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A Rate Adaptive Bit-loading Algorithm for a DMT Modulation System for in-Building Power-Line Communications

Simone Morosi, Dania Marabissi, Enrico Del Re, Romano Fantacci, Nicola Del Santo.
Dipartimento di Elettronica e Telecomunicazioni - Università degli Studi di Firenze
Via di S. Marta 3, I-50139 Firenze, Italy - Tel.: +390554796271 - Fax.: +390554796485
Corresponding author's e-mail address: morosi@lenst.det.unifi.it

Abstract- **This paper deals with a variable rate Discrete Multi-tone Modulation (DMT) system for broadband Power-Line Communications (PLCs), based on the bit-loading algorithm proposed by Leke and Cioffi. In the proposed system a suitable LMS channel estimator is considered, which is based on the insertion of a Training Sequence (TS). The proposed approach will be compared with the ideal channel estimates, showing its effectiveness. Moreover, different TS lengths will be compared. The performance of the system, expressed in terms of Bit-Rate and Bit Error Rate (BER), with and without an estimation error is derived by simulation under the assumption of frequency-selective multipath fading channel and additive colored Gaussian noise, according to the in-building networks model.**

I. INTRODUCTION

In recent years a great attention has been devoted to the low voltage grid as a potential medium for the "last mile" access network, due to its capability to offer a low cost solution to carry the Internet, voice and data services, video on demand, and other services which may require high bit-rates [1]. Moreover, all the applications known as in-home services could be enhanced by the use of power line grid as a local IP network. Besides, the recent advances in communication and modulation technologies, as well as in adaptive signal processing and error detection and correction, have opened the way for new and effective medium access control and physical layer protocols which allow the PLC networks to operate at speeds which are comparable to the traditional wired and more recent wireless Local Area Networks (LANs) [13]. In particular, high data-rate PLC specifications have been defined by the HomePlug 1.0 standard [12] which ideally can provide a raw data rate equal to 14 Mb/sec. HomePlug 1.0 implements a built-in QoS protocol, which results to be attractive for real-time applications, but does not use bit-loading techniques; the same coding rate and modulation method are used for all indicated carriers.

Recently, the HomePlug Alliance announced the evolution of the standard, called HomePlug AV, which will provide low cost solutions for entertainment applications such as HDTV and Home Theater, being capable of a data rate of 200 Mb/sec at the physical layer. The final specifications release is expected for the second quarter of 2005, and the first chips and products will be probably seen in 2006. Moreover, a third standard, Home Plug BPL (Broadband over Power Line) is intended to provide last mile access solutions. For this standard, the first goal will be the development of Market Requirements Document (MRD) during 2005.

Therefore, these standards end up making PLC competitive with all the alternatives. Ethernet is a proven technology, reliable, secure, and can support data rates of 100Mb/sec or higher, but has the drawback of the lack of ubiquity: it requires new wires. Phone Lines, used with the future HomePNA 3.0, will support data rates up to 100Mb/sec, but neither phone jacks are ubiquitous. Wireless solutions are currently the most popular. IEEE 802.11b can achieve data rates up to 11Mb/sec, 802.11g up to 54Mb/sec; and the incoming 802.11n has as target data rate 100Mb/sec. The main drawback of 802.11 solutions is their incapability of providing QoS, due to the limitation of the 802.11 MAC, which makes it unsuitable especially for in-Home entertainment solutions. Power Lines are easily accessible, no new wiring is needed, and with the next generation HomePlug AV standard, will provide data rates of 80Mb/sec at the MAC layers, with a built-in QoS support. They are indeed a good choice for in-home networking. On the other hand, PLC channel constitute a rather hostile medium for data transmission; particularly, it is characterized by frequency selective phenomena, echoes, colored noise, narrow band interference and impulsive noise. This challenging environment claims for highly sophisticated communications techniques [2].

Different communication systems have been proposed for broadband downstream Power-Line Communications (PLCs) [3]: particularly, Discrete Multi Tone (DMT) techniques have proved to be the proper solution due to their capabilities in facing channel impairments, while affording high capacity. In these system the overall transmitted information is separated in many parallel independent sub-streams, by supporting variable bit-rates; moreover, a guard interval or a cyclic prefix is included, to eliminate the *Intersymbol Interference* (ISI) resulting from multipath propagation. Finally, the Bit-loading techniques [4] permit the DMT systems to achieve a capacity which is close to the theoretical limit, at the cost of a complexity increase. As a result, intense studies have been focused on the DMT

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modulations, also because of their immunity to the noise and the channel conditions and their flexibility and capability to achieve high data rates over hostile frequency-selective channels. Particularly, a DMT system provides a fine data-rate granularity: in a DMT system, the bits to be transmitted are mapped in symbols belonging to the appropriate constellation, and time domain symbols are obtained using Inverse Fourier Transform of 2N Hermitean complex symmetric values, obtained from N complex symbols, with the addition of a cyclic prefix (CP) at the beginning of the symbol; the prefix is obtained by repeating the last few bits of the same symbol; the CP length must be at least equal to delay spread of the channel. At the receiver, demodulation is obtained by skipping the cyclic prefix, and then applying Direct Fourier Transform. After DFT operation, a 1-tap equalizer is required to perform coherent demodulation. With Bit –loading techniques, more bits are transmitted on the sub-channels characterized by better Signal-to-Noise Ratio (SNR) values, while the sub-bands whose SNR results to fall below a certain threshold are completely turned off. Therefore, in DMT systems, the Channel State Information (CSI) must be known both at the transmitter - in order to perform bit-loading algorithms – and, obviously, at the receiver; hence, the SNR value of each sub-channel must be determined *a-priori* by the receiver, and fed back to the transmitter. In the case of a slow time-varying channel, as the PLC channel, a simple data-aided method can be used for channel estimation: particularly, a known training sequence can be sent, without remarkable performance and throughput loss. The most crucial aspect for the DMT system design can be identified in the law which is used to distribute power and bits to all the subchannels, i.e., the *Bit-loading algorithm*. As it is known, the optimal Bit-loading scheme is based on the *water-pouring* distribution: it can be shown that water-filling algorithm converges to Shannon channel capacity as the bandwidth of the subchannels gets smaller [6]. In practical applications, however, the optimal solution cannot be determined, because it assumes infinite granularity in constellation sizes and in sub band division, which are not realizable: several sub-optimal bit-loading algorithms have been recently investigated in literature [4, 7-10].

In this paper, a variable rate DMT modulation system for broadband Power-Line Communications (PLCs), based on the bit-loading algorithm [4] is proposed. In order to evaluate performance loss due to channel estimation, a modified LMS channel estimator has been introduced [5], which is based on the insertion of a Training Sequence (TS). Moreover, different TS lengths will be compared.

II. Bit-loading Algorithms

The performance of a DMT system strongly depends on the effectiveness of the Bit-loading technique which is adopted. As it is known, the bit-loading algorithms belong to two main families, namely *Margin Adaptive* and *Rate Adaptive*. While the Margin Adaptive algorithms minimize the probability of error, for a given Bit-Rate, the Rate Adaptive ones maximize the Bit-Rate, assuming a given P_e as a system constraint. The three algorithms that are mainly in use today belong all to the first category: Hughes-Hartogs, Chow, and Fischer.

Hughes-Hartogs [7] algorithm generates and uses a table of incremental energies, which has to be renewed at each step, for any additional bit assigned to a particular subchannel. The computational complexity is the weak point of this algorithm that results to be impractical when the number of subchannels and the number of bits per symbol are large, as in the PLC environments.

Chow's algorithm [8][9] was proposed in early 90's, for ADSL systems: it is based on the fact that the difference between optimal water-filling energy distribution and flat-energy one is minimal. As a consequence, the same amount of energy is assigned to each subchannel turned on, while the number of bits to be assigned is computed by a logarithmic law, depending on the desired value of Bit Error Rate (BER) and on the estimates of the SNR value of the subchannel.

Fischer's algorithm [10] is the most recent of three. It aims at minimizing the probability of error in each subchannel: this algorithm relies on a set of iterative equations, which lead to a flat-energy distribution, and show a slight improvement in SNR over Chow solution.

However, the evolution of broadband communications focuses the interest on the highest achievable rate; hence, the Rate adaptive approach results to be more promising. In this paper a DMT system with a Rate Adaptive Bit-loading algorithm is analyzed; in particular the algorithm proposed by Leke & Cioffi has been considered [4]. This algorithm, proposed in 1997, relies on the following assumption: the most crucial aspect of the Bit-loading algorithms is the determination of the subchannels that have to be turned off and on; particularly, if a subchannel which should be turned off is used for the transmission, the BER that characterizes it increases and ends up raising the overall BER.

Therefore, the first step consists in determining which subchannels have to be turned on and off. Let H_m and σ_m^2 represent respectively the gain and the noise variance of the m^{th} subchannel; it will be turned on if:

$$\Gamma \cdot \frac{\sigma_m^2}{\left|H_m\right|^2} > \frac{1}{N_{on}} \cdot \left(\varepsilon + \Gamma \cdot \sum_{n=1}^{N_{on}} \frac{\sigma_n^2}{\left|H_n\right|^2} \right) \qquad (1)$$

where ε is the total energy budget, N_{on} is the number of subchannels turned on, while the parameter Γ, defined as the *SNR gap*, indicates how far the system is from the maximum achievable capacity. The *SNR gap* is a function of the target probability of error; if an uncoded M-QAM system is considered, Γ can be evaluated as:

$$\Gamma = 10\log_{10}\left(\frac{\left\lfloor Q^{-1}\left(\frac{P_e}{3}\right)^2\right\rfloor}{3}\right) \quad (2)$$

The second step consists in distributing the energy over the subchannels which have been turned on. The optimal water-filling distribution of energies can easily be obtained by few operations. The energy in each subchannel is given by:

$$\varepsilon_n = \frac{1}{N_{on}} \cdot \left(\varepsilon + \Gamma \cdot \sum_{n=1}^{N_{on}} \frac{\sigma_n^2}{|H_n|^2}\right) \quad (3)$$

for $n = 1, \quad , N_{on}$.

In the final step the algorithm provides the number of bits per each subchannel which is equal to:

$$b_n = \frac{1}{2} \cdot \log_2\left(1 + \frac{\varepsilon_n \cdot g_n}{\Gamma \cdot \gamma_m}\right) \quad (4)$$

where γ_m is the target margin, and g_n is the processing gain: $g_n = \frac{|H_n|^2}{\sigma_n^2}$. The number of bits which is determined by (4) has to be rounded to an integer value, and energies re-scaled accordingly. The Leke-Cioffi algorithm is well suited for slowly varying channels, and for *bursty* application, such as the IP communications and generic packet data transmissions, where it is important to afford transmission at the maximum achievable data rate.

III. Channel Estimation

A bitloading algorithm requires the knowledge of *Channel State Information* (CSI), so involving the problem of Channel Estimation. Moreover, the Channel State Information must be known both at the transmitter, which has to distribute bits and energies according with the described scheme, and at the receiver, to realize the demodulation and the 1-tap equalization. Therefore, the system must provide a training sequence, and a feedback line, to feed back the CSI from the receiver to the transmitter. The slowly time varying channel characteristics permit very precise estimates.

We have introduced an LMS channel estimator [5], implementing a low-complexity pilot-aided estimation scheme, in order to evaluate the performance decay of the system, in the presence of an estimation error. With the chosen algorithm, slightly different from classical LMS, a training sequence which is composed of M symbols, is periodically sent, and the subchannel estimated coefficient h_n is obtained iteratively.

In classical LMS the prediction error $e(t)$ is evaluated by comparing the filtered received sample with the expected one, then multiplied by the correction factor μ and used to adjust the estimated coefficient of the filter. In the considered AMSE scheme, the received signal on the nth subcarrier, is normalized to the transmitted pilot-sequence signal known at the receiver.

The resulting signal, is compared with a coefficient, $h_n(m)$: the difference between these two signals is the error, $e(m)$. This error is used to determine a correction factor for the new coefficient:

$$h_n(m+1) = \frac{1}{L}\sum_{l=0}^{L-1} h_n(m-l) + \mu e^*(m) \quad (6)$$

where m indicates the position of the bit within the Training Sequence (TS). Note that in (6) the updating of the coefficients $h_n(m)$ is performed by using the mean of L previous coefficients, while in the classical LMS only the value of coefficient at the previous step is considered; the parameter L has to be less or equal than the TS length M ($L \le M$). The coefficient $h_n(m)$ represents the channel estimate and, after a few iterations, produces a value close to the real channel value: particularly, the AMSE algorithm refines the estimates achieved by the length-M pilot signal. This scheme is based on the assumption that the parameter remains constant over the duration of the pilot sequence; therefore, it is effective with a slowly fading channel.

The effectiveness of the proposed approach will be shown in the section of the simulation results.

IV. Proposed System and Working Conditions

It is worth stressing that Discrete Fourier Transform (DFT) utilization permits us to perform modulation and demodulation by base-band processing: particularly, modulation operation, that is symbol mapping upon a single sub-carrier, is accomplished by inverse Fourier Transforming (IDFT) of 2N Hermitean complex symmetric values; these values are generated from N complex symbols in order to transmit a real signal modulated upon N sub-carriers. During propagation, ISI arises because of delay spread while channel distortion takes to loose orthogonality between sub-carriers, so creating Inter-Channel Interference (ICI). Both impairments can be addressed by introducing a cyclic prefix (CP): in particular, if the CP length is chosen to be at least equal to delay spread ν, bandwidth efficiency loss is equal to ν / (ν + 2N). CP introduction takes to obtain at the receiver cyclic convolution of Channel Impulse

Response and transmitted signal so that it is relatively easy to eliminate CP from the received signal. Moreover, after DFT, only a 1-tap equalizer is required for the received signal.

The propagation environment which is considered in this paper is the wired communication channel inside of buildings as described in [11]. The impedance of the Power Line (PL) channel is highly varying with frequency, ranging between a few Ohms and a few kOhms. The load condition changes and the discontinuities in branch cables can cause reflection and echoes. The peaks in the impedance characteristics may occur at certain frequencies. As a result, the PL channel can be considered as a multipath propagation environment with deep narrow-band notches in the frequency response. The power lines noise spectrum is highly varying with frequency and time; in the considered environment three kinds of noise can be identified: Additive Colored Gaussian Noise with spectral power density decaying with frequency, narrow-band interference which can be modeled as single tones in frequency domain, and impulse noise: in particular, impulse noise is composed by strong peaks whose duration could be equal to some ms and mean time between occurrence to several s. During such strong peaks, information bits are damaged so that proper coding and interleaving schemes are needed to avoid remarkable performance loss. In this paper, uncoded data flow is taken into account so that this kind of noise is not considered. Finally, the channel characteristics are assumed not to show fast variations in time with respect to the bit epoch so that the channel can be considered as quasi-stationary. In order to effectively represent channel characteristics, the set of echo model parameters provided in [11] has been adopted. In the simulations, the following working conditions have been assumed:

- Frequency ranging from 1 to 21.480 MHz.
- Coherent phase modulation.
- Perfect power matching (i.e., ideal power transfer).
- Number of sub-channels equal to 256.
- Maximum number of bits per symbol equal to 7 (i.e., 128 signal constellation).

V. RESULTS

In this section, the performance of the proposed systems is described in different environment conditions and for several system load configurations: it is expressed both in terms of Bit-Rate and BER, derived by simulations under the assumptions of frequency-selective fading channel and additive colored gaussian noise, according to the model [11].

First, we have considered the case of perfect knowledge of the channel state information. The transmitted bit rate increases as the value of the SNR increases, as shown in Fig. 1: this effect is due to the adoption of a rate-adaptive algorithm. It can be observed that the algorithm tends to saturate the best channels, i.e., the ones with better channel conditions. Moreover, since the number of bits per symbol cannot be greater than 7, overall bit-rate tends to an asymptotic value, which in our case was about 70 *Mb/s*. Consequently, Bit Error Rate decreases slowly while SNR increases, as shown in Fig. 2, until the system reaches its saturation: in the case of 7 bits per symbol this saturation condition is reached for a SNR equal to 30 dB. When saturation occurs, the transmission bit-rate becomes constant, and BER drops rapidly.

On the other hand, Figs. 1 and 2 show that the algorithm can not maintain the probability of error on the desired value: the BER starts decaying slowly as SNR increases, and more rapidly when the system reaches its saturation. This effect is due to the characteristics of PL channel, which is heavily spectrally shaped: as a result, several favorable subchannels reach rapidly the maximum number of bits that the encoder is able to allocate. To get better performances, the system should so provide wider sets of signals, meaning a significant complexity increase.

Comparing the system using estimated coefficients with the one with ideal channel knowledge, BER and Bit-Rate performance, respectively in Figs. 3 and 4, can be seen to be extremely near. We can deduce that the estimation problem is not crucial in this context because of the channel quasi-stationarity and the short distances involved: particularly, in this environment, the estimation process is not to be repeated so often; therefore, the training sequences which are used in the proposed system most be long enough to reach a good approximation. Figs. 3 and 4 show the results which have been obtained by training sequences of 250 and 500 DMT symbols.

VI. CONCLUDING REMARKS

In this paper, a DMT transmission system, dedicated to power line applications has been analyzed. The proposed systems uses a Rate Adaptive Bitloading scheme based on the one proposed by Leke and Cioffi; in the system an LMS channel estimator has been introduced and used to compare performance with ideal channel estimation case.

The simulation results showed that estimation problem is not crucial in such a context and that the Bitloading algorithm permits us to achieve an extremely high data-rate, if higher order constellations have been allowed. The complexity required by the proposed system is comparable with the one required by the DSL communications.

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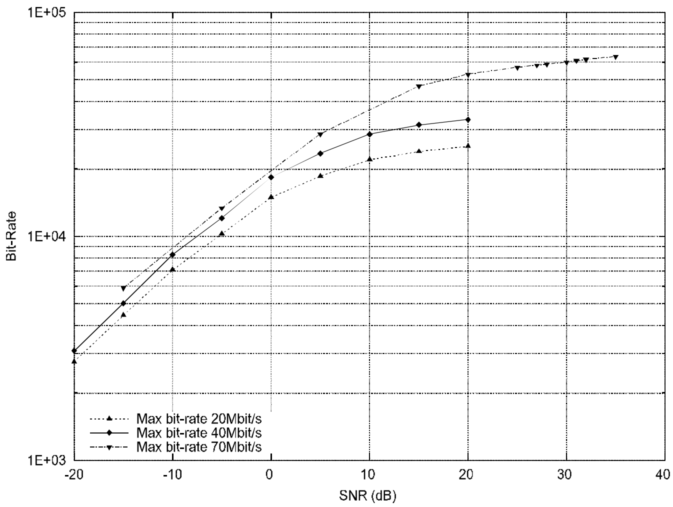

Fig. 1: Bit-rate comparison.

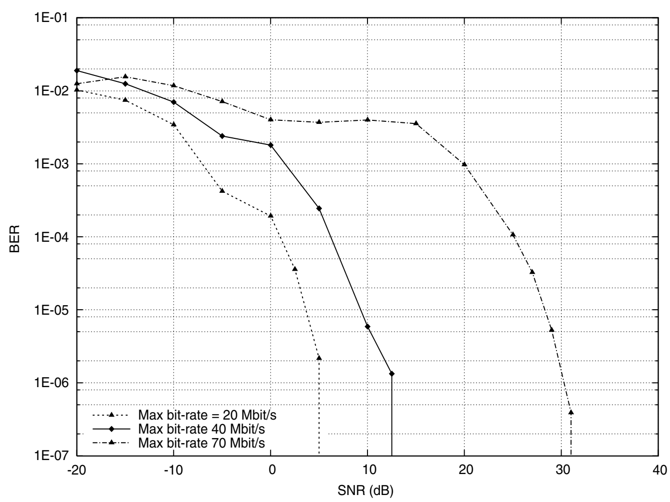

Fig. 2: BER comparison.

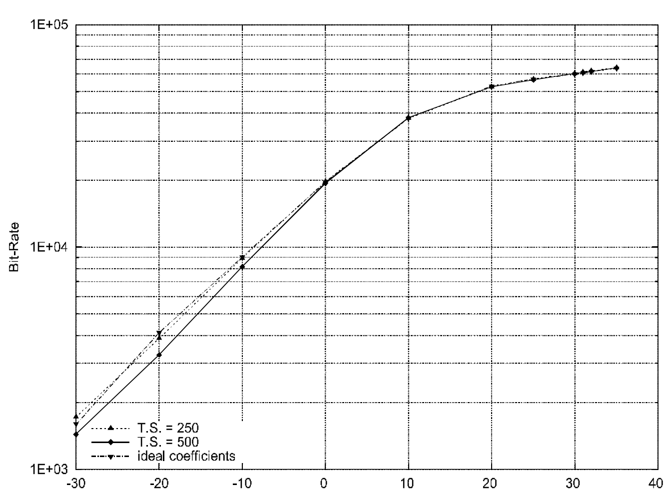

Fig. 3: Bit-rate comparison.

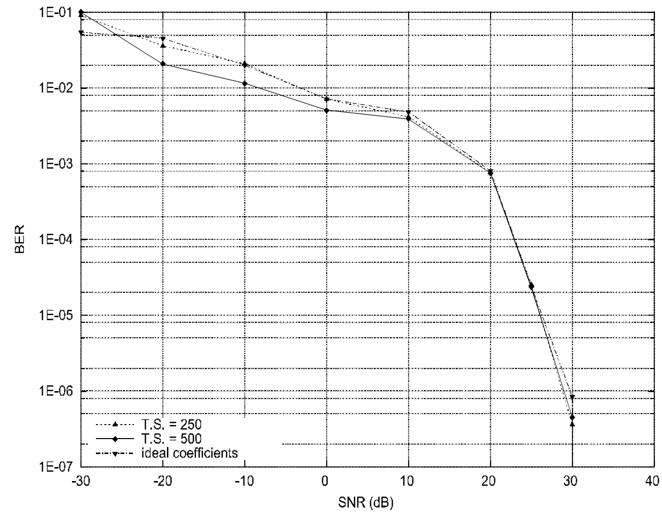

Fig. 4: BER comparison.

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