I. Title of the Study

Application of MUSIC, ESPRIT and SAGE Algorithms for Narrowband Signal Detection and Localization

II. Abstract

This research involves the application of a super-resolution direction of arrival estimation using Multiple Signal Classification (MUSIC), Estimation of Signal Parameters via Rotational Invariance Principles (ESPRIT) and Space Alternating Generalized Expectation Maximization (SAGE) algorithms using data gathered from empirical measurements. The study uses an antenna testdbed composed of an antenna receiver, radio frequency (RF) front-end receiver circuitry, data acquisition block and personal computer interface. The algorithms will then process the data and display the results offline.

The antenna testbed uses a 1.9 GHz receiver front end connected to a 2-element antenna array. The received signal will be downconverted to an intermediate frequency (IF). This will be then connected to one of the input ports of the Altera fpga board. The Altera fpga evaluation board digitizes the signal at IF and serves as the interface to the personal computer. Hence, the data consists of actual off-the-air received signals at 1.9 GHz. A vector signal generator is used as the source producing a modulated output at 1.9 Ghz. While an empty room will then serve as the wireless channel.

Application of the algorithm requires data from an antenna array. In this study, a virtual antenna array can be created by calibrated displacement of the 2-element antenna array. For this research, a 4-element and an 8-element linear array will be emulated. Due to computational complexity and hardware limitations, the algorithms are implemented offline. After processing, the results will then be plotted and displayed on a pc.

Calibration of the testbed will be done before actual data gathering. Performances of the algorithms are evaluated based on the ability of the system to accurately estimate the direction of arrival for different set-ups. Variation of the set-up includes the number and spacing of elements in the antenna array, the array structure and the signal power of the source. The effect of obstructions in the empty room will also be investigated to get reflections, scattering and multipath profiles.

III. Statement of the Problem

The ever growing demand for mobile communications is increasing the need for better coverage, improved capacity and higher quality of service. The concept of space-time processing promises to improve mobile radio performance significantly. The principles of space-time processing can be used to develop “smart antennas” that use antenna arrays and adaptive processing. Smart antenna systems communicate directionally by forming specific antenna beam patterns. This can be done adaptively and is aptly called “adaptive beamforming.” Adaptive beamforming needs information on the desired signal direction-of-arrival for it to direct its main lobe accordingly while forming nulls on interference signals. There has been a lot of study done to derive algorithms to estimate signal direction-of-arrivals. However, most of the studies use simulation. Hence, there is a need to investigate the performance of DOA estimation algorithms using empirical data to determine the viability of applying these algorithms for practical communications system.

IV. Objectives

This study aims to realize the viability of three direction-of-arrival estimation algorithms for practical communications system using available hardware and software.

The specific objectives are:

- To be able to successfully implement and evaluate MUSIC, ESPRIT and SAGE algorithms using empirical measurements;
- To be able to implement a physical testbed for experimental verification of published theoretical results using MUSIC, ESPRIT and SAGE algorithms;
- To be able to investigate the effects of varying experimental set-up parameters including the number of elements in the antenna array, the inter-element spacing, the structure of the antenna array eg. linear vs. circular, signal power of the source, obstructions in the wireless channel;

V. Background

The use of an array of sensors allows many advantages over the use of single receiver-typically weaker signals can be detected and their localization or directions of arrival estimated.

The first approach to achieving this with an array of sensors was conventional beamforming. It is based upon the addition of receiver outputs, weighted with a fixed set of weights, such that the outputs of the receiver due to a plane wave from a chosen direction were in phase. This could be done by time delays for broadband signals or by phase shifting for narrowband signals. For this class of beamformers the array gain achieved and the ability to resolve closely spaced signals are determined by the array aperture.

The next close of beamformers, such as the so called optimum and maximum entropy beamformers, derived an optimum set of weights based on various statistical criteria and showed how the aperture of the array no longer limits on determining the resolution and interference rejection performance of an array. For this class of beamformers the signal-to-noise ratio (SNR) was the main limiting parameter. However, other issues such as integration times and array calibration could significantly affect the performance of the array.

Some typical example of arrays are shown below.

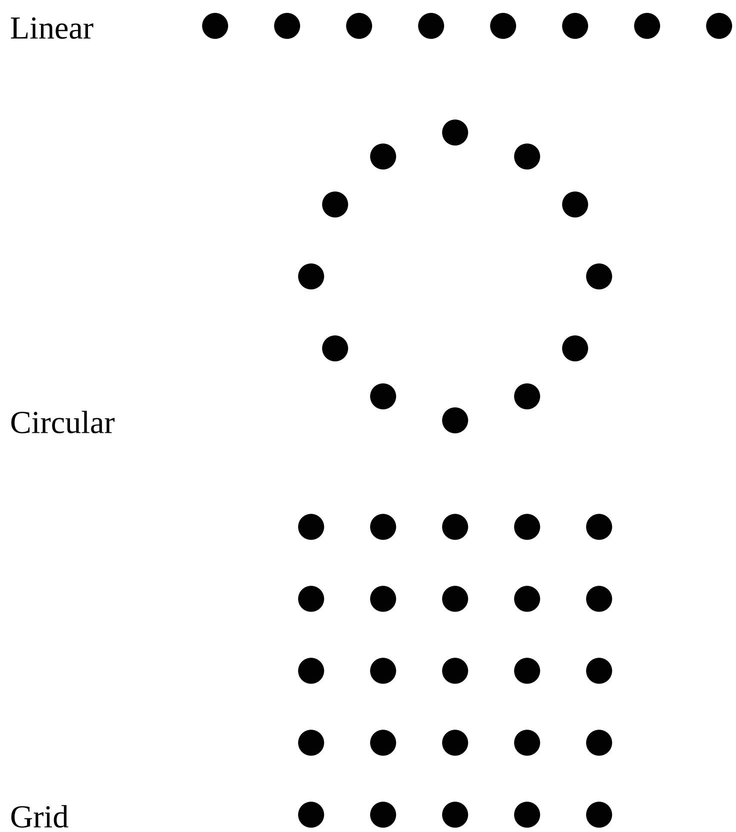

Figure 1. Typical antenna array structures

An important use of arrays is the estimation of the direction-of-arrivals (DoA) of plane wave signals incident upon an antenna array and the formation of beams steered in selected directions. These techniques have provided a formal analytic framework that has encouraged further theoretical development and a computational framework for practical applications.

V. Review of Related Literature

A. Introduction

Nowadays, the ever growing demand for mobile communications is constantly increasing the need for better coverage, improved capacity and higher quality of service. Space-time processing promises to improve mobile radio performance significantly. The principles of space-time processing can be used to develop "smart antennas" that use adaptive arrays of antenna sensors. Therefore, the smart antenna concept has become very interesting to the mobile communications industry.

Multi-user wireless communication systems, such as cellular, are prone to interference and poor voice quality due to the omni-directional transmission of radio frequency (RF) signals. In traditional antenna systems, the wide area dispersement is necessary because the user's location is unknown. The dispersed transmission pollutes the electromagnetic environment by radiating most of the transmitted power in unnecessary directions.

In contrast, smart antenna systems determine a user's location and attempt to focus and receive energy only in desired directions. At the same time it is possible to place spatial nulls in the direction of unwanted interferences. This capability can be used to improve the performance of a mobile communication system. This may be done adaptively and is called "adaptive beamforming."

Adaptive beamforming needs information on the desired signal direction-of-arrival for it to direct its main lobe accordingly while forming nulls on interference signals. There has been a lot of study done to derive algorithms to estimate signal direction-of-arrivals. These methods can be classified into two groups: Maximum Likelihood (ML) Method and Subspace-Based Method. Multiple Signal Classification (MUSIC) and Estimation of Signal Parameters via

Rotational Invariance Principles (ESPRIT) algorithms are subspace-based methods while Space Alternating Generalized Expectation Maximization (SAGE) algorithm was derived from the maximum likelihood method.

B. Direction of Arrival Estimation and its Applications

Smart antenna systems are rapidly emerging as one of the key technologies that can enhance overall wireless communications system performance. By making use of the spatial dimension, and dynamically generating adaptive receive and transmit antenna patterns, a smart antenna can greatly reduce interference, increase the system capacity, increase power efficiency as well as reduce overall infrastructure costs [7]. It is also possible to multiplex channels in the spatial dimension just as in the frequency and time dimensions. This is often referred to as Spatial Division Multiple Access (SDMA) [11]. A wide range of wireless communication systems may benefit from smart antenna systems including high mobility cellular systems, low mobility short range systems, wireless local loop applications, satellite communications and wireless LAN. It is predicted how smart antennas will have a transition from a largely unknown technology to a valuable option for all wireless communication applications by 2005 [9].

The figure below shows the general smart antenna architecture.

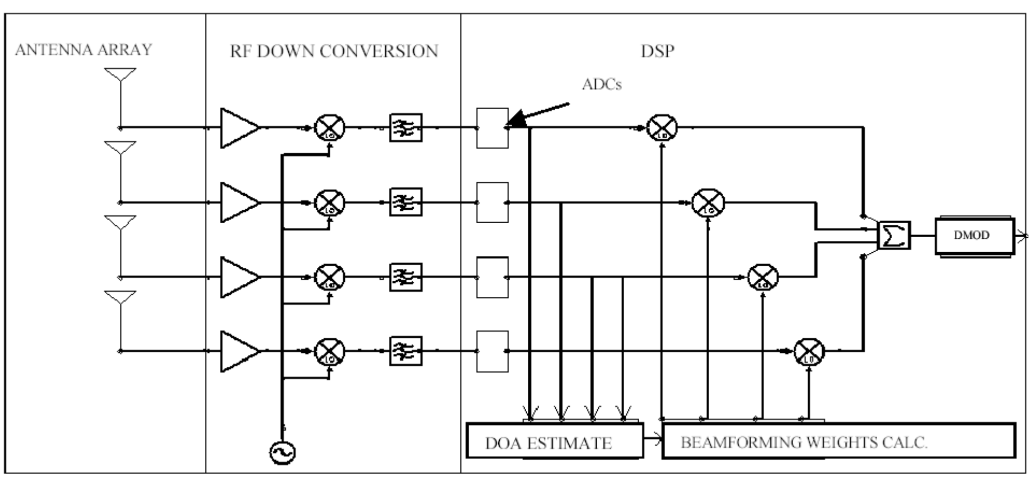

Figure 2. General smart antenna architecture

It has been proven both theoretically and experimentally that smart antennas can provide the benefits stated above, but possibly the most challenging problem related to adaptive antennas is their practical implementation [10]. Digital signal processing (DSP) algorithms related smart antennas come at a high computational expense making their real time implementation difficult. Adaptive antennas use multiple antenna elements (antenna array), increasing the RF hardware complexity of the receiver, and making a fully functional multichannel transceiver more expensive. But with the emergence of high speed analog to digital converters (ADC), digital signal processors and the dramatic reduction in cost of microwave components the smart antenna has become a feasible system to design and implement.

There are two approaches for smart antenna strategies: switched beam and adaptive antenna arrays. Both systems attempt to increase gain according to the location of the user; however, only the adaptive system provides optimal gain while simultaneously identifying, tracking, and minimizing interfering signals. It is the adaptive system's active interference strategy and additional gain that provide substantial performance advantages and flexibility over the more passive switched beam approach [8]].

Smart antenna systems communicate directionally by forming specific antenna beam patterns. When a smart antenna directs its main lobe with enhanced gain in the direction of the user, it naturally forms side lobes and nulls or areas of medium and minimal gain respectively in directions away from the main lobe. Different switched beam and adaptive smart antenna systems control the lobes and the nulls with varying degrees of accuracy and flexibility.

The figure below illustrates the beam patterns that each system might choose in the face of a signal of interest and two co-channel interferers in the positions shown.

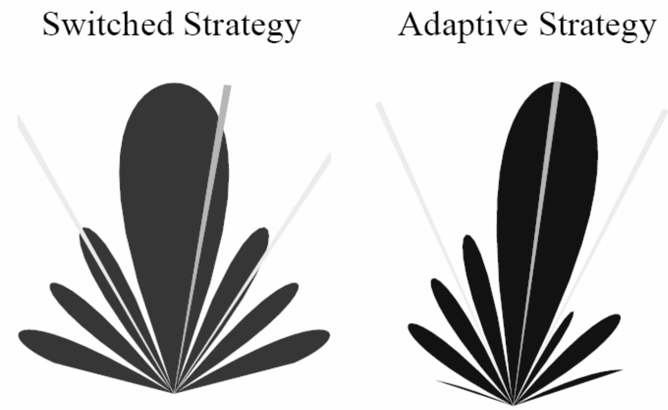

Figure 3. Switched Strategy vs. Adaptive Strategy

Both systems have directed the lobe with the most gain in the general direction of the signal of interest, although the adaptive system has chosen more accurate placement, providing greater signal enhancement. Similarly, the interfering signals arrive at places of lower gain outside the main lobe, but again the adaptive system has placed these signals at the lowest possible gain points and better insures that the main signal received maximum enhancement while the interfering signals receive maximum suppression.

C. Direction of Arrival Estimation Algorithms

There are different methods of uplink and downlink processing available for use in 'smart' antenna systems. For the uplink processing, the methods are Spatial Structure, Training Signal and Temporal Structure. While for the downlink processing, the methods are Frequency Division Duplex (FDD) and Time Division Duplex (TDD) [1].

The spatial structure methods describe the techniques to be used in this research. Spatial structure methods exploit the information in the steering vector used to estimate the direction of arrivals (DOAs) of the signals impinging on the sensor array. The estimated direction of arrivals are then used to determine the weights in the patternforming network. This is called beamforming. Spatial structure methods only exploit spatial structure and training signals and the temporal structure of the signals is ignored.

A key issue in array beamforming is the estimation of the DoA's of plane wave signals incident upon an array. Many popular eigen subspace approaches to this problem are based on eigen analysis of the cross spectral matrix of the narrowband receiver outputs and the utilization of the properties of suitably defined projection matrices to calculate DoA estimates. This is the basis of MUSIC algorithm for narrowband signals in [24] and ESPRIT [20].

All the subspace based methods are based on the eigenvector decomposition of the covariance matrix

$$\mathbf{R}_{xx} = E\left(\mathbf{x}(t)\mathbf{x}^{H}(t)\right)$$

Equation 1. Covariance matrix

If uncorrelated noise is present, then the covariance matrix can be expressed as

$$\mathbf{R}_{xx} = \mathbf{A}E\left(\mathbf{s}(t)\mathbf{s}^{H}(t)\right)\mathbf{A} + E\left(\mathbf{n}(t)\mathbf{n}^{H}(t)\right)$$

Equation 2. Covariance matrix with uncorrelated noise

where A is the steering matrix, s(t) is the received signal and n(t) is the noise with power equal to its variance.

The singular value decomposition is then applied to the covariance matrix. If a signal vector is orthogonal to A, then it is an eigenvector of R_{xx} with eigenvalue equal to the variance of noise. Hence the eigenvectors R_{xx} with eigenvalue σ^2 belong to the nullspace of A. From here, the smallest eigenvalues are

$$\lambda_{L+1} = \lambda_{L+2} = \ldots = \lambda_N = \sigma^2$$

Therefore it is possible to partition the eigenvectors into noise eigenvectors and signal eigenvectors and thus

$$\begin{aligned} \mathcal{R}[\mathbf{U}_s] &= \mathcal{R}[\mathbf{A}] \\ \mathcal{R}[\mathbf{U}_n] &= {}^{\perp}\mathcal{R}\left[\mathbf{A}^H\right] \end{aligned}$$

$\mathcal{R}[\mathbf{U}_s]$ is called the signal subspace and $\mathcal{R}[\mathbf{U}_n]$ is called the noise subspace.

The projection operators onto these signal and noise subspaces are defined as

$$\begin{aligned} \mathbf{P_A} &= \mathbf{AA}^+ = \mathbf{U}_s\left(\mathbf{U}_s^H\mathbf{U}_s\right)^{-1}\mathbf{U}_s^H = \mathbf{U}_s\mathbf{U}_s^H \\ \mathbf{P_A^\perp} &= \mathbf{I} - \mathbf{AA}^+ = \mathbf{U}_n\left(\mathbf{U}_n^H\mathbf{U}_n\right)^{-1}\mathbf{U}_n^H = \mathbf{U}_n\mathbf{U}_n^H \end{aligned}$$

where A+ is the pseudo-inverse of A.

Equation 3. Projection operators onto signal and noise subspaces

The simplest of the algorithms that are based on the above stated subspace decomposition is the MUSIC (Multiple SIgnal Classification) algorithm. For example, assume L signals are impinging on the sensor array. A(θ) is projected into the noise subspace. The projection gives the vector

$$\mathbf{z} = \mathbf{P_A^\perp}\mathbf{a}(\theta)$$

The magnitude squared of z can be written as

$$f(\theta) = \mathbf{z}^H\mathbf{z} = \mathbf{a}^H(\theta)\mathbf{P_A^\perp}^H\mathbf{P_A^\perp}\mathbf{a}(\theta) = \mathbf{a}^H(\theta)\mathbf{U}_n\mathbf{U}_n^H\mathbf{a}(\theta)$$

Therefore, the array manifold is searched; f(θ) is evaluated for all θ as DOA estimates the points at which f(θ) = 0. A major breakdown of the MUSIC algorithm is when the signals are coherent or correlated signals. Such cases make the autocorrelation matrix not full rank.

ESPRIT on the other hand estimate signal arrival directions by exploiting the rotational invariance of the signal subspaces of subsets of the array receivers. It avoids the searching of the manifold and does not require complete knowledge of the array manifold as is the case for the MUSIC algorithm.

The SAGE algorithm meanwhile is a technique derived from the Maximum Likelihood (ML) principle which allows for high resolution determination of the incident angle, the propagation delay and the complex amplitude. The Space Alternating Generalized Expectation maximization (SAGE) Algorithm [23] updates the parameters sequentially by replacing the high dimensional optimization process necessary to compute the joint maximum likelihood estimate of the parameters, by several separate, low dimensional maximization procedures, which are performed sequentially.

Given M samples {x(1), x(2)…x(M)}, the likelihood function for the vector channel model assumed in is given by

$$p\left(\mathbf{x}(k);\boldsymbol{\theta}\,,\mathbf{s}(k),\sigma^2\right)=\prod_{k=1}^{M}\left(\pi\sigma^2\right)^{-N}\exp\left(-\frac{1}{\sigma^2}\left\|\mathbf{x}\left(k\right)-\mathbf{A}\mathbf{s}\left(k\right)\right\|^2\right)$$

Equation 4. Maximum likelihood function

where θ is the directional information, s(k) is the transmitted signal and σ^2 is the variance of the noise process. The ML estimates of these unknowns are calculated as the maximising arguments of $p(x(k); \theta, s(k), \sigma^2)$, the rationale being that these values make the probability of the observations as large as possible.

The estimate for the signal waveform is

$$\hat{\mathbf{s}}(k)=\mathbf{A}^{+}\mathbf{x}(k)$$

Equation 5. Signal waveform estimate where A+ is the pseudo-inverse of A.

Maximum likelihood estimation is a parametric method and hence its resolution is not limited as is the case for the conventional beamformer. However, a multidimensional search is required to find the estimates, resulting in a high computational complexity. The ML estimator can be classified as a deterministic ML estimator, because the impinging multipath rays of both, the desired signal and the interferers, are modeled deterministically. It is also possible to model the interfering sources as colored Gaussian noise. Furthermore, for Gaussian signal sources, the stochastic ML estimator attains the Cramer-Rao lower bound (CRB), since all unknowns in the stochastic model are estimated consistently. For the deterministic model, the number of signal waveform parameters grows as the number of samples increases, implying that they cannot be estimated consistently.

For coherent multipath signals, the subspace based methods do not work properly, because the signal subspace and the subspace spanned by the steering matrix are not equivalent in that case. There exists a technique called spatial smoothing [6] that is able to mitigate this disadvantage. Spatial smoothing means that the array is split into identical subarrays, the covariances of which are averaged. The rank of the averaged covariance matrix can be shown to increase by 1 with probability 1 for each additional subarray [15]. The drawback of spatial smoothing is that the effective aperture of the array is reduced, since the subarrays are smaller than the original array. The other possibility is to use single snapshot methods or the computationally more complex ML estimation method, both of which do not have any problems with coherent signals.

The number of DOAs that can be estimated is smaller than the number of antenna elements. This is a major disadvantage in environments suffering from large angle spread. If large angle spread is present, then the point source model is not valid and inevitably many different DOAs correspond to a single signal source. In that case spatial structure methods require more antenna elements than the total number of impinging signals and their multipaths. But since the number of sources for this research is limited and that a virtual array will be used, then the number of DOA that can be estimated will not be of concern.

In practical antenna arrays, the antenna elements are not identical and they are mutually coupled. Spatial structure methods explicitly exploit the knowledge of the steering vector $a(\theta)$ Therefore, mutual coupling and difference of antenna elements have to be included into the steering vector, if spatial structure methods are to work properly. Because usually this data is not known beforehand, it has to be estimated very accurately. This is called array calibration. This research will use a two antenna receivers to create a virtual array by making calibrated displacements of the antenna.

The difficulty in implementing subspace methods in DOA estimation lies in the computational complexity in solving eigenvalue problems. A proposed method can solve eigenvalues and eigenvectors in a very short time, only 1.74% of the time by the conventional method, by regarding the eigenproblem as solving a fourth-order algebraic polynomial [5]. It is also confirmed that the proposed algebraic approach does not make the accuracy worse when it is implemented by finite word-length processors like digital signal processor (DSP) or field programmable gate array (FPGA). This paper is very helpful for real-time implementation of DOA estimation. Even if the algorithms will be implemented offline, the mathematical approach can be utilized to implement the algorithms using the fpga.

D. MUSIC Algorithm

Multiple signal classification (MUSIC), proposed by Schmidt, is an improvement of Piserenko's minimum eigenvalue algorithm. It provides asymptotically unbiased estimates of (1) the number of signals, (2) directions of arrival (or emitter locations), (3) strengths and cross correlations among the incident waveforms, and (4) the strength of noise/interference [24].

This method has been applied for wideband acoustic detection and tracking [12]. Given adequate SNR, the incoherent wideband MUSIC algorithm performed very well, produced accurate DOA estimates, and yielded sharp distinct peaks in the beampattern. Other studies simulated MUSIC algorithm for estimating moving targets [17] and DS-CDMA. In all these applications, MUSIC was able to give an accurate DOA estimates by yielding a MUSIC spectrum which exhibits sharp peaks in the vicinity of the true path delays.

For a realization as a dedicated hardware component or on a fixed-point DSP, the sensitivity of the MUSIC algorithm in general and especially the eigenvector decomposition (EVD) with respect to a fixed-point representation of the data needs to be quantified.

E. ESPRIT Algorithm

Estimation of Signal Parameters via Rotational Invariance Principles (ESPRIT) approximates signal arrival directions by exploiting the rotational invariance of the signal subspaces of subsets of the array receivers [4]. It is popular as it avoids the searching for the manifold and does not require complete knowledge of the array manifold as is the case for the MUSIC algorithm.

Some studies applied ESPRIT for parametric localization of distributed sources for both coherently and incoherently distributed source models. In the coherently distributed case, the central angle of the sources are estimated by using two closely spaced subarrays. It was showed that the rotational invariance exists approximately and ESPRIT was used to estimate the central angles [20]. In the incoherently distributed case, each source is modeled as a 2-D subspace in the observation space. The source central angles are estimated by ESPRIT using a dimension of 2 for each source. It was shown that the array covariance matrix can be approximated by the second moment of the source angular power density and the source powers. Hence, the source parameters can be estimated by a least-squares covariance matrix fitting.

F. SAGE Algorithm

The expectation-maximization (EM) method can facilitate maximizing likelihood functions that arise in statistical estimation problems. In the classical EM paradigm, one iteratively maximizes the conditional log-likelihood of a single unobservable complete data space, rather than maximizing the intractable likelihood function for the measured or incomplete data. EM algorithms update all parameters simultaneously, which has two drawbacks: 1)slow convergence, and 2) difficult maximization steps due to coupling when smoothness penalties are used [23].

The space-alternating generalized EM (SAGE) method updates the parameters sequentially by alternating between several small hidden data spaces defined by the algorithm designer. [23] proves that the sequence of estimates monotonically increases the penalized-likelihood objective, derives asymptotic convergence rates, and provides sufficient conditions for monotone convergence in norm. Two signal processing applications illustrate this method: estimation of superimposed signals in Gaussian noise, and image reconstruction from Poisson measurements. In both applications, the SAGE algorithms easily accommodate smoothness penalties, and converge faster than the EM algorithms. Simulations have shown the performance of SAGE for providing accurate DOA estimates [3] [22]. The influence of coupling in antenna arrays in the SAGE algorithm was investigated [16]. It was shown that the SAGE algorithm performed well for reasonable coupling values between antenna elements.

G. DOA Estimation Implementation

The existence of mutual coupling in spatial reference adaptive arrays degrades severely the performance of the array. This effect is particularly harmful in direction finding algorithms. The minimization of mutual coupling effects in DOA algorithms can be done in two ways: via array processing or via electromagnetic (EM) analysis. The first one deals with signal processing techniques and is based on modifications of the corresponding eigenstructure (ES) algorithms. The second deals with the electromagnetic model of the array and seeks to correct the actual antenna voltages for the mutual coupling and then process these corrected voltages to determine the angle of arrival [15].

Direction of arrival (DOA) estimation of narrowband wavefronts impinging on an array of sensors has long been of great research interest. In [2], it is shown that for a uniform circular array (UCA), the accuracy of DOA estimation deteriorates more than exponentially as the radius of the UCA is reduced below a certain limit. It is further shown that UCAs with radii (in wavelengths) coinciding with the zeros of the Bessel functions will result in DOA estimates that are somewhat degenerate. The well-known multiple signal classification (MUSIC) algorithm is used as an example. The results of this paper will serve as a guide for the circular array set-up to be used.

Studies show that all the common algorithms namely the beamformer methods, MUSIC, Min-Norm, JoDeG, ESPRIT, and SAGE, generate an ambiguous error in the estimated direction-of-arrival results, when the antenna element spacing is more than half the carrier wavelength [14]. Here, the source of this ambiguity is identified by applying the algorithms on the real measured data in a controlled environment. Inter-element spacing in an antenna array is usually at half wavelength or less. This research will also investigate the effects of inter-element spacing in the array to determine any ambiguities resulting from this set-up.

IV. Methodology

Experimental verification of MUSIC, ESPRIT and SAGE involves the following stages:

1) Design and Implementation of the antenna elements for the virtual antenna array;

2) Implementation of the analog RF front-end receiver circuitry for the single antenna;

3) Configuration of the data acquisition blocks and interface to the personal computer (PC);

4) Testing of individual sub-systems;

5) Characterization of the directional antenna for transmission;

6) Data acquisition;

7) Offline processing of empirical data using MUSIC, ESPRIT and SAGE.

8) Calibration of the complete system;

These stages are described in detail in the succeeding paragraphs.

Antenna Block

The receiver block will be implemented using an omnidirectional monopole antenna. The monopole antenna was chosen as the radiating element of the system due to its relatively small size and ease of fabrication as compared to other types of antennas. It would be designed to be compatible with the carrier frequency of the modulated signal which is 1.9 GHz. This antenna will be used to create a virtual linear antenna array of 4 and 8 elements. Hence, the position of the receiving antenna during data acquisition is such that it will emulate a linear antenna array.

The calibrated displacement of the antenna for the next data acquisition will be half wavelength, measured at the center frequency of the operating bandwidth.

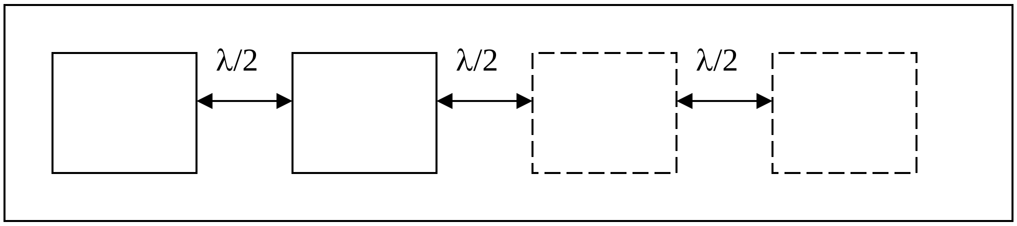

Figure 4. Virtual antenna array

Analog Radio Frequency Receiver Front End

Since the energy of the received signal is typically small, it will have to be amplified without contributing too much noise. RF Micro Devices' RF2489 evaluation board will be used to serve as a Low Noise Amplifier (LNA) and Mixer. The LNA will be connected to the output of the single antenna receiver through coaxial cable.

After amplification, the signal will then be down converted to an Intermediate Frequency (IF). An RF signal generator will serve as the local oscillator (LO) of the frequency mixer. The output of the mixer will be signals whose frequencies are the sum and difference of the local oscillator frequency and the RF frequency. Given that the desired frequency is 1900 MHz, the desired down converted frequency is at 110 Mhz and the image frequency is at 2120 Mhz. The output of the mixer will be passed through a low pass filter to eliminate the image frequency. The cut-off frequency of the low-pass filter should be high enough to pass the desired frequency but low enough to attenuate the image frequency.

Data Acquisition Block and PC Interface

The received signal, after being amplified, down converted and filtered, will then be digitized using an analog-to-digital converter (ADC). The data acquisition bock to be used is the Altera DSP Development Kit, Stratix Edition. This fpga board has two 12-bit, 125 Mhz ADC's. Sub-sampling will be employed since the sampling rate is below the Nyquist rate, 220 MHz. The digitized data will then be transmitted to a personal computer for data storage.

Integrated System

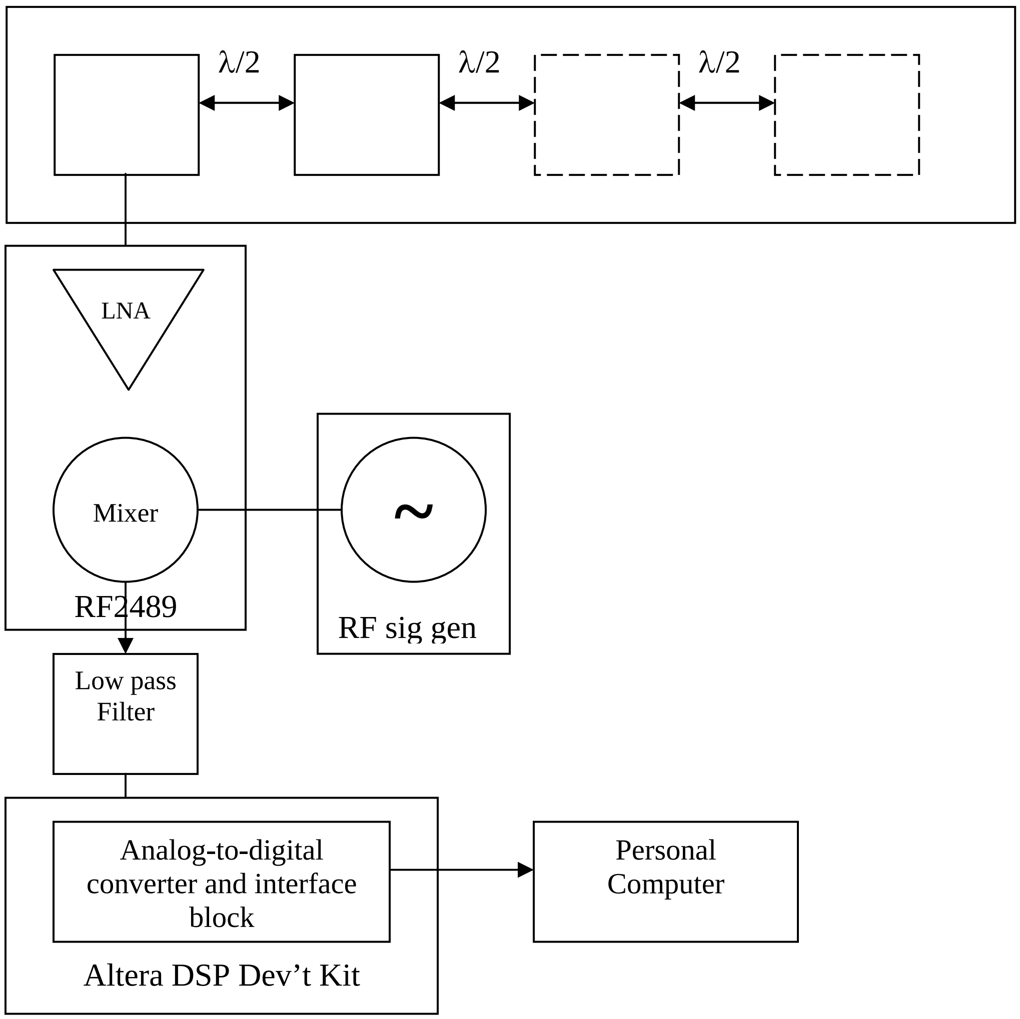

Figure 5. Block Diagram of Completed Testbed

Testing

A brief summary of the testing procedures for the system is given below.

1. Antenna sub-block

    A. The input impedance and phase response of each antenna element will be measured using a network analyzer;

B. The spatial response of the antenna element will be measured using open area field-testing through an antenna radiation pattern measurement system (ARPMS).

2. RF analog front-end receiver sub-block

A. The input impedance, gain and phase response of the low noise amplifier (LNA) and downconverters after each antenna element will be measured using a network analyzer to verify correct amplitude and phase balance of each receiver path;

B. The outputs of the frequency downconverters will be measured to ensure dynamic range compatibility with the analog-to-digital converters (ADC);

C. Since the study focuses on the source localization, the antenna for the transmitter will be characterized. The antenna radiation pattern measurement system (ARPMS) will be used to measure the spatial response.

D. The spatial response of the entire antenna array system together with the analog RF receiver front-end will be measured using the APRMS. This involves using an antenna to transmit a wave towards the phased array antenna system, which will be at a certain distance from the transmitter. The receiver system will be rotated 360° while the received power is measured using a spectrum analyzer and tabulated. From the tabulated results, the radiation pattern of the antenna array in receive mode can be plotted.

3. IF digital sub-block

A. The outputs of the analog-to-digital converters (ADC) will be measured using logic analyzers;

B. The PC interface to the fpga board will be tested for successful communication and exchange of data;

Data acquisition

Empirical measurements will be done to emulate a linear 4-, and 8-element antenna array. A modulated signal will be transmitted with a carrier frequency of 1.9 GHz. The receiving testbed is located at a certain distance from the transmitter. The first set-up is such that the source has a 0° azimuth with respect to the virtual array. Hence after data acquisition, the receiving antenna will be displaced at half wavelength from the previous position and the measurement is repeated.

Recorded data will then be processed offline. Since the direction of the source is known, the result may be used to calibrate the system if the results will not agree with the localization of the source.

The variables in the set-up are the number of elements in the virtual array, the inter-element spacing in the virtual array, the transmitter signal power and the direction of the source.

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