PERFORMANCE ANALYSIS OF A HIGH-SPEED ULTRA-WIDEBAND WPAN MAC

Sergey D. Andreev, Andrey M. Turlikov and Alexey V. Vinel
Department of Information Systems
State University of Aerospace Instrumentation
Bolshaya Morskaya, Saint-Petersburg, Russia
E-mail: vinel@ieee.org

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ABSTRACT

The current paper addresses the problem of the throughput-delay performance of a contemporary WPAN MAC standard. A brief overview of the standard functionality is first presented that allows a system model derivation. Two different acknowledgement policies are described under which the channel operation is considered. Two possible input traffic models are also considered one of them being saturation conditions under which the performance analysis is done that is further verified by means of the simulation. The obtained results show the system behavior as the number of channel users increase and allow the tuning of the protocol parameters to improve the performance.

INTRODUCTION

With the advent of new technologies the wireless data networks attract more and more attention in the modern world. As a consequence the interest for the analysis of such networks grows steadily. This paper is aimed at the performance analysis of a newly introduced wireless personal area network (WPAN) standard. This standard (ECMA 2005) considers the ultra-wideband physical layer and offers unrivaled data rates which will ensure its applicability. As noticed in (Vishnevsky et al. 2006b) the attention to this standard, especially to its Medium Access Control (MAC) sublayer, is underpaid which leads to limiting its capabilities as 'fine tuning' is yet to be done.
A brief overview of the standard is given in (Vishnevsky et al. 2006a). The current paper extends the understanding of the MAC features and tracts their relations down to the IEEE 802.11 and IEEE 802.16 standards. It is structured as follows. The "MAC SIMPLIFIED DESCRIPTION" section provides a review of the standard-defined mechanisms for the channel access. It is followed by the "PROBLEM STATEMENT" section where the traditional performance metrics are discussed and the necessary questions are posed.
"ANALYTICAL RESULTS" provide more insight into the MAC performance analysis mainly following the approach of (Bianchi 2000) for IEEE 802.11 and of (Vinel et al. 2005) for IEEE 802.16, considering regenerative system behavior, which was noticed in (Vishnevsky and Lyakhov 2002). "NUMERICAL RESULTS" section addresses the verification of the obtained results by means of the simulation. It shows that the information about the number of users in the network, which is available at the MAC layer, can be used to increase the overall system performance. "CONCLUSION" summarizes the main contributions of the paper.

MAC SIMPLIFIED DESCRIPTION

The current paper considers the MAC sublayer of the OSI Reference Model for the high-speed personal ultra-wideband (UWB) wireless networks (ECMA 2005) (referred to as Standard below). The Standard is flexible and supports numerous features such as user mobility, power management, advanced security, range measurement and many more. Therefore only the basic functionality that is relevant to the point of the current paper will be addressed below.
The channel operation in time may be considered as a sequence of adjacent *superframes* which start times (BPSTs) are known to all the channel users. Structurally, each superframe consists of the *beacon period* (BP) immediately followed by the *data period* (see Figure 1). In the BP only *beacons* are transmitted by users with beacon being a type of broadcast frame for the managerial purposes. The beacon includes the necessary service data including the user's id thus enabling all the neighbors to know exactly the total number of users in the system.

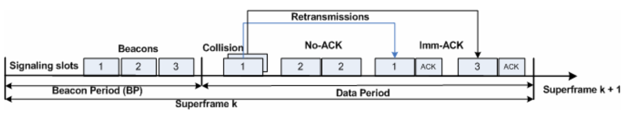

Figure 1: Channel Operation Snapshot

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In the data period no beacons are transmitted. Instead, three frame types, namely *data* (referred to as *packet* below), *control* and *command* frames may be sent by a user. Alike beacons, control and command frames serve managerial purposes. All the superframes have the same size (65 ms approximately) which means that the more users send beacons in the BP, the fewer data frames may be transmitted in the data period. Should a new user wish to join an existing group of users (beacon group) it first announces its intention in any of the two beacon slots (*signaling slots*) left empty by the other users. The Standard has a robust mechanism ensuring all the users have the same view of the channel operation, i.e. the same BP length and the number of participating users since a so-called hidden terminals problem may occur.

Consider now the data period in more detail. The Standard defines two basic mechanisms of channel access, namely, Distributed Reservation Protocol (*DRP*) and Prioritized Contention Access (*PCA*). The former is a reservation technique enabling users to schedule the channel time for further use through negotiation. The latter is a randomized access scheme which is a generalization of the popular CSMA/CA (Carrier Sense Multiple Access with Collision Avoidance) algorithm. Since several reservation types are introduced by the Standard each having complex functional rules it is clear that the proper DRP operation require a sophisticated resource allocation policy to be developed for every channel user type. Therefore one can expect that in practice PCA operation will be more popular than that of DRP. For this reason the primary focus of the current paper will be set on the PCA channel access scheme.

The Standard offers a good variety of the acknowledgement (*ACK*) policies, i.e. different sets of rules according to which the successful receipt of a transmitted packet is verified. The most traditional is Imm-ACK policy, when the acknowledging frame (which is a control frame) is sent back to the sender immediately (as soon as possible) following the successful receipt of a single data frame. This policy is sensible to set when the transmitting data is delay-sensitive and should be delivered with maximum reliability. If the sender receives no ACK frame it retransmits the same packet until in the end its receipt is verified or the limit of retransmission tries is reached when the frame is discarded. The next policy, No-ACK, requires no acknowledgement to be sent. Thus the sender treats each frame that is sent successfully as successfully received. No-ACK is chosen when the data is the most delay-sensitive but tolerate packet drops due to the channel collisions or the background noise. The last possible policy is B-ACK when a sender transmits a block of packets and upon a special request the receiver responds with an ACK vector to acknowledge those packets that are received successfully. The sender may retransmit the packets that dropped.

PROBLEM STATEMENT

Traditionally, in the multiple access theory and its applications, two main metrics are used for the performance evaluation of a MAC protocol. The terms and definitions of these metrics may differ, depending on the problem features and practical issues. In particular, the first metric indicates the efficiency of the channel resources usage and is further referred to as *throughput* being the ratio of channel time spent to transmit the packet payload information. By contrast, the *delay* is the random variable, which characterizes the efficiency of a protocol from the point of view of an arbitrary user. The delay can be measured for any packet in the system as the time interval from the moment the packet arrives at the user queue till it is successfully transmitted. The computation of the throughput and the mean delay values for a typical UWB network scenario is done in the rest of the paper. The following practical questions are to be answered during the analysis:

- What are the values for the mean delay and the throughput when the protocol operates in some typical scenario?
- What is the maximal number of users the protocol can support, provided reasonable values of mean delay and throughput are observed?
- How can the system performance be improved by means of tuning the MAC protocol parameters?

The answers to these questions are found by means of both analytical techniques and interactive simulation.

SYSTEM MODEL

The total of n *users* share the same broadcast channel. As it is baseless to predict how would n change in practice it is considered that the total number of users is constant, which implies no user arrives to or departs from the system. The channel conditions are *ideal*, i.e. there is no background noise (noiseless channel) and all the users can receive each other's transmissions ensuring no hidden terminal is present. Therefore, the three possible situations, namely, successful, collision or empty channel, are distinguished by all the users. The duration of each situation depends solely on the acknowledgement policy considered. Note that the B-ACK policy is parameterized since an implementer must define a number of successive packet transmissions which are verified by a single acknowledgement vector. To avoid implementation issues only the *Imm-ACK* and *No-ACK* policies were chosen to illustrate the system behavior.

For simplicity reasons the time axis is slotted into equal simulation slots. A simulation slot time duration is set to a user clock resolution value as defined in the Standard. As this value is rather small ($1\,\mu s$) such a synchronization can be used for the simulation of the transmissions in the data period.

Two different models of *input traffic* are considered. For the first one the packet arrivals to the system represent Bernoulli process with the constant overall *intensity* $\lambda = \xi n$ packets per superframe, where ξ is the probability that a user generates a packet in a simulation slot. The second traffic model is that of the *saturation conditions* ensuring a user always has a packet that is prepared for transmission. All the packets have the same size of L bytes in total including the necessary headers and trailers. The arrived packet is queued by the user until the moment it is transmitted successfully. The queues are unbounded so that no packet drops would be possible due to overflow.

The PCA protocol defined in the Standard, as mentioned above, is a generalization of the renowned CSMA/CA channel access scheme for the four increasing priorities (categories) of the input traffic, which are Background, Best Effort, Video and Voice respectively. It is based on the truncated binary exponential backoff (*BEB*) conflict resolution protocol. If the channel in *not* sensed busy during the defined Arbitration Inter-Frame Space (*AIFS*) interval the user transmits a pending packet immediately. Otherwise, the user monitors the channel until it is *not* sensed busy and performs a so-called backoff by setting the *backoff counter* value as described below. If collision is sensed (no immediate acknowledgement returned within the SIFS time after the packet transmission) a user also delays the further retransmission for some future time by setting the new backoff counter.

The value of the backoff counter is sampled uniformly in the range $[0, W-1]$, where W is the current value of the *contention window*. The backoff counter value is decreased by a unity after the channel is not sensed busy for AIFS and afterwards every time the empty slot is detected and remains unchanged ('frozen') otherwise (in case of collision or success). When the backoff counter reaches zero a user transmits. In the initialization stage and every time the transmission is successful the user sets its W value to the predefined constant $W_{\min}$. In case of collision the contention window value is doubled until it reaches the upper bound of $W_{\max} = 2^m W_{\min}$ to stop growing (the value of m is referred to as the *maximal backoff stage*). Note that if No-ACK policy is used then the W value is never increased and remains equal to $W_{\min}$ throughout the operation.

One important innovation that extends the classical formulation of the BEB algorithm is the transmission opportunity (*TXOP*) concept. A TXOP is the amount of time for which a user 'captures' the channel and within which it transmits its packets with an interval of SIFS only and without performing the backoff. More specifically, a user transmits pending packets, if any, until all of them are transmitted successfully, collision is sensed, or it reaches a given TXOP limit (Θ). In every outcome a user backoffs according to the BEB rules. Clearly, in the saturation conditions users always transmit maximum packets until TXOP limit provided no collision is sensed. Each traffic priority is defined by the specific values of AIFS, $W_{\min}$, $W_{\max}$ and TXOP limit (see Table 1).

Table 1: System parameters

Parameter name	Parameter variable	Parameter value
Simulation slot time	mClockResolution	1 μs
Traffic priority	mPriority	AC_BK (Background)
Data rate	mRate	53.3 Mbps
Total number of users	n	*Variable*
Total superframe intensity	Λ	*Variable* packets per superframe
Total simulation slot intensity	λ	Λ / T_{SF} packets per simulation slot
User simulation slot intensity	ξ	λ / N packets per simulation slot
Packet length	L	1000 bytes
Superframe duration	mSuperframeLength, T_{SF}	65536 μs
Beacon slot duration	mBeaconSlotLength	85 μs
Beacon period (BP) duration	T_{BP}	$(N+2) \cdot$ mBeaconSlotLength μs
Empty slot duration	pSlotTime	9 μs
SIFS duration	pSIFS	10 μs
AIFS duration	AIFS	$7 \cdot$ pSlotTime + pSIFS
Minimum contention window value	mCWMin, $W_{\min}$	15
Maximum contention window value	mCWMax, $W_{\max}$	1023

Parameter name	Parameter variable	Parameter value
Packet duration	$T_{packet}(L)$	$\left\lceil (42+6\cdot\left\lceil {(8\cdot L+38)}/{100}\right\rceil)\cdot 0.3125\right\rceil\ \mu s$
Payload duration	$T_{payload}(L)$	$6\cdot\left\lceil {(8\cdot L+38)}/{100}\right\rceil\cdot 0.3125\ \mu s$
Acknowledgment duration	T_{ack}	$14\ \mu s$
Transmission opportunity limit	mTXOPLimit, Θ	$512\ \mu s$

ANALYTICAL RESULTS

Due to the complexity of the system defined by the Standard, the analytical results are obtained only for the asymptotic case, when overall traffic intensity is very small ($\lambda \to 0$) and for the saturation case model.

In saturation conditions an approach similar to that of (Binachi 2000) for IEEE 802.11 can be applied. Notice that the main differences in the system model formulated above from the model in (Bianchi 2000) are beacon periods, TXOP limits and No-ACK policy. Therefore, the following assumptions are introduced of which the last two repeat those of Bianchi:

- Only the data period is first considered. The existence of the beacon period will be taken into account in the further analysis.
- The time axis is slotted into non-equal slots. All the users know the slots borders. The duration of a slot depends on the situation in the channel (empty, success or collision). Asynchronous transmissions that occur during the data period are regarded as the synchronous ones.
- The probability that a user starts sending in a slot depends neither on the previous history nor on the behavior of the other users and is denoted as π.

In the framework of the above assumptions, the operation of an arbitrary user is considered. Denoting p as the conditional collision probability of a user making an attempt to transmit in some slot, it is easy to obtain:

$$p = 1-(1-\pi)^{n-1}. \tag{1}$$

Observing the user states in the beginning of each slot, a sequence of states can be represented by a two-dimensional Markov-chain:

$$\{s(t),\ c(t)\}, \tag{2}$$

where $s(t)$ is the ratio $W/W_{\min}$ for the user at the beginning of a slot starting at the moment t and $c(t)$ – the value of the user backoff counter at the moment t.

The approach for the computation of π is based on the observation, that the process (2) is a renewal one. Indeed, one can show that according to the binary exponential backoff rules, the moments of user successful transmission are the renewal points.

Consider a process of a packet transmission by a user. Let $\overline{N}$ be the mean number of the packet transmission attempts and $\overline{K}$ be the mean number of slots the user defers the transmission for during this process. Then, the probability π is computed as follows:

$$\pi = \frac{\overline{N}}{\overline{N}+\overline{K}}. \tag{3}$$

It is easy to see, that the number of the packet transmission attempts is distributed geometrically, thus:

$$\overline{N} = \sum_{i=1}^{\infty} i(1-p)p^{i-1} = \frac{1}{1-p}. \tag{4}$$

Let $\overline{K_i}$ be the mean number of slots the user has been deferring its transmission for, conditioning that exactly i attempts were made to successfully transmit the packet, then

$$\overline{K} = \sum_{i=1}^{\infty} \overline{K}_i (1-p)p^{i-1}. \tag{5}$$

One can show that the following equations hold:

$$\overline{K}_i = 2^{i-1}W - \frac{W+i}{2},\ \text{for } 1 \le i \le m+1, \tag{6}$$

$$\overline{K}_i = 2^{m}W\frac{i-m+1}{2} - \frac{W+i}{2},\ \text{for } i > m+1. \tag{7}$$

Substituting (4) – (7) into (3) and after some algebraic simplifications it can be obtained, that:

$$\pi = \frac{2(1-2p)}{(1-2p)(W+1)+pW(1-(2p)^m)}. \tag{8}$$

The equations (1) and (8) represent the system of two non-linear equalities with two unknowns, π and p, which can be solved numerically to compute π for Imm-ACK. For No-ACK policy the contention window is never increased, thus $m = 0$, what leads to a simple expression for the slot transmission probability:

$$\pi = \frac{2}{W+1}. \tag{9}$$

Following the above way to compute π the durations of slots in the system for Imm-ACK and No-ACK polices can be summarized in Table 2.

Table 2: Expressions for the Slots Durations

Situation in the channel	Slot duration for Imm-ACK	Slot duration for No-ACK
Success	$T_s^{(1)} = T_{packet}(L) + pSIFS + T_{ack} + AIFS$	$T_s^{(2)} = \Theta + AIFS$
Empty	$T_e = pSlotTime$	
Collision	$T_c^{(1)} = T_{packet}(L) + AIFS$	$T_c^{(2)} = \Theta + AIFS$

Finally, the throughput S can be computed using the following formula:

$$S = \frac{E[data\ period\ length]}{T_{SF}} \cdot \frac{E[payload\ per\ slot]}{E[slot\ duration]}, \qquad (10)$$

which leads to

$$S = (1 - \frac{T_{BP}}{T_{SF}}) \cdot \frac{E[payload\ per\ slot]}{T_e(1-\pi)^n + T_s n\pi(1-\pi)^{n-1} + T_c(1 - n(1-\pi)^{n-1} - (1-\pi)^n)}, \qquad (11)$$

where the values for the slots durations are taken from Table 2. The expressions

$$E[payload\ per\ slot] = \left\lfloor \frac{\Theta}{T_{packet}(L) + pSIFS} \right\rfloor \cdot T_{payload}(L) n\pi(1-\pi)^{n-1} \qquad (12)$$

for Imm-ACK and

$$E[payload\ per\ slot] = \left\lfloor \frac{\Theta}{T_{packet}(L) + 2pSIFS + T_{ack}} \right\rfloor \cdot T_{payload}(L) n\pi(1-\pi)^{n-1} \qquad (13)$$

for No-ACK, finish the derivation of the throughput.

For $\lambda \to 0$ the mean delay for the packet transmission D_0 can be computed observing a packet arrival at the empty system, which leads to

$$D_0 = (1 - \frac{T_{BP}}{T_{SF}}) T_s^{(1)} + \frac{T_{BP}}{T_{SF}} (T_{BP}/2 + T_s^{(1)}) \qquad (14)$$

for Imm-ACK policy, and

$$D_0 = (1 - \frac{T_{BP}}{T_{SF}})(AIFS + T_{packet}(L)) + \frac{T_{BP}}{T_{SF}}(T_{BP}/2 + AIFS + T_{packet}(L)) \qquad (15)$$

for No-ACK policy.

The first terms of the equations (14) and (15) correspond to the case, when a new packet arrives during the data period, while the second terms do to the case, when the arrival takes place during the BP.

NUMERICAL RESULTS

An accurate simulation program has been developed to validate the obtained results. It uses event-driven simulation with a slotted time axis as discussed in the "SYSTEM MODEL" section. The parameters for the simulation runs are summarized in Table 1.

In Figure 2 the saturation throughput versus the user number is demonstrated. Notice that the analytical results give a good approximation of the system performance. The two acknowledgement policies, namely, Imm-ACK and No-ACK, are compared for the standard-defined parameters. Notice that as the number of users increase the throughput performance of the No-ACK policy degrades dramatically.

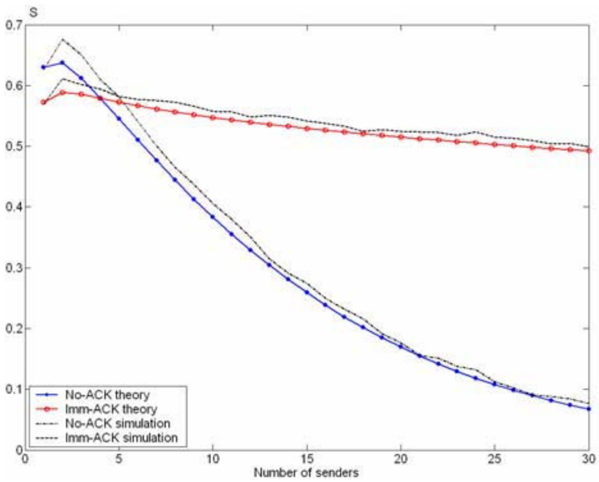

Figure 2: Saturation Throughput Analysis

Low No-ACK throughput is particularly noticeable since the given values are the upper bounds for the real channel throughput because in the presence of the background noise the channel operation suffers further degradation. This effect is basically due to the fact that the users never increase their contention window values. By contrast, the Imm-ACK policy is shown to give higher throughput values for all the user number range considered even despite the ACK overhead.

As mentioned above, there is the information available at the MAC layer about the exact number of users in the system (n). The improvement of the collision resolution protocol on the basis of this information can be done to maximize the saturation throughput. In Imm-ACK case a derivative of (11) should be calculated and set equal to zero. The resulting equality is easy to solve under the assumption that $\pi \ll 1$, which implies $(1-\pi)^n \approx 1 - n\pi + \frac{n \cdot (n-1)}{2} \cdot \pi^2$ and $\pi \approx (n \cdot \sqrt{T_c / 2 \cdot T_e})^{-1}$. Further through (1) p is obtained, which is used in (8) (or in (9) in No-ACK case) together with π.

The pair (W, m) should next be found which is the solution of the derived equality that maximizes the throughput. Generally, throughput maximization does not lead to the delay minimization. However, as it is illustrated in Figure 3, for the UWB MAC such a choice of parameters reduces the mean delay for the high intensities.

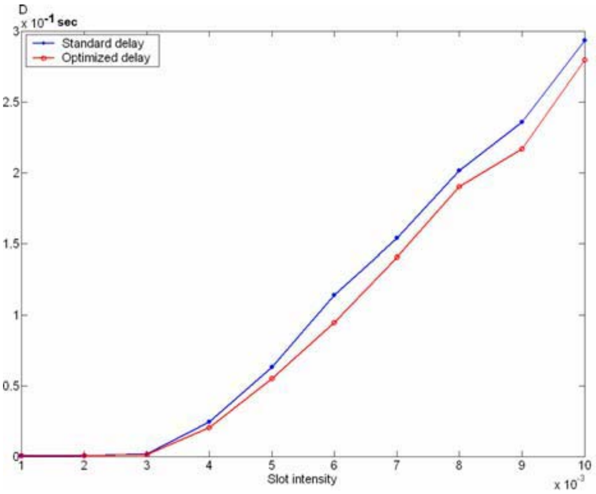

Figure 3: Mean Delay Analysis

CONCLUSION

The main contributions of this paper are twofold. Firstly, the analytical model is developed for the performance analysis of the UWB MAC. Secondly, some interesting numerical results are obtained for the performance metrics of the network and a way to improve the system performance is shown. The further development of the above model is the current research activity of the authors.

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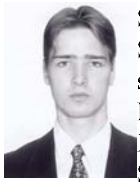

SERGEY D. ANDREEV was born in Saint-Petersburg, Russia, in 1984. He studied Computer Science at Saint-Petersburg State University of Aerospace Instrumentation, from where he received a Specialist's degree in 2006. As Ph.D. student at the same university, he continues a research in the field of the random multiple access and is currently interested in wireless telecommunications.

ANDREY M. TURLIKOV is an associate professor of information systems department of Saint-Petersburg State University of Aerospace Instrumentation, where he graduated from in 1980 and received his Ph.D. degree in 1986. Since 1987 he has been involved in teaching work. Dr. Turlikov is the author of more than 60 publications. His research interests include multiple access systems and real-time data transmission protocols.

ALEXEY V. VINEL was born in Saint-Petersburg, Russia in 1983. He has received his bachelor and master degrees in information systems from Saint-Petersburg State University of Aerospace Instrumentation, (where he is a Ph.D. student now), in 2003 and 2005 respectively, both with honors. Since October 2004 till March 2005 Mr. Vinel did an internship at Siemens AG, Munich, Germany. Mr. Vinel's research interests include random multiple access algorithms analysis and performance evaluation of wireless networks.