Direction of Radio Finding via MUSIC (Multiple Signal Classification) Algorithm for Hardware Design System

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Abstract. Concept of radio direction finding systems, which use radio direction finding is based on digital signal processing algorithms. Thus, the radio direction finding system becomes capable to locate and track signals by the both. Performance of radio direction finding significantly depends on effectiveness of digital signal processing algorithms. The algorithm uses the Direction of Arrival (DOA) algorithms to estimate the number of incidents plane waves on the antenna array and their angle of incidence. This manuscript investigates implementation of the DOA algorithms (MUSIC) on the uniform linear array in the presence of white noise. The experiment results exhibit that MUSIC algorithm changed well with the radio direction.

1. Introduction

A radio direction finding (DF) system [1-3] is a receiver arranged in a combination to determine the azimuth angle of a distant emitter. Basically, all DF systems derive the emitter location from an initial determination of the angle-of-arrival (AOA) [4-6]. Radio direction finding techniques have classically been based on multiple-antenna systems employing multiple receivers [7-9]. And MUSIC use simultaneous phase information from each antenna to estimate the angle-of-arrival of the signal of interest. Multiple receivers are impractical. Thus, single channel techniques are of interest, particularly in mobile scenarios. Although the amount of existing research for single channel DF is considerably less than for multi-channel direction finding, single channel direction finding techniques have been previously investigated. DOA (Direction of arrival) estimation is a key research area in array signal processing and many engineering applications, such as wireless communications, radar, radio astronomy, sonar, navigation, tracking of various objects, earthquake, medicine and other emergency assistance devices that need to be supported by direction of arrival estimation. In modern society, DOA (Direction of arrival) estimation is normally researched as a part in the field of array processing, so many works highlight radio direction finding. Over the past ten years, Wireless Local Area Networks (WLANs) have increased quickly because of its flexibility and convenience. In order to satisfy the requirements of advanced services, a high-speed data rate is necessary. Owing to the excessive use of the low end of the spectrum, people begin to search for the higher frequency bands for more applications. With higher user density, higher frequency and higher data rate, multipath fading and cross interference become the main issues. In order to solve these problems and get higher communication capacity, smart antenna systems are proved to be very effective in suppression of the interference and multipath signals

Signal processing in smart antenna systems concentrates on the development of efficient algorithms for Direction of Arrival (DOA) estimation and adaptive beam forming. However, there are many limitations if DOA (Direction of arrival) estimation uses a fixed antenna. Antenna main-lobe beam width is inversely proportional to its physical shape. It is not a practical option to improve the accuracy

of angle measurement in accordance with an increase in the physical aperture of the receiving antenna. Some systems such as missile seeker or aircraft antenna have limited physical size, so they are sufficiently wide in beam width of the main lobe to correspond. They do not have a good resolution and if there are multiple signals falling in the antenna's main lobe, it becomes too difficult to distinguish between them.

There are many kinds of super resolution algorithms such as spectral estimation, Bartlett, Capon, ESPRIT [10], and Min-norm [11]. One of the most popular and widely used subspace-based techniques to estimate the DOA (Direction of arrival) of multiple signal sources is the ESPRIT algorithm. Large numbers of computations are needed to search for the spectral angle. When using the ESPRIT algorithm, so in real applications its implementation can be difficult. Compared with spectral ESPRIT algorithm, the MUSIC method has better performance with reduced complexity computation. It can only be used in uniform linear array (ULA) or non-uniform linear array whose arrays are restricted to a uniform grid [12] [13]. MUSIC [14] was the most promising and a leading candidate for further study and actual hardware implementation. However, although the performance advantages of MUSIC are substantial, they are achieved at a cost in computation (searching over parameter space) and storage (of array calibration data) [15]. In this thesis, I will focus on the MUSIC algorithm.

2. MUSIC (Multiple Signal Classification) Algorithm

Multiple Signal Classification (MUSIC) algorithm was proposed by Schmidt and his colleagues in 1979 [16]. It has created a new era for spatial spectrum estimation algorithms. The promotion of the structure algorithm characterized rise and development, and it has become a crucial algorithm for theoretical system of spatial spectrum. The basic idea of MUSIC algorithm is to conduct characteristic decomposition for the covariance matrix of any array output data, resulting in a signal subspace orthogonal with a noise subspace corresponding to the signal components. Then these two orthogonal subspaces are used to constitute a spectrum function, be got though by spectral peak search and detect DOA (Direction of arrival) signals.

MUSIC estimates the frequency substance of a signal or autocorrelation matrix using an Eigen space method. This method assumes that a signal, $x(n)$, consists of p complex exponentials in the presence of Gaussian white noise. Given an $M \times M$ autocorrelation matrix, R_x, if the eigenvalues are sorted in decreasing order, the eigenvectors corresponding to the largest eigenvalues (i.e. directions of largest variability) span the signal subspace. The remaining $M - p$ eigenvectors span the orthogonal space, where there is only noise. Note that for $M = p + 1$, MUSIC is identical to Pisarenko harmonic decomposition. The general idea is to use averaging to improve the performance of the Pisarenko estimator. The frequency estimation function for MUSIC is $\widehat{P_{MN}}(e^{j\omega}) = \frac{1}{\sum_{i=p+1}^{M}|e^H v_i|^2}$, where v_i are the noise eigenvectors and $e = [1 \ e^{j\omega} e^{2j\omega} \cdots \ e^{(M-1)\omega} \ ]^T$.

3. Numerical and Experimental Results

We estimate implementation of MUSIC algorithms and display that MUSIC algorithm is as precise having appreciably simply structure. Figure 1shows that root locus diagrams produced, for a case when one QPSK emitter was impinging on the 30-element uniform linear arrays (ULA) with λ/2 inter element spacing. Root locus diagram produced by MUSIC algorithm is obviouslyovercrowdedhaving22 pairs of roots in spilt of the fact that only one emitter was impacting on the ULA. Figure 2show mean square error (MSE) in DOA (Direction of arrival) estimation as a function of the SNR value. The MUSIC was estimated when one emitters were impinging on the four-element ULA. The first simulation shows how two signals are recognized by the MUSIC algorithm. There are two independent narrow band signals, the incident angle is 75°, those signal is not correlated, the noise is ideal Gaussian white noise, the SNR is 20dB, the element spacing is half of the input signal wavelength, array element number is 10, the number of snapshots is 200. The sample size was kept constant at T=1000 samples. Emitters were located at 75°.
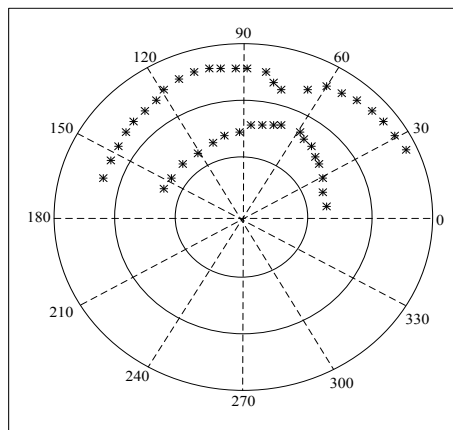
Figure 1 Root locus diagram of the MUSIC algorithm for one source and 30-element ULA.
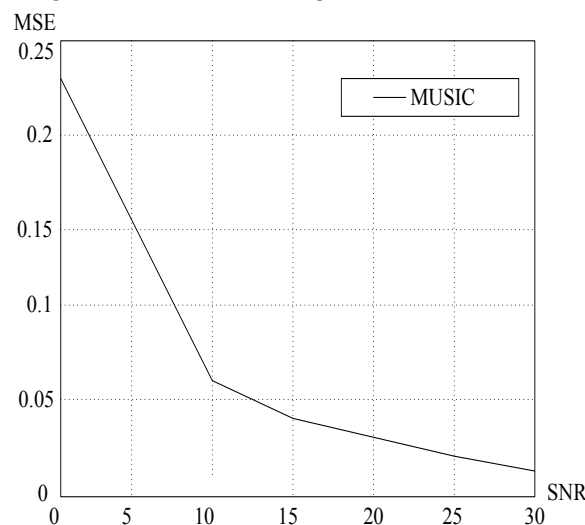

Figure 2 MSE in DOA estimation as a function of the SNR for one source and four-element ULA with sample size T=1000 samples.

In addition, Figures 1-2 shows the polar diagram of the two reciprocal roots provided by the MUSIC algorithm. The MUSIC antenna array as seen in Figure 3. The main board consists of a uniform circular array system with four ADMP504 MEMS microphones (Analog Devices, Norwood, MA, USA), a ADSP21375 (Analog Devices, Norwood, MA, USA) as the core processor, MAXIM MAX11043, 4-Channel, 16-Bit, Simultaneous-Sampling ADCs (Maxim Integrated Products, Sunnyvale, CA, USA) and supplemental hardware circuits. The MAX11043 contains a versatile filter block and programmable-gain amplifier (PGA) per channel. The extended board contains a CSR BC6415 Bluetooth module (Cambridge Silicon Radio, Cambridge, UK), a data acquisition interface and debug interface. The ZYBO is a feature-rich, ready to use, entry level embedded software and digital circuit development platform built around the smallest member of the Z-7101. The Z-7101 is based on the Xilinx All Programmable system on chirp architecture, which tightly integrates a dual- core ARM Cortex-A8 processor with Xilinx 7 Series Field Programmable Gate Array logic. When coupled with the rich set of multimedia and connectivity peripherals available on the ZYBO, the Z-7101 can host a whole system design. The on-board memories, video and audio I/O, dual-role USB, Ethernet, and SD slot will have your design up and ready with no additional hardware needed. Figure 3 illustrates the hardware components that make up the system. Additionally, six connectors are available to put any design on an easy growth path. The SNR error and execution time required for processing uniform linear array with no of elements varying from 7 to 14 is tabulated. we can see that the same SNR value both numerical and experimental approaches gave the same result verifying that with an equivalent

physical array aperture of only 1.5 wavelength the DOA (Direction of arrival) in a decoupled array environment can be estimated with errors smaller than 0.01°. The result as shown in Table. 1.
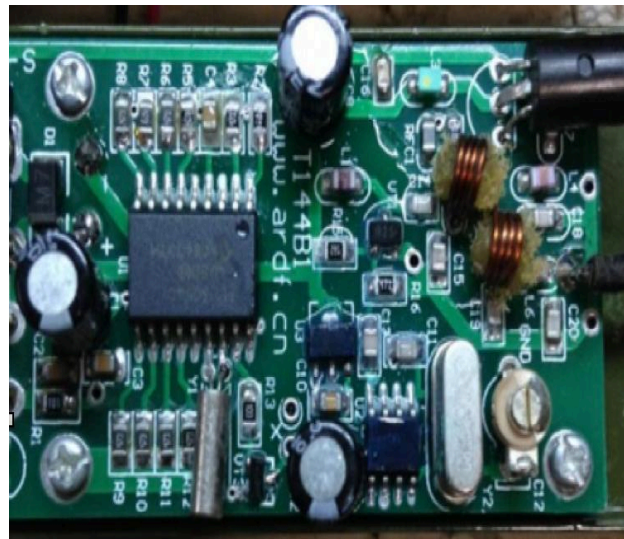

Figure 3 MUSIC antenna arrayfor hardware design system

Table1 Experiment data for searchng accuracy

Test Number	30°	45°	60°	75°
1	29.89°	45.04°	59.60°	74.87°
2	30.05°	45.01°	60.02°	75.12°
3	29.94°	44.57°	60.03°	75.89°
4	30.17°	44.98°	60.01°	75.67°
5	29.95°	44.87°	60.06°	75.95°

4. Conclusion

The MUSIC algorithm was derived. It coined modified root polynomial estimates the DOAs from the polynomial of degree 2L, where L represents number of emitters. The MUSIC algorithm estimates DOAs from the polynomial of degree 2N-2, where N represents number of antennas by selecting L roots that is closest to the unit circle in order to estimate DOA. We have shown by extensive numerical performance evaluation that both algorithms have the same accuracy. Our analysis, based on both numerical and experimental data, shows that with an equivalent physical array aperture of only 1.5 wavelength the DOA in a decoupled array environment can be estimated with errors smaller than 0.01°.

5. References

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