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A Novel Framework for Medium Access Control in Ultra-Wideband Ad Hoc Networks

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Abstract. Ultra-wideband (UWB) communication is becoming an important technology for future Wireless Personal Area Networks (WPANs). A critical challenge in UWB system design is that a receiver usually needs tens of micro-seconds or even tens of milliseconds to synchronize with transmitted signals, known as timing acquisition problem. Such a long synchronization time will cause significant overhead, since the data rate of UWB systems is expected to be very high. In this paper, we address the timing acquisition problem at the medium access control (MAC) layer, and propose a general framework for medium access control in UWB systems; in this framework, a transmitting node can aggregate multiple upper-layer packets into a larger burst frame at the MAC layer. Furthermore, we design an MAC protocol based on the framework. Compared to sending each upper-layer packet individually, which is a typical situation in exiting MAC protocols, the proposed MAC can drastically reduce the synchronization overhead. Simulation results show that the proposed MAC can significantly improve the performance of UWB networks, in terms of both throughput and end-to-end delay.

Keywords. ultra-wideband (UWB), MAC, timing acquisition, throughput, delay

1 Introduction

According to Federal Communications Commission (FCC), an *Ultra-wideband* (UWB) system is defined as any radio system that has a 10-dB bandwidth larger than 20 percent of its center frequency, or has a 10-dB bandwidth equal to or larger than 500 MHz [1]. To enable the deployment of UWB systems, FCC allocated an unlicensed frequency band $3.1 - 10.6$ GHz for indoor or hand-held UWB communication systems [1].

In the past few years, UWB communication has received considerable attention in both academia and industry. Compared to traditional narrow band systems, UWB can provide high data rate (> 100 Mb/s) with very low-power emission (less than -41 dBm/MHz) in a short range. In addition, UWB can also support multiple access. These features make UWB particularly suitable for *wireless personal area network* (WPAN) applications. Currently, IEEE

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802.15.3 working group is studying the use of UWB as an alternative physical layer technique. Several implementation schemes have been proposed recently [2, 3], and extensive research has been conducted.

Despite the salient features, to successfully implement a UWB system, a number of challenges must be addressed [10, 15]. One of the critical issues is timing acquisition [8–10], which is a process of synchronizing the receiver's clock with the transmitter's clock so that the receiver can determine the boundary between two transmitted symbols. In practice, timing acquisition is usually performed by sending a preamble before information bits [2, 3]. Depending on the receiver design, the duration of a preamble varies from tens of micro-seconds to tens of milliseconds [4]. Evidently, for high data rate applications, the overhead of preambles will significantly reduce the efficiency of UWB networks [10].

Existing works that consider the timing synchronization issue in UWB MAC layer design include [5, 6]. In [6], the authors assumed that the UWB network can provide multiple channels through different time-hopping (TH) codes. One of the TH codes is used for a control channel while all the rest are used for data channels; in addition, carrier-sensing multiple access/collision avoidance (CSMA/CA) based MAC protocols were employed to resolve collision in the control channel. Specifically, signaling packets such as request-to-send (RTS) and clear-to-send (CTS), which are used in CSMA/CA, are transmitted through the control channel, while data packets of a connection are transmitted through a data channel (specified by a unique TH code). To reduce the timing synchronization overhead, [6] 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 achieves good performance in the simulation, there are still some critical issues unclear in [6]. For instance, it is not clear how a node selects a TH code for its data channel in a distributed manner so as to avoid using the same TH code as that used by another node. Another potential problem is that the link maintenance scheme will increase the transmission time of the transmitter, thereby reducing the battery life and introducing extra interference.

In [5], the authors studied the impact of long synchronization time on the performance of CSMA/CA and TDMA schemes used in UWB networks. However, the paper did not provide solutions to mitigate the timing acquisition problem.

In this paper, we propose a framework for UWB medium access control to mitigate the timing acquisition problem. 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, under which each upper-layer packet is delivered individually, transmitting multiple upper-layer packets in one frame will significantly reduce the synchronization overhead.

Our framework consists of five major components. The first component is a packet classification policy that determines how to classify incoming upper-

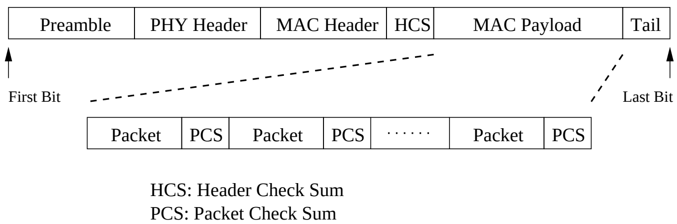


Figure 1: An example of the burst frame.

layer packets according to their destination and quality of service (QoS) requirements. The second component is a buffer management policy that provides QoS and/or fairness among different flows. The third component is a packet assembly policy that dictates how to assemble packets into a burst frame, which should take into account synchronization overhead, physical layer constraints, QoS, and fairness among different nodes. The fourth component is an acknowledgement policy that specifies the acknowledgement procedure at the receiver side. The last component is a packet error control policy, which describes the method to mitigate packet errors.

Furthermore, we design an MAC protocol based on the proposed framework. Extensive simulation results show that, compared to sending each upper-layer packet individually, which is a typical situation in exiting MAC protocols, the proposed scheme can significantly improve throughput and delay performance of a UWB network under different data rates, various synchronization time, and different bit error rates.

The rest of the paper is organized as follows. In Section 2, we first present a general framework for medium access control in UWB networks, and then design an MAC protocol based on the framework. Simulation results are shown in Section 3. Finally, Section 4 concludes the paper.

2 A Framework for UWB MAC

In this section, we present a novel framework for medium access control in UWB networks. The key idea of this framework is that a transmitting node can aggregate multiple upper-layer packets into a burst at the MAC layer and transmit the burst to a destination node, instead of transmitting each upper-layer packet individually. In this manner, the overhead of timing synchronization is reduced. In this paper, we focus on the MAC design for single-hop, single channel UWB networks, and leave multi-hop and/or multi-channel UWB networking for future study.

Our research is motivated by the limitation that a UWB receiver needs a long synchronization time. Since the data rate of a UWB channel is expected to be rather high (usually in the order of 100 Mb/s), the overhead of timing synchronization can be significantly large. For example, suppose the data rate of a UWB channel is 100 Mb/s; then 10 μs synchronization time will incur an overhead of 1000 bits, which is much larger than the length of most control packets in existing MAC protocols (*e.g.*, 160-bit RTS and 112-bit CTS at the MAC layer).

The rest of this section is organized as follows. Section 2.1 presents our general framework for UWB MAC while Section 2.2 describes an MAC protocol based on the framework.

2.1 A General Framework for UWB MAC

In this section, we describe a general framework for the UWB MAC design. Our framework inherits most features of the IEEE 802.11 MAC protocol, which has been widely deployed. In addition to a data frame and control frames such as RTS, CTS, and ACK (acknowledgement), we also define a new type of frame, called *burst frame*. Unlike the existing data frame (in IEEE 802.11 MAC) that contains only one upper-layer packet, a burst frame may consist of multiple upper-layer packets, and is transmitted as one unit. An example of the burst frame structure is illustrated in Fig. 1. Next, we present our framework, which consists of five major components as below.

1. **Packet classification policy**: Incoming upper-layer packets should first be classified and put into different queues so as to achieve the target QoS (see Fig. 2). Hence, a policy that specifies the packet classification method is needed.

2. **Buffer management policy**: Since different queues share the same physical memory/buffer (see Fig. 2), a buffer management policy is required to manage the buffer in a way that achieves QoS requirements and/or fairness among different flows.

3. **Packet assembly policy**: Since multiple upper-layer packets can be assembled into a burst frame, we need a policy that describes the packet assembly process; the policy should take into account synchronization overhead, physical layer constraints, QoS, and fairness among different nodes. Note that a packet assembler also serves as a scheduler for multiple queues in the buffer (see Fig. 2).

4. **Acknowledgement policy**: When a receiver successfully receives a burst frame, an acknowledgement should be sent to the transmitter. So we need an acknowledgement policy to specify the acknowledgement procedure at the receiver side.

5. **Packet error control policy**: Since packet errors are unavoidable, error control must be in place. Hence, a policy that dictates what error control scheme to be used, is needed.

In the rest of this section, we elaborate on the above issues.

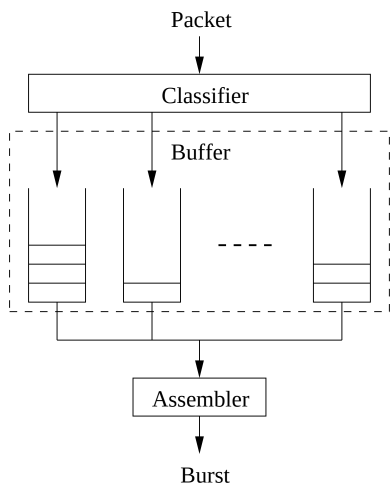


Figure 2: Architecture of the MAC layer at the transmitter.

2.1.1 Packet Classification Policy

A packet classification policy specifies whether the packet classification of incoming upper-layer packets is based on the destinations and/or the QoS classes of the packets. If the packet classification is based on both the destination and the QoS class, then incoming upper-layer packets can be classified and put into $N \times C$ queues (see Fig. 2), where N is the total number of destinations including possible broadcast and multicast addresses, and C is the total number of QoS classes.

2.1.2 Buffer Management Policy

To provide QoS, each queue may maintain its unique control parameters such as:

- The maximum number of packets in the queue;
- The maximum value of the total length of all packets in the queue;
- The arrival time and expected departure deadline of each packet.

With the above parameters, we can apply a buffer management policy such as push-out, tail-dropping, complete sharing, complete partitioning, partial sharing of the total buffer among all the queues [7, 12, 13] so as to achieve QoS requirements and/or fairness among different flows.

2.1.3 Packet Assembly Policy

A packet assembly policy specifies how packets are assembled and how to schedule the packets in different queues (see Fig. 2) while achieving the QoS requirements and/or fairness among flows destined to different destinations, which poses a major challenge in the UWB MAC design. In general, an assembly policy can be determined based on one or several of the following criteria:

- The maximum and minimum size of a burst frame

  Given the maximum and minimum size of a burst frame (in bits), denoted by L_{max} and L_{min}, respectively, an assembly policy can determine when and how to assemble a burst. For example, when the length of a queue (in bits) is greater than or equal to L_{min}, a burst, the size of which is not greater than L_{max}, will be assembled.

  Note that the larger size of a burst frame, the smaller the synchronization overhead (relative to the burst frame). However, a larger L_{min} value will also lead to a longer packet assembly delay, which may affect delay sensitive services. On the other hand, the size of a burst frame may be limited by constraints such as the transmission/reception buffer size. Hence, in determining the value of L_{max} and L_{min}, factors such as synchronization overhead and assembly delay, need to be considered.

- The maximum and minimum number of packets in a burst

  In addition to the maximum and minimum size of a burst frame, the maximum and minimum number of packets in a burst frame, denoted as B_{max} and B_{min}, respectively, can also be specified. Depending on the acknowledgement policy, the value of B_{max} may affect the length of an acknowledgement packet, which will be discussed later in Section 2.1.4.

- Delay constraints of packets

  An assembly policy can be based on the delay constraints of packets in the assembly queues. In addition, burst assembly can be triggered by a combination of the delay constraints of packets in assembly queues, the total number and size of packets in assembly queues.

- Destinations of packets

  An assembly policy can be based on the destinations of packets to be assembled. Obviously, packets that have the same destination can be assembled into a burst. In addition, when a transmitter is sending data with an omni-directional antenna, all neighbor nodes within the transmission range are able to receive the transmission; hence, by utilizing this feature, we can assemble packets that have different destination addresses, which further reduces the synchronization overhead. Nevertheless, such an approach may cause some other issues. For example, the transmitter may have to wait a longer time for the acknowledgements from all possible destinations.

- Prediction of packet arrivals to the MAC layer

  An assembly policy can be based on the prediction of packet arrivals to the MAC layer. For example, if RTS/CTS are employed for collision avoidance, then it is possible that during the RTS/CTS process, a number of new packets arrive and these newly arrived packets may be added to the current burst, so that the synchronization overhead is further reduced. If packet arrivals to the MAC layer can be predicted, the transmitter can over-reserve the resource (*e.g.*, setting a larger value for the 'duration' field in the RTS frame than the current burst size), so that some new packets can be added to the current burst when CTS is received. In this way, both the end-to-end delay of packets and the synchronization overhead can be reduced. Obviously, the bandwidth efficiency will be lower if the over-reserved resource is not used due to less packet arrivals than expected.

Furthermore, an assembly policy may specify what scheduling discipline to be used to achieve the QoS requirements and fairness among flows destined to different destinations. The possible scheduling disciplines include round robin, earliest deadline first (EDF), and weighted fair queueing. For example, if the round robin is used, the round robin scheduler first selects a queue and then assemble packets from the queue into a burst; for the next burst, the same procedure is repeated.

2.1.4 Acknowledgement Policy

An acknowledgement policy specifies the acknowledgement procedure at the receiver side. Depending on the packet assembly policy and the QoS requirement of packets, the following policies can be selected.

- Acknowledgement for a burst frame with multiple destinations

  If all packets in a burst frame have the same destination, then the acknowledgement procedure can be the same as that of IEEE 802.11. However, if packets in a burst frame have different destinations, then the MAC must be re-designed. Particularly, in the RTS/CTS scheme, the RTS frame must contain the total number of destinations to be reached and a list of identifiers of these destinations, the order of which determines the priority of a destination node in sending a CTS. Similarly, the burst frame must also contain the same information to dictate how receivers send the ACK frames. Fig. 3 illustrate an acknowledgement policy for a burst that has multiple destinations.

- Acknowledgement for packets in a burst

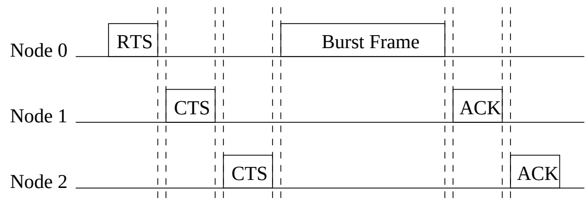


Figure 3: An example of a burst that has multiple destinations, each of which has a different priority to send a CTS.

If all the packets in a burst are destined to the same receiver, only one acknowledgement for the whole burst is required. However, the drawback of this scheme is that, a single bit error of the burst may cause retransmission of all the packets in the burst. To reduce the potential overhead of retransmission, it may be desirable for the receiver to indicate the delivery status of each packet in a burst. To achieve this, the following MAC can be used.

1. A check sum must be created for each packet in the burst to determine which packet contains a bit error. An example of the frame structure can be found in Fig. 1, where each packet has a packet check sum (PCS).
2. To perform per-packet acknowledgement, the format of a burst and an ACK frame must be redefined. Specifically, if *go-back-N* Automatic Repeat Request (ARQ) [11] is used, then the sequence number of the last successful received packet is contained in the ACK frame. If the selected ARQ is used, then the sequence numbers of all erroneous packets in the burst must be identified.

- Pipelining acknowledgement

  In IEEE 802.11, a receiver acknowledges any unicast packet that it has successfully received. In the literature, such scheme is known as *stop-and-wait* protocol since the transmitter will wait for an acknowledgement for the previous transmitted packet before it can send the next packet [11]. To reduce the overhead of timing synchronization, we can apply the well-known *pipelining acknowledgement* scheme [14], which has been widely used in communications networks to improve the transmission efficiency in situations where the round-trip transmission time is large. In this scheme, instead of sending one ACK frame for each burst frame, the receiver may defer the ACK till receiving a certain number of burst frames. In this manner, the overhead of synchronization time can be reduced. Nevertheless, this approach may increase the buffer usage at the transmitter since the transmitter must buffer all unacknowledged packets.

- Piggyback acknowledgement

  In the literature, piggyback acknowledgement is also widely applied to reduce control overhead. The main approach of piggybacking is to transmit acknowledgement information in a data frame, instead of generating an ACK frame. Assuming an omni-directional antenna is used, the acknowledgement can ride on any outgoing packet.

2.1.5 Packet Error Control Policy

To mitigate packet errors, the following options could be chosen.

- Varying spreading factor

  From the physical layer perspective, a UWB system can vary the spreading factor in the DS code [2] to adapt to the channel condition. Specifically, when the packet error ratio is high, a transmitter can use a larger spreading factor to increase SNR; when the packet error ratio is low, a transmitter can use a smaller spreading factor to decrease SNR. Note that varying the spreading factor changes the transmission data-rate [2].

- Channel coding
- ARQ
- hybrid ARQ and channel coding

2.2 An MAC Protocol for UWB Networks

To evaluate the performance of the aforementioned framework, we design a simple MAC protocol based on the framework.

1. Packet classification policy

   We consider only one class of traffic for each destination. Suppose there are N nodes in a UWB network; then we can implement N packet queues in each node, among which $N-1$ queues are dedicated to other nodes and one queue is use to buffer broadcast packets.

2. Buffer management policy

   We use tail-dropping when there is a buffer overflow.

3. Packet assembly policy

   We require that all the packets in a burst frame have the same destination. Then the implementation can be simplified since we can keep all the frame structures and most of the control procedure in IEEE 802.11.

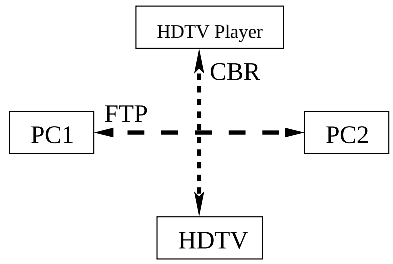


Figure 4: A simulation scenario.

We assume that a burst frame will be generated if the channel is idle. In other words, we let the $L_{min} = 0$ and $B_{min} = 1$ so that there is no packet assembly delay. To achieve the fairness among destinations, a simple round-robin scheme will be employed. When a burst assembly is finished, the burst frame will be stored in a buffer and waiting for transmission.

4. Acknowledgement policy

   A receiver will send one ACK frame to the transmitter if a burst frame is correctly received.

5. Packet error control policy

   For simplicity reason, we do not use any error control scheme.

3 Simulation and Numerical Results

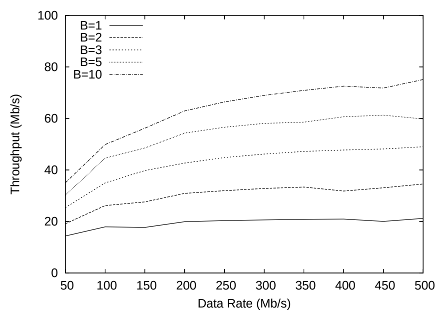
In this section, we evaluate the proposed MAC scheme by extensive simulations. We conduct all experiments under the following setting:

- Two personal computers (PC), one high definition television (HDTV), and one HDTV player are placed within an $4m \times 4m$ area, as shown in Fig. 4.
- A constant bit rate (CBR) video stream is established from the HDTV player to the HDTV set. We further assume that packets of the video stream arrive isochronously with a constant packet inter-arrival time of $0.5ms$ and that each packet consists of 1250 bytes.
- A large file is transferred from PC1 and PC2 with standard FTP and TCP configuration.

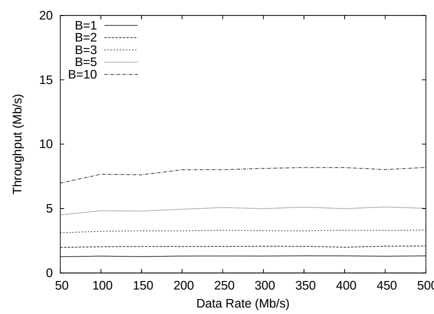
We implement the proposed MAC protocol on the NS-2 network simulator. Except the *data rate* and the *length of the preamble*, all configurations

Table 1: Simulation setting

Minimum contention window	31
Maximum contention window	1023
Slot time unit	20 μs
SIFS	10 μs
RTS threshold	0
Long retry limit	4



(a) $T = 10\mu s$



(b) $T = 1ms$

Figure 5: Throughput vs. data rate.

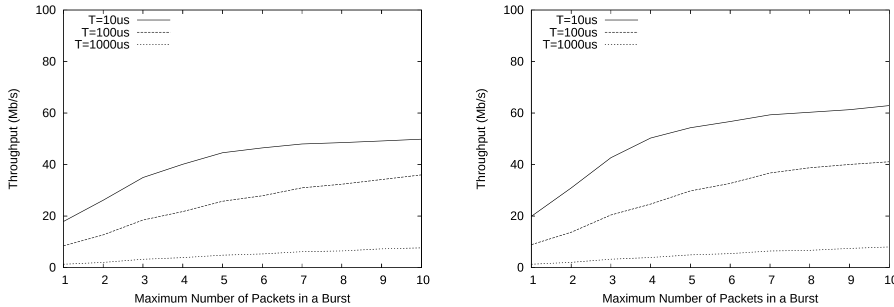
are identical to the default setting for IEEE 802.11 in NS2, which is given by Table 1. Note that in this setting, the RTS/CTS access scheme will always be used since the *RTS threshold* is 0.

To simplify the discussion, we define the following parameters:

- R: physical-layer transmission rate (in bits/s);
- T: synchronization time ($R \times T$ is the length of the synchronization preamble in a packet);
- B: the maximum number of packets in a burst (i.e. B_{max}).

In the simulations, we compare the proposed MAC to the benchmark case where $B = 1$, i.e., sending each upper-layer packet individually, which is a typical situation in exiting MAC protocols.

Fig. 5(a) and (b) show the total throughput for the two connections vs. data rate R for different values of B under $T = 10\mu s$ and $T = 1ms$, respectively. The total throughput is defined by the total number of successfully received bits (for both the CBR and the FTP connection) divided by the duration of a simulation run. We choose $T = 10\mu s$ as a typical value used by IEEE 802.15.3a [2, 3] and choose $T = 1ms$ as a typical value achievable by the current technology [4]. From Fig. 5(a), we observe that, compared to the benchmark case where $B = 1$, the proposed MAC scheme can significantly



(a) $R = 100$ Mb/s (b) $R = 200$ Mb/s

Figure 6: Throughput vs. the maximum number of packets in a burst.

improve the throughput, and the improvement is more significant with the increase of the data rate R. Specifically, if $R = 50$ Mb/s, setting $B = 10$ will double the total throughput, as compared to the case where $B = 1$; if $R = 500$ Mb/s, the throughput for $B = 10$ is 3.5 times of that for $B = 1$.

In addition, from Fig. 5(a) and (b), it can be observed that the throughput for $T = 1ms$ is much less than that for $T = 10\mu s$. This is due to a much larger synchronization overhead involved in the case where $T = 1ms$. Since we apply the RTS/CTS scheme in our simulation, a successful data packet transmission requires four packet exchanges, *i.e.*, RTS, CTS, DATA, and ACK. Therefore, for $T = 1ms$, the synchronization overhead is $4ms$ per successfully-received data packet. Consequently, we can send at most 250 packets or bursts per second. Thus the throughput is significantly limited when the synchronization time is long.

Another important observation from Fig. 5 is that, if we fix B, then the throughput increases with data rate R only when R is small. If R is greater than a certain threshold, the throughput does not increase with R. For example, under $B = 5$ and $T = 10\mu s$, the throughput is about 60 Mb/s for all values of R that are greater than 300 Mb/s. This indicates there exists a saturated throughput.

Fig. 6(a) and (b) show the throughput vs. the maximum number of packets in a burst B under $R = 100$ Mb/s and $R = 200$ Mb/s, respectively. From Fig. 6(a), it can be seen that the throughput increases with B. Particularly, for $T = 100\mu s$ and $T = 1ms$, the throughput increases almost linearly with B ($1 \leq B \leq 10$). However, for $T = 10\mu s$, the throughput increases slightly with B when $B > 5$. A similar trend can be also observed in Fig. 6(b). These results demonstrate that the proposed MAC scheme can significantly improve the throughput, compared to the benchmark case where $B = 1$. Furthermore, Fig. 6 shows that the increase of the synchronization time significantly reduces the throughput.

So far all discussions are based on the assumption that packet losses are only due to collision. In reality, there may be bit errors. In Fig. 7, we

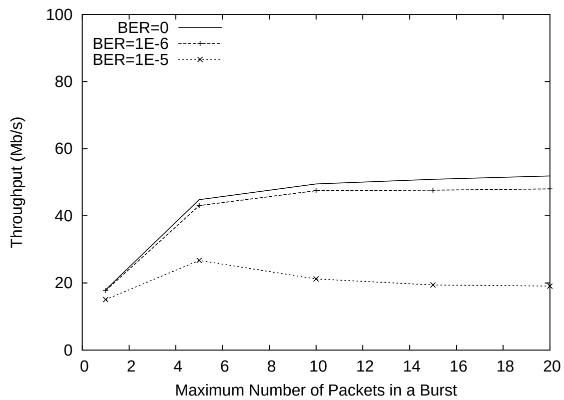


Figure 7: Throughput vs. the maximum number of packets in a burst under different channel bit error rate ($T = 10\mu s, R = 100$ Mb/s).

show the throughput of the proposed MAC scheme under various bit error conditions, and with $R = 100$ Mb/s and $T = 10\mu s$. In this case, we assume that no channel coding is employed at the physical layer and that all packets in a burst will be dropped if any bit error occurs in the burst. We study the throughput performance under three bit error rates, *i.e.*, BER = 0, 10^{-6}, and 10^{-5}. Note that for a packet with a length of 1250 bytes (as set in our simulator), bit error ratio 10^{-6} and 10^{-5} will lead to about 1% and 10% packet loss ratio, respectively. From Fig. 7, it can be observed that the throughput for BER= 10^{-6} is slightly less than the throughput for BER=0. In contrast, when the BER is 10^{-5}, the result is quite different; it can be seen that the throughput increases with B for $B \leq 5$ while the throughput decreases with B for $B > 5$. To mitigate the reduction of throughput in this case, a sophisticated error control scheme may be needed. The results in this set of simulations show that, even if there are bit errors, the proposed MAC scheme can still significantly outperform the benchmark case where $B = 1$, in terms of throughput.

Finally, we investigate the end-to-end packet delay performance of the proposed MAC scheme. 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. In the simulations, we measure the average delay of the CBR traffic for $R = 100$ Mb/s and 200 Mb/s, respectively. As shown in Fig. 8, the average end-to-end delay decreases with the increase of B. This shows that the proposed MAC scheme achieves smaller end-to-end delay than the benchmark case where $B = 1$.

In summary, the simulation results demonstrate that, for $B > 1$, the proposed MAC protocol outperforms the benchmark case where $B = 1$, in terms of both throughput and end-to-end delay.

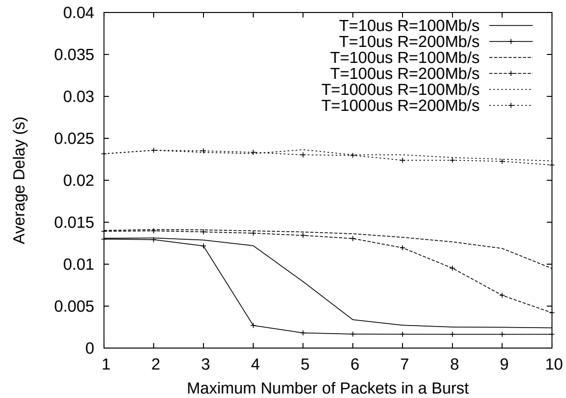


Figure 8: Delay vs. the maximum number of packets in a burst.

4 Conclusions

In this paper, we studied the MAC design issue in high data rate UWB ad hoc networks. Our objective is to mitigate the timing synchronization problem, which causes serious performance degradation in UWB communication systems. To address this problem, we proposed a general framework for medium access control in UWB systems; under this framework, a transmitting node can aggregate multiple upper-layer packets into a larger burst frame at the MAC layer. In addition, we designed an MAC protocol based on the framework. Extensive simulation results show that, compared to sending each upper-layer packet individually, which is a typical situation in exiting MAC protocols, the proposed MAC can significantly improve throughput and delay performance of a UWB network. This is due to the fact that the proposed MAC can drastically reduce the synchronization overhead.

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