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AN OVERVIEW OF CHANNEL ESTIMATION APPROACHES FOR UWB IMPULSE RADIO

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***Abstract*—Aside from being a very hot research topic from some time, Ultra Wideband (UWB) impulse radio is quickly becoming a reality due to the standardization efforts leading by the IEEE 802.15.3a working group for wireless personal area networks. However, the issue of increasing the receiver performance is still open due to the tough engineering challenges that face its design. Among all the problems that are present in receiver design, two arise as the most difficult to address with a moderate cost-complexity solution: synchronization and channel estimation using extremely narrow pulses in the order of nanoseconds. Timing requirements are rigorous because even small misalignments render symbol detection impossible. Also channel estimation becomes critical for coherent detection but the large number of parameters together with the formidable sample rates required, motivates the search for alternatives to the traditional methods developed for narrowband and spread-spectrum methods. This article focus on channel estimation and it intends to review what are the challenges and some of the approaches that have been proposed to tackle the issue. The work will be further complemented with a report of the topics covered by the NEWCOM summer school on Estimation theory for wireless communications to be held at the Ecole Nationale Supérieure des Télécommunications, Paris, France in October 2005.**

***Index Terms*—ultra wideband communications, UWB, IEEE 802.15, channel estimation, wireless personal area networks, WPAN.**

I. INTRODUCTION

THE usage of ultra-wideband (UWB) impulse radio technology for civil and commercial applications is leaving research labs to become soon a commercial reality, aside of radar and military fields were it has been widely used for many years. A clear signal comes at no surprise from consulting companies (e.g. Gartner) whose clients are generally the CEOs and CTOs of the largest companies of the world, which are starting to talk widely about this technology, and projecting the benefits that the companies entering early in the UWB market could harvest.

Despite its natural application for wireless personal area networks (WPAN), UWB technology is also envisioned as an enabler for the massive deployment of sensor networks, consisting of a large number of networks nodes that are usually thought as being spread over an ample geographical area. The requirements for sensor networks nodes include low cost and power where in this particular case energy provisioning is more limited than in WPANs because of the fact that it is usually more difficult to recharge the batteries of a large number of sparsely distributed nodes. For example, comparing UWB with Bluetooth, the energy requirements and the higher cost of the latter makes the former a better candidate for this kind of application [1].

Medical diagnostics, surveillance, ocean underground exploration, and even radar imaging systems can benefit from the pulses of UWB radios which are generally shorter that the dimensions of the object under study and thus offer higher resolution and pronounced sensitivity to scattering. Also the precise localization capabilities potentially offered by UWB radios, improves positioning accuracy, leveraging its future usage for vehicular radar, collision-avoidance and cruise control systems. The final stimulus was given by the United States (U. S.) Federal Communications Commission (FCC) when they open bandwidth between 3.6 to 10.1 GHz, where devices using UWB transceivers are allowed to operate almost at noise floor, coexisting with current RF systems and using low-power and ultra-short pulses to signal the information. Although the U. S. is pioneering this movement similar regulatory processes are currently being carried out by other countries worldwide. This new huge spectrum availability along with its tight exploitation rules poses over UWB radios stringent requirements in almost every component design, something that is even more tough adding the fact that commercially massive production of UWB radios calls for a low cost and preferably simple solution, a big challenge that has not been faced in deployments thought for military utilization. Fortunately, the low cost and high performance of today's digital signal processors (DSP) come as a great help, but even though, power always comes at the cost of battery energy, something that still imposes a severe limit on the complexity of problems that can be tackled with mobile devices.

Among all the issues that involve the development of efficient and low cost UWB radios, this article focuses on channel

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estimation. The estimation of the parameters that characterizes the channel is of paramount importance to increase the performance of UWB coherent detectors. This technique has been successfully used in other transmission schemes, such as DS-CDMA, and although there were efforts to adapt them to solve UWB estimation, the formidable higher sampling rates in the latter call for different approaches.
The rest of article is organized as follows: section II, describes concept and the main problems behind the idea of channel estimation; section III, IV and V concentrate on some methods that are being explored along with their achieved results, finally the appendix will include a report of the topics covered in the NEWCOM summer school on Estimation theory for wireless communications to be held at the Ecole Nationale Supérieure des Télécommunications, Paris, France in October 2005.

II. THE CHANNEL ESTIMATION PROBLEM

A. *General Estimation Concepts*

A conceptual communication model is depicted in Fig.1. The received waveform y not only depends on the transmitted symbols x but also on a set of parameters related to the transmitter (θ_T) and to the channel (θ_C). They are unknown to the receiver and in order to be able to retrieve the symbol sequence x, it must estimate these parameters which can be considered unwanted since they somehow corrupt the signal that transport the sequence x. Once estimated, the parameters are then used as if they were true values. The block diagram of Fig. 2 explicitly shows the estimation process conceptualizing it via a *Channel Estimator* block. In the simplest case of an additive noise channel the parameters $\theta = \{\theta_T, \theta_C\}$ may be assumed constant but unknown. The parameter set θ includes, for example, the phase φ or the fractional time delay ε. In this case, the task of the channel estimator of Fig. 2 is to estimate a set of unknown parameters in a known signal corrupted by noise. Then the parameter adjustment can be done by shifting the phase $\varphi(t)$ of a voltage-controlled oscillator such that the estimation error $\phi(t) = \varphi(t) - \hat{\varphi}(t)$ is minimized.

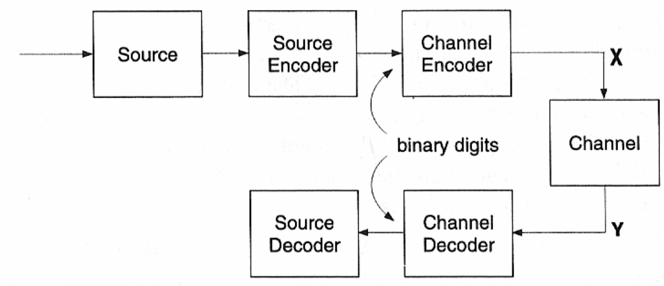

Fig. 1. A conceptual communications system model as it is usually conceived in information theory. [3]

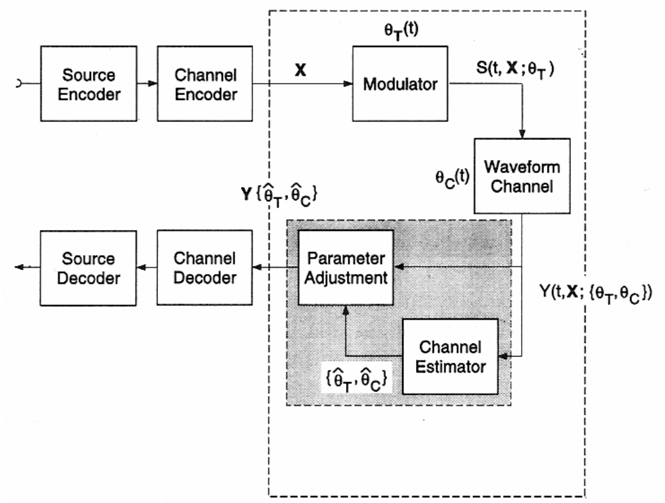

Fig. 2. Physical Communication Model. The receiver comprises a part to estimate the channel parameters and another for decoding the data. The dashed rectangle encloses a block which is characterized by a probability p(y/x). The shaded area has the task to produce an output such that the performance of the channel decoder + source decoder is as close as possible to the ideal of perfect channel knowledge. [3]

However, the channel model discussed above is not applicable to mobile communications, because in the latter case the channel varies with time. In a time-varying channel, the task of the channel estimator consists of estimating a time-variant set of parameters, and since this variation is random, mathematically that means one has to solve the problem of estimating a random signal in a noisy environment, where this random signal is in fact, the channel impulse response (CIR). Therefore, the parameter adjustment block has to perform more complex tasks such as the calculation of the taps of a matched filter or an equalizer, that is to say, the computation of channel-dependent parameters.

Recalling that, in general, the received signal $y(t, x; \theta)$ contains two random sequences: the useful data x and the randomly varying channel parameters θ ; Fig. 2 suggests that it is possible to separate the estimation of unwanted parameters θ and the estimation of the useful data sequence x. Unfortunately this is not the case and in principle both must be jointly estimated, which is a very complex task when not aided by some auxiliary procedures as it will be shown later.

B. *Estimation particularities of UWB signals*

Briefly the main characteristics of UWB systems can be summarized as follows:

- Very short pulses (≈1ns)
- Multipath channel
- Time hopping pulse trains
- Very large bandwith (≥ 500MHz)
- Baseband transmission of pulse trains

The multipath environment in which UWB devices are expected to work on, calls for the utilization of sophisticated receivers that can take advantage of this characteristic. There are two designs that can do the job: one known as multiuser detection [2], which is an optimal solution whose complexity increases with the number of users something that renders it impractical for the usages that are being envisioned for UWB. The other possible approach is the utilization of the well known Rake receiver which, in spite of being a suboptimal approach, it offers a good tradeoff between high-performance and low-complexity, something that was explicitly mentioned earlier as one of the requirements of UWB technology.

In a Rake receiver, also known as correlation receiver, each received signal, i.e. including echoes, is correlated with a local replica of the transmitted pulse train which is then combined into a single test variable for a final decision. Fig. 3 depicts such as system. Each correlator accounts for a multipath component. The number L of multipath components taken into consideration will depend on the channel model adopted. In order to successfully detect each echo, the local replicas have to be appropriate delayed. Also, each of the attenuations $\hat{\gamma}_l$ incurred by the various echoes must be known to maximize the signal-to-noise ratio (SNR) at the decision point (Z'). Therefore, it can be easily concluded that in order to have a successful Rake receiver it is necessary to have the best possible delay and attenuation approximations of the channel. The delay and the attenuation jointly constitute the channel parameters that need to be estimated. As it will be showed there are several techniques to approach the problem, and the purpose of the present article is to overview some of them.

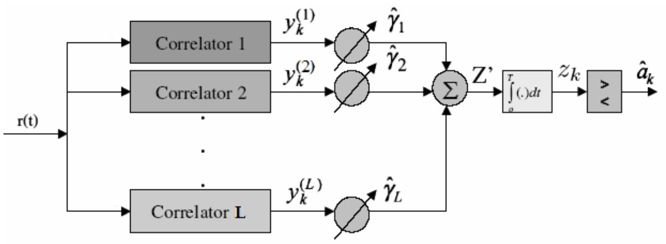

Fig. 3. Block diagram of a Rake receiver composed of L correlators usually called Rake-L receiver. Each correlator detects a time-shifted version of the original transmission.

$r(t)$: Composite waveform at the output of the receiver antenna

${y_k}^{(l)}$: Output of the l correlator for the k^{th} symbol. $l:1..L$

$\hat{\gamma}_l$: Weighting coefficients ($l:1..L$). If the power or SNR is small out of a particular correlator, it will be assigned a small weighting factor.

$Z' = \sum_{l=1}^{L} \hat{\gamma}_l \cdot {y_k}^{(l)}$ if the maximal-ratio combining is used.

Fig. 4 shows the indoor channel model (CIR) adopted by the IEEE 802.15.3a channel modeling sub-committee. Multipath components arrive at the receiver in groups (clusters). The number of clusters and multipath components may theoretically extend over infinite time, but since they exhibit exponentially decaying power profile, they practically vanish after some time. However, the number of multipath components is 315, distributed over a delay spread of about 50 ns. Recalling that the receiver has to estimate attenuation and delay for each multipath component, such a large number of them, formidable increases the complexity of the channel estimation, far beyond those developed for DS-CDMA. Most of the estimation methods developed so far increase their computational complexity as the number of multipath compents increase, rendering them practically unfeasible for realistic UWB application, and this poses one of the major challenges that are still being researched

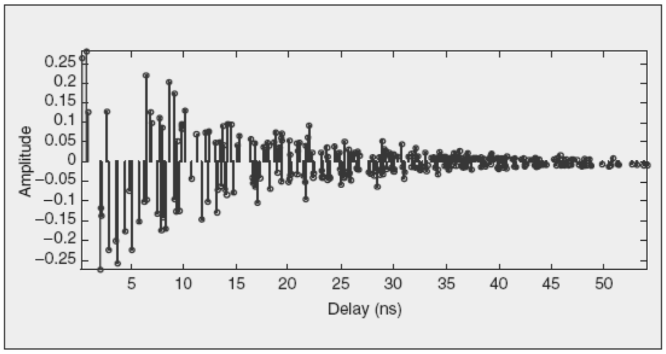

Fig. 4. UWB indoor NLOS channel impulse response model with 0-4 m transmitter-receiver distance, standardized by the IEEE P802.15-02/368r5-SG3a working group for WPAN in November 2002. It is basically a modified version of the well-known Saleh-Valenzuela (S-V) indoor channel model that was established back in 1987, based on measurements utilizing low power ultra-short pulses (of width 10 ns and center frequency 1.5 GHz) in a medium-size, two-story office building. [1]

III. MAXIMUM-LIKELIHOOD APPROACH

Using the maximum-likelihood (ML) criterion, the derived optimum receiver structure is the outcome of a mathematical optimization problem. Reference [2] is a good example of this kind of approach, where the impulse response is estimated either with the aid of training symbols (data-aided), or without them using just unknown information-bearing symbols (nondata-aided). To render the manipulations simpler, multiuser interference (MUI) is treated as white Gaussian noise under the assumption that the central limit theorem holds if there are many users that transmit at similar powers.

A. *Data-Aided Estimation*

The user's pulse position modulated baseband UWB signal can be modeled as [2]:

$$s(t) = \sum_i b(t - iNT_f - a_i\Delta)$$

where

$$b(t) = \sum_{j=0}^{N-1} g(t - jT_f - c_jT_c)$$

$g(t)$: Information bearing pulse. Usually a Gaussian monocycle (first derivative of a Gaussian pulse).

T_f : Frame duration.

T_c : Duration of an addressable time bin. (chip)

$\{c_j\}$: User's specific time-hopping (TH) code sequence. Integers taking values in the range $0 \le c_j \le N_h - 1$; being N_h the length of the TH code.

N : Number of frames in which a single symbol is spread (number of frames/symbol). In UWB transmission, multiple frames comprise a symbol. Therefore a symbol has a duration $T_s = NT_f$.

$b(t)$: Block of monocycles with period $T_s = NT_f$.

a_i : Data symbols at time i modeled as binary (0 or 1) independent and identically distributed (i.i.d.) random variables.

Δ : Time shift impressed by a unit data symbol on the monocycles of a block. Modulation index.

Summarizing, the transmitted signal consists of a sequence of $b(t)$ -shaped position-modulated blocks.

Once several TH signals are simultaneously transmitted over a channel with L_c multipath components, the compound received signal can be written as [2]:

$$r(t) = \sum_{l=1}^{L_c} \gamma_l s(t - \tau_l) + w(t)$$

where

$s(t)$: Desired user's signal.

γ_l : Attenuation affecting the l^{th} path signal replica.

τ_l : Delay affecting the l^{th} path signal replica

$w(t)$: Thermal noise + multiuser interference.

Assuming the number of paths L_c are known, the parameters $\{\gamma_l\}$ and $\{\tau_l\}$ are not known a priori and must be estimated to maximize the SNR at the decision point. Applying the ML criterion and stating the log-likelihood function of the pair (γ, τ), after some manipulations, [2] arrives to the expression:

$$\log[\Lambda(\tilde{\gamma}, \tilde{\tau})] = 2\sum_{l=1}^{L_c} \tilde{\gamma}_l \sum_{k=0}^{M-1} z_k(\tilde{\tau}_l, a_k) - ME_b \sum_{l=1}^{L_c} \tilde{\gamma}_l^{\,2} \tag{1}$$

where

$\tilde{\gamma}$: Trial value of γ

$\tilde{\tau}$: Trial value of τ

M : Number of pilot symbols of the training sequence. Large enough to guarantee adequate estimation accuracy.

$z_k(\tilde{\tau}_l, a_k)$: Response of the matched filter at $t = kNT_f + \Delta a_k + \tilde{\tau}_l$

Eb : Energy of $b(t)$

Since the $\{a_k\}$ are part of the training sequence that are known by both transmitter and receiver, $z_k(\tilde{\tau}_l, a_k)$ contains sufficient statistics (i.e. all information of the received signal), thus (1) is ready to compute the ML estimates of (γ, τ). Therefore, the last step is to maximize $\log[\Lambda(\tilde{\gamma}, \tilde{\tau})]$ as a function of $\tilde{\gamma}$ and $\tilde{\tau}$, to obtain the estimators $\hat{\gamma}_l$ and $\hat{\tau}_l$.

Fig. 5 shows a possible arrangement where the bit stream is organized in frames each composed of training sequences (shown shaded) and useful data information. Fig. 5 shows a variable preamble, whereas in practice it is generally composed of a fixed number of M pilot symbols followed by useful data. Using the known "pilot" symbols the channel parameters are estimated. Conversely, during the transmission of data symbols, the channel parameters are assumed to be known and the information bits are decoded. Both [2] and [3] agree to notice that the frame structure can be matched to the channel characteristics to meet two conflicting requirements: efficiency of the channel use and accuracy of the estimate.

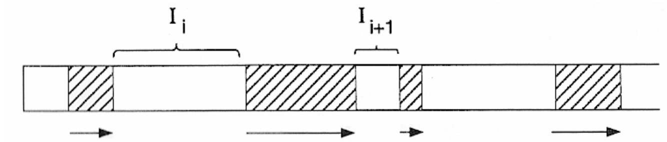

Fig. 5. Frame structure. Shaded areas are the pilot symbols used for data-aided channel estimation. I_i is the information [3].

B. Non Data-Aided Estimation

Non data-aided estimation (NDA) is useful in broadcast networks where training sequences would hinder the transmitter when new users enter the network. In this case everything remains the same but for the fact that now the symbols are completely unknown and therefore they can be considered unwanted. Therefore the idea is to produce a marginal likelihood function that depends only of γ, τ , by averaging the likelihood function $\Lambda(\tilde{a}, \tilde{\gamma}, \tilde{\tau})$ over the probability density of $\tilde{a}$ [2], an by assuming a low SNR. The resulting function $\Lambda(\tilde{\gamma}, \tilde{\tau})$ is quite similar to (1) and therefore the same maximization method as before can be applied.

C. Performance Comparison

The performance of both methods have been assessed in [2] finding that DA is superior to NDA as expected. They found that using DA, up to 20 users can be accommodated with limited BER degradations, whereas the performance of NDA is rather poor and could only accommodate about 10 users. This might have prove the success of DA estimation, however the computational complexity of the method increases as the number of multipath components increases, and becomes unaffordable for the large number of multipaths such as those considered in the UWB indoor channel model of Fig. 4. Additionally, the assumption of subpulse sampling rates in the order of 12.5 to 25 samples per monocycle yields a staggering sampling range between 17.9 ~ 35.7 GHz which requires additional hardware and accurate timing control, thus increasing the overall cost of the system.

IV. LEAST SQUARES ESTIMATION APPROACH

This approach is also based on the outcome of a mathematical optimization problem, but in this case as a result of the application of a least square (LS) fit. The idea is to minimize the squared Euclidean distance between the samples of the received signal with a noiseless sampled version of the CIR which is assumably known. Reference [4] and [5] provide a good example of this kind of approach. Reference [5] proposes a least-squares method and a subspace technique which finds channel sparseness and then exploits this structure for final channel estimation. The authors also showed symbol and frame synchronization using least squares methods. They assume that the delays values are equally spaced claming the advantage that once the propagation delay along the line of sight (LOS) has been estimated, the associated channel coefficients can be estimated in closed form since they appear linearly in the received signal model. Among the possible risks is the fact that being the delays of the UWB organized in clusters, this can lead to over-parameterization with the estimation of many zero-valued taps, nevertheless in [4] the authors proposes a possible solution to this problem.

Towards the application of the LS method, [5] starts supposing the channel impulse response structure and number of effectively nonzero paths, L are known. After some extra assumptions and some mathematical manipulations the Euclidean distance between the received signal samples and the estimated CIR trials is given by:

$$D^2(\tilde{\mu},\tilde{h}) = \sum_{m=0}^{M_f-1}\sum_{l=0}^{L-1}\left[x_{m,|l+\tilde{\mu}|_N} - \tilde{h}[l]\right]^2 \qquad (2)$$

where,

$\tilde{\mu}$: Timing parameter associated with the start of individual frame trials, being $\mu = \lfloor \tau_{LOS} / t_s \rfloor$ the integer part of the propagation delay along the LOS τ_{LOS} .

$\tilde{h} = \left[\tilde{h}(0), \tilde{h}(1), \ldots, \tilde{h}(L-1)\right]^T$: Trial vector with the sampled version of the CIR.

$N_f = T_f / t_s$ is the integer number of samples per frame taken at t_s [ns]

$x_{m,|l+\tilde{\mu}|_{N_f}}$: Vector containing m samples of the received signal $x_m = [x_{m,0}, x_{m,1}, \ldots, x_{m,N_f-1}]$, when observed over M_f frames $m = 0,1,\ldots,M_f - 1$

The minimization is then performed operating on $\tilde{h}$ and $\tilde{\mu}$ iteratively, first by fixing $\tilde{\mu}$ to find the minimum over $\tilde{h}$. Then minimum is substituted into (2) to obtain $\hat{\mu}$. Finally the CIR $\hat{h}[l]$ is estimated setting $\tilde{\mu} = \hat{\mu}$ in the expression of $\tilde{h}$.

As a conclusion, it is easy to see that the computational complexity in terms of flops of the estimation algorithm proposed in [5] is moderate since the approach has a straightforward implementation because it basically consists of an arithmetic average over the received samples. Nevertheless, in addition to over-sampling in the order of GHz, this approach requires assumptions which include the knowledge of the channel impulse response structure along with the number of paths, and that the time of the first arrival τ_0 , as well as the difference between the first and the last arrival $\tau_L - \tau_0$ have to be less than the frame duration T_f ; something that can be considered somewhat restrictive.

V. ESTIMATION OF THE AGGREGATE ANALOG CHANNEL

The two methods described before are computationally complex, the first probably more than the second but even though, both require exhaustive computations. Another thing that characterize both approaches is the high sampling rate they need to reach the desired accuracy, e.g. [2] explicitly mentions significant degradation of their technique below 9 GHz, which is still high but even though, not enough.

To avoid these problems one possibility is to estimate the aggregate analog channel $y(t) = (g * h)(t)$, following the idea of the training sequences mentioned in section III *A*, but implemented in a slightly different way. The approach belongs to those termed Transmitted-Reference (TR) signaling, in which information conveying pulses are transmitted together with unmodulated, also known as pilot pulses to assist the process of estimation. Instead of coupling each information pulse with a pilot, something that could reduce the performance in 50%, the training pulses are inserted for every UWB symbol, i.e. with a period $T_s = NT_f$, as shown in Fig. 6 [1]. The received pilot pulses are averaged, and used as the correlator template to decode the received information bearing pulses, improving template matching performance. The method is designed so as to minimize the channel's mean-square estimation error while at the same time maximizes its average capacity, thus taking into account performance considerations as well as bandwidth efficiency. They achieve that by optimizing the placement, the number and the energy allocation of pilot waveform pulses. These parameters determine not only the channel estimation and symbol demodulation performance but also the effective data rate. The nice thing about this technique is that it can use simple integrate-and-dumb demodulation at a frame rate, i.e. ≈ 10 MHz considering with a typical frame of T_f = 100 ns, thus solving one of the problems stated at the beginning. Reference [6] shows that PWAM is applicable to both pulse amplitude modulation (PAM) as well as pulse position modulation (PPM), which is the model that has been used here.

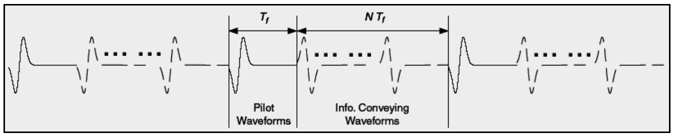

Fig. 6. In Pilot Waveform Assisted Modulation (PWAM), pilot pulses are inserted for every UWB symbol in order to facilitate and increase the accuracy of channel estimation parameters [1].

Finally, Fig. 7 shows what the possible approaches for the placement of pilots are. Regarding bandwidth efficiency, the usage of a training sequence (preamble) has also the same performance as PWAM, but the fact that pilot waveforms are distributed throughout each burst, makes the latter design more immune to slow channel variation. Since PWAM allocates the number of pilot pulses according to channel coherence time, when the channel varies rapidly, it trades in information rate for improved channel estimation performance. Comparing this behavior with that of traditional TR systems it comes out that the PWAN scheme provides more flexibility, since it can adapt to the conditions of the channel whereas TR is inherently more robust but at the cost

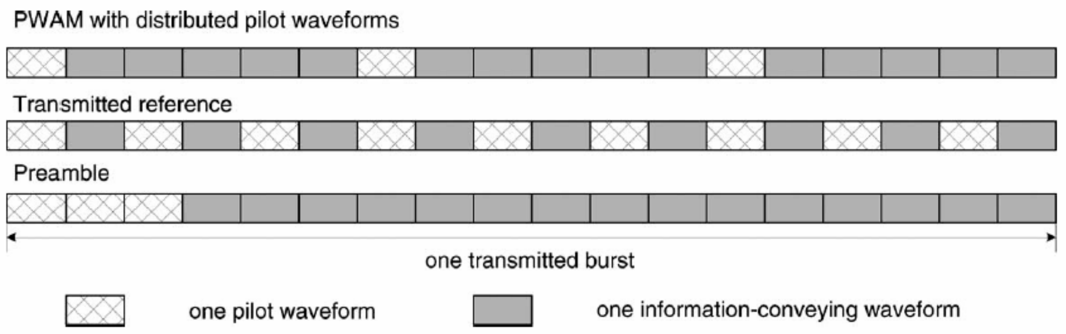

Fig. 7. Comparison of pilot distribution in for different schemes: PWAM, transmitted reference (TR), preamble. Different placements benefit different aspects of the overall system design. Inserting the pilot waveforms equi-spaced throughout a burst is beneficial especially when the channel is varying slowly over time, while gathering all pilot waveforms at the beginning of each burst like in preamble, will yield the smallest delay in symbol decoding [6].

of sacrificing information rate when the channel does not vary over a group of symbols. The estimation robustness of the system was assessed thru several simulations that confirmed its capabilities. The analysis was carried out assuming a single-user link over a quasi-static multipath channel, which can be rendered equivalent to a multiple user setting just by choosing the appropriate user-specific spreading codes. In the presence of timing offset, the BER performance degradations were significant when a single pilot waveform was used, but negligible when 25 pilot waveforms were averaged.

VI. CONCLUSION

The transmission of information using UWB technology involves ultrashort pulses, and the UWB channel is rich in multipath diversity. Rake receivers can resolve may paths but collecting them requires a large number of fingers. Being a coherent technique, Rake reception relies on channel knowledge for efficient detection. For UWB this knowledge comes thru the utilization of a parameterized model of the indoor channel. The number of multipath components are given by the model but their attenuations and delays has to be estimated. This article summarizes the main techniques that are being proposed to tackle this problem along with a description of their performance as well as their advantages and disadvantages.

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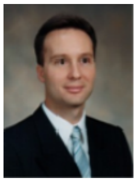

Carlos G. Bilich was born in Santa Fe, Argentina in 1971. He graduated as Electro-Mechanical technician from the Superior Industrial School in 1990. Then he moved to Rosario to study Electronic Engineering at the National University of Rosario, and graduated in 1997. He was working in industry even before graduation as a network system engineer at Carrefour S.A. Soon after his graduation he was appointed as an automation engineer at Techint Corporation for his flat steel division in Argentina, named SIDERAR, where Mr. Bilich implemented several control models for the electric furnaces and water cooling lines of the hot steel rolling mill. In 1999 he was recipient of a Fulbright scholarship award to do his master degree in Telecommunications at the University of Pittsburgh, USA. In 2001 he was back at SIDERAR as a semi-senior network engineer of the systems department. He was involved in performance measuring and tuning of the company intranet as well as working closely with software developers to aid in the design of distributed applications that can meet the stringent performance standards of the company. By that time he was also chief architect of the automatic measuring system that was developed to monitor the performance of the J2EE and Microsoft distributed applications developed in SIDERAR. Recently, in 2004 he won a scholarship to pursue a Ph. D. in computer networks and mobility from the University of Trento, Italy. While doing his Ph. D. Mr. Bilich also works as a junior scientist at the Center for REsearch And Telecommunication Experimentation for NETworked communities, CREATE-NET; an International Research Center founded by a group of leading Universities and Research Centers of Europe and America, leading by the renowned Professor Dr. Imrich Chlamtac. Mr. Bilich primary research interests include: Pervasive computing technologies for healthcare, wireless networks, distributed applications and breakthrough technologies for all packet switching photonic networks.