Performance of an Aggregation-Based MAC Protocol for High-Data-Rate Ultrawideband *Ad Hoc* Networks

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***Abstract*—Ultrawideband (UWB) communication is an emerging technology that promises to provide high data rate communication for wireless personal area networks. One of the critical challenges in UWB system design is the timing acquisition problem, i.e., a receiver needs a relative long time to synchronize with transmitted signals. Clearly, the timing acquisition overhead will significantly limit the throughput of high data rate UWB *ad hoc* networks. To resolve the timing acquisition problem, the authors proposed a general framework for medium access control (MAC) protocols in their previous work (K. Lu, D. Wu, and Y. Fang, "A novel framework for medium access control in ultra-wideband ad hoc networks," *Dynamics of Continuous, Discrete and Impulsive Systems (Series B)*, vol. 12, no. 3, pp. 427–441, Jun. 2005); under the framework, a transmitting node can aggregate multiple upper layer packets into a burst frame at the MAC layer. In this paper, the authors propose an aggregation-based MAC protocol within the framework. Besides packet aggregation, they also design a novel retransmission scheme which is suitable for error-prone wireless environment, in which only the packets that encounter transmission errors will be retransmitted. To evaluate the performance of the protocol, they develop a three-dimensional Markov chain model for the saturated throughput performance. In addition, they also analyze the end-to-end delay performance through simulation. Extensive numerical and simulation results show that, compared to existing MAC protocols, in which upper layer packets are transmitted one by one, the proposed protocol can drastically reduce the timing acquisition overhead. Consequently, both the throughput and the end-to-end delay performance can be significantly improved.**

***Index Terms*—Bit error rate (BER), burst frame, delay, medium access control (MAC), throughput, timing acquisition, ultrawideband (UWB).**

I. Introduction

DEFINED by Federal Communications Commission (FCC), an ultrawideband (UWB) system is a radio system that has a 10-dB bandwidth larger than 500 MHz or larger than 20% of its center frequency [2]. To promote the development of UWB communications systems, the FCC allocated an unlicensed frequency band 3.1–10.6 GHz for UWB systems in 2002 [2].

Compared to existing narrowband wireless communication systems, UWB can provide high data rate ($>$ 100 Mb/s) in a short range, typically less than 10 m, with much less power emission (less than -41 dBm/MHz). These features are particularly suitable for wireless personal area network (WPAN) applications. Consequently, UWB communication has been studied extensively in both academia and industry in the past few years. For example, the IEEE 802.15.3a working group is currently focusing on UWB as the physical layer technique for providing data rate as high as 480 Mb/s. Implementation schemes for IEEE 802.15.3a have been proposed recently [3], [4] by leading companies such as Motorola, Intel, and TI.

To successfully deploy UWB *ad hoc* networks, a number of challenges have been identified and must be addressed [5], [6]. One of the critical issues is that a receiver usually needs a relative long time to synchronize its clock with transmitted signals, known as the timing acquisition problem [5], [7], [8]. From a physical layer perspective, timing acquisition is generally performed by sending a preamble before information bits [3], [4]. Depending on the receiver design, the duration of a preamble varies from about 10 μs [3], [4] to as high as tens of milliseconds [9]. Obviously, the overhead of preambles will significantly reduce the efficiency of high date rate UWB networks [5].

In the literature, most existing studies addressed the timing acquisition problem through the physical layer design. To reduce the timing acquisition overhead, we recently proposed a general framework for UWB medium access control (MAC) protocols in [1]. The main idea of our framework is to assemble multiple upper layer packets into one burst frame at the MAC layer. In contrast to the traditional approach, where each upper layer packet is delivered individually, transmitting multiple upper layer packets in one frame will significantly reduce the synchronization overhead.

Our framework in [1] is based on the well-known carrier-sensing multiple access with collision avoidance (CSMA/CA) protocol, which is used in both WPAN and wireless local area networks (WLANs) [21]. In the literature, the performance of CSMA/CA protocol has been studied theoretically in a number of existing works. Most of them are based on a saturated assumption, which means that every node in the network has packet to send at any time [10], [11]. In [10], Bianchi first introduced a two-dimensional Markov chain model to evaluate the CSMA/CA protocol with binary exponential backoff scheme.

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Wu *et al.* [11] improved the model to take into account the retransmission limit in the protocol. Although the saturated model cannot be used to analyze the throughput when the incoming traffic load is light and cannot be used to study the end-to-end delay performance since it does not consider the impact of the MAC layer queue, the saturated model can efficiently provide insightful analytical results. Therefore, the saturated model has been widely used in the literature and has been extended to analyze improved CSMA/CA protocols [19], [20].

The main contribution of this paper is to design and analyze the performance of an aggregation-based MAC protocol, within the framework [1]. Specifically, the proposed protocol has two major features: 1) a packet aggregation scheme at the MAC layer and 2) a novel frame retransmission scheme, in which only the packets that encounter transmission errors will be retransmitted. With these salient features, our protocol can significantly improve the performance of UWB *ad hoc* networks even in an error-prone wireless environment.

To evaluate the performance of the protocol, we develop a three-dimensional Markov chain model for the saturated throughput performance. We also evaluate the end-to-end delay performance through simulation, where the end-to-end packet delay is the duration from the epoch that the packet enters the MAC layer at the source node to the epoch that the packet is successfully received at the destination node. Extensive simulation and analysis results show that, compared to sending each upper layer packet individually, which is typical in existing MAC protocols, the proposed scheme can significantly improve throughput and delay performance of a UWB *ad hoc* network under different data rates and different bit error rates (BERs).

The rest of this paper is organized as follows. In Section II, we briefly overview the existing MAC designs for UWB networks. In Section III, we design an aggregation-based MAC protocol for high data rate UWB *ad hoc* networks based on the framework in our previous work [1]. In Section IV, we analyze the saturation throughput performance of the MAC protocol. Extensive simulation and numerical results are shown in Section V. Finally, Section VI concludes this paper.

II. Related Work

The UWB MAC design issue has been studied in a number of previous works [13]–[17]. However, we note that the timing acquisition problem was not adequately addressed in these works. Particularly, the synchronization overhead has only been studied in [13] and [15]. In [13], the authors provided several CSMA/CA based MAC protocols for UWB *ad hoc* networks. An important assumption in this paper is that the UWB communications system can provide multiple channels through different time-hopping (TH) codes. To reduce the timing synchronization overhead, the authors proposed a link maintenance scheme in which the data channel is maintained by transmitting low-rate control packets when there is no data packet to transmit. Although the link maintenance scheme can achieve good performance in the simulation, there are still some critical issues unclear. One of the main issues is that it is not clear how a node can assign a TH code for its data channel in a distributed manner, such that existing data channels are not compromised by the interference of the new code. Another potential problem is that the link maintenance scheme will increase the transmission time of the transmitter, thereby introducing extra interference and reducing the battery life.

In [15], the authors studied the impact of long synchronization time on the performance of CSMA/CA and time-division-multiple-access schemes used in UWB networks. However, this paper did not provide solutions to mitigate the timing acquisition problem.

To reduce the timing acquisition overhead, we recently proposed a general framework for UWB MAC protocols in [1]. The main idea of our framework is to assemble multiple upper layer packets into one burst frame at the MAC layer. In contrast to the traditional approach, where each upper layer packet is delivered individually, transmitting multiple upper layer packets in one frame will significantly reduce the synchronization overhead. It is worth noting that the framework in [1] is based on the well-known CSMA/CA, which has also been assumed in [13] and [15] and which is required by IEEE 802.15.3 [3], [4].[1] Consequently, our framework is also suitable for other CSMA/CA-based high data rate MAC protocols, such as IEEE 802.11n [18].

It is important to note that, CSMA/CA is not the only MAC protocol in IEEE 802.15.3 [22], which is an important protocol for high data rate WPAN. In IEEE 802.15.3, a delayed acknowledgment (dly-ack) scheme is provided to reduce the overhead of control messages. In the dly-ack scheme, the source node will send a sequence of data frames, and the destination node will only reply with a single acknowledgement (ACK) frame. In this scheme, we note that 1) each data frame has its own preamble and 2) two consecutive frames must be separated at least by an interframe space, i.e., short interframe space or minimum interframe space. Clearly, compared to the dly-ack scheme, our approach can further reduce the synchronization overhead; and thus can provide better throughput performance.

III. Aggregation-Based MAC Protocol for High Data Rate UWB *Ad Hoc* Networks

In this section, we provide an aggregation-based MAC protocol based on the framework in [1]. The key ideas of this protocol are as the following. First, a transmitting node can aggregate multiple upper layer packets into a burst frame at the MAC layer and transmit the burst frame to a destination node, instead of transmitting each upper layer packet individually. In this manner, the overhead of timing synchronization is reduced. Second, if errors are detected in one or more packets in a burst frame, the receiving node will notify the transmitting node such that only the packets that encounter transmission errors will be retransmitted in the next frame. In this manner, the efficiency of retransmission can be improved in error-prone wireless environment.

In the framework of [1], five types of policies can be specified to tune the aggregation-based protocol. Specifically, the packet classification policy is used to classify incoming upper layer packets according to their destination and quality of service (QoS) requirements; the buffer management policy is utilized to

[1]The carrier sense requirement is defined in [3] and [4] as the receiver CCA performance, where CCA stands for clear channel assessment.

provide QoS and/or fairness among different flows; the packet assembly policy is responsible for frame aggregation, where synchronization overhead, physical layer constraints, QoS, and fairness among different nodes will be considered; the ACK policy is introduced to specify the ACK procedure at the receiver side; and finally, the packet error control policy is in charge for mitigating packet errors.

In this paper, we consider a fundamental protocol that supports only one QoS class of traffic for each destination. Therefore, an incoming packet can be classified according to its destinations, and then be put into a corresponding packet queue. Suppose there are N nodes in a network; then we can implement N packet queues in each node, among which $N-1$ queues are used for buffering packets destined to other $N-1$ nodes, and one queue is used for buffering broadcast packets. Since there is only one class for each queue, the tail-dropping policy is applied. In the rest of this section, we will focus on the packet assembly policy and the frame retransmission scheme.

A. Packet Assembly Policy

In this policy, it is required that all the packets in a burst frame have the same destination so that most existing functions of IEEE 802.11 can be reused in our protocol. To achieve the fairness among destinations, a simple round-robin scheme is employed. When a burst assembly is finished, the burst frame will be stored in a buffer and waiting for transmission.

To assembly a burst frame, the total number of packets in the queue must exceed a threshold $B_{\min}$ and the server is idle (i.e., there is no other burst frame waiting for transmission). In addition, we assume that the total number of packets in a burst frame must be smaller than or equal to a preset value $B_{\max}$. Specifically, the burst frame assembly procedure is described by the following two algorithms, where N_q denotes the number of packets in the buffer (excluding packets in the server).

Algorithm 1 Burst assembly policy at the transition instance from server busy to server idle

```
procedure BURST ASSEMBLY
  if N_q ≥ B_min then
    if N_q ≤ B_max then
      Assemble a burst frame with N_q packets,
      and move the frame to transmission buffer;
    else
      Assemble a burst frame with B_max packets,
      and move the frame to transmission buffer.
    end if
  end if
end procedure
```

Algorithm 2 Burst assembly policy at a packet arrival when the server is idle

```
procedure BURST ASSEMBLY
  if N_q = B_min then
    Assemble a burst frame with B_min packets,
    and move the frame to transmission buffer.
  end if
end procedure
```

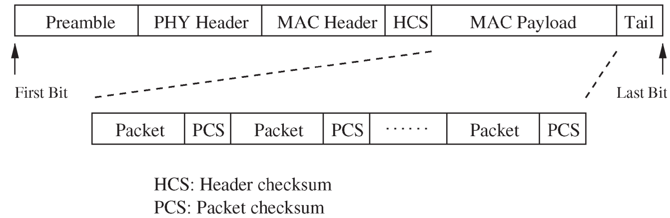


Fig. 1. Format of a frame.

B. Frame Retransmission Scheme

We now discuss the frame retransmission scheme. The main idea of the scheme is to retransmit only the packets that have detected transmission errors. For example, if five (5) packets are transmitted in a frame and only three (3) of them are successfully received without any errors, then in the next frame, the sender only needs to retransmit two (2) packets, which encountered transmission errors in the previous attempt. Compared to retransmitting all packets in a frame, the new scheme can significantly improve the efficiency when the wireless channel is error prone.

To realize the new retransmission scheme, we first need to define the data frame structure as in Fig. 1, where each packet in the frame has its own checksum, i.e., the packet checksum. In addition, the total number of packets in the frame and the length (in bytes) of each packet can be stored in the MAC header.

At the sender side, once a burst frame is aggregated, the checksum of each packet can be calculated and the packet with its checksum can be stored in a transmission buffer. Since the maximum number of packets in a frame is $B_{\max}$, the transmission buffer size can be set as the production of $B_{\max}$ and the maximum length of one packet plus the length of checksum. Before the frame is transmitted, the number of packets in the frame and the length of each packet will be put in the MAC header.

At the receiver side, a decoder is responsible for detecting the transmission error packet by packet. If a packet is correct, then it will be temporarily stored in a reception buffer; otherwise, the packet will be discarded and the index of the dropped packet will be saved. Since the maximum number of packets in a frame is $B_{\max}$, the reception buffer size can be set as the production of $B_{\max}$ and the maximum length of one packet (no checksum). After checking all packets, if none of them has errors, then all the packets in the reception buffer will be forwarded to the upper layer and a positive ACK frame will be returned to the sender.[2] Consequently, the sender will release all the packets in the transmission buffer.

On the other hand, if some of the packets have errors, then the receiver will use the negative acknowledgement (NACK) frame to inform the sender that a set of packets need to be retransmitted. If the sender can correctly received the NACK frame, it will remove all the packets that have been confirmed,

[2] In the traditional CSMA/CA protocol [21], only positive ACK will be used, which means that the receiving node will transmit the ACK frame only if the received frame is correct.

and then try to retransmit the rest of the packets in the next frame. Notice that the MAC header and payload of the next frame must be updated accordingly.

Finally, the transmission buffer will be emptied if the sender has retransmitted a frame for a maximum number of attempts. On the other side, the receiver will forward all received packets in the reception buffer to the upper layer and reset the buffer once it receives a frame with a new sequence number.

IV. Saturation Throughput Analysis

In this section, we develop a three-dimensional Markov chain model to evaluate the saturation throughput performance of the MAC protocol proposed in Section III. To conduct the analysis, we make the following assumptions.

1) There are N identical nodes in a network, in which any two nodes in the network can directly communicate with each other. In other words, we only consider the single-hop scenario.
2) To simplify the notation, we assume that the physical-layer transmission rate for every message is fixed at R (in bits per second).
3) The probability that a frame transmission is failed due to collision, denoted as p, is fixed and is independent of the backoff stage.
4) After a burst frame is successfully delivered (i.e., the transmitter receives a positive ACK) or is removed from the sending buffer after the maximum number of retries has been reached, a node can always assemble a new burst frame that has $B_{\max}$ packets. To simplify the notation, we let $B = B_{\max}$ hereafter.
5) Bit errors can occur uniformly in the payload of one MAC frame and the BER is fixed at a value denoted by ϵ. Similar to [12], we assume that the frame headers are error free.[3]
6) The probability that a packet transmission error occurs, denoted as q, is fixed.
7) The effect of propagation delay is negligible. This assumption is valid in a typical WPAN scenario, which has a communication rage less than 10 m, or a typical WLAN scenario, which has a communication rage less than 300 m (corresponding to 1-μs propagation delay).
8) We ignore the overhead for extra checksum for each packet in a burst frame and the overhead for the indication in the NACK frame.
9) The packet length of every packet follows the same distribution, denoted as $f(n)$, where n is the size of the packet in bytes.

Since the proposed MAC keeps all the frame structures and most of the control procedure in IEEE 802.11, we can use the Markov modeling technique introduced in [10]. In particular, we can partition the continuous time axis into intervals of length $X(t)$, where t is the integer index of an interval. In addition, two consecutive intervals are delimited by the event of a value change in the backoff counter. We can then obtain a three-dimensional discrete time embedded Markov chain with state $\{s(t), b(t), p(t)\}$, where $s(t)$ is the index of the backoff stage at t, $b(t)$ is the value of the backoff counter at t, and $p(t)$ is the number of packets in the burst frame at t.

We now consider the possible states in this three-dimensional Markov chain. Let M be the maximum index of backoff stages, which is the same as the maximum number of retries for a packet. Consequently, the possible value of $s(t)$ is in $\{0, 1, 2, \ldots, M\}$.

Let W and $2^K W$ denote the minimum and maximum backoff window size, respectively. According to the binary exponential backoff scheme in IEEE 802.11, we note that the backoff window size at stage m can be defined as

$$W_m = \begin{cases} 2^m W, & m \le K \\ 2^K W, & m > K \end{cases}. \tag{1}$$

Clearly, for stage $s(t) = m$, the possible value of $b(t)$ is in $\{0, 1, 2, \ldots, W_m - 1\}$.

Based on our assumptions, we can see that the total number of packet in the frame must be B if $s(t) = 0$. On the other hand, if $s(t) > 0$, then the number of packets in a frame can be in $\{1, 2, \ldots, B\}$.

Define the steady-state probability of state $\{s(t) = m, b(t) = i, p(t) = k\}$ as

$$b_{m,i,k} = \lim_{t \to \infty} \Pr\left[s(t) = m, b(t) = i, p(t) = k\right]. \tag{2}$$

To calculate the steady-state probabilities, we can first consider the state transition probability $p_{\xi\eta}$ as the following:

1) $\xi = (m, i+1, k), \eta = (m, i, k), 0 \le i \le W_m - 1$

$$p_{\xi\eta} = 1 \tag{3}$$

2) $\xi = (m, 0, k), \eta = (m+1, i, k), 0 \le m < M$

$$p_{\xi\eta} = \frac{1}{W_{m+1}}\left[p + (1-p) \cdot q^k\right] \tag{4}$$

3) $\xi = (m, 0, k), \eta = (m+1, i, j), j < k, 0 \le m < M$

$$p_{\xi\eta} = \frac{1}{W_{m+1}}(1-p) \cdot \binom{k}{j} \cdot q^j \cdot (1-q)^{k-j} \tag{5}$$

4) $\xi = (m, 0, k), \eta = (0, i, B), 0 \le m < M$

$$p_{\xi\eta} = \frac{1}{W_0}(1-p) \cdot (1-q)^k \tag{6}$$

5) $\xi = (M, 0, k), \eta = (0, i, B)$

$$p_{\xi\eta} = \frac{1}{W_0}. \tag{7}$$

[3]The MAC header is much shorter than the payload and in general will be transmitted at a lower data rate [21], compared to the data rate for sending payload. Consequently, the probability that errors occur in the header is generally smaller than the probability that errors occur in the payload of the frame.

$$\begin{cases} b_{m,i,k} &= \frac{W_m - i}{W_m} \times b_{m,0,k} \\ b_{1,0,B} &= \left[p + (1-p)q^B\right] \times b_{0,0,B} \\ b_{1,0,k} &= (1-p)\binom{B}{k} q^k (1-q)^{B-k} b_{0,0,B}, \quad 1 \le k < B \\ b_{m+1,0,k} &= p \cdot b_{m,0,k} + (1-p) \\ & \quad \times \sum_{j=k}^{B} \left[\binom{j}{k} q^k (1-q)^{j-k} \times b_{m,0,j}\right], \quad 0 < m \le M. \end{cases} \tag{8}$$

Similar to [10], we can then derive the relationships of the steady-state probabilities as in (8), where parameter q can be estimated by using

$$q = 1 - \sum_{n=0}^{\infty} f(n)(1-\epsilon)^{8n}. \tag{9}$$

Since the steady-state probabilities must satisfy

$$1 = \sum_{m,i,k} b_{m,i,k} \tag{10}$$

we can calculate all steady-state probabilities $(b_{m,i,k})$ numerically given p and q.

Now, let p_τ be the probability that a node will transmit a frame in a interval $X(t)$. According to the procedure in binary exponential backoff, in which a transmission is initiated if and only if $i = 0$, we can derive that

$$p_\tau = b_{0,0,B} + \sum_{m=1}^{M} \sum_{k=1}^{B} b_{m,0,k}. \tag{11}$$

Since a frame collision can occur only if there are two or more nodes transmit frame in the same interval, we can calculate p through

$$p = 1 - (1 - p_\tau)^{N-1}. \tag{12}$$

By using the relationship in (11) and (12), we can obtain both p and p_τ numerically.

To calculate the saturation throughput, we define the following parameters.

1) $\widetilde{X}(t)$ be the time duration in $X(t)$ that is used for transmitting error-freed packets.
2) S denotes the normalized throughput at one node, which is defined as the fraction of time that the channel is used to successfully transmit data packets, i.e.,

$$S = \frac{E\left[\widetilde{X}(t)\right]}{E\left[X(t)\right]}.$$

3) p_t denotes the probability that there is at least one packet transmission in an interval of length $X(t)$, which can be calculated by

$$p_t = 1 - (1 - p_\tau)^N. \tag{13}$$

4) p_s denotes the probability that there is only one packet transmission in an interval of length $X(t)$, which can be calculated by

$$p_s = N p_\tau (1 - p_\tau)^{N-1}. \tag{14}$$

5) L denotes the mean size of a packet, which can be expressed with $f(n)$ as

$$L = \sum_{n=0}^{\infty} 8n f(n). \tag{15}$$

6) $\widetilde{L}$ denotes the mean size of an error-freed packet, which can be expressed by

$$\widetilde{L} = \sum_{n=0}^{\infty} 8n f(n)(1-\epsilon)^{8n}. \tag{16}$$

7) σ denotes the length of a preset fixed time duration. When there is no packet transmission, we have $X(t) = \sigma$. For instance, in IEEE 802.11 direct sequence spread spectrum mode [21] $\sigma = 20\ \mu$s.
8) $T_s = E[X(t)|\text{burst frame received without collision}]$.
9) $\widetilde{T}_s = E[\widetilde{X}(t)]$.
10) $T_c = E[X(t)|\text{burst frame collision}]$.

With the above parameters, we can calculate S by

$$S = \frac{p_s \widetilde{T}_s}{(1-p_t)\sigma + p_s T_s + (p_t - p_s) T_c}. \tag{17}$$

We now discuss the calculation of T_s, $\widetilde{T}_s$, and T_c in (17). Since the propagation delay can be ignored, we can obtain T_s by

$$T_s = T_s^{o} + \frac{1}{p_\tau} \sum_{m=0}^{M} \sum_{k=1}^{B} \left(b_{m,0,k} \cdot \frac{kL}{R}\right) \tag{18}$$

where T_s^o denotes the overhead of a frame transmission, which consists of interframe spacings, the time to transmit physical and MAC layer headers, and the time to transmit control packets.

To calculate $\widetilde{T}_s$, we can use

$$\widetilde{T}_s = \frac{1}{p_\tau} \sum_{m=0}^{M} \sum_{k=1}^{B} \left[b_{m,0,k} \cdot \frac{k\widetilde{L}}{R}\right]. \tag{19}$$

Next, we derive the calculation for T_c. For the request to send/clear to send (RTS/CTS) scheme, T_c is independent of the size of burst frame, and thus can be calculated by the same method in [23]. For the basic access scheme, a comprehensive method has been proposed in our previous work [23], in which the probability mass function of the payload size is required. Let $\widetilde{f}(n)$ be the probability mass function of the length of any burst frame, and let $f_k(n)$ be the probability mass function of the length of a burst frame that consists of k packets. Clearly, $f_k(n)$ can be calculated by using

$$f_k(n) = \underbrace{f(n) \otimes f(n) \otimes \cdots \otimes f(n)}_{k} \tag{20}$$

TABLE I
SETTING OF THE MAC PROTOCOL

minimum contention window size	8
maximum contention window size	256
σ	2 μs
SIFS	1 μs
DIFS	5 μs
long retry limit	4
short retry limit	7
buffer size	50

where $\otimes$ denotes convolution. We can then derive $\widetilde{f}(n)$ as

$$\widetilde{f}(n) = \frac{\sum_{m=0}^{M}\sum_{k=1}^{B} b_{m,0,k} f_k(n)}{\sum_{m=0}^{M}\sum_{k=1}^{B} b_{m,0,k}}. \tag{21}$$

V. SIMULATION AND NUMERICAL RESULTS

In this section, we evaluate the performance of the proposed MAC protocol by simulation and numerical studies. Table I gives the values of the control parameters used in the simulation and numerical analysis.

We implement the proposed MAC protocol on the NS-2 network simulator and conduct simulations under the following setting.

1) All nodes are located in a 4 $\times$ 4-m area.
2) Packet arrivals to any node are modeled by a Poisson process with the same rate λ.
3) The physical-layer transmission rate, denoted by R, is the same for all messages.
4) The synchronization time, denoted by $T_{\rm sync}$, is the same for all messages. In most of our experiments, we let $T_{\rm sync} = 10$ μs, which is similar to the setting in the two proposals for IEEE 802.15.3a.[4]
5) Unless otherwise specified, we assume that the total arrival rate of data (in bits per second or b/s) is equal to R. In other words, we let the traffic load be 1 Erlang.
6) Unless otherwise specified, we assume BER is 0.
7) Unless otherwise specified, we assume that the RTS/CTS scheme is used.

In this paper, we focus on two major performance metrics: throughput and average end-to-end delay. The throughput is defined by the total number of successfully received bits divided by the duration of a simulation run. For a given packet, the end-to-end delay is the duration from the epoch that the packet enters the buffer at the MAC layer to the epoch that the packet is successfully received. Thus, the end-to-end delay may include queueing delay, assembly delay, burst transmission time, propagation delay, and backoff delay. The queueing delay is the duration between the epoch that the packet arrives to the buffer and the epoch that the packet is eligible for assembly. The assembly delay is the duration between the epoch that the packet is eligible for assembly and the epoch that a burst frame is formed. The average end-to-end delay is calculated by the sum of the end-to-end delay of all successfully received packets divided by the total number of successfully received packets.

In the simulations and numerical analysis, we compare the proposed MAC to the benchmark case when $B_{\max} = 1$, i.e., sending each upper layer packet individually, which is a typical situation in existing MAC protocols.

Fig. 2 shows the throughput and average end-to-end delay versus $B_{\max}$ for both the basic and the RTS/CTS scheme, with $T_{\rm sync} = 10$ μs, $R = 50$ Mb/s, $N = 10$, and packet size is fixed to 1000 B. In the figure, "Ana" and "Sim" denote analytical results and simulation results, respectively; the same notations will be used in other figures. The analytical results of the saturation throughput is calculated by $S \times R$, where S is computed by (17). We first observe that the saturation throughput (achieved from the analytical model) increases with $B_{\max}$ for both the basic and the RTS/CTS scheme. Specifically, if the basic scheme is used, the saturation throughput of $B_{\max} = 10$ is about 6 Mb/s larger than that for $B_{\max} = 1$; if the RTS/CTS scheme is used, the saturation throughput of $B_{\max} = 10$ is about 15 Mb/s larger than that for $B_{\max} = 1$. In addition, Fig. 2(a) also shows that the RTS/CTS scheme outperforms the basic scheme in terms of the saturation performance.

For the simulation, we compare the performance of two cases: 1) $B_{\min} = 1$ and 2) $B_{\min} = B_{\max}$, when the load is 1 Erlang. We can see that, the throughput of the second case always performs better than the first case. In addition, the throughput of the second case can be accurately predicted by the analysis [see Fig. 2(a)].

From Fig. 2(b), we can observe that the proposed MAC scheme can also significantly reduce the average end-to-end delay, compared to the benchmark case when $B_{\max} = 1$. For the RTS/CTS scheme, setting $B_{\min} = B_{\max} = 10$ can decrease the delay by more than 70 ms, compared to the case when $B_{\max} = 1$.

Fig. 3 shows the throughput versus $B_{\max}$ with different BER and packet size distribution conditions where we let $T_{\rm sync} = 10$ μs, $B_{\min} = B_{\max}$, and $N = 10$. In Fig. 3(a), we compare the proposed retransmission schemes, denoted as packet-based retransmission (PR), and a previous scheme [23], denoted as frame-based retransmission (FR), where all the payload in a frame will be retransmitted if a transmission is failed. Clearly, if the BER is 0, the two retransmission schemes will achieve the same throughput performance. However, when the BER is 10^{-5}, then the performance of the two schemes are quite different. Specifically, when FR is used, the maximum throughput (about 60 Mb/s) can be achieved if $B_{\max}$ is 3. And the throughput will decrease with the increase of $B_{\max}$ if $B_{\max} > 3$. On the other hand, when PR is used, we can see that the throughput can still increase with the increase of $B_{\max}$. Moreover, we can see that the overall throughput for $B_{\max} = 20$ is about 86 Mb/s, which is significantly larger than the maximum throughput of FR. These results indicate that the proposed retransmission scheme can significantly improve the throughput performance in error-prone wireless channels.

In Fig. 3(b), we study the impact of packet size distribution on the throughput performance, where we let BER be 10^{-5} and let $R = 200$ Mb/s. Specifically, we tested four distributions: 1) packet size is fixed to 1000 B (denoted as "Fixed"); 2) packet

[4]The standard preamble duration in [4] is 9.375 μs, and the preamble duration in [3] is 9 μs plus an unspecified duration for training sequence.

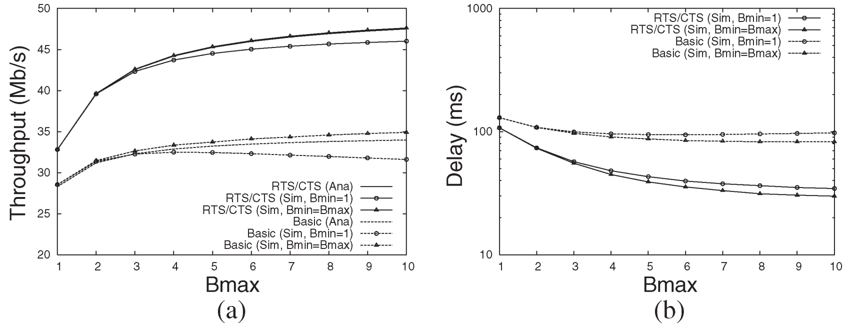


Fig. 2. Performance versus $B_{\max}$ with different access schemes ($T_{\rm sync} = 10\ \mu$s, $R = 50$ Mb/s, $N = 10$). (a) Throughput. (b) End-to-end delay.

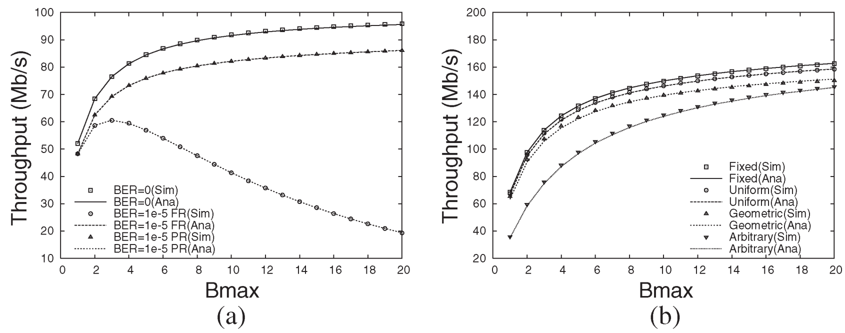


Fig. 3. Performance versus $B_{\max}$ with different BER and packet size distribution conditions ($T_{\rm sync} = 10\ \mu$s, $N = 10$). (a) Impact of BER($R = 100$ Mb/s). (b) Impact of packet size distribution (BER $= 10^{-5}$, $R = 100$ Mb/s).

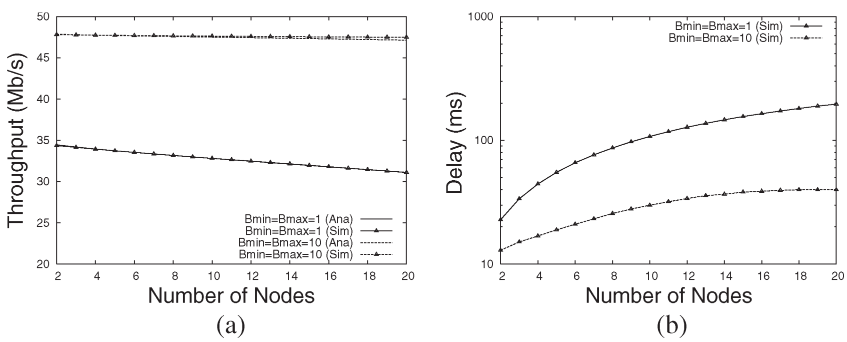
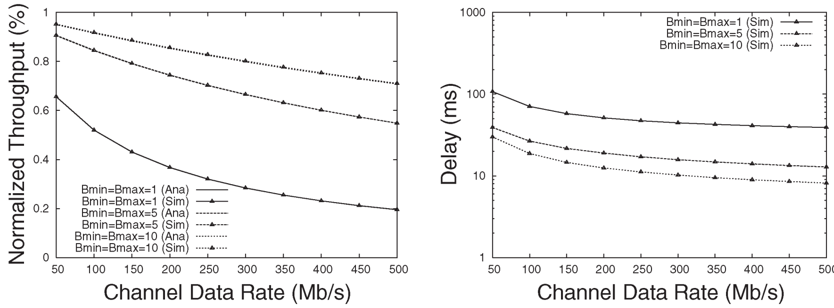


Fig. 4. Performance versus number of nodes ($T_{\rm sync} = 10\ \mu$s, $R = 50$ Mb/s). (a) Saturation throughput. (b) End-to-end delay.

size is uniform from 1 to 1999 B (denoted as "Uniform"); 3) packet size follows a geometric distribution with average packet size be 1000 B (denoted as "Geometric"); and 4) packet size follows an "arbitrary" distribution (50% packets are 44 B, 20% packets are 552 B, 20% packets are 576 B, 10% packets are 1500 B), which is a simplified model based on the Internet traffic characteristics [24]. From Fig. 3(b), we can observe that, the proposed analytical model is highly accurate for different packet size distribution, and the proposed protocol can dramatically improve the throughput performance. In addition, we can also observe that the performance are rather similar for distribution 1), 2), and 3), which have the same average packet size as 1000 B. For this reason, we will fix the packet size to 1000 B in the rest studies.

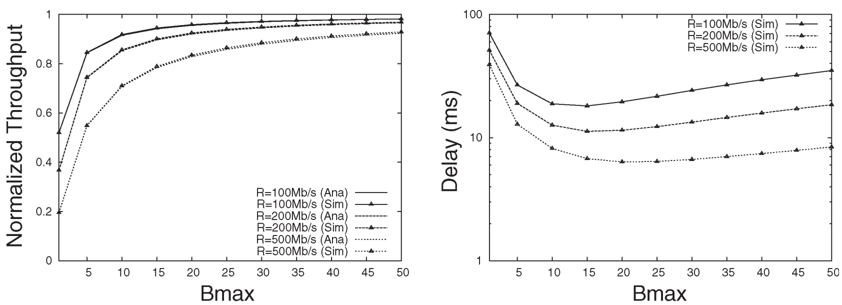
Fig. 4 shows the saturation throughput and average end-to-end delay versus the number of nodes N when $T_{\rm sync} = 10\ \mu$s and $R = 50$ Mb/s. From Fig. 4(a), we can observe that the saturation throughput for $B_{\min} = B_{\max} = 10$ is significantly higher than that for $B_{\max} = 1$ with different N. We can also observe that the saturation throughput decreases with the increase of the number of nodes. It is important to note that the saturation throughput of $B_{\min} = B_{\max} = 10$ is less sensitive to the increase of N than that of the benchmark case. Specifically, if the number of node increases from 2 to 20, the decrease of the saturation throughput is 3.3 Mb/s for the benchmark case. In comparison, if $B_{\min} = B_{\max} = 10$, the decrease of the saturation throughput is only about 0.7 and 0.3 Mb/s for the saturated analysis and simulation, respectively. Like the previous results, the saturation throughput can also be accurately predicted by the analysis [see Fig. 4(a)].

In Fig. 4(b), we see that the average end-to-end delay increases with respect to the total number of nodes N. Fig. 4(b) also shows that, the average end-to-end delay for $B_{\min} = B_{\max} = 10$ is much smaller than that of $B_{\min} = B_{\max} = 1$, which implies that the assembly delay is not an major component of the end-to-end delay in the saturated condition.



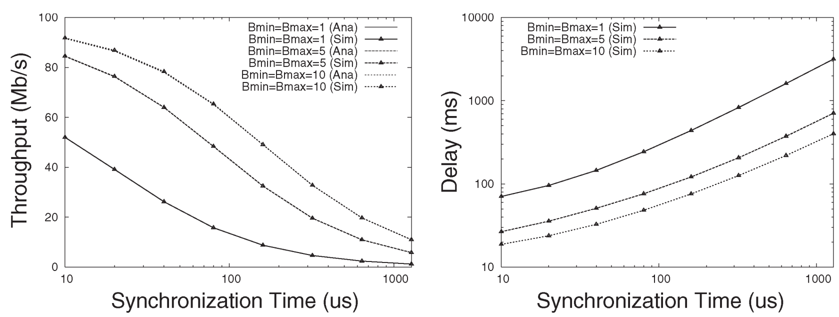
(a) (b)

Fig. 5. Performance versus channel data rate ($T_{\mathrm{sync}} = 10\ \mu$s, $N = 10$). (a) Normalized throughput. (b) End-to-end delay.



(a) (b)

Fig. 6. Performance versus $B_{\max}$ with different R ($T_{\mathrm{sync}} = 10\ \mu$s, $N = 10$). (a) Normalized throughput. (b) End-to-end delay.



(a) (b)

Fig. 7. Performance versus T_{sync} ($N = 10, R = 100$ Mb/s). (a) Throughput. (b) End-to-end delay.

Fig. 5 shows the performance versus channel date R for $T_{\mathrm{sync}} = 10\ \mu$s and $N = 10$. From Fig. 5(a), we observe that the simulation results for throughput exactly match the throughput given by the saturated analysis. We can also see that the normalized throughput S decreases with the increase of R (even though the actual throughput $S \times R$ still increases). In addition, we find that the reduction of throughput is less significant if $B_{\max}$ is larger. Specifically, S decreases by about 45% if $B_{\max} = 1$. In contrast, S only shrinks by about 22% if $B_{\max} = 10$.

From Fig. 5(b), we also observe that the average delay decreases with the increase of R. For example, if $B_{\max} = B_{\min} = 10$, the average delay for $R = 50$ Mb/s is about 30 ms, while the average delay for $R = 500$ Mb/s is about 8 ms. On the other hand, we note that the average delay can be reduced if we allow more packets to be assembled into a burst frame.

In the previous discussions, we observe that increasing $B_{\max}$ can significantly improve the throughput performance. However, we also notice that the increase of $B_{\max}$ will also increase the packet assembly delay and the burst transmission time, which may degrade the end-to-end delay performance. To better understand such phenomena, Fig. 6 shows the delay and throughput versus $B_{\max}$ with different R, where we let $B_{\min} = B_{\max}$. We can observe that the throughput increases significantly if $B_{\max}$ changes from 1 to 10. However, the improvement is slight if $B_{\max}$ increase from 40 to 50. On the other hand, from Fig. 6(b), we can see an optimum $B_{\max}$, denoted as $B'_{\max}$, which will lead to the minimal end-to-end delay. Moreover, the optimum $B'_{\max}$ increases with R. For instance, the value of $B'_{\max}$ is 15 for $R = 100$ and 200 Mb/s; while it is 20 for $R = 500$ Mb/s. Since the throughput always increases with $B_{\max}$, we can see that both the throughput and the delay performance can be improved if $B_{\max}$ increases from 1 to $B'_{\max}$; but, there is a tradeoff between the throughput and delay performance if $B_{\max}$ is larger than $B'_{\max}$.

In Fig. 7, we evaluate the impact of the synchronization time T_{sync}. We can see that, the throughput is significantly reduced with the increase of T_{sync}. For example, if $T_{\mathrm{sync}} = 80\ \mu$s, the throughput is only 15 Mb/s for $B_{\max} = 1$, which is significantly lower than the transmission rate $R = 100$ Mb/s. In contract, setting $B_{\min} = B_{\max} = 10$ can still yield about 65 Mb/s throughput in the same condition.

In summary, the simulation and numerical results demonstrate that, by setting an appropriate $B_{\max}$ $(B_{\max} > 1)$, the proposed MAC protocol can significantly outperform the benchmark case where $B_{\max} = 1$, in terms of both throughput and end-to-end delay. Moreover, our analytical results match the simulation results well, indicating the accuracy of our analysis.

VI. Conclusion

In this paper, we proposed and analyzed the performance of an aggregation-based MAC protocol for high data rate UWB *ad hoc* networks. The protocol has two major features: 1) a packet aggregation scheme at the MAC layer and 2) a novel frame retransmission scheme, in which only the packets that encounter transmission errors will be retransmitted. Consequently, the performance of UWB *ad hoc* networks can be significantly improved even in an error-prone wireless environment. To evaluate the performance of the protocol, we develop a three-dimensional Markov chain model for the saturation throughput performance. In addition, we also evaluate the end-to-end delay performance through simulation. Extensive simulation and analysis results show that, compared to sending each upper layer packet individually, which is typical in existing MAC protocols, the proposed scheme can significantly improve both that throughput and the delay performance of a UWB *ad hoc* network under different data rates and different BER conditions.

References

[1] K. Lu, D. Wu, and Y. Fang, "A novel framework for medium access control in ultra-wideband ad hoc networks," *Dyn. Contin. Discrete Impuls. Syst., Ser. B*, vol. 12, no. 3, pp. 427–441, Jun. 2005.

[2] *First Report and Order in the Matter of Revision of Part 15 of the Commission's Rules Regarding Ultra-Wideband Transmission Systems*, Apr. 2002. ET Docket 98-153. Federal Communications Commission (FCC 02-48), Std.

[3] IEEE P802.15-04/0137r3, *DS-UWB Physical Layer Submission to 802.15 Task Group 3a*, Jul. 2004. Project: IEEE P802.15 Working Group for Wireless Personal Area Networks (WPANs).

[4] IEEE P802.15-04/268r3, *Multi-Band OFDM Physical Layer Submission to 802.15 Task Group 3a*, Mar. 2004. Project: IEEE P802.15 Working Group for Wireless Personal Area Networks (WPANs).

[5] S. Roy, J. R. Foerster, V. S. Somayazulu, and D. G. Leeper, "Ultra-wideband radio design: The promise of high-speed, short-range wireless connectivity," *Proc. IEEE*, vol. 92, no. 2, pp. 295–311, Feb. 2004.

[6] W. Zhuang, X. Shen, and Q. Bi, "Ultra-wideband wireless communications," *Wirel. Commun. Mob. Comp.—Special Issue Ultra-Broadband Wireless Communications for the Future*, vol. 3, no. 6, pp. 663–685, Dec. 2003.

[7] R. L. Peterson, R. E. Ziemer, and D. E. Borth, *Introduction to Spread Spectrum Communications*. Englewood Cliffs, NJ: Prentice-Hall, 1995.

[8] Y. Ma, F. Chin, B. Kannan, and S. Pasupathy, "Acquisition performance of an ultra wide-band communications system over a multiple-access fading channel," in *Proc. IEEE Conf. Ultra Wideband Syst. and Technol.*, 2002, pp. 99–103.

[9] S. Aedudodla, S. Vijayakumaran, and T. F. Wong, "Rapid ultra-wideband signal acquisition," in *Proc. IEEE WCNC*, Mar. 2004, pp. 1148–1153.

[10] G. Bianchi, "Performance analysis of the IEEE 802.11 distributed coordination function," *IEEE J. Sel. Areas Commun.*, vol. 18, no. 3, pp. 535–547, Mar. 2000.

[11] H. Wu, Y. Peng, K. Long, S. Cheng, and J. Ma, "Performance of reliable transport protocol over IEEE 802.11 wireless LAN: Analysis and enhancement," in *Proc. IEEE INFOCOM*, Jun. 2002, pp. 599–607.

[12] Z. Hadzi-Velkov and B. Spasenovski, "Saturation throughput—Delay analysis of IEEE 802.11 DCF in fading channel," in *Proc. IEEE ICC*, May 2003, pp. 121–126.

[13] S. S. Kolenchery, J. K. Townsend, and J. A. Freebersyser, "A novel impulse radio network for tactical military wireless communications," in *Proc. IEEE MILCOM*, 1998, pp. 59–65.

[14] F. Cuomo, C. Martello, A. Baiocchi, and F. Capriotti, "Radio resource sharing for *ad hoc* networking with UWB," *IEEE J. Sel. Areas Commun.*, vol. 20, no. 9, pp. 1722–1732, Dec. 2002.

[15] J. Ding, L. Zhao, S. Medidi, and K. Sivalingam, "MAC protocols for ultra wideband (UWB) wireless networks: Impact of channel acquisition time," in *Proc. SPIE ITCOM*, Jul. 2002.

[16] B. Radunovic and J. Y. Le Boudec, "Optimal power control, scheduling and routing in UWB networks," *IEEE J. Sel. Areas Commun.*, vol. 22, no. 7, pp. 1252–1270, Sep. 2004.

[17] J. Y. Le Boudec, R. Merz, and J. W. B. Radunovic, "A MAC protocol for UWB very low power mobile *ad-hoc* networks based on dynamic channel coding with interference mitigation," EPFL Tech. Rep. ID: IC/2004/02.

[18] Y. Xiao, "MAC performance analysis and enhancement over 100 Mbps data rates for IEEE 802.11," in *Proc. IEEE VTC—Fall*, Oct. 2003, pp. 1869–1873.

[19] J. Zhao, Z. Guo, Q. Zhang, and W. Zhu, "Performance study of MAC for service differentiation in IEEE 802.11," in *Proc. IEEE GLOBECOM*, pp. 778–782.

[20] J. W. Robinson and T. S. Randhawa, "Saturation throughput analysis of IEEE 802.11e enhanced distributed coordination function," *IEEE J. Sel. Areas Commun.*, vol. 22, no. 5, pp. 917–928, Jun. 2004.

[21] *Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications*, 1999. ISO/IEC 8802-11:1999(E). ANSI/IEEE Std. 802.11.

[22] *Wireless Medium Access Control (MAC) and Physical Layer (PHY) Specifications for High Rate Wireless Personal Area Networks (WPANs)*, 2003. IEEE Std. 802.15.3.

[23] K. Lu, D. Wu, Y. Fang, and R. C. Qiu, "On medium access control for high data rate ultra-wideband ad hoc networks," in *Proc. IEEE WCNC*, New Orleans, LA, Mar. 2005, vol. 2, pp. 795–800.

[24] K. Thompson, G. J. Miller, and R. Wilder, "Wide-area internet traffic patterns and characteristics," *IEEE Netw.*, vol. 11, no. 6, pp. 10–23, Nov./Dec. 1997.

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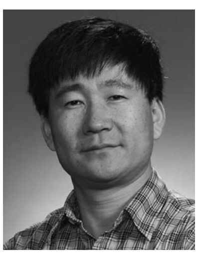

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