A New Adaptive Algorithm for Bit Allocation in DMT Modems

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Abstract: - This paper presents a new algorithm for distributing bits to the subchannels of a DMT modem. The so-called Inverse Robin Hood algorithm converges rapidly and can effectively optimise the overall bit error rate in a non-Gaussian channel.

Key-words: - Discrete multi-tone modulation, adaptive bit-loading, DSL modems, non-Gaussian channels.

1 Introduction

Discrete Multi-Tone (DMT) modems are increasingly being used in many telecommunications systems, from ADSL modems to the 802.11a and Hiperlan wireless standards. They use an efficient form of Frequency-Division Multiplexing (FDM), which performs modulation and demodulation by Inverse and Forward Fast Fourier Transforms (FFTs) respectively. Typically a simple frequency-domain equaliser is used, and in the case of wireless systems, equalisation may be performed using an adaptive decision-feedback equaliser.

DMT modems use different symbol alphabets in different subchannels, each with 2_i^N, $N \in \mathbb{Z}$ symbols. Traditionally, the choice of N_i is governed by the Signal to Noise Ratio (SNR) measured in each subchannel [1]. The objective of the optimisation algorithm used in most existing DMT modems is to achieve the same Bit Error Rate (BER) across all subchannels. However, this is based on an assumption which, in general, is incorrect - namely, that the only noise present is Additive White Gaussian Noise (AWGN).

In practise, most telephone lines suffer from Impulse Noise (IN) impairments as well as AWGN[2], [3]. Under these noise conditions, the BER may be significantly higher than for a purely Gaussian channel for a given SNR. Therefore, analytic results relating BER to SNR will not be appropriate for such channels, although they may give a reasonable initial estimate. This would also be true for many other channels over which DMT modems operate.

2 The Inverse Robin Hood Algorithm

The bit-allocation algorithm presented in this paper offers a number of advantages over the conventional algorithm. When used in a non-Gaussian channel, it is still able to generate a bit allocation table which provides approximately constant BER across all subchannels. It also converges relatively quickly, with a modest degree of computational complexity.

The conventional bit-allocation algorithm constructs a so-called incremental power matrix, in which the elements of successive rows contain the level of extra power required to transmit an additional bit in the subchannel which corresponds to that column. The bit allocation is then performed by searching for the minimum incremental power in successive rows, and assigning an extra bit to that subchannel until either all bits are used up or alternatively, when the total power is just less than the maximum allowed transmission power. However, this algorithm implicitly assumes a Gaussian channel, and therefore will not necessarily provide an optimal bit allocation for a channel subject to non-Gaussian noise.

The Inverse Robin Hood[1] (IRH) algorithm takes the simpler approach of measuring the BER in each subchannel (since this is what we really wish to optimise). The modem is initialised as follows:

- Initialise M to some integer between 1 and half the total number of channels;
- Initialise the bit allocation table with a uniform number of bits per subchannel, i.e. each subchannel is allocated a number of bits equal to the total bits per macro-symbol divided by the number of subchannels;

The allocation process then continues according to the following algorithm:

1. Transmit a number of training symbols known to both receiver and transmitter;
2. Measure the actual BER in each subchannel;
3. Identify the best M subchannels and the worst M subchannels;
4. Deduct one bit per symbol from each of the M 'bad' subchannels;
5. Credit one bit per symbol to each of the M 'good' subchannels;
6. If $M > 1$, re-calculate M according to the formula

$$M_{i+1} = M_i \frac{\log TBER_{i-1}}{\log TBER_i} \tag{1}$$

   where $TBER_i$ is the total BER measured in iteration i. M is only adjusted in the second and successive iterations.

This is repeated until the overall BER falls below the desired threshold value, or when repeated iterations provide no improvement in BER. Changes in bit allocation are discarded if they do not result in a better average - and if two 'good' or 'bad' subchannels have an approximately equal bit error rate, the channel to be deducted or credited is selected at random.

[1]Because it takes bits from the poor BER subchannels and gives them to the 'rich' BER subchannels

The logarithmic function used to scale back M was found to allow the algorithm to converge rapidly, without needlessly swapping around bits between subchannels as the bit allocation approaches the optimum distribution.

3 Results

For the following experiments we have used a 256-channel DMT modem similar to that defined by the ADSL standard [4]. The channel model is as described in [5] (essentially a distributed RLC ladder network with a distributed impulse noise + AWG noise source). The RLC parameters for the channel are 0.5 km of 0.63 mm cable with 113 Ω/km, 700 μH/km and 45 nF/km, followed by 1.5 km of 0.5 mm cable with 280 Ω/km, 587 μH/km and 50 nF/km. The DMT modem transmits 768 bits per macro-symbol (an average of three bits per subchannel). The overall SNR is 15 dB.

The training process is illustrated in Figure 1. BER calculations are averaged over approximately 200 macro-symbols for each iteration.

The eventual bit allocation is shown in Figure 2.

4 Conclusion

The IRH algorithm converges quickly and is highly effective at providing a uniform BER across all subchannels on a non-Gaussian channel. Since there is no implicit assumption about the characteristics of the channel, this algorithm has potential applications in many terrestrial data transmission environments.

References:

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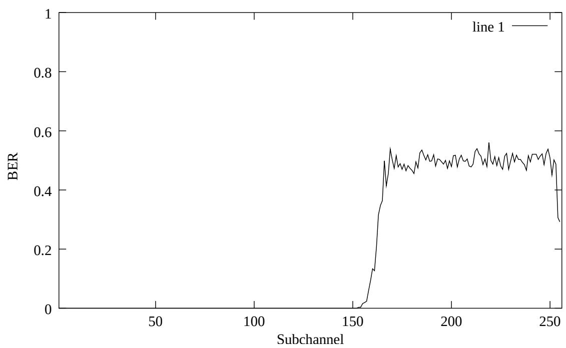
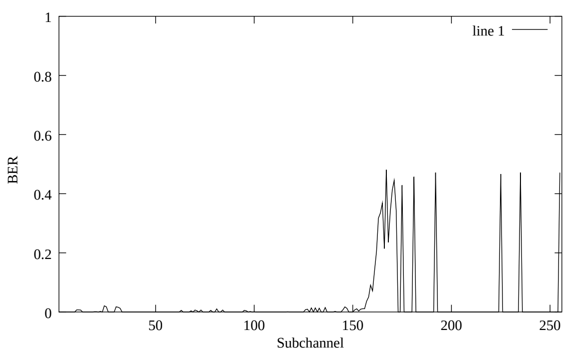
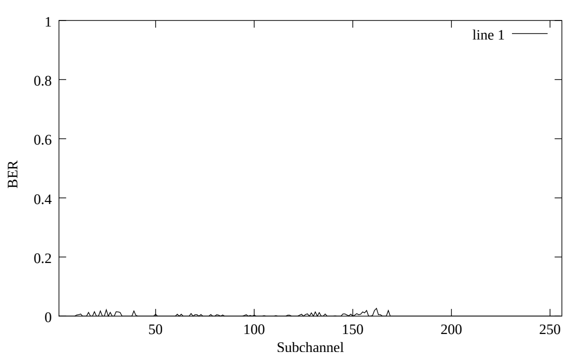
[2] Jr. James T. Aslanis and John M. Cioffi, "Achievable Information Rates on Digital Subscriber Loops: Limiting Information Rates with Crosstalk Noise,"



(a) Initial BER/subchannel



(b) After eight iterations



(c) After ten iterations

Figure 1: Convergence of IRH algorithm

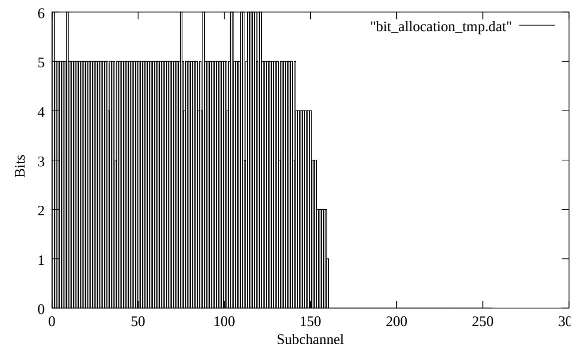


Figure 2: Final Bit Loading

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