International Research Journal of Engineering and Technology (IRJET) e-ISSN: 2395-0056
Volume: 02 Issue: 04 | July-2015 www.irjet.net p-ISSN: 2395-0072

SELECTIVE LISTENING OF UNDERWATER ACOUSTIC TARGETS USING MVDR BEAMFORMING

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Abstract - *Beamforming has achieved wide success in passive sonar applications such as the detection and localization of sound sources. This paper presents minimum variance distortionless response beam forming-the commonly used technique in antenna beamforming- for the selective listening of underwater acoustic target signal. The algorithm performance is compared to conventional beamforming method delay and sum. For simulations three acoustic sources are used to verify algorithm performance. The result shows that the adaptive MVDR method is capable of improving the SINR using limited no of hydrophones.*

Key Words:*Adaptive Beamforming, Conventional beamforming, Delay and sum, Minimum Variance Distortionless Response*

1. INTRODUCTION

Sound is considered the best form of radiation that travels through the sea. This is because sound is the most robust form of radiation against attenuation by underwater conditions especially when compared to other sources of radiations such as electromagnetic waves. The applications of underwater sound include acoustic devices for navigation and localization, remote control and monitoring of underwater equipments, location and identification of underwater mammals and emergency communications [1].

The technique that uses sound propagation under water to navigate, communicate or to detect objects underwater is called SONAR. Sonar systems can be active or passive. In passive sonar detection and tracking, the sonar sensor receives a signal generated by the target. The detection process involves the recognition of target signals in the presence of background noise. Passive acoustic source localization in underwater environments uses beamforming techniques in combination with hydrophone arrays. One of the main challenges in any beamforming technique is to suppress interferering sources of sounds. When the location of an interferer is known in advance, spatial nulling can be used to suppress contributions to the beamformer output from that specific direction. There exists a large set of algorithms to nullify the effects of interferers.

Conventional beamforming uses a set of weights and time delays to combine the signals from the sensors in an array. The most commonly used conventional beamforming is delay-and-sum (DSB) method [2]. The drawback of the DSB is that, the algorithm does not adapt to the direction of the noise signal. Adaptive beamforming is a powerful technique to enhance a signal of interest while suppressing the interference signal and noise at the output of the array sensor. Adaptive beamforming alters the direction pattern in according to the changes in the acoustic environment, thus provides a better performance than conventional beamforming. In this paper the performance of an adaptive beamformer, minimum variance distortionless response (MVDR) is evaluated [3]-[6]. The MVDR beamformer was originally introduced for passive systems. The minimum variance distortionless response (MVDR) beamformer, uses the recorded wave field to compute a set of optimal weights to be applied to each sensor, before coherently adding the sensor outputs. The weights are chosen such that the variance of the output is minimized while maintaining unit gain in the view direction. The MVDR beamformer offers improved resolution and image quality compared to the conventional delay-and-sum (DSB) beamformer.

2. PROBLEM STATEMENT

In most cases single sensor measurements can only be used to extract a very limited set of data. On the other hand an array of sensors can be used to yield useful information by making it act as a spatially-selective filter [2]. The spacing between elements determines the highest operating frequency. If the wavelength of the incoming signal is less than the spacing between the elements, then spatial aliasing occurs [7]-[8].what the class of target it belongs to and other important parameters describing the content.

Figure 2 shows the result obtained by varying the no of elements and spacing in a uniform linear array (ULA). The arrays are set to receive a 500 Hz wave at 60 degrees. We can see that, increasing the number of sensors gives a higher directivity. Also as the spacing increases, the directivity gets narrower. However, after a certain critical

International Research Journal of Engineering and Technology (IRJET) e-ISSN: 2395-0056
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spacing, a new main beam is introduced. The critical spacing, for a region between 0 and 180 degrees, is equal to 0.35m or λ/2. The main beam gets wider as we approach 0º and 180º, and are more distributed around the desired reception angle. In many scenarios, linear arrays are not suitable because they do not provide a 360 degree field of view. Using more than one array, in a specific geometrical disposition (rectangular, circular) we can solve these problem [7].

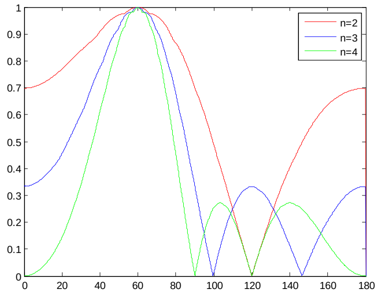


Fig -1: Effect of increasing element spacing

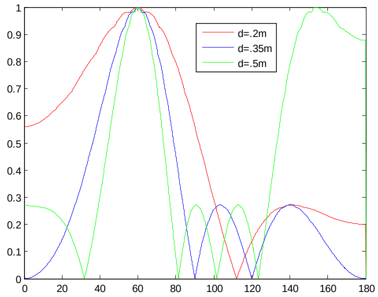


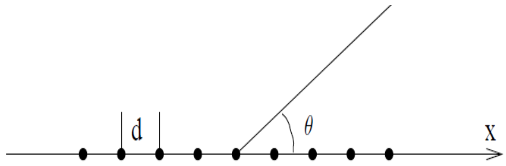
Fig -2: Effect of increasing number of elements

3. ALGORITHM

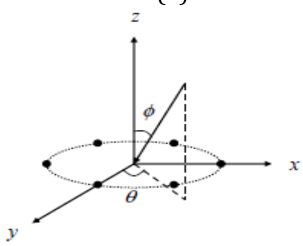
3.1 Delay and Sum Beamforming

This type of beamforming optimizes the hydrophone in a particular direction and does not change the direction as the incident source signal changes. The beam is optimized for the direction of desired source while suppressing the sound fromother directions as much as possible. Thus the directional response of the array is fixed to particular angle of incidence.

Consider a uniform linear array consisting of N number of sensors. The signal entering from an angle needs to travel an additional distance, Δu to the next sensor. Δu is proportional to distance between the elements and angle of incidence. Target source is positioned in the far field region so that the incoming wave front can be assumed to be planar.



(a)



(b)

Fig -3: (a) Uniform Liner Array (b) Circular Array

Time delay of arrival of signal between the elements

$$T_{delay}= d \sin(\theta) / c \quad (1)$$

where c is the speed of sound.

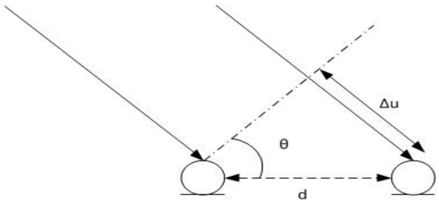


Fig -4:Sound arrival between the elements from far field

3.2 Minimum Variance Distortionless Response (MVDR)

MVDR primarily aims to minimize the total output power and to maintain a distortionlessmainlobe response towards the desired signal. Consider a uniform linear array (ULA), which consists of M number of elements. The signal and interference DOAs are θ0 and θj respectively, with corresponding steering vectors a (θ0) and a (θj).

$$y(k) = w^{H}x(k) \quad (2)$$

$$x(k) = s(k)+i(k)+n(k) \quad (3)$$

$$x(k) = s(k)a(\theta_0)+\sum_{j=1}^{q} i_j(k)a(\theta j)+n(k) \quad (4)$$

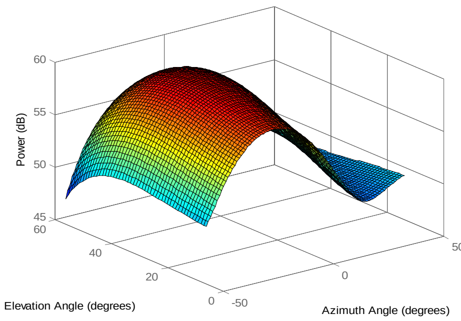
where s(k) is the desired source signal, i(k) is the interfering signal and n(k) is the noise signal.

Based on the desired goal the weight vector can be chosen on the basis of

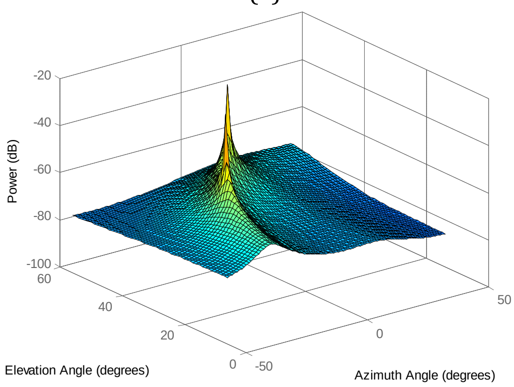
$$\text{Min } w^{H}Rw \quad \text{Subject to } w^{H} a(\theta_0)=1$$

The conventional delay and sum beamforming (DSB) has lots of limitations. The algorithm does not adapt to the direction of the noise signal, so for certain angles the suppression of noise will not be high or even negligible.

Another disadvantage of the DSB beamformer is that it operates in a limited frequency range. At a signal with low frequencies the DSB performs poorly. It has a limitation in resolving two closely-spaced signals. When two signals arrive from directions separated by less than the beamwidth, the above DSB will fail to estimate the directions of the signals. The only advantage of the DSB beamformer over MVDR is a lower computational complexity. Minimum variance distortionless response (MVDR) is similar to DSB but uses anMVDR beam. The peaks of the output spatial spectrum indicate the DOAs of the received signals. To resolve the closely-spaced signals, we can use the minimum variance distortionless response (MVDR) algorithm. To illustrate this limitation, simulate two received signals from 0 and -20 degrees in azimuth and 45 and 30 degrees in elevation.



(a)



(b)

Fig -5: (a) DSB Spatial Spectrum (b) MVDR Spatial Spectrum

4. SIMULATION

The simulation for selective listening of underwater target has been implemented, developed graphical user interface (GUI) using Matlab. Assumption were made that sound is coming from far field region, where circular array having 3 elements and radius 2m is selected. Three sound sources from different directions are seal [-90;0], beluga whale[-20;10] and blue whale[-10,0] are used. Seal sound is selected as the desired sound.

Delay and sum method gave poor SINR, classifier misclassified the outputs, Seal sound is interpreted as Beluga Whale sound. MVDR method proposed, overrides the performance of delay and sum method, correctly classifies the different classes.

Next evaluate the performance using circular and uniform linear array. 3 signals from different directions, beluga whale sound, blue whale and a sound of seal are used. The incident direction of the first signal is -30 degrees in azimuth and 0 degrees in elevation. The direction of the second signal is -10 degrees in azimuth and 10 degrees in elevation. The thirdsound comes from 20 degrees in azimuth and 0 degrees in elevation.

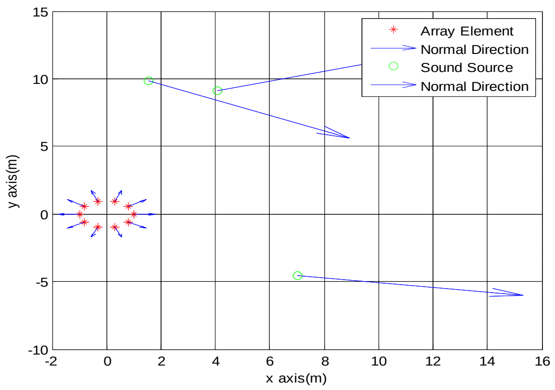


Fig -6: Location of sound sources with respect to array

The received signal is stored in a N-column matrix where each column of the matrix represents the signal collected by one hydrophone. Following figures (figure 7, 8, 9) represent the signal obtained using circular array having 10 no of sensors and radius 1m.

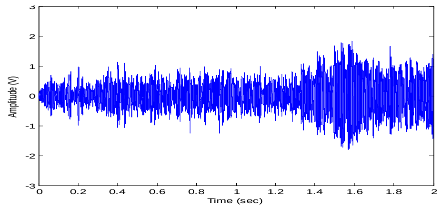


Fig -7: Sensor Output

Here beluga whale is considered as the desired sound source and delay and sum beamformer is formed. Circular array gain obtained is 11.2274. By attaching FIR filters to each sensor, the MVDR beamformer is formed. In the steering direction, the MVDR beamformer uses distortionless constraints and the array gain obtained is 17.1559.

Figure 10 shows the effect of varying no of elements in a uniform linear array in a beamformer performance. Element spacing is 0.5m. MVDR beamformer gives better SINR value.

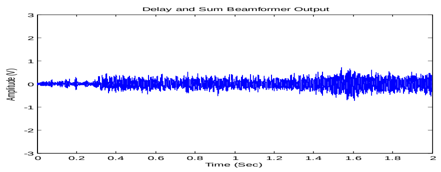


Fig -8: DSB Output

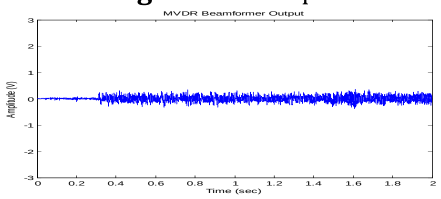


Fig -9: MVDR Output

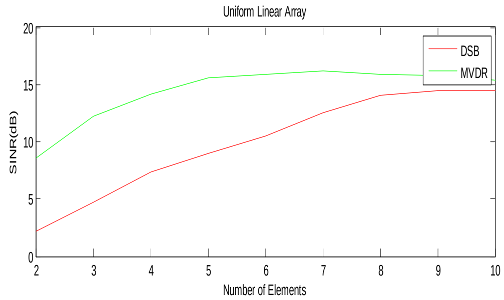


Fig -9: SINR obtained by varying the no of elements in a ULA

5. CONCLUSION

In this paper minimum variance distortionless response beamforming for the selective listening of underwater acoustic target signal is described. MVDR posses better ability to block noise source apart from steering a unity gain response to the target direction. So better classification accuracy of targets is obtained. For the simulation, sounds of marine species from different directions are used. Improved SINR has been obtained using MVDR beamforming than the conventional delay and sum method.

ACKNOWLEDGEMENTS

The authors gratefully acknowledge Naval Research Board, New Delhi for the financial assistance and the Department of Electronics, Cochin University of Science and Technology for extending all the facilities for carrying out this work.

REFERENCES

[1] Ulf Michel,"History of acoustic beamforming",Berlin Beamforming Conference (BeBeC), November 2006

[2] Steven Bell, "Acoustic Beamforming: System Theory and Requirements", Student Member, IEEE and Nathan West, Student Member, IEEE.

[3] Stergios Stergiopoulos,"Advanced Beamformers", Defence R&D Canada Technical Report,DRDC Toronto TR, 2008-101,September 2008

[4] M G Smith, "Noise source identification using microphone arrays",Proceedings of the institute of acoustics

[5] Mohammad A. Dmour, Student Member, IEEE, and Mike Davies, Member,"A New Framework for Underdetermined Speech Extraction Using Mixture of Beamformers" IEEE Transactions On Audio, Speech, And Language Processing, Vol. 19, No. 3, March 2011 445

[6] S S V Sumanth Kotta, Bulli Koteswararao Kommineni, "Acoustic Beamforming for Hearing Aids Using Multi Microphone Array by Designing Graphical User Interface",Master's Thesis, Blekinge Institute of Technology ,January-2012.

[7] Jacob Benesty, Jingdong Chen, Yiteng Huang,"Microphone Array Signal Processing",Springer Topics in Signal Processing,Springer-Verlag 2008

[8] Svend Oscar Petersen,"localization of sound sources using 3D microphone array".University of Southern Denmark-Sepetember 2004

BIOGRAPHIES

Shameer K. Mohammed is working as Project Scientist in the Department of Electronics, CochinUniversity of Science and Technology 2012 onwards. His areas of interest include, Underwater Acoustics, Artificial Intelligence, pattern recognition, Signal Processing, Automatic Target Recognition, Hidden Markov Model etc.

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