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A method of multi-channel reference signals acquiring in broadband ANC

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Abstract: In a flank array on an unmanned underwater vehicle (UUV), self-generated noise which has broadband and colored spectrum property in frequency and spatial domain is the main factor affecting the performance of weak signal detection, so the technique of adaptive noise cancellation (ANC) as well as physical denoising and active noise cancellation are often used in practice. Because ANC is based on correlations, improvements in performance come from better correlation between reference signals and primary signals. Taking full advantage of the characteristics of flank arrays and the characteristics of information obtained from hydrophones, a new method for reference signal acquisition for adaptive noise cancellation is proposed, in which the multi-channel reference signals are obtained by accurate delaying for a given direction of arrival (DOA) and differencing between adjacent outputs of array elements. The validity of the proposed method was verified through system modeling simulations and lake experiments which showed good performance with little additional computational burden.

Keywords: weak signal detection; adaptive noise cancellation; multi-channel reference signal
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1 Introduction

The target acoustic signal received by unmanned underwater vehicle (UUV) flank array is inevitably polluted by the marine environmental noise and self-generated noise. Since the amplitude distribution of marine environmental noise is Gaussian at general depth, and the self-generated noise is broadband, time-variant, non-stationary and colored noise field which has a certain space direction[1], the capabilities of the weak target signal detection are affected extremely. Especially, with the development of modern noise control technology, stealth technology and the emergence of new dynamic technique, the self-generated noise has turned into major noise source in received signals and the main factor in capability of weak signal detection. In practice system, adaptive noise cancellation (ANC) technology is usually used besides many methods such as physical noise reduction and active noise reduction methods. To acquire better results in noise cancellation, the choice of reference signals and the full use of their information (multi-channel system) are essential. The direct output of additional noise sensor (accelerometer) near the source is generally considered as better method[2], but there are a lot of noise sources (mainly including mechanical noise, propulsion system noise and hydrodynamic noise, etc.). Furthermore, the output reflects only the vibration strength in UUV flank system. In fact, the part of self-generated noise in the signal received from the hydrophone is a complex result, which is decided by transmission property of different noise sources and the system structures. For the results of adaptive noise cancellation are determined by the correlation between primary and reference signals, the cancellation would be limited if the direct output of accelerometer is used as a reference signal.

Taking into account the fact that the output of hydrophone is the best indicator of self-generated noise, furthermore, the spatial difference exists between far field target and self-generated noise, according to the noise cancellation theory[3-5], the more the reference sources, the better the canceling results can be gained. In this paper, a new method of multi-channel reference signals obtained in ANC is presented, which uses the delaying and differencing of adjacent channel signals. The results of simulation and lake experiment show that the effect of self-generated noise cancellation is increased remarkably comparing with conventional methods.

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2 System modeling and theory analysis

2.1 System modeling

The outputs of UUV flank array hydrophone include three kinds of signals, i.e. self-generated noise, marine environmental noise and far field target signal. A system model can be built, as an example where three independent self-generated noise sources (certainly it also expands to more sources) and M hydrophones are assumed. Let n_j be the jth noise source, T_i be the ith hydrophone, m_{0i} be the marine environment noise from the ith hydrophone, $s_i(n)$ be the far field target signal received by the ith hydrophone. Denote the unit impulse response of the jth noise source to the ith hydrophone transmission system by $g_{ji}(n)$, i=[1, 2, 3,⋯,M]. So the received signal of T_i can be expressed as

$$y_i(n) = s_i(n) + g_{1i}(n) * n_1(n) + g_{2i}(n) * n_2(n) + g_{3i}(n) * n_3(n) + m_{0i}, \tag{1}$$

where "*" denotes linear convolution. Fig.1 gives structure of the system model, which represents the realization of only one hydrophone noise cancellation system, where noise source is a wide stationary process, correlation among different noise sources and that between signal and noise do not exist, and all noise transmission paths are equivalent to linear time-invariant systems. From the Fig.1, it can be easily found out that the reference signals are obtained through time delay (based on the DOA of target) and difference between adjacent hydrophone outputs. By differencing, the target signal in references is eliminated, and the disturbance on the useful signal is avoided during the adaptive process.

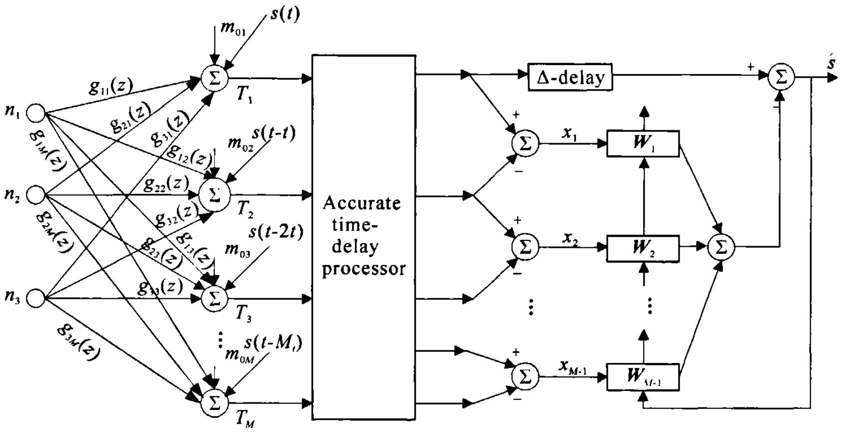


Fig.1 The system model of multi-channel differencing adaptive noise cancellation

2.2 Theory analysis

Suppose the incidence direction of far field target signal is in the normal (the other incidence directions can be weighted by array steering vector, the results are the same). To simplify the analysis, the impact of marine environmental noise is ignored. So from Eq.(1), the following is derived:

$$y_1(n) = s_1(n) + g_{11}(n) * n_1(n) + g_{21}(n) * n_2(n) + g_{31}(n) * n_3(n);$$

$$y_2(n) = s_2(n) + g_{12}(n) * n_1(n) + g_{22}(n) * n_2(n) + g_{32}(n) * n_3(n);$$

$$\cdots$$

$$y_M(n) = s_M(n) + g_{1M}(n) * n_1(n) + g_{2M}(n) * n_2(n) + g_{3M}(n) * n_3(n).$$

where $s_1(n)$=$s_2(n)$=$s_3(n)$=,⋯,=$s_M(n)$=$s(n)$, the outputs of adjacent channel differencing are

$$x_1(n) = [g_{11}(n) - g_{12}(n)] * n_1(n) + [g_{21}(n) - g_{22}(n)] * n_2(n) + [g_{31}(n) - g_{32}(n)] * n_3(n);$$

$$x_2(n) = [g_{12}(n) - g_{13}(n)] * n_1(n) + [g_{22}(n) - g_{23}(n)] * n_2(n) + [g_{32}(n) - g_{33}(n)] * n_3(n);$$

$$\cdots$$

$$g_{1M-1}(n) - g_{1M}(n)] * n_1(n) + [g_{2M-1}(n) - g_{2M}(n)] * n_2(n) + [g_{3M-1}(n) - g_{3M}(n)] * n_3(n).$$

Let $g_{ijmn}(n)$=$g_{ij}(n)$-$g_{mn}(n)$, $X(n)$=$[x_1(n),\cdots,x_{M-1}(n)]^{\mathrm{T}}$, $N(n)$=$[n_1(n),n_2(n),n_3(n)]^{\mathrm{T}}$, and the z domain transform of $g_{ijmn}(n)$, $X(n)$, $N(n)$ (for the acquired sampling series) are $g_{ijmn}(z)$, $X(z)$ and $N(z)$, then

$$X(z) = N^{\mathrm{T}}(z)G(z), \tag{2}$$

where $G(z)=\begin{bmatrix} g_{1112}(z) & g_{1213}(z) & \cdots & g_{1M-11M}(z) \\ g_{2122}(z) & g_{2223}(z) & \cdots & g_{2M-12M}(z) \\ g_{3132}(z) & g_{3233}(z) & \cdots & g_{3M-13M}(z) \end{bmatrix}$.

Conduct multi-channel noise cancellation to $y_1(n)$, using $X(n)$ as the reference signals, then the best available approximation of the target signal of the converged system can be found. The following is the solution of unconstrained optimal weight vector.

$$\boldsymbol{\Phi}_{xx}(z)=\boldsymbol{G}^{\mathrm{H}}(z)\boldsymbol{\Phi}_{nn}(z)\boldsymbol{G}(z), \tag{3}$$

where $\boldsymbol{\Phi}_{xx}(z)$ is the power spectrum matrix of the M-1 reference signals of the adaptive filter, and

$$\boldsymbol{\Phi}_{nn}(z)=\begin{bmatrix}\Phi_{n_1n_1}(z) & 0 & 0\\ 0 & \Phi_{n_2n_2}(z) & 0\\ 0 & 0 & \Phi_{n_3n_3}(z)\end{bmatrix},$$

$\boldsymbol{\Phi}_{n_in_i}(z)$ is the power spectrum density matrix of the ith noise source, i=1,2,3. Also, the cross-spectrum vector between the reference input $X(n)$ and the original input $y_1(n)$ can be derived:

$$\boldsymbol{\Phi}_{xy1}(z)=\boldsymbol{G}^{\mathrm{H}}(z)\,\boldsymbol{\Phi}_{nn}(z)\boldsymbol{G}_1(z), \tag{4}$$

where $\boldsymbol{G}_1(z)=[g_{11}(z), g_{21}(z), g_{31}(z)]^{\mathrm{T}}$.

Define the best transfer function $\boldsymbol{W}_{\mathrm{opt}}(z)=[\boldsymbol{W}_{1\mathrm{opt}}(z)\cdots\boldsymbol{W}_{\mathrm{M-1opt}}(z)]^{\mathrm{T}}$, where $\boldsymbol{W}_{i\mathrm{opt}}(z)$ is the z transform of $\boldsymbol{W}_{i\mathrm{opt}}(n)$, $\boldsymbol{W}_{i\mathrm{opt}}=[w_{i1\mathrm{opt}}\ w_{i2\mathrm{opt}}\cdots w_{iN\mathrm{opt}}]^{\mathrm{T}}$, $i=1,2,\cdots,M-1$. N is the order of channel filters, so the unconstrained Wiener solution is:

$$\begin{gathered}\boldsymbol{W}_{\mathrm{opt}}(z)=\boldsymbol{\Phi}_{xx}^{\ -1}(z)\boldsymbol{\Phi}_{xy1}(z)=\\ [\boldsymbol{G}^{\mathrm{H}}(z)\boldsymbol{\Phi}_{nn}(z)\boldsymbol{G}(z)]^{-1}\boldsymbol{G}^{\mathrm{H}}(z)\boldsymbol{\Phi}_{nn}(z)\boldsymbol{G}_1(z).\end{gathered} \tag{5}$$

The best weighted vector demonstrates the overall minimum MSE of the performance surface. In a special case, if $\boldsymbol{G}$ is a square matrix, i.e. the number of independent noise sources is the same as the number of input channels, and the matrix is invertible, from Eq.(5), it can be derived that

$$\boldsymbol{W}_{\mathrm{opt}}(z)=\boldsymbol{G}^{-1}(z)\boldsymbol{G}_1(z), \tag{6}$$

$$\hat{s}(z)=y_1(z)-\boldsymbol{W}_{\mathrm{opt}}(z)\boldsymbol{X}(z)=s(z). \tag{7}$$

It is obvious that the noise in original input would be cancelled completely by the adaptive filter, and signal doesn't become corrupted in ideal condition, thus the output SNR is enhanced.

In practice, even the direction of target is unknown, the method would be still applicable in the way demonstrated as follows. Given the incidence of target in turn with the step of half beamwidth in the whole searching fan range, the delay of the array output would be determined. In the case of the given direction that is different from the true one, the reference signals obtained by delaying and differencing also include target signal. As a result, self-generated noise and target signals would be cancelled during the ANC, where the amplitude of beam forming in given direction is very small. On the other hand, when the true and given incidence directions are consistent, after delaying and differencing, only the linear combination of output of self-generated noise from hydrophones is included in the reference signals. Then cancellation only reacts on self-generated noise and there is no harm to target signal, and the direction corresponding to the maximum output of beam forming is in the direction of target.

3 Performance analysis

In the actual system, due to the similarity in the installation position of hydrophone and that in the independent transmission of different noise sources, as well as the complexity of transmission and that of the multi-source action simultaneous on the UUV, the constructed simulation system adopting the same noise source uses the same type of transfer function approximated by a higher order FIR filter. By changing the filter parameters, the imitation of different characteristics of transmission channel is achieved. In the simulation, aiming at the gain of SNR, the mono-frequency signal is used as target signal, and LMS algorithm is adopted. The coherence function is performed to measure the correlation between input of reference x_i and the primary $y_1(n)$ under different circumstances, which is defined as

$$\begin{gathered}\Phi_{x_ix_i}(z)=\Phi_{n_1n_1}(z)\left|g_{1i1i+1}(z)\right|^2+\Phi_{n_2n_2}(z)\left|g_{2i2i+1}(z)\right|^2+\\ \Phi_{n_3n_3}(z)\left|g_{3i3i+1}(z)\right|^2;\end{gathered}$$

$$\begin{gathered}\Phi_{x_iy_1}(z)=g_{11}^{*}(z)g_{1i1i+1}(z)\Phi_{n_1n_1}(z)+g_{21}^{*}(z)g_{2i2i+1}(z)\Phi_{n_2n_2}(z)+\\ g_{31}^{*}(z)g_{3i3i+1}(z)\Phi_{n_3n_3}(z);\end{gathered}$$

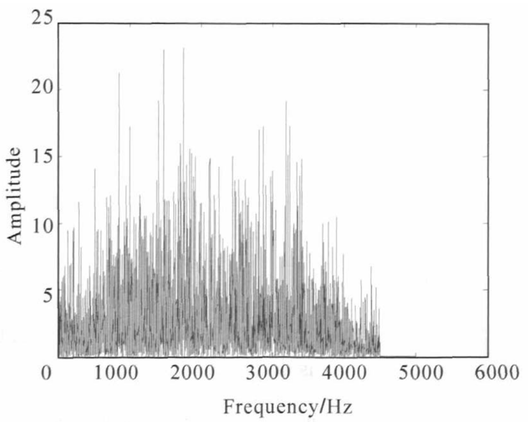
$$\begin{gathered}\Phi_{y_1y_1}(z)=\Phi_{ss}(z)+\left|g_{11}(z)\right|^2\Phi_{n_1n_1}(z)+\left|g_{21}(z)\right|^2\Phi_{n_2n_2}(z)+\\ \left|g_{31}(z)\right|^2\Phi_{n_3n_3}(z).\end{gathered}$$

$$\gamma_{x_iy_1}^2(z)=\frac{\left|\Phi_{x_iy_1}(z)\right|^2}{\Phi_{x_ix_i}(z)\Phi_{y_1y_1}(z)}, \tag{8}$$

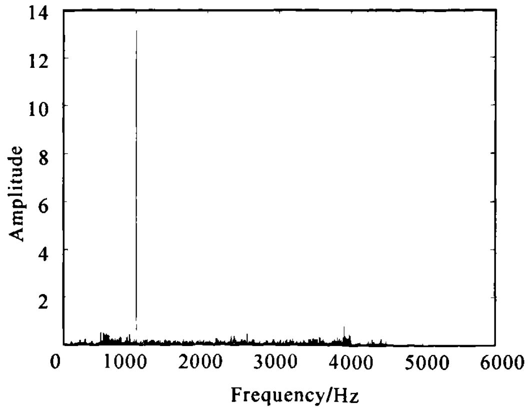
where $\Phi_{x_ix_i}(z)$, $\Phi_{y_1y_1}(z)$ and $\Phi_{n_in_i}(z)$ denote power spectra of x_i, y_1 and n_i, respectively, and $\Phi_{x_iy_1}(z)$ denotes cross-power spectrums of x_i and y_1.

In the simulation, the sinusoid centering on 1 kHz, which is embedded in three kinds of independent Gaussian white noise, impinges on the uniform line array with five omnidirectional hydrophones in the normal incidence direction, where elements are half wavelength apart. The adaptive filter with order N=8 and step size μ=0.02 is adopted. SNR of the input/output signal is defined as the rate between module of 1 kHz spectral line and the module sum of the rest of spectral lines in frequency domain. The SNR subtraction of output from input is defined as SNR gain. Table 1 shows the relationship between SNR gain with the transfer performance of channels, where the amplitude of input sinusoidal signal is 0.1. For the same system, table 2 shows the change of SNR gain vs. the SNR of input signal.

Fig.2 illustrates a typical waveform in the simulation by the proposed method, where the amplitude of input signal is 0.1, input SNR is −62.037 6 dB; SNR gain is 15.662 4 dB.



(a) Power spectrum of input signal



(b) Power spectrum of output signal

Fig.2 Power spectrum of input and output signals

The results of simulation show that the use of differencing signals as reference is helpful for counteracting the noise of input signal effectively. Table 1 shows that SNR gain increases (corresponding to better canceling effect) with the mean value enhancement of the coherence function, when taking x_1 as an example, the transfer function is adjusted. The increase of coherence function reflects enhancement of correlation, so in practice, this function can be used as a standard when trying to choose the reference signal (if selection is available). Table 2 shows the value of the mean coherence function changes with input SNR in the experiment conditions. In fact, the correlation determines the ultimate degree of canceling, the SNR gain decreases with the increase of input SNR, which proves the fact.

Table 1 SNR gain in the condition of the coherence changed

Input SNR/dB	Output SNR/dB	Mean of coherence function of x_1	SNR gain/dB
−62.179 6	−52.566 7	0.601 2	9.612 9
−65.166 0	−55.156 7	0.688 1	10.009 3
−63.528 1	−52.066 1	0.778 2	11.462 1
−63.564 4	−51.974 6	0.780 3	11.589 9
−63.275 8	−49.382 7	0.865 8	13.893 1
−62.585 6	−47.661 0	0.912 5	14.924 6
−62.932 0	−44.882 0	0.979 2	18.050 0

Table 2 SNR gain in the condition of the input SNR changed

Input SNR/dB	Output SNR/dB	Mean of coherence function of x_1	SNR gain/dB
−64.181 2	−48.873 3	0.875 3	15.307 9
−54.603 5	−39.636 1	0.871 3	14.967 4
−48.440 3	−34.766 4	0.872 4	13.674 0
−45.481 7	−32.889 2	0.868 5	12.592 5
−43.755 4	−31.909 4	0.872 8	11.845 9
−40.131 3	−30.416 9	0.866 4	9.714 4
−35.663 0	−29.302 9	0.852 7	6.360 1

4 Lake experiments

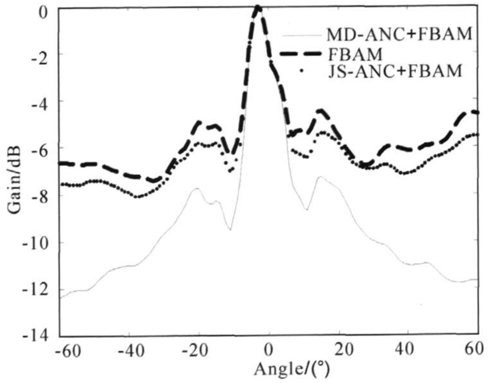
In order to verify the proposed method, data from lake experiments was processed. Both the self-generated noise and target signal were broadband noise (2~15 kHz). After shifting and differencing, 16 channels of reference signal were acquired, the output of each hydrophone was processed by ANC, and the detection was carried in beam domain. In the experiment 1, the target was located at the angle of −30°, and 50 Monte Carlo experiments with 6 000 snapshots under the assumption that false alarm probability is defined as 0.05 were implemented. Table 3 shows the statistical characteristics of the results.

Table 3 Statistical characteristics (experiment 1)

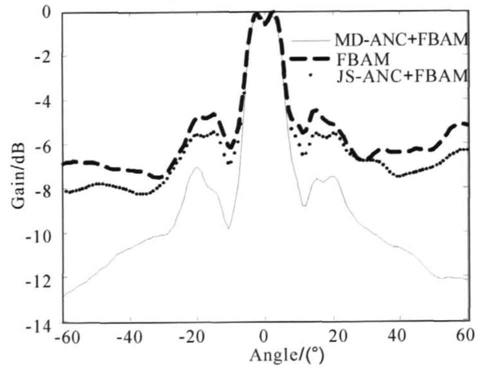
Input SNR /dB	−32	−30	−28	−26	−24
P_{d1}	0.050	0.050	0.050	0.125	0.900
P_{d2}	0.125	0.700	1.000	1.000	1.000

Notes: P_{d1} denotes detection probability where ANC was not carried before broadband beam forming; P_{d2} denotes detection probability where ANC was carried before broadband beam forming.

Fig.3 illustrates typical beam pattern of lake experiment 2 when different approaches were adopted.



(a) Spatial angle spectrum 1



(b) Spatial angle spectrum 2

Fig.3 Typical beam pattern

In Fig.3, JS-ANC+FBAM denotes broadband beam forming, where ANC was implemented and the reference signals were acquired by the output of accelerometers near the self-generated noise source; FBAM denotes broadband beam forming without ANC; MD-ANC+FBAM denotes broadband beam forming with ANC, in which the reference signals was acquired by the proposed method.

From the results of experiment above, a conclusion can be drawn that the effect of ANC can be improved greatly by the new multi-channel reference signal and the sidelobe of beam forming is reduced significantly. As a result, the detection probability increased.

5 Conclusions

In order to enhance the UUV's capability of detecting weak signal in a complicated environment, a new method of the multi-channel reference signals obtained in ANC by use of delaying and differencing is proposed. Simulation and the lake experiment show that the proposed method performs better in the self-generated noise canceling. In the same experimental conditions, lots of experimental results show that the minimum detectable SNR could be reduced about 4~5 dB compared to the untreated signal, and the SNR could be increased about 3~5 dB in beam domain.

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