A Rate Adaptive Bit-loading Algorithm for a Discrete Multi-tone Modulation System in Downstream Power Line Communications

Simone Morosi, Enrico Del Re, Romano Fantacci, Dania Marabissi, 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) communication system for broadband Power-Line Communications (PLCs), based on the bit-loading algorithm proposed by Leke and Cioffi. In the proposed system a proper LMS channel estimator will be considered, which is based on the insertion of a Training Sequence (TS). The proposed approach will be compared with ideal channel estimation case, so 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 the recent years a great attention has been devoted to 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 Internet, voice and data services, video on demand, and other services which require high bit-rates. Moreover, all applications known as in-home services could be enhanced by the use of power line grid as a local IP network. Conversely, the recent advances in communication and modulation technologies, as well as in adaptive signal processing and error correction and detection, have opened the way of new and effective medium access control and physical layer protocols that can permit to the PLC networks to operate at speeds comparable to the traditional wired and more recent wireless Local Area Networks (LANs).

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 resulted to be the proper solution due to their capabilities in facing channel impairments, while affording high capacity.

In such a system overall transmitted data are separated in many parallel independent sub-streams, by supporting variable bit-rates; moreover, a guard interval or a cyclic prefix is included, to eliminate *Intersymbol Interference* (ISI) resulting from multipath propagation. Finally, the bit-loading techniques permit DMT system to achieve capacity near to theoretical limit, at the cost of an increase in system complexity.

Recently, intense studies focus on the DMT modulations, because of their immunity to the noise and the channel conditions, their flexibility and capability to achieve high data rates over hostile frequency-selective channels[1]. Particularly, a DMT system provides fine data-rate granularity: in a DMT system, the bits to be transmitted are mapped in proper 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; prefix is obtained repeating last few bits of the same symbol. CP length must be at least equal to delay spread of the channel. At

---

[1] DMT techniques have been chosen by ANSI (*American National Standards Institute*) and by ETSI (*European Telecommunications Standard Institute*) for ADSL systems, because of its capability to reject FEXT (*far-end crosstalk*, the most common form of noise in DSL)

the receiver, demodulation is obtained skipping the cyclic prefix, and then applying Direct Fourier Transform. After DFT operation, a 1-tap equalizer is required to perform coherent demodulation. Hence, 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 be 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 used, 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 to be emitted and the 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.

In this paper, a variable rate Discrete Multi-tone Modulation (DMT) communication 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, an 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.

This paper is organized as follows: in section II the main bit-algorithms are reviewed and the proposed technique is thoroughly described while the channel estimator is described in the Section III. The behavior of the proposed detectors is discussed in Section IV together with the working conditions. Finally, the simulation results are presented in Section V, before the concluding remarks that are given in Section VI.

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 m*argin 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 of fixed data-rate algorithms: 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, 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 Bitloading 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 end 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 mth 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:

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

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 $n = 1, \ldots, N_{on}$.

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 to obtain a highly accurate 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 presence of an estimation error. With the chosen algorithm, slightly different from classical LMS, a training sequence which is composed by M symbols, is periodically sent, and the subchannel estimated coefficient $h_{n,i}$ is obtained iteratively, using the mean of L previous coefficient.

In classical LMS prediction error $e(t)$ is evaluated by comparing received sample with expected one, then multiplied by a correction factor μ and used to adjust estimated coefficients $h(t)$:

$$h_{n+1}(t) = h_n(t) + \mu e^*(t) \quad (5)$$

This algorithm, called *Adaptive Mean Square Error* (AMSE), refines the estimate during whole duration of training sequence:

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

where M is a parameter, that must obviously satisfy the condition $L \le M$. This scheme is based on the assumption that the parameter remains constant over the duration of the pilot sequence; therefore, it is effective with 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

The proposed system has the structure shown in Fig.1. It is worth stressing that Discrete Fourier Transform (DFT) utilization permits to perform modulation and demodulation by base-band processing: particularly, modulation operation, that is symbol mapping upon 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 faced by cyclic prefix (CP) introduction: in particular, if CP length is chosen to be at least equal to delay spread ν, bandwidth efficiency loss is equal to $\nu / (\nu + 2N)$. 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.
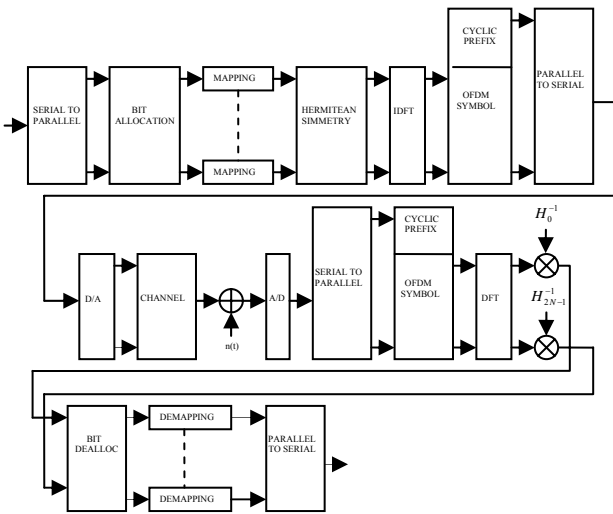

Fig.1: DMT block diagram.

The propagation environment considered in this paper is the wired communication channel inside of buildings as described in [11]. Power Line channel impedance is highly varying with frequency, ranging between a few Ohm and a few kOhm. Load condition changes and discontinuities in branch cables can cause reflection and echoes. Peaks in the impedance characteristics may occur at certain frequencies. As a result, PL channel can be considered as a multipath propagation environment with deep narrow-band notches in the frequency response. 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 modelled 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, channel

characteristics are assumed not to show fast variations in time with respect to the bit epoch so that 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 are 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 multipath 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. 2: 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. 3, until the system reaches its saturation: in the case of 7 bits per symbol this saturation condition is reached at a SNR of 30 dB. When saturation occurs, the transmission bit-rate becomes constant, and BER falls down rapidly.

On the other hand, Figs. 2 and 3 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.
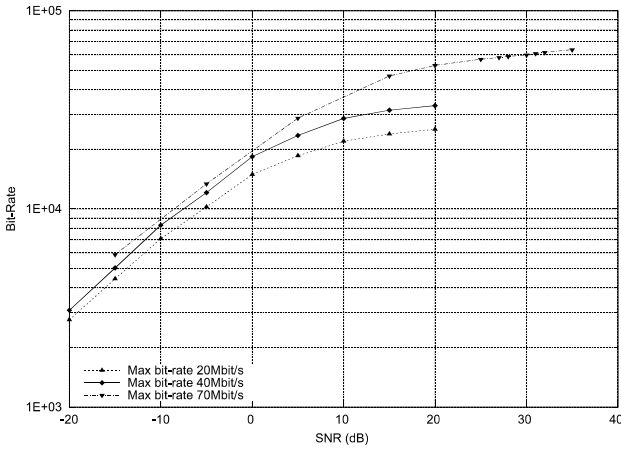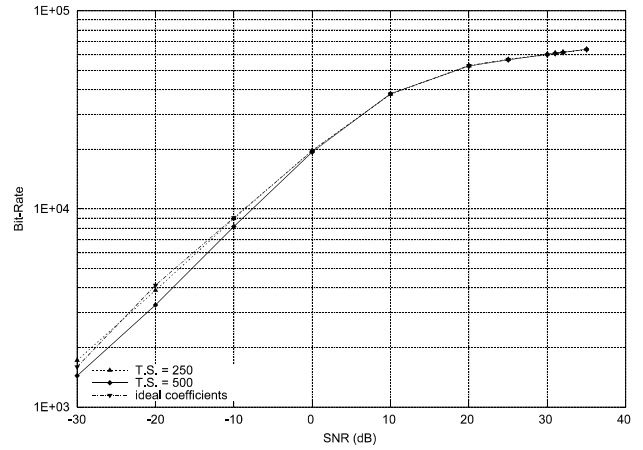
Fig. 2: Bit-rate comparison.
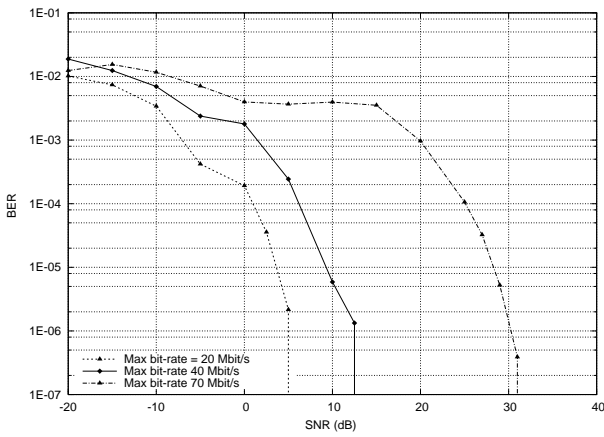

Fig. 4: Bit-rate comparison.
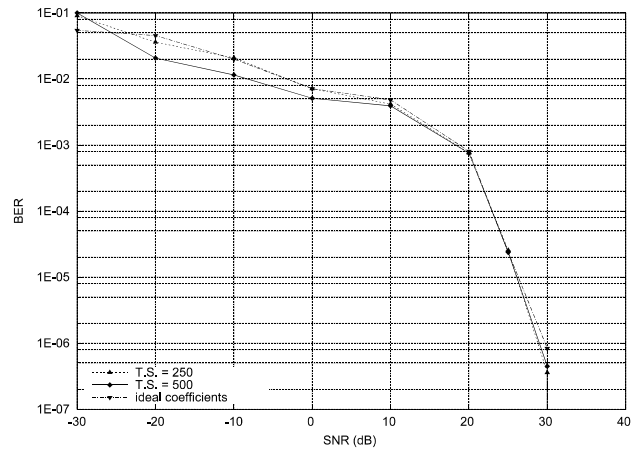

Fig. 3: BER comparison.

Fig. 5: BER comparison.

Comparing the system using estimated coefficients with the one with ideal channel knowledge, BER and Bit-Rate performance, respectively in Figs. 4 and 5, 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, allowing to use a training sequence long enough to reach a good approximation. Figs. 4 and 5 show results obtained by using training sequences of 250 and 500 DMT symbols.

VI. Concluding Remarks

In this paper, a DMT transmission system, dedicated to power line context, and implementing a Rate Adaptive Bitloading scheme based on the one proposed by Leke and Cioffi, has been analyzed; in the system an LMS channel estimator has been introduced and used to compare performance with ideal channel estimation case. The simulations results showed that estimation problem is not crucial in such a context, and that bitloading algorithm permits to achieve 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 comunications.

References

[1] Pavlidou N., Vinck A. J. H., Yazdani J., Honary B., "Power Line Communications: State of the Art and Future Trends", *IEEE Communications Magazine*, VOL 41, NO. 4, Apr. 2003, pp. 34-40.

[2] E. Biglieri, "Coding and Modulation for a Horrible Channel", *IEEE Communications Magazine*, May 2003.

[3] Del Re E., Fantacci R., Morosi S., Seravalle R., "Comparison of CDMA and OFDM Techniques for Downstream Power-Line Communications on Low Voltage Grid", *IEEE Transactions on Power Delivery*, VOL 18, NO. 4, Oct. 2003.

[4] Leke A., Cioffi J. M. "A Maximum Rate Loading Algorithm For Discrete Multitone Modulation Systems", *Global Telecommunications Conference, GLOBECOM*, 1996

[5] Fantacci R., Marabissi D., Michelini M., Bergamini G. "Low Complexity Pilot-Aided Data Detection in MC-CDMA Systems", unpubl. 2003.

[6] R.G. Gallagher, *Information Theory and Reliable Communication*, Wiley, 1968.

[7] D. Hughes Hartogs, "Ensemble Modem Structure for Imperfect Transmission Media", U.S. Patents, Nos. 4,833,706.

[8] J.S. Chow, J.C. Tu, J.M. Cioffi, "Discrete Multitone Transceiver System for HDSL Applications", *IEEE Journal on Selected Areas in Communications*, Aug. 1991.

[9] J.S. Chow, J.M. Cioffi, J. Bingham, "A Practical Discrete Multitone Transceiver Loading Algorithm for data Transmission Over Spectrally Shaped Channels", *IEEE Transaction on Communications*, Feb/Mar/Apr, 1995.

[10] R. F. H. Fischer and J. B. Huber. "A New Loading Algorithm for Discrete Multitone Transmission", *Proc. IEEE GLOBECOM '96*, pp. 724-728, London, UK, Nov. 1996.

[11] H. Phillips, "Modeling of powerline communication channels", in *Proc. 3rd Int. Symp. Power Lines Commun. Its Applicat.*, Lancaster, U.K., Mar. 1999.