ADAPTIVE EQUALIZATION FOR DIGITAL MOBILE RADIO SYSTEMS

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Abstract

In digital mobile radio systems the time dispersion due to multipath propagation is a problem, when the bit time is of the same order as the delay spread. The resulting intersymbol interference (ISI) can be quite severe. To combat the ISI one needs some form of equalizer. Here we study the performance of the adaptive decision feedback equalizer (DFE) on a fading dispersive channel. It is shown that the multipath in fact give diversity effects by using a DFE. Furthermore the improvement from two branch antenna diversity is shown.

1 INTRODUCTION

When the data rate on the mobile radio channel is above 100 kbits/s the ISI due to multipath propagation can be severe. The channel is furthermore subject to fading due to movement of the vehicles. To handle these problems one must use an adaptive equalizer, designed for modulation methods suitable for mobile communications.

The system we will study operates in the 900 MHz band and uses time division multiple access (TDMA) with ten users per carrier. The user data rate is approximately 30 kbits/s (speech- and channel coding). The net data rate is therefore 300 kbits/s, i.e. the bit time T_b is roughly 3 μs. Each user transmits the bits in bursts of approximately 1 ms. The burst consists of guardtime, synchronization sequence and information bits. In this paper the synchronization sequence is 30 bits and the number of information bits is 220. The modulation method is gaussian minimum shift keying (GMSK) with $BT_b = 0.25$. The described system is similar to the GSM system, in fact it is almost identical to the GSM proposal in [3].

According to [4] the delay spread can be as high as 10 μs, in fact up to 20 μs has been measured in the alps. The ISI in the described system will therefore be quite severe, up to several bit times. Due to fading the channel changes so fast that it is not constant from burst to burst, in fact it is not even constant during the burst. Therefore we must have an *adaptive* equalizer to take care of the ISI and to track the channel.

The derivation of the adaptive algorithms and equalizer structures, and a more detailed discussion of the results are given in [1] and [2].

2 THE EQUALIZER

One of the reasons for choosing GMSK modulation, or other argument modulation methods, is that the envelope of the transmitted signal is constant. These methods are generally nonlinear which makes it difficult to design a traditional equalizer, i.e. a DFE. It is though well known that GMSK is approximately equal to staggered quadrature phase shift keying (SQPSK). In [1] the DFE structure as well as adaptive algorithms are derived for SQPSK. This equalizer can then be used also for GMSK, although a minor degradation due to the approximation will result. Instead of going into details here we just discuss the general principles of the DFE, shown in Figure 1.

In the literature on adaptive equalizers one is almost always assuming quadrature amplitude modulation (QAM), which can be seen as a generalization of QPSK. The modulator for QPSK splits the input data in two streams, which are pulse shaped for spectrum efficiency. These signals (I- and Q-signals) are then amplitude modulated with the carrier in quadrature, i.e. multiplied by $\cos(2\pi f_c)$ and $-\sin(2\pi f_c)$ respectively. The difference for SQPSK is that the Q-signal is delayed by T_b. In the demodulator the received signal is mixed with the carrier in quadrature. After lowpass filtering the two I- and Q-signals are sampled with bitrate. The resulting complex valued signal $y(n)$ is fed to the equalizer, see Figure 1. Note that by sampling with bit rate we prepare for fractionally spaced equalization (FSE). In the SQPSK case one of course offsets the Q-signal by T_b in the decision making, in the same way as for the transmitter.

The DFE consists of two transversal filters, the feed forward (FF) and the feed back (FB) filters. The number of taps in the two filters are N_1 and N_2 respectively. The DFE can therefore be described by (N_1,N_2). The objective of the DFE is to minimize the error signal $e(n)$. Note that during the training sequence the decisions $d_D(n)$ are known in the equalizer. The taps in the DFE can be adjusted either by block methods or recursively. Here the equalizer is adjusted recursively with the fast converging recursive least squares (RLS) algorithm. This algorithm minimizes the exponentially weighted sum of squared errors cost function each iteration. It can be shown that when the equalizer is optimally adjusted, i.e. minimum mean squared error (MMSE), the FB filter taps are exactly the causal part of the ISI from the channel convolved with the FF filter.

A completely different type of equalizer is the maximum likelihood sequence estimator (MLSE), also known as the Viterbi equalizer. The performance on the fading channel is roughly the same as for the DFE, given similar implementation complexity. The Viterbi equalizer is not further considered here.

3 SIMULATION RESULTS

In the simulations the channel model is the so called two ray channel. This model consists of one direct path and one delayed reflected path, both subject to (independent) Rayleigh fading. Here the mean power of the two rays are equal and the doppler frequency is 100 Hz. The channel is therefore specified by the delay between the rays. Furthermore white gaussian noise (WGN) with single sided spectral density N_0 is added. This gives an SNR of E_b/N_0, where E_b is the bit energy.

In [2] it is shown that one has to use the RLS algorithm, at least during the training sequence. The simpler least mean squares (LMS) algorithm is not fast enough to converge during the short training sequence. In this paper the RLS algorithm with a forgetting factor of 0.97, giving a memory of approximatley 30 symbols (60 bits in the system described here).

Before the bit error rates (BER's) are analyzed it should be mentioned that most of the bursts are received without errors. When the burst contain errors it is a high probability for many errors, i.e. a catastrophic burst. The reason is error propagation in the FB-filter and in the adaptive algorithm. This information is very important when designing the error correcting codes.

In Figure 2 the performance for three equalizer structures are given, for different amount of ISI. The BER, or probability of bit error, versus delay of reflected ray is shown, for an SNR of $E_b/N_0 = 18.5$ dB. For zero delay we have the so called flat fading case. One linear equalizer (8,0), i.e. no FB filter, and two DFE's (6,4) and (10,8) are considered. The conclusion is that for realistic channels the (6,4) equalizer performes best. The (8,0) equalizer can not handle large time dispersion as well as the other two. The reason why the (10,8) equalizer, i.e. the most complex one, is the worse is because of the short training sequence and noise in the weights (from the algorithm).

The most interesting conclusion from Figure 2 though is the diversity effects one gains, due to the multipath. That is, for a moderate time dispersion the equalizer actually performes better than without time dispersion. In fact the zero time disperion case, i.e. flat fading, is a very difficult case which the equalizer can not handle well.

The remedy to handle flat fading under normal circumstances, i.e. lower bit rates, is to use for instance antenna diversity. In Figure 3 it is shown that this is a very good method also when we have an equalizer. Here we have a simple selection diversity, i.e. the branch with the largest power is chosen, on a burst to burst basis. The two branches are assumed to be totally uncorrelated. This is of course not true, but gives an indication of the possible performance improvement. In this simulation we used the equalizer (6,4) and the SNR was $E_b/N_0 = 15$ dB.

4 CONCLUSION

A DFE usable for GMSK modulation is derived. For a bit rate of 300 kbits/s one must use an equalizer. Due to the short training sequence (30 bits or 15 symbols) one is forced to have the fast converging RLS algorithm. It is shown that one actually gets diversity effects due to multipath from the equalizer. The flat fading case is though very difficult to handle. One method to solve this is to use for example antenna diversity, which is shown to give very large gains especially in the flat fading case.

References

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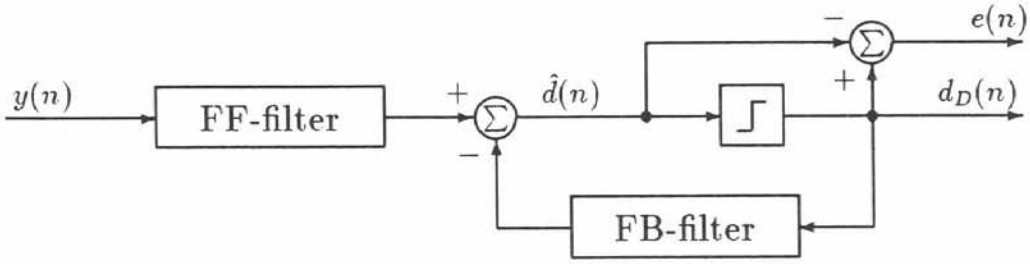

Figure 1: Structure of the Decision Feedback Equalizer (DFE)

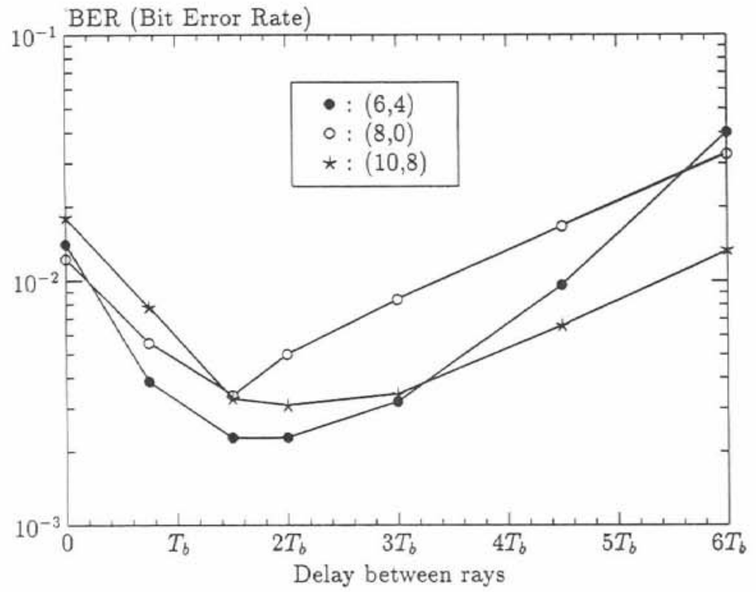

Figure 2: Different DFE-configurations, $E_b/N_0 = 18.5$ dB.

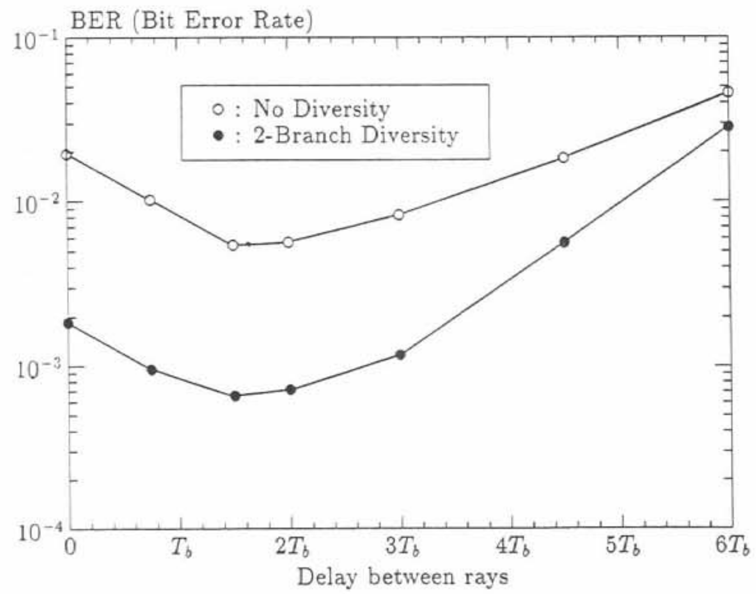

Figure 3: Antenna diversity, $E_b/N_0 = 15$ dB.