A fast vector array adaptive beam forming method

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Abstract. Based on model features of the vector sensor array signals, the paper transforms the time delay of broadband signals in time domain into the phase shift of different sub-bands in frequency domain to realize accurate time delay, and uses Hilbert Transform to construct analytic signals to form a fast vector array adaptive beam forming algorithm flow. The verification result with experimental data shows that this algorithm has much better target resolution capability than conventional beam forming algorithm. With the increase of 4-6dB in target detection capability, it has bright application prospect.

1. Introduction

The vector sensor is made up of sound pressure hydrophone and particle velocity hydrophone, which could be able to concurrently, synchronously and independently measure each quadrature component of the sound pressure scalar and the particle velocity vector in the sound field [3]. The single vector hydrophone has frequency-irrelevant dipole directivity under small dimensions, which is applicable to measuring and detecting of low (ultra) frequency signals. Because the sound pressure and vibration velocity (three components) in the coherent source sound field are completely coherent, but the sound pressure and vibration velocity in the isotropic noise field are mutually independent, so the number independent dimensions of the vector hydrophone increases to 4. For the vector array having the same number of hydrophones as the sound pressure hydrophone array, the number of independent array elements is four times that of sound pressure hydrophone. Additionally, the vector array could be able to eliminate the problem of port/starboard ambiguity that may occur to conventional line array and correctly detect the port/starboard of the target in the detection distance. Therefore, the vector array is significant for detection and parameter estimation of weak targets.

For the arrays constituted by multiple hydrophones arranged in different locations, the transmission time delays reaching each sensor in the plane wave are different. The conventional beam forming method (CBF) dose the time delay compensation and then add them and output. And CBF is greatly robust and proved to be the best air space matching filter [5] in the homogenous noise field in case of single target. Based on the signal model features of vector sensor array, the paper proposes the algorithm by which the time domain time delay of broadband signals is transformed to frequency domain phase shift on different sub-bands, to realize accurate time delay of signals. A fast and high-resolution adaptive beam forming method applicable to vector hydrophone array is formed, whose superiority is verified when used in marine recorded data.

2. Signal model of acoustic vector sensor array

Acoustic vector sensor consists of 2D and 3D vector sensors which could be used to pick up the sound pressure and vibration velocity of one point in the sound field. Taking 2D vector sensor as example, the DOA is defined as $\theta \in [0, 2\pi]$. In far field, vibration velocity component is only the cosine weight of

the sound pressure transmission direction. Their waveforms and phases are the same and completely coherent [6].
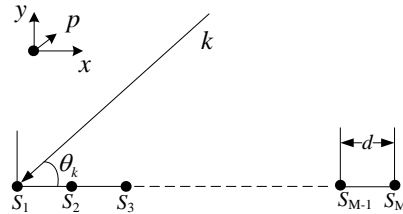

Figure 1. 2D vector sensors array model

Taking 2D acoustic vector sensor as example, M vector sensors are uniformly arranged on X axis with their directions consistent with the positive direction of X axis. The distance between array elements is d and there are P broadband signal sources in the space whose DOA is $\{\theta_1, \theta_2, \cdots, \theta_p\}$. Then the output of No. m array element is:

$$y_m(t) = \sum_{i=1}^{P} a_m(\theta_i) \otimes a(\theta_i) x_i(t) + n_m(t) \tag{1}$$

Wherein $a(\theta_i) = [1, \cos\theta_i, \sin\theta_i]^T$, $n_m(t) = [n_{pm}(t), n_{xm}(t), n_{ym}(t)]^T$.

Based on conventional time delay beam forming, the output signal of each array element is inserted with time delay $\tau_m(\theta_i) = md\cos\theta_i / c$, $m = 1, \cdots, M-1$, becoming the time delay relative to reference array element, which is the time delay when No. i signal source reaches No. m array element. It is only relevant to the target signal bearing. After time delay, the output is:

$$y_m(t) = \sum_{i=1}^{P} a_m(\theta_i) \otimes a(\theta_i) x_i(t - \tau_m(\theta_i)) + n_m(t - \tau_m(\theta_i)) \tag{2}$$

Wherein, $x(t)$ is the target signal, θ is target bearing, $p(t)$, $v_x(t)$, $v_y(t)$ is sound pressure velocity signals; $n_p(t)$, $n_x(t)$, $n_y(t)$ is the noise corresponding to each channel which is of zero mean and of spatial decorrelation.

3. Signal model of acoustic vector sensor array

3.1 Basic principle

In processing of array signals, the time delay accuracy of different array element signals will directly affect the signal processing performance [8]. In traditional digital sonar system, the multi-channel time delay is realized by digital delay line [9]. The time delay is integer times of sampling interval. The digital signal that is discretized after sampling is transmitted to one shift register whose drive period is exactly equal to the signal sampling period T_s. The length of tapped delay line is equal to the max delay time required for forming beam and then the signal is read out at one tap. Driven by drive signal, the signal in each period is moved one bit and repeat to obtain delayed time sequence digital signal. But the actual signal delay time is not the integer times of the sampling interval which can cause the time delay error.

The fractional time delay methods such as the linear interpolation, Sinc function interpolation, and Farrow filter etc. can eliminate the influence of time delay error. But these methods adopting fractional time delay filter are based on real signals processing in time domain which brings huge load to the equipment. In processing of digital signals, the common method is to adopt digital convolution filtering which can bring sharp increase of computing load [10].

According to the time shift character of Fourier Transform, if $x(n)$ corresponds to $X(k)$, then $x(n+m)$ corresponds to $X_1(k) = W_N^{-km} X(k)$. Wherein, $W_N = \exp(-j2\pi/N)$ can express formula (2) as frequency domain discretization type, i.e.:

$$Y_m(f_k) =$$
$$\sum_{i=1}^{P} a_m(\theta_i) \otimes a(\theta_i) X_i(f_k) \cdot \exp(-j2\pi f_k \tau_m(\theta_i)) \qquad (3)$$
$$+ N_m(f_k) \cdot \exp(j2\pi f_k \tau_m(\theta_i))$$

The formula above is the frequency domain type of the array signal. We found that the time delay of the signal in time domain can be realized by phase shift in frequency domain, i.e. insert phase shift factor $\exp(j2\pi f_k \tau_m(\theta_i))$ to compensate the inconsistency of signals in different channels caused by compensation angles, so as to realize different weightings.

3.2 Algorithm flow
Based on previous analysis on signal model and accurate time delay of vector array, we apply Fourier Transform to broadband beam formation of vector array, so as to establish a fast vector array adaptive beam forming algorithm (FVABF). Its main flow is as follows:
(1) Process the discretized real signals of 3M channels by filtering (this step can be finished by sonar);
(2) Perform FFT processing on the fast beat data of N point length which has been discretized and filtered;
(3) Multiply the orthogonal frequency of signals with the phase factor to eliminate the influence of time delay $\exp(j2\pi f_n \tau_m(\theta_i))$;
(4) Transform the signals in frequency domain of 3M channels into time domain to constitute analytic signals in time domain that are subjected to accurate time delay.
(5) Construct covariance matrix R for time the signal data in domain analytic;
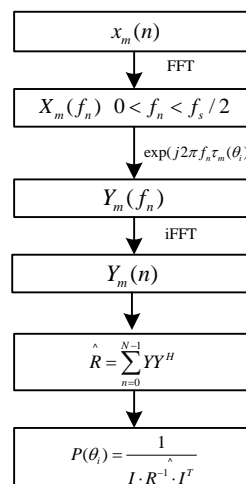Use covariance matrix R to calculate the output power $P(\theta_i)$;

$$x_m(n)$$

FFT

$$X_m(f_n) \quad 0 < f_n < f_s/2$$

$$\exp(j2\pi f_n \tau_m(\theta_i))$$

$$Y_m(f_n)$$

iFFT

$$Y_m(n)$$

$$\hat{R} = \sum_{n=0}^{N-1} YY^H$$

$$P(\theta_i) = \frac{1}{I \cdot \hat{R}^{-1} \cdot I^T}$$

Figure 2. flow for fast vector array adaptive beam forming algorithm

4. Data verification
In order to check FVABF performance index, marine tests are conducted by randomly catching non-cooperative targets at the sea that is about 60 meters deep. The test array is made up of 32 elements that are evenly and horizontally arranged. The 32 channel digital recorder is used to record voltage (sound pressure) data at the output end of hydrophones with sampling frequency being 12kHz. The data are processed with CBF and FVABF with working frequency range being [400 700] and the number of one time fast beat data points is 1024. The data processing results are compared with the data on course display to show the recorded data are valid.

Figure 3 is comparison diagram of CBF and FVABF for the data of the same moment. It can be seen from Figure 3 that min. detection angle of FVABF is smaller than that of CBF while array process gain of FVABF is 4-6dB higher than that of CBF.
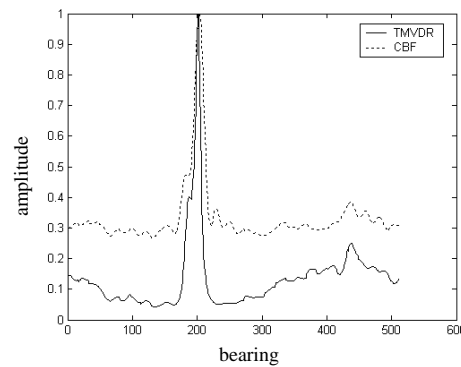

Figure 3. Comparison diagram of CBF and FVABF

4.1 Track 1 record
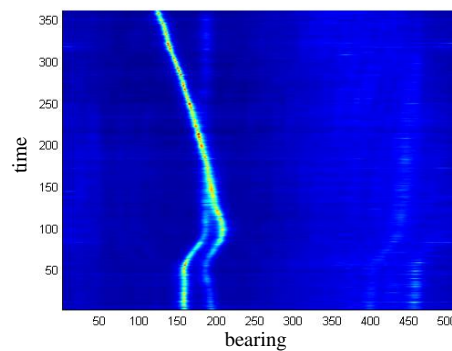Figure 4 is course bearing-time record.

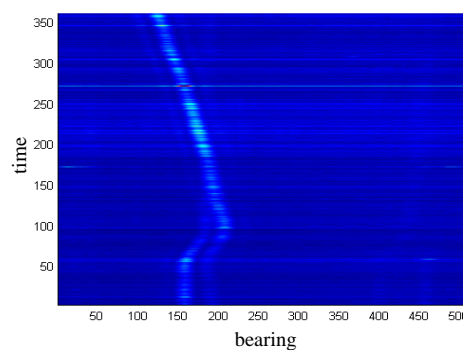**Figure 4.** (a) History of CBF processing results

Figure 4. (b) History of FVABF processing results

It can be seen that there are four batches of targets in the course (bearing angles are 150, 200, 400 and 450 respectively) in the processing results of FVABF; and there are only two batches of targets (bearing angles are 150 and 200 respectively) in the processing results of CBF. This shows that FVABF has higher detection capability than CBF.

When FVABF is at Y-coordinate 130, it could be able to detect the targets at bearings of 150 and 200. When CBF is at Y-coordinate 110, it is unable to detect such two targets. This shows that FVABF has higher bearing resolution capability than CBF.

4.2 Track 2 record

Figure 5 is the bearing-time records of track 2. Three targets was seen in the route based on the processing results of FVABF. The starting points of three targets are located in 200, 350 and 400 nearby, and the latter two targets' bearing cross each other at the location in 250-550.Compare the processing results from FVABF and CBF, we can reach the following conclusion:

(1) It can be seen from the definition of three targets' track that FVABF has the higher processing gain than CBF.

(2) It can be seen from the targets whose bearing are 350 and 400 that FVABF has the higher detecting ability than CBF.
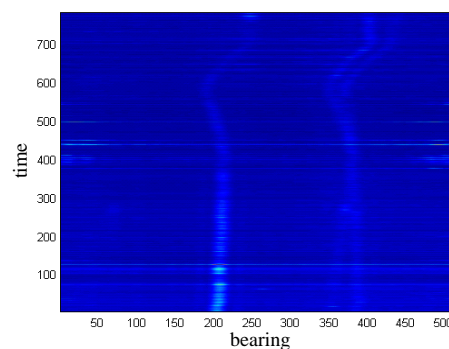

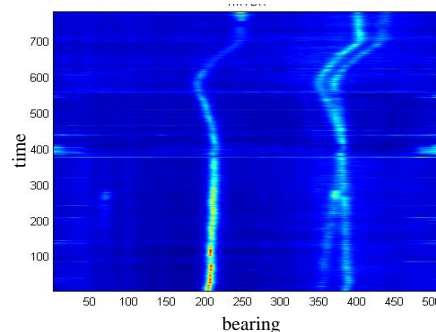**Figure 5.** (a) History of CBF processing results

Figure 5. (b) History of FVABF processing results

5. Conclusion

The adaptive beam forming technology has wide application in high resolution of sonar array. But it is prevented from being applied to engineering practices due to the large computing load. And this problem is also one of the major issues that are faced in the research of adaptive beam forming technology.

The fast adaptive beam forming algorithm designed by this paper is to apply Fourier Transform to construct analytic signals in time domain to obtain a optimal weight vector of plurality in one fast beat by Minimum Variance Distortionless Response (MVDR). Its signal detection and bearing resolution performances are both better than those of CBF. Additionally, less computing load will further bring it into wider engineering applications.

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