Mode separation and direction of arrival estimation in HF links

F. Arikan,[1] N. Yilmaz,[2] O. Arikan,[3] and M. K. B. H. Miled[4]

Received 24 January 2002; revised 15 October 2002; accepted 25 November 2002; published 5 February 2003.

[1] Estimation of arrival angles and incoming signals is a challenging problem for HF channels where the signals are correlated and the separation between the signals can be as low as a couple of degrees. In this paper, a new algorithm, Multipath Separation-Direction of Arrival (MS-DOA), is developed to estimate both the arrival angles in elevation and azimuth and the incoming signals at the output of the reference antenna with very high accuracy. The MS-DOA algorithm provides reliable angle and signal estimates even with small separation of arrival angles and for low SNRs. The minimum number of antennas that are required by the algorithm is only one more than the number of incoming signals. In a narrowed down region of interest and for a few incoming signals, the computational search time for MS-DOA is only a couple of minutes in a standard PC. *INDEX TERMS:* 6974 Radio Science: Signal processing; 6979 Radio Science: Space and satellite communication; 2494 Ionosphere: Instruments and techniques; *KEYWORDS:* direction finding, Multipath Separation, DOA estimation

Citation: Arikan, F., N. Yilmaz, O. Arikan, and M. K. B. H. Miled, Mode separation and direction of arrival estimation in HF links, *Radio Sci.*, *38*(1), 1007, doi:10.1029/2002RS002606, 2003.

1. Introduction

[2] Ionospheric channels exhibit random space, time and frequency variations which cause various degrading effects on the transmitted signals including multipath and polarization fading. Depending on the frequency spectrum of the transmitted signal, the structure of the ionosphere and Earth's magnetic field, the electromagnetic wave entering the ionosphere may split into ordinary (O) and extraordinary (X) modes and also travel through different paths each having its own time delay, polarization, doppler spread, attenuation, wavenumber, group delay and phase shift [*Goodman*, 1992]. Thus when these modes from different layers and/or ground reflections are collected at the receiving antenna array, the signals may add up destructively or constructively. This situation is generally referred to as multipath fading. The O and X modes undergo Faraday rotation and due to the time variation of the electron content, the rotation angle changes with time causing polarization fading at the receiver array.

[3] For proper recovery of the transmitted signals, the modes and mutipath components need to be successfully separated at the receiver. There have been various efforts to separate the modes and overcome the degrading effects of fading. One major direction is to apply diversity techniques including angle of arrival (AOA), polarization, frequency and time diversity to cope with multipath and polarization fading. Although all of the diversity methods have certain advantages, due to time, space and frequency variant structure of the ionosphere, none of the above listed diversity techniques is a universal solution to the fading problem.

[4] One of the most commonly used adaptive receiver for the fading multipath channel of HF has been the RAKE receiver which is formed by two tapped delay lines (or matched filters) and the outputs are compared for decision making [*Proakis*, 1995]. The tap gains of the delay lines are adjusted by cross correlating the received signal by the reference signals at the receiver. Although this kind of receiver has been in use for some time, it has been demonstrated that its performance for channels with even moderate Intersymbol Interference (ISI) is unacceptable. Also, according to the data provided by the International Telecommunications Union (ITU) on the structure of the ionosphere, the time delay between two

[1]Department of Electrical and Electronics Engineering, Hacettepe University, Ankara, Turkey.
[2]Turkish Navy, NATO Snake-pit Headquarters, Ankara, Turkey.
[3]Department of Electrical and Electronics Engineering, Bilkent University, Ankara, Turkey.
[4]Department of Electrical and Computer Engineering, Northeastern University, Boston, Massachusetts, USA.

0048-6604/03/2002RS002606

modes can be as high as 5 ms and as low as 1 μs [*International Telecommunications Union-Radiocommunications* (*ITU-R*), 1992]. Thus in order to accommodate and separate the modes, the total length of the tapped delay line should be extremely large. This requirement increases the computational complexity and memory load and thus the system performance degrades accordingly.

[5] The ordinary (O) and extraordinary (X) modes of the electromagnetic wave in the ionosphere have orthogonal polarizations. Thus, ideally, by employing two perpendicular antennas oriented according to the Earth's magnetic field components, these modes can be resolved [*Edjeou et al.*, 1993a; *Afraimovich et al.*, 1999; *Compton*, 1981; *Erhel et al.*, 1994]. In the ionosphere, O and X modes travel with different wave vectors, different paths, different time delays and different frequency shifts causing the modes to suffer different amplitude, phase and polarization variations. The variation of the polarization is also known as Faraday Rotation. The Faraday Rotation is proportional to both the distance traveled and the difference in wave numbers of the ordinary and extraordinary modes. The wave number is a function of the relative dielectric constant in which electron concentration is the major parameter. In order to apply this technique successfully and separate the modes exactly at the receiver site, the polarization properties of the waves, the electron concentrations at the exit of the ionosphere, and directions of the Earth's magnetic field components are need to be known at the receiver with very high accuracy. Moreover, in the techniques that separate the modes using the polarization components on the orthogonal antennas, the components separated are not necessarily the orthogonal components of the modes at the exit of the ionosphere. Thus, although separation of modes with this method is possible theoretically, the results are not very reliable when various multimode signals impinge on the antennas from multilayer ionosphere [*Afraimovich et al.*, 1999; *Compton*, 1981]. The mode separation using the orthogonality of polarization method also produces erroneous results when the polarization of the modes are linear (quasi transverse propagation) instead of being circular [*Edjeou et al.*, 1993a; *Erhel et al.*, 1994].

[6] Adaptive Direction Finding (DF) algorithms are more commonly used to separate the signals arriving on to the antenna array from various directions. All the modes exiting the ionosphere arrive to the receiving antenna with different elevation and azimuth angles. Adaptive DF algorithms are used to determine these arrival angles and thus separate the signals accordingly. Although various methods are reported in the literature for separation of multipath signals [*Godara*, 1997], eigenstructure methods such as Multiple Signal Classification (MUSIC) [*Schmidt*, 1986], CLOSEST [*Buckley and Xu*, 1990] and Estimation of Signal Parameters via Rotational Invariance Techniques (ESPRIT) [*Roy and Kailath*, 1989] are widely used since they can separate the angles with high resolution. Although these techniques are advantageous over the classical DF methods, with typical homogeneous array apertures, the algorithms fail to distinguish signals which are highly correlated (like multipath signals) and the resolution capability may be a couple of degrees [*Godara*, 1997; *Pillau*, 1989; *Roy and Kailath*, 1989; *Chenu-Tournier et al.*, 2000]. In order to cope with these disadvantages, preprocessing techniques like forward-backward smoothing are employed [*Godara*, 1997; *Pillau*, 1989; *Williams et al.*, 1988]. Yet, in order to use these preprocessing methods, the number of antennas that are utilized in the receiving array has to be doubled and also the computational complexity increases [*Pillau*, 1989].

[7] Algebraic methods for deterministic source separation and direction of arrival estimation have certain advantages over the adaptive techniques [*Van Der Veen*, 1998]. These methods act on a block of data and do not exploit the source statictics. The sources impinging on the antenna array are determined as a collection of eigenvalues and eigenvectors utilizing the subspace properties of the antenna array response matrix [*Van Der Veen*, 1998]. In this paper, we adopted the basic algebraic subspace methods for blind source estimation and developed an algorithm which can separate the multipath modes successfully and find their arrival angles with high accuracy. With typical array apertures, the resolution capability of the developed method (which will be called MS-DOA for Multipath Separation-Direction of Arrival) can get as low as 0.2 degrees without the help of any preprocessing techniques. For homogeneous arrays, the number of antennas that are required in the array has to be one more than the number of incoming signals. The developed technique also allows the user to recover the multipath signals with very high accuracy. In MS-DOA, both the array output vector and incoming signal vector are expanded in terms of a basis vector set. A linear system of equations equation is formed using the coefficients of the basis vector for the array output vector, the incoming signal vector and the array manifold. The angles of arrival in elevation and azimuth are obtained as the maximizers of the sum of the magnitude squares of the projection of the signal coefficients on the column space of the array manifold. Once the array manifold is estimated then the incoming signals can also be determined using the basis vectors and signal coefficients. For certain array configurations, the search for maximizing angles can be eliminated by using closed form solutions of the constructed linear system [*Yilmaz*, 2000].

[8] In section 2, the channel simulation model which is used to obtain synthetic signals is discussed. The simu-

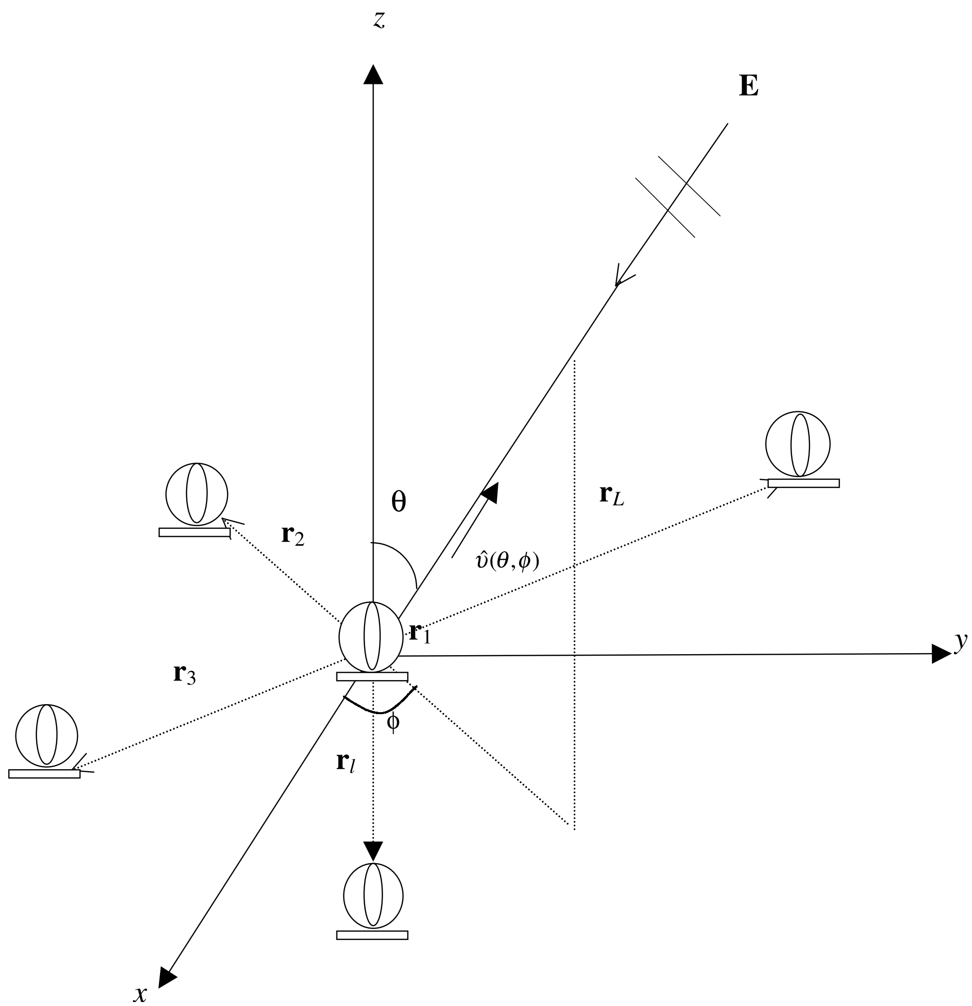


Figure 1. Incoming electric field and the coordinate system for the receiving array.

lated signals are then used to test the performance of the new MS-DOA algorithm described in section 3 and the results are provided in section 4.

2. HF Channel Simulation

[9] The reliability and error performance of direction finding (DF) algorithms can be tested by making use of sufficiently realistic synthetic signals obtained from a computer simulation program which includes ionospheric multipath, free space propagation and receiving antenna effects. In this simulation model, the receiving array consists of L identical antennas and the sensors are arranged in a coordinate system as shown in Figure 1. The reference antenna is placed at $\mathbf{r}_1$ which is the origin of this coordinate system. In order to obtain synthetic HF signals, we implemented a channel model which considers each propagation mode as a separate signal. Thus, the transmitted signal $m(t)$ is received from a total of K modes which are assumed to travel in K different paths as denoted in Figure 2. The propagation of the wave is examined in five different stages as denoted in Figure 2. In the first stage, for each mode k, the transmitted signal $m(t)$ is convolved with the impulse response of the kth mode to give the signal at the output of the ionosphere, $C_k(t)$. The second stage of the simulation consists of the free space propagation path from the exit of the ionosphere to the reference antenna of the receiving array. The effect of this free-space propagation path is modeled by the function $\mathbf{F}_k(r_k)$, which denotes the attenuation of the magnitude of the wave and the phase shift at a distance r_k for the kth mode. In this model the signal $C_k(t)$ is considered to be radiated omnidirectionally with a polarization assigned according to the Earth's magnetic field. This polarization vector is also included in $\mathbf{F}_k(r_k)$. In the third stage, white Gaussian noise, $\mathbf{n}_k(t) = n_k(t)\hat{\mathbf{n}}_k$, is

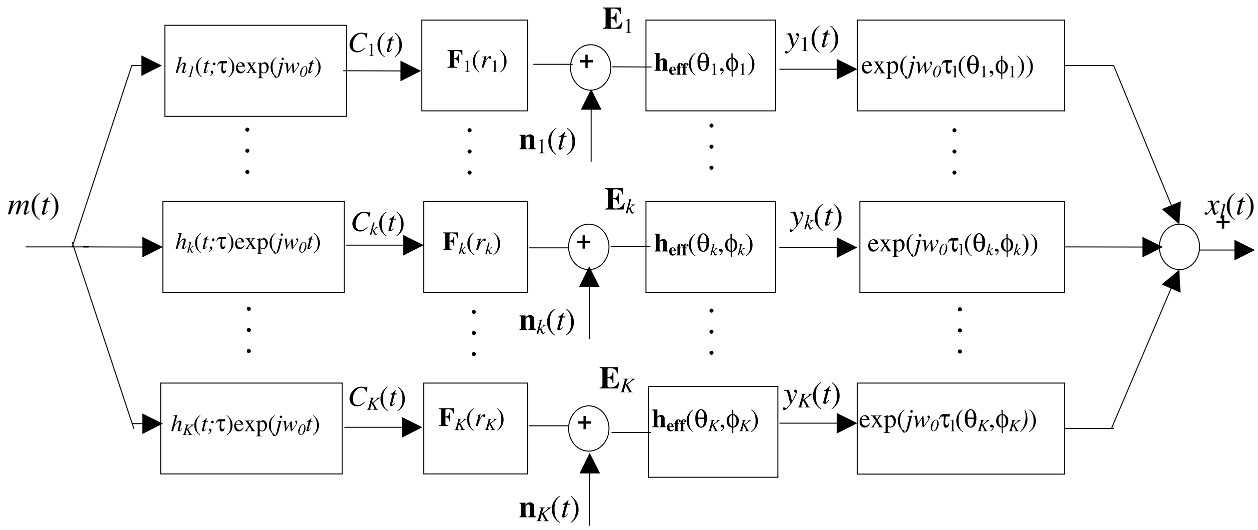


Figure 2. Outline of the simulation program.

added to the electromagnetic wave. Here, $\hat{\mathbf{n}}_k$ is the unit vector for the noise added on the kth mode as shown in Figure 2. The fourth stage consists of the reception of the incoming wave by the reference antenna. The open circuit voltage at the output of the reference antenna is obtained by the dot product of $\mathbf{E}_k$, the incident electric field of the kth mode and the effective length of the antenna, $\mathbf{h_{eff}}(\theta_k, \phi_k)$. Effective length is a function of antenna polarization and antenna pattern and summarizes the characteristics of the receiving antenna [*Collin*, 1985]. Using the schematic model provided in Figure 2, the open circuit voltage at the reference antenna due to the kth mode, $y_k(t)$, can be expressed as follows:

$$y_k(t) = \{C_k(t)\mathbf{F}_k(r_k) + \mathbf{n}_k(t)\} \cdot \mathbf{h_{eff}}(\theta_k, \phi_k). \qquad (1)$$

In the last stage of simulation, the voltage waveforms at the output of each antenna in the sensor array are obtained. The output signal of the lth antenna (where $1 \leq l \leq L$) due to K incoming signals (Figure 2) can be expressed as

$$x_l(t) = \sum_{k=1}^{K} y_k(t) e^{jw_0\gamma_l(\theta_k,\phi_k)}, \qquad (2)$$

where $y_k(t)$ is the open circuit voltage at the output of the reference antenna due to kth impinging signal and $\gamma_l(\theta_k, \phi_k)$ is the time delay between the lth sensor and the reference sensors for a plane wave arriving from the direction (θ_k, ϕ_k). The time delay $\gamma_l(\theta_k, \phi_k)$ is equal to

$$\gamma_l(\theta_k, \phi_k) = (\mathbf{r}_l/c).\hat{v}(\theta_k, \phi_k), \qquad (3)$$

$\mathbf{r}_l$ is the position vector of lth antenna, c is the speed of light in vacuum and $\hat{v}(\theta_k, \phi_k)$ is the unit vector in the direction of (θ_k, ϕ_k) as depicted in Figure 1.

[10] In equation (1), $C_k(t)$ denotes the output signal of the ionosphere due to the kth mode and it is defined as

$$C_k(t) = [h_k(t;\tau) * m(t)] \exp(j2\pi f_0 t), \qquad (4)$$

where $h_k(t; \tau)$ is the time varying low-pass HF channel response of the kth mode; * denotes convolution and f_0 is the carrier frequency. $\mathbf{F}_k(r_k)$ represents the propagation of the electromagnetic wave from the exit point at the ionosphere to the receiver array. If the modes are considered to be point sources at the exit of the ionosphere, then the propagation function is the free space Green's function which can be given as follows:

$$\mathbf{F}_k(r_k) = F_k(r_k)\mathbf{f}_k, \qquad (5)$$

where $F_k(r_k)$ is the free space Green's function

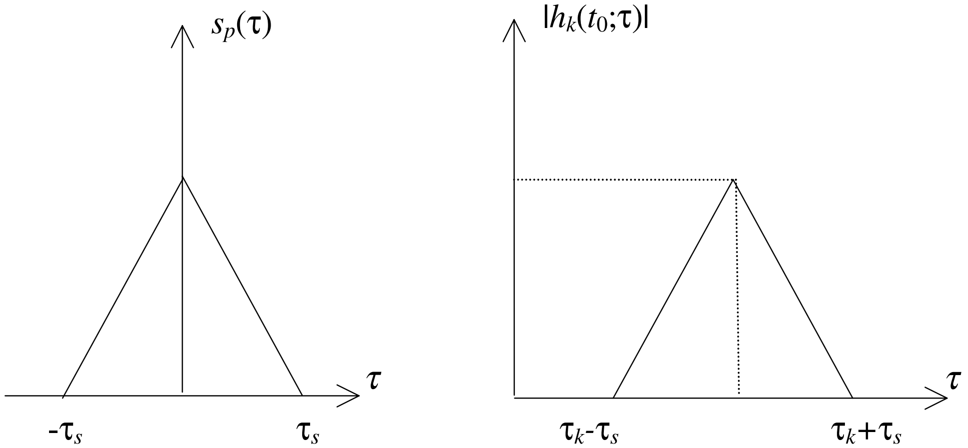
$$F_k(r_k) = jk_0 Z_0 \frac{e^{-jk_0 r_k}}{4\pi r_k}, \qquad (6)$$

and $\mathbf{f}_k$ is the polarization vector of the wave which is defined according to the model given by *Bertel et al.* [1989]. In the above relations, k_0 is the wave number; Z_0 is the free space impedance; and r_k is the distance from the exit of the ionosphere to the reference antenna for the kth mode. The polarization vector $\mathbf{f}_k$ is defined as

$$\mathbf{f}_k = [0 \quad jp_k \quad 1]^T, \qquad (7)$$

where the polarization coefficient p_k represents the polarization of the kth mode. The superscript T denotes the transpose.

[11] For the time varying HF channel impulse response various alternatives are available including *Watterson et al.* [1970], *ITU* [1998], and *Bertel et al.* [1996]. Any of



a) b)

Figure 3. (a) An example of spread function. (b) The magnitude of channel impulse function at a given time t_0 after the spread function in part (a) is applied.

these models can be used in the simulation program. In this study, we have chosen the model proposed by *Bertel et al.* [1996] since the amplitude and polarization ratios for each mode are defined using experimental values. This expression for the impulse response is then modified by adding a spread function to model the effect of thickness of the ionospheric layers. Thus $h_k(t;\tau)$ for the kth mode can be written as

$$h_k(t;\tau) = A_k s_p(\tau - \tau_k)\exp(j[\Delta w_{dk}(t)]t), \quad (8)$$

where A_k denotes the attenuation factor of the kth mode and it is usually taken as a constant for all modes, $s_p(\tau)$ is the spread function and an example is given in Figure 3a. The group delay of the kth mode is denoted by τ_k and the time varying Doppler shift is given as

$$\Delta w_{dk}(t) = 2\pi\left[\Delta f_{ok} + aE_{lk}\cos\left(\frac{2\pi t}{T_o} + \psi_k\right)\right]. \quad (9)$$

In the above equation, Δf_{ok} is the average doppler shift due to daily movements of the ionosphere, a is the amplitude proportionality coefficient in Hz and for an HF link of 1000 km, it has a typical value of 1; E_{lk} is the elevation angle of the kth mode and from Figure 1, $E_{lk} = 90 - \theta_k$; T_o denotes the perturbation period of the ionosphere due to the gravity waves with a typical value of 15 minutes; ψ_k denotes the phase origin of these perturbations and it is considered to be the same for all modes. For a fixed observation time t_0, $|h_k(t_0;\tau)|$ is given in Figure 3b.

[12] The signal model provided in equation (1) can be used to realize various multimode/multipath scenarios. The simulation data obtained from such a channel model is used in section 4 to test the performance of the proposed MS-DOA algorithm and compare the results with other DF algorithms like MUSIC. The simulation parameters for the scenarios are provided in section 4.

3. Proposed Method for Multipath Separation and Direction of Arrival Estimation

[13] Our goal is to determine the angle of arrivals of K modes in elevation and azimuth (θ_k; ϕ_k) using the L antenna output signals and then to estimate the impinging signals on the reference antenna as depicted in Figure 4. Yet, these estimated signals from the K modes do not provide the initial transmitted signal $m(t)$. In order to obtain $m(t)$, further processing is necessary to estimate the channel response. By properly equalizing the channel impulse response corresponding to the kth mode, channel input signal can be estimated as detailed by *Miled and Arikan* [2000].

[14] As shown in Figure 1, assuming a total of K electromagnetic waves impinging on an array of L antennas, the angles of arrival of each wave will be estimated by the MS-DOA algorithm by using a block processing technique. Following a down-conversion stage, the baseband output of the sensors are sampled with a period of T_s. Then, blocks of length N_s samples are formed as

$$\mathbf{x}_{l,q} = [x_l((q-1)N_s) \ \ldots \ x_l(qN_s)]^T, \quad (10)$$

where q denotes the block number. Likewise, let $\mathbf{y}_{k,q}$ defined as $\mathbf{y}_{k,q} = [y_k((q-1)N_s) \ldots y_k(q N_s)]^T$ denote the

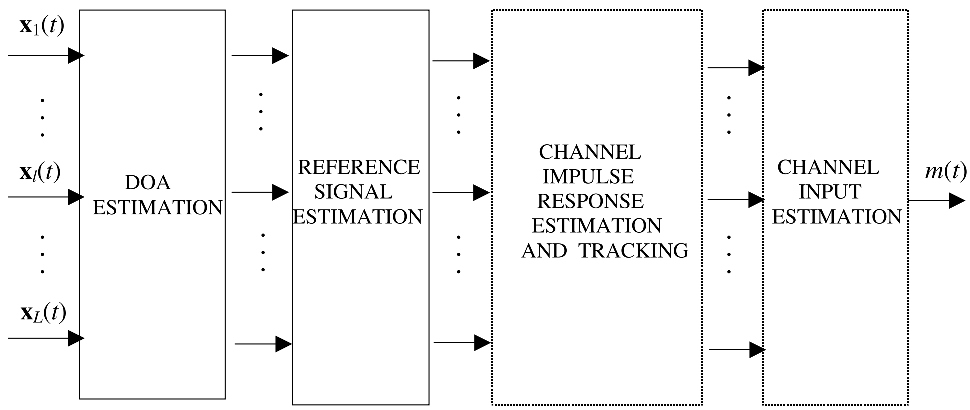


Figure 4. MS-DOA estimates the arrival angles and the incoming signals at the output of the reference antenna for all modes. From this information, channel impulse response and channel input can also be estimated as a possible extension.

samples of the qth sample block of output of the reference antenna for the kth impinging wave. In the following, the block numbers are suppressed for notational simplicity. The measurement model of the impinging waves given in equation (2) can be rewritten in the following matrix-vector form:

$$\mathbf{X} = \mathbf{Y}\mathbf{A}^T, \tag{11}$$

where

$$\mathbf{X} = [\mathbf{x}_1 \ \ldots \ \mathbf{x}_l \ \ldots \ \mathbf{x}_L] \tag{12}$$

$$\mathbf{Y} = [\mathbf{y}_1 \ \ldots \ \mathbf{y}_k \ \ldots \ \mathbf{y}_K] \tag{13}$$

and

$$\mathbf{A}^T = \begin{bmatrix} A_1(\mathbf{a}_1) & A_2(\mathbf{a}_1) & \cdots & A_L(\mathbf{a}_1) \\ \vdots & \vdots & \vdots & \vdots \\ A_1(\mathbf{a}_K) & A_2(\mathbf{a}_K) & \cdots & A_L(\mathbf{a}_K) \end{bmatrix}, \tag{14}$$

where $A_l(\mathbf{a}_k) = e^{jw_0\gamma_l(\theta_k,\phi_k)}$ and $\mathbf{a}_k = [\theta_k \ \phi_k]^T$. Since $\mathbf{x}_l$ values are linear combinations of $\mathbf{y}_k$ values, the rank of $\mathbf{X}$ can be at most K. Thus we can find K basis vectors $\mathbf{b}_k$ to span the signal subspace which $\mathbf{x}_l$'s belong to. If $\mathbf{X}$ and $\mathbf{Y}$ are expanded into such basis vectors, the following expressions are obtained:

$$\mathbf{X} = [\mathbf{b}_1 \ldots \mathbf{b}_k \ldots \mathbf{b}_K][\mathbf{X}_1 \ldots \mathbf{X}_k \ldots \mathbf{X}_K]^T \tag{15}$$

$$\mathbf{Y} = [\mathbf{b}_1 \ldots \mathbf{b}_k \ldots \mathbf{b}_K][\mathbf{Y}_1 \ldots \mathbf{Y}_k \ldots \mathbf{Y}_K]^T, \tag{16}$$

where $\mathbf{X}_k = [X_{1k} \ X_{2k} \ldots X_{Lk}]^T$ and $\mathbf{Y}_k = [Y_{1k} \ Y_{2k} \ \ Y_{Kk}]^T$. One possible way to choose the basis vectors is to use singular value decomposition on the antenna output matrix $\mathbf{X}$ as

$$\mathbf{X} = \mathbf{U}\mathbf{\Sigma}\mathbf{V}^H, \tag{17}$$

where

$$\mathbf{U} = [\mathbf{u}_1 \ \ \ldots \mathbf{u}_l \ \ \ldots \mathbf{u}_L] \tag{18}$$

$$\mathbf{V} = [\mathbf{v}_1 \ \ \ldots \mathbf{v}_l \ \ \ldots \mathbf{v}_L] \tag{19}$$

$$\mathbf{\Sigma} = \begin{bmatrix} \sigma_1 & 0 & \cdots & 0 \\ 0 & \sigma_2 & \cdots & 0 \\ \vdots & \vdots & \ddots & \vdots \\ 0 & 0 & \cdots & \sigma_L \end{bmatrix}. \tag{20}$$

[15] Although there are L singular values, only K of those are effective (or dominant) due to the fact that the rank of matrix $\mathbf{X}$ is K. Thus the basis vectors are chosen as the effective first K column vectors of $\mathbf{U}$ as

$$[\mathbf{b}_1 \ \ldots \ \mathbf{b}_k \ \ldots \ \mathbf{b}_K] = [\mathbf{u}_1 \ \ \ldots \mathbf{u}_k \ \ \ldots \mathbf{u}_K]. \tag{21}$$

Then the effective singular value matrix $\mathbf{\Sigma}_{\boldsymbol{eff}}$ and effective matrix $\mathbf{V}$, $\mathbf{V}_{\mathbf{eff}}$ are used to form the coefficient matrix in equation (15) as

$$\mathbf{X} = [\mathbf{b}_1 \ldots \mathbf{b}_k \ldots \mathbf{b}_K][\mathbf{X}_1 \ldots \mathbf{X}_k \ldots \mathbf{X}_K]^T \tag{22}$$

and

$$[\mathbf{X}_1 \ldots \mathbf{X}_k \ldots \mathbf{X}_K]^T = \underbrace{\begin{bmatrix} \sigma_1 & 0 & \cdots & 0 \\ 0 & \sigma_2 & \cdots & 0 \\ \vdots & \vdots & \ddots & \vdots \\ 0 & 0 & \cdots & \sigma_K \end{bmatrix}}_{\Sigma_{eff}} \mathbf{V}_{\mathbf{eff}}{}^H. \tag{23}$$

Once $[\mathbf{X}_1 \ \ldots \ \mathbf{X}_k \ \ldots \ \mathbf{X}_K]$ is determined, $\mathbf{X}$ can be obtained from equations (21) and (22).

[16] If the definitions in equations (15) and (16) are inserted back into equation (11), we obtain

$$\begin{aligned}&[\mathbf{b}_1 \ldots \mathbf{b}_k \ldots \mathbf{b}_K][\mathbf{X}_1 \ldots \mathbf{X}_k \ldots \mathbf{X}_K]^T \\ &\quad = [\mathbf{b}_1 \ldots \mathbf{b}_k \ldots \mathbf{b}_K][\mathbf{Y}_1 \ldots \mathbf{Y}_k \ldots \mathbf{Y}_K]^T \mathbf{A}^T. \end{aligned} \quad (24)$$

Since $\{\mathbf{b}_1 \ldots \mathbf{b}_k \ldots \mathbf{b}_K\}$ is a basis, equation (24) implies the following equality:

$$\underbrace{\begin{bmatrix} \mathbf{A} & \mathbf{0} & \cdots & \mathbf{0} \\ \mathbf{0} & \mathbf{A} & \cdots & \mathbf{0} \\ \vdots & \vdots & \vdots & \vdots \\ \mathbf{0} & \mathbf{0} & \cdots & \mathbf{A} \end{bmatrix}}_{\mathbf{A_g}} \underbrace{\begin{bmatrix} \mathbf{Y}_1 \\ \mathbf{Y}_2 \\ \vdots \\ \mathbf{Y}_K \end{bmatrix}}_{\mathbf{y_g}} = \underbrace{\begin{bmatrix} \mathbf{X}_1 \\ \mathbf{X}_2 \\ \vdots \\ \mathbf{X}_K \end{bmatrix}}_{\mathbf{x_g}}. \quad (25)$$

The above equation defines a linear system where $\mathbf{y_g}$ has to be determined. A solution can be obtained when the column vector $\mathbf{x_g}$ is in the subspace formed by the column vectors of the matrix $\mathbf{A_g}$. Because of noise, $\mathbf{x_g}$ has components outside the signal subspace, therefore, an exact solution may not exist. Thus, rather than an exact solution, the minimizer of the following cost function can be used:

$$J\left(\mathbf{a}_1; \ldots; \mathbf{a}_K; \mathbf{y_g}\right) = \| \mathbf{A_g y_g} - \mathbf{x_g} \|^2, \quad (26)$$

where $\| \, . \, \|$ denotes L_2 norm. Using equation (25), the above cost function can be written as the sum of individual cost functions as

$$J\left(\mathbf{a}_1; \ldots; \mathbf{a}_K; \mathbf{y_g}\right) = \sum_{k=1}^{K} J_k = \sum_{k=1}^{K} \| \mathbf{A Y}_k - \mathbf{X}_k \|^2 . \quad (27)$$

We investigate the values $\mathbf{a}_k$ and $\mathbf{y_g}$ which will minimize J. Because of the orthogonality property of the least squares cost function, the individual J_k's are minimized when the projection of $\mathbf{X}_k$'s onto the range space of $\mathbf{A}$ are maximized. The projections are defined as

$$\mathbf{P}_k(\mathbf{a}_k) = \mathbf{A}\left(\mathbf{A}^H \mathbf{A}\right)^{-1} \mathbf{A}^H \mathbf{X}_k, \quad (28)$$

where the superscript H denotes the Hermitian operator and $1 \leq k \leq K$. Therefore, the optimal solution can be obtained as the maximizer of the following function $\mathcal{M}$:

$$\mathcal{M}(\mathbf{a}_1; \ldots; \mathbf{a}_K) = \sum_{k=1}^{K} \| \mathbf{P}_k \|^2 . \quad (29)$$

Once the arrival directions are estimated as the maximizer of $\mathcal{M}$, then $\mathbf{Y}_k$'s can be obtained as

$$\begin{aligned}\mathbf{Y}_k = &\left(\mathbf{A}^H(\tilde{\mathbf{a}}_1; \ldots; \tilde{\mathbf{a}}_K)\mathbf{A}(\tilde{\mathbf{a}}_1; \ldots; \tilde{\mathbf{a}}_K)\right)^{-1} \\ &\cdot \mathbf{A}^H(\tilde{\mathbf{a}}_1; \ldots; \tilde{\mathbf{a}}_K)\mathbf{X}_k. \end{aligned} \quad (30)$$

[17] The computed $\mathbf{Y}_k$'s, the output signals of the reference antenna for the kth mode, are then inserted into the equation (16) to obtain $\mathbf{Y}$. Thus, with MS-DOA algorithm, not only the arrival angles of the incoming signals are estimated but also the incoming signals themselves at the output of the reference antennas are determined. As shown in Figure 4, the impulse response of the channel, $h_k\,(t; \tau)$ and the initial transmitted signal $m(t)$ can be estimated by further processing algorithms such as those proposed by *Miled and Arikan* [2000].

[18] The proposed MS-DOA algorithm can be categorized in the deterministic blind signal separation. Thus, its performance is closely related to the assumed parametric channel model described in equations (1)–(9). Since the assumed channel model is a realistic one, such a deterministic approach has been preferred over the class of stochastic techniques such as MUSIC. MS-DOA algorithm assumes the slow variation of channel and its parameters within a block of data which usually chosen shorter than the stationarity period of ionospheric channel. Thus if the channel and its parameters change significantly within the time duration of a block of data, then the MS-DOA algorithm will fail like all other approaches mentioned in the Introduction section.

[19] Unlike the MS-DOA algorithm, the MUSIC algorithm operates on the estimated correlation matrix of the received source signals of equation (2), which is given by

$$\mathbf{R_x} = \mathcal{E}\left\{\mathbf{x}_n \cdot \mathbf{x}_n^H\right\}, \quad (31)$$

where $\mathcal{E}$ is the expectation operator; superscript H denotes the Hermitian; and

$$\mathbf{x}_n = [\mathbf{x}_1(n) \; \ldots \; \mathbf{x}_l(n) \; \ldots \; \mathbf{x}_L(n)]^T \quad (32)$$

is the received antenna outputs at time sample n. Note that, in practice, the expectation operator is replaced by time averaging on a recent block of array outputs. Therefore, in MUSIC, the signal and noise subspaces are separated based on their differences in the power spectral domain. Such a treatment provides robust estimates to the direction of arrivals of impinging signals when the SNR is above a certain threshold and the impinging waveforms are not strongly correlated [*Godara*, 1997]. A more detailed comparison between the statistical and deterministic array processing approaches is presented by *Van Der Veen* [1998] and *Swindlehurst et al.*, [1997].

[20] In the above derivation of the MS-DOA algorithm, it is assumed that the singular values corresponding to the signal subspace are dominant (equation (23)). When some of the source signals are relatively weaker with respect to noise, and/or angle of arrivals of some of the source signals are closer than the resolution power of the sensor array, the number of dominant singular values may be identified to be less than the number of source

signals. In such cases, the MS-DOA algorithm (like all other previously mentioned methods) provides reliable estimates only for the angle of arrivals of stronger source signals. In the next section, the performance of MS-DOA and MUSIC algorithms will be compared for the test scenarios.

4. Results

[21] In this section, the performance of the developed algorithm, MS-DOA, is tested for various signal-to-noise ratios (SNRs), angle and signal estimation errors for typical antenna array configurations. As discussed in detail in the Introduction, the RAKE algorithm is not widely used in practice due to its limitations in computational complexity and high demands on memory. The method of separation of signals using their orthogonal polarization properties has limited application area. These algorithms produce highly erroneous results when the polarizations of the incoming waves are quite linear or when the number of signals incoming from the ionospheric paths increase [*Compton*, 1981; *Erhel et al.*, 1994; *Afraimovich et al.*, 1999]. As discussed previously in this paper, MUSIC is the most commonly used DF method in the literature for high resolution spatial analysis due to its ease in implementation [*Rogier et al.*, 1991]. In spite of its drawbacks for HF DF estimation, it forms a basis of comparison with other methods. Thus, the performance of the new method is compared with MUSIC out of all the other possible DF methods mentioned in the Introduction. In order to compare the performance of MS-DOA and MUSIC for the same scenarios (the same simulated signal and the minimal antenna configuration), plain MUSIC (without any preprocessing/spatial smoothing) algorithm is chosen. As mentioned in the previous sections, with preprocessing/spatial smoothing, the performance of the MUSIC algorithm improves significantly. Yet, the number of antennas required for these improved MUSIC algorithms are twice as many as those required by plain MUSIC. According to the results obtained in this section, it is observed that the developed algorithm (MS-DOA) is highly advantageous in estimation of arrival directions for low SNR and small angle separation situations when compared with plain MUSIC.

[22] The synthetic signals that are used as input to the DF algorithms are generated from the simulation model provided in section 2. As shown in Figure 2, the transmitted signal $m(t)$ is received from a total of K paths. According to the equations (1), (2), and (4), the simulation model, includes the ionospheric multipath, free space propagation and receiving antenna effects. The HF scenarios that are realized by the simulation program are set based on the HF DF experiments by *Edjeou et al.* [1993a, 1993b], *Rogier et al.* [1991], and *Bertel et al.* [1989]. Thus, various parameters that are required by the simulation program (appearing in the transmitted signal, HF channel model, free-space propagation path and the receiving array) are set by making use of these real life scenarios. The noise used in the simulations is modeled as Additive White Gaussion Noise which is commonly used in various communication scenarios although it might not be the best model for HF noise. For experimental setups, and for other possible HF noise models, a prewhitening filter would guarantee the best performance of MS-DOA algorithm.

[23] In the simulations, the input signal $m(t)$, is chosen to be a bit sequence, generated at 2400 baud. In order to obtain synthetic HF signals, the value of the group delay, τ_k, in equation (8) is based on the information provided by *ITU* [1998] and *ITU-R* [1992] for good ionospheric channel model. The other HF channel model parameters in equations (8) and (9) are set as suggested by *Bertel et al.* [1996] and *Edjeou et al.* [1993b]. Namely, in equation (9), Δf_{ok} is set to 0.1 Hz, $a = 1$, $T_o = 15$ minutes and ψ_k is set to 0 for all modes. The values for the other parameters of the simulation scenarios such as the attenuation factor, the polarization, and the arrival angles of the modes in equations (2), (6), (7) and (8) are obtained from the experimental results of *Bertel et al.* [1996], *Edjeou et al.* [1993a, 1993b], *Rogier et al.* [1991], and *Bertel et al.* [1989].

[24] In the simulation model, the receiving array consists of L identical antennas and the sensors are arranged in a coordinate system as shown in Figure 1. The reference antenna is placed at at $\mathbf{r}_1$ which is the origin of this coordinate system. The circular crossloops are used as sensors in the arrays The effective length of the crossloop antenna is obtained from *Lemur et al.* [1997] for the passive sensors. As described in equation (1), the incoming electric field is projected onto the effective length of the antennas to obtain the open circuit voltage of each sensor. Finally, the received signals at the output of antennas, $x_l(t)$, are demodulated to baseband and sampled at a rate of 9600 samples/sec.

[25] Due to space and time variations of ionospheric layers, modes can arrive to the receiving array with a wide range of angle separation. The difference between the arrival angles of O and X modes can be as low as a couple of degrees and modes arriving from various layers or from different origins can be separated by tens of degrees. Differentiating closer modes is a challenge when the angle separation and SNR are low due to the correlation of modes from the same origin. Thus, the first performance criterion for a DOA algorithm is how well it can differentiate the incoming signals for various SNRs. The separation of angles of the incoming signals, whether they are correlated or not, number and configuration of the receiving antennas and SNR are some of the factors that are effective in the performance of the DOA algorithms.

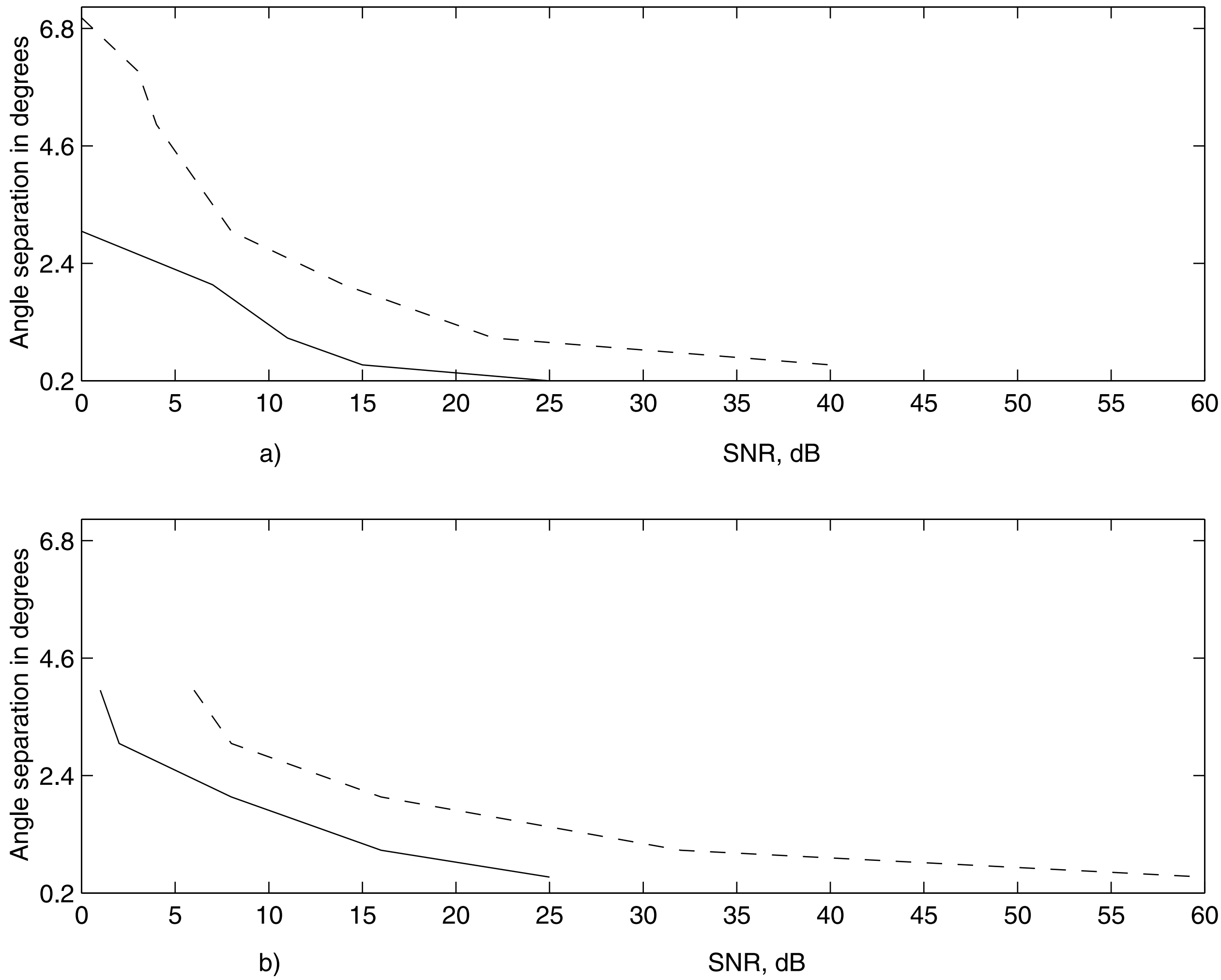


Figure 5. Angle separation in elevation versus SNR for MS-DOA (solid line) and MUSIC (dashed line) (a) for a 1×3 linear array and (b) for a 2×2 planar array.

In order to test the performance of the DOA algorithm, the worst possible case for correlated signals is realized with a test scenario where two synthetic signals are generated from the same original signal corresponding to O and X modes with equal amplitudes. The azimuthal angles of these modes are taken equal to each other and signals are allowed to vary only in elevation. Since there are only two incoming signals impinging on the array varying only in elevation, to minimize the effect of antenna number and configuration, we set up a linear array made up of three antennas separated by 0.5 λ. For homogeneous arrays, the minimum number of antennas required by both MS-DOA and plain MUSIC is one more than the number of incoming signals. For each SNR, we checked the lowest angle separation in degrees that the DOA algorithm can successfully differentiate. The results are presented in Figure 5a. For this set-up, MS-DOA (indicated by solid line) can successfully separate signals for very low SNRs. For example, at 0 dB SNR, MS-DOA can separate signals with only 3 degrees of difference in elevation whereas MUSIC can differentiate signals with 7 degrees of separation. As SNR increases to 25 dB, MS-DOA can determine the arrival angles when the separation is as low as 0.2°. Even at 40 dB SNR, MUSIC can only separate signals with 0.5 degrees of difference in elevation. For 1 degree of separation, MS-DOA provides about 12 dB improvement over MUSIC.

[26] In order to observe the variations in the performance of MS-DOA and MUSIC algorithms when the incoming signals and antenna configuration are changed, we realized another test scenario using the parameters provided in Table 1. The azimuthal angles are kept

Table 1. Some of the Parameters Used in the Simulation Scenario to Observe Normalized Standard Deviation of Error in θ, ϕ, $\mathbf{y}_k$

Mode	Path, k	Frequency	θ_k	ϕ_k	Group Delay, τ_k	Virtual Height
X	1	6.175 MHz	32°	122°	0 ms	209.2 km
O	2	6.175 MHz	34°	123°	0.2 ms	235.6 km

constant for the two incoming modes and the elevation angles are allowed to vary for each SNR as discussed in the previous paragraph. To receive the signals that are incoming with two different azimuthal angles, the array configuration is enlarged to an 2×2 planar array whose sensors are separated by 0.5 λ. Again for each SNR, we checked the lowest angle separation in degrees in elevation only that the DOA algorithm can successfully differentiate. In Figure 5b, with planar array, it is observed that the MS-DOA algorithm can separate signals with 0.5 degrees difference in elevation angles for 25 dB SNR and for the same situation, MUSIC requires 59 dB for a successful separation. Thus, the MS-DOA can differentiate signals significantly better than plain MUSIC for situations where the signals have high correlation, low separation in angle and low SNRs.

[27] The second important criterion in the performance of the DOA algorithms is how well they can estimate the arrival angles. Thus, in order to observe the amount of error both in estimation of arrival angles and in the estimation of the received signals at the output of the reference antenna, the sensor outputs, that are obtained for a simulated test scenario, are processed by both the MS-DOA and MUSIC algorithms. Then, the errors in the arrival angles and in the input sequence are computed for both DF algorithms. For statistically sound observation of error, the experiment has to be repeated N times and the normalized standard deviation of errors should be calculated for various SNRs. For N runs, the error between the desired angle ($\theta_{k,d}$) and estimated angle ($\theta'_{k,n}$) in elevation for the kth path and for each n (where $n = 1, \ldots, N$) can be given as $e_{\theta_k,n} = \theta_{k,d} - \theta'_{k,n}$. The normalized standard deviation of $e_{\theta_k,n}$ is obtained by

$$\sigma(\theta k) = \frac{1}{\sqrt{N}} \left[\sum n = 1^N \left(e_{\theta_k,n} - m_{\theta_k} \right)^2 \right]^{1/2}, \qquad (33)$$

where m_{θ_k} is the mean of the estimation error, i.e., $m_{\theta_k} = \left(\sum n = 1^N e_{\theta_k,n} \right)/N$. For the azimuth, the error between the desired and estimated angles for N runs is defined by $e_{\phi_k,n} = \phi_{k,d} - \phi'_{k,n}$ and the normalized standard deviation is given by

$$\sigma(\phi k) = \frac{1}{\sqrt{N}} \left[\sum n = 1^N \left(e_{\phi_k,n} - m_{\phi_k} \right)^2 \right]^{1/2}, \qquad (34)$$

where $m_{\phi_k} = \left(\sum n = 1^N e_{\phi_k,n} \right)/N$. The normalized standard deviation of estimation errors of incoming signals at the output of the reference antenna versus various SNRs can be computed by defining the error between the desired signal sequence $\mathbf{y}_{k,d}$ in equation (10) as $\mathbf{e}_{\mathbf{y}_k,n} = \mathbf{y}_{k,d} - \mathbf{y}'_{k,n} \cdot \mathbf{y}'_{k,n}$ is the estimated sequence in nth run for the kth path. The normalized standard deviation is given by

$$\sigma(\mathbf{yk}) = \frac{1}{\| \mathbf{y}_k \|} \frac{1}{\sqrt{N}} \left[\sum n = 1^N \| \mathbf{e}_{\mathbf{y}_k,n} - \mathbf{m}_{\mathbf{y}_k} \|^2 \right]^{1/2}, \qquad (35)$$

where $\mathbf{m}_{\mathbf{y}_k} = \left(\sum n = 1^N \mathbf{e}_{\mathbf{y}_k,n} \right)/N$ and $\| \, . \, \|$ denotes L_2 norm.

[28] In order to observe the normalized standard deviations of the errors in the estimation of arrival angles and in the estimation of reference signals, a simulated scenario is generated on the computer using the parameters provided in the beginning of this section and also those in Table 1. Again, for maximum correlation between the incoming signals, the two incoming signals are chosen to be O and X modes that are originated from the same source. In this scenario, there is a 2° difference in elevation and 1° difference in azimuth between paths 1 and 2 as given in Table 1. In order to maximize the selectivity both in elevation and in azimuth, the receiving antennas are arranged as a V shaped array on the x-y plane as given in Figure 1. The antenna locations are [(0, 0); (0.5λ, 0.5λ); (λ, λ); (0.5λ, 1.5λ); (0, 2λ)]. For various SNRs, for this scenario, the elevation and azimuth angles are estimated with both MS-DOA and MUSIC algorithms N = 20 times. The normalized standard deviations of arrival angle errors both in azimuth and elevation are provided in Figures 6a to 6d. In estimation of all four angles, (θ_1; ϕ_1 and θ_2; ϕ_2), MUSIC algorithm could not estimate the values when SNR is less than 10 dB for the elevation and 15 dB for the azimuth. Thus, the results for the MUSIC algorithm(denoted by the dashed line) in Figure 6 do not have the same range in SNR as MS-DOA. As shown in Figure 6, the MS-DOA can successfully estimate all four angles with SNRs as low as couple of dBs. As SNR increases, the normalized standard deviation of all angle errors are drastically reduced to zero and thus the accuracy of estimation of directions of arrival is significantly improved with MS-DOA compared to MUSIC. The normalized standard deviation of estimation errors of incoming signals at the output of the reference antenna versus various SNRs are plotted in Figure 7. Since MUSIC algorithm can not estimate the incoming signals, only the results for MS-DOA are presented in Figure 7. As can be observed from Figure 7, the normalized standard deviations of the signal errors decrease with increasing SNR. Even with low

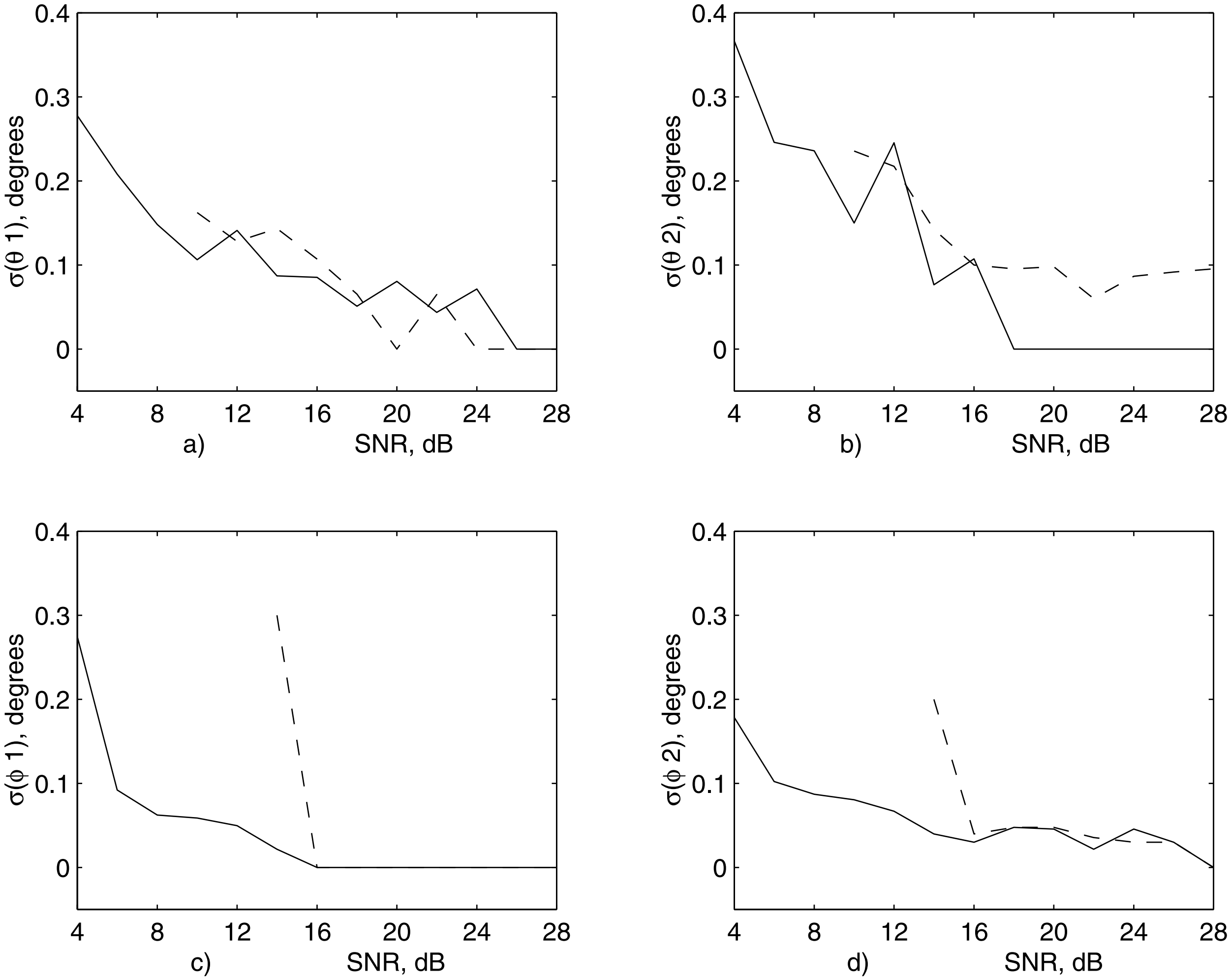


Figure 6. Normalized standard deviations for angle error in θ and ϕ versus SNR for two paths for MS-DOA (solid line) and MUSIC(dashed line) (a) $\sigma(\theta 1)$ versus SNR, (b) $\sigma(\theta 2)$ versus SNR, (c) $\sigma(\phi 1)$ versus SNR, and (d) $\sigma(\phi 2)$ versus SNR.

SNRs, MS-DOA provides a very accurate and reliable signal estimation.

[29] The number of antennas and how they are configured in the receiving array coordinate system are important parameters in the performance of the DF algorithms. As discussed in the previous section and by *Godara* [1997], for the basic performance of both MS-DOA and plain MUSIC algorithms, the number of antennas that are used in homogeneous arrays should be one more than the number of incoming signals. As the number of antennas increase, while the number of incoming signals is kept constant, the performance both DF algorithms in separation of signals and in arrival angle estimation improves and errors decrease. In the simulated scenarios, we chose the most basic array configurations to minimize the effect of the antenna arrangement. For real experimental scenarios, the optimum number of antennas and array configuration should be determined according to the requirements and statistics of the desired HF link. Even though, we have chosen the most basic configurations with minimum number of antennas, MS-DOA has outperformed plain MUSIC both in mode separation and estimation of arrival angles.

[30] As it is observed from the discussions and examples in this section, MUSIC can only separate modes and estimate arrival angles if the correlation between the modes is less; if the separation angle between the modes is large; and if the SNR is high. For low SNRs, the performance of plain MUSIC

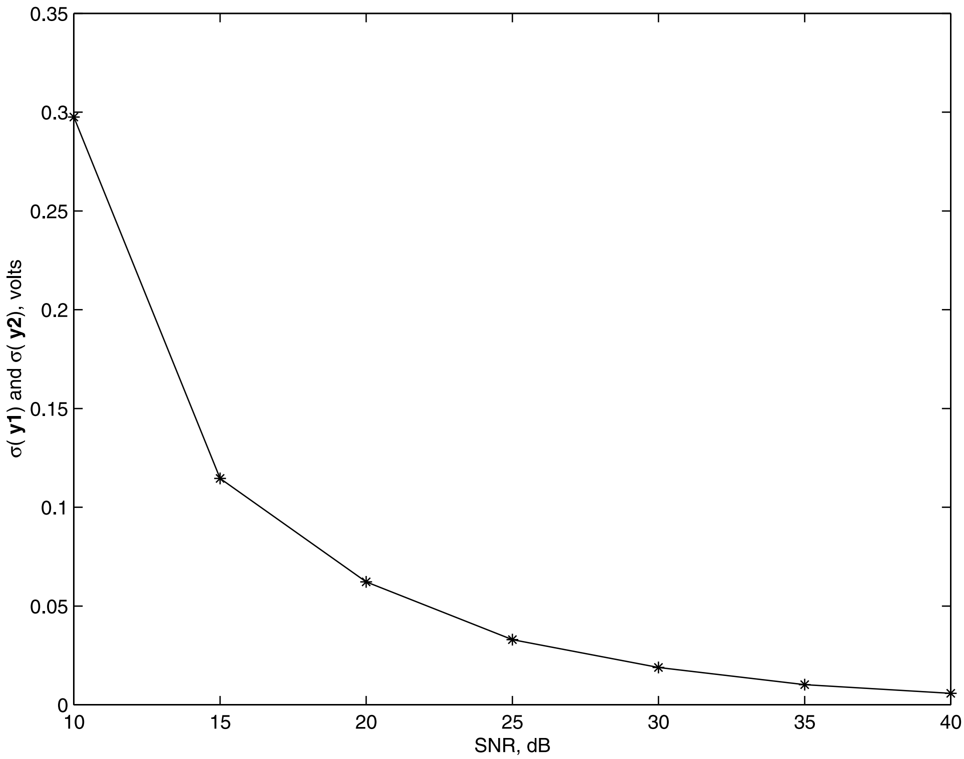


Figure 7. Normalized standard deviations for the incoming signal at the output of the reference antenna versus SNR for two paths for MS-DOA; σ(**y1**) versus SNR is denoted by a solid line, σ(**y2**) versus SNR is denoted by asterisks.

degrades for all scenarios. The performance of the proposed MS-DOA algorithm is closely related to the assumed parametric channel model. As long as the HF channel varies slowly and its parameters do not vary significantly within a block of data, even with minimum number of sensors and low SNRs, the MS-DOA algorithm outperforms plain MUSIC in mode separation and arrival angle estimation. With MS-DOA algorithm, the estimation of reference signals is also possible, a property which is not available in MUSIC algorithm.

[31] The estimation of arrival angles in MS-DOA algorithm is performed by a search for the angles which are the minimizers of the cost function J as given in equations (26) to (29). The search takes only a few minutes on a standard PC for a limited angle range and for a few incoming signals. If the angle range is enlarged and number of incoming signals increases, the MUSIC algorithm can not separate signals but it can be used as a first stage to roughly narrow down the regions of interest. Then, in those regions, MS-DOA will successfully estimate the arrival angles with minimum computational time. The structure of array manifold given in equations (11) and (25) is a function of array configuration and number of incoming signals. Thus, for a predetermined array, it is possible to solve equation (25) in closed form. In this manner, there is no need to search for the arrival angles which are minimizers of the cost function. An example of this closed form solution is provided by *Yilmaz* [2000] for an 2×2 planar array and two incoming signals.

5. Conclusions

[32] In this paper, a new algorithm, Multipath Separation-Direction of Arrival(MS-DOA), is developed to estimate arrival angles in elevation and azimuth for signals incoming from various ionospheric paths. The signals at the output of the reference antenna can also be identified with high accuracy. This method forms a basis for further estimation of HF channel and input signal. In MS-DOA, both the array output vector and incoming signal vector are expanded in terms of a basis vector set. A linear equation is formed using the coefficients of the basis vector for the array output vector and the incoming signal vector and the array

manifold. The angles of arrival in elevation and azimuth which maximize the sum of the magnitude squares of the projection of the signal coefficients on the range space of the array manifold are the required separation angles. Once the array manifold is estimated then the incoming signals can also be determined using the basis vectors and signal coefficients. The search for maximizing angles can be eliminated by solving the above mentioned system in closed form. The performance of the MS-DOA is a function of the array configuration and number of antennas in the receiving array. For homogeneous arrays, the minimum number of antennas that are required by the algorithm is only one more than the number of incoming signals. As the number of receiving sensors increase the performance of the MS-DOA improves. The optimum array configuration should be determined according to the statistical structure of the desired HF link. In this paper, the performance of MS-DOA is compared with plain MUSIC, for test scenarios which are formed with for minimum array configurations and number of sensors. According to our results, MS-DOA provides very accurate estimates of arrival angles both in elevation and azimuth even at low SNRs and small angle separations. With MS-DOA, it is also possible to estimate the incoming signals at the output of the reference antenna successfully. This feature is not available in MUSIC. Thus, MS-DOA provides significant improvement over MUSIC algorithm. In a narrowed down region of interest, the computational search time for MS-DOA is comparable to that of MUSIC for a few incoming signals. Such a search takes only couple of minutes in a standard PC and even that can be eliminated by solving for the arrival angles in closed form.

References

Afraimovich, E. L., V. V. Chernukhov, V. A. Kobzar, and K. S. Palamartchouk, Determining polarization parameters and angles of arrival of HF radio signals using three mutually orthogonal antennas, *Radio Sci.*, *34*(5), 1217–1225, 1999.

Bertel, L., J. Rojas-Varela, D. Cole, and P. Gourvez, Polarization and ground effects on HF receiving antenna patterns, *Ann. Telecommun.*, *44*, 345–440, 1989.

Bertel, L., P. Parion, and D. Lemur, Model of narrowband signal used in ionospheric high frequency (3–30 MHz) channel, in *Journees d'etudes SEE 96: Communications Numeriques en Presence de Multi-Trajets*, SEE Club 29, Paris, France, March 1996.

Buckley, K. M., and X. L. Xu, Spatial spectrum estimation in a location sector, *IEEE Trans. Acoust. Speech Signal Process.*, *38*, 1842–1852, 1990.

Chenu-Tournier, M., P. Larzabal, J. P. Barbot, and J. Grouffaud, Blind, high-resolution, space-time separation of multipaths in an ionospheric propagation, *Radio Sci.*, *35*(1), 235–246, 2000.

Collin, R. E., *Antennas and Radiowave Propagation*, McGraw-Hill, New York, 1985.

Compton, R. T., Jr., On the performance of a polarization sensitive adaptive array, *IEEE Trans. Antennas Propag.*, *29*(5), 718–725, 1981.

Edjeou, A., D. Lemur, and L. Bertel, Design and realization of a goniopolarimeter: Application to tilts study, paper presented at Second South Pacific Set Workshop, Univ. of Newcastle, Newcastle, N. S. W., Australia, July 1993a.

Edjeou, A., L. Bertel, and V. Massot, Identification des modes et trajets de propagation sur des liasons HF a l'aide d'un goniopolarimetre, paper presented at 53rd AGARD-EPP Symposium and Panel Meetings, Adv. Group for Aerosp. Res. and Dev., Rotterdam, Netherlands, 4–8 Oct. 1993b.

Erhel, Y., A. Edjeou, and L. Bertel, Contribution of the polarization diversity in HF direction finding systems, paper presented at International Symposium, Soc. of Photo-Opt. Instrum. Eng., San Diego, Calif., 24–29 July 1994.

Godora, J., Application of antenna arrays to mobile communications, part II, Beam forming and direction of arrival considerations, *Proc. IEEE*, *85*, 1193–1245, 1997.

Goodman, J. M., *HF Communications-Science and Technology*, Van Nostrand Reinhold, New York, 1992.

International Telecommunications Union (ITU), Wideband high frequency channel simulation system, *ITU Doc. 9C Temp/4E*, Geneva, 1998.

International Telecommunications Union-Radiocommunications (ITU-R), CCIR recommendation 520-2: Use of high frequency ionospheric channel simulators, recommendations and reports, Int. Radio Consult. Comm., Geneva, 1992.

Lemur, D., A. Edjeou, and L. Bertel, HF active loop antenna response, paper presented at Tenth International Conference on Antennas and Propagation (ICAP'97), Inst. of Electr. Eng., Edinburgh, UK, 14–17 April 1997.

Miled, M. K. B., and O. Arikan, Input sequence estimation and blind channel identification in HF communication, paper presented at International Conference on Acoustics, Speech and Signal Processing (ICASSP'2000), Inst. of Electr. and Electr. Eng., Istanbul, Turkey, 5–9 June 2000.

Pillau, S. U., *Array Signal Processing*, Springer-Verlag, New-York, 1989.

Proakis, J. G., *Digital Communications*, 3rd ed., McGraw-Hill, New York, 1995.

Rogier, J. L., G. Multedo, L. Bertel, and V. Baltazart, Ionospheric multipaths separation with a high resolution direction finding algorithms mapped on an experimental system, paper presented at Fifth International Conference, HF Radio Systems and Techniques, Inst. of Electr. Eng., Edinburgh, UK, July 1991.

Roy, R., and T. Kailath, ESPRIT-Estimation of signal parameters via rotational invariance techniques, *IEEE Trans. Acoust. Speech Signal Process.*, *37*, 984–995, 1989.

Schmidt, R. O., Multiple emitter location and signal parameter estimation, *IEEE Trans. Antennas Propag.*, *34*, 776–780, 1986.

Swindlehurst, A. L., M. J. Goris, and B. Ottersten, Some experiments with array data collected in actual urban and suburban environments, paper presented at Workshop on Signal Processing Advances in Wireless Communications, Inst. of Electr. and Electr. Eng., Paris, France, April 1997.

Van Der Veen, A., Algebraic methods for deterministic blind beamforming, *Proc. IEEE*, *86*(10), 1987–2008, 1998.

Watterson, C. C., J. R. Jurashek, and W. D. Bensema, Experimental confirmation of an HF channel model, *IEEE Trans. Commun.*, *18*, 792–803, 1970.

Williams, R. T., S. Prasad, A. K. Mahalanabis, and L. H. Sibul, An improved spatial smoothing technique for bearing estimation in multipath environment, *IEEE Trans. Acoust. Speech Signal Process.*, *36*, 425–432, 1988.

Yilmaz, N., A new algorithm for HF multipath separation and computation of arrival angles (in Turkish), M.S. thesis, Hacettepe Univ., Ankara, Turkey, 2000.

F. Arikan, Department of Electrical and Electronics Engineering, Hacettepe University, Beytepe, 06532 Ankara, Turkey. (arikan@hacettepe.edu.tr)

O. Arikan, Department of Electrical and Electronics Engineering, Bilkent University, Bilkent, 06533 Ankara, Turkey. (oarikan@ee.bilkent.edu.tr)

M. K. B. H. Miled, Department of Electrical and Computer Engineering, Northeastern University, 360 Huntington Ave., Boston, MA 02115, USA. (mkmiled@ece.neu.edu)

N. Yilmaz, Turkish Navy, NATO Snake-pit Headquarters, Elmadag, 06780 Ankara, Turkey. (nyilmaz@themail.com)