Implementation of a Smart Antenna System with an Improved NCMA Algorithm

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Abstract — **This paper presents the possibility of using adaptive algorithms for digital beamforming purposes. A normalized constant modulus algorithm (NCMA) is implemented in a standard FPGA. In this way, a simple and non-hardware-intensive, smart antenna system in combination with a software-defined radio (SDR) has been realized for mobile FM reception. A new kind of algorithm initialization, leads to an improvement in startup behavior. The quality in signal separation makes the NCMA algorithm also suitable for MIMO purposes. The NCMA algorithm increases the reception quality for mobile communication systems dramatically.**

I. INTRODUCTION

A common problem of mobile communication systems is the influence of multi-path propagation and the Doppler effect that leads to the deep fading effect. The quality of reception crucially depends on the propagation channel's condition. Fig. 1 shows a typical receiving characteristic of a channel with several scattering objects. It can be seen that the quality of reception changes enormously with the position of the receiver.

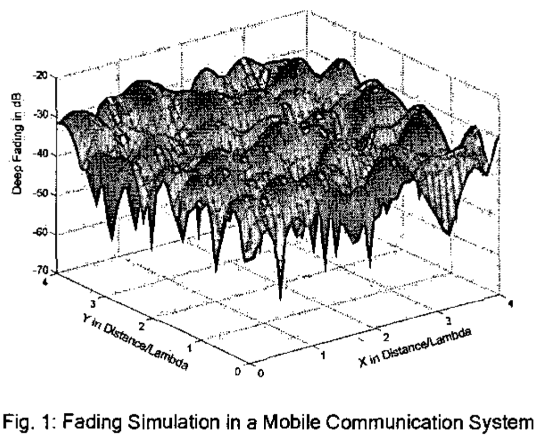

Fig. 1: Fading Simulation in a Mobile Communication System

In conventional analog techniques, like mobile FM radio reception, the antenna diversity [5] is a solution for this problem. This technique is nowadays implemented in many cars. There are several receiving antennas mounted on different positions on the car, so that a controller can switch to the antenna with the best receiving level. Other solutions like smart antennas, beamforming techniques and adaptive algorithms are almost impossible to implement with common analog techniques.

The development of digital receiver techniques, like SDR (software defined radio), nowadays allows the practical realization of smart antenna systems with all imaginable beamforming and equalizer techniques. The big advantage is that the information from all antennas can be used, so that the combined receiving signal is better than the signal from each single antenna.

This paper describes the implementation of an adaptive beamforming algorithm for mobile FM radio reception. Furthermore, it shows the possible enhancement of signal reception that can be achieved using this adaptive digital beamforming system. The algorithm and a software defined radio are implemented in a standard FPGA.

II. SDR – SOFTWARE DEFINED RADIO

The demand for communication interfaces between a car and the environment increases as a result of the large number of stationary and mobile applications. In addition there is a need for a worldwide uniform car equipment in order to reduce costs. All this leads to the need for a software configurable platform that is capable of handling standards like AM, FM, GSM, UMTS, digital broadcasting standards (DAB, Sirius, XM-Sat Radio), analog and digital television and other data links. Only a software configurable receiver can decrease cost, size and volume while increasing the number of receivable radio standards.

The use of full digital reception techniques can also lead to an enhancement of signal quality, because new algorithms like channel estimation, channel diversity [6] and MIMO like techniques can be used even for older analog standards.

The presented algorithms are implemented on an RF sampling receiver for FM reception [3]. The analog input stage consists of a low-noise amplifier with tunable amplification and an anti-aliasing filter with a passband

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of approximately 85 to 115 MHz. The signal is sampled at a frequency of 80 MS/s so the signal appears after sampling between 5 and 35 MHz (see Fig. 2). The downconversion, decimation and channel filtering is performed in the digital domain.

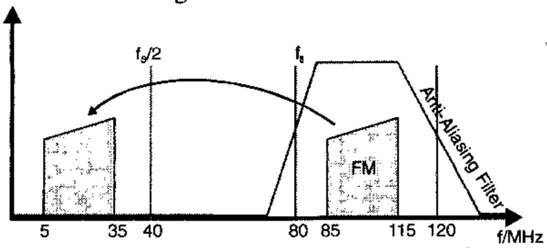

Fig. 2: Frequency plan of the RF sampling receiver

For the reception of the signal from multiple antennas this sampling head is replicated several times. Each data path downconversion and decimation is implemented separately. The algorithms described in the following sections use the complex baseband signal.

III. DIGITAL BEAMFORMING ALGORITHM

The main purpose of digital beamforming is to add or combine coherently the received signal from the different antennas, in a way that the effects of multipath propagation and Doppler are eliminated [8]. It is possible to create an antenna characteristic in a way that only the wanted beams are received and the unwanted ones are suppressed. Therefore, the received and digitized complex input signal $x_i(k)$ from each antenna i is weighted with a complex factor, the so-called weighting factor w_i or weighting vector w (see equation (2)). This makes it possible to shift the phase and to vary the amplitude of the receiving signal x_i or vector x.

$$x(k) = \begin{pmatrix} x_1(k) \\ x_2(k) \\ \vdots \\ x_n(k) \end{pmatrix} \quad ; \quad w(k) = \begin{pmatrix} w_1(k) \\ w_2(k) \\ \vdots \\ w_n(k) \end{pmatrix} \tag{1}$$

$$y(k) = w^H(k)\,x(k) \tag{2}$$

In a mobile environment, where all the characteristics are changing dynamically, the weighting vector of the antenna has to be recalculated successively.

For modulation types like FM radio, where no training sequence or reference signal for spatial channel estimation is provided, a blind beamforming algorithm has to be chosen. These blind algorithms use the so called property restoring method, which means they use a known property like the constant modulus of the modulation type and try to restore it.

The classical formulation of the CMA algorithm [7] is used for time domain filtering the blind equalization. The constant modulus algorithm family is suited for all signals with a constant complex envelope like FM-, PM-, PSK-, or FSK-modulation. The algorithm uses the magnitude, which is not used for information purposes, to calculate a cost function. A main advantage of the constant modulus family is its independence of antenna arrangement.

Two CMA formulations, the LSCMA [1] and NCMA [2] are very suited for spatial filtering purposes, i.e. adaptive beamforming.

The normalized CMA algorithm (NCMA), which offers an essential improvement in stability and adaptation speed, was firstly presented by Hilal and Duhamel in 1992 [2]. The stability of NCMA depends not on the input signal as is the case for CMA. Another advantage of NCMA is its simplicity of implementation in hardware.

LSCMA (least-square-CMA) [1], which offers even a higher adaption speed than NCMA, is not chosen because of the significantly higher processing power required for matrix inversions. Furthermore, LSCMA is prone to numerical instability, and, in addition, a noise capture phenomena may occur if the input SNR is low.

IV. THE NCMA – ALGORITHM

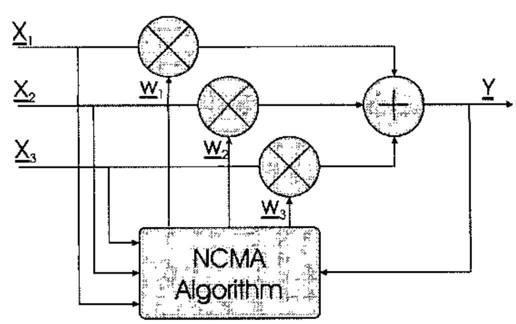

Fig. 3: Schematic Illustration of the NCMA Algorithm

The new weighting vector calculation for the NCMA algorithm is given in equation (3) and a schematic illustration of the algorithm can be seen in Fig. 3.

$$w(k+1) = w(k) + \mu \frac{x(k)}{x^H(k)\,x(k)} \frac{y^*(k)}{|y(k)|}\left(1 - |y(k)|\right) \tag{3}$$

The algorithm uses a stochastic gradient method to minimize the cost function (1 - |y(k)|). The step size μ offers the possibility to control the convergence speed and the mean CM rest error in a specific range. The

NCMA works in a stable manner with $0<\mu<2$ and reaches its maximum convergence speed with $\mu=1$ [4].

A main task is the initialization of the NCMA algorithm. A wrongly chosen vector w(0) may lead to noise capturing over several seconds. Simulations showed up that this behavior depends on the constellation of the input signals. One solution for this problem is to normalize the input vector with the highest possible input value, so that the magnitude of the normalized input signals is smaller than one.

Another possibility is to pre-calculate the weighting vector initialization (4) in a way that the algorithm starts up in convergence.

$$w(0)=\frac{x(0)}{x^{H}(0)\,x(0)} \tag{4}$$

In this way, the cost function is zero for the first input vector and the algorithm is in convergence. Simulation showed that this is the fastest startup, is independent of the input values and no noise capturing occurs.

V. Results

First of all, a vector channel model was created in MATLAB to verify the algorithm. The result showed an excellent increase in reception quality. In order to verify the theoretical results, the algorithm was tested with real data provided through several test drives under different conditions. The test drives were done with a measurement van, where three $\lambda/4$ monopole antennas were positioned at different triangular distances on the roof. The reception was done with three software radios, which were connected to a logic analyzer for data recording.

The recorded data was then verified under MATLAB. The demodulated signal of the three input antennas and the combined signal were investigated. The signal shown in Fig. 4 was recorded by driving through a tunnel at 80 km/h. On all three antennas crackle interference can be heard, but on the combined signal all of the interference is eliminated.

This excellent increase in reception quality is shown by the signal spectrum of the demodulated signals in Fig. 4. An improvement of around 18 dB SNR can be seen between the best receiving single antenna and the combined signal. Signal power is assumed to be the power of the 19kHz carrier, whereas the noise power is assumed to be the noise power measured between 16 and 18kHz. This is not the real signal-to-noise ratio for FM transmission, but in this case only the improvement relative to the received signals needs to be shown.

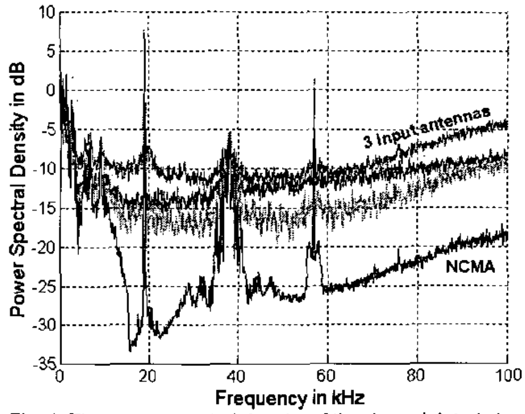

Fig. 4: The power spectral density of the demodulated signal

The visualized data show the average improvement over the measurement period and not at discrete times. Therefore, the distribution function of the signal-to-noise ratio is calculated. The cumulated distribution function of the same data set is shown in Fig. 5. The probability to have less SNR can be seen as percentage of time in which the SNR is below the indicated value.

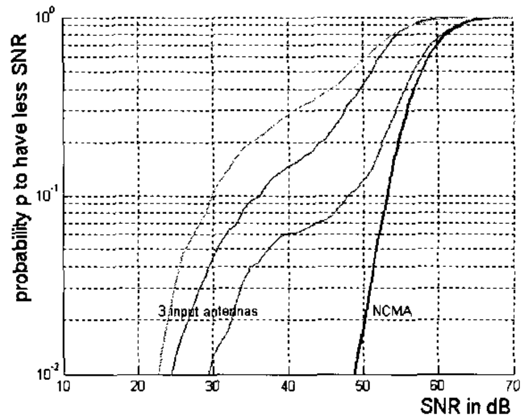

Fig. 5: The improvement with three worse receipted signals

One can see for example that the probability to have a SNR of less than 50dB is around 2% for the combined signal. The SNR from the best antenna is less than 50dB about 12% of the time, whereas the signal of the worst antenna has a SNR of less than 50dB for 60% of the time.

On the other hand, it is also important what happens with excellent received signals. As can be seen in Fig. 6, the combined signal is slightly better than the signal from the best antenna. The data for Fig. 6 were recorded static with a line of sight connection to the FM transmission station, in order to get a transmission condition with nearly no fading.

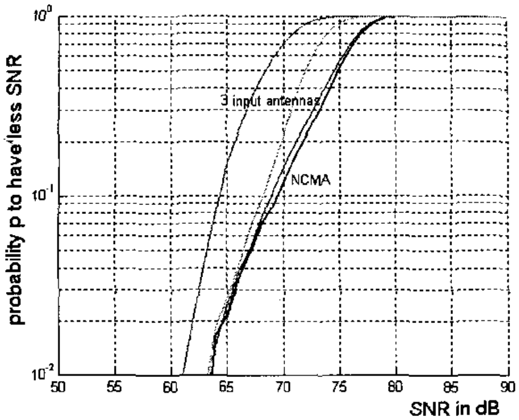

Fig. 6: The improvement with three excellent received signals

Similar increase in reception quality has been achieved with all test drives under different conditions and with different antenna arrangements.

VI. Implementation in an FPGA

There are several possibilities to implement the NCMA algorithm in hardware. The FPGA (field programmable gate array) solution has several advantages over a DSP implementation. First of all, one has only to implement the required hardware for the special purpose. Furthermore, the final implementation is very cheap, because it is only a small step to go from a FPGA solution to an ASIC.

The main task of the implementation is not to deal with the accuracy, but to handle the needed data range. The algorithm contains several complex multiplications in series, where the result can be very small or even very large. Therefore, the easiest method of implementation is to use floating-point numbers. Thus, it is not a simple push-button implementation. A modular DSP structure has been implemented in the FPGA, with only the necessary commands. The big advantage for this solution is that only the required hardware has to be realized and that the modular structure offers the possibility to change or extend the algorithm relatively easily.

The final implementation of the whole algorithm for three receiving antennas was done within 100k Gates in a standard SPARTAN II FPGA from XILINX. The verification of the solution showed similar behavior to the MATLAB simulation of the algorithm.

In this way, the enlargement for more antennas needs only more chip area, because the calculations are paralleled.

VII. Conclusion

The implementation of an adaptive beamformer for a mobile communication system with a NCMA algorithm has been presented. The pre-calculation of the initial weighting vector, improves the algorithm's startup behavior enormously. The excellent quality of the smart antenna system in increasing reception quality has been shown by using an analog broadcast standard. Scaling to microwave frequencies is straight forward.

The implementation has been done in a standard FPGA, so that the system can be realized very efficiently and cheaply. In this way, a one-chip solution of beamformer and software radio is possible.

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