# An Accurate Reduced-Order Polynomial Solution for Root-MUSIC Source Localization Using Displaced Sensor Arrays

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***Abstract –*** This paper proposes a modified Root-MUSIC direction finding algorithm for source localization using a displaced sensor array (DSA) configuration which utilizes two parallely-displaced arrays in the vertical plane. It is shown that the proposed configuration utilizes the spatial displacement of array sensors in both the horizontal and azimuth directions together with the symmetry of the two parallel arrays in order to reduce the rank of the spatial covariance matrix. This results in a reduced-order Root-MUSIC polynomial for which the complex roots correspond to the desired directions of the radiating sources to be localized. Simulation results show that the developed algorithm outperforms the standard Root-MUSIC algorithm for conventional uniform linear arrays (ULAs) in terms of computational efficiency, numerical accuracy, and angular resolution.

***Keywords:*** Smart antennas, source localization, direction finding, and Root-MUSIC.

## I. INTRODUCTION

The development of personal communication devices is a challenging topic in modern electromagnetic research. The number of users that can actually interact at the same time with the base stations is very high. It is therefore necessary to develop efficient methods, which are able to track the desired users and mitigate the effects of interference signals. The use of multiple antennas seems to be very helpful in enhancing the performance of transmit/receive systems in the communication networks. In particular, multiple antennas offer several advantages: They allow increasing the channel capacity; reducing channel fading by using the spatial diversity of the antenna array; and, finally, mitigating co-channel and inter-symbol interferences.

Spatial filtering methods using advanced antenna techniques, smart or adaptive antennas, have received much attention over the last few years. Filtering in the spatial domain can separate spectrally and temporally overlapping signals from multiple mobile users, and hence the performance of a system can be significantly improved. Particular interest in such adaptive antennas has been shown with regard to code-division multiple-access (CDMA) systems. In CDMA systems, all users communicate simultaneously in the same frequency band, and hence multiple-access interference (MAI) is one of the major causes of transmission impairment [1].

Furthermore, since antenna arrays generate beams with a maximum toward the desired users and nulls in the directions of interferences, they play an important role in improving the performance of both the base stations and the mobiles. To this end, an essential step is the source localization or estimation of the directions of arrival (DOAs) of the waves that impinge on the antenna array. Several methods have been proposed in the literature for source localization. Among them, subspace eigenanalysis-based methods such as MUSIC, and its derivative version Root-MUSIC [2], seem to be popular due to their high resolution capability and low computational complexity.

A smart antenna system at the base station of a cellular mobile system is depicted in Fig. 1. It consists of a uniform linear antenna array for which the currents are adjusted by a set of complex weights using an adaptive beamforming algorithm. The adaptive beamforming algorithm optimizes the array output beam pattern such that maximum radiated power is produced in the directions of desired mobile users and deep nulls are generated in the directions of undesired signals representing co-channel interference from mobile users in adjacent cells. Prior to adaptive beamforming, the directions of users and interferes must be obtained using a source localization or direction-of-arrival estimation algorithm [3].

The concept of displaced sensor array (DSA) was introduced by the author in [4-6] where it was shown that such a configuration, which is composed of two parallely-displaced sensor arrays, can improve the performance of the smart antenna system. The proposed DSA configuration has several other advantages. First, it maintains almost the same radiation aperture as the conventional uniform linear array yet it can handle more signals from users and interferers because it has more array sensors when compared to the conventional uniform linear array. Second, the horizontal displacement between the two parallel arrays allows for resolving correlated signals encountered in multipath propagation environment

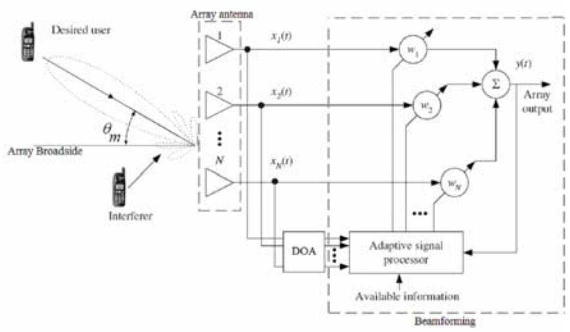


Fig. 1. A functional bolock diagram of a smart antenna system.

without having to apply spatial smoothing techniques. Moreover, the vertical separation between the two parallel arrays allows for resolving signals arriving in the vertical plane at the endfire direction.

In this paper we show that using the proposed DSA configuration results also in a reduced-order Root-MUSIC polynomial. This in turn leads to a more efficient source localization when compared to conventional Root-MUSIC for uniform linear arrays.

The paper is organized as follows: Section II presents the theory of Root-MUSIC source localization for uniform linear array (ULA). Section III derives a reduced-order modified Root-MUSIC polynomial using the proposed displaced sensor array (DSA) configuration. Section IV presents simulation results showing the performance improvement obtained when the modified Root-MUSIC algorithm proposed in this paper is used for cases involving signals incident close to array broadside as well as endfire direction. Finally, some conclusions are given in Section V.

## II. ROOT-MUSIC FOR UNIFORM LINEAR ARRAY (ULA)

Let a uniform linear array (ULA) be composed of $N$ sensors, and let it receive $M$ ($M < N$) narrowband sources $s_m(t)$ impinging from directions $\theta_1, \theta_2, ....\theta_M$, as shown in Fig. 2. Assume that there are $K$ snapshots $\mathbf{x}(1), \mathbf{x}(2), ..., \mathbf{x}(K)$ available. The $N \times 1$ array observation vector is modeled as,

$$\mathbf{x}(t) = \sum_{m=1}^{M} \mathbf{a}_{ULA}(\theta_m) s_m(t) + \mathbf{n}(t) \quad (1a)$$

$$= \mathbf{A}_{ULA}\mathbf{s}(t) + \mathbf{n}(t), \quad (1b)$$

where $\mathbf{A}_{ULA}$ is the $N \times M$ matrix of the signal direction vectors and is given by,

$$\mathbf{A}_{ULA} = [\mathbf{a}_{ULA}(\theta_1), .\mathbf{a}_{ULA}(\theta_2), ...., \mathbf{a}_{ULA}(\theta_M)] \quad (2)$$

$\mathbf{a}_{ULA}(\theta_m)$ is the $N \times 1$ steering vector of a ULA given by,

$$\mathbf{a}_{ULA}(\theta_m) = [1, e^{j(2\pi/\lambda)d\sin\theta_m}, ..., e^{j(2\pi/\lambda)(N-1)d\sin\theta_m}]^T \quad (3)$$

where $1 \leq m \leq M$. In addition $\mathbf{s}(t)$ is an $M \times 1$ vector of source waveforms; $\mathbf{n}(t)$ is an $N \times 1$ vector of white sensor noise; $\lambda$ is the wavelength; $d$ is the inter-element spacing; and $(.)^T$ is the transpose. Equation (3) can be rewritten as,

$$\mathbf{a}_{ULA}(z_m) = \begin{bmatrix} 1 & z_m & . & . & z_m^{N-1} \end{bmatrix}^T \quad (4a)$$

$$= \begin{bmatrix} 1 & e^{j\psi_m} & . & . & e^{j(N-1)\psi_m} \end{bmatrix}^T \quad (4b)$$

$$= [\exp\{(2\pi/\lambda)(n-1)d\sin\theta_m\}]^T, \quad (4c)$$

where $z_m = e^{j\psi_m}$, $\psi_m = (2\pi/\lambda)d\sin\theta_m$, and $1 \leq n \leq N$.

The conventional (forward only) estimate of the covariance matrix is defined as,

$$\mathbf{R} = E\{\mathbf{x}(t)\mathbf{x}^H(t)\} \quad (5)$$

where $E\{.\}$ represents the ensemble average; and $(.)^H$ is the Hermitian transposition operator. Equation (5) can be approximated by applying temporal averaging over $K$ snapshots (or samples) taken from the signals incident on the sensor array. This averaging process leads to forming a spatial correlation (or covariance) matrix $\mathbf{R}$ given by, [7]

$$\mathbf{R} = \frac{1}{K}\sum_{k=1}^{K} \mathbf{x}(k)\mathbf{x}^H(k). \quad (6)$$

Substituting for $\mathbf{x}(t)$ from equation (1) in equation (6) yields,

$$\mathbf{R} = \mathbf{A}_{ULA}\mathbf{R}_{ss}\mathbf{A}_{ULA}^H + \sigma_n^2\mathbf{I} \quad (7)$$

where $\mathbf{R}_{ss} = E\{\mathbf{s}(t)\mathbf{s}^H(t)\}$ is an $M \times M$ source waveform covariance matrix, $\sigma_n^2$ is the noise variance, and $\mathbf{I}$ is an identity matrix. The matrix $\mathbf{R}$ is centro-Hermitian if, [7]

$$\mathbf{R} = \mathbf{J}\mathbf{R}^*\mathbf{J} \quad (8)$$

where $\mathbf{J}$ is the exchange matrix with ones on its anti-diagonal and zeros elsewhere, and $(.)^*$ stands for complex conjugate. The covariance matrix $\mathbf{R}$ in equation (8) is known to be centro-Hermitian if and only if $\mathbf{S}$ is a diagonal matrix, i.e., when the signal sources are uncorrelated.

A common a subspace based DOA estimation algorithm is MUSIC (MUltiple SIgnal Classification) [3]. This method is based on the eigen-decomposition of the covariance matrix $\mathbf{R}$ to into a signal subspace having $M$ eigenvalues with corresponding eigenvectors $\mathbf{v}_1$, $\mathbf{v}_2$ ,......., $.\mathbf{v}_M$, and noise subspace having $(N-M)$ eigenvalues with corresponding eigenvectors $\mathbf{v}_{M+1}$, $\mathbf{v}_{M+2}$ ,......., $.\mathbf{v}_N$. Let $\mathbf{V}s$ be the matrix whose columns are the source subspace eigenvectors, and $\mathbf{V}_n$

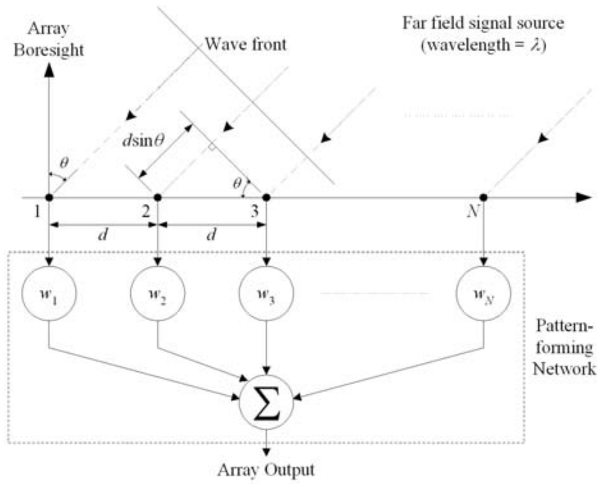


Fig. 2. Geometry of a uniform linear array (ULA).

be the matrix whose columns are the noise subspace eigenvectors, i.e.,

$$\mathbf{V}s = [\mathbf{v}_1, \mathbf{v}_2, ..., \mathbf{v}_M] \tag{9}$$

$$\mathbf{V}_n = [\mathbf{v}_{M+1}, \mathbf{v}_{M+2}, ..., \mathbf{v}_N]. \tag{10}$$

The covariance matrix $\mathbf{R}$ can then be expressed as, [7]

$$\mathbf{R} = \mathbf{V\Pi V} \tag{11a}$$
$$= \mathbf{V}_s\mathbf{\Pi}_s\mathbf{V}_s + \sigma^2\mathbf{V}_n\mathbf{V}_n^H \tag{11b}$$

where the subscripts $s$ and $n$ stand for signal and noise subspace, respectively. In equation (11a) $\mathbf{\Pi}_s$ is a diagonal matrix given by $\mathbf{\Pi}_s = diag\{\pi_1, \pi_2, ..., \pi_M\}$. Hence, the normalized MUSIC angular spectrum is defined as, [7, 8]

$$P_{ULA}(\theta) = \frac{\mathbf{A}_{ULA}^H\mathbf{A}_{ULA}}{\mathbf{A}_{ULA}^H\mathbf{V}_n\mathbf{V}_n^H\mathbf{A}_{ULA}}. \tag{12}$$

By examining the denominator in equation (12) it is evident that peaks in the MUSIC angular spectrum occur at angles $\theta$ for which the array manifold matrix $\mathbf{A}_{ULA}$ is orthogonal to the noise subspace matrix $\mathbf{E}_n$. Those angles $\theta$ define the desired angles of arrival of the of the narrownband source signals impinging on the ULA. A comprehensive performance evaluation of the MUSIC algorithm for DOA estimation can be found in [9, 12].

The denominator in equation (12) can be rearranged to form the conventional Root-MUSIC polynomial as, [8]

$$G_{ULA}(z) = \mathbf{a}_{ULA}^T(1/z)\mathbf{V}_n\mathbf{V}_n^H\mathbf{a}_{ULA}(z). \tag{13}$$

The roots of the polynomial $G_{ULA}(z)$ defined in equation (13) occur in pairs ($z$ and $1/z$) with the same argument in the $z$-plane. The roots $z_m$ which lie on the unit circle (or close to it) will correspond to the directions of arrival $\theta_m$ of the incident signals while the other roots are spurious roots. The directions of arrival $\theta_m$ are thus given by,

$$\theta_m = \sin^{-1}\left[(\frac{\lambda}{2\pi d})\arg(z_m)\right]. \tag{14}$$

## III. MODIFIED ROOT-MUSIC FOR DISPLACED SENSOR ARRAY (DSA)

The DSA configuration consists of two parallel ULAs displaced by a horizontal distance $d = \lambda/4$ and vertical separation $s = \lambda/2$, as shown in Fig. 3. Each of the two parallel ULAs consists of $N$ linear equispaced omnidirectional sensors with inter-element spacing $d = \lambda/2$.The two parallel ULAs are positioned along the $x$ axis with an azimuth angle $\theta_m$ measured with respect to the $z$ axis. It is assumed that the DSA configuration receives $M$ narrowband source signals $s_m(t)$ from incidence directions $\theta_1, \theta_2, ....\theta_M$. At a particular instant of time $t = 1, 2, ..., K$, where $K$ is the total number of snapshots taken, the desired users signal vector $\mathbf{x}(t)$ is given by,

$$\mathbf{x}(t) = \sum_{m=1}^{M}[\mathbf{a}_1(\theta_m) + \mathbf{a}_2(\theta_m)]\,s_m(t) + \mathbf{n}(t) \tag{15}$$

where $\mathbf{n}(t)$ is the sensor noise vector modeled as temporally white and zero mean complex Gaussian process, $\mathbf{a}_1(\theta_m)$ and $\mathrm{a}_2(\theta_m)$ are the steering (or response) vectors for the two parallel arrays with respect to $\theta_m$, which represents the angle of arrival of the $m$ th signal. The first steering vector $\mathbf{a}_1(\theta_m)$ has dimensions $N \times 1$ and represents the space factor of the first array with respect to direction $\theta_m$. It is given by,

$$\mathbf{a}_1(\theta_m) = \{\exp[j(n-1)\psi_m]\}^T, \qquad 1 \leq n \leq N \tag{16}$$

where $\{\}^T$ is the transposition operator, and $\psi_m$ represents the electrical phaseshift from element to element along the array defined as $\psi_m = 2\pi(d/\lambda)\sin\theta_m$, where $d$ is the inter-element spacing and $\lambda$ is the wavelength of the received signal. The second steering vector $\mathbf{a}_2(\theta_m)$ has dimensions $N \times 1$ also and represents the space factor of the second array with respect to direction $\theta_m$. It is given by,

$$\mathbf{a}_2(\theta_m) = \mathbf{a}_1(\theta_m).F_s(\theta_m).F_\Delta(\theta_m) \tag{17}$$

where $F_s(\theta_m)$ and $F_\Delta(\theta_m)$ represent the space factors due to the vertical separation $s$ and horizontal displacement $\Delta$ of the two parallel arrays, respectively. These are given by,

$$F_s(\theta_m) = \exp\left[-j2\pi\left(\frac{s}{\lambda}\right)\cos\theta_m\right] \tag{18}$$

$$F_\Delta(\theta_m) = \exp\left[-j2\pi\left(\frac{\Delta}{\lambda}\right)\sin\theta_m\right]. \tag{19}$$

Equation (15) can be written as,

$$\mathbf{x}(t) = \sum_{m=1}^{M} \mathbf{a}_{DSA}(\theta_m)s_m(t) + \mathbf{n}(t) \qquad (20)$$

where $\mathbf{a}_{DSA}(\theta_m)$ is the DSA steering vector given as the superposition of the steering vectors of the two parallely-displaced ULAs, i.e.,

$$\mathbf{a}_{DSA}(\theta_m) = \mathbf{a}_1(\theta_m) + \mathbf{a}_2(\theta_m). \qquad (21)$$

The combination of all possible steering vectors forms the array manifold (or steering vector) matrices $\mathbf{A}_1$ and $\mathbf{A}_2$ of size $N \times M$ each, i.e.,

$$\mathbf{A}_1 = [\mathbf{a}_1(\theta_1), .\mathbf{a}_1(\theta_2), ...., \mathbf{a}_1(\theta_M)] \qquad (22)$$

$$\mathbf{A}_2 = [\mathbf{a}_2(\theta_1), .\mathbf{a}_2(\theta_2), ...., \mathbf{a}_2(\theta_M)]. \qquad (23)$$

The received signal vector $\mathbf{x}(t)$ of equation (15) can then be written as,

$$\mathbf{x}(t) = \mathbf{A}_{DSA}\mathbf{s}(t) + \mathbf{n}(t) \qquad (24)$$

where $\mathbf{A}_{DSA}$ is the overall array manifold matrix and is given by,

$$\mathbf{A}_{DSA} = \mathbf{A}_1 + \mathbf{A}_2. \qquad (25)$$

The covariance matrix for the DSA configuration can then be expressed as,

$$\mathbf{R} = \mathbf{A}_{DSA}\mathbf{R}_{ss}\mathbf{A}_{DSA}^H + \sigma_n^2\mathbf{I}. \qquad (26)$$

Using the expression for $\mathbf{A}_{DSA}$ as given in (24), the denominator in equation (12) results in the following modified Root-MUSIC polynomial,

$$G_{DSA}(z) = \mathbf{a}_{DSA}^T(1/z)\mathbf{V}_n\mathbf{V}_n^H\mathbf{a}_{DSA}(z). \qquad (27)$$

$G_{DSA}(z)$ given in equation (27) for the proposed DSA has a reduced order compared to the order of the conventional Root-MUSIC polynomial $G_{ULA}(z)$ derived in equation (13) for ULA.

## IV. SIMULATION RESULTS

### *A. Detection of Signals Incident Closer to Array Broadside Direction*

The results for Root MUSIC were obtained using a ULA configuration with $N = 12$ elements in the array, and compared them to those obtained using the modified Root-MUSIC using the DSA configuration with $N/2 = 6$ elements in each array. Inter-element spacing of $d = \lambda/2$ is maintained in both configurations. This is essential to

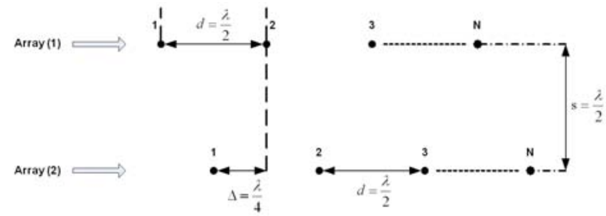


Fig. 3. Proposed displaced sensor array (DSA) configuration consists of two parallely-displaced uniform linear arrays (ULAs) in the vertical plane displaced horizontally by a distance $\Delta = \lambda/4$ and separated vertically by a distance $s = \lambda/2$.

reduce the effects of inter-element mutual coupling. We have assumed a signal-to-noise ratio $SNR = 10$dB and the number of snapshots $K = 100$. There are $M = 5$ incoming signals to be detected arriving at directions $\theta_m$ close to the array broadside direction ($\theta = 0^o$). These directions are: $\theta_1$=30$^o$, $\theta_2$=20$^o$, $\theta_3$=10$^o$, $\theta_4$=-20$^o$, and $\theta_5$=-40$^o$.

Results are presented in Table 1 showing that more accurate results with less percentage error are obtained for the DOA estimates when the modified Root-MUSIC algorithm is used. These results are also plotted in Fig. 4 which clearly demonstrates that the modified Root-MUSIC for DSA outperforms conventional Root-MUSIC for ULA in terms of numerical accuracy coupled with computational efficiency. The latter feature is evident from the reduced-order complex-root polynomial obtained as a result of the modified Root-MUSIC algorithm proposed for DSA of this paper.

### *B. Detection of Signals Incident Closer to Array Endfire Direction*

The results for Root MUSIC were obtained using a ULA configuration with $N = 14$ elements in the array, and compared them to those obtained using the modified Root-MUSIC using the DSA configuration with $N/2 = 7$ elements in each array. Inter-element spacing of $d = \lambda/2$ is maintained in both configurations. We have assumed a signal-to-noise ratio $SNR = 10$dB and the number of snapshots $K = 100$. There are $M = 6$ incoming signals to be detected arriving at directions $\theta_m$ close to the array endfire direction ($\theta = \pm 90^o$). These directions are: $\theta_1$=-85$^o$, $\theta_2$=-80$^o$, $\theta_3$=-75$^o$, $\theta_4$=75$^o$, $\theta_5$=80$^o$, and $\theta_6$=85$^o$.

Results are presented in Table 2 showing that more accurate results with less percentage error are obtained for the DOA estimates when the modified Root-MUSIC algorithm is used. These results are also plotted in Fig. 5 which clearly demonstrates that the modified Root-MUSIC for DSA outperforms conventional Root-MUSIC for ULA in terms of numerical accuracy coupled with computational efficiency. The latter feature is evident from the reduced-order complex-root polynomial obtained as a result of the modified Root-MUSIC algorithm proposed for DSA of this paper.

Table 1. Comparison of Root-MUSIC and Modified Root-MUSIC (incidence close to array broadside direction) ($M = 5$, $d = 0.5\lambda$, $SNR = 10$dB, and $K = 100$).

| | Root-MUSIC | | Modified Root-MUSIC | |
|---|---|---|---|---|
| $\theta_m^{exact}$ | $\theta_m$ | % error | $\theta_m$ | % error |
| $30^o$ | $30.78^o$ | 2.60% | $30.11^o$ | 0.37% |
| $20^o$ | $20.53^o$ | 2.65% | $20.21^o$ | 1.05% |
| $10^o$ | $10.16^o$ | 1.60% | $10.02^o$ | 0.20% |
| $-20^o$ | $-20.33^o$ | 1.65% | $-20.07^o$ | 0.35% |
| $-40^o$ | $-40.6^o$ | 1.50% | $-40.12^o$ | 0.3% |

Table 2. Comparison of Root-MUSIC and Modified Root-MUSIC (incidence close to array endfire direction) ($M = 6$, $d = 0.5\lambda$, $SNR = 10$dB, and $K = 100$).

| | Root-MUSIC | | Modified Root-MUSIC | |
|---|---|---|---|---|
| $\theta_m^{exact}$ | $\theta_m$ | % error | $\theta_m$ | % error |
| $-85^o$ | $-76.2^o$ | 10.35% | $-80.9^o$ | 4.82% |
| $-80^o$ | $-73.1^o$ | 8.63% | $-76.8^o$ | 4.00% |
| $-75^o$ | $-70.9^o$ | 5.47% | $-73.5^o$ | 2.00% |
| $75^o$ | $71.9^o$ | 4.13% | $73.8^o$ | 1.60% |
| $80^o$ | $75.3^o$ | 5.88% | $77.2^o$ | 2.25% |
| $85^o$ | $77.5^o$ | 8.82% | $82.9^o$ | 2.47% |

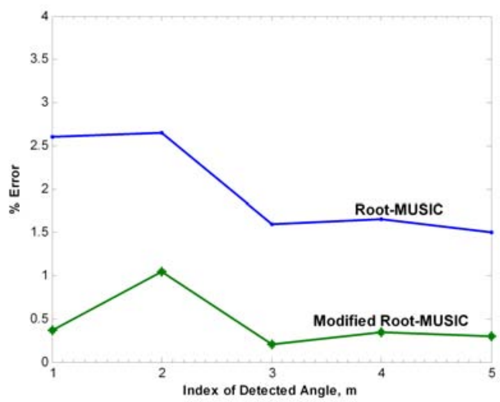


Fig. 4. Comparison of Root-MUSIC and Modified Root-MUSIC for incidence close to the array broadside direction.

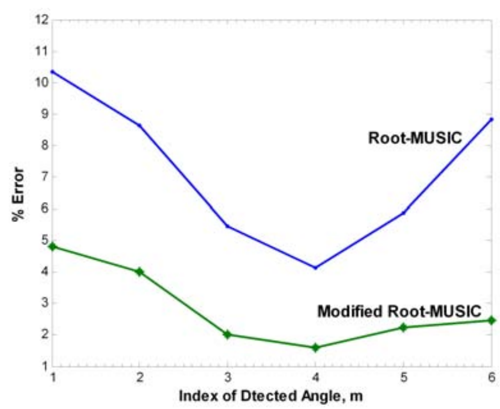


Fig. 5. Comparison of Root-MUSIC and Modified Root-MUSIC for incidence close to the array endfire direction.

## V. CONCLUSIONS

We have used a parallely-displaced sensor array configuration to derive a reduced-order polynomial expression for the Roo-MUSIC source localization method. The derived reduced-order complex-root polynomial makes the proposed method for displaced sensor arrays much more efficient in terms of computational complexity when compared to the conventional Roo-MUSIC method used in conjunction with uniform linear arrays. Moreover, the horizontal displacement in the displaced sensor array configuration between the two parallel arrays allows for resolving correlated signals encountered in multipath propagation environment without having to apply spatial smoothing techniques [11]. Moreover, the vertical separation between the two parallel arrays allows for resolving signals arriving in the vertical plane at the endfire direction. Results were presented to show the improved performance of the proposed method in terms of computational efficiency and numerical accuracy.

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