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THE INFLUENCE OF MAC BUFFER ON THE CONTENTION-BASED ACCESS SCHEME WITH BURSTING OPTION FOR IEEE 802.11E WIRELESS NETWORKS

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Abstract

Wireless LANs are increasingly being used for inelastic applications. Currently, there is little support for quality of service in the IEEE 802.11 MAC protocol, and IEEE task group E has defined the 802.11e MAC extension. Enhanced distributed channel access (EDCA) is a contention-based scheme of the 802.11e standard. To allow a station to transmit more than one frame from a single contention, an optional feature known as controlled frame-bursting (CFB) is introduced in the standard. In this paper, we initially performed an average analysis to determine a suitable burst duration limit. Then, a detailed evaluation and comparison of the EDCA protocol with the CFB option is carried out through simulation to quantify its performance gain. The impact of the MAC transmit buffer size is also incorporated. Accordingly, we have proposed a suitable approach to guide the configuration of the burst duration limit. It is demonstrated that an optimized CFB configuration allows the MAC protocol to achieve 30% more capacity than the basic EDCA scheme.

Keywords: IEEE 802.11e, MAC, EDCA, Frame-Bursting , Quality of Service.

Nomenclature

AC	Access category
AIFS	Arbitration IFS
AP	Access point
CSMA/CA	Carrier sense multiple access/Collision avoidance
CW	Contention window
CW_{min}	Minimum contention window
CW_{max}	Maximum contention window
DCF	Distributed coordination function
DIFS	DCF interframe space
N	number of WSTAs
p	Collision probability
PF	Persistence factor
R	Network capacity
S	Saturation throughput
SIFS	Short IFS
T_{cycle}	Time between two successful burst transmissions
TXOP	Transmission opportunity
TXOPLimit	TXOP duration limit
WSTA	Wireless station

1. Introduction

Wireless local area networks (WLANs) are seeing more deployment as a competing technology to the next generation (3/4G) cellular networks in the provision of data services. Since the standardization of WLAN protocol stack by the IEEE 802.11 subcommittee in 1997, the deployment of these networks is occurring at a great pace. The original standard provided data rates up to 2 Mbps at the unlicensed ISM (2.4 GHz) band. Later, the IEEE 802.11 working group published several enhanced physical layer (PHY) specifications, namely 802.11b PHY with data rate up to 11 Mbps in the ISM band, 802.11a PHY that can achieve a data rate up to 54 Mbps using orthogonal frequency division multiplexing (OFDM) in the 5 GHz unlicensed national information infrastructure (UNII) band and 802.11g PHY with similar rate as 802.11a but works at the ISM band. All these PHYs have the same IEEE 802.11 MAC protocol specified for channel access. With the standards supporting relatively higher data rates, WLANs became widely installed at homes, corporate building, campuses, airports and hot-spots.

As these networks become more ubiquitous, mobile users are increasingly using common multimedia applications predominantly used on desktop computers with wired access to the networks. When there exists various traffic mix with different requirements on these networks, there is a serious need to provide service differentiation known as quality of service (QoS) to enable inelastic traffic types like telephony calls, to be treated with urgency against bursty traffic type of web browsing-like applications in the face of shared bandwidth contention. In the wired world, IETF has defined the integrated services (IntServ) and differentiated services (DiffServ) models for IP networks with QoS requirements. For the wireless networks, the QoS provision is more critical as the wireless bandwidth is significantly more limited. Wireless medium is also subject to fast changes in signal-to-noise ratio, which affects the bit error rate experienced by the wireless stations. Thus, it is difficult to provide hard QoS guarantees in such networks. Instead, the wireless networks should minimally provide *soft* QoS guarantees for the support of these multimedia applications.

There is a significant number of research efforts to address the issues of providing QoS in WLANs prior to IEEE 802.11 standardization and thereafter. MAC protocols could generally be categorized as centralized or distributed. Centralized protocols, such as time-division multiple access, reservation-based schemes and polling schemes, enable more stringent QoS guarantees through a single point of coordination [1]. The coordinator (known as base station or access point) is able to carry out admission control, scheduling and channel access control, and thus, is more suited to *infrastructure* type WLANs (versus *ad hoc* type). Some examples of centralized protocols are point coordination function (PCF) of 802.11 and the hybrid coordination function (HCF) of the 802.11e [2] where both employ polling mechanisms, HIPERLAN/2 of European Telecommunications Standards Institute (ETSI) and many wireless asynchronous transfer mode (ATM) proposals. The acceptance of these protocols has generally been lukewarm due to high overhead, high cost/complexity and limited scalability. On the contrary, the distributed protocols are simpler to implement and require only limited overhead. They also have applications in domains where infrastructure mode is infeasible or difficult to be built like in battlefields, temporary festive or business venues and historic sites.

Some proposals prior to 802.11 standardization were Aloha, CSMA, MACA [3] and MACAW [4]. After the standardization of 802.11 protocol stack that was based on a variant of CSMA, there were a number of proposals to extend distributed coordination function (DCF) to support service differentiation by varying random backoff periods, contention windows or both. Some of these main proposals were Distributed Fair Sharing (DFS) [5], Blackburst [6] and enhanced distributed coordination access of IEEE 802.11e. All these distributed protocols are contention-based schemes and thus, need to address the contention and collision problems efficiently.

The suitability of the proposed schemes in 802.11e and their enhancements had already been investigated [7-12]. It was shown in [9] that EDCA works well for service differentiation and priority access. EDCA was demonstrated to assure better performance for high priority classes but at the cost of lower priority classes [10], and HCF was able to assure better QoS support when the medium is heavily loaded [11]. However, HCF involves state at the access point (AP) making it a less robust protocol. EDCA was also shown to support simultaneously a large number of voice and video flows in hot-spots and home network scenarios [12].

Within the IEEE 802.11e standard, there is also an option to use the controlled frame-bursting (CFB) mechanism to achieve better medium utilization through reduced collisions [2]. Once a wireless station (WSTA) contends and wins access to the channel, the CFB option would enable it to send more than one data frame without further contention during the current transmission opportunity. The WSTA is allowed to transmit as many frames as permitted within a certain limit specified in the MAC MIB. Since uncontrolled bursting may increase the frame delay variation, judicious use of the feature is necessary. It was reported that the use of CFB might enhance the performance and achieve better utilization [13].

In this paper, we investigate the effectiveness of the CFB feature in detail. The effect of the bursting mechanism is studied in a network with various mix of multimedia traffic and different offered load levels. It is also expected that the MAC transmit buffer size will have a critical effect on the CFB mechanism. As such, the impact of the MAC buffer size is also investigated. Based on these investigations, we also propose a guideline for the configuration of the burst duration limit to achieve an optimized EDCA protocol performance. Initially, the performance analysis is based on a mathematical model. This model follows the average-case analysis technique as in [14]. It is simple yet effective. We derive closed-form expression for the saturation throughput based on the average backoff expressions given in [14] and [15]. For further detailed investigation, simulation is used.

The remainder of this paper is organized as follows. In next section, we describe the basic EDCA scheme and the CFB mechanism in detail. Our simulation methodology and the adopted network scenario are described in Section 3. Section 4 presents the relevant performance metrics for each traffic class with its tolerable limits. Subsequently, the simulation results from the system experimentation are analyzed. The paper concludes by summarizing main findings in the final section, while highlighting extensions to this work.

2. The Enhanced Distributed Channel Access Scheme

The IEEE 802.11e MAC protocol is conceived as a compatible extension of the previous IEEE 802.11 MAC. It includes both a contention-based scheme known as EDCA and a more complete polling-based scheme known as HCF. Before we proceed further, let us define the nomenclature used further.

EDCA is governed through a distributed mechanism very similar to the existing DCF that relies on CSMA/CA protocol. EDCA adds prioritization by allowing different traffic classes to be mapped to *access categories* (ACs). As specified in the this standard, EDCA supports up to four traffic classes. One or more traffic classes can be mapped to each AC. Each AC has a separate queue and its associated contention and backoff values. In DCF, the backoff counter of a WSTA only commences when the wireless medium has been idle for at least a DCF interframe space (DIFS) time interval of fixed length. In EDCA, the interval can be different for each AC and is designated as Arbitration IFS (AIFS). An AIFS may be equal or greater than a DIFS.

Each AC contends for medium access as a separate DCF instance within a WSTA using its own contention parameters, AIFS, minimum contention window (CW_{min}) and maximum contention window (CW_{max}). When the medium is sensed idle for an AIFS period, a WSTA waits for an additional random period known as a backoff period. This period is computed from the contention window (CW) value. CW is initially set to the CW_{min}. The backoff period (in multiple of time slots) is determined uniformly from the interval [0, CW). After sensing the medium idle for AIFS, a WSTA decrements its backoff timer by one for each time slot elapsed while the medium remains idle. If the medium is found busy before the backoff timer expires, the timer is frozen and the WSTA has to abstain from any transmission for the duration of the current usage. When the medium becomes idle again, this WSTA restarts the frozen timer. When the timer reaches zero, the frame is transmitted. If an external collision is detected, CW is increased by a factor known as *persistence factor* (PF) as follows:

$$CW_i = \min(PF*CW_{i-1}, CW_{max}) \tag{1}$$

where *i* is the number of attempts to transmit the frame. PF can be varied too for each AC to provide further differentiation. In 802.11 MAC, its value defaults to two, representing a binary exponential backoff scheme.

ACs with smaller values for these contention parameters will generally experience lower mean waiting delay; thus, getting higher priority to the medium. For each WSTA, multiple ACs may set off in parallel leading to internal virtual collision. Virtual collisions between ACs within a WSTA are resolved such that the MAC frames from the higher priority AC receives the transmission opportunity (TXOP) and the lower priority AC(s) backs off without counting the collision like an external collision, i.e. does not affect its CW value.

2.1. The controlled frame-bursting mechanism

In IEEE 802.11e, there is also the possibility to use the option of CFB mechanism to enhance the performance and achieve improved channel utilization. Through this option, once a WSTA wins the TXOP to the wireless medium under normal contention, it is allowed to transmit more than one frame without further contention of the channel during its current TXOP. This operation is illustrated in Fig. 1. After obtaining access to the channel, the WSTA is allowed to transmit a number of frames from the same AC queue within a specified duration. The total duration is bounded by the dot11EDCATableLimit (shortened here as TXOPLimit for simplicity) MIB variable. This value is set based on either the most recently received QoS Parameter Set element of a beacon frame from the AP in an infrastructure setting or pre-configured in the WSTA's MIB in an ad hoc setting. To ensure that no other WSTA interrupts the burst, the interframe space between the reception of an acknowledgement and the subsequent transmission of the next data frame in the burst is a short IFS (SIFS) interval.

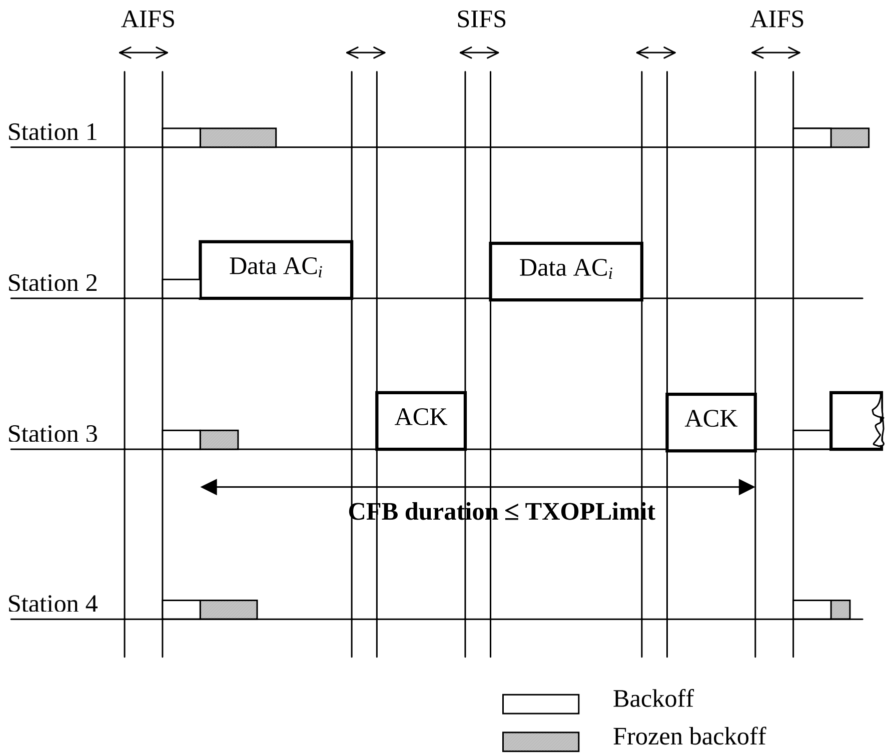

Fig. 1. The EDCA controlled frame-bursting mechanism.

Even though it is felt that CFB would increase the medium utilization, the selection of an appropriate value for the TXOPLimit is necessary especially when considering delay-sensitive flows being present in the networks. When a relatively large burst limit is used, a WSTA may continue holding the channel for a long time resulting in larger delay variations for sensitive flows like interactive voice and video in other WSTAs. Also, setting the value to a relatively small value allowing only the transmission of a few frames in a burst may reduce its operation to basic EDCA. Thus, the choice of an appropriate limit value is crucial for an optimized operation of the network with a particular traffic mix. To evaluate and verify the choice of the TXOPLimit value, an average analysis is performed and is presented further.

3. The Analytic Model

It was shown in Ref. [14] that the performance of 802.11 DCF protocol is mainly dependent upon CW_{min} and the number of WSTAs (N). These parameters in turn affect the collision probability p. Since EDCA is extended from DCF, similar assumption will hold here. For the tractability of this analysis, we will consider the network operation in saturation. Each AC queue will always have a frame to transmit while in saturation, thus every transmission is preceded by a backoff. Since the backoff of each AC is different due to their specific contention parameters, we will represent the average WSTA backoff case, $\overline{W}$. Extending from [14] for the multiple ACs, the collision probability p is given as:

$$p = \frac{1}{2}\left(1 + \frac{4}{g} - \sqrt{1 + \left(\frac{4}{g}\right)^2}\right), \text{ where } g = \frac{\overline{\mathrm{CW}_{\min}}}{N-1} \tag{2}$$

$\overline{\mathrm{CW}_{\min}}$ represents the average $\mathrm{CW}_{\min}$ of the ACs. From [14] and [15], the average backoff for a WSTA is adapted as:

$$\overline{W} = \frac{1 - p - p(2p)^m}{1 - 2p} . \frac{\overline{\mathrm{CW}_{\min}}}{2} \tag{3}$$

where m is used to compute $\mathrm{CW}_{\max}$ as $m \times \mathrm{CW}_{\min}$. Next, we need to derive the saturation throughput as the main performance metric for this analysis. The transmission episodes occur at rate $1/T_{\mathrm{cycle}}$, where T_{cycle} is the time between two successful burst transmissions. The number of frames transmitted in a burst for an AC is:

$$L_i = \left\lfloor \frac{\mathrm{TXOPLimit}}{T^i_{\mathrm{data}} + 2T_{\mathrm{SIFS}} + T_{\mathrm{ACK}}} \right\rfloor \tag{4}$$

where T^i_{data} is the time to transmit a data frame of AC_i, T_{SIFS} is the time duration of SIFS and T_{ACK} is ACK transmission time. Since this analysis uses average wherever possible, $\overline{L}$ represents the average number of frames from any ACi queue and $\overline{T_{\mathrm{frame}}}$ is used to denote the average transmission time [involves the denominator of eqn. (4)]. Since T_{cycle} comprises the average burst transmission, the average AIFS delay (represented as $\overline{T_{\mathrm{AIFS}}}$) and the average backoff delay [eqn. (2)], we could approximate it as:

$$T_{\mathrm{cycle}} = \overline{T_{\mathrm{frame}}}.\overline{L} + \overline{T_{\mathrm{AIFS}}} + \frac{\overline{W}}{N+1}.T_{\mathrm{slot}} \tag{5}$$

Let r_{succ} represent the rate of successful transmissions and r_{tx} the rate of transmission (including collisions). Assuming these are geometrically distributed, these rates are adopted from [14]:

$$r_{\mathrm{succ}} = \frac{2(1-p)}{2-p}.\frac{1}{T_{\mathrm{cycle}}}$$

$$r_{\text{tx}} = \frac{2}{2-p} \cdot \frac{1}{T_{\text{cycle}}} \tag{6}$$

We could now derive the expression for the saturation throughput (S_i) for each ACi as:

$$S_i = \frac{1}{3} r_{\text{succ}} . L_i . T^i_{\text{data}} . R = \frac{2(1-p)}{3(2-p)} \cdot \frac{L_i . T^i_{\text{data}}}{T_{\text{cycle}}} . R \tag{7}$$

where R is the network capacity.

This analysis is used to determine the optimal TXOPLimit to be used further in the various simulation experiments. The adopted simulation scenario and traffic characteristics for the simulation study are discussed further.

4. The Simulation Model

In order to evaluate the performance of the EDCA MAC protocol, a discrete-event simulator has been developed. The simulator is built using C++ and models a WLAN with an 802.11a PHY and 802.11e MAC layers. The simulator is developed using the object-oriented approach. Its overall software architecture is given in Fig. 2 as a simplified class diagram. The symbols used on the diagram represent the standard UML notations.

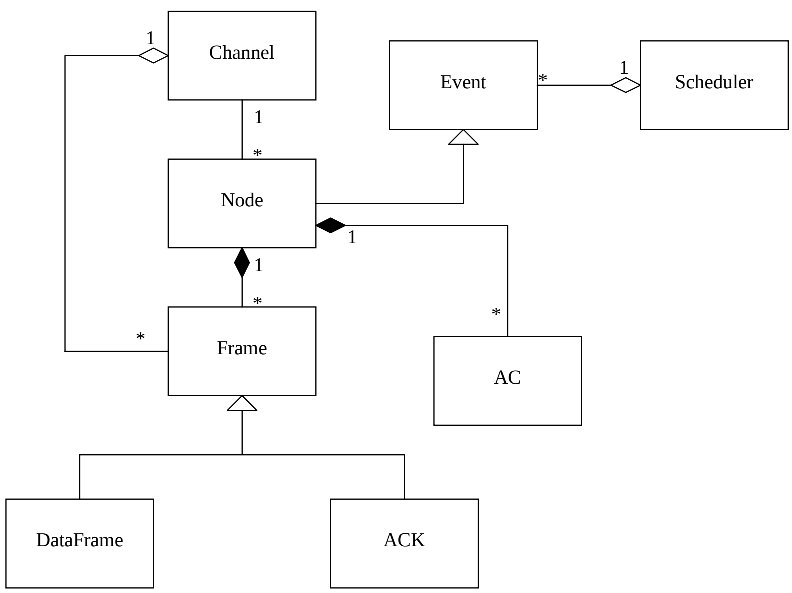

Fig. 2. The simplified class diagram of the simulator.

The support classes for this discrete-event simulator are the Scheduler and Event classes. The scheduler maintains the global simulation clock and a list of Event objects in chronological order. Its execution is controlled by the specified simulation duration. The single active entity in the real system, i.e. a WSTA (represented by the Node class), also represents the most complicated class in this simulator. The Node class implements the EDCA mechanism, which controls the AC selection and virtual collision resolution, the backoff mechanism, the frame exchange, and the external collision resolution. Since the nodes generate almost all significant events, it is made a subclass of the Event class. Since only the nodes generate the frames, it has a composition relationship with the Node class. Since a node implements more than one AC, this is represented as the next composition relationship with multiple AC objects.

There are two types of frames generated, namely the data and ACK frames represented by the DataFrame and ACK classes, respectively. The frame objects maintain the typical frame header information, mainly the source and destination addresses, and the duration field. The Channle class represents the shared wireless channel to be accessed by the Node objects. When there is only one frame in transit, the transmission would be successful. Otherwise, the presence of multiple frames is an indication of a collision event. The Channel class is also the appropriate class to represent the radio propagation behaviour and the channel bit error rate. The resulting architecture is highly cohesive and loosely coupled enabling easy extension for many other future works.

The adopted topology comprises a number of WSTAs configured in an ad hoc setting as shown in Fig. 3. All WSTAs are located within a basic service set (i.e. a cell) such that every WSTA is able to detect a transmission from any other WSTA, and they are static in the simulations. It is also assumed that the wireless medium is error-free. The adopted IEEE 802.11a PHY parameters are presented in Table 1.

Fig. 3. The single-BSS ad hoc network topology.

Table 1. IEEE 802.11a PHY parameters

Parameters	Values
aSlotTime	9 μs
SIFS	16 μs
Channel bit rate	54 Mbps
Propagation delay	500 *n*s

Our simulations involve a few different traffic sources to represent the typical traffic mix on the network. Voice signals are known to have a two-state ON/OFF behavior, where talkspurts are followed by silence periods. The G.729 speech codec [16] has been selected to model the voice calls, with voice packets of 60 bytes generated every 20 ms during a talkspurt, which corresponds to 24 kbps bit rate. The talkspurt and silence duration times are exponentially distributed with a mean value of 1 sec and 1.35 sec, respectively [17]. As for the video sources, we have considered low-quality video application with 500 kbps data rate, which emulates an MPEG downlink streaming service. We assume that these flows are generated by constant bit rate sources. A video packet length of 1500 bytes has been selected. Finally, it is assumed that the bursty data source, emulating an FTP application, follow a Poisson arrival with packet size of 1000 bytes and bit rate of 200 kbps. Two AC queues are used in each WSTA to enqueue voice and FTP packets or video and FTP packets. It is assumed that there is equal number of such WSTAs in the network.

These sources' parameters and their corresponding AC's contention parameter values are summarized in Table 2. The contention parameter values are chosen to ensure sufficient differentiation among the ACs to minimize collision, and were validated through simulations in [10]. To emulate the increase of system load, we gradually increase the number of WSTAs.

Table 2. MAC and source traffic parameters.

Parameters	Voice	Video	Bursty Data
AIFS (SIFS+*x**aSlotTime)	2	4	7
CW_{min}/CW_{max}	5/200	15/511	31/1023
Packet size (bytes)	60	1500	1000
Packet Interval (ms)	20	24	-
Bit rate (kbps)	24	500	200

Based on the adopted simulation scenario, simulations are performed and results are presented and discussed in the following section.

5. Performance Evaluation

In order to select the optimal TXOPLimit for the adopted scenario, we performed numerical evaluation of the given mathematical model. Initially, the collision probability (see Fig. 4) and saturation throughput (see Fig. 5) are evaluated against number of nodes. The curves in each graph are plotted by solving eqns. (2) and (7), respectively. The overall behaviour is consistent with the results presented in [14]. In Fig. 4, it is evident that CW_{max} has minimal effect on *p*. On the contrary, CW_{min} has significant influence whereby a larger CW value substantially reduces the collision probability. However, using a large CW_{min} (i.e. 127 and 255) had an unfavourable effect on throughput as shown in Fig. 5. This shows the oppressive effect of increasing CW, which unnecessarily increases backoff and thus underutilising the network capacity. Also, we could observe that CW_{max} (represented by m*CW_{min}) again had almost negligible effect even in bursting situation.

To optimize the TXOPLimit, we varied this bursting limit for different network sizes, but fixed $\overline{CW_{min}}$ = 31 and m = 3 as depicted in Fig. 6. It is evident that increasing TXOPLimit beyond 5 ms has minimal improvement in throughput across different network sizes. The initial rise in throughput is due to the smaller collision rate. However, further increase in throughput is bounded by the available capacity and the collision probability. We did not derive the mean transfer delay as will be extremely high as the network operates in saturation. However, the delay metric will be computed through simulation further. Intuitively though, a large TXOPLimit will result in higher delay and jitter experience of the delay sensitive traffic such as voice. Therefore, the remaining simulation results will only use limit not larger than 10 ms.

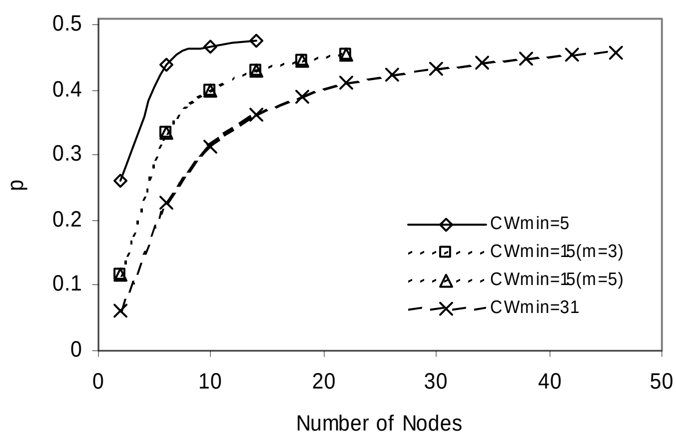

Fig. 4. The collision probability (*p*) vs. number of nodes for different CW_{min} and *m*.

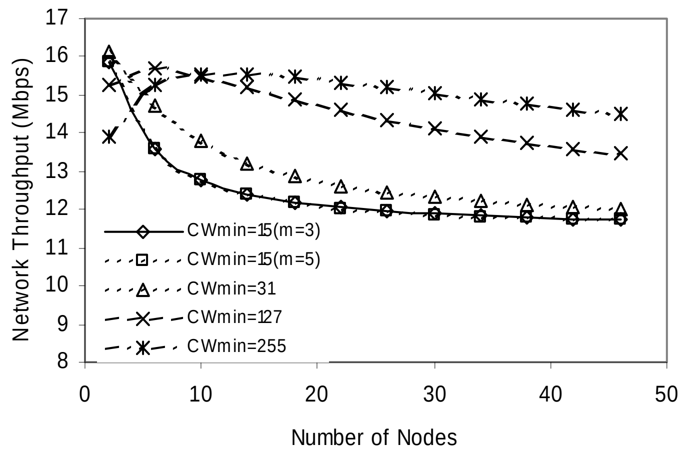

Fig. 5. Network throughput (*S*) vs. number of nodes for different CW_{min} and *m*.

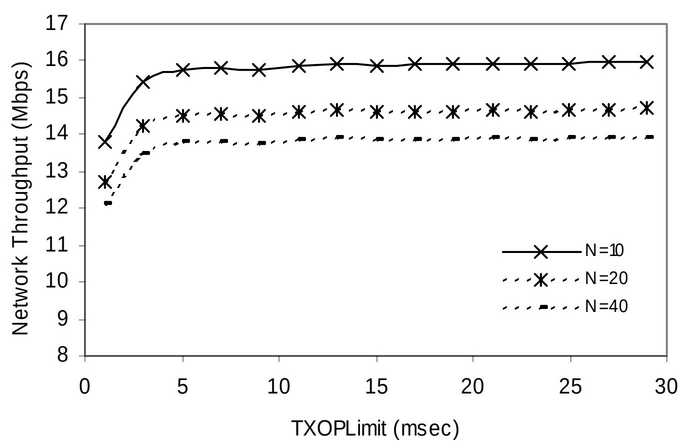

Fig. 6. Network throughput (*S*) vs TXOPLimit for different network sizes.

To evaluate the effect of the CFB mechanism on each traffic class through simulation, the following metrics are used:

- **Mean transfer delay:** the average time taken to transmit a packet from the time it is generated to the time the source receives a successful acknowledgement.
- **Delay jitter:** the variation of frame transfer delay.

- **Throughput:** the offered load that is actually delivered to the destination, which excludes the MAC and PHY overheads.
- **Packet loss ratio:** the fraction of discarded to generated packets.

The main performance metrics for voice traffic are frame transfer delay, jitter and packet loss [18]. In order to preserve the user-perceived voice quality, commonly accepted maximum values in an end-to-end connection over an IP network are 150 ms for the one-way delay, several milliseconds for delay jitter and 3% packet loss. However, since a WLAN is likely to represent only the first or last hop of an end-to-end connection, we have instead chosen 20 ms as a maximum acceptable value, as suggested in [12]. As for the video traffic, we have adopted 25 ms as the maximum tolerable one-way delay [19]. Throughput degradation has more serious consequence on the video performance when adaptive algorithms and jitter buffers are employed.

The above metrics are evaluated for different network sizes, *N*. In order to see the impact of MAC transmit buffer size on the protocol performance, we have adopted three different buffer sizes, namely 32 KB, 64 KB and 128 KB. In the forthcoming graphs, these MAC buffer sizes are shown in the legend as 32, 64 and 128, respectively. For the initial investigation, we have assumed that the TXOPLimit value is fixed at 10 ms.

Figures 7 and 8 depict the network performance in terms of mean delay and jitter for the three different buffer sizes. Consistent results are obtained for both metrics. It is observed that the smallest buffer size ensures the minimal average delay and jitter for the applications. The difference becomes more evident when the network operates at heavy loads. As more loads are introduced, nodes with bigger buffer have more queued packets to be transmitted. Subsequently, these packets wait longer periods to gain media access resulting in longer transfer delays and jitters. All the considered source types exhibited similar behavior. For example, considering the voice application, it experienced an increase of more than 70% in both delay as well as jitter when buffer size is increased from 32 KB to 128 KB for the network of 100 nodes.

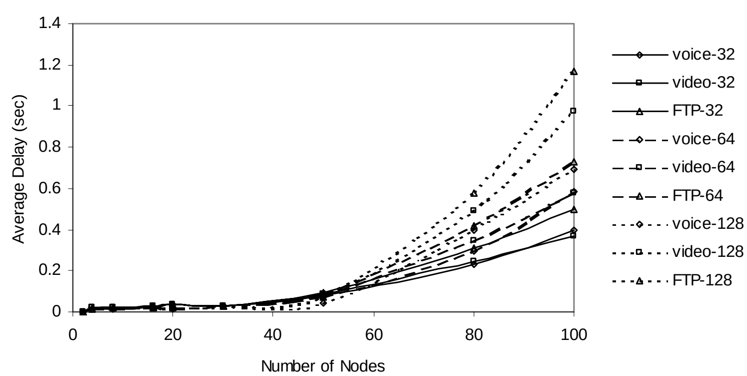

Fig. 7. Mean frame transfer delay vs. number of nodes for different MAC transmit buffer sizes.

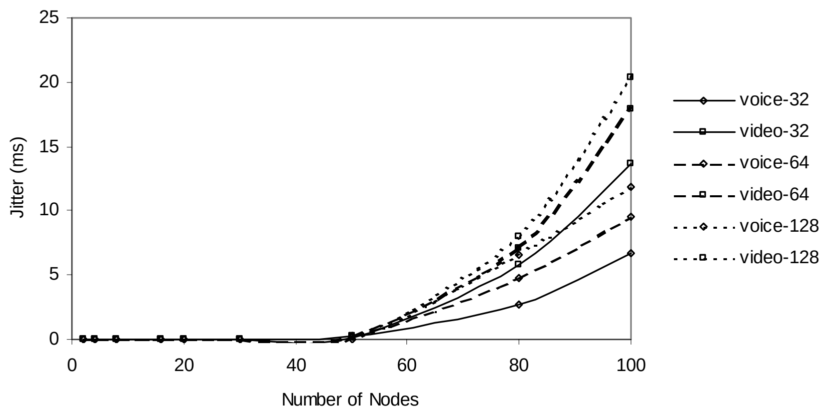

Fig. 8. Delay jitter vs. number of nodes for different MAC transmit buffer sizes.

In Fig. 9, the throughput performance is plotted against network size for the different buffer sizes. Contrary to the above delay performances, the nodes with smaller buffer size exhibit lower throughput especially when the network has more than 50 nodes. This behavior is due to the increased contentions at higher loads (see Fig. 4), which results in packets waiting longer periods to gain access. As the rate of packet transfer decreases, the throughput growth slows and saturates at a certain level. This effect is more pronounced for the smallest buffer size. Video sources enjoy the highest throughput due to their data rates and access priority. Consistent results to supplement the throughput behavior is shown in Fig. 10. It is seen that packet loss probability is minimal when limited nodes exist in the network. As the network grows, the packets tend to wait longer, and newly arriving packets may be dropped as the buffer space runs out. This is more evident for nodes with smaller transmit buffer as expected. Thus, it is obvious that in order to sustain a higher throughput and a lower packet loss probability, a bigger buffer is necessitated. However, as seen earlier, bigger buffers tend to increase mean delay and jitter values, which may be intolerable for certain inelastic applications like voice and video. A single buffer size to fit all cases may not be possible. Instead, we may use separate queue thresholds for each AC, and any incoming packets exceeding its queue threshold may be proactively dropped, as its deadline might not be met. Such a move might reduce the transmission of useless packets, while achieving a fairer utilization among the nodes.

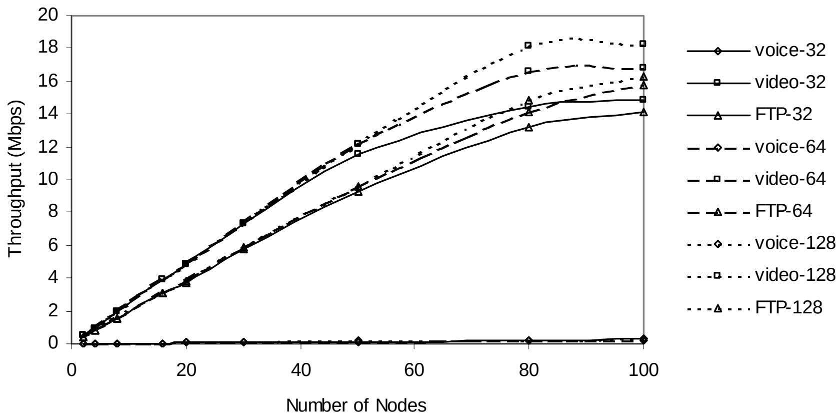

Fig. 9. Throughput vs. number of nodes for different MAC transmit buffer sizes.

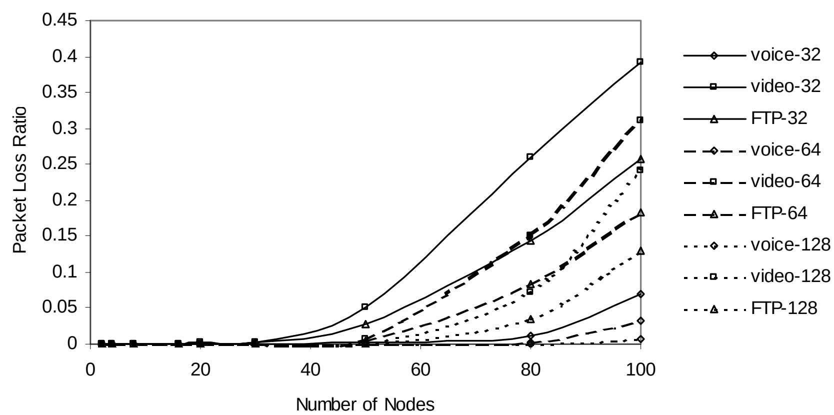

Fig. 10. Packet loss ratio vs. number of nodes for different MAC transmit buffer sizes.

In the further study, we chose to use 64 KB as the buffer size, and vary TXOPLimit to see its impact on the network performance. Based on the analysis performed earlier, we have accordingly selected 0.5 ms, 5 ms and 10 ms for this limit. The following plots show the chosen limits as part of their legend.

Figures 11 and 12 show the mean delay and jitter performance for various TXOPLimits. It is observed in Fig. 11 that mean delay increases significantly for all the cases when *N* is more than 50 nodes. EDCA with the smallest TXOPLimit duration (i.e. 0.5 ms) exhibits a wide-varying delay especially for larger networks as each TXOP only occupies the medium for a short period. Its delay performance is unpredictable, and behaves more like the basic EDCA scheme without the CFB option. For larger limits, nodes are able to transmit more frames without contentions, while achieving more controlled and predictable use of the media. Between 5 ms and 10 ms, there is no significant difference in delay for all the traffic sources. The jitter metric comparison is shown in Fig. 12. Jitter values for all the sources are mostly within acceptable range for the different burst limits.

Figure 13 displays the results of throughput metric for different network sizes. Here, it is evident that the CFB option enables the EDCA protocol to achieve significantly higher throughput in an error-free condition. With frame-bursting, the network is able to efficiently utilize the network and achieve higher throughput especially at higher loads. However, any increase of the burst duration beyond 5 ms has only negligible improvement on the throughput. A consistent result is also obtained for the loss probability as shown in Fig. 14. As before, the smallest burst-limited EDCA protocol faces significant frame losses especially when *N* goes beyond 50 nodes. Again, there is little improvement in frame loss probability when the TXOPLimit is increased beyond 5 ms. Thus, the 5-ms limit ensures a more controlled sharing of the media. By prohibiting nodes from over or under occupying the medium, we are able to ensure an optimized performance and controlled collision rate. And this observation is entirely consistent with the numerical results presented earlier, thus validating the mathematical model.

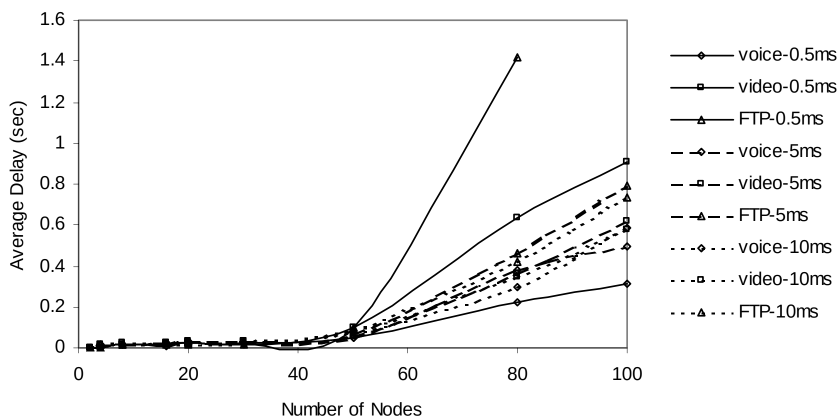

Fig. 11. Mean frame transfer delay vs. number of nodes for different TXOPLimit values.

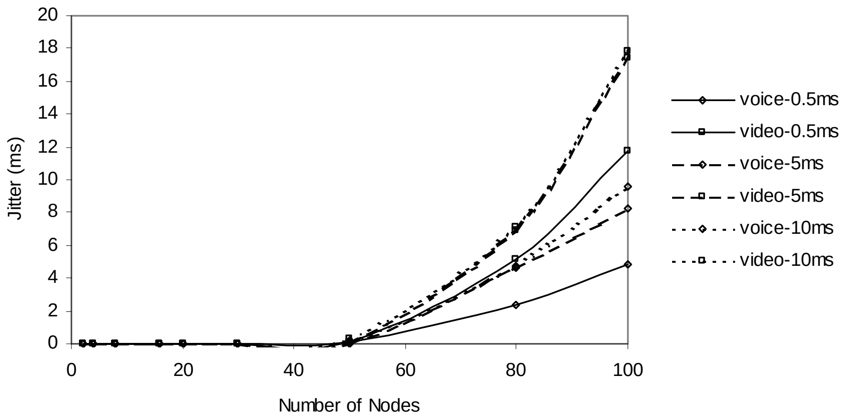

Fig. 12. Delay jitter vs. number of nodes for different TXOPLimit values.

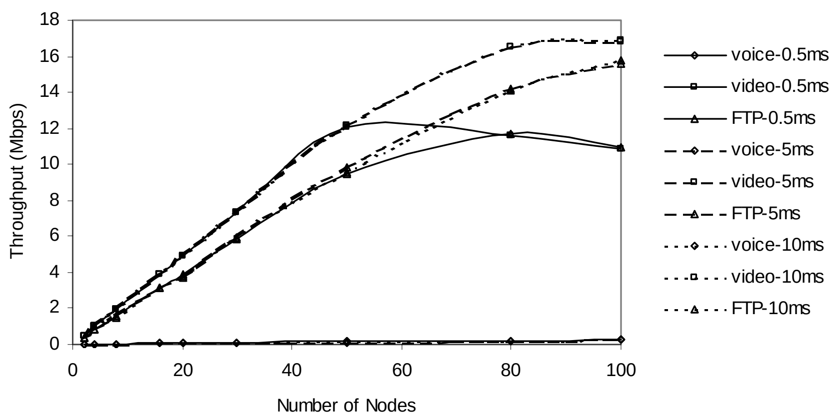

Fig. 13. Throughput vs. number of nodes for different TXOPLimit values.

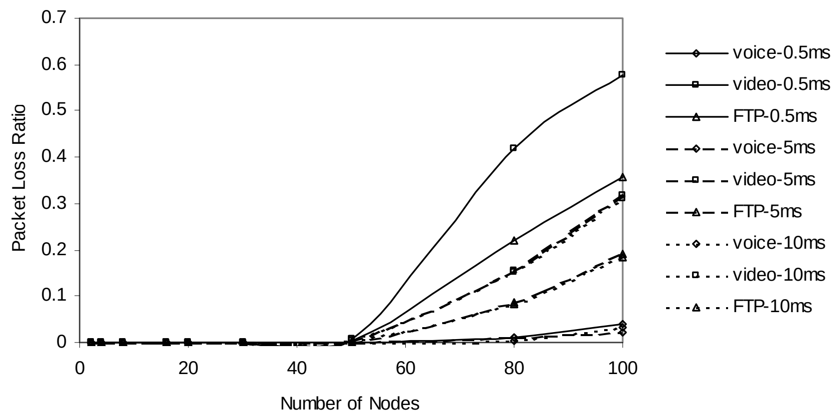

Fig. 14. Packet loss ratio vs. number of nodes for different TXOPLimit values.

To illustrate the significant improvement obtained using the CFB feature, we plotted the overall network throughput against network size for the same TXOPLimits, shown in Fig. 15. It is evident that the bursting option allows the MAC protocol to achieve substantially higher utilization especially in larger networks. For a shared network, the basic EDCA scheme may only be suitable for small to medium-sized networks. Any BSS with more than 40 nodes will suffer significantly when only the basic EDCA scheme is employed. It is found that the CFB feature utilizing a 5-ms limit achieves a throughput increase of about 50% than the limited bursting option of 0.5 ms when the channel is error-free and absent of hidden/exposed terminal problems. This feature allows the MAC protocol to realize more than 60% of the physical data rate. Setting the limit beyond 5 ms does not result in any further improvement.

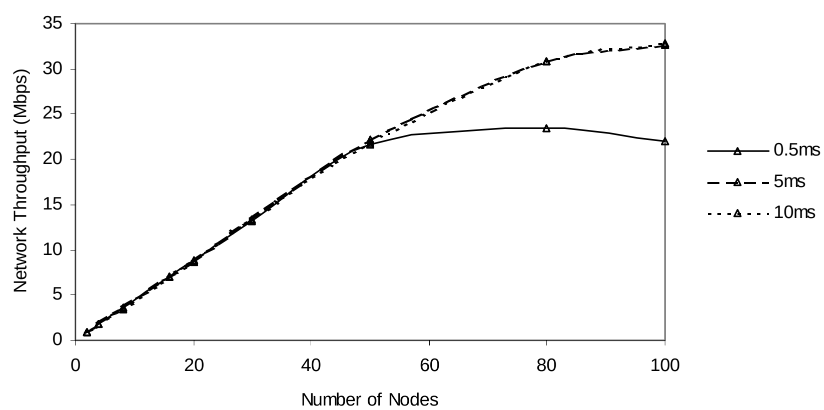

Fig. 15. Network throughput for different TXOPLimit values.

Therefore, it is conclusive that employing the frame bursting option is indeed beneficial especially in terms of EDCA scalability and its support for various traffic mix. It is also evident that proper configuration of the TXOPLimit variable is necessary for an optimized operation of the EDCA protocol. Furthermore, the optimized CFB configuration allows the MAC protocol to realize 30% more capacity of the available physical data rate.

6. Conclusions and Future Work

Distributed QoS support for 802.11 networks is seeing a lot of interests from the researchers as the technology becomes more accessible to the general users. As the users begin to use various multimedia applications, these wireless networks support of such applications is crucial as their bandwidths are more limited. EDCA is one such protocol proposed to address the QoS issue in the upcoming IEEE 802.11e standard.

It is clearly demonstrated here and elsewhere that the basic EDCA operation fails to scale well. It was proposed in the standard that EDCA could use the frame bursting technique to increase utilization when a station wins access contention. In this paper, we have shown that the frame-bursting option can indeed be used to achieve improved performance and network utilization. However, it is shown that for an optimized operation, the proper configuration of the TXOPLimit variable is crucial along with the MAC transmit buffer size. It is also shown that a bigger buffer size may not necessarily improve the inelastic applications' performance. It may be helpful to adopt a separate queue threshold for each traffic class to guide a proactive packet discard to realize a performance trade-off.

As a future work, we would introduce a bit-error model into the analysis to investigate its influence on the bursting mechanism. This study would allow us to use the link status to guide the burst limit that would bounded by the current error-free TXOPLimit.

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