

# DESIGN AND PERFORMANCE EVALUATION OF A NON-PREEMPTIVE PRIORITY BASED ACCESS CONTROL SCHEME FOR IEEE 802.11 DCF ACCESS METHOD<sup>+</sup>

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## ABSTRACT

IEEE 802.11 is a standard for wireless LANs. The basic access method in its medium access control (MAC) layer protocol is the distributed coordination function (DCF) for the ad hoc networks. It is based on the mechanism of carrier sense multiple access with collision avoidance (CSMA/CA). DCF is used to support asynchronous data transmission. However, it cannot support service discipline of integrated multimedia traffic since it does not include any priority and access control policy. With a little bad luck, a station might have to wait arbitrarily long to send a frame. In this paper, we propose a pragmatic non-preemptive priority based access control scheme for IEEE 802.11 DCF access method. Under such a scheme, modifying the CSMA/CA protocol in the contention period supports many levels of priorities such that real-time applications can be supported. Simulations are conducted to analyze the proposed scheme. The results show that it provides a good performance in IEEE 802.11 DCF environments.

**Key words:** Wireless LAN, CSMA/CA, Priority, Multimedia Applications

## 1. INTRODUCTION

Wireless Local Area Network (WLAN) is a rapidly emerging field of activity in computer network because it supports mobility. Currently, there are two emerging WLAN standards: the European Telecommunications Standards Institute (ETSI) High Performance European Radio LAN (HIPERLAN) [1] and the IEEE 802.11 WLAN [2]. There are also several other proposals under study [3-4]. Most draft standards cover the physical layer and medium access control (MAC) sublayer of the open systems interconnection (OSI) seven-layer reference model.

The IEEE standard for WLANs was initiated in 1988 as IEEE 802.4L, a part of the IEEE 802.4 token bus wired LAN standard. In 1990 the IEEE 802.4L changed its name to IEEE 802.11 to form a WLAN standard in the IEEE 802 LAN standards organization. The scope of the standard is "to develop a MAC sublayer and Physical Layer (PHY) specification for wireless connectivity for fixed, portable and moving stations within a local area". The standard defines the basic media and configuration issues, throughput requirements, transmission procedure, and

range characteristics for WLAN technology.

However, frames in DCF, the basic access method in IEEE 802.11 MAC layer protocol, do not have priorities, and there is no other mechanism to guarantee an access delay bound to the stations. To put it another way, real-time applications like voice or live video transmission may suffer with this protocol. Since the demand for transferring delay-sensitive data in wireless environment is evident from the evolution of new data communication applications, several works have been investigated and discussed in this area [5-10]. MACA/PR (multiple access collision avoidance with piggyback reservation) [10], an asynchronous network based on the collision avoidance MAC scheme employed in the IEEE 802.11 standard, provides QoS guarantees to real-time traffic by reserving bandwidth in the ensuing contention-free period. However, this scheme deviates from pure carrier sensing methods in that every node has to construct channel state information based on reservation requests carried in packets sent onto the channel. The black-burst (BB) contention mechanism [7] gives higher priority and periodical access to real-time traffic and ensures collision-free transmission of real-time packets. One important element that is still missing from this architecture is that round-robin discipline among real-time nodes is good for transmitting deterministic traffic. However, constant peak cell rate (PCR) based bandwidth allocation for variable bit rate (VBR) traffic results in unused traffic slots and channel utilization is degraded. Besides, this approach fails when hidden terminals exist as those hidden terminals may have experienced the same delay and each BB contention period is not guaranteed to produce a unique winner, thus real-time data packets will still suffer from collisions [5]. A simple priority schemes for IEEE 802.11 DCF access method has been proposed in [9]. A high priority station has a shorter waiting time when accessing the medium. Performance for transporting integrated traffic is also examined.

In this paper, we propose an advanced, pragmatic, and yet more flexible method to modify the DCF access method such that station's multi-levels priorities can be supported along with the real time applications in an ad hoc network. Under such a scheme, a high priority station has a shorter waiting time when accessing the medium. Furthermore, when collision occurs, a high priority station can also have advantage in accessing the medium. Besides, the proposed adaptive contention window mechanism and

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adaptive bandwidth management strategy not only dynamically expand and contract the contention window size according to the current load but also separate admitted inactivated real-time traffic from newly requesting access traffic and data traffic to maintain an efficient bandwidth utilization and guarantee a minimum bandwidth for data traffic. The proposed scheme is performed at each station in a distributed manner, and it can be implemented in the present IEEE 802.11 standard with relatively minor modifications. Performance for integrated traffic is examined in detail. The results show good performance improvements over the original DCF protocol.

The remainder of this paper is organized as follows. In Section 2, we explain some terminology in IEEE 802.11 standard and the CSMA/CA protocol on which our proposed scheme is based. Section 3 describes the proposed scheme in detail. Simulation and its results are shown in Section 4. Section 5 concludes this paper.

## 2. PRELIMINARIES

The IEEE 802.11 standard considers two network topologies: ad hoc and infrastructure-based. In an ad hoc configuration, the mobile terminals communicate with each other in an independent basic service set (BSS) without connectivity to the wired backbone network. In an infrastructure network, mobile terminals communicate with the backbone network through an access point (AP). The AP is a bridge supporting range extension by providing the integration points necessary for network connectivity between multiple BSSs, thus forming an extended service set (ESS).

The fundamental access method of the IEEE 802.11 MAC protocol for ad hoc networks is the distributed coordination function (DCF), also known as carrier sense multiple access with collision avoidance (CSMA/CA). CSMA/CA works by a "listen before talk" scheme. In brief, to transmit a station must sense the medium to determine if another station is transmitting and must ensure that the medium is idle for the specified distributed coordination function interframe space (DIFS) duration before transmitting. Three interframe space (IFS) intervals are specified in the IEEE 802.11 standard: short IFS (SIFS), PCF-IFS (PIFS), and DCF-IFS (DIFS). The SIFS interval is the smallest IFS, followed by PIFS and DIFS respectively. As a result, PCF traffic has high priority over DCF traffic. Control frames, which wait SIFS before transmission, have the highest-priority access to the communication media.

The CSMA/CA protocol allows for options that can minimize the amount of bandwidth wasted when collision occurs by using request to send (RTS), clear-to-send (CTS), data and acknowledge (ACK) transmission frames in a sequential fashion. Communications is established when one of the wireless nodes sends a short message RTS frame. The message duration, NAV, alerts all others in the medium to back off for the duration of the transmission. The receiving station issues a CTS frame which echoes the senders address and the NAV. If the CTS frame is not received, it is assumed that a collision occurred and the

RTS process starts over. After the data frame is received, an ACK frame is sent back verifying a successful data transmission. A common limitation with wireless LAN systems is the "Hidden Terminal" problem [11]. This can disrupt 40% or more of the communications in a highly loaded LAN environment [12]. It occurs when there is a station in a service set that cannot detect the transmission of another station to detect that the media is busy. The use of RTS, CTS, Data and ACK sequences can also help the prevention of the disruptions caused by this problem.

Despite these precautions, collision can still occur. The collision avoidance portion of CSMA/CA is performed by a random backoff procedure. If a station with a frame to transmit initially senses the channel to be busy; then the station waits until the channel become idle for DIFS period, and then computes a random backoff time to wait before sensing again to verify a clear channel on which to transmit. If the channel becomes busy before time out, the station freezes its timer. This process is repeated until the waiting time approaches zero and the station is allowed to transmit. The idle period after a DIFS period is referred to as the contention window (CW). For more information about IEEE 802.11 standard, see [2] and [13-15].

## 3. THE PROPOSED NON-PREEMPTIVE PRIORITY BASED ACCESS CONTROL SCHEME

In this section, we delineate the proposed scheme in detail. The proposed scheme is simple, efficient, flexible, scalable, and also easy to implement in comparison to PCF (The PCF protocol is extremely complex and has substantial delay at low load, i.e., stations must always wait for the polling, even in an otherwise idle system). Our method can be divided into three parts: enforcing priorities for DCF access, the adaptive contention window mechanism, and the adaptive bandwidth management strategy.

### 3.1 Enforcing Priorities for DCF Access

In WLANs, the medium access control (MAC) protocol is the key component that provides the efficiency in sharing the common radio channel among various multimedia traffic, each requiring different QoS performance; thus, imposing the necessity of prioritization. In this section, we proposed a novel method to modify the CSMA/CA protocol such that station's multi-levels priorities can be supported along with the real time applications in an ad hoc network.

As mentioned earlier, the collision avoidance portion of CSMA/CA is performed through a random backoff procedure. The random backoff time is an integer value that corresponds to a number of time slots. Initially, a station computes a back-off time in the range 0-7. If a station with a frame to transmit initially senses the channel to be busy, it waits until the channel becomes idle, and then, the station decrements its backoff timer until the medium becomes busy again or the timer reaches zero. If the timer has not reached zero and the medium becomes busy, the station freezes its timer. When the timer finally decrements to zero, the station transmits its frame. If two or more stations decrement to zero at the same time, a collision will

occur, and each station will have to generate a new backoff time in the range 0-15. For each retransmission attempt, the backoff time grows as  $\lfloor \text{ranf}() \cdot 2^{2+i} \rfloor \cdot \text{Slot\_Time}$ , where  $i$  is the number of consecutive times a station attempts to send a frame,  $\text{ranf}()$  is a uniform variate in (0,1), and  $\lfloor x \rfloor$  represents the largest integer less than or equal to  $x$ . For more information about CSMA/CA protocol, see [2].

The basic idea of this method is that priority access to the wireless medium is controlled through different backoff time. The shorter backoff time a mobile waits, the higher priority this mobile will get. Therefore, we first change the backoff time generation function to  $\lfloor \text{ranf}() \cdot \frac{2^{2+i}}{2} \rfloor$  for high priority mobiles and  $\frac{2^{2+i}}{2} + \lfloor \text{ranf}() \cdot \frac{2^{2+i}}{2} \rfloor$  for low priority mobiles. This technique divides the random backoff time into two parts:  $0 \sim \frac{2^{2+i}}{2} - 1$  and  $\frac{2^{2+i}}{2} \sim 2^{2+i} - 1$ . The high priority mobiles use the former and the low priority mobiles use the latter. For example, initially, the high priority mobiles generate a backoff time in the range 0-3, and the low priority mobiles generate a backoff time in the range 4-7. Thus, the former will have higher priority in contending the channel.

Then, to support multiple-level priorities, the backoff time generation function is changed to  $\lfloor \text{ranf}() \cdot 2^{1+i} \rfloor + k \cdot 2^{1+i}$ , where  $k$  is the level of priority. The lower  $k$  a mobile has, the higher priority this mobile will get.

Consecutive time (i)	Backoff slot numbers			
	1 <sup>st</sup>	2 <sup>nd</sup>	3 <sup>rd</sup>	4 <sup>th</sup>
Admitted voice traffic (0, 1, 1)	0-3	0-7	0-15	0-31
Admitted video traffic (1, 1, 1)	4-7	8-15	16-31	32-63
New quest/Data traffic (2, 2, 1)	8-15	16-31	32-63	64-127

Figure 1. Examples of backoff time of individual traffic.

However, for fixed backoff range, the probability of collisions in the same priority level will increase if the number of contended mobiles with the same priority increases. In other words, this scheme should have the ability to expand or contract the backoff range arbitrarily. That is, we allow different backoff ranges for different priority levels in this scheme. To this end, we change the backoff time generating function to  $\lfloor \text{ranf}() \cdot 2^{m+i} \rfloor + k \cdot 2^{n+i}$ , where  $k$  is the level of priority, and  $m$  and  $n$  are the parameters used to decide the number of slots in individual priority levels and the number of slots between each priority levels respectively. In this paper, the admitted voice traffic have the highest priority among all other requests, and the second priority

class is admitted activated/inactivated video traffic. The new requests and data traffic will have the lowest priority level, as an example shown in Fig. 1. Note that we give wider range to the lowest priority level since there is more traffic that contends in this priority level.

It is also noteworthy that when a mobile decrements its backoff timer and the medium becomes busy, the mobile freezes its timer. This means that a mobile will raise its priority automatically after several times of transmission failure. Hence, starvation will not occur in this method.

### 3.2 The Adaptive Contention Window Mechanism

The collision avoidance portion of CSMA/CA is performed by a variable time-spreading of the users' access. However, collisions still occur if two or more stations select the same backoff slot, then these stations have to reenter the competition with an exponentially increasing CW parameter value, and the increase of the CW parameter value after collisions is the reaction that the CSMA/CA provides to make the access mechanism adaptive to channel conditions. This strategy avoids long access delays when the load is light because it selects an initial (small) parameter value of CW by assuming a low level of congestion in the system. However, it causes a high collision probability and channel utilization is degraded in bursty arrival or congested scenarios. In other words, this strategy might allocate initial size of CW, only to find out later that it is not enough when the load increased. The size of CW must be reallocated with a larger size, but each increase of the CW parameter value is obtained paying the cost of a collision (bandwidth wastage). Besides, after a successful transmission, the size of CW is set again to the minimum value without maintaining any knowledge of the current channel status.

There has been several works reported in the literature about the enhancement of the IEEE 802.11 DCF protocol to increase its performance when it is utilized in WLANs [16-20]. Most of them were on the simulation study. Some studies try to approximate the knowledge about the number of users involved in the access by exploiting the history of the system. However, it is worth observing how this investigation could result expensive, difficult to obtain and subject to significant errors, especially in high contention situations [20]. In the previous section we mentioned that our scheme has the ability to expand or contract the backoff range arbitrarily by changing the parameters  $k$ ,  $m$  and  $n$ . In this section, we propose an adaptive contention window mechanism for IEEE 802.11 DCF protocol to dynamically expand and contract the contention window size according to the current load and achieve the theoretical capacity limits. Our scheme is based on the results of the capacity analysis model of IEEE 802.11 protocol originally proposed by L. Bononi et al. in [19-20], and we also use the concept introduced in [20-21]. The proposed scheme is simple and efficient, and it does not suffer from the problem of harmful fluctuation occurred in [18]. In addition, the proposed scheme is based only on carrier sensing and can be implemented in the present IEEE 802.11 standard without requiring any complicated computation or additional hardware support.

In order to exploit the early and meaningful information about the actual state of congestion of the channel, we start by defining the utilization factor  $\alpha$  of a contention window to be the number of busy slots,  $s_b$ , observed in the latest  $2^{m+i}$  slots divided by the size (number of slots) of the current contention window ( $2^{m+i}$ ). For the use we considered to make with, the value of  $\alpha$  has to be updated in every backoff interval to reflect the actual state of the channel. It is intuitive to observe that the value of utilization factor,  $\alpha$ , provides a lower bound to the actual number of stations trying to access the channel during the last contention window because some stations may transmit in the same slot. However, in order to maximum the utilization of every slot in a contention window, we still need to engineer the tight upper bound of  $\alpha$  to help us to complete this scheme.

Assume there are  $M$  stations working in asymptotic conditions in this system. The stations transmit frames whose size are i.i.d. (independent and identically distributed) sampled from a geometric distribution with parameter  $q$ . Specifically, the size of a frame is an integer multiple of the slot size,  $t_{slot}$ , and hence the mean frame space is  $\frac{t_{slot}}{1-q}$ . To simplify the analysis, we assumed a geometrically distributed backoff instead of the uniform sampled backoff of IEEE 802.11. That is to say, at the beginning of an empty slot a station starts the transmission of a frame with probability  $p$ , and defers the transmission with probability  $1-p$ . Hence, the average size of contention window is  $1 + \frac{2}{p}$ .

Let  $t_f$ ,  $t_v$  and  $t_s$  denote, respectively, the average frame transmission time, the average temporal distance between two consecutive successful transmission, also referred to as the average virtual transmission time, and the average time required for a successful transmission. Hence, the protocol capacity,  $\rho$ , is  $\frac{t_f}{t_v}$ . Also, from the

geometric backoff assumption, all the processes which define the occupancy pattern of the channel are regenerative with respect to the sequence of time instants corresponding to the completion of a successful transmission. Therefore, we can analytically investigate the value of the  $p$  parameter that minimizes  $t_v$ , named the optimal  $p$  value,  $p_{opt}$ . In other words,  $p_{opt}$  is the  $p$  value that maximizes the protocol capacity,  $\rho$ .

In [19], it is shown that  $p_{opt}$  is closely approximated by the  $p$  value that guarantees a balance, in a virtual transmission time, between collision and idle periods. Furthermore, results presented in [20] indicate that, if the stations utilize the optimal  $p$  value, i.e.,  $p_{opt}$ , the collision probability is low. Hence, it reveals that  $M \cdot p_{opt}$  is a tight upper bound of  $\alpha$  in a system operating with the optimal channel utilization level.

It is worth nothing that  $p_{opt}$  is highly affected by the  $M$  value, however, by fixing a given value for the frame size parameter  $q$ , the value of  $M \cdot p_{opt}$  is almost constant [20], and can be computed off-line by exploiting

the analytical model presented above. Consequently,  $M \cdot p_{opt}$  can be used as a measure of the network contention level when the network utilizes the optimal contention window size corresponding to the ongoing network and traffic configuration. In other words, if the utilization factor,  $\alpha$ , is bounded above by the constant,  $M \cdot p_{opt}$ , the system will be kept in a better situation from the viewpoint of channel utilization since the lower rate of collisions will result in a significant increase in throughput and decrease in mean access delay.

Thus, when the utilization factor,  $\alpha$ , exceeds the value of  $M \cdot p_{opt}$ , we can expand the contention window by allocating more slots than the old contention window had. On the other hand, it is also desirable to contract the contention window when the utilization factor becomes too small, so that the wasted bandwidth is not exorbitant.

A natural strategy for expansion and contraction is to allocate a new contention window size that has twice as many slots as the old one by increasing the value of  $m$  when the utilization factor exceeds the value of  $M \cdot p_{opt}$ , and halve the size of contention window when the utilization factor becomes less than half of  $M \cdot p_{opt}$  by decreasing the value of  $m$ . This strategy guarantees that the utilization factor of the contention window never drops below  $\frac{1}{2}(M \cdot p_{opt})$ . However, such a common heuristic

would conduct the size of contention window to fluctuate rapidly between expansion and contraction. To avoid this behavior, we can improve upon this strategy by allowing the utilization factor of contention window to drop below  $\frac{1}{2}(M \cdot p_{opt})$ . Specifically, we continue to double the size of contention window when the utilization factor exceeds  $M \cdot p_{opt}$ , but we halve the size of contention window when the utilization factor becomes less than  $\frac{1}{4}(M \cdot p_{opt})$ , rather than  $\frac{1}{2}(M \cdot p_{opt})$  as before. That is, the utilization factor is therefore bounded below by  $\frac{1}{4}(M \cdot p_{opt})$ . Please note that the parameters,  $m$  and  $n$ , of other traffic should be adjusted at the same time while changing the value of  $m$ .

### 3.3 The Adaptive Bandwidth Management Strategy

In section 3.1, we have mentioned that starvation will not occur in this method since a mobile will raise its priority automatically after several times of transmission failure. However, we still need guarantee a minimum bandwidth for data traffic to maintain a reasonable bandwidth usage. In this section, we propose an adaptive bandwidth management strategy. Our strategy not only tries to maximize the bandwidth utilization but also guarantee a minimum bandwidth for data traffic. In addition, this strategy is very simple and easy to implement without any extra computation.

As shown in Fig. 2, the total bandwidth is divided into two parts: channel I and channel II. We allocate channel I to real-time/data traffic and channel II to new-call/data call traffic. Beside, we allow admitted real-time traffic to use

bandwidth exclusively with preemptive priority over data traffic in channel I to let the real-time traffic has a larger share of pie, and data traffic sometimes has precedence for using network resources than new-call traffic in channel II to guarantee a minimum bandwidth for data traffic. The algorithm to manage the bandwidth is summarized in the following.

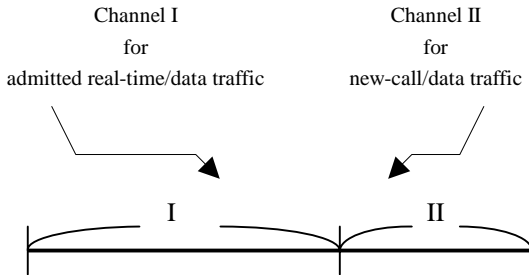


Figure 2. Proposed bandwidth partition.

### Function Adaptive\_Bandwidth\_Management

**IF** monitored bandwidth of data traffic < size of allocated bandwidth II **THEN**

**IF** bandwidth utilization <  $\mu$  **THEN**

$$n_{new\_call} = n_{data}$$

**ELSE**

$$n_{new\_call} = n_{data} + 1$$

As the pseudo-code illustrates, the monitored bandwidth of data traffic and bandwidth utilization are the measures used to manage the bandwidth. If the monitored bandwidth of data traffic becomes less than the size of allocated bandwidth II, and the bandwidth utilization is not good enough (less than the threshold value  $\mu$ ), it implies that there is not so much real-time traffic. Hence, we let the new-call traffic has the same priority with data traffic by keeping the value of parameters of new-call traffic as the same as data traffic. Otherwise, we guarantee a minimum bandwidth for data traffic by delaying the new-call traffic in high load. Please note that we can also guarantee a proper bandwidth for voice/video traffic in the same way. This algorithm can be run periodically.

## 4. SIMULATIONS AND PERFORMANCE EVALUATION

In this section, we evaluate the performance of the proposed scheme.

### 4.1 Simulation Model

The simulation models are built using the Simscript tool [22]. The model represents an ad hoc network, where all stations in the BSS (Basic Service Set) are capable of directly communicating with all other stations in the BSS. Several assumptions have been made to reduce the complexity of the model. First, the “hidden terminal” and “exposed terminal” problems [11] are not addressed in the simulation model. Second, no mobiles operate in the “power-saving” mode. Third, no interference is considered from nearby BSSs. Finally, the probability that the frame is transmitted successfully is calculated as:  $p_r\{\text{success}\} =$

$(1 - BER)^n$ , where  $n$  is the number of bits transmitted in the frame and BER denotes bit error rate.

Three types of traffic are considered in the simulation.

#### 1. Pure data

The arrival of data frames from a mobile’s higher-layer to MAC sublayer is Poisson. Frame length is assumed to be exponentially distributed with mean length 1024 octets.

#### 2. Voice traffic

Voice stream is characterize by two parameters ( $\gamma_c, \delta$ ), where  $\gamma_c$  is the rate of the source and  $\delta$  is the maximum tolerable jitter (packet delay variation) for this stream. Frames of voice traffic that are not successfully transmitted within its maximum jitter constraint is assumed to be lost. The voice stream is modeled as an two state Markov on/off process, where stations are either transmitting (on) or listening (off). The amount of time in the off or on state is exponentially distributed, where the mean value of the silence (off) period is 1.5 s, and the mean value of the talk spurt (on) period is 1.35 s. Each connection duration is exponentially distributed with mean time 3 minutes.

#### 3. Video traffic

Video stream is characterized by three parameters ( $r_v, \beta, d$ ), where  $r_v$  is the average rate of the source,  $\beta$  is the maximum burstiness of the source, and  $d$  is the maximum tolerable delay (packet transfer delay) for this stream. We use a Source Model in [23]. The bit rate of a single source for the  $n$ th frame,  $\lambda(n)$ , is defined by the recursive relation:  $\lambda(n) = a\lambda(n-1) + bw(n)$  [bit/pixel], where  $a=0.8781$ ,  $b=0.1108$ , and  $w(n)$  is a sequence of independent Gaussian random variables which have mean 0.572 and variance 1. Like voice frames, video frames that are not successfully transmitted within its maximum tolerable delay,  $d$ , is assumed to be lost.

Assume video, voice and data are mixed in the ratio of 1:1:1. Performance is measured in terms of average access delay, loss probability, bandwidth utilization, etc. The default values used in the simulation are listed in Table 1. The values for the simulation parameters are chosen carefully in order to closely represent the realistic scenarios as well as make the simulation feasible and reasonable.

### 4.2 Simulation Results

We compare the proposed scheme with the conventional IEEE 802.11 DCF protocol, and the simulation results are shown below in the form of plots. Figure 3 and 4 show the value of utilization factor,  $\alpha$ , and the collision probability versus the offered load for the proposed scheme and conventional DCF protocol. As shown in the figure, we can see that the value of utilization factor of conventional DCF protocol increase steadily as the load increases, and generally greater than the values of proposed scheme. On the other hand, the collision probability of the conventional DCF protocol degrades severely when the offered load become larger than 0.5 in contrast to the smoothness of the proposed scheme. Understandably, the decrease of the collision probability is achieved by our adaptive contention window mechanism. Since the value of utilization factor is bounded below by a constant. The collision probability will

be kept under the threshold.

Table 1. Default attribute values used in the simulation.

Attribute	Value	Meaning & Explanation
Channel rate	2 Mb/s	Data rate for the wireless channel
Mobiles	10	10 mobile hosts in a basic service set
Slot_Time	20 $\mu$ S	Time needed for each time slot
SIFS	10 $\mu$ S	Time needed for each short interframe space
DIFS	50 $\mu$ S	Time needed for each DCF interframe space
MAC header	272 bits	Header length of MAC layer header
PHY header	128 bits	Header length of physical layer header
RTS	160 bits + PHY header	Frame length of each request-to-send frame
CTS	112 bits + PHY header	Frame length of each clear-to-send frame
ACK	112 bits + PHY header	Frame length of each Acknowledgement
BER	$10^{-6}$	Bit error rate
$\mu$	0.8	Minimum bandwidth utilization wanted
$\Pi$	5 %	Size of allocated bandwidth $\Pi$
$r_c$	32 kb/s	Voice source data rate
$\delta$	32 ms	Tolerable jitter for voice source
$\beta$	5	Maximum burstiness
$d$	50 ms	Maximum packet delay for video source
$M \cdot p_{opt}$	0.2	The theoretical upper bound used to adjust the contention window size

Figures 5 and 6 show average access delay of voice and video traffic respectively. Note that the average access delay of the proposed scheme remains low when the offered load is high, but the conventional DCF protocol shows sharp rise as the load increases. For voice and video traffic, the allowable loss probability is about  $10^{-2}$  [24] and  $10^{-3}$  [25] respectively. With this criterion, the proposed scheme can tolerate an offered load of 0.9. The simulation results indicate that the proposed scheme can be used to transmit high priority real-time applications such as voice and video traffic.

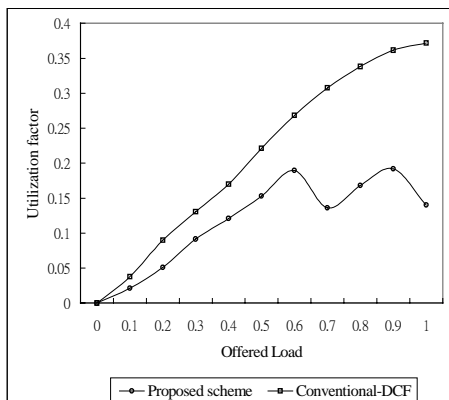


Figure 3. The value of utilization factor versus offered traffic.

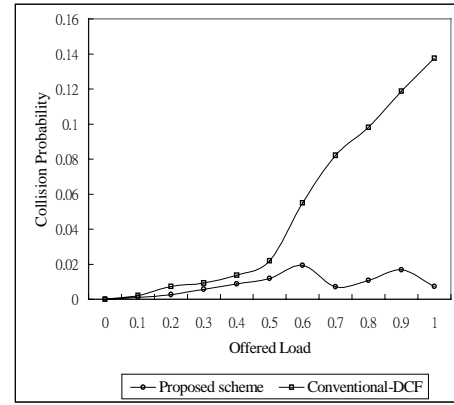


Figure 4. The collision probability versus offered traffic.

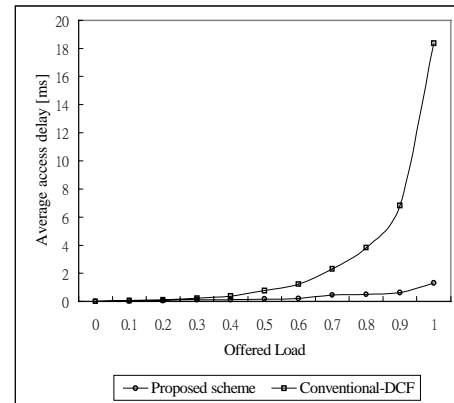


Figure 5. Average access delay of voice traffic.

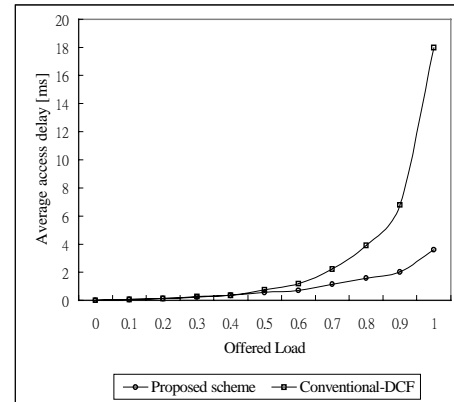


Figure 6. Average access delay of video traffic.

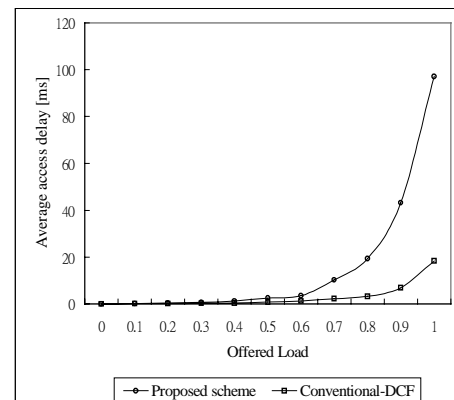


Figure 7. Average access delay of data traffics.

Figure 7 shows the average access delay of data traffic under multimedia traffic condition. As expected, the average access delay of data traffic in the proposed scheme is worse than the conventional DCF protocol since it is of low priority. However, the lower priority traffic can have the bandwidth it needs in light load, so it is not wasted. Besides, we still might guarantee a minimum bandwidth for data traffic to maintain a reasonable bandwidth usage by using the function of adaptive bandwidth management in section 3.3.

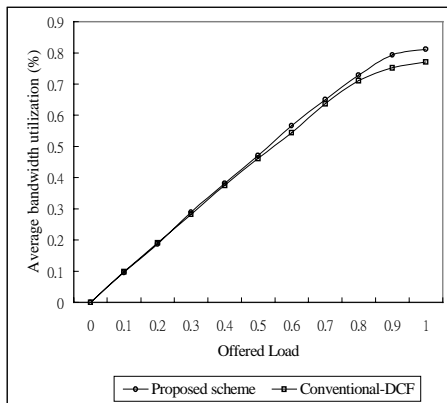


Figure 8. Average bandwidth utilization.

Figure 8 presents the average bandwidth utilization as a function of the offered load. Average bandwidth utilization is the percentage of the bandwidth actually being used in the total bandwidth. As illustrated in Fig. 8, the average bandwidth utilization is higher for the proposed scheme in a highly loaded system. It reveals a better performance for our proposed scheme under heavy load because the lower rate of collision probability results in a significant increase in throughput and decrease in mean access delay.

## 5. CONCLUSIONS

The design of priority-sensitive network protocols continues to be an important problem, and broadband wireless links constitute a subclass where prioritization is key to optimizing overall performance. In this paper, we proposed a pragmatic non-preemptive priority based access control scheme built on well-know protocols and offered easily implemented and yet flexible criteria for traffic prioritization in a wireless environment. We also demonstrate the performance in a quantitative way.

The new generation wireless technologies should support universal wide-band access to a variety of services such as cordless telephony, Internet access, multimedia conference, remote audio, and flexible positioning of audio system. This means that various QoS requirements are needed in the future. Thus, multilevel priorities, bandwidth allocation, connection admission control, and traffic policing all need to be considered together to satisfy various QoS flows in the future networks.

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