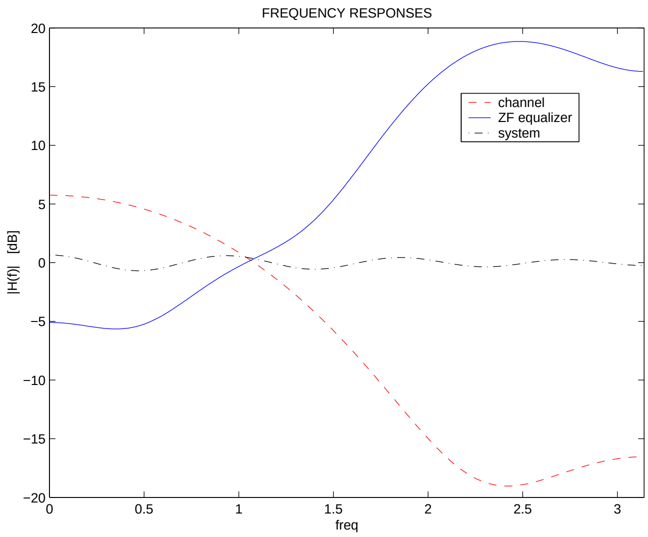
# S-72.3250

# Laboratory Works in Radiocommunications

# RECEIVER



**Prerequisites: S-72.3281 (or S-88.2211), knowledge of MATLAB.**

**See the course home page for more information.**

**Student laboratory is in the room E306.**

**Good luck!**

# RECEIVER FOR BANDLIMITED CHANNEL

## 1 Introduction

In this laboratory work we concentrate on receiver for a bandlimited channel that causes intersymbol interference (ISI) to the communications signal. The purpose is to implement a receiver in MATLAB and detect data transmitted through such a channel.

The main blocks of the receiver are channel estimation and equalization units. All practical communication systems also use channel coding, but we exclude it in this laboratory work, partly because it is a theoretical topic that is not easily fitted into laboratory work format. Furthermore, MATLAB is not the ideal tool for processing tens of millions of symbols required in error-control coding simulations.

Channel estimation can be considered a part of system identification and/or estimation theory. Estimation theory is covered in detail in [Kay93], and partly in [Koi01] and [Paj00]. In this laboratory work we only need a tiny fragment of that material, to be found in [Puk00] and as a preliminary problem.

Channel equalization is also a wide-ranging topic. In this work we treat it as a linear filtering problem. Necessary background material can be found in [Häg01] or [Laa01].

The purpose of this introductory text is not to reproduce material that can be found in hundreds of books, scientific papers, and lecture notes. Instead, certain key points, important in this laboratory work, are discussed briefly and appropriate references are given.

## 2 A bandlimited communication system (abridged)

The simplest[1] version of a receiver for AWGN channel consists of a matched filter and a decision device. If the channel induces ISI to the signal, it has to be removed before making symbol decisions. Thus, we introduce an equalizer to the receiver structure. The equalizer, however, needs to know the channel impulse response to be able to cancel its effect. Hence, we also need a channel estimator in our receiver. These blocks are briefly reviewed in the next two sections.

[1] Assume phase synchronization, symbol timing etc are known.

# 3 Channel estimation

Many types of channel estimators have been reported in literature. We can separate them in two categories: blind estimators and non-blind estimators.

Blind estimators - which are not considered in this lab work - extract the channel estimate from the received signal without any known symbols inserted in the transmitted signal. The obvious advantage is that bandwidth is saved. However, given a fixed number of basic computation operations their performance is, in general, worse than non-blind estimators'.

Non-blind estimators use *a priori* known symbols to compute an estimate of the channel IR. The known symbols are typically called training sequence, or pre/mid/postamble. For instance, in the GSM/GPRS system each transmitted burst contains 26 training symbols.

When channel noise is white Gaussian noise it can be shown that the minimum-variance unbiased estimator is linear, and the problem simplifies considerably. In this special case we can utilize a simple least squares (LS) type channel estimator for optimum performance. Reference [Puk00, pp. 3-5] and the preliminary problem P1 include the necessary background material you need about LS estimation in this laboratory work.

# 4 Zero-forcing equalizer, discrete-time case

## 4.1 General

Many types of equalizers can be used to cancel ISI of the received signal. The Viterbi equalizer[2] provides maximum-likelihood sequence out of all possible transmitted sequences, and it is thus optimal in that sense. A less-known algorithm, which minimizes the symbol-error probability, is the maximum-a-posteriori (MAP) algorithm [Pro95]. The Viterbi detector and the MAP detector are the optimal[3] detectors in error-minimizing sense but so far their computational complexities have proven to be prohibitive in applications with fast data rate.

Decision-feedback equalizer (DFE) minimizes mean-square error or peak distortion and provides reasonably good performance with less complexity than the optimal detectors. It is a nonlinear equalizer structure, and has better performance than a linear equalizer with the same optimizing criterion (MSE or ZF).

Linear MMSE or ZF equalizers are the simplest types of equalizers. In the following we concentrate on the ZF equalizer.

[2] In fact, the Viterbi algorithm gives "only" a very good approximation of the ML sequence.
[3] It is assumed that noise is white with Gaussian pdf and channel impulse response is known perfectly.

### 4.2 ZF solution for the equalizer coefficients

Continuous-time ZF solution of the equalizer transfer function has been discussed in [Häg01] and [Laa01], see the material therein. We next focus on the equivalent discrete-time ZF equalizer, which is the one used in practice.

Assumption: Channel impulse response $\mathbf{h} = \begin{bmatrix} h_{-L} & h_{-L+1} & \cdots & h_L \end{bmatrix}^T$, or its estimate, is known and it has odd length. We want to find the odd-length equalizer impulse response $\mathbf{c} = \begin{bmatrix} c_{-N} & c_{-N+1} & \cdots & c_{N-1} & c_N \end{bmatrix}^T$ such that its convolution sum with $\mathbf{h}$ produces an impulse. Thus $\mathbf{c}$ must satisfy $h(n) * c(n) = \delta(n)$. Written out as a group of linear equations we get

$$q_m = \sum_{n=-N}^{N} c_n h_{m-n} = \begin{cases} 1, & m = 0 \\ 0, & m = \pm 1, \pm 2, \ldots, \pm N. \end{cases} \tag{1}$$

In other words the impulse response of the cascade of the channel and the equalizer is forced to be an impulse, hence canceling the ISI induced by the channel. Equation (1) can be put into matrix form

$$\mathbf{Hc} = \mathbf{q} \tag{2}$$

from which the equalizer coefficients are solved by inverting the square matrix $\mathbf{H}$.

### 4.3 Example

Let $\mathbf{h} = \begin{bmatrix} 0.5 & 0.85 & 0.2 \end{bmatrix}^T$ and $N = 2$. Equation (2) becomes

$$\begin{bmatrix} 0.85 & 0.5 & 0 & 0 & 0 \\ 0.2 & 0.85 & 0.5 & 0 & 0 \\ 0 & 0.2 & 0.85 & 0.5 & 0 \\ 0 & 0 & 0.2 & 0.85 & 0.5 \\ 0 & 0 & 0 & 0.2 & 0.85 \end{bmatrix} \begin{bmatrix} c_{-2} \\ c_{-1} \\ c_0 \\ c_1 \\ c_2 \end{bmatrix} = \begin{bmatrix} 0 \\ 0 \\ 1 \\ 0 \\ 0 \end{bmatrix}. \tag{3}$$

The solution is $\mathbf{c} = \begin{bmatrix} 0.69 & -1.18 & 1.73 & -0.47 & 0.11 \end{bmatrix}^T$. Notice how zeros were appended on both sides of the vector $\mathbf{h}$ to create a square matrix $\mathbf{H}$. The system impulse response is plotted in Figure 1. The two samples on both sides of the reference sample have been forced to zero, and reference sample has unit amplitude as dictated by the equation (3). However, some ISI remains in the samples outside the equalizer's length. Adopting a larger $N$ may or may not alleviate the problem.

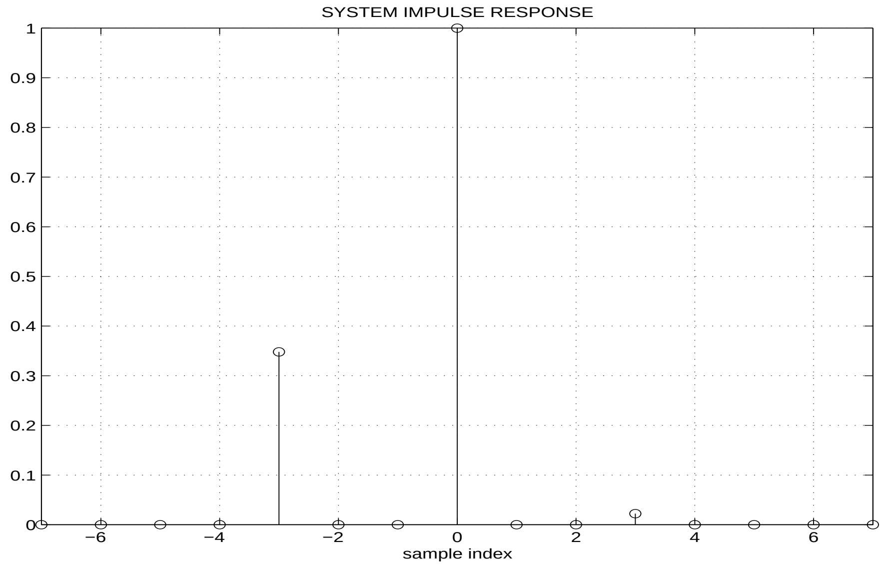



*Figure 1. The impulse response of the cascade of the channel and the equalizer.*

# Appendices

1. **General information on PROPSIM+ radio channel simulator**

2. **Technical data on PROPSIM+**

# References

**Puk00** M. Pukkila, "Channel estimation modeling", seminar presentation, 19.12.2000. Available at http://www.comlab.hut.fi/opetus/260/lab2.html.

**Koi01** V. Koivunen, Lecture notes of S-88.200, fall 2001

**Pro95** J.G. Proakis, *Digital Communications*, 3rd edition, McGraw-Hill, Singapore, 1995

**Häg01** S.G. Häggman, Lecture notes of S-72.232, spring 2001

**Paj00** P. Pajunen, Lecture notes of Tik-61.238, fall 2000

**Kay93** S.M. Kay, *Fundamentals of Statistical Signal Processing, Volume I: Estimation Theory*, Prentice-Hall, 1993.

**Laa01** T.I. Laakso, Lecture notes of S-88.211, spring 2001