# On Medium Access Control for High Data Rate Ultra-Wideband Ad Hoc Networks

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Abstract-A critical challenge in ultra-wideband (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, and analyze its saturation throughput performance. 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. Numerical and 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.

## I. 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 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 [4],

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[5]. One of the critical issues is timing acquisition [4], [6], [7], 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 microseconds to tens of milliseconds [8]. Evidently, for high data rate applications, the overhead of preambles will significantly reduce the efficiency of UWB networks [4].

Existing works that consider the timing synchronization issue in UWB MAC layer design include [9], [10]. In [9], 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. To reduce the timing synchronization overhead, [9] 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 [9]. One of the potential problems is that the link maintenance scheme will increase the transmission time of the transmitter, thereby reducing the battery life and introducing extra interference. In [10], 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* upperlayer packets into *one* burst frame at the MAC layer. In contrast to the traditional approach, under which each upperlayer 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-layer packets according to

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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.

Based on the proposed framework, we design an MAC protocol and analyze its saturation throughput performance. Extensive simulation and analysis 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 and different bit error rates.

The rest of the paper is organized as follows. In Section II, we first present a general framework for medium access control in UWB networks, and then design an MAC protocol based on the framework. In Section III, we analyze the saturation throughput performance of the MAC protocol. Extensive simulation and numerical results will be shown in Section IV. Finally, Section V concludes the paper.

#### II. 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 upperlayer packets into a burst at the MAC layer and transmit the burst to a destination node. 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.

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

## A. A General Framework for UWB MAC

In this section, we describe a general framework for the UWB MAC design. Our framework is based on the CSMA/CA<sup>1</sup> MAC protocol, which is easier to be implemented than TDMA in practice, and which is the basic MAC scheme supported by both IEEE 802.11 and IEEE 802.15.3. In this framework, besides a data frame and control frames such as RTS, CTS, and ACK, 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. Next, we present our framework, which consists of five major components as below. 1) Packet Classification Policy: A packet classification policy specifies how to classify incoming upper-layer 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, where N is the total number of destinations including possible broadcast and multicast addresses, and C is the total number of QoS classes.

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 so as to achieve QoS requirements and/or fairness among different flows.

3) Packet Assembly Policy: A packet assembly policy specifies how packets are assembled and how to schedule the packets in different queues 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 size of a burst frame may be limited by constraints such as the transmission/reception buffer size.
- The maximum and minimum number of packets in a burst In addition to the 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.

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.

4) Acknowledgement Policy: An acknowledgement policy specifies the acknowledgement procedure at the receiver side.

<sup>&</sup>lt;sup>1</sup>Although UWB signals in general do not have a "carrier", a receiver must be able to sense if the channel is busy. Such a requirement is defined as the *receiver CCA performance* [2], [3].

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.

· Acknowledgement for packets in a burst

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.

• Pipelining acknowledgement

In IEEE 802.11, a receiver acknowledges any unicast packet that it has successfully received. To reduce the overhead of timing synchronization, we can apply the well-known *pipelining acknowledgement* scheme [11], 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 [12].

• 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.

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. Note that varying the spreading factor changes the transmission data-rate [2].

- Channel coding
- ARQ
- Hybrid ARQ and channel coding

## B. An MAC Protocol for UWB Networks

To evaluate the performance of the aforementioned framework, we design a simple MAC protocol based on the framework. In this protocol, we consider only one class of traffic in the network. For buffer management, we use tail-dropping when there is a buffer overflow. To assemble a burst frame, we require that all the packets in a burst frame have the same destination. Moreover, we assume that a burst frame will be generated if the channel is idle and if the total number of packets in the queue is at least  $B_{min}$ . In addition, the number of packets in a burst shall not exceed  $B_{max}$ . A receiver will send one ACK frame to the transmitter if a burst frame is correctly received. An finally, for simplicity reason, we do not use any error control scheme.

#### **III. SATURATION THROUGHPUT ANALYSIS**

In this section, we provide an analytical model to evaluate the saturation throughput performance of the MAC scheme proposed in Section II-B. Note that the proposed MAC keeps all the frame structures and most of the control procedure in IEEE 802.11. Using the Markov modelling technique in [13], [14], we partition the continuous time axis into intervals of length X(t), where t is the integer index of an interval; two consecutive intervals are delimited by the event of a value change in the backoff counter; then, we obtain a twodimensional discrete time embedded Markov chain with state  $\{s(t), b(t)\}$ , where t is discrete time index or the index of an interval, b(t) is the value of the backoff counter at t, and s(t)is the index of the backoff stage at t. To conduct the analysis, we make the following assumptions:

- There are N identical nodes in an ad hoc network.
- Any two nodes in the network can directly communicate with each other. In other words, we consider single-hop scenario only.
- The physical-layer transmission rate for every message is fixed at *R* (in bits/s).
- For any packet, the probability that its transmission is not successful, is independent of the backoff stage s(t) and has a fixed value p.
- After a packet is successfully delivered (i.e., the transmitter receives a positive acknowledgement), or is discarded after the maximum number of retries has been reached, a node can assemble a new burst that has  $B_{max}$  packets.
- Bit errors can occur uniformly in the payload of an MAC frame and the bit error rate is fixed at a value denoted by *ε*. The frame headers are assumed to be error free to simplify the analysis.
- The effect of propagation delay is negligible. This assumption is valid in a typical UWB WPAN scenario, which has a communication range less than 10 m (corresponding to 33 nanoseconds propagation delay).

Under the above assumptions, a 2-D embedded Markov chain  $\{s(t), b(t)\}$  can be formulated. Let M be the maximum index of backoff stages; or equivalently, the maximum number of retries for a packet. Let W and  $2^K W$  denote the minimum and maximum backoff window size, respectively. Then the backoff window size at stage m can be defined as

$$W_m = \begin{cases} 2^m W & m \le K \\ 2^K W & m > K;. \end{cases}$$
(1)

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

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

$$p_{\tau} = \begin{cases} \frac{2(1-2p)(1-p^{M+1})}{(1-2p)(1-p^{M+1})+W(1-p)(1-(2p)^{M+1})} & m \le K \\ \frac{2(1-2p)(1-p^{M+1})}{(1-2p)(1-p^{M+1})+W(1-p)(1-(2p)^{K+1})+W2^{K}p^{K+1}(1-2p)(1-p^{M-K})} & m > K \end{cases}$$
(3)

With the 2-D Markov chain, we can derive a closed-form solution for all  $b_{m,i}$ . Let  $p_{\tau} = \sum_{m=0}^{M} b_{m,0}$ . Then, assuming that a transmission is initiated if and only if i = 0, we can obtain the relationship between  $p_{\tau}$  and p, characterized by Eq. (3).

Since an successful packet delivery means that there is no collision and all data bits are received correctly, we can calculate p through

$$p = 1 - (1 - p_{\tau})^{N-1} \times (1 - p_e)$$
(4)

where  $p_e$  denotes the probability that a burst transmission fails due to bit errors.

Let  $B = B_{max}$  denote the total number of packets in a burst frame and  $f_B(n)$  denote the probability mass function of the total size n (in bits) of B packets. Based on the assumption that bit errors occurs uniformly, we can calculate  $p_e$  by

$$p_e = 1 - \sum_{n=0}^{\infty} f_B(n)(1-\epsilon)^n$$
 (5)

With  $0 and <math>0 < p_{\tau} < 1$ , we can calculate p and  $p_{\tau}$  numerically through Eq. (4) and Eq. (3).

To calculate the saturation throughput, we define the following parameters

• S denotes the normalized throughput, which is defined as the fraction of time that the channel is used to successfully transmit data packets, i.e.,

$$S = \frac{\text{E[correctly-received-payload transmission time]}}{\text{E}[X(t)]}$$

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

$$p_t = 1 - (1 - p_\tau)^N \tag{6}$$

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

$$p_s = N p_\tau (1 - p_\tau)^{N-1}$$
(7)

•  $L_B^s$  denotes the mean size of a successfully received burst, which can be expressed with  $f_B(n)$  as

$$L_B^s = \sum_{n=0}^{\infty} n f_B(n) p_s (1-\epsilon)^n \tag{8}$$

•  $\sigma$  denotes the length of a preset fixed time duration. When there is no packet transmission, we have  $X(t) = \sigma$ . In 802.11b direct sequence spread spectrum mode,  $\sigma = 20\mu s$ .

•  $T_s = E[X(t)|$ burst received without collision].

•  $T_c = E[X(t)|$ burst collision].

With the above parameters, we can calculate S by

$$S = \frac{L_B^s/R}{(1 - p_t)\sigma + p_s T_s + (p_t - p_s)T_c}$$
(9)

We now discuss the calculation of  $T_s$  and  $T_c$  in Eq. (9). Since the propagation delay can be ignored, we can obtain  $T_s$  for the basic access scheme by

$$T_s^{basic} = 2T_{sync} + T_{SIFS} + T_{DIFS} + \frac{2L_{PH} + L_{MH} + L_{ACK} + L_B}{R}$$
(10)

where  $T_{sync}$  denotes the synchronization time,  $T_{SIFS}$  denotes the time duration of SIFS,  $T_{DIFS}$  denotes the time duration of DIFS,  $L_{PH}$  denotes the length of physical frame header in bits (excluding the synchronization preamble),  $L_{MH}$  denotes the length of MAC frame header in bits,  $L_{ACK}$  denotes the length of ACK frame in bits, and  $L_B$  is the mean length of the burst payload in bits, i.e.,

$$L_B = \sum_{n=0}^{\infty} n f_B(n).$$

For the RTS/CTS access method, we have

$$T_s^{rts} = T_s^{basic} + 2T_{sync} + 2T_{SIFS} + \frac{2L_{PH} + L_{RTS} + L_{CTS}}{R}$$
(11)

where  $L_{RTS}$  and  $L_{CTS}$  denote the length of RTS and CTS frame in bits.

Since the propagation delay is negligible, the ACK timeout is same as EIFS, the length of which is given by

$$T_{EIFS} = T_{SIFS} + T_{DIFS} + T_{sync} + \frac{L_{PH} + L_{ACK}}{R}$$
(12)

Hence, for the RTS/CTS scheme,  $T_c$  is given by

$$T_{c}^{rts} = 2T_{sync} + T_{SIFS} + T_{DIFS} + \frac{2L_{PH} + L_{RTS} + L_{CTS}}{R}$$
(13)

Next, we derive  $T_c$  for the basic access scheme. Assume there are J ( $J \ge 2$ ) nodes transmitting in an interval of length X(t). Denote n(k) the length of the burst sent by node k. Denote  $n_{max}$  the maximum of n(k), for  $k = 1, \dots, J$ . Then the conditional probability that  $n_{max} = n$  given there are Jnodes transmitting, denoted by  $q_{n|J}$ , can be expressed by

$$q_{n|J} \stackrel{\triangle}{=} Pr\{n_{max} = n|J\} = \left[\sum_{l=0}^{n} f_B(l)\right]^J - \left[\sum_{l=0}^{n-1} f_B(l)\right]^J \tag{14}$$

# TABLE I

SETTING OF THE MAC PROTOCOL.

minimum contention window size	8
maximum contention window size	256
σ	$2 \ \mu s$
SIFS	$1 \ \mu s$
DIFS	5 µs
long retry limit	4
short retry limit	7
buffer size	50
$T_{sync}$	$10 \ \mu s$
packet size	1000 Bytes

Let  $L_B^c = E[n_{max}]$ . Then, we have

$$L_B^c = \sum_{J=2}^N \frac{\binom{N}{J} p_{\tau}^J (1 - p_{\tau})^{N-J}}{p_t - p_s} \times \left(\sum_{n=0}^\infty n \cdot q_{n|J}\right) \quad (15)$$

With  $L_B^c$ , we can obtain  $T_c^{basic}$  by

$$T_c^{basic} = T_s^{basic} + \frac{L_B^c - L_B}{R} \tag{16}$$

### IV. 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. Due to limited space, we only present results under the following setting:

- All nodes are located in a 4 m  $\times$  4 m area.
- The physical-layer transmission rate, denoted by *R*, is the same for all messages.
- Packet arrivals to any node are modelled by a Poisson process with the same rate λ. We further assume that the total arrival rate of data (in bits/s or b/s) is equal to R. In other words, we let the traffic load be 1 Erlang.
- Unless otherwise specified, we assume BER is 0.
- Unless otherwise specified, we assume that the RTS/CTS scheme is used.

In this study, 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.

Fig. 1 shows the throughput and average end-to-end delay vs.  $B_{max}$  for both the basic and the RTS/CTS scheme, with R = 50 Mb/s and N = 10. The saturation throughput is calculated by  $S \times R$ , where S is computed by Eq. (9). We first observe that the saturation throughput 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



Fig. 1. Performance versus  $B_{max}$  with different  $B_{min}$  (R = 50 Mb/s, N = 10)

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. 1 (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. 1(a)). From Fig. 1(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  $B_{max} = 1$ . Comparing case (1) and (2), we note that the average end-to-end delay is reduced if we assemble more packets into a burst.

Fig. 2 shows the throughput and average end-to-end delay vs. the number of nodes N when R = 50 Mb/s. Here we compare two cases: 1)  $B_{min} = B_{max} = 1$  (benchmark) and 2)  $B_{min} = B_{max} = 10$ . From Fig. 2(a), we observe that the saturation throughput for case (2) is significantly higher than that for the benchmark case. Specifically, the saturation throughput for case (2) is 16 Mb/s larger than that for case (1) when N = 20. In Fig. 2(b), we see that the average end-to-



Fig. 2. Performance versus the number of nodes (R = 50 Mb/s)

end delay for case (2) is smaller than that of case (1), which implies that the assembly delay is not a major component of the end-to-end delay in the saturated conditions.

Fig. 3 shows the throughput performance of the proposed MAC under different BER conditions, where R = 100 Mb/s, N = 10, and  $B_{min} = B_{max}$ . We observe that, the saturation throughput for BER= $10^{-7}$  is almost the same as that for BER=0. As the BER increases to  $10^{-6}$ , the saturation throughput is slightly reduced. A significant change occurs when BER= $10^{-5}$ , where we see that the throughput decreases with the increase of  $B_{max}$  if  $B_{max} > 3$ . Note that BER= $10^{-5}$  will lead to a packet error rate of about 10% if the size of the packet is 10000 bits. From Fig. 3, we see that the proposed scheme performs better than the benchmark case where  $B_{max} = 1$  even under BER > 0.

#### V. 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,



Fig. 3. Throughput performance versus  $B_{max}$  with different BER (R=100 Mb/s,  $N=10,\,B_{min}=B_{max})$ 

and analyzed its saturation throughput performance. Extensive simulation and numerical 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 under different data rates and different bit error rates.

#### REFERENCES

- First report and order in the matter of revision of part 15 of the commissions rules regarding ultra-wideband transmission systems, Federal Communications Commission (FCC 02-48), Std., Apr. 2002, ET Docket 98-153.
- [2] IEEE P802.15-04/0137r1, "DS-UWB physical layer submission to 802.15 Task Group 3a," Mar. 2004, Project: IEEE P802.15 Working Group for Wireless Personal Area Networks (WPANs).
- [3] 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).
- [4] S. Roy, J. R. Foerster, V. S. Somayazulu, and D. G. Leeper, "Ultrawideband radio design: The promise of high-speed, short-range wireless connectivity," *Proc. IEEE*, vol. 92, no. 2, pp. 295–311, Feb. 2004.
- [5] W. Zhuang, X. Shen, and Q. Bi, "Ultra-wideband wireless communications," Wireless Communications and Mobile Computing, Special issue on Ultra-Broadband Wireless Communications for the Future, no. 6, Dec. 2003.
- [6] R. L. Peterson, R. E. Ziemer, and D. E. Borth, *Introduction to Spread Spectrum Communications*. Prentice Hall, 1995.
- [7] 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 Conference on Ultra Wideband Systems* and Technologies'2002, 2002, pp. 99–103.
- [8] S. Aedudodla, S. Vijayakumaran, and T. F. Wong, "Rapid ultra-wideband signal acquisition," in *Proc. IEEE WCNC'04*, Mar. 2004.
- [9] 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, 1998, pp. 59–65.
- [10] 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*'2002, July 2002.
- [11] A. S. Tanenbaum, Computer Networks, 4th ed. Prentice-Hall, 2003.
- [12] 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.
- [13] G. Bianchi, "Performance analysis of the ieee 802.11 distributed coordination function," *IEEE J. Select. Areas Commun.*, vol. 18, no. 3, pp. 535–547, Mar. 2000.
- [14] 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*, June 2002, pp. 599–607.