Sound source localization based on improved adaptive beamforming

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Abstract: In adaptive systems, the least mean square error (LMS) adaptive algorithm is widely used. The variable step size LMS algorithm (SVSLMS) based on the Sigmoid function has greatly improved the convergence speed, and the normalized LMS algorithm is stable; Sexual performance has certain advantages. Therefore, improvement is made on this basis, the SVSLMS algorithm is normalized, and the iteration speed is increased by changing the iteration factor, and the stability of the algorithm is also improved. This paper uses MATLAB simulation tools to verify the superiority of the new algorithm. Finally, the improved algorithm is applied to beamforming to estimate the azimuth of arrival of the simulated sound source signal at the partial discharge point of the submarine power cable.

1. Introduction

Submarine power cables operate underwater for a long time and are prone to corrosion, leakage, partial discharge and other phenomena \cite{1}, which have a greater impact on themselves and the marine environment\cite{2}. Among them, in the ocean, partial discharge points will produce ultrasonic, electromagnetic waves and other sound waves \cite{3}. Therefore, it is necessary to accurately locate the discharge point to reduce loss. At present, for the localization of partial discharge points, there are acoustic measurement method, acousto-magnetic synchronization method and audio-frequency induction method in the classification of precise positioning \cite{4}. But for the analysis of underwater acoustic signal positioning, currently the most used is the sonar system, that is, the analysis and positioning of beamforming and arrival estimation in the array signal \cite{5}.

Adaptive beamforming will use an adaptive algorithm to improve the accuracy of beamforming, mainly based on the least mean square error (LMS) algorithm\cite{6}. The only disadvantage of this algorithm is the slow convergence speed. In the improved LMS algorithm, Han Guangchao \cite{7} et al. found in practice that adding the Sigmoid function (SVSLMS) to the variable step size LMS can improve the convergence rate, but the disadvantage is that the stability is not high; while the normalized LMS algorithm does not converge quickly but Have strong stability. Therefore, a new LMS algorithm is proposed, SVSLMS performs normalization processing, so that the adaptive beamforming technology
has high stability and speeds up the convergence. The array signal simulated by MATLAB is mainly used to analyze the beamforming and the direction of arrival. Compared with different low signal-to-noise ratios, the improved LMS algorithm adaptive beamformer is used to analyze and compare the advantages and disadvantages of the improved algorithm. And in beamforming, the change of spatial directivity and its influence on the estimation of wave arrival.

2. Acoustic passive positioning technology

2.1. GPS

Underwater acoustic passive positioning technology is one of the main technologies in sonar systems. Sonar can be divided into active sonar and passive sonar. Passive sonar system refers to the system that passively receives the signal from the target in the water in the ocean, and after processing each function in the system, it calculates the distance and direction of the target signal to determine the position of the target signal. The partial discharge point of the submarine power cable is small and the marine environment is complicated. Active sonar sends signals to find the partial discharge point signal, and then feedbacks the positioning, but the signal interference is too large, so the passive sonar system is selected. The partial discharge point of the submarine power cable propagates in the ocean in the form of ultrasound, and collects the signal through the hydrophone (array element) arranged in different combinations. Figure 1 is the positioning flowchart.

Firstly, the collected signal is preprocessed, and the signal is output by each array element. After the direction of arrival is estimated, the undetermined azimuth is estimated, and the adaptive beamforming technology is used for delay, weighting, summation and other steps to automatically adjust the received sound source signal. The characteristics of the direction, assign different weights to the received signal of the array element, compare the received signals from various directions, strengthen the effective signal, and play an anti-interference role, so that the output is superimposed in a desired direction to produce the main lobe the beam improves the gain and directivity of the system, making the spatial directivity the more clear.

2.2. Direction of arrival estimation

The direction of arrival is also called direction finding calculation, DOA estimation, spatial spectrum estimation, direction of arrival estimation, etc. It is estimated by processing the target signal received by the base array. The relationship between each element of the "spectrum" at different positions in space. Therefore, the estimation of the direction of the sound source will be converted into the estimation of
the spectrum, which usually includes the power spectrum of the beam or the spectrum peak of the MUSIC algorithm. This system uses a uniform linear array to establish a model of the received signal (Figure 2), the number of array elements is M, the distance between the array elements is set to d, the incident signal is s(t), and the incoming wave direction is set to θ.

The marine environment is changeable and there are multi-party interference signals. Suppose the desired signal is \( d(t) \), there are N interference signals \( i_N(t) \), its direction of arrival is \( \theta_N \), its direction vector is \( a(\theta_N) \), noise signal \( n(t) \), have the same variance. Therefore, the signal output at time t received on the array element:

\[
y(t) = a(\theta_d) d(t) + \sum_{N=1}^{N} a(\theta_N)i_N(t) + n(t)
\]

Output power at this time:

\[
P(w) = E[|d(t)|^2]|w^H a(\theta_d)|^2 + \sum_{N=1}^{N} E[|i_N(t)|^2]|w^H a(\theta_N)|^2 + \sigma^2||w||^2
\]

\( \omega \) is the vector of adaptive filter weight coefficients, and \( H \) represents transpose. Then its power spectrum formula is

\[
P(\theta) = w^H R_{yy} w = \frac{1}{a(\theta)R_{yy} a(\theta)}
\]

Where: \( R_{yy} \) is the yt autocorrelation matrix. Therefore, according to formula (3), the direction with the strongest power is found by scanning all the angles in the traversal space, which is the sound source direction.

3. Adaptive beamforming

Adaptive beamforming is a method that allows the hydrophone (array element) to continuously adjust its own parameters in the complex ocean environment to adapt to the surrounding environment, suppress surrounding interference signals, and highlight the method to obtain effective signals. Beamforming is the process of delay-weighting-summation. After the elements receive the signal, they are sent to the beamformer. The beamformer weights the signal received by each element to compensate for the propagation time of each element. Extension, and finally output and sum in one of the arrays in the desired direction to form the processed signal. Adaptive beamforming is shown in the figure.
Using 16-element linear array, the received signal vector of each element \( x(t) \) is represented by a linear algebra matrix. Order vector \( \mathbf{w} = [w_1, w_2, \ldots, w_M] \), the output can be expressed as

\[
y(t) = \mathbf{w}^H \mathbf{x}(t) = \sum_{m=1}^{M} w_m^* x_m(t) \tag{4}
\]

### 3.1. Adaptive beamforming based on LMS algorithm

In the above process, due to the complexity of the marine environment, it is necessary to realize the optimization of the weight set through an adaptive algorithm under a certain optimal criterion, and adjust the weight set to the best position in real time. Here, the least mean square MSE criterion and LMS algorithm are used to realize adaptive beamforming, which is \( y(t) - d_q(t) \) the mean square value of is minimized. Rewrite (4), get

\[
J(\mathbf{w}) = E \left[ |w_q^d \mathbf{x}(t) - d_q(t)|^2 \right]
\]

it is called the cost function. After expanding and simplifying it, the cost function is:

\[
J(\mathbf{w}) = E \left[ \mathbf{x}(t) \mathbf{x}^H(t) \mathbf{w}_q - E[\mathbf{d}(t)\mathbf{x}^H(t)] \mathbf{w}_q 
- w_q^d E[\mathbf{x}(t)d_q^*(t)] + E[d(t)d_q^*(t)] \right] \tag{5}
\]

Taking the partial derivative of \( \mathbf{w}_q \):

\[
\frac{\partial}{\partial \mathbf{w}_q} J(\mathbf{w}) = 2 R_{xx} \mathbf{w}_q - 2r_{xd} \tag{6}
\]

Where: \( R_{xx} \) is \( \mathbf{x}(t) \) autocorrelation matrix; \( d_q^*(t) \) is the complex conjugate, \( r_{xd} \) is \( \mathbf{x}(t) \) cross-correlation vector with expected signal \( d_q(t) \), \( q \) is the number of signals.

Let (6)\( =0 \) get: \( \mathbf{w}_q = R_{xx}^{-1} r_{xd} \) is the array weight vector.

Update weight:

\[
\mathbf{w}_q(k+1) = \mathbf{w}_q(k) - \frac{1}{2} \mu \mathbf{V} \tag{7}
\]

In the formula: \( \mu \) is the convergence factor, which controls the convergence speed and stability of the adaptive algorithm; \( \mathbf{V} \) is the formula (6).

Replace the mathematical expectation in the formula with the instantaneous value to get \( \mathbf{V} = \mathbf{x}(t)\mathbf{e}_q(t) \), \( \mathbf{e}_q(t) = \mathbf{x}^H(t)\mathbf{w}_q(t) - d_q^*(t) \).

The final weight is updated to:

\[
\mathbf{w}_q(t+1) = \mathbf{w}_q(t) - \mu \mathbf{x}(t)\mathbf{e}_q(t) \tag{8}
\]

After the weight is updated, the output signal is finally obtained as:

\[
y(t) = \mathbf{w}^H(t+1)\mathbf{x}(t) = \sum_{m=1}^{M} w_m^* x_m(t) - \mu \mathbf{x}(t)\mathbf{e}_q(t) \mathbf{x}_q(t) \tag{9}
\]

In summary, in the LMS algorithm, the selection of the convergence factor \( \mu \) will affect the stability and convergence speed of the algorithm. If \( \mu \) is too large, the convergence speed is fast but the steady-state error will increase; if \( \mu \) is too small, the stability is good, and the convergence speed is reduced. There is a certain inconsistency between the two. Therefore, the following improvement methods are proposed to improve the convergence. Speed can increase stability.

### 3.2. Adaptive beamforming based on improved LMS algorithm

Error and convergence speed are the two most critical indicators in the above-mentioned LMS adaptive algorithm. Usually, the convergence factor \( \mu \) determines the performance of the algorithm. In the fixed-step LMS algorithm, that is, the normalized LMS algorithm, the steady-state error and the convergence speed restrict each other. Therefore, the Sigmoid function is added to the improved LMS algorithm, and then normalized to improve the performance of the LMS algorithm, and the accuracy can also be improved through adaptive beamforming. Refer to the Sigmoid function LMS algorithm (SVSLMS algorithm) mentioned in [7], so that

\[
\mu(t) = \beta(1 / (1 + \exp(-\alpha|e(t)|)) - 0.5) \tag{10}
\]

Where: \( \alpha \) controls the shape of the Sigmoid function and determines the speed of the curve's ascent; \( \beta \) controls the value range of the Sigmoid function. There is a certain relationship between \( \mu(t) \) and \( e(t) \) in the Sigmoid function. In the initial convergence stage of the algorithm, the error \( e(t) \) is
relatively large, and the corresponding convergence the larger the factor, the faster the convergence speed, but the stability will decrease. Therefore, combined with the normalized LMS algorithm to improve the stability, the iterative formula of the algorithm is

\[
\mathbf{w}(t+1) = \mathbf{w}(t) - \mu(t) \frac{1}{\mathbf{x}(t)\mathbf{x}(t)^H} \mathbf{x}(t) \mathbf{e}(t)
\]

(11)

Therefore, in formula (11), change the convergence factor \( \mu \), and finally get the output signal

\[
y(t) = \mathbf{w}^H(t+1)\mathbf{x}(t) = \sum_{m=1}^{M} \mathbf{w}_m(t) - \mu(t) \frac{1}{\mathbf{C} + \mathbf{x}(t)\mathbf{x}(t)^H} \mathbf{x}(t) \mathbf{e}(t) \mathbf{x}(t)^H \mathbf{e}(t)
\]

(12)

In the formula: \( \mathbf{C} \) is a constant.

Combine normalization and Sigmoid function to get a new LMS algorithm. It will automatically adjust as the input signal power changes. The convergence speed changes with the change of \( \mu \), the convergence speed has been improved, and the stability has been enhanced.

3.3. Simulation algorithm comparison

By comparing and analyzing the influence of the SVSLMS algorithm and its improved algorithm in adaptive beamforming, the differences between the algorithms can be understood. The simulation signal takes 1,500 times as the number of iterations, loops 200 times, and the signal-to-noise ratio is 0.

Figure 4 shows the comparison of the convergence factor between the SVSLMS algorithm and the improved algorithm. The same is the variable step size LMS algorithm. After normalization, the convergence factor changes quickly and has strong stability. Figure 5 compares its convergence speed. For the black curve, the optimal parameters selected by the SVSLMS algorithm are \( \alpha = 10 \), \( \beta = 0.3 \) respectively. It can be seen that the improved algorithm is better than SVSLMS in terms of convergence rate.

4. Simulation positioning

4.1. Pretreatment

Different submarine cable materials have different requirements on the depth of the seabed. Most of them are concentrated in the ocean at 2,000 m. At the same time, to avoid damage due to mechanical external forces, the submarine cable is usually buried 1~1.5 m below the seabed, so local collection. When the signal of the discharge point is discharged, the signal will be interfered by different degrees of water, gravel, organisms, etc. Usually, the ultrasonic signal is above 20 kHz, and the signal emitted by the organisms in the ocean is at 52 Hz, so the impact on the organism is negligible, but the signal is still affected by the fluctuation, which is collectively referred to as ocean noise here. Use 20 kHz as the simulation signal to simulate non-stationary signals, and add two kinds of noises, Gaussian partial white noise and uniformly distributed noise, to make the simulation more realistic. The horizontal axis is the number of iterations, and the vertical axis is the signal amplitude. Figure 6 is a comparison chart before and after signal processing.
4.2. Direction of Arrival Estimation and Positioning

In the simulation experiment, the target analog signal is taken as a uniform 16-element linear array. The distance between every two array elements is \( d = 0.5 \), the signal-to-noise ratio is 0, the number of samples is 2,000, the incident signal is \( s(t) \), the traversal angle is between \(-90^\circ \) to \(90^\circ \), and the interference direction is added at the same time \(-45^\circ \) and \(30^\circ \), the noise is Gaussian white noise. Under such conditions, the analog signal is processed through an improved beamforming algorithm to obtain a beam pattern. Compared with the application of the improved algorithm in beamforming, the strongest signal is obtained through the power spectrum.

![Direction of Arrival Estimation and Positioning Chart](image)

Figure 8 Direction of arrival estimation and positioning
It can be seen from Figure 7 that in the 0° direction, the main lobe is obvious, highlighting directivity, while in the interference direction -45° and 30°, the improved algorithm has obvious nulling phenomenon, which is well suppressed. Figure 8 uses an improved algorithm to calculate the power spectrum. Through the power spectrum, the direction with the highest power can be obtained and determined as the sound source direction. Combined with Figure 7, the beamforming algorithm is used to extract the signal. For the interference signal, it is obvious. Compare the height of the peaks to exclude other directions.

5. Conclusion
The direction of arrival estimation is usually post-calculated in the sonar system. It is used to detect the existence of large objects, and it is easy to judge the general direction. The beamforming algorithm can strengthen and extract useful direction signals. This paper mainly uses the adaptive beamforming algorithm to estimate the direction of arrival, and converts the direction estimation into power spectrum estimation to obtain the sound source position.

In the simulated seabed environment, the non-stationary signal adopts adaptive beamforming, combined with the pattern of Figure 7, the main lobe protrudes into the desired direction signal. In the power spectrum of Figure 8, the direction of the sound source has the largest amplitude, which has a certain effect on signal positioning. The propelling effect. The array element arrangement can be further explored. The simulation experiment uses a 16-element uniform linear array as the receiving base array model, but the number of sensors makes the overall system cost higher. If the array element is changed and the number is reduced, the accuracy of the positioning of the partial discharge point of the submarine power cable can be further explored.

References
[1] Zhang Yenning. Overview of fault location methods for overhead-cable hybrid transmission lines[J]. Electric Power Engineering Technology. 2020, (6): 44-51.
[2] Shen Zhiyi, Yang Liuhui, Song Liang. Research on power cable fault location method[J]. Modern Industrial Economics and Information. 2019, (7): 132-133.
[3] Qiao Mei Gao, Yu Lin Qi. Power Cable Fault Locator Based on Synchronization Principle of Sound and Magnetic[J]. Advanced Materials Research, 2013, 2584: 2337-2340.
[4] Gao Jianping. Technical analysis and system design of power cable fault location[D]. Hebei Province: North China Electric Power University, 2010. 50
[5] Liu Yongjia. Research on underwater acoustic passive positioning technology based on array signal processing [D]. Guangdong Province: South China University of Technology, 2011.87
[6] Ge, Huamin, Zhao, Weiwei. Improved Variable Step-Size and Variable Parameters LMS Adaptive Filtering Algorithm[J]. Sensors & Transducers,2013, 158(11):369-373.
[7] Han Guangchao, Wang Feng, Zhao Heming, et al. A new variable step size LMS adaptive filtering algorithm and its application[J]. Journal of North University of China: Natural Science Edition. 2017, (2): 140-144.
[8] Jiang Guojun. Research on DOA estimation algorithm of circular array based on beam space and differential co-array domain [D]. Heilongjiang Province: Harbin Institute of Technology, 2020.142