# THE INFLUENCE OF COHERENT SIGNALS ON ALGORITHMS FOR ADAPTIVE ANTENNAS IN MOBILE COMMUNICATION

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## ABSTRACT

In this paper a few different algorithms for adaptive beamforming are compared. The algorithms are LMS, RLS, SMI, MUSIC and WSF. Computer simulations are carried out to determine the algorithms robustness against fading. The simulations show that the algorithms differ when the received signal (desired or interfering), from each transmitter, consists of several coherent signals that origins from the vicinity of the transmitter. This is a very common case in a mobile telephony system in urban areas.

## 1. INTRODUCTION

An adaptive antenna consists of an array of spatially distributed elements. Impinging on the array are waveforms from several transmitters at different locations. By properly combining the antenna outputs, it is possible to extract individual signal waveforms from the received superposition. Even if the signals occupy the same frequency band they can be extracted if they have different angles of arrival to the antenna array. The operation may be seen as a spatial (angular) filtering [4] and [12].

Mobile communication scenarios in urban areas usually involve many coherent multipaths from each transmitter. The number of reflections may be much larger than the number of resolvable direction-of-arrival's (DOA) [1] and [5].

Adaptive antennas may be used at the base-station and/or at the mobile-unit. In this paper we consider an adaptive receiving antenna based at the base-station.

## 2. ANTENNA ARRAY

Multiple antenna elements can be used in mobile communication as a diversity scheme to reduce fading [9]. The elements must then be spaced sufficiently apart (>10 wavelengths) in order to obtain signals at different elements that fade independently. Unlike diversity antennas, we consider an array of antennas spaced sufficiently close (about half a wavelength) in order to obtain signals at different antennas that differ only by a phase factor.

In the simulation study we have considered a linear array consisting of *ten* (10) omnidirectional antenna elements at a relative distance of half a wavelength.

## 3. SIMULATION MODEL

Consider a typical mobile-radio communication link that consists of an elevated base-station antenna and an antenna mounted on a mobile. Due to terrain configuration and man-made obstacles in the vicinity of the mobile, there is seldom a direct line-of-sight propagation path between the mobile and the base antenna. Hence multipath propagation is a predominant characteristic, where signal transmission is through always changing multiple paths between the moving vehicle and the fixed base-station.

The signals received at the base-station from a mobile unit can be considered as resulting mainly from reflections of the radio waves. If all multipaths stem from the vicinity of the mobile the receiver level varies quickly as the mobile moves, according to the phase differences among the reflections. This is often referred to as *Rayleigh fading* (short-term fading) [8]. Variations in the local terrain give rise to a more slowly time-varying trend, *log-normal fading* (long-term fading) [8]. Herein, only the effects of Rayleigh fading are considered. To consider all of the reflections to origin from the vicinity of the mobile is a realistic assumption since the base-station antenna usually is mounted high above all possible reflectors (this assumption is not true in a micro-cell). Each reflection is travelling its own path with its specific distance. The differences in path-distance in these simulations are not big enough to give rise to inter-symbol interference.

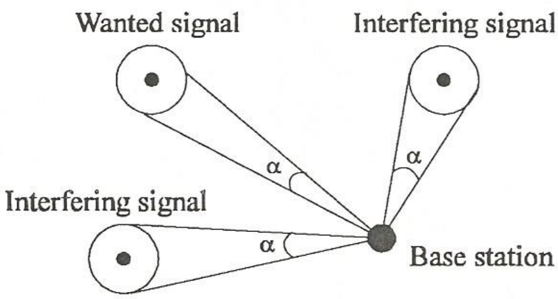


Fig.1 Model used in the simulation study. The signal from each transmitter consists of a number of the reflections from randomly distributed places inside a circle surrounding the transmitter. The reflection angle $\alpha$ is defined.

In the simulation study, *one* desired signal and *two* interfering signals are considered. The received signals, from each transmitter (desired or interfering) at the base station is approximated by a sum of a finite number of coherent reflections, with random phases and amplitudes which give rise to fading. As mentioned above the reflections are assumed to accrue nearby of the mobile. The reflection points are randomly distributed inside a circular surrounding the mobile, see fig.1. Since the mobile is moving the received waveform at the base-station will wary in time.

In a stationary surrounding it is the velocity of the mobile that gives rise to the fading phenomena. The parameters deciding the fading are the frequency of the signal, the velocity of the mobile, the number of reflections and the DOA's of the signals to the base-station. In the simulations the frequency is set to 1800 MHz, the number of reflections of the signal from each transmitter is *twenty* (20). The velocity of the mobile and the radius of the reflections circle are altered. The simulations also take the doppler effect [8] into account.

In the simulations the transmitted signal is GMSK modulated with a flow of 270kbit/s. The sampling rate is set to 270 kHz i.e. the array elements are sampled every 3,7 $\mu$s.

## 4. ALGORITHM CLASSES

The adaptive algorithms considered herein can be divided into classes [2].

The reference signal methods requires a replica (a reference signal) of the desired wave. If the wanted signal is known (i.e. one have some knowledge of the signal that makes it unique) a signal similar to the desired can be used as a reference when calculating the weights. The reference signal could be the TDMA synchronisation sequence (the training sequence) or the code sequence in a CDMA system. Examples of algorithms using a reference signal are the Least Mean Square (LMS, [1]) and the Recursive Least Square (RLS, [1]) algorithm. since these algorithms are using a reference signal the DOA can be unknown. LMS is the algorithm that has the lowest computational requirements but the convergence rate is much lower than e.g. RLS.

Another class of algorithms is based on high-resolution direction finding. The number of impinging signals is estimated and the DOA's can then be computed. These angles are used to construct the beamformer. In our case we use the Linear Least Squares Estimate (LLSE). The term high-resolution origins from the fact that these algorithms can resolve different transmitters even within the conventional lobe-width. Examples of algorithms are the MUltiple SIgnal Classification (MUSIC, [7]) and the Weighted Subspace Fitting (WSF, [6]) algorithms. These algorithms have much higher computational requirements than the reference signal methods and the array manifold has to be familiar. If the array manifold ([6]) is unknown it is necessary to calibrate the array. Even small errors in the calibration procedure may considerably degrade the performance.

An algorithm that is very simple is the Sample Matrix Inversion (SMI, [1]) algorithm. SMI can be implemented as a reference method or without a reference signal if the DOA of the wanted signal is known in advance. In the simulation study the SMI algorithm was implemented in a recursive form as in [10] with a reference signal.

## 5. IMPORTANT PARAMETERS IN DIFFERENT SYSTEMS

There are three dominant classes of telecommunication systems used today - FDMA, TDMA and CDMA.

In a FDMA system e.g. NMT, each mobile transmits at an unique frequency. The signals are relatively continuous which results in a more static environment than in a

TDMA system. A drawback is the reluctance of an obvious reference signal to adapt on.

In a TDMA system e.g. GSM, the convergation rate and the complexity of the algorithm are important parameters. Since several mobiles are transmitting on the same frequency but in different time-slots the adaptive antenna must be able to change its radiation pattern from time-slot to time-slot. The reference signal could be the a synchronisation or training sequence.

A CDMA system e.g. Qualcomm, uses orthogonal signals in the same frequency band [9]. The pseudo-noise signals can be used as continuous reference signals.

## 6. SIMULATION RESULTS

The algorithms considered in these simulations are the Least-Mean-Squares (LMS), the Recursive-Least-Squares (RLS), the Sample-Matrix-Inversion (SMI), the MUltiple-SIgnal-Classification (MUSIC) and the Weighted-Subspace-Fitting (WSF) algorithms. The convergation time for the LMS algorithm can be reduced if the antenna signals are pre-processed in a Butlermatrix. The Butlermatrix is a passive reciprocal and relatively broadband beamforming network generating a number of orthogonal beams [1]. The LMS simulation are carried out both with and without a Butlermatrix.

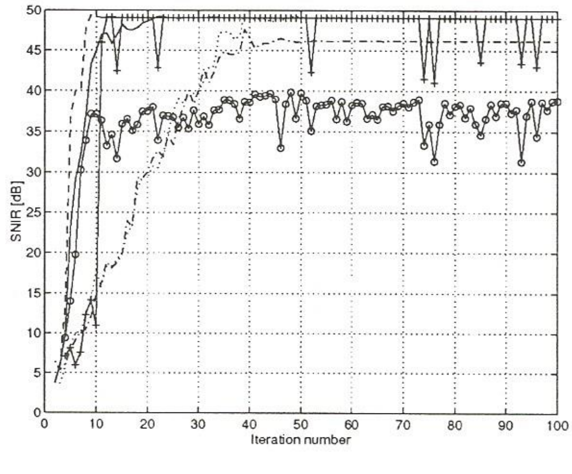


Fig.2 SNIR versus iteration number. Velocity of mobile = 0, reflection angle $\alpha$ = 1 degree. Curve types: sold = RLS, dashed = SMI, dotted = LMS, dashdot = LMS with Butler, solid with plus = WSF, solid with circle = MUSIC.

The parameter of interest in the simulations is the Signal-to-Noise-plus-Interference Ratio (SNIR). Fig.2 shows the values of SNIR versus the iteration number. RLS, SMI and WSF convergates quickly to a high value. LMS convergates slower than the others and reaches the saturated value, about 3dB lower than the former, after 40 iterations. The MUSIC algorithm convergates quickly but to a low value.

The MUSIC algorithm calculates estimates the DOA's from the eigenvalues of the sample covariance matrix. Therefore it can only resolve signals that are incoherent. If multiple coherent signals are impinging on the array with different DOA's the algorithm will not succeed.

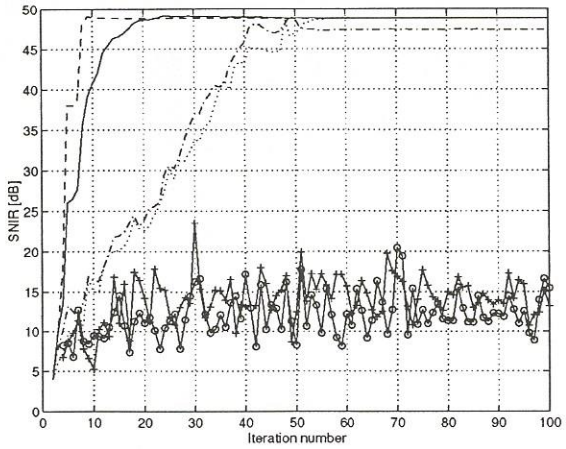


Fig.3 SNIR versus iteration number. Velocity of mobile = 0, reflection angle $\alpha$ = 20 degrees. See fig.2 for description of curves.

An N-element array has only N-1 degrees of freedom in its radiation pattern. If the number of reflections is larger than the degrees of freedom, then the high-resolution algorithms can not resolve the DOA's. That's why MUSIC and WSF gets low values of SNIR when the circle radius increases, see fig.3. The reference signal methods are much more robust against multiple coherent signals.

If the velocity of the mobile is increased, the SNIR for the reference methods will go down, see fig.4. The received signals resemblance with the reference signal will diminish due to heavy fading.

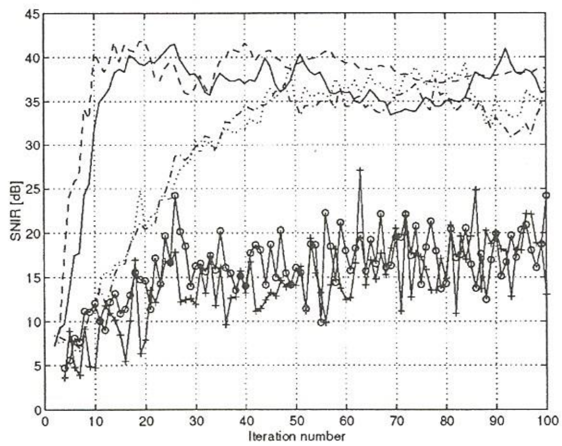


Fig.4 SNIR versus iteration number. Velocity of mobile = 100km/h, reflection angle $\alpha$ = 20 degrees. See fig.2 for description of curves.

The figures show the mean-value of 10 simulations. In order to get smoother behaviour one should take the mean over a larger number of simulations.

## 7. DISCUSSION

When the circle containing reflections increases its radius, there will be a limit when intersymbol interference will start to become a problem. To take this into consideration is the next step in the simulation study. Intersymbol interference is not a problem that can be separated from the adaptive antenna and be solved by the equalizer alone. In order to get a optimized result the array weights and the equalizer parameters have to be calculated together.

Both the reference signals and the high-resolution methods have their disadvantages. All systems don't have an obvious reference signal and to assume the array manifold to be known (even with calibration) is a optimistic condition. A way to get by them can be the use of higher order statistics [11]. With higher order cumulants it is possible to get blind estimation of the source steering vector.

## 8. ACKNOWLEDGEMENT

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