Semi-blind estimation method in DMT-based Transmission on Indoor Power Line

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Keywords: Channel estimation, DMT, semi-blind estimation, pilot carriers.

Abstract:
This paper deals with the optimization and the performances evaluation of multi carrier transmissions on indoor power lines, using semi-blind channel estimation. Various transmission schemes are proposed to mitigate the effects of the stationary noise, the channel selectivity and the sudden change of the channel transfer function. These different techniques are optimized for a transmission on an indoor power line and their performances, expressed in terms of bit error rate, are compared.

1. Introduction

One of the main features of the radio channel using a low voltage power line network as a physical link between the transmitter and the receiver is the presence of important noise. To cope with the effect of impulsive noise, various techniques have been proposed [1], [2]. Among them, it is well-known that the DMT modulation scheme offers an important robustness in the presence of a narrowband noise induced on the power lines and which are mainly due to broadcast transmitters.

As the channel is frequency selective, a simple one-tap equalization technique is currently used, but the difficulty lies in the fact of finding the best estimation of the complex channel transfer function in the transmission bandwidth. In [3], [4], authors proposed a method based on pilot symbols assisted modulation (PSAM). Nevertheless, since the channel can suddenly change when an electrical appliance is connected or disconnected to the power network, the number of pilot symbols must be large enough to rapidly detect a sudden change of the channel characteristics. Another solution consists of using a differential coding, but its performance decreases rapidly with the increase of the size M of the constellation.

In this paper, we propose new adaptive channel estimation based on a semi-blind algorithm which immediately detects any change of the channel and can thus continuously estimate the channel transfer function. To ensure a good trade off between performance and complexity, the proposed solution is only dedicated for $M < 64$.

The first part of this paper describes a semi-blind channel estimation for an OFDM transmission technique. An example of its performance in a time-varying frequency selective channel is presented and the results are compared with those given by well-known estimation algorithms.

In a second part, we describe the main steps needed to adapt the proposed method to the DMT process and few examples will show the advantage of using this new way of estimating the channel characteristics.

2. Semi-blind channel estimation for an OFDM technique

In this part we briefly recall the block diagram of an OFDM transmission link and the model used for simulating the channel. Then the semi-blind channel estimation is presented and applied to examples for testing the performance of this new algorithm in terms of the number of erroneous bits.

2.1. Description of the OFDM process

The OFDM process described in [5] is one of the basic modulation scheme proposed for PLT and its block diagram is shown in Figure 1. Its main feature is to transmit the information in parallel with N subcarriers owing to an IFFT. The major advantage is to cope with the frequency selectivity of the channel by dividing the available bandwidth into N equal subbands in which a flat channel can be assumed, allowing the use of a simple equalization algorithm using, for example, the zero-forcing criterion. Furthermore, a cyclic prefix of 32 samples is added for improving the immunity against inter symbol interference due to multipath propagation. We have considered typical values of 256 subcarriers, a 4-QAM (4 Quadrature Amplitude Modulation) modulation, and a bit rate of 10 Mbit/s, leading to a transmission frequency band width of 5.5 MHz.

The transfer function is modeled by a tap delay line filter. The complex multiplication factors are given by the amplitudes of each “tap” of the impulse response, assuming that the phase angles of these taps are uniformly
distributed. The amplitudes are determined from a statistical analysis [6] based on an experimental approach carried out in a 1 MHz - 30 MHz frequency band, and on a specific installation. It comprises a line, few ten meters long, with a large number of power outlets in which various appliances can be plugged and unplugged. The additive noise, which has been measured in this frequency band and which always been used through this paper, is a superposition of an AWGN having a power spectral density (PSD) of -140 dBm/Hz and of a narrow band noise corresponding to various "peaks" reaching -110 dBm/Hz due to broadcast transmitters [7].

![Figure 1: OFDM transmission schema](image)

### 2.2. Semi-blind channel estimation

The principle of a blind estimation [8] is based on a particular statistical characteristic of symbols D of the M-QAM constellations. If we consider the set of M symbols \( \{D\}_1:M \) available in the M-QAM constellation, one can find an integer “\( J_M \)” which satisfies the following relation:

\[
E[D] = \begin{cases} 
\frac{\alpha_M}{M} & j \in [1, J_M - 1] \\
J_M & j = J_M 
\end{cases}
\]  

(1)

In this formula, \( \alpha_M \) is a coefficient which can be expressed in terms of the expected value of all symbols raised to the power \( J_M \) and divided by M. Note that \( \alpha_M \) is specific for each M-QAM modulation. For example, for 4-QAM, \( \alpha_4 = -4 \) and \( J_M=4 \).

Assuming that the channel is time invariant during a frame, the channel estimation \( \hat{H} \) of the frequency response \( H \), can be related to the received symbol \( Y_{t,k} \) in frame \( t \), and for the sub-carrier \( k \), when T frames are used for this estimation, by the following expression:

\[
\hat{H}_{k}^{J_M} = \frac{M}{\alpha_M} \frac{1}{T} \sum_{t=1}^{T} Y_{t,k}^{J_M} H_{k}^{J_M}
\]  

(2)

The amplitude of \( \hat{H} \) is easily deduced from eq (2). However, since the estimate \( \hat{H} \) is raised to the power \( J_M \), there is an uncertainty on the phase value.

To remove this ambiguity, we suggest estimating the phase angle by introducing a single pilot tone per frame, and by periodically changing the subcarrier, for example every two frames. Owing to a linear interpolation of the phase variation between two pilot tones and by comparing, at a given frequency, this interpolated phase angle to the \( J_M \) possible values deduced from (2), the most probable phase angle, among the \( J_M \) values, can be determined. In the following, such an approach will be called a "semi-blind" method.

The remaining problem to be solved is related to the sudden change of the propagation channel during the T frames used for estimating the transfer function. If such a change occurs, this will lead to an error burst, its maximum duration being in the order of the duration of the T frames. To minimize the number of errors, it is interesting to detect as soon as possible a change in the channel state. This can be achieved, as already proposed in [6], by calculating a variable \( \rho \), defined by eq. (3), and by detecting an important variation of its amplitude.

\[
\rho = \frac{\alpha_M}{M * N} \sum_{k=1}^{N} \left( \frac{Y_{t,k}}{\hat{H}_{k}} \right)^{J_M}
\]  

(3)

This variable depends on the received symbol \( Y_{t,k} \) and on the estimated channel transfer function. From this equation, it appears that \( \rho \) will vary suddenly if a channel modification occurs, the real and estimated values of the transfer function becoming very different. On contrary, if any channel modification occurs, the value of \( \rho \) remains nearly equal to 1.

### 2.3. Simulation results

In this part, the aforementioned semi-blind estimation algorithm and the conventional method such as PSAM methods or the recursive least square (RLS) algorithm are successively introduced in the OFDM link to compare their performance in a succession of different time invariant channels, corresponding to abrupt changes of the channel characteristics. The equalization is done by applying the zero forcing criterion. In each configuration, the number of pilot symbols is the same, so that the effective transmission rate remains constant.

For the PSAM method, we consider two conventional pilot insertion techniques, the first one being based on a filtered decision feedback estimator [3] using two training frames every 512 frames of information, while the other one uses sixteen pilot tones [4] associated with a low pass
filter in time domain to significantly reduce the noise in the received signal pilots. Another method proposed in [9] is the Recursive Least Square (RLS) algorithm for determining the transfer function $H_k$ of the channel, corresponding to the subcarrier of subscript $k$. From a preliminary study, we have seen that the optimum number of training frames is 4. We have thus kept this value in all simulations.

In the following examples, we assume that 1024 OFDM frames are transmitted, one frame containing 512 bits, and that the channel suddenly changes at time $T_0$ when sending the 262nd frame. This is simulated by generating a second channel impulse response in our model. Curves in Figure 2 show the evolution of the number of erroneous bits in each frame for the different methods. Curve (0) deduced from the RLS algorithm, shows that the number of errors reaches 350 when the channel transfer function suddenly changes and then remains nearly constant. Indeed, the RLS algorithm, during its adaptive phase, calculates its coefficients from the decision on previous symbols. But after a sudden channel variation, these coefficients are not suited to the new channel and the algorithm diverges. The result of the estimation based on the pilot frames (curve 1) shows that the number of errors is important in the transition zone and, since a new initialization frame is needed to adjust the estimation coefficients, this second method is not well-adapted for our environment too.

Figure 2: Number of erroneous bits for different methods of channel estimation, $E_b/No=14$ dB

Curve 3 corresponds to the performances of the semi-blind estimation, while curve 4 deals with the results of a method combining this semi-blind estimation with a detection of a change in the channel state. These curves clearly show the advantage of using this last technique. Nevertheless, it is important to mention that no error appears in curve (4) only because the change of the channel transfer function occurs just at the beginning of a new frame. In the general case, the frame sent during a change of the channel state, will be lost.

3. Semi-blind channel estimation applied to DMT technique

After briefly recalling the principles of the DMT technique, the different possibilities for applying an optimized version of the semi-blind estimation method previously proposed are described. The different solutions are compared with the conventional estimation methods in non-stationary channel.

3.1. Description of the DMT process

The principle of the DMT process is that the size of the subcarriers constellation is allocated as a function of the signal to noise ratio in each sub-band. For the bit allocation, the Fischer-Huber algorithm [10] is used, the principle being to transmit a predetermined number of bits per second (512 bits in our examples). Furthermore, the bit error probability of the transmission is minimized owing to a recursive stage in the process.

3.2. Extension of the semi-blind estimation

The extension of the semi-blind channel estimation to the DMT modulation is not straightforward. Indeed, as previously recalled, the binary allocation algorithm chooses the size of the constellation in each sub-band depending on the signal to noise ratio in this band and it may thus often happen that few subbands are unused.

The semi-blind estimation requiring pilot carriers equally spaced in the frequency domain, the pilot arrangement must be carefully chosen. Furthermore, one can easily imagine that using “good” subbands for the pilot tones, i.e. subbands having an excellent signal to noise ratio, a good estimation of the channel response will be obtained. However, the drawback of such a choice is an important decrease of the effective bit rate, since such “good” sub channels may support a large size constellation.

If a “group” is defined as a set of adjacent used sub carriers in the frequency domain, each group being thus separated by an unused sub band, the objective of the proposed Pilot Tone Selection (PTS) is to optimize the number and the distribution of the pilot tones depending on the size (number of carriers) of each group.

The different steps of the PTS algorithm are detailed in this paragraph. First of all, after allocating bits in each sub band with the Fischer-Huber algorithm, the different groups are identified. Each group requires at least one pilot tone, due to the principle of the semi-blind estimation. Then the pilot tones are distributed in the group...
proportionally to the number of subcarriers in each group. After the pilot allocation, the pilot position can be optimized. Two solutions have been tested: The pilot tones are chosen on the sub carriers having either the smallest or the largest size of constellation.

In both cases, the optimization of the pilot position is the same. The aim is that, in a group, the pilot tones are the most equally spaced as possible. To achieve this goal, an iterative calculation based on the distance between two pilots is done. An example of the choice of the PTS is shown in Figure 3 for the DMT process with 16 subcarriers and 4 pilot tones. The constellation size is given versus the number of subcarriers. The conventional algorithm with equally spaced pilots is compared to the PTS method.

3.3. Simulation results

To illustrate the semi-blind approach and the PTS algorithm, a parametric study has been performed on the estimation accuracy of the channel transfer function and its consequence on the bit error rate (BER), by introducing different criteria in the PTS algorithm. Let us consider the transmission of 512 kbits in 1000 different frequency selective channels, each of them suddenly and randomly changing one time. The upper part of Figure 4 represents the cumulative distribution of the bit error rate for three configurations. The curve noted “Uni.” has been deduced from a simulation assuming that the pilot tones are equally-spaced within the transmission bandwidth, while curves “PTS min size” and “PTS max size” correspond to an optimal choice of their distribution by selecting pilot sub carriers having either the smallest or the largest size of constellation, respectively. In this example, the number of frames \(T\) for estimating the channel response is 45 and the number of pilot tones \(P\) is equal to 9. This Figure shows that the BER has been improved by adequately choosing the position of the pilot tones.

It is also important to know the spectral efficiency of each pilot allocation method. Therefore, let us consider a fixed transmission bandwidth of 5.5 MHz and assuming, as a reference, that two bits per frame are dedicated as pilots. The three methods can lead to positive or negative variation of the bit rate around 10 Mbits/s.

The bars in the lowest part of Figure 4 give the minimum, mean and maximum percentage of the effective bit rate for the three techniques. It appears, for example, that selecting subcarriers for which the DMT algorithm would allocate a constellation of minimum size, avoiding a decrease of the effective bit rate, (or an increase of the channel bit rate) leads to satisfactory results.

Finally, the performances of a DMT transmission scheme which includes the proposed PTS algorithm, are compared with those either of a DMT process with the PSAM method, based on pilot tones, or of an OFDM process with a semi-blind estimation.

For the simulation, the BER is computed over 1000 different non time variant selective channels, with one sudden change, 512 kbits being sent in each case. Furthermore, the maximum transmitted DSP for each carrier is equal to -60 dBm/Hz and the mean attenuation of the channel is equal to 50 dB. Curves plotted in Figure 5, representing the BER cumulative distribution, shows that the DMT scheme associated with the semi-blind channel estimation gives better results than the two others, if the
average signal to noise ratio (SNR) is low, typically smaller than 20 dB. A parametric study has shown that, for higher SNR, the DMT and OFDM gives similar results.

Figure 5: Cumulative distribution of BER for different techniques: OFDM, DMT, channel estimation

5. Conclusion

An approach of multicarrier transmissions on indoor power lines, using an association of a semi-blind estimation with a state change detection process has been proposed. The way of positioning pilot tones have been optimized and a comparison between the proposed estimation method and the classical methods like those based on a conventional pilot insertion has been presented.

References