data(DATA2 in Fig.3) receiver--- the original source station. Then the destination could transfer the data packet (DATA2 in Fig.3) to the source station, and the source will reply with a normal ACK. Note that in Fig.3, all the NAVs setting in the receiving range of source station is shown above the horizontal line and NAVs setting in the destination receiving range is below the line.

Note that the first ACK in the procedure acts as an RTS sending by the destination station; therefore, the second data transfer from the destination to the source always deals with the hidden terminal issue as in RTS/CTS access method.

Also, the first ACK in the procedure is a normal ACK for the source destination if the source station only supports 802.11 DCF, not DCF+. This frame must be an ACK because

in 802.11, it states that in DCF when a station transmit a data frame, it must receive a MAC ACK frame from the destination station, all the other frames will discarded even if it is a data frame from the destination with some enhancement, e.g., MAC ACK piggybacked in the data frame. Therefore, in our DCF+, we choose to use an ACK frame after the first data frame to keep the backward compatibility with 802.11 DCF.
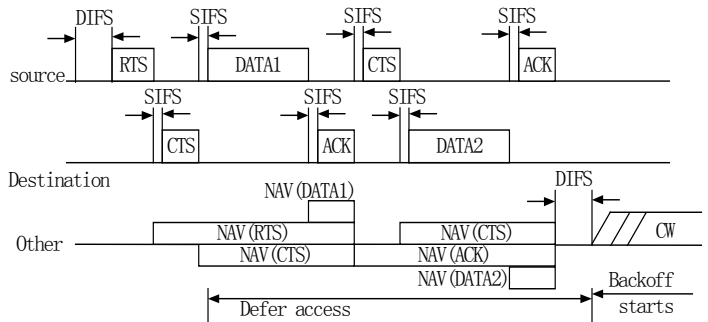

Fig.4 DCF+ starts with RTS/CTS access mechanism

If the frame exchange starts with RTS/CTS access method, the procedure is similar, which is shown in Fig.4.

Since all the frames introduced in DCF+ has been standardized in 802.11 DCF, therefore, even if other stations only support DCF, not DCF+, the frame exchange will not be disturbed and the performance will not be degraded. Meanwhile, stations only supporting DCF and stations supporting DCF+ still can exchange frames by using DCF. Therefore, the backward compatibility is guaranteed. However, two issues are still non-trivial and we would discuss as following.

First, in DCF+ we assume that the destination station has a data frame ready to be transmitted to the source, but that is not always the situation. The destination station will always send an ACK after it receives the DATA1 frame correctly. Therefore, upon receiving an ACK, the source station using DCF+ must determine whether it needs to send the CTS to reserve channel for the second data frame. In this paper, it is assumed that by examining the duration field of the ACK frame received; the source can determine whether the destination station has a data frame ready to send.

Second, consider a scenario where the destination station uses DCF+, but the source station only supports DCF. Supposing that whether or not the source station supports DCF+ is unknown at the destination station, then the destination station may reserve the channel by the ACK and the bandwidth may be wasted. We propose to solve this issue by the following alternative ways: 1) A DCF+ station can make a record to determine whether another station is DCF+ capable. We assume that a station only makes record for the stations with which it has data exchanges. 2)Some reserved fields in the data frame can be used to indicate the source station is DCF+ capable, otherwise it is not. For example, the reserved subtype value for a data frame can be used to fulfill this function.

IV. PERFORMANCE ANALYSIS FOR DCF

In this paper, we focus on the *saturated throughput*, which is also examined in paper [11]. This is a fundamental

performance figure defined as the limit reached by the system throughput as the offered load increases, and it represents the load that the system can carry in stable conditions.

The key contribution of this paper is the analytical evaluation of the saturated throughput, in the assumption of ideal channel conditions. Also, the Markov model in paper[11] does not consider the frame retry limits, thus it may overestimate the throughput of 802.11. Our model is based on that in paper[11]. In our analysis, we assume a fixed number of stations, each one always has a packet available for transmission. To make it easy to compare with the model in paper [11], we use the same symbols and variables used in it. The analysis includes two parts: 1)With a Markov chain, the behavior of a station is examined, which we use to get the stationary probability τ that the station transmit a packet; 2)The throughput of both basic and RTS/CTS access methods is examined.
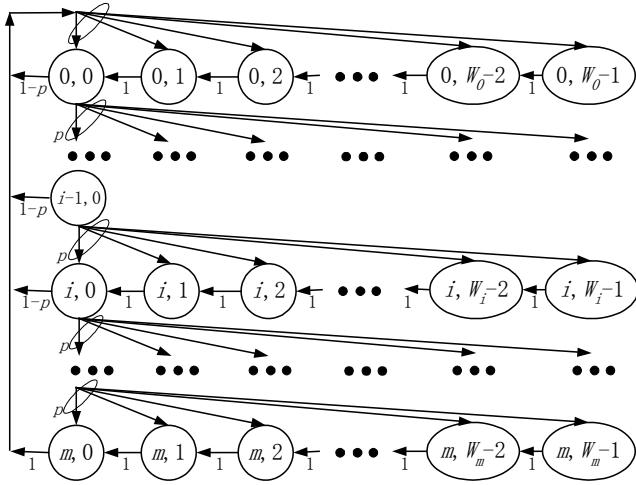
A. Markov Chain Model

Fig.5 Markov chain model of new back-off window scheme

We use the same assumption in paper [11] for our analysis. The contending stations are supposed to be a fixed number, *n*. Let $b(t)$ be the stochastic process representing the back-off window size for a given station at slot time *t*. Note that the slot time is referred to as the constant value σ and the variable time interval between two consecutive backoff time counter decrements[11]. As in paper [11], the key approximation in this model is that the probability *p* that a transmitted packet collides is independent on the state $s(t)$ of the station. Thus, the bi-dimensional process $\{s(t), b(t)\}$ is a discrete-time Markov chain, which is shown in Fig.5.

This paper will use all the parameters assigned for Direct Sequence Spread Spectrum(DSSS) PHY in 802.11; for other PHY layers, the analysis process is similar. In DSSS, CW_{min} and CW_{max} equal to 31 and 1023 respectively. Therefore, we have

$$\begin{cases} W_i = 2^i W & i \leq m' \\ W_i = 2^{m'} W & i > m' \end{cases} \quad (1)$$

where $W=(CW_{min}+1)$, and $2^{m'}W = (CW_{max}+1)$, so for DSSS, we have $m' = 5$.

Unlike paper [11], here we use *m* to represent *maximum backoff stage*. As specified in 802.11[1] this value could be larger than *m'*, while the CW will be hold after that, which is shown is equation (1). In fact, here *m* also means the maximum retransmission count, which is different for data frame and RTS frame, i.e., 5 and 7 respectively[1]. Paper [11] does not distinguish those two cases. The key difference between paper [11] and this one is that the Markov chain models are different, which is because our model considers the effects of frame retransmitting limit.

In this Markov chain, the only non-null one-step transition probabilities are[2]

$$\begin{cases} P\{i,k \mid i,k+1\} = 1 & k \in [0, W_i-2] \; i \in [0,m] \\ P\{0,k \mid i,0\} = (1-p)/W_0 & k \in [0, W_0-1] \; i \in [0,m-1] \\ P\{i,k \mid i-1,0\} = p/W_i & k \in [0, W_i-1] \; i \in [1,m] \\ P\{0,k \mid m,0\} = 1/W_0 & k \in [0, W_0-1] \end{cases} \quad (2)$$

These transition probabilities account, respectively, for: 1)the decrements of the backoff timer; 2)after a successful transmission, the backoff timer of the new packet starts from the backoff stage 0; 3)an unsuccessful transmission makes the backoff stages increase; 4)at the maximum backoff stage, the CW will be reset if the transmission is unsuccessful or restart the backoff stage for new packet if the transmission is successful.

Let $b_{i,k}$ be the stationary distribution of the Markov chain. First note that

$$b_{i-1,0} * p = b_{i,0} \qquad 0 < i \leq m \quad (3)$$

we have $\qquad b_{i,0} = p^i b_{0,0} \qquad 0 \leq i \leq m \quad (4)$

Since the chain is regular, so for each $k \in (0, W_i-1)$, we have

$$b_{i,k} = \frac{W_i-k}{W_i} \begin{cases} (1-p)\sum_{j=0}^{m-1} b_{j,0} + b_{m,0} & i = 0 \\ \\ pb_{i-1,0} & 0 < i \leq m \end{cases} \quad (5)$$

With (4) and transitions in the chain, equation (5) can be simplified as

$$b_{i,k} = \frac{W_i-k}{W_i} b_{i,0} \quad 0 \leq i \leq m \quad (6)$$

Therefore, by using the normalization condition for stationary distribution, we have

$$1 = \sum_{i=0}^{m} \sum_{k=0}^{W_i-1} b_{i,k} = \sum_{i=0}^{m} b_{i,0} \sum_{k=0}^{W_i-1} \frac{W_i-k}{W_i} = \sum_{i=0}^{m} b_{i,0} \frac{W_i+1}{2} \quad (7)$$

Using equation (1)(6)(7), we have equation (8), which is shown on top of next page.

Now the probability τ that a station transmits in a randomly chosen slot time can be expressed as,

$$\tau = \sum_{i=0}^{m} b_{i,0} = \frac{1-p^{m+1}}{1-p} b_{0,0} \quad (9)$$

where $b_{0,0}$ can be obtained from equation (8).

$$b_{0,0} = \begin{cases} \dfrac{2(1-2p)(1-p)}{W(1-(2p)^{m+1})(1-p)+(1-2p)(1-p^{m+1})} & m \leq m' \\[4mm] \dfrac{2(1-2p)(1-p)}{W(1-(2p)^{m'+1})(1-p)+(1-2p)(1-p^{m+1})+W2^{m'}p^{m'+1}(1-2p)(1-p^{m-m'})} & m > m' \end{cases} \quad (8)$$

In the stationary state, a station transmits a packet with probability τ, so we have

$$p = 1-(1-\tau)^{n-1} \qquad (10)$$

Therefore, equations (8)(9) and (10) represent a nonlinear system in the two unknowns τ and p, which can be solved by numerical results. Note that we must have $p \in (0,1)$ and $\tau \in (0,1)$.

Since the Markov chain transitions in Fig.5 are different from that in paper [11], the results obtained for $b_{0,0}$ is different from that in paper [11], so do τ and p.

B. Throughput Analysis

Let P_{tr} be the probability that there is at least one transmission in the considered slot time. And let P_s be the probability that a transmission is successful, given the probability P_{tr}. So we have

$$P_{tr} = 1-(1-\tau)^n \qquad (11)$$

$$P_s = \frac{n\tau(1-\tau)^{n-1}}{P_{tr}} = \frac{n\tau(1-\tau)^{n-1}}{1-(1-\tau)^n} \qquad (12)$$

Now we are able to express the normalized system throughput S as the ratio,

$$S = \frac{E[\text{Payload Information in a slot time}]}{E[\text{Length of a slot time}]}$$
$$= \frac{P_s P_{tr} E[P]}{(1-P_{tr})\sigma + P_s P_{tr} T_s + (1-P_s)P_{tr}T_c} \qquad (13)$$

where we use the same symbols as those in paper [10]. Here, T_s and T_c are the average time the channel is sensed busy because of a successful transmission or a collision respectively. The $E[P]$ is the average packet length and σ is the duration of an empty slot time.

Let packet header be $H = PHY_{hdr} + MAC_{hdr}$ and let propagation be δ. Then we must have the following expression, which is different from that in paper[11] because we consider the ACK timeout effect.

$$\begin{cases} T_s^{bas} = DIFS + H + E[P] + \delta + SIFS + ACK + \delta \\ T_c^{bas} = DIFS + H + E[P^*] + SIFS + ACK \end{cases} \qquad (14)$$

where *bas* means basic access method and $E[P^*]$ is the average length of the longest packet payload involved in a collision. In all our cases, all the packets have the same fixed size, therefore, we have $E[P]=E[P^*]=P$.

For the RTS/CTS access method, assuming that all the station use the RTS/CTS for the data frame for simplicity, then we have

$$\begin{cases} T_s^{rts} = DIFS + RTS + SIFS + \delta + CTS + SIFS + \delta + H \\ \qquad + E[P] + SIFS + \delta + ACK + \delta \\ T_c^{rts} = DIFS + RTS + SIFS + CTS \end{cases} \qquad (15)$$

where *rts* means RTS/CTS access method. Note that here we suppose collision only occurs between RTS frames and T_c^{rts} is different from that in paper [11] because we consider the CTS timeout effect.

V. MARKOV MODEL VALIDATION

This paper uses the well-known simulation tool *NS*-2[12] from Lawrence Berkeley National Laboratory. To validate our model, we will compare the results with that obtained in paper [11].

Also, this paper assumes each station has enough data to send to obtain the saturated throughput performance of the new backoff scheme. We will vary the number of stations to see the effect of throughput degradation due to increased collision probability.

All the parameters used in analytical model and our simulations follow the parameters in paper [2] for DSSS, and are summarized in table.1. Note that we assume the application data payload is 1000bytes, IP header and UDP header are 20 and 8 bytes, so packet payload at MAC layer is 1028bytes.

Packet Payload	8224bits
MAC header	224bits
PHY header	192bits
ACK	112bits+PHY header
RTS	160bits+PHY header
CTS	112bits+PHY header
Channel bit rate	1Mbps
Propagation delay	1us
Slot time	20us
SIFS	10us
DIFS	50us

Tabel.1 System parameters for MAC
and DSSS PHY Layer

Our MAC Markov model equations are independent of the parameters; so it does not matter when choosing parameters for different PHY layers.

A. Simulation results for basic access method

First we see the results of basic access method, which is shown in Fig.6. Here we use *new model* to represent the model in this paper and *old model* to represent the model in paper [11]. For a fixed number of stations, we run 10 simulations with different random seed. Each symbol "+" represents a simulation result. Note for some simulation series, some symbols are superposed because those results are very close to each other.

From the figure we are able to see that the analytical model of this paper is more accurate than that in paper [11]. The model in paper [11] overestimates the results of 802.11 because it does not consider the retry limit in the Markov chain transitions and timeout in the throughput analysis. On the

country, our analysis results match the simulation results closely especially when the number of stations is large, which follows our assumption when the Markov chain is formed, i.e., that the probability p that a transmitted packet collides is independent of the state $s(t)$ of the station is accurate when the number of stations is large.
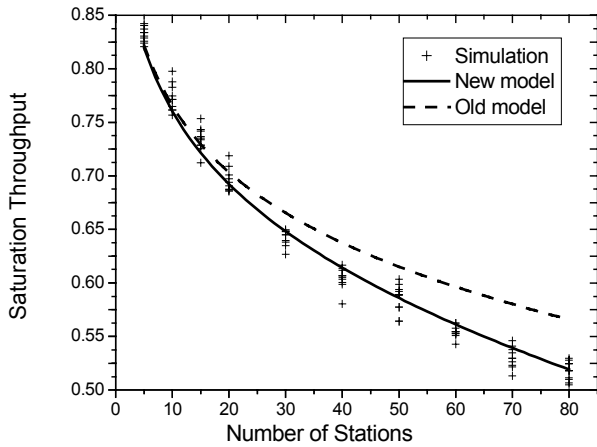

Fig.6 Analysis versus simulations: basic access method

B. Simulation results for RTS/CTS access method

The results comparison of RTS/CTS access method is similar to that of basic method. Note that in Fig.7, the vertical axis scale is different to that in Fig.6. From this figure, we are able to conclude that RTS/CTS access method is useful to compensate the performance degradation due to collision, whose probability increases with the number of stations. Note that we can get these results because in this paper the packet payload length, 1028bytes, is large enough to compensate the overheads introduced by RTS/CTS. Note that in Fig.7 our model still overestimates the throughput. It is because there are some routing packets, which are transmitted by broadcast and does not use RTS/CTS handshaking. The number of routing packets increases with the number of the stations.
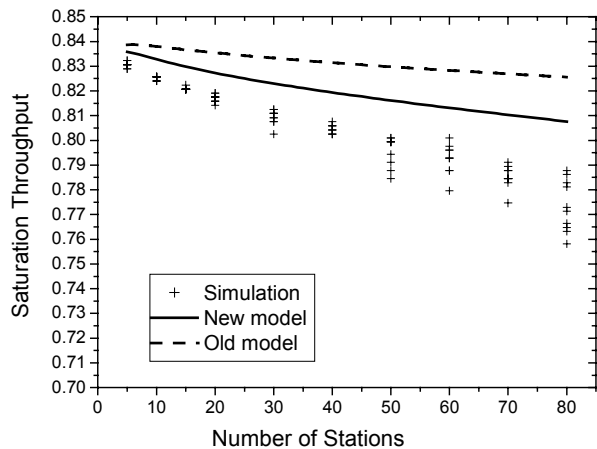

Fig.7 Analysis versus simulations: RTS/CTS access method

Therefore, the accuracy of our Markov model has been validated by simulations, we will use it as a tool to analyze the performance of DCF+.

VI. DCF+ ANALYSIS

In this section, we will use the Markov chain model to analyze the performance of DCF+. Note that the destination station does not always has a packet for the source station. In such scenario, the access procedure is the same as that in DCF. For analysis simplicity, here we assume that the destination always has such a packet to transfer. Therefore, the DCF+ throughput performance achieved in this section is actually the upper bound of DCF+ for two-way traffic. We will examine the real scenario where TCP over DCF and DCF+ and compare the results in the next section.

This paper will use TCP as the analysis reliable transport protocol and suppose there is no delay ACK used in the destination, that is, a TCP data packet always trigger a TCP ACK packet transfer on the backward direction. The application data packet is segmented at the TCP layer, each segment contains 1000 bytes, so a TCP data packet arrives from the IP layer to the link layer is 1040 bytes, 40 bytes for IP and TCP header overheads totally. The TCP ACK packet is supposed to be 40 bytes long, with no overhead introduced for options.

Suppose the packet length arrives from the high layer to the MAC layer has an probability distribution function(PDF) $F(x)$, for simplicity we assume that TCP sending window is large enough, thus the probability of data packet arriving at MAC layer and ACK packet arriving is the same, then in our cases, we have

$$F(x) = \begin{cases} 0 & x < 40 \\ 1/2 & 40 \le x < 1040 \\ 1 & x \ge 1040 \end{cases} \qquad (16)$$

For simplicity, supposing the probability of three or more packets simultaneously colliding can be neglected, then the longest packet length for two packets in collision has the PDF as following

$$F*(x) = F^2(x) \qquad (17)$$

Then the analysis procedure can be repeated similarly as those for DCF in section III. This paper gives the analysis and simulation results for DCF+ in Fig.8 and Fig.9, for access starting with basic access and RTC/CTS exchange. Note that here we use UDP to generate enough traffic to satisfies all our assumptions for analysis simplicity. The performance of real TCP over DCF+ and DCF will be examined by elaborate simulations in the next section.

From the results we can see that our scheme DCF+ can improve the throughput performance of WLAN. Also from figures we can see the results of DCF have much larger variation than those of DCF+, especially in the RTC/CTS exchange case. Therefore, we can conclude that our DCF+ scheme has more stable performance comparing with DCF; meanwhile, the throughput has been enhanced.
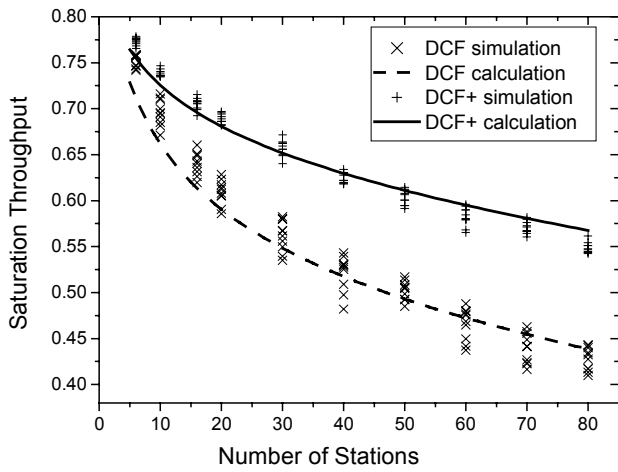
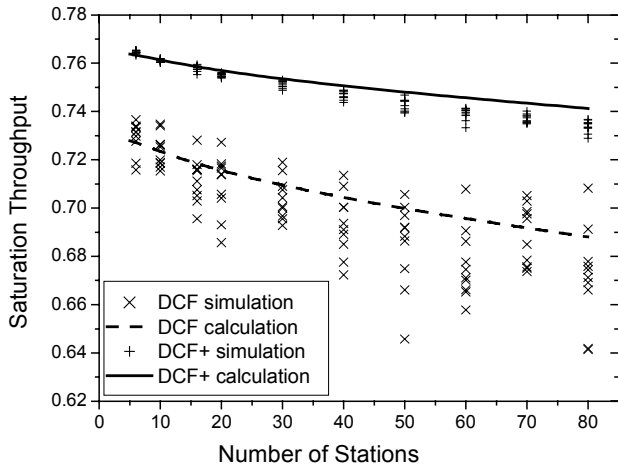Fig.8 Simulation and analysis for DCF+ starting with basic access mechanism

Fig.9 Simulation and analysis for DCF+ starting with RTS/CTS access mechanism

VII. PERFORMANCE OF TCP OVER WLAN

In this section, we will examine the performance of TCP over WLAN, both over DCF and DCF+. The performance is classified into three categories: goodput, fairness and delay. The TCP segment size is set to 1460bytes, so the packet arriving at MAC layer is 1500bytes, including IP and TCP header. The bandwidth of WLAN is set to 2Mbps. Other parameters are kept the same as those in table.1.

A. Goodput of TCP over WLAN

First we examine the throughput performance of TCP. For a number of stations, we run 10 simulations with different seed, and each result is marked by a symbol, i.e., "+" for DCF+ and "x" for DCF. The average value of those 10 series simulations are linked by line. Note that here the goodput is collected at application layer, so it does not take the retransmission traffic into account. Therefore, it is called as *goodput* to distinguish it from the throughput.

From Fig.10 and Fig.11, we can see that DCF+ can improve the performance of TCP over WLAN. Comparing the results in Fig.10-11 to those in Fig.8-9, we can see that our analysis model is accurate to predict the results of DCF+, especially in basic access method. Also, we find that due to increased collision probability, the performance of TCP over WLAN degrades when the number of stations increases, especially when RTS/CTS is used.

Therefore, we can conclude that although the MAC layer retransmissions can reduce the affect of collisions to high layer reliable transport protocols, the performance of TCP still degrades fast when the competition at MAC increases, especially in the case of RTS/CTS. Note that RTS/CTS has been considered a way to deal with hidden terminals; it can also be used for collision resolution. In our case, it cannot hide the MAC layer competition to high layers well, when TCP is running over WLAN.
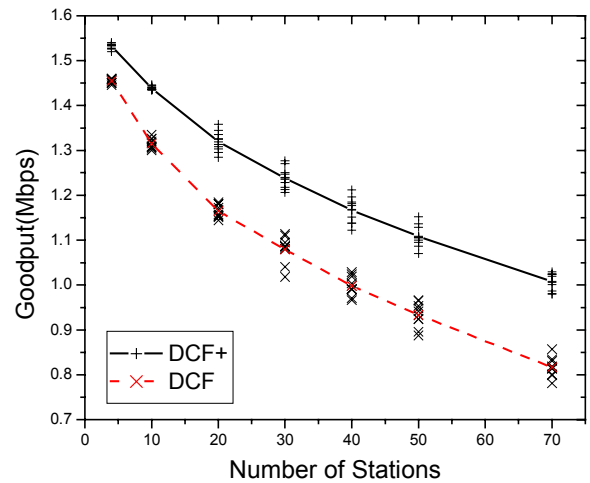

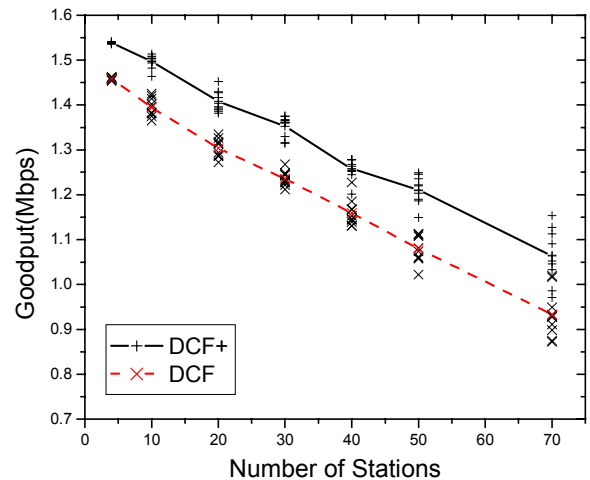Fig.10 Goodput of TCP over WLAN: basic access

Fig.11 Goodput of TCP over WLAN: RTS/CTS

B. Fairness of TCP over WLAN

Here we use a metric called *fairness index*[21] for the goodput measured at the receiver. The fairness index, *f*, is defined as follows: if there are *n* concurrent connections in the

network and the goodput achieved by connection i is equal to x_i, and $1 \le i \le n$, then

$$f = (\sum_{i=1}^{n} x_i)^2 / (n \sum_{i=1}^{n} x_i^2) \qquad (18)$$

The fairness index is always a non-negative value and lies between 0 and 1. The closer the value is to 1, the better fairness.
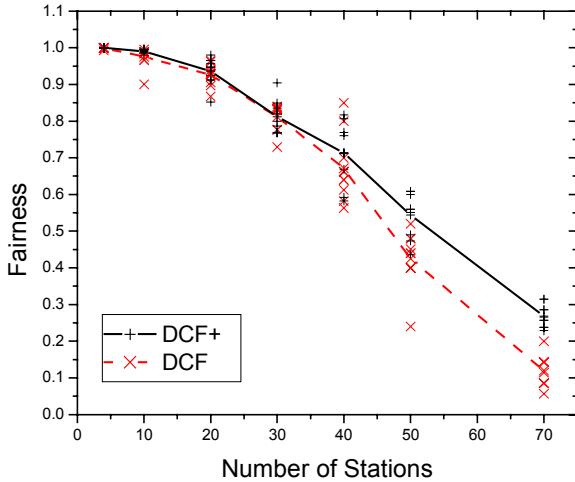

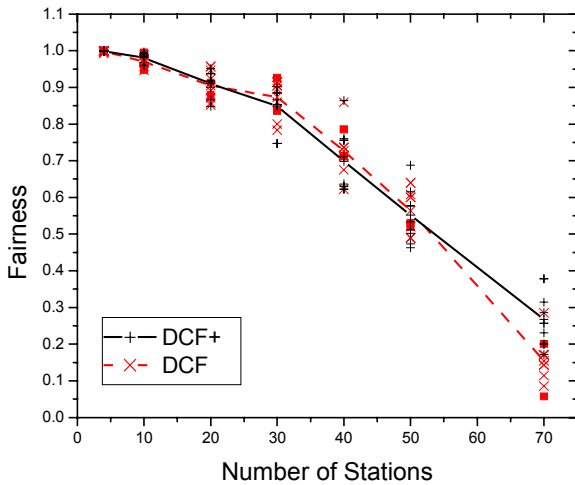Fig.12 Fairness index of TCP over WLAN: basic access

Fig.13 Fairness index of TCP over WLAN: RTS/CTS

From Fig.12-13, we can see the fairness index for the goodput achieved by each TCP connection. The fairness index under DCF+ is a little bit higher than that under DCF, although both of the results are poor when the number of stations is large. Note that the fairness of TCP over WLAN with large number of connections is very poor because some connections are almost starved by other connections. When the connection number increase, the average goodput for each connection decreases. Thus, the normalized variation increases greatly and the fairness degrades severely.

Although RTC/CTS can improve the goodput performance when comparing with basic access method, it cannot improve the fairness index simultaneously.

Therefore, we can conclude that the fairness that can be achieved is related to the competition at the MAC layer. When the number of stations increases, the fairness index performance degrades severely, both for DCF and DCF+.

C. Delay introduced at MAC layer

When DCF+ is introduced, people may worry that our DCF+ may increase the delay for other stations waiting for transmission because the data exchange procedure is longer in DCF+ than that in DCF.

The delay introduced can be classified into two categories: 1) MAC layer access delay, which also includes the delay for data transmissions and retransmissions; 2) delay at Interface Queue(IFQ), which is the queueing delay introduce at link layer(LL) queue.
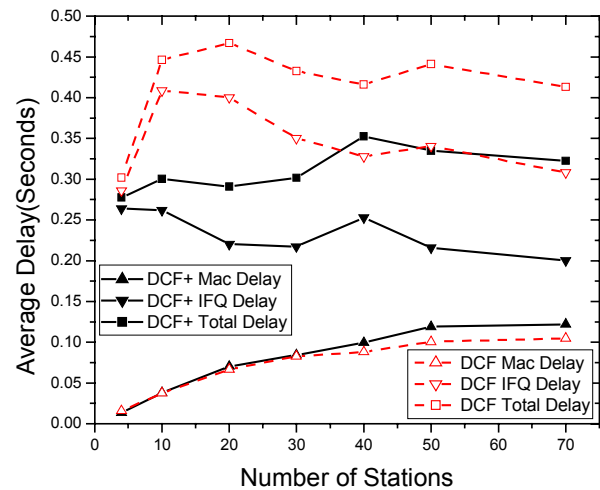

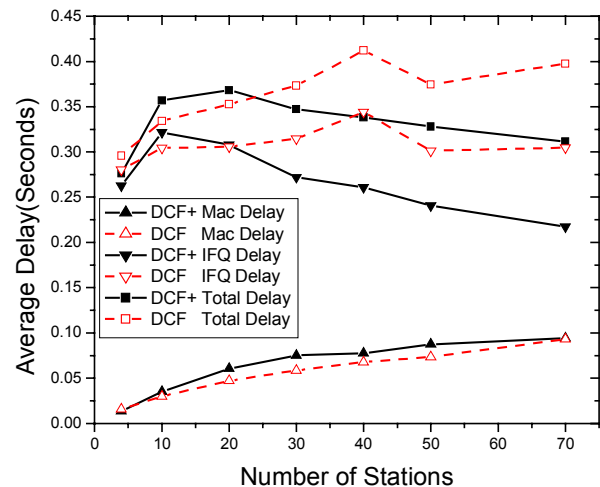Fig.14 Delay at MAC of TCP over WLAN: basic access

Fig.15 Delay at MAC of TCP over WLAN: RTS/CTS

From Fig.14-15, we can see the delay when TCP is over WLAN. Clearly, the IFQ delay is much more larger than that of MAC access delay. Although our DCF+ slightly increases the MAC access delay, the IFQ delay is reduced greatly at the same time. Therefore, the total delay introduced by DCF+ is smaller than that of DCF.

From the figures we can see that when the number of stations increase, the MAC access delay increases, which is because the collision probability increases. On the other hand, the IFQ delay first increases then decreases with the number of stations increases. This is because the TCP is affected by the collisions at MAC layer and the goodput decreases with the increased station number. Therefore, with increased delay and degraded good, TCP send less number of packets to IFQ, and finally the IFQ delay decreases correspondingly.

Therefore, by elaborate simulations, we conclude that DCF+ can improve the performance of TCP over WLAN at all the three metrics we examine in this paper: goodput, fairness index and delay.

VIII. Conclusions

IEEE 802.11 MAC is proposed to support packet transmission over WLAN and DCF is the basis of 802.11. Due to its own protocol characteristics, such as MAC ACK, MAC retransmissions, are different from those of traditional wireless medium, the performance of reliable transport protocol, such as TCP, over WLAN needs careful studies.

TCP needs the transport layer acknowledgement (TCP ACK) on the backward direction. In the scenario of TCP over WLAN where a shared channel is used for multiple access, the forward TCP data and the backward TCP ACK will compete the channel, which may cause collisions and degrade the performance. Based on these observations, this paper proposes a scheme named DCF+ to enhance the performance of TCP over WLAN.

This paper also proposes a new and simple analytical model based on Markov chain to compute the throughput performance of IEEE 802.11 DCF and our proposed DCF+. This model can be used for both the basic access method and the RTS/CTS access method in DCF. Comparisons with simulations as well as the model presented in paper [11] show that this model is accurate in predicting the 802.11 system throughput. Our modeling results for DCF+ are also verified by elaborate simulations.

The performance of TCP over WLAN is examined by elaborate simulations, both for DCF and DCF+. Also, another contribution of this paper is the analysis of the simulation results of TCP over WLAN. Finally we conclude that DCF+ can be used to enhance the performance of TCP over WLAN in all the three metrics we examine in this paper: goodput, fairness index and delay.

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