On Equalization and Beamforming for Mobile Radio Applications

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Submitted in partial fulfillment of the requirements for the degree of Teknologie Licenciat.

Abstracts

Chapter 2

Design equations are presented for robust and realizable decision feedback equalizers, for IIR channels with colored noise. Given a probabilistic measure of model uncertainty, the mean MSE, averaged over a whole class of possible models, is minimized. A second type of robustification, which reduces the error propagation due to the feedback, is also introduced. The resulting design equations define a large class of equalizers, with DFE's and linear equalizers based on nominal models being special cases.

Chapter 3

Design equations are presented for a robust realizable decision feedback equalizer, for FIR channels with uncertain channel coefficients and white noise. The mean MSE averaged over the class of channels is minimized. An example using a robust DFE for a fading GSM-channel is presented.

Chapter 4

The purpose of this paper is to investigate the use of combined spatial and temporal equalization, in particular for short training sequences. The motivation for combining spatial and temporal equalization is the existence of multipath propagation and co-channel interference. Our main concern is to obtain good performance yet low complexity. We will suggest a low complexity algorithm utilizing a circular antenna array. Although it has inferior performance in an asymptotic sense, it turns out to be superior to the general solution for short training sequences. This conclusion is supported by simulations where a number of algorithms are evaluated for different scenarios involving co-channel interference.

Chapter 5

Combined spatial and temporal equalization using an antenna array combined with a decision feedback equalization scheme is investigated. In particular a TDMA type system with a relatively short training sequence is considered. Three algorithms are introduced. The first two algorithms are based on indirect schemes, where the channels to each receiver antenna element are identified. The identified channels and the correlations of the residuals are then used for the tuning of the beamformer/equalizer coefficients. The spatio-temporal correlations of the residuals are used in the first algorithm while in the second algorithm only the spatial correlations of the residuals are considered. The third algorithm forms a number of beams by using mixtures of different delayed versions of the training sequence as reference signals. It then performs temporal equalization by combining the outputs from the different beamformers, with appropriate delays. This latter algorithm requires less computations for the tuning of the equalizer, at the expense of a performance degradation in general. The algorithms are evaluated with simulations of multipath scenarios involving co-channel interference.

Chapter 6

Combined spatial and temporal equalization with a multi-antenna decision feedback equalizer is considered for a TDMA mobile radio system. Unless the time frame allocation is synchronized for all surrounding base stations, the co-channel interferers may start to interfere at arbitrary time instances during a base stations reception. A co-channel interferer may thus be present during the data sequence of the time frame while it is not present during the training sequence. In order to handle this situation, adaptation to changing interference environment is introduced. Two versions of the equalizer are considered. The covariance matrix estimates needed for tuning the equalizer parameters are partitioned into a signal part and a noise plus interference part. The signal part is derived from identified signal channels, while the noise plus interference part is constructed by only considering the spatial correlations of the residuals from the identification procedure. The first algorithm is adapted by updating the noise plus interference part of the covariance matrix estimate, combined with a retuning of the equalizer at appropriate intervals. The second algorithm performs the adaptation by tuning a number of separate beamformers. This adaptation requires considerably less computations, at the price of a potential performance degradation. For the investigated scenario, the performance degradation is however minor.

Chapter 7

Beamforming with the Sample Matrix Inversion method (SMI), using an antenna array, is considered for a GSM-signal with possible sampling offset. The problem of generation of the reference signal from the training sequence is considered. As the proper reference signal to be used varies with the sampling offset, a modified SMI method is proposed that uses a "variable" reference signal. The performance of the standard SMI beamforming method is shown to degrade when there is a sampling offset. The proposed modified SMI method, however, is shown to be immune to sampling offsets up to (at least) 0.5 symbol intervals, meaning that it will not matter where in the symbol interval the sampling takes place.

Chapter 1

Summary

1.1 Introduction

The use of wireless mobile radio communications among cellular phones grows rapidly. Since the number of users also increases rapidly, improved capacity and quality become fundamental issues, and efficient use of the allocated frequency band is crucial. In cellular systems, this is achieved by dividing the physical space into cells. Adjacent cells are allocated different frequencies, but a given frequency can be re-used by more remote cells. The received signals in cellular systems are often disturbed by co-channel interferers, i.e. by transmitters using the same frequency a few cells away. It is therefore of interest to suppress co-channel interferers. As the demand for capacity increases, it is of interest to shorten the frequency reuse distance, or even to plan for more than one mobile serviced by the same frequency in the same cell. This would increase the importance of suppressing co-channel interferers.

As the co-channel interferers are of the same nature as the desired signal, each co-channel interferer can be modeled as M-ary noise filtered through an FIR filter. Signals on neighbouring frequencies may also cause interference in the received signal. Additionally, a thermal noise component will always be present. The noise can be modeled as Gaussian, and possibly colored.

In order to combat intersymbol interference and suppress interfering noise, an equalizer is incorporated into the receiver. The purpose of the equalizer is to estimate the transmitted symbols from the received signal, in spite of intersymbol interference and interfering noise. Two possible ways to accomplish this is to either use a decision feedback equalizer (DFE) or a maximum likelihood sequence estimator (MLSE).

The multiplexing between different users can be realized with different schemes. In this report, the case of time division multiple access (TDMA), on narrow band channels, is generally considered. Each user is here assigned a time slot, during

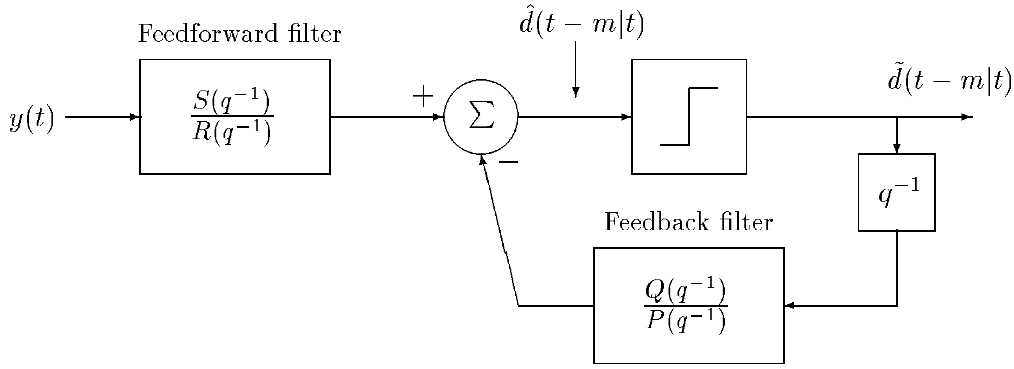

Figure 1.1: Decision Feedback Equalizer (DFE). The equalizer consists of two linear filters, and a static nonlinear decision device. The filters are here represented as rational functions in the backward shift operator q^{-1}.

which data is transmitted in a burst. The burst typically consists of an unknown data sequence combined with a known training sequence. The training sequence can be used for tuning of the equalizer.

1.2 The Decision Feedback Equalizer (DFE)

The DFE consists of a feedforward and a feedback filter, see Figure 1.1. The DFE:s considered here is confined to have linear filters. For a given user selected smoothing lag, m, the feedforward and the feedback filters are tuned such that their combined output, $\hat{d}(t-m|t)$, constitute a minimum mean square error (MSE) estimate of the transmitted symbol $d(t-m)$. Based on $\hat{d}(t-m|t)$, estimates of the discrete transmitted symbols, $\tilde{d}(t-m|t)$, are decided in a decision device, by choosing the discrete symbol closest to $\hat{d}(t-m|t)$. The decided symbol estimates, $\tilde{d}(t-m|t)$, are then fed back into the feedback filter.

The DFE is intended to work in the following way. The feedforward filter is tuned such that the combination of the transmission channel and the feedforward filter, has impulse response coefficients before tap number m close to zero, while tap number m should be close to 1. The impulse response after tap m can be of arbitrary shape, since it will be cancelled by the feedback filter (if all decided symbols are correct). The design of the filters is a compromise between performing equalization (combating intersymbol interference) and suppressing interfering noise.

A drawback with the DFE is that it normally performs decisions on a symbol-by-symbol basis rather than on sequences of symbols. A decision error may also cause additional errors, and lead to a burst of errors.

1.3 Maximum Likelihood Sequence Estimation (MLSE)

The MLSE estimates the transmitted sequence by estimating the most likely transmitted symbol sequence, based on the whole sequence of received signal samples. If the interfering noise is white, estimation of the transmitted sequence is conceptually easy. We can compute the likelihoods for every possible transmitted sequence and select the sequence with the largest likelihood. These likelihoods are obtained as the product of the probabilities of receiving each signal sample, given that particular transmitted sequence. Although the method is conceptually simple, it would be next to impossible to compute the estimate without the Viterbi algorithm. The Viterbi algorithm utilizes the fact that the channel for the transmitted signal is of finite length, much shorter than the symbol sequence (burst). The complexity of the algorithm is reduced to computing NM^L probabilities and updating the likelihoods of possible sequences. Here N is the number of data symbols, M is the number of symbols in the alphabet and L is the number of taps in the channel, see [?]. Without the Viterbi algorithm, M^N likelihoods would have to be computed.

If the interfering noise is colored, the received signal can first be whitened, if the interfering noise color is known. The noise at the output of the whitening filter will then be white. A MLSE detector can subsequently be applied to the new signal. The channel will now, however, consist of the original one combined with the whitening filter. Usually this new channel will have a larger number of taps than the original one. This will increase the complexity of the Viterbi algorithm.

1.4 The DFE versus MLSE

If the noise is white and if the correct noise distributions are used in the calculations of the probabilities, the MLSE is optimal. The DFE equalizer is optimized to give minimum mean square error before the decision device (here restricted to linear filtering). The estimated transmitted sequence is then formed on a symbol-by-symbol basis from the signal, ocuring before the decision device. This is in general a suboptimal procedure, as compared to the MLSE. The DFE can, however, be modified such that its decisions are based on more than one data sample, employing a MISO or a MIMO decision device. See for example [?]. The DFE can also be complemented with schemes guarding against decision errors. This can improve the performance of the DFE, at the expense of an increased amount of computations. See for example [?].

The computational complexity of the MLSE equalizer, implemented with the Viterbi algorithm grows as M^L. As the length of the channels increase, the complexity of the Viterbi algorithm grows exponentially. The computational complexity for executing the DFE, however, increases only linearly with the channel length. For channels with many taps, the complexity of the Viterbi algorithm will be considerably higher than for the DFE.

In the case of colored noise, one can for the MLSE choose to whiten the noise as described above, before applying the MLSE. However, as the whitening alters and usually increases the length of the channel, the complexity of the Viterbi algorithm will increase and the combined filtering is no longer necessarily optimal. The DFE on the other hand can handle colored noise directly, with optimal linear filtering minimizing the MSE before the decision device.

Another drawback with the MLSE *may* appear in the case of adaptation. The nature of the MLSE algorithm suggests that it may require a longer smoothing lag in order to perform better than the DFE. If this is the case, then it could be advantageous to use the DFE in situations where adaptation to a time varying scenario is of interest.

The above discussion justifies further investigation of the use of the DFE.

1.5 Robust Decision Feedback Equalizers

For time varying channels, improved results can be obtained by adapting the DFE to the channel variations. If the channel variations are small, however, one can instead of adaptation choose to design the DFE such that it works well for a wider range of channels, covering the expected time variations. In chapter 2, the design of DFE:s that are robust with respect to uncertainties in the channel and noise models and to uncertainty in the correctness of the decided feedback data, is discussed.

Design equations are presented for robust and realizable decision feedback equalizers, for IIR channels with colored noise. Given a probabilistic measure of model uncertainty, the mean MSE, averaged over a whole class of possible models, is minimized. A second type of robustification, which reduces the error propagation due to the feedback, is also introduced. The resulting design equations define a large class of equalizers, with DFE's and linear equalizers based on nominal models being special cases.

In chapter 3, the special case of designing decision feedback equalizers that are robust with respect to time variations in the channel is discussed. The particular case of an FIR channel, typical for mobile radio applications, is discussed. The performance of the robust decision feedback equalizer is exemplified for a GSM-channel with typical time variations.

1.6 Combined Temporal and Spatial Equalization

If only one antenna element is available, it is only possible to perform temporal equalization to estimate the transmitted sequence. If several antenna elements are available it becomes possible to achieve beamforming by modifying the phase and amplitude of the received signals at the individual antenna elements prior to summation. It is then possible to perform spatial filtering by forming beams in the direction of a desired signal. Noise and interference and also delayed signals which would result in intersymbol interference, can be suppressed if they arrive from other directions. This can be viewed as spatial equalization. The beamforming concept can be combined with temporal equalization, resulting in spatio-temporal equalization. It is then possible to effectively make use of the energy in delayed signals arriving from several directions while suppressing the signals from co-channel interferers. Spatio-temporal equalization can be performed by generalizing the single-input-single-output (SISO) DFE to a multiple-input-single-output (MISO) DFE. Arrays of antenna elements are only practical at base stations. This technique is therefore primarily applicable in the transmission from the mobiles to the base stations.

In chapter 4 and chapter 5 the use of a MISO DFE for spatio-temporal equalization is investigated. In particular, the case of TDMA systems with relatively short training sequences is studied.

A MISO DFE will by necessity contain a larger number of adjustable parameters than a SISO DFE. This leads to two potential problems.

- The adjustment of many filter parameters, based on short training sequences, is sensitive to noise. Misadjustments may lead too poor performance.

- The computational complexity of the algorithm will increase.

Two key issues are therefore investigated in chapter 4 and chapter 5, for different filter structures and different filter adjustment schemes: The performance of filters designed from short training data series and the computational complexity of the algorithms.

When tuning equalizers to suppress co-channel interferers by the use of a training sequences, a problem arises. A co-channel interferer may not be present during the training sequence, but start its transmission during the data sequence. The equalizer may then perform poorly when this new interferer appears. In chapter 6, the use of adaptation for MISO DFE:s, in order to suppress such co-channel interferers, is discussed.

1.7 Beamforming with the SMI Method for a GSM Signal with Sampling Offset

Consider a receiver with several antenna elements. Beamforming can then be used in order to optimally receive the desired signal while suppressing co-channel interferers. One method for doing this is to use the sample matrix inversion method (SMI), [?]. In this method the weights corresponding to the different antenna elements are adjusted in order to make the resulting received signal as close as possible to a reference signal. In a TMDA system, the training sequence can be used for constructing a reference signal. The GSM system, however, uses Gaussian minimum shift keying, which results in intersymbol interference already at the modulation stage. A consequence is that the transmitted signal, resulting from a known symbol sequence, is not completely known at the receiving end, unless the exact timing of the sampling is known. The design of a reference signal corresponding to the transmitted signal is thus ambiguous.

In chapter 7, the problem of performing beamforming with a modified sample matrix inversion method for a GSM signal with sampling offset, is addressed.

1.8 Publications

The contents of chapters 2 to 7 are taken from the following publications:

Chapter	*Reference*
2	M. Sternad, A. Ahlén and E. Lindskog, “Robust decision feedback equalizers”, in *Procceedings of Int. Conf. on Acoustics, Speech and Signal Processing*, vol. 3, Minneapolis, Minnesota, U.S.A., April 1993, pp. 555-58.
3	E. Lindskog, M. Sternad and A. Ahlén, “Designing decision feedback equalizers to be robust with respect to channel time variations”, in *Proceedings of Nordic Radio Symposium seminar*, Uppsala Sweden, November 10-11 1993.
4	E. Lindskog, A. Ahlén and M. Sternad, “Combined spatial and temporal equalization using an adaptive antenna array and a decision feedback equalization scheme”, in *Proceedings of Int. Conf. on Acoustics, Speech and Signal Processing*, Detroit, Michigan, U.S.A., May 8-12 1995.
5	E. Lindskog, A. Ahlén and M. Sternad, “Spatio-temporal equalization for multipath environments in mobile radio applications”, in *Proceedings of the 45th IEEE Vehicular Technology Conference*, Rosemont, Illinois, U.S.A., July 26-28 1995, To appear.
6	E. Lindskog, “Indirect spatio-temporal equalization and adaptive interference cancellation for multipath environments in mobile radio applications”, in *Proceedings of IEEE/IEE Workshop on Signal Processing Methods in Multipath Environments*, Glasgow, UK, April 20-21 1995, pp. 115-24. Revised version.
7	E. Lindskog, “Beamforming with the sample matrix inversion method for a GSM signal with sampling offset”, Submitted to The Sixth International Symposium on Personal, Indoor and Mobile Radio Communications, Toronto, Canada, 27-29 September 1995.