Chapter 1

Introduction

1.1 Background

First generation mobile communication systems (e.g. NMT and AMPS) are based on analog transmission techniques, whereas second generation systems (e.g. GSM and D-AMPS) are digital. In digital systems, more efficient use of the available spectrum is achieved by digital encoding of the speech data. Third generation systems are now under investigation, providing services like wireless access to the Internet and high data rate applications like real time video transmission. To cope with these high-bandwidth services and the enormous increase in the number of users, more efficient use of the radio spectrum is required. One proposal is to exploit the spatial dimension by means of smart antennas. The concept of smart antennas is viewed as a means to obtain significantly improved spectral efficiency, better quality of service and higher capacity, while at the same time achieving considerable savings in base station costs [41]. International projects like the European Community-funded TSUNAMI II aim to identify, evaluate and develop adaptive antenna technologies applicable to third generation systems [14,15].

The introduction of array antennas at the base station site can be exploited in different ways: The power transmitted by the terminal can be decreased significantly due to the higher directivity provided by the array antenna. Also, the frequency reuse distance in the cellular system can be decreased by active suppression of co-channel interference from adjacent cells. Even fully developed SDMA (spatial division multiple access) based systems may be an option. More than one user can then be allocated to the same frequency and timeslot in a single cell, provided that the users are assigned different training sequences.

In 1994 the Signals and Systems Group at Uppsala University, together with Ericsson Radio Access AB commenced a project concerning adaptive antennas, sponsored by NUTEK (the National Board for Industrial and Technical Development) and Ericsson Radio Access AB. The aim of the project was to design an adaptive antenna for the uplink according to the DCS-1800 standard, and to investigate the benefits and drawbacks of SDMA. To be able to use commercially available base stations, the beamforming was implemented in hardware by means of microwave phase shifters and attenuators, in conjunction with passive combining. Two parallel SDMA-channels where implemented. During the summer and fall of 1996, the performance of the adaptive antenna was evaluated by laboratory measurements and outdoor field-trials. A description of these experiments and the obtained results form a part of the present thesis.

The practical work on adaptive antennas has highlighted several algorithmic issues which require further investigation: How can limited training data and *a priori* information best be utilized to tune an adaptive array with high accuracy? How do non-idealities in the array implementation affect the performance of the array, and how can such effects be reduced by automatic calibration? The objective of a major part of the thesis is to provide partial answers to these questions.

1.2 Outline

For mobile communications, equalization based on indirect methods turns out to be superior as compared to direct methods: In general, the number of channel parameters are less than the number of equalizer parameters. Also, with a fading mobile channel, the changes in the equalizer parameters are more abrupt than the changes in the channel parameters, see e.g. [46]. To perform equalization based on indirect methods, an estimate of the channel from the mobile to the base station is required. Often the channel is modeled as an FIR-filter, and is identified in a least-squares fashion utilizing an *a priori* known stream of symbols, the training sequence. In Chapter 2, a method for multi-user estimation, i.e. the simultaneous estimation of channels from several users, is presented. Making the assumption that we know the pulse shaping in the transmitter and in the receiver filters, this information can be utilized in order to improve the channel estimate. Only the ”air-interface” part of the channel then needs to be modeled and identified, utilizing the pulse-shaped symbols as input to the system. Due to the multipath nature of the channel, it is not likely that the delayed versions of the transmitted signal will arrive exactly at the sampling instants. Delayed versions of the pulse shaping filter are therefore used in the channel model. Multipath components arriving in between the sampling instants may then be approximated by a linear combination of the pulse shaping functions.

The channel estimate can be further improved by parameterizing the multipath components of the channel in terms of angles and relative gains. This is the topic of Chapter 3. The channel estimate of Chapter 3 is obtained by a projection in a spectrum norm sense onto the parameterized subset of the set of impulse response coefficients (channel taps). The proposed method is compared to other identification methods in terms of BER, when utilized in a multidimensional MLSE detector.

In order to increase the capacity of cellular systems, the spatial dimension can be utilized by means of an array antenna. One approach is to do spatio-temporal equalization, using for example a multidimensional MLSE detector, implemented via the Viterbi algorithm [43]. Another approach is to form the beampattern in an adaptive manner. Equalization is then performed on the beamshaped signal. In Chapter 4 an adaptive antenna built for uplink use according to the DCS-1800 standard is evaluated both by laboratory measurements and by outdoor field-trials. The beampattern of the adaptive antenna is formed using the SMI- (sample matrix inversion) algorithm and a hardware beamforming network.

The methods for channel identification presented in Chapters 2 and 3 can be implemented in the digital parts of the adaptive antenna system described in Chapter 4. If the signal processing capacity is upgraded, it is also possible to implement a real-time spatio-temporal equalizer utilizing the estimated channel. Another possibility is to utilize the estimated channel to do indirect beamforming. The hardware beamforming network can then be used.

Practical implementation of adaptive antennas is associated with several quantizations of the involved signals. Prior to the signal processing, the signals of the antenna elements must be sampled and digitized by A/D converters. The weighting and combining of the signals will also introduce quantization errors especially if the beamforming is accomplished in hardware (digitally controlled phase shifters and attenuators). These effects are investigated in Chapter 5 by means of theoretical analysis, simulations and measurements using the antenna described in Chapter 4.

Some algorithms based on angle estimation need a calibrated array. This is not the case using the SMI-algorithm. The antenna described in Chapter

4 does however require a calibration since the beamforming is performed in hardware. The weights are calculated based on the signals at the A/D converters, but are applied to the signals at the hardware weighting units. Thus, the attenuation and phase shift between the A/D converters and the weighting units need to be calibrated for. Since active components in receivers and weighting units tend to drift with temperature, the calibration prior to operation will not be optimal after some time. In Chapter 6 two methods are proposed to mitigate this problem. The first method utilizes the weights calculated by the SMI-algorithm to form a reference signal. An LMS-like algorithm then adjusts the hardware weights so that the output of the adaptive antenna follows the reference. In the second method proposed, a more traditional identification approach is taken. There the drift due to temperature is tracked, and utilized when the SMI-weights are to be steered out to the hardware weighting units. The performance of the proposed algorithms are investigated by means of simulations.

The results of the investigations performed in Chapters 2 through 6 are discussed and concluded in Chapter 7.

1.3 Contributions

The material presented in this thesis has been discussed previously in the papers presented below. The main parts of Chapters 2 and 3 can be found in the conference papers

E. Lindskog and J. Strandell, ”Multi-User Channel Estimation Exploiting Pulse Shaping Information”, *9th European Signal Processing Conference*, Island of Rhodes Greece, 8-11 Sept. 1998. Submitted.

J. Strandell and E. Lindskog, ”Separate Temporal and Spatial Parametric Channel Estimation”, *9th European Signal Processing Conference*, Island of Rhodes Greece, 8-11 Sept. 1998. Submitted.

The material presented in Chapter 4 can be found in the conference papers listed below, where the first paper presents laboratory measurements and some of the field-trial results. Some conclusions on the benefits of using the adaptive antenna in a cellular system are also presented in terms of the spectral efficiency gain. The second paper is more focused on the outdoor field-trials.

J. Strandell, M. Wennström, A. Rydberg, T. Öberg, O. Gladh, L. Rexberg, E. Sandberg, B. Andersson and M. Appelgren, ”Experimental Evaluation of an Adaptive Antenna for a TDMA Mobile Telephone System”, in *Proceedings of the IEEE Personal Indoor and Mobile Radio Communications conference,* Helsinki, Finland, pp. 79-84, 1997.

J. Strandell, M. Wennström, A. Rydberg, T. Öberg, O. Gladh, L. Rexberg and E. Sandberg, ”Design and Evaluation of a Fully Adaptive Antenna for Telecommunication Systems”, in *Proceedings of the Antenn 97 conference*, Gothenburg, pp. 357-366, 1997.

The material of Chapter 5 has been submitted for publication as

M. Wennström, J. Strandell, A. Rydberg and T. Öberg, ”Analysis of Quantization Effects in Adaptive Antennas for Cellular Systems”, submitted to *IEEE Transactions on Vehicular Technology*.