Enhancing performance of the IEEE 802.11 Distributed Coordination Function via packet bursting

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Abstract - **During the past few years, Wireless Local Area Networks (WLANs) have become extremely popular. The IEEE 802.11 protocol is the dominating standard for WLANs employing the Distributed Coordination Function (DCF) as its essential medium access control (MAC) mechanism. This paper presents a simple and accurate analysis using Markov chain modelling to compute IEEE 802.11 DCF performance, in the absence of hidden stations and transmission errors. This mathematical analysis calculates in addition to the throughput efficiency, the average packet delay and the packet drop probability for both basic access and RTS/CTS medium access schemes. The derived analysis, which takes into account packet retry limits, is validated by comparison with OPNET simulation results. The mathematical model is used to study the effectiveness of the RTS/CTS scheme at high data rates and the performance improvements of transmitting a burst of packets after winning the contention for medium access. Packet bursting considerably increases both throughput and packet delay performance but lowers the short-term fairness on medium access.**

I. Introduction

In recent years, Wireless Local Area Networks (WLANs) play a key role in the data communications and networking areas, having witnessed significant research and development. Technological and regulatory progress has allowed the issues of high prices, low data rates and licensing requirements to be addressed driving the popularity of wireless LANs to grow significantly [1]. With wireless networking, regardless of where end users are, they can have network connectivity being a mouse-click away from key information and applications. Recent advances in wireless technology and mobile communications have provided wireless capabilities to portable devices including palmtop computers, laptops and personal digital assistants (PDAs). IEEE 802.11 is the de facto standard utilized by most WLANs worldwide and provides physical and medium access layer specifications.

The IEEE 802.11 protocol [2] incorporates two Medium Access Control (MAC) methods; the compulsory contention-based Distributed Coordination Function (DCF) and the optional centrally controlled Point Coordination Function (PCF). DCF supports asynchronous data transfer on a best effort basis and is best suited to delay insensitive data. PCF has QoS support and address delay sensitive data communications. DCF employs a Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) access scheme using binary exponential backoff and defines a basic access as well as an optional Request-To-Send/Clear-To-Send (RTS/CTS) mechanism for packet transmission. The latter uses small RTS/CTS packets exchanged at the basic control rate to reserve the medium before large packets are transmitted in order to reduce the duration of a collision. Moreover, the RTS/CTS reservation scheme is utilized to combat the hidden station problem.

Considerable research has been carried out to model and study the IEEE 802.11 protocol performance either by simulation [3][4] or analytical modeling of DCF [5]-[8]. Bianchi in [5] and Wu in [6] employ Markov chain models to analyze DCF operation and calculate the saturated throughput of 802.11 MAC protocol. In particular, Bianchi [5], models the idealistic assumption that packet retransmissions are unlimited and a packet is being retransmitted continuously until its successful reception. Wu in [6] extends Bianchi's analysis to include the finite packet retry limits as specified in the IEEE 802.11 standard. In [7] we report average packet delay, for the case of infinite retry limits [5]. In [8] we provide a new performance analysis of the 802.11 protocol, which is based on the Markov chain model developed in [6], and allows the calculation of the average packet delay, the packet drop probability and the packet drop time.

Other papers [9]-[10] have attempted to improve IEEE 802.11 performance. Sheu in [9] suggests concatenating several data packets in a large packet by introducing modifications in certain packet formats. Sadeghi in [10] proposes another approach by transmitting a burst of packets for a single RTS/CTS handshake that considerably improves performance.

In this paper, we present a complete DCF performance modelling that includes Wu's [7] throughput model and the

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analysis [8] that calculates the average packet delay and packet drop probability. The average packet delay is calculated by considering the average number of slots required for a successful packet transmission. This method is elegant, very intuitive, leads to simple equations for the average packet delay and its accuracy is validated by OPNET simulation outcome. Based on this analysis, we identify that when the packet size increases there is significant increase on throughput but packet delay increases as well. We also explore the effectiveness of RTS/CTS scheme in respect to the basic access at high data rates and when no hidden stations are present. Finally, we propose a performance improvement by employing packet bursting in order to reduce overhead costs (i.e. backoff time and less RTS/CTS exchanges). The main idea of packet bursting is based on the transmission of more than one data packets when a station attains control of the medium while retaining the long-term fairness provided by the 802.11 protocol.

II. Overview of IEEE 802.11 DCF

This section briefly introduces the components of the binary exponential backoff mechanism employed in DCF utilized by the mathematical analysis that follows. Readers can refer to the IEEE standard [2] or to [4]-[8] for further details on the IEEE 802.11 DCF.

DCF is based on the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) technique and adopts a slotted Binary Exponential Backoff (BEB) scheme to reduce collisions due to stations transmitting simultaneously. Each station waits a random backoff interval before initiating a packet transmission (this is the Collision Avoidance feature of the protocol). Moreover, all stations utilize the Network Allocation Vector (NAV) for virtual carrier sensing, by updating its value with the duration of other stations transmissions. Thus, stations know when the current transmission ends and the channel is idle again. DCF describes two techniques for packet transmission; the mandatory basic access and the optional Request-To-Send/Clear-To-Send (RTS/CTS) reservation scheme. The RTS/CTS reservation scheme should be used if the collision probability is high and the packet size is longer than a threshold in order to shorten collision duration and cope with hidden stations. In this case, short RTS and CTS packets are exchanged to reserve the medium prior to the transmission of the long data packet.

Under DCF, a station willing to transmit a data packet senses the channel to determine its state. If the channel is detected idle, the station waits for a DCF inter-frame space (DIFS) time interval. If no other transmission takes place during the DIFS period, the station proceeds with its packet transmission. If the medium is sensed busy, the station defers transmission and initializes its random backoff timer to minimize the probability of collision with packets being transmitted simultaneously by other stations. The backoff timer is decremented when the medium is idle, is frozen when the medium is sensed busy and resumes again only after the medium has been idle for longer than DIFS. In addition, to avoid channel capture, a station must wait a random backoff time between two consecutive packet transmissions, even if the medium is sensed idle for more than DIFS after a successful packet transmission. Note that each station is allowed to transmit only when its backoff timer reaches zero and at the beginning of a slot time. The value of the backoff timer for each station is uniformly chosen in the interval $[0, W_i -1]$, where W_i is the current contention window (CW) size, i is the backoff stage, $i \in [0,m]$ and m represents the station short retry count. The value of W_i depends on the number of unsuccessful transmissions of a packet; at the first transmission attempt, CW is equal to the minimum backoff window size $CW_{min} = W_0 = W$. After each retransmission due to a packet collision, W_i is doubled until a maximum backoff window size value is reached, $CW_{max} = W_{m'} = W \cdot 2^{m'}$ where m' identifies the maximum number of backoff stages. Once W_i reaches CW_{max}, it will remain at this value until it is reset to CW_{min} after the successful data packet transmission or when the retry limit for this packet is reached. Upon the successful reception of a packet, the destination station sends back an immediate positive acknowledgment (ACK) after a time interval equal to Short Inter-Frame Space (SIFS). Explicit transmission of an ACK is required since, in the wireless medium, a transmitter cannot determine if a packet is successfully received by listening to its own transmission. If the source station does not receive an ACK, the data packet is assumed to have been lost and a retransmission is scheduled according to the previous backoff rules. According to IEEE 802.11 standard [2], every station maintains a retry count that indicates the number of retransmission attempts of a data packet. If the retry count reaches the specified limit, retry attempts cease and the packet is discarded.

III. Legacy IEEE 802.11 DCF

This section first employs the Markov chain model of [6][8] to report simple equations for saturation throughput and then presents an elegant model for calculating the average packet delay. The accuracy of the new model is validated by comparing analytical results with OPNET simulation outcome. We then utilize the mathematical model to explore the effectiveness of the RTS/CTS reservation scheme compared to the basic access at high data rates.

A. Analytical modeling

We make use of the same assumptions as in [5][6][8]; all stations always have a packet available for transmission (saturation case), the channel is error-free and no hidden stations exist. The probability p that a transmitted packet collides is assumed to be constant and independent of the number of collisions the station has experienced in the past and is given by:

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

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$$\tau = \begin{cases} \dfrac{2(1-2p)(1-p^{m+1})}{W(1-(2p)^{m+1})(1-p)+(1-2p)(1-p^{m+1})} & , \ m \le m' \\ \\ \dfrac{2(1-2p)(1-p^{m+1})}{W(1-(2p)^{m'+1})(1-p)+(1-2p)(1-p^{m+1})+W\,2^{m'}\,p^{m'+1}(1-2p)(1-p^{m-m'})} & , \ m > m' \end{cases} \quad (2)$$

where n is the number of contending stations, τ is the transmission probability of a packet given by equation (2), m is the retry limit, indicating that a packet will be discarded after an unsuccessful transmission at the m stage. Equations (1) and (2) form a non-linear system with two unknowns p and τ which can be easily solved by utilizing numerical methods.

The saturation throughput S, defined as the fraction of time the channel is used to transmit useful payload, is given by:

$$S = \frac{P_{tr} P_S\, l}{E[slot]} = \frac{P_{tr} P_S\, l}{(1-P_{tr})\sigma + P_{tr} P_S T_S + P_{tr}(1-P_S) T_C} \quad (3)$$

where $E[slot]$ is the average length of a slot time, l is the payload packet length, σ is the duration of an empty slot time, $P_{tr} = 1-(1-\tau)^n$ is the probability that there is at least one packet transmission in the considered slot time, $P_S = n\tau(1-\tau)^{n-1} / P_{tr}$ is the probability that an occurring packet transmission is successful, T_C and T_S are the average durations the medium is sensed busy due to a collision and a successful transmission respectively.

The values of T_C and T_S depend on the medium access mechanism and are given for the basic access and the RTS/CTS access mechanisms by:

$$\begin{cases} T_S^{bas} = DIFS + T_{DATA} + SIFS + T_{ACK} + 2\delta \\ T_C^{bas} = DIFS + T_{DATA} + SIFS + T_{ACK} + 2\delta \end{cases} \quad (4)$$

$$\begin{cases} T_S^{RTS} = DIFS + T_{RTS} + T_{CTS} + T_{DATA} + 3\,SIFS + T_{ACK} + 4\delta \\ T_C^{RTS} = DIFS + T_{RTS} + SIFS + T_{CTS} + 2\delta \end{cases} \quad (5)$$

where δ is the propagation delay, T_{DATA} , T_{ACK} , T_{RTS} and T_{CTS} is the time required to transmit the DATA, ACK, RTS and CTS packets, respectively.

We next present an elegant method, which calculates the average delay for a successfully transmitted packet, which is defined as the time interval from the instance a head-of-queue packet is ready for transmission until its successful reception. When retry limits are considered, packet delay cannot be simply obtained from throughput (like in [7]); the calculation of the average number of slot times needed for a successful packet transmission is necessary. If a packet has reached its retry limit and is dropped, it will not be included in the calculation of the average packet delay.

The average packet delay $E[D]$ is given by:

$$E[D] = E[X]\, E[slot] \quad (6)$$

where $E[X]$ is the average number of time slots needed for a successful transmission. $E[X]$ is calculated by multiplying the number of time slots d_i the packet is delayed in each backoff stage by the probability q_i for the packet to utilize this backoff stage:

$$E[X] = \sum_{i=0}^{m} d_i\, q_i \quad (7)$$

where d_i is given by:

$$d_i = \frac{W_i + 1}{2} \quad , \ i \in [0, m] \quad (8)$$

In order to calculate the conditional probability q_i , we first work out the packet drop probability for a head-of-queue packet, which is defined as the conditional probability that a packet is dropped when its retry limit is reached. Since a packet reaches the last backoff stage m, if it encounters m collisions in the previous stages, this packet will be dropped if it experiences another collision. Therefore, the packet drop probability is equal to:

$$p_{drop} = p^m\, p = p^{m+1} \quad (9)$$

The conditional probability q_i that a successfully transmitted packet utilizes the i backoff stage can be computed as:

$$q_i = \frac{(p^i - p^{m+1})}{1 - p^{m+1}} \quad , \ i \in [0, m] \quad (10)$$

since packets that are not dropped (with probability $1-p^{m+1}$) reach the i stage with probability $\left(p^i - p^{m+1}\right)$ (we have to deduct the probability p^{m+1} of dropped packets from the probability p^i of the total number of packets reaching the i stage). By combining (7), (8) and (10), E[X] is given by:

$$E\,[X] = \sum_{i=0}^{m} \left[\frac{(p^i - p^{m+1})\, \frac{W_i + 1}{2}}{1 - p^{m+1}} \right] \quad (11)$$

Finally, packet delay can be found by combining (6), (11) and (3).

B. Performance evaluation

Unless otherwise specified, the values reported in the following figures have been obtained using the system parameters in table I and are based on the Direct Spread Sequence Spectrum (DSSS) physical layer used in 802.11b standard [2].

Parameter	Value
Packet payload, l	8184 bits
Slot time, σ	20 μs
MAC header	272 bits
PHY header, l_{PHY}	192 bits
RTS packet	160bits + l_{PHY}
CTS packet	112bits + l_{PHY}
ACK packet	112bits + l_{PHY}
DIFS	50 μs
SIFS	10 μs
Data rate, C	2, 5.5, 11 Mbit/s
Minimum CW, W_0	32
Number of CW sizes, m'	5
Short retry limit, m	6

Table I The system parameters used to obtain numerical results

Fig. 1 plots throughput and packet delay against the number of contending stations for data rate of C=2 Mbit/s. Results obtained from the analytical model are compared to simulation outcome by means of our IEEE 802.11 simulator developed with the OPNET™ Modeler simulation software package. The figure validates the analytical model and modeling assumptions since an almost exact match is observed between analytical results (lines) and simulation outcome (symbols)[1]. Moreover, the figure illustrates that analytical modeling that considers retry limits predicts very accurately DCF throughput performance, a conclusion not drawn in [6] which added retry limits in the analytical model in [5].

The effect of packet payload size on performance is illustrated in fig. 2; throughput and packet delay are plotted against packet size for a congested network (n=50), C=11 Mbit/s and for both access mechanisms. Fig. 2 shows that when the highest data rate of 11 Mbit/s is utilized combined with the low control rate of 2 Mbit/s, the basic access scheme outperforms RTS/CTS even for relatively large packet values (l<8000 bits). This surprising result demonstrates the deficiency of the RTS/CTS scheme for high data rates (C=11 Mbit/s); only very large packet size values render the RTS/CTS beneficial even when the collision probability increases considerably as a result of the large number of contending stations. An interesting observation is that increasing the packet size, packet delay increases as well. This can be easily explained due to the fact that higher packet size values actually denote a longer transmission time duration.

[1] Simulation results are acquired with a 95% confidence interval lower than 0.002.

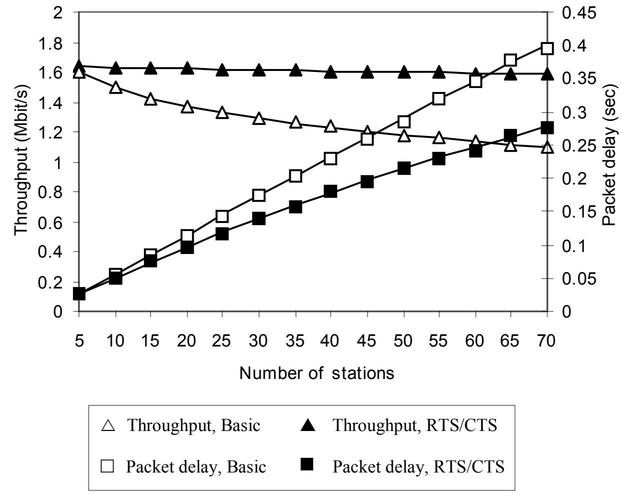

Fig. 1 Throughput and packet delay: analysis versus simulation, for C=2 Mbit/s

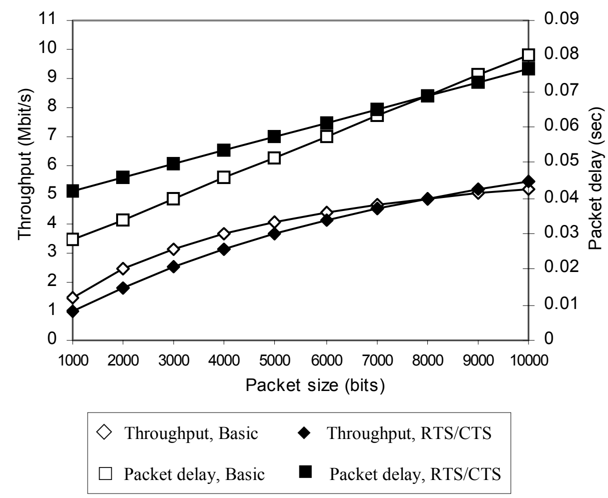

Fig. 2. Packet delay and throughput against packet size, for n=50 and C= 11 Mbit/s

IV. Performance enhancement through packet bursting

The concept of transmitting more than one data packets after winning DCF contention is called packet bursting. It is included in the latest 802.11e draft specification and has been discussed in [10]. The number of pending data packets that a station will transmit with packet bursting depends on the data and control rate it is employing. The advantage of packet bursting is the increased throughput due to the reduction of contention periods and RTS/CTS exchanges at the cost of short-time unfairness.

A. Implementation Issues of packet bursting

Fig. 3 and 4 illustrate how packet bursting is applied to both basic access and RTS/CTS schemes. The presented implementation of packet bursting is based on the fragmentation mechanism of the IEEE 802.11 protocol discussed in [10]. This mechanism provides a simple and practical way for stations to hold the medium for multiple packet transmissions when high data rates are utilized. A station that implements packet bursting transmits a burst of *ppb* packets before releasing the medium. The receiving station individually acknowledges every DATA packet by sending an ACK packet after a SIFS interval and the transmitting station sends the next DATA packet upon reception of this ACK (again after SIFS). If any DATA packet transmission fails (an ACK is not received) the burst is terminated and the station shall attempt to contend for the medium and retransmit the failed DATA packet and the packets following it. Since the SIFS interval is shorter than the DIFS, it is ensured that the sender retains control over the medium and that no other station can go into contention and start transmitting until all the packets that belong to the burst are transmitted.

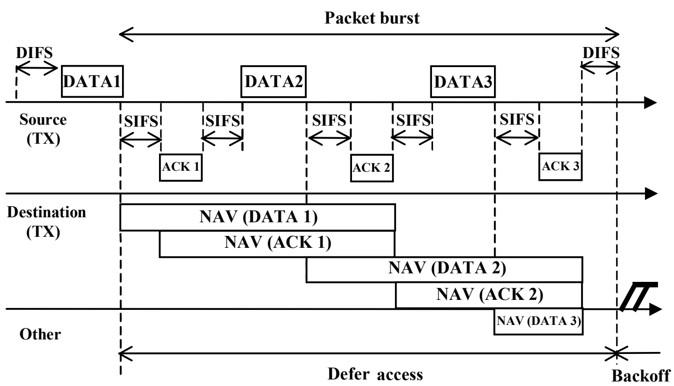

Fig. 3 Implementation of packet bursting to basic access scheme (*ppb*=3)

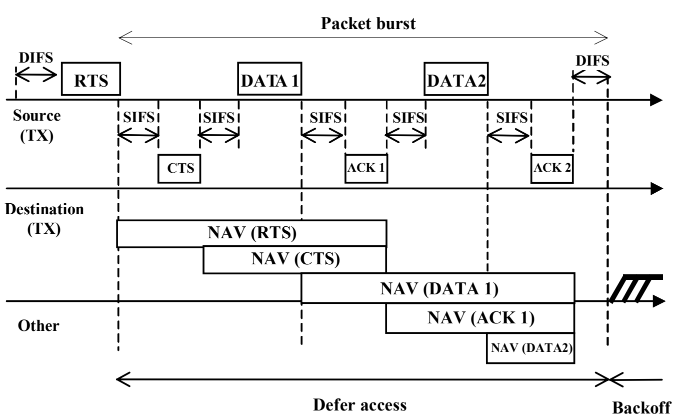

Fig. 4 Implementation of packet bursting to RTS/CTS scheme (*ppb*=2)

A detailed description for the NAV usage in packet bursting when the RTS/CTS mechanism is employed (fig. 4) is given next. The duration information included in the RTS and CTS packets is used to update the NAV of the stations to indicate that the channel is busy until the end of ACK 1. Both DATA 1 and ACK 1 packets contain duration information to update the NAV of all receiving stations to indicate a busy channel until the end of ACK 2. This carries on until the last DATA packet, which carries the duration of one ACK time plus one SIFS time in its duration field. The ACK for the last DATA packet has the duration field set to zero. Thus, each DATA/ACK pair acts as virtual RTS/CTS for the next DATA/ACK exchange and no further RTS/CTS packet exchange is necessary. Also every DATA packet (except the last one) has the *more fragments* flag in the MAC header set to 1 in order to indicate the use of the fragmentation mechanism. The MAC header of the DATA packets also carries the packet number that is used by the destination to arrange the order of the DATA packets (in the case of a single packet transmission, this field is set to 0).

An alternate mechanism to transmitting a specific number of packets after winning the DCF contention is to allow stations to transmit consecutive packets provided that the total access time does not exceed a certain limit (TXOP limit). This mechanism is introduced in IEEE 802.11e and the implemented number of packets per burst depends on the transmission rate and on the signal quality at the receiver. As stations implementing the packet bursting mechanism utilise the standard backoff procedure and thus experience the same delays but transmit more information after winning the contention for the medium, it is expected that packet bursting should improve performance. When a station that implements packet bursting has only one packet available in the station's queue, normal DCF procedures are used and the system has the same performance as without packet bursting.

B. Analytical modelling of packet bursting

The saturation throughput S in the case of packet bursting is computed as:

$$S_{burst} = \frac{P_{tr} P_S \; ppb \; l}{E'[slot]} \frac{P_{tr} P_S \; ppb \; l}{(1-P_{tr})\sigma + P_{tr} P_S T'_S + P_{tr}(1-P_S)T'_C} \quad (12)$$

where *ppb* is the number of packets per burst employed by all stations, $E'[slot]$ is the average slot time when packet bursting is used, T'_S and T'_C are the average durations the medium is sensed busy due to a collision and a successful transmission respectively for packet bursting transmissions. The values of T'_S and T'_C for the basic and the RTS/CTS access mechanisms are given by (13)-(14), respectively.

$$\begin{cases} T_S'^{\,bas} = DIFS + ppb \; T_{DATA} + (2\,ppb - 1)\,SIFS + ppb\,T_{ACK} + 2\,ppb\,\delta \\ T_C'^{\,bas} = DIFS + T_{DATA} + SIFS + T_{ACK} + 2\,\delta \end{cases} \quad (13)$$

$$\begin{cases} T_S'^{\,RTS} = DIFS + T_{RTS} + T_{CTS} + ppb\;T_{DATA} + (2\,ppb+1)SIFS + ppb\,T_{ACK} + (2\,ppb + 2)\delta \\ T_C'^{\,RTS} = DIFS + T_{RTS} + SIFS + T_{CTS} + 2\delta \end{cases} \quad (14)$$

By utilizing the approach discussed in section III, we can calculate the average delay for a successfully transmitted packet that belongs to a burst of *ppb* packets as:

$$E[D]_{burst} = \frac{E[X]\,E'[slot]}{ppb} \qquad (15)$$

where $E[X]$ and $E'[slot]$ are given by (11) and (12), respectively.

C. Performance evaluation of packet bursting

Fig. 5, 6, 7 and 8 illustrate the substantial improvement of packet bursting on performance; they plot throughput and packet delay versus network size for different burst size values and data rates for both basic access and RTS/CTS schemes. All figures clearly show that packet bursting substantially enhances performance by increasing throughput and reducing packet delay. This is explained by considering that packet bursting reduces the overhead by amortizing the cost of the contention period and RTS/CTS packet exchange over several packets.

It is quite interesting to study why and how packet bursting increases performance in different scenarios. When no packet bursting is implemented (*ppb*=1) and the basic access is used (fig. 5), throughput considerably decreases for all data rates when the network size increases due to the increased packet collision probability. When *ppb*=1 and the RTS/CTS scheme is used, throughput is not significantly affected from network size increase for *C*=2 Mbit/s because the increased packet collision probability does not degrade performance due to the short collision duration. However, for higher data rates (*C*=5.5 Mbit/s and *C*=11 Mbit/s), throughput degrades with network size increase because the collision duration is high compared to the data rate as the RTS and CTS control packets are always transmitted at the lower control rate of 2 Mbit/s.

When packet bursting is utilized (*ppb*=3 and *ppb*=5) for the basic access scheme, throughput considerably increases, especially for large networks with increased collision probability, mainly because packet bursting shortens the duration of collisions as compared to the duration of successful transmissions! Collisions involve only the first DATA packet of the packet burst because the lack of the first ACK packet forces the transmitting stations to contend again for medium access; successful medium accesses last much longer as they involve the transmission of a burst of packets.

When packet bursting is utilized (*ppb*=3 and *ppb*=5) for the RTS/CTS scheme, throughput is not significantly increased for *C*=2 Mbit/s due to the relatively short RTS/CTS collision duration. However, at higher data rates, (*C*=5.5 Mbit/s and *C*=11 Mbit/s), throughput is considerably increased because packet bursting reduces the number of medium reservations that involve the transmission of RTS and CTS packets at the low data rate of *C*=2 Mbit/s.

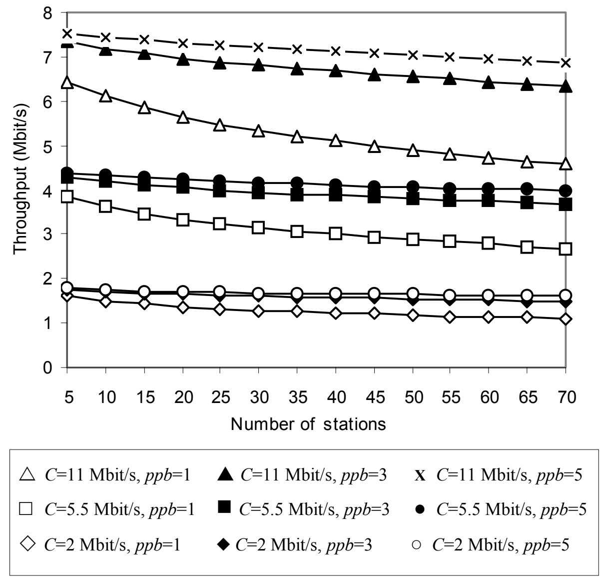

Fig. 5 Throughput enhancement of packet bursting (basic access)

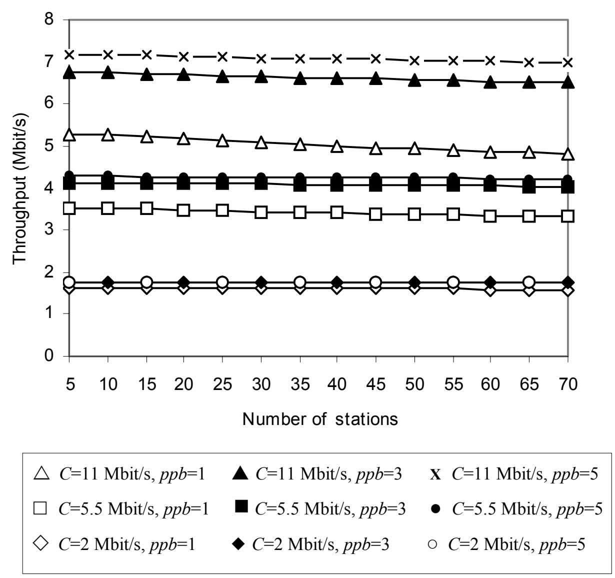

Fig. 6 Throughput enhancement of packet bursting (RTS/CTS)

D. Fairness issues

The main purpose of a successful packet bursting implementation is the selection of a reasonable packet burst size value that improves performance and, at the same time, prevents stations from capturing the medium for long periods. Medium capture is undesirable and creates fairness problems. The fairness of a protocol is measured in terms of how resources are assigned to different stations over a period of time. Based on the length of this time period, the fairness can be measured on short-term or on long-term basis.

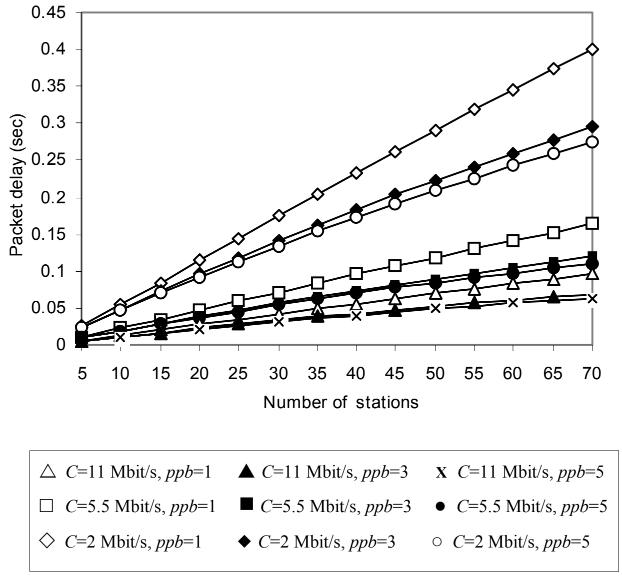

Fig. 7 Packet delay reduction of packet bursting (basic access)

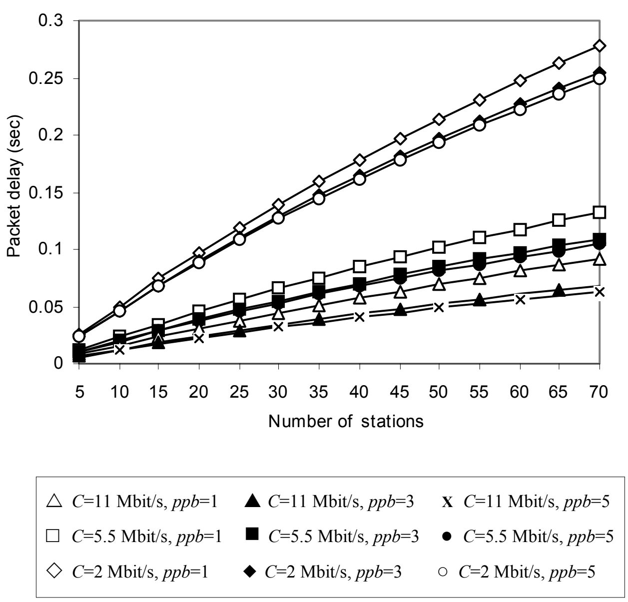

Fig. 8 Packet delay reduction of packet bursting (RTS/CTS)

Intuitively, short-term fairness of a protocol refers to its ability to allocate the channel bandwidth equally to competing stations over short time periods; long-term fairness, in contrast, measures the same ability over longer time periods. The short-term fairness automatically implies long-term fairness, but not the vice versa [11].

To measure fairness, this work utilizes the average fairness index proposed by Jain [12]:

$$F_J = \frac{\left(\sum_{i=1}^{n} x_i\right)^2}{n \sum_{i=1}^{n} x_i^2} \qquad (16)$$

where n is the number of stations and x_i is the throughput of station i during the considered window size of w successful packet transmissions. Absolute fairness is achieved when $F_J = 1$ (all stations equally share the medium) and absolute unfairness (a station monopolizes the channel) is achieved when $F_J = 1/n$.

In fig. 9, we examine the fairness of packet bursting (utilizing the average Jain's fairness index) by considering two window size values that represent a short-term scale (w=1000 packets) and long-term scale (w=10000 packets). The figure reveals the weak fairness of both the packet bursting and the legacy IEEE 802.11 on a short-term scale (a small window size exhibits high unfairness). In fact, the fairness index is considerably lower than one when packet bursting is not utilized, especially for large network size values. However, fairness improves in both cases when the window size used for measurement is increased, ensuring long-term fairness (in long-term all contending stations experience on average the same number of collisions).

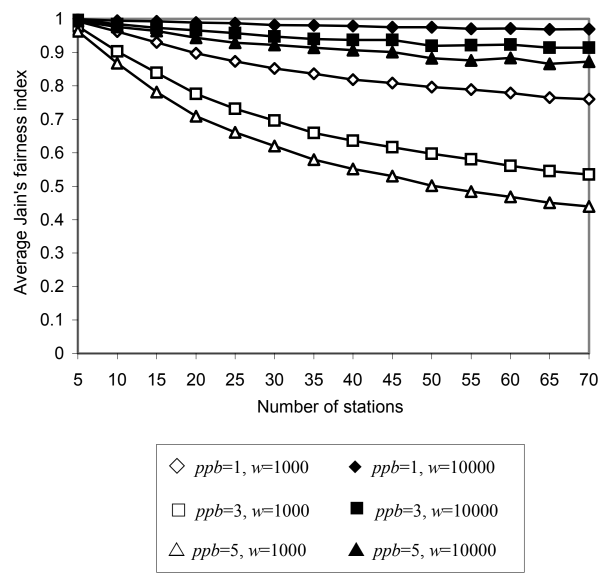

Fig. 9 Fairness of packet bursting over short and long time scale (Basic access, C=2 Mbit/s)

V. Conclusions

This paper reviews mathematical modelling of IEEE 802.11 DCF assuming ideal channel conditions. A simple an intuitive mathematical model is presented that considers packet retry limits and calculates throughput, average packet delay and packet drop probability. The model is utilized to study the effectiveness of RTS/CTS reservation scheme at high data rates and identified that the RTS/CTS scheme outperforms basic access only for high packet size values because the RTS and CTS control packets are transmitted at a much lower control rate. We also extended the mathematical model to consider packet bursting, an approach in which a station transmits more than one data packets when it gets hold of the medium. Results obtained for different scenarios showed that the application of packet bursting significantly enhances performance (a) in large networks utilizing basic access because it shortens collision duration and (b) in high data rate networks utilizing RTS/CTS scheme because it reduces the number of medium reservations that involve the transmission of the RTS and CTS control packets at the low control rate. Furthermore, fairness was explored for both legacy DCF and packet bursting cases; packet bursting experiences weak fairness in short-time scale while retaining the long-term fairness provided by the backoff mechanism of the 802.11 protocol. The benefits of enhanced performance and easy implementation through the fragmentation mechanism makes packet bursting available without difficulty to any station employing IEEE 802.11a, 802.11b or 802.11g technologies. Finally, if packet bursting is combined with priority mechanisms, it can provide a complete solution for enhancing performance in QoS applications.

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