Estimating Angle of Arrival and Output Signals to Noise Ratio for Wideband Signals

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ABSTRACT

Multiple Signal Classification (MUSIC) algorithm is the most popular algorithm to estimate angle of arrival (AOA) for narrowband signals. This algorithm is based on the use of subspace noise Eigenvectors (Q_{n}) matrix related to the smallest set of Eigen values of received signal covariance matrix (ϕ_{xx}). These set of Eigen values are corresponding to thechannels thermal noise. It is found that when the received signal has none zero bandwidth the distribution of Eigen Values magnitudes is changed according to the value of received signals bandwidth. The Eigenvalues magnitudes which are relating to the received sources are spreading on expense of noise sub space Eigen values .This new distribution causes a change in the relating eigenvectors, and now the Eigenvalues corresponding to thermal noise is no longer exist .Since Music algorithm is based on the use of these eigenvectors relating to the thermal noise Eigenvalues, and all these thermal noise subspace Eigenvalues are occupied by wideband signals, Music algorithm is no longer work. In this paper we suggest a new approach to estimate angle of arrival (AOA) as well as output signal to noise ratio (SNR) for wide band signals by using linear adaptive array antenna with transversal filter as a wide band beam-former processor in conjunction with Linearly Constrained Minimum Variance beam-former algorithm (LCMV). A general mathematical model for M array sensors with J taps behind each sensor and with a delay time between taps equivalent to a phase shift (90^{o}) at center frequency is presented. The proposed approach is simulated and tested in different cases for different input levels and different received signals relative bandwidth (10%, 15%, 20% up to 40%) the result shows that with a proper number of taps behind each sensor the system is excellently Compensate the effect of signals bandwidth and estimate the angles of arrivals well the output SNR's of all received signals simultaneously with high accuracy. Under the same assumptions MUSIC technique is tested with transversal filter as a broadband processor, the results were very poor.

Keywords: Wideband signal, Angle of arrivals estimation (AOA), MUSIC algorithm, Adaptive array Antenna, Transversal Filter, LCMV algorithm

تخمين زاوية الوصول ومستوى نسبة الاشاره الى الضوضاء في الخارج للاشارات ذات الحزمة العريضة

الخلاصة

خوارزمية Multiple Signal Classification (MUSIC) هي الاكثر شهرة في تخمين زاوية وصول الاشارة(Angle of arrival) والتي تستند على استخدام متجهات الايجن (Eigen Vectors)الخاصة بضوضاء قنوات الاستلام لمصفوفة الهوائيات(Qi)والعائده الى قيم الايجن(Eigen Values)لمصفوفة مجموعة الاشارات المستلمة(φxx) .تبين ان توزيع قيم الايجن في محور المصفوفة(φxx) يعتمد على مقدار قدرة وعرض حزمة

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الاشاره الراديوية المستلمة. لاحظنا عند زيادة عرض حزمة الاشارة المستلمه فان ذالك سيؤدي الى ظهور قيم ايجن جديده على حساب قيم الايجن الخاصه بضوضاء قنوات الاستلام وهذا بدوره يؤدي الى ظهور قيم جديده لمتجهات الايجن والتي تعتبر الاساس في تشكل نموذج الاشعاع النهائي لمصفوفة الهوائيات.وحيث ان خوارزمية ميوزك تعتمد في عملها على تشكيل دالة رياضية اساسها متجهات الايجن لضوضاء القنوات لحساب زاوية الوصول للاشارات المستلمة فان تغير قيم متجهات الايجن يجعل خوارزمية ميوزك عاجزه عن قراءة الاتجاه للاشارات ذات الحزمة العريضة.

في بحثي هذا تم اقتراح طريقة جديدة لمعالجة ايجاد زاوية الوصول للاشارات العريضة الحزمة عن طريق استخدام مصفوفة هوائيات متكيفة وباستخدهم المرشحات المتعامده (Transversal filter) مع كل هوائي من هوائيات المنظومة للحصول على معالج اشارة عريض الحزمة.تم بناء نموذج رياضي ونموذج برمجي للنظام المقترح.عند فحص المنظومة في حا لات مختلفه ومتعددة للاشارات المستلمه وبعرض حزم مختلفة ومقارنتها مع اشارات الحزمة الضيقة تبين ان المنظومة اعطت نتائج رائعة واستطاعت التغلب على مشاكل عرض الحزمة وخمنت زوايا الوصول بدقة مساوية لحالات الاشارات ذات الحزمة الضيقة.تحت نفس الفرضيات تم فحص خوارزمية الميوزك بعد ربطها الى منظومة المرشحات المتعامده وكانت النتائج سيئه كون ان خوارزمية الميوزك تعتمد في اساس عملها على استخراج الاتجاهات من متجهات الايجن للضوضاء والتي تغيرت بشكل كلي مع زيادة عرض الحزمة التردديةللاشارات المستلمة.

INTRODUCTION

Adaptive array antennas are mainly used for minimizing the reception level of strong interference sources by introducing nulls toward them. From theoretical background, the adaptive array antenna can be modified in a way to estimate Angle of Arrival (AOA) of received signals. The problem of determining AOA has received considerable attention in array signal processing fields such as radar, sonar, and wireless communication system [1, 2]. Most of the high-resolution directions finding methods such as MUSIC [3], ESPRIT [4], and others [5, 6] have been studied in the context of 1-D and 2-D cases. Conventional implementations of these techniques are based on either Eigenvalue decomposition of the covariance matrix or singular value decomposition (SVD) of the array data matrix. These methods are computationally intensive and suitable for narrow band signals so that they cannot respond to a wideband signals even with wideband processors so that they might not suite for modern radar and communication techniques uses wideband techniques for high bit rate. Moreover they are also not suited for real-time applications where the required signals are to be tracked on-line. It is found that the presence of single complex adaptive weight in each element channel of array is sufficient for processing narrowband signals [7, 8, and 9], so for array antennas operating over a wideband, some sort of frequency dependent is required at each element channel. To process broadband signal transversal filter in conjunction with digital beam forming processor is required to employ in each channel in order to have channel transfer function H(jw) function to the input frequency [10, 11, and 12].

This paper proposes simultaneous AOA and output SNR estimation by using linear adaptive array antenna of M sensors and a transversal filter behind each element of J taps with a fixed delay time Δ unit in conjunction with a LCMV beam forming algorithm. The aim is to design an accurate and effective AOA and output SNR estimator for broadband signals. Compared with MUSIC method, the proposed method completely compensates the effect of signal bandwidth and gives response equivalent to the case of zero bandwidth without any change in the accuracy of estimation, while MUSIC method exhibits poor results even with the use of transversal filter.

Music Method

The MUSIC method is simple, popular, high resolution and efficient Eigen structure method for AOA estimation only. The MUSIC spatial spectrum can be expressed as [13]

$$DF(\theta) = \frac{1}{C^{T(\theta)Qn\,Qn^{H}}C^{*}(\theta)} \quad \ldots (1)$$

C (θ) is a spatial vector given by $C(\theta) = [g_z(\theta) * e^{-jz\beta d \sin(\theta)}]^T, z = 1, 2, \ldots M$

Where $g_z(\theta)$ is the element field pattern and θ is a spatial angle belonging to $[0,2\pi]$. Q_n is a thermal noise sub space eigen vector related to the set of equal and smallest eigen value of orthogonalized received signals covariance matrix (ϕ_{xx}).

Mathematical Formulation for Proposed Approach.

Figure(1) is a broadband beam former with(M)omnidirectional array sensors separated by distance (d) and (J) taps of (J-1) delays behind each sensor with

Constant time delay (Δ) seconds. The case with J=1 corresponds to a narrowband beam former. The element signal vector containing signals at mth element of J taps is

$$X_m = [x_{m,1}(t),\ x_{m,2}(t), \dots\dots\ x_{m,J}(t)\ ]^T \quad \dots (2)$$

The total signal vector for the whole array is

$$X = [X_1^T, X_2^T, \dots\dots, X_M^T\ ]^T \quad \dots (3)$$

The corresponding element and total weight vector are

$$W_m = [w_m,\ w_{m,2}, \dots\dots, w_{m,J}\ ]^T \quad \dots (4)$$

$$W = [W_1^T, W_2^T, \dots\dots,\ W_M^T\ ]^T \quad \dots (5)$$

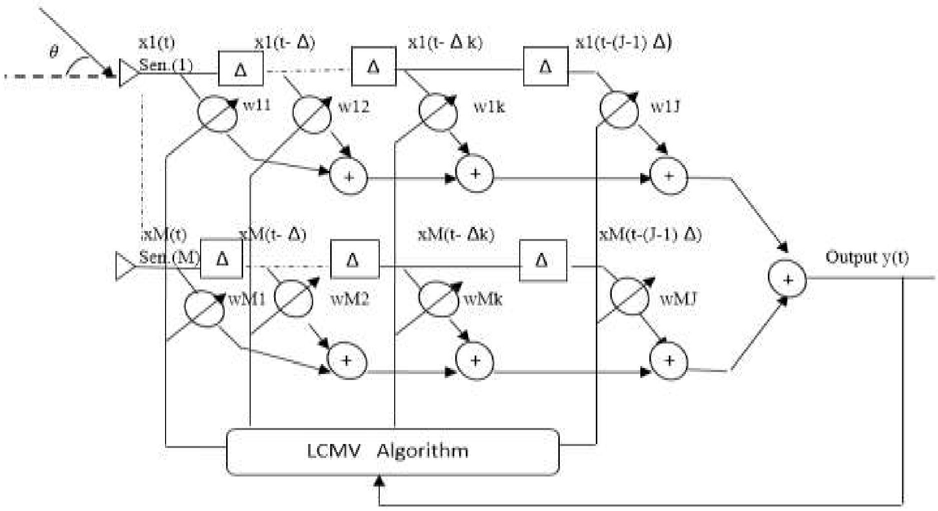

Figure (1) Wideband adaptive array AOA estimator

The LCMV beam former applies linear constrains on the beam former weights so as when the array spatial vector angle (θ) is coincide with received signal angle (θ_d), the received signal pass with specified gain according to the following.

The output power from system is

$$\{|y(Bw,\theta)|^2\} = W^{\dagger}\, \phi_{xx}\,(Bw,\theta)\, W \quad \dots (6)$$

Where †is a transpose conjugate, Bw is a relative bandwidth, and ϕ_{xx} is a covariance matrix of broad band received signals plus channels thermal noise power of $(MJ \times MJ)$ dimensions. The LCMV beam former [14] applies

$$\min_w W^{\dagger}\, \phi_{xx}\,(Bw,\theta)\, W \quad \dots (7)$$

Subjected to

$$C^{\dagger}\,(Bw,\theta)\, W = 1 \quad \dots (8)$$

C is a constraint spatial vector of dimension $(MJ \times 1)$ with angle span (0,2π), the constraint spatial vector at mth element of J taps is

$$C_m(Bw,\theta) = [c_{m,1}, c_{m,2}, \dots, c_{m,J}]^T \quad \dots (9)$$

The total spatial vector for the whole array is

$$C(Bw, \theta) = [C_1^T, C_2^T, \dots\dots\dots.., C_M^T]^T \qquad \dots (10)$$

The solution to (9) and (10) can be computed using Lagrange multiplier [14].

$$W_{opt} = \phi_{xx}^{-1} C (C^{\dagger} \phi_{xx}^{-1} C)^{-1} \qquad \dots (11)$$

The received signal at each sensor is assumed consisting two uncorrelated component: desired signal $S_d(t)$ under estimation process from unknown angle θ_d and noise channel component $n(t)$. The signal correlation matrix ϕ_{xx} can be decomposed to desired signal ϕ_d and noise componentϕ_n, respectively

$$\phi_{xx} = \phi_d + \phi_n \qquad \dots (12)$$

if the desired signal is assumed to have flat power spectral density amplitude of $(2\pi p_d / \Delta\omega_d)$ over a limited bandwidth $\Delta\omega_d$centered at ω_o as shown in fig.(2),the correlation function$R_d(\tau)$ of the desired signal is then given by[15]

$$R_d(\tau) = P_d \, \text{sinc}(\frac{\Delta\omega_d \tau}{2}) e^{j\omega_o \tau} \qquad \dots (13)$$

Where τ is time delay. Let T_e be the unit propagation delay between adjacent array elements and T_o be the delay unit time between adjacent element taps, the correlation of the desired signal between pair of element signal vector S_{dm} and S_{dn} is then

$$\phi_{d(m,n)} = E[S_{dm}^* S_{dn}^T] \qquad \dots (14)$$

Where E [.] is the expected value

The ith rows and kth column of$\phi_{d(m,n)}$ is found by following equation

$$[\phi_{d\,(m,n)}]\, i, k = R_d[(m-n)T_e + (i-k)T_o] \qquad ... (15)$$

The inter-element delay time $T_e$is equal to

$$T_e = (d/c)\sin(\theta_d) \qquad \dots (16)$$

Where d is the interelement spacing and c is the wave propagation velocity

If we chose T_o equivalent to a time required by desired signal to cross a distance L equal to a quarter wave lengths at center frequency then we have

$$T_o = L/c = \lambda o / 4c \qquad \dots (17)$$

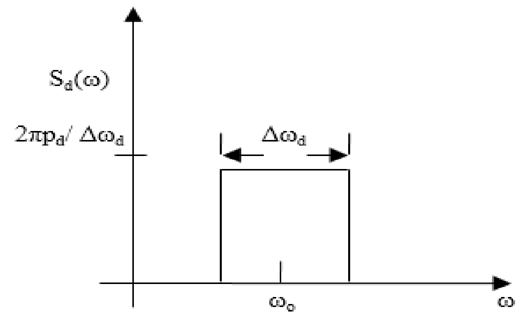

Figure (2) power spectrum density of wideband signal

Now the equivalent phase delay between signals at different taps due to delay time $T_o$at a center frequency is

$$\varphi_o = T_o \omega_o = \pi/2 \qquad \dots (18)$$

Substituting (13), (16), and (17) into (15) leads to $[\phi_{d\,(m,n)}] i, k = P_d \, \text{sinc}\left(\frac{\Delta\omega_d}{2}\{(m-n)T_e + (i-k)T_o\}\right) * e^{j\omega_o[(m-n)T_e + (i-k)T_o]}$

… (19)

The product of $\Delta\omega_d T_e$ can be normalized over a center frequency ω_o to have relative bandwidth rather than absolute as follows

$$\Delta\omega_d T_e = (\Delta\omega_d/\omega_o) * \omega_o T_e = Bw_d \varphi_e \quad \dots (20)$$

Where Bw_d is the desired signal relative bandwidth and φ_e is the inter element phase delay at center frequency, using Eq. (16), φ_e can be expressed as a function of inter element spacing and desired signal angle of arrival as

$$\varphi_e = \beta d \sin(\theta_d) = (2\pi/\lambda_o)\, d \sin(\theta_d) \quad \dots (21)$$

For $d = \lambda_o/2$ equation (21) becomes

$$\varphi_e = \pi \sin(\theta_d) \quad \dots (22)$$

By the same way

$$\Delta\omega_d T_o = (\Delta\omega_d/\omega o) * \omega o T_o = Bw_d \varphi_o \quad \dots (23)$$

Where φ_o is a phase delay due to tap time delay T_o. Using Eq. (20), Eq. (22), Eq. (23) and Eq. (18), then Eq. (19) becomes

$$[\phi_{d\,(m,n)}]i,k = P_d\, sinc\left(\frac{Bw_d}{2}\tau_d\right) * e^{j\tau_d} \quad \dots (24)$$

Where the desired delay time is equal to

$$\tau_d = [\pi(m-n)\sin(\theta_d) + (\pi/2)(i-k)] \quad \dots (25)$$

Where $m,n = 1,2,\dots.M\ sensors\ ;\ i,k = 1,2,\dots,J\ Taps$

The channels thermal noise components are assumed to be uncorrelated between channels and correlated for the same channel due to the existence of taps delay behind each sensor. If the noise component has a flat spectrum density amplitude $(2\pi\sigma^2/\Delta\omega_n)$ over noise band width Bw_n, then the noise correlation matrix can be derived using the same steps as for desired signal, only final equations will be listed as follows

$$[\phi_{n(m,n)}]i,k = \begin{cases} Rn[(i-k)T_o] \dots & for\ m = n \\ 0\ \dots\dots\dots\dots. & for\ m \neq n. \end{cases} \quad \dots (26)$$

$$[\phi_{n(m,m)}]i,k = \sigma^2\, sinc\left(\frac{\Delta\omega_n}{2}(i-k)T_o\right) * e^{j\omega_o}(i-k)T_o \quad \dots (27)$$

Where σ^2 is the variance of thermal noise component.

$$[\phi_{n(m,m)}]i,k = \sigma^2 sinc\left(\frac{Bw_n}{2}\tau_n\right) * e^{j\tau_n} \quad \dots (28)$$

τ_n is the noise delay time and equal to

$$\tau_n = [(\pi/2)\,(i-k)] \quad \dots (29)$$

Now all elements of ϕ_{xx} can be calculated and only one think is rest, the spatial vector ($\boldsymbol{C}$), by using the same steps, constraint spatial vector of dimension $(MJ \times 1)$ for whole array can be written as

$$[C_{m,1}]i,1 = sinc\left(\frac{\Delta\omega_c}{2}\{(m-1)T_e + (i-1)T_o\}\right) * e^{j\omega_o[(m-1)T_e+(i-1)T_o]} \quad \dots (30)$$

$$[C_{m,1}]i,1 = sinc\left(\frac{Bw_c}{2}\tau_c\right) * e^{j\tau_c} \quad \dots (31)$$

Where τ_c is a spatial vector delay time equal to

$$\tau_c = [\pi\,(m-1)\sin(\theta c) + (\pi/2)(i-1)] \quad \dots. (32)$$

Where $m,n = 1,2,\dots.M\ sensors\ ;\ i,k = 1,2,\dots,J\ Taps$

θ_c is a spatial angle belongs to $[0,2\pi]$ and Bw_c is the relative bandwidth of spatial vector, and its value is taken to be equal to the designed system bandwidth. Then W_{opt} given by Eq. (11) can now be found as a function of desired and noise

Band width as well as a function of desired signal angle of arrival. According to constraint given by Eq. (8) when a spatial vector angle θ_c is equal to the angle of i^{th} received signals (θ_{id}), the output of the system will be maximum from this directions which represents AOA estimation. With the optimized weight, the output powers due to desired signals and noise (P_d and P_n) can be found as a function of bandwidth and angle of arrival as follows

$$P_d(\theta_d, Bw_d) = W_{opt}{}^{\dagger}\phi_{d\ ((\theta_d, Bw_d))} W_{opt} \quad \ldots (33)$$
$$P_n(Bw_n) = \sigma^2 W_{opt}{}^{\dagger} W_{opt} \quad \ldots (34)$$

Finally the output signal to noise ratio (SNR) as a function of signal bandwidth and angle of arrival is equal to

$$SNR(BW_d, \theta_d) = \frac{P_d(Bw_d, \theta_d)}{P_n(\ Bw_n)} \quad \ldots (35)$$

Simulation Results

All simulation programs were written in MATLAB 7.10 and the following assumptions are considered:-

Omnidirectional linear array antenna of six elements uniformly distributed with $0.5\lambda_o$ inter element spacing .The received signals are considered to be statistically independent and uncorrelated with the channel thermal noise components. Three scenarios will be present to show the capability of proposed approach comparing with MUSIC technique.

A- Scenario1:- A single transmitter source emits from angle 60°.The received power from this source at antenna field is 0dB.

Figure (3) shows a MUSIC response for different signal relative bandwidth values. It can be seen that the result is excellent when the bandwidth is zero while for 0.4 relative bandwidth ,it is severely deteriorate even with the use of 3 and 6 tap delay unit behind each sensor (1 tap is a case of narrowband processor),this is because the increase in signal bandwidth cause spreading in the distribution of covariance matrix Eigenvalues and the subspace Eigenvalues set related to the thermal noise channels is no longer exist, this leads to a set of corresponding Eigenvector (Q_n) which are no longer orthogonal with eigenvector related to the received source, and then the real direction to the received source is lost.

Figure (4) shows the response of proposed system with and without tap delay units. It can be seen that for 0.4 relative band width with 1 tap (no delay unit attached)

AOA estimation is completely worsen but with 3 taps behind each array element the AOA estimation as well as output SNR are significantly improved but SNR level is still less than the level of zero bandwidth, while when 6 taps are used behind each array elements the processor exhibits a performance like the case of narrow band signal for both SNR and AOA, which means that the new broadband processor completely compensate the signal bandwidth effects.

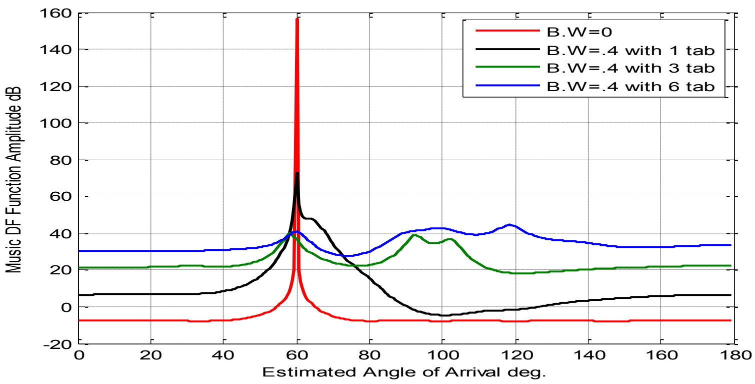

Figure (3): Estimated AOA by MUSIC algorithm

response which means that the wide band processor is succeeded in compensation the effects of bandwidth and perform its job like the case of narrow band signals.

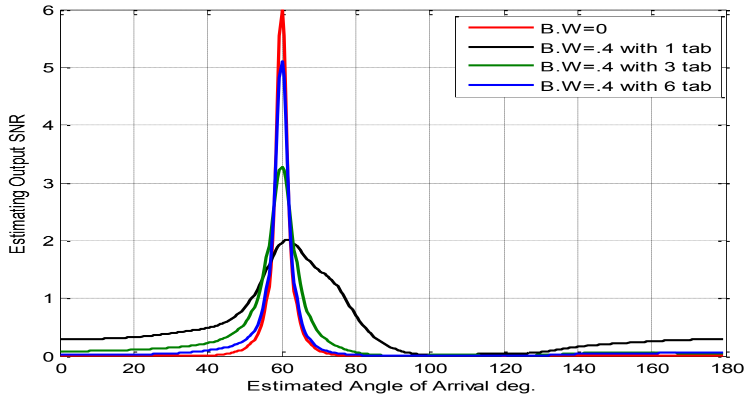

Figure (4) Estimated AOA &SNR by proposed system

B- Scenario2: Two transmitting sources emits from angles 60o and 120o with different transmitting power, the received powers are assumed to be 0dB form source 1 and 3dB form source 2 in order to test the output SNR response of the system for different input levels.

Figure (5) shows the MUSIC DF response for the case of narrowband (zero bandwidth) and (0.4) relative bandwidth for two sources. The response is excellent when the bandwidth is zero and severely worsens for 0.4 bandwidth even with use of 3 and 6 taps behind array elements that is because the subspace Eigenvalues related to thermal noise are used by adaptive processor to face the wideband sources by creating multi-nulls in front of them and no more Eigenvalues are available for thermal noise channel.

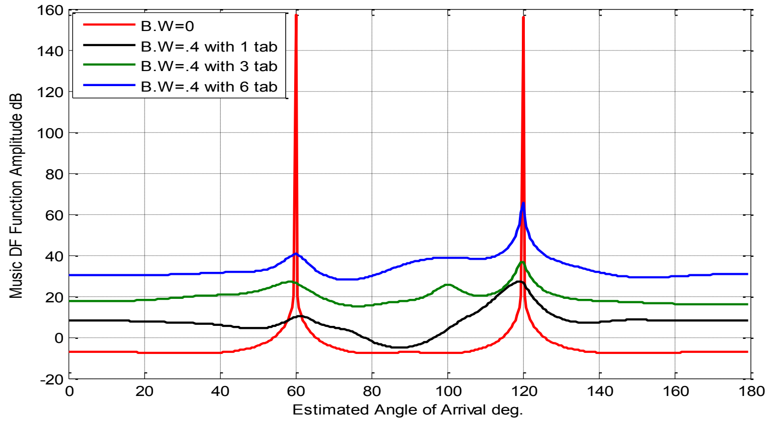

Figure (5): Estimated AOA by MUSIC Algorithm for two sources

Figure (6) shows the response of proposed system for the two transmitting sources with zero and 0.4 relative bandwidth in two cases when the adaptive processor with and without tap delay unit. It can be seen that the output SNR for zero bandwidth is proportional to the input power levels received from sources and also the estimated AOA's are correct and accurate. For the case when the relative bandwidth is increased to 0.4 there are no estimation results for both SNR's and AOA's when no delay units attached array sensors but when 3-taps are used behind each element the system response is significantly improved for the AOA estimation and partially improved for SNR and with 6-tap the system response reached the case of narrowband signals.

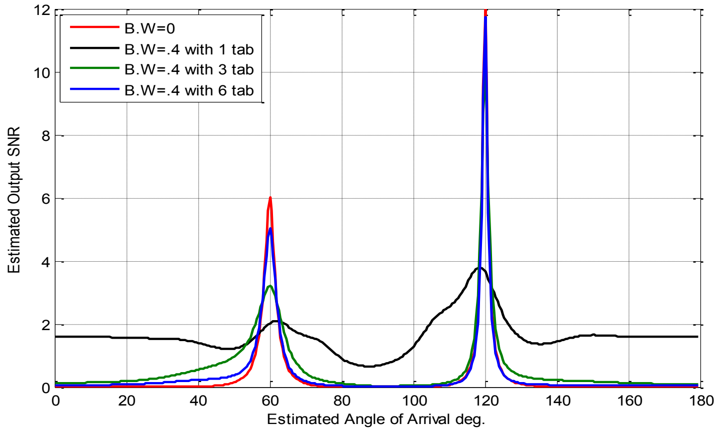

Figure (6): Estimated AOA&SNR for two sources by proposed system

C- Scenario3:- Three transmitting sources are at angles 40o, 70o and 150o with relative bandwidths 0.1, 0.2, and 0.15 and with received input powers 4, 6, and 8dB respectively.

Figures (7) shows that when the relative bandwidths of received signals are increased from zero to 0.1, 0.2 and 0.15 the estimated SNR's and AOA's are severely deteriorated because the narrowband processor cannot respond well to the wideband signals.

Figure (8) shows that the amount of improvement occurred increases with the increase in the number of taps behind array processor from 3 to 6. For six taps the output SNR's and AOA's for the three sources are equivalent to the case of zero bandwidth as shown in figure (7), which means that the proposed system is able to handle different input levels as well as different relative bandwidths.

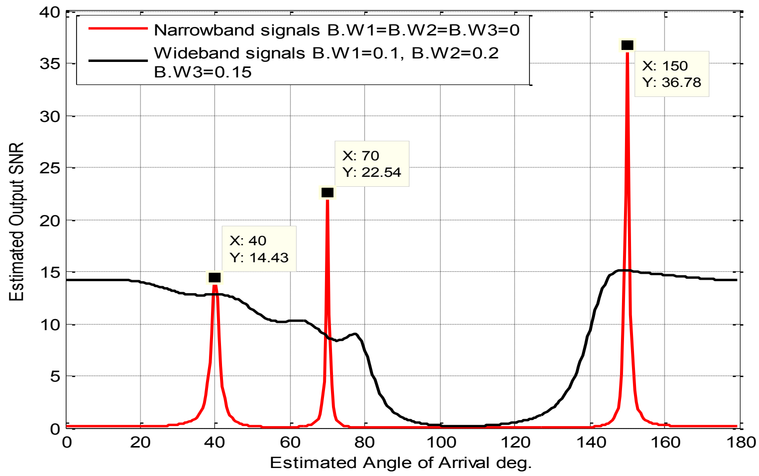

Figure (7): Estimated AOA&SNR by narrowband estimator

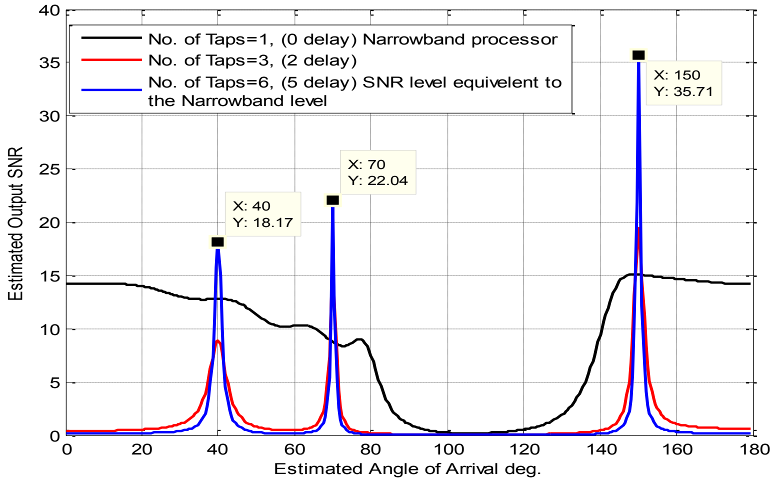

Figure (8): Estimated AOA&SNR by proposed wideband processor with different tap numbers.

CONCLUSIONS

A detailed analysis of the bandwidth performance of proposed system constructed from linear array antenna in conjunction with adaptive processor followed by tap delay units and governing by LCMV algorithm has been provided in terms of AOA and output SNR. Compared with MUSIC method , the tap delay unit-base beam-forming structure performs better in many aspects; It can simultaneously estimate both output SNR and AOA for received signals while MUSIC can only estimate AOA for narrowband signals, delay time units adaptive processor excellently compensate the bandwidth effect of received signals and exhibit accurate estimation for both output SNR and AOA while MUSIC even with use of delay units behind array element is failed to give any results since it is based on Eigen structure technique. The required numbers of delay time units behind array elements increase with increasing of signal bandwidth. Finally the proposed system in same time can handle multi sources with different bandwidths and transmitting powers.

REFERENCES

[1] H. L. Trees, Optimum array processing, Part IV: Detection, Estimation, and Modulation Theory. Wiley, New York, 2002.

[2] Xin and A. Sano, "Computationally efficient subspace-based method for direction-of-arrival estimation without Eigen decomposition" IEEE Trans. Signal Process., vol. 52, pp. 876-893, Apr. 2004.

[3] Elkamchouchi, H. and Mofeed, M. A. E. “Direction of arrival Methods and Time Difference of Arrival Position Location Technique” Proceeding of Twenty Second National Radio Science Conference IEEE, Pages: 173-1822, 2005.

[4] Naftali (Tuli) Herscovici, Christos Christodoulou"Uniform Circular Arrays for SmartAntennas” IEEE Antennas and Propagation Magazine, vol. 47, No.4, August 2005.

[5] GU. J., Stark and Yang Y., “Design of Tapped Delay Line Antenna Array Using Vector Space Projection”, Trans. Ant. and Propag. IEEE, No.12, pp. 4178-4182, Dec.2005

[6] Performance of Linearly Constrained Minimum Variance Beam-formers”,Proc. Of the 2007 15^{t} Intl. Conf. on Digital Signal Processing (DSP 2007)

[7] Lin N., Liu W. and Langley R., "Performance Analysis of a two-Element Linearly Constrained Minimum Variance Beam-former with Sensor Delay Line Processing", Proc. Of the 2007 15t Intl.

[8] Raed M. Shubair, Mahmoud A. Al- Qutaryri and Jassim M. Samhan " A Setup for the Evaluation of Music and LMS Algorithms for Smart Antenna System" Jornal of Communication ACADEMY Vol.2 Issue: 4 pages: 71-77 2007.

[9] Md. Bakhar, Dr. Vani. R.M and Dr. P.V. Hunagund "Eigen Structure Based Direction of Arrival Estimation Algorithms for Smart Antenna Systems" IJCSNS International Journal of Computer Science and Network Security, VOL.9 No.11, November 2009.

[10] Huilong Zhang, Xiurong Ma, Yunxiang Cheng and Yuan Bai "Study on Direction of Arrival Estimation Algorithm for Smart Antenna of Uniform Circular" International Conference on Computer Application and System Modeling (ICCASM 2010)) IEEE VOL.. 15 Pages: 627-631 2010.

[11] H. Z. X. Ma , Y. Cheng and Y. Bai"Study of DOA Estimation Algorithm for Smart Antenna of Uniform Circular" Int. Conf. on Microwave and Millimeter Wave Technology IEEE pp. 1138-1141, 2010.

[12] Goncharova, A. A., Weber R.J. and Hung, Y. "Sparse Array DOA Estimation in the Presence of Unknown Non-Uniform Noise" Aerospace Conference IEEE Pages: 1-8 2011.

[13] Sinan M. Abdula Star "Improvement of DOA Estimation Based on Lanczos Algorithm"Eng.&Tech.Journal, Vol.32, Part(A), No.7,2014.

[14] Laiyth M. Al-Rawi,Dhafer R. Zaghar & Ekhlas H. Karam "Design and Implementation of a Mobile Target Tracking System using FPGA" Eng.&Tech.Journal, Vol.32, Part(A), No.11,2014.

[15] Adheed H. Sallomi and Hussien A. Hussien "Adaptive Antenna Capabilites in GSM Systems performance Improvement" Eng.&Tech.Journal, Vol.33, Part(A), No.3,2015.