# A Noble Equalizer Structure with the Variable Length of Training Sequence for Increasing the Throughput in DS-UWB

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## ABSTRACT

The training sequence with the appropriate length for equalization and initial synchronization is necessary before sending the pure data in the burst transmission type DS-UWB system. The length of the training sequence is one of the factors which make throughput decreased. The noble structure with the variable length of the training sequence whose length can be adaptively tailored according to the channel conditions (CM1,CM2,CM3,CM4) in the DS-USB systems is proposed.

This structure can increase the throughput without sacrificing the performance than the method with fixed length of training sequence considering the worst case channel conditions. Simulation results under IEEE 802.15.3a channel model show that the proposed scheme can achieve higher throughput than a conventional one with the slight loss of BER performance. And this structure can reduce the computation complexity and power consumption with selecting the short length of the training sequence.

**Key Words :** Ultra-WideBand(UWB), equalizer, dense multipath, wireless personal area networks (WPANs)

## Ⅰ. Introduction

Recently a demand for the short-range wireless communication with low cost, low power and high datarate wireless access is increasing.

Ultra-wideband(UWB) is one of the promising techniques for supporting the high datarate multimedia service. UWB systems transmits very short pulses(usually shorter than 1ns) with extremely wide bandwidths(upto7.5GHz)[1],[2]

The UWB systems can be divided into two groups: one is single band system and the other is multi-band system. In this paper, we study a single band UWB system based on DS-CDMA technique, namely DS-UWB system. In a DS-UWB system, the pulses are transmitted continuously using a pseudorandom sequence for the spreading of information bits.

Single-user coherent rake receiver scheme with IEEE802.15a channel model has been proposed for DS-UWB system[5]. The scheme with rake receivers requires channel estimation to perform diversity receiver by the complex channel estimation algorithm. In order to capture enough signal energy from dense multipath components, it is necessary to increase the number of rake fingers. However it causes the hardware implementations to be complex and the computation to be increased.

Another approach to overcome the above-mentioned disadvantages is based on the non-coherent reception techniques, such as differential receivers [4] and adaptive LMS and RLS receivers [6]. These techniques do not require channel estimation and allow capturing a large amount of the received energy.

However, differential receivers represent a sub-optimal solution compared with coherent rake receivers and are respected to work only at rather

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high signal-to-noise ratios [6]. Adaptive LMS and RLS receivers perform with simple structure and easiness of implementation. In the adaptive LMS and RLS receivers, it is necessary to send the training sequence for the equalization as well as the initial synchronization. Normally in the burst transmission type DS-UWB system, the longer the length of the training sequence, the lower the throughput. In this paper, the noble equalizer structure with the variable length of training sequence according to the channel conditions instead of the fixed length of training sequence is proposed.

We evaluate performance on convergence rate, complexity and robustness for the proposed equalizer structure and examine their BER performance under the realistic IEEE802.15.3a channel model. And also the throughput was compared with normal scheme through the simulation.

## Ⅱ. The structure of the proposed equalizer

### 2.1 The Conventional Adaptive Equalizer Structure

Fig.1. shows a general adaptive equalizer configuration, where k is the iteration number. $x_i(k)$, i. 0,1,...,N. denotes the input signal, y(k) is adaptive filter output signal, and d(k) defines the desired signal. The error signal is calculated as d(k)-y(k). The error signal e(k) is then used for the adaptive algorithm in order to determine the appropriate updating of the filter coefficients $w_i(k)$, i= 0,1, ...,N.

### 2.2 The Proposed Equalizer Structure

In this section, we propose a novel equalizer structure shown in Fig.2.

In a conventional adaptive transceiver structure, a training sequence of fixed length is employed during the training phase [6]. However, neither the channel state nor its variation is known in advance actually. In other words, we cannot determine how long the training sequence should be employed to perform training. If the length is taken too short, it may give rise to insufficiency of the tap coefficients' convergence and result in degradation of the BER performance. On the other hand, too long training sequence may induce loss of the information throughput. In the conventional UWB systems, the longest training sequence should be used considering the worst case of channel condition.

In order to capture enough multipath energy, the observation window of the adaptive filter is typically longer than one bit interval and therefore, windows overlap in time.

Some adaptive filter algorithms, such as LMS, RLS [6], can be applied to perform updating the tap coefficients of the filter. However, we should note that the characteristics of an adaptive filter algorithm such as convergence rate, complexity and robustness, etc, must be considered when it is applied to DS-UWB systems. Recently, some paper

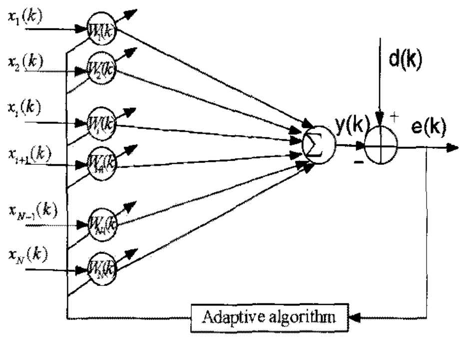


Fig. 1. The conventional equalizer structure

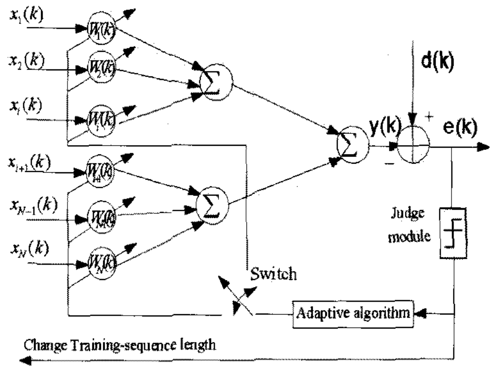


Fig. 2. The proposed equalizer structure

had discussed this problem. The recursive least squares (RLS) algorithm has also been considered in DS-UWB systems [6]. While it can typically provide more rapid convergence than the conventional LMS algorithm, a major drawback of the RLS algorithm is its computation complexity. For instance, the complexity of the LMS algorithm is O(Np), where "big O" notation is used to indicate that the algorithm complexity is proportional to the product of N and p. N is the symbol span of the filter and p is the number of samples per symbol. Whereas the complexity of the RLS algorithm becomes $O(N^2p^2)$ [7].

To cope with this disadvantage, now we propose a novel adaptive structure shown in Fig.2.

The training data are inputted into the proposed adaptive equalizer and after each filtering, the square of error (SE) is computed and compared with a threshold δ by judge module in Fig.2. The judge module can transform the training number in the training phase and change the running taps number in the transmission. In this proposed structure, the number of taps is decreased as half of the total number of taps when the MSE converges to the appropriate threshold. Because in CM1, equalizer arrives at convergence earlier than CM4, it is possible to make the number of taps reduced. The novel structure can economize the cost of transmitting training sequences for adaptation phases and reduce the complexity of the adaptive algorithm. For instance, the complexity of the LMS algorithm becomes O(Np/2), whereas the complexity of the RLS algorithm becomes $O((N/2)^2P^2)$ shown in Table 1.

Table. 1. Computational Complexity

| Algorithm | Computational Complexity | | |
|---|---|---|---|
| | *All-Taps* | *Half-Taps* | *Proposed* |
| LMS | $O(NP)$ | $O(\frac{N}{2}P)$ | $O(NP) \rightarrow O(\frac{N}{2}p)$ |
| RLS | $O(N^2P^2)$ | $O\left((\frac{N}{2})^2P^2\right)$ | $O(N^2P^2) \rightarrow O\left((\frac{N}{2})^2p^2\right)$ |

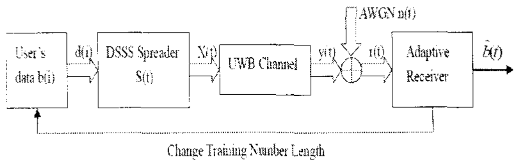


Fig. 3. the proposed transceiver architecture block diagram for DS-UWB System

## Ⅲ. System description

We consider a single-user DS-UWB system signaling through UWB multipath channel, where the user's data are modulated by BPSK direct-sequence spread-spectrum(DSSS) modulation. The proposed adaptive transmit-receive scheme is illustrated in Fig.3. We can see that it consists of a transmitter, a proposed adaptive receiver scheme.

### 3.1 Transmitter Module

The UWB transmitter is shown in the left part of Fig.3. The user's data are filled with an equiprobable binary bit stream $\{b(i)\} \in \{1,-1\}_{i=0}^{p-1}$ (P denotes the number of bits per packet) and a bipolar pseudorandom sequence is generated and stored as training sequence. The DSSS spreader's input d(i) is switched between the data bit stream and the training sequence according to the link's status. At the spreader's output, the transmitted signal is given by

$$x(t) = \sum_{i=0}^{P-1} d(i) \bullet S(t - iT) \tag{1}$$

where d(i) could be either one information bit or a training bit that is known to the receiver. $T_b$ denotes the duration of one bit signal and S(t) is the normalized spreading-spectrum waveform given by

$$S(t) = \sum_{j=0}^{N-1} n(j) \bullet w_{tr}(t - jT_p) \tag{2}$$

where n(j) is a pseudorandom spreading code that takes values {1, −1} and has a period longer than N. It is assumed that each information bit consists of N pulses and each pulse's duration is $T_p$. $w_{tr}(t)$ denotes the UWB pulse and generally has the expression of the second derivative of Gaussian pulse [5]

$$w_{tr}(t) = \left[1 - 4\pi(\frac{t}{\tau_m})^2\right] \cdot \exp\left[-2\pi(\frac{t}{\tau_m})^2\right] \tag{3}$$

where $\tau_m$ is the exponential factor that dominates the pulse's duration $T_p$ .

### 3.2 Channel Model

The IEEE 802.15.3a channel model is based on a modified Saleh-Valenzuela model where multipath components arrive in clusters and each cluster could contain several components namely rays [8]. The channel impulse response can be described as

$$h(t) = X\sum_{l=0}^{L-1}\sum_{m=0}^{M-1}\alpha_{m,l}\,\delta(t - T_l - \tau_{m,l}) \tag{4}$$

where the real-valued multipath gain is defined by $\alpha_{ml}$ for cluster l and ray m. The l th cluster arrives at $T_l$ and its mth ray arrives at $\tau_{m,l}$ which denotes the delay of the m th component relative to the first part in the l th cluster i.e. $\tau_{0,l}$ =0. The amplitude $|\alpha_{ml}|$ has a log-normal distribution and the phase $\angle\alpha_{ml}$ is chosen from {0, π} with equiprobability. The arrivals of each cluster and each ray in a cluster can be described as Poisson processes with rate and Λ and λ respectively. The inter arrival time between two clusters $T_{l+1}$ - $T_l$ or two rays within one cluster $\tau_{m+1,l}$ - $\tau_{m,l}$ is exponentially distributed. L and M denote the numbers of clusters and rays with in a cluster respectively. Finally, the Log-normal shadowing of the total multipath energy is modeled with X.

There are four indoor channel models namely CM1~CM4 for different channel characteristics. CM1 denotes the propagation environment with line of sight (LOS) and 0~4m propagation distance. CM2~CM4 denote three non-line of sight (NLOS) propagation environments with different propagation distance or delay spread (see [8] for more detail about the channel models and characteristics parameters).

After the transmitted signal is passed through the multipath channel, the multipath-impaired signal can be written as

$$y(t)=x(t)\otimes h(t) \tag{5}$$

where symbol ⊗ denotes convolution.

### 3.3 Receiver Module

The UWB receiver is shown in the right part of Fig.1. At the receiver's input, the total received signal can be written as

$$r(t) = y(t) + n(t) \tag{6}$$

where n(t) denotes the zero-mean additive white Gaussian noise (AWGN).

The adaptive receiver can be regarded as a sampling filter followed by an adaptive filter. The sampling filter samples the total received signal r(t) at least as fast as the Nyquist rate and its output is fed to the adaptive filter's input. The adaptive filter is a finite-impulse response(FIR) filter that essentially acts as a linear equalizer with MMSE criterion which minimizes the mean square error(MSE) no matter what type of noise may be present.

The judge module in the receiver module can transform the training number in the training phase and change the running taps number in the transmission. The process is introduced in the part II. During each bit decision period, a bit decision b ˆ(t) is made at the output of the adaptive filter and is then fed back to the adaptive filter to compute MSE. In order to capture enough multipath energy, the observation window of the adaptive filter is typically longer than one bit interval and therefore, windows overlap in time. Some adaptive filter algorithms, such as LMS, RLS and so on, can be applied to perform updating the tap coefficients of the filter.

For all adaptive algorithms, during the transmission phase, the analytic BER for single-user at the output of the adaptive filter can be given by the following expression [6]:

$$P_{su} = Q\left(\sqrt{\frac{w_o^H E_s}{|w_o|\delta^2}}\right) \tag{7}$$

, where $w_o$ denotes the optimal solution of the tap coefficients. $E_S$ represents the $N_w$ samples of multipath-impaired signal, y(t) which is dropped into the observation window and is determined by the UWB user's spreading waveform and the impulse response of UWB channel. $\delta^2$ denotes the power spectral density of the AWGN and Q(.) denotes the unit Gaussian tail function. Equation (7) shows that the BER is only determined by the tap coefficients for the given UWB signal power and the noise power. Therefore, so long as the output MSE of different adaptive algorithms is comparable, they can show the same BER performance.

## Ⅳ. Simulation Results

In order to evaluate the behavior of the proposed structure described in the previous section, we have simulated a single-user DS-UWB system with the proposed transmit-receive architecture. All the simulations in this paper were conducted under the following assumptions:

- The user's uncoded data rate $R_b$ is 200Mbits/s.
- The modulation format is Binary Phase Shift Keying (BPSK) and the length of one packet during transmission is 1000bits.
- The transmitted UWB pulse $w_{tr}(t)$ is the second derivative of Gaussian pulse with the duration $T_p$ of 0.333 ns. So the pulse rate or chip rate is equal to 3GHz
- A Gold sequence with a period of 31 is used as the spreading code and the spreading factor is 15, namely each information bit consists of 15 UWB pulses.
- The adaptive filter is sampled at twice of chip rate namely 6GHz (as fast as Nyquist rate) and the observation window width is taken to be the duration of two received symbols' interval.
- The sampling interval of multipath channel is 0.167 ns and we assume the channel is stationary during the transmission phase.
- The distortion from the antenna and nonlinear hardware are neglected.

There are four kinds of channel models, CM1, CM2, CM3, and CM4. CM1 denotes the propagation environment with line of sight (LOS) and 0~4m propagation distance and CM2~CM4 denote three non-line of sight (NLOS) propagation environments with different propagation distance or delay spread. The Bit Error Rate (BER) performance was simulated under all of four channel models.

In the BER simulations, 1Mbits (1000 packets) had been transmitted, and then the average of the 100 realizations' BER is taken as the BER for the SNR.

Fig. 4. presents the convergence profiles of the two types of adaptive algorithm LMS and RLS. It shows RLS have the better convergence performance than LMS. The MSE of the receiver with RLS converges more quickly than the receiver with LMS at the beginning. Later, they converge at the MSE level finally and on the steady level. In the conventional equalizer scheme, the length of training sequence is fixed and too long (more 500 training bits) [6]. Whereas the proposed equalizer scheme can be duly switched from training phase to transmission state as soon as the Judge module decides the arrival of convergence state.

Fig. 5. shows the output BER of the conventional equalizer scheme with different adaptive algorithm for all the four channel models. We can see that the BER performance on CM1 is the best among the four channel models owing to the presence of LOS and the BER on CM4 is the worst because of the extreme NLOS. After arriving at the convergence with the same MSE level, they show

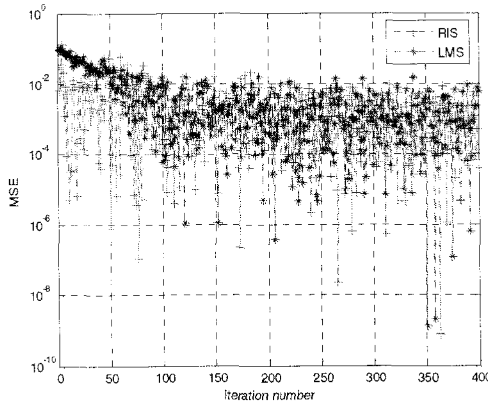


Fig.4. MSE convergence profiles for receiver with different adaptive algorithm

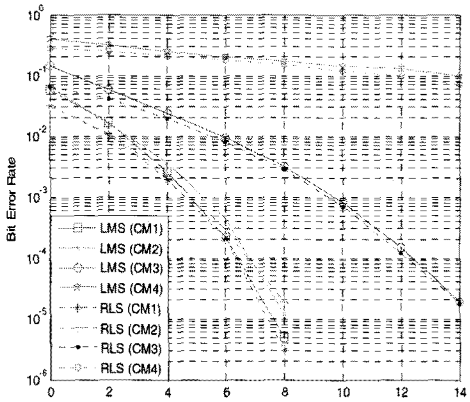


Fig.5. BER comparison for the conventional structurewith different adaptive algorithm.

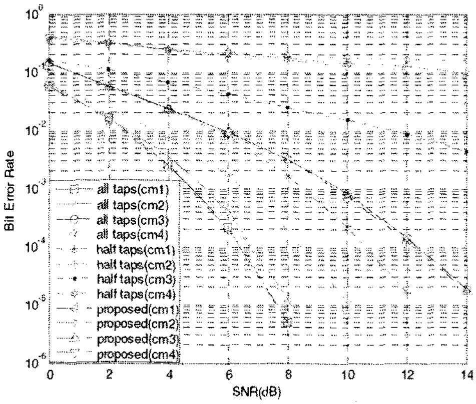


Fig. 6. BER comparison for the conventional and the proposed structure

similar BER curves. And this result is consistent with the analysis of equation (7).

Fig.6. presents the BER comparison between the transceiver scheme employing the conventional equalizer with all taps, the conventional equalizer with half taps and the proposed equalizer with the variable taps according to the four channel models. We can see that for CM1, at BER=$10^{-3}$ , the conventional equalizer with all taps and the proposed equalizer give about a 4dB advantage relatively over the conventional equalizer with half taps. The proposed equalizer gets a better performance than the conventional equalizer with half taps, and close BER with the conventional equalizer with all taps on CM2~CM4. In the simulation, the proposed equalizer scheme can

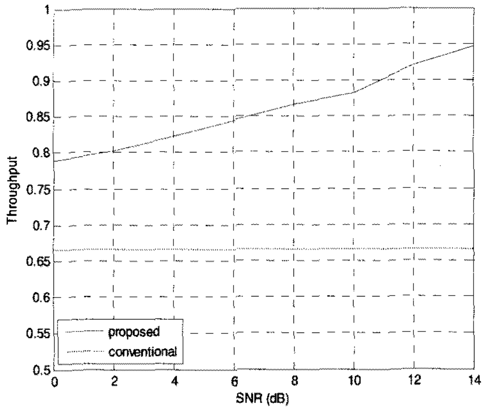


Fig.7. Throughput comparison for the conventional and the proposed structure according to the channel variation as $T < 0.001, Channel(CM3, CM1, CM2, CM2, CM4, CM1)$

exhibit a comparable BER performance compared with the conventional equalizer with all taps and better performance than the conventional equalizer with half taps. The advantage of the proposed scheme is to make it reduced the overhead during the training phase, which can increase the throughput as shown in Fig.7. And owing to the reduced number of taps, this scheme can reduce the computation complexity without inducing loss of the BER performance.

## V. Conclusions

In this paper, a transceiver scheme which has the structure with the variable length of training sequence according to CM1,2,3,4 in the DS-UWB systems was proposed. The proposed scheme is capable of improving the system's throughput with a novel equalizer. The proposed scheme can reduce the overhead during the training phase and reduce the computation complexity without inducing loss of the BER performance. Theoretical analysis and simulation results show that, for its low complexity and high throughput without loss of the BER performance, can be considered as a promising candidate for low cost, low power consumption and low complexity for indoor UWB application scenarios.

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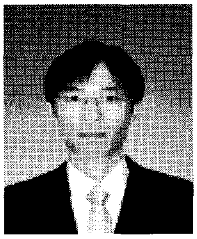

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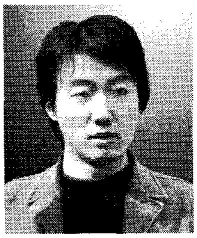

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