Passive-Range Estimation Using Dual Focused Beamformers

Ki-Man Kim, In-Sic Yang, Seung-Yong Chun, and Won-Tchon Oh

Abstract—The passive-range estimation technique using a single focused beamformer has been studied under the sea. However, there are not many more methods in the case of closed multisource environments. In this paper, we propose the technique using dual focused beamformers to estimate the ranges of a closed multisource, and compare the proposed technique with the previous method. The proposed method is verified via computer simulation under the simplified multipath underwater channel model.

Index Terms—Dual focused beamformers, multisource, passive-range estimation.

I. INTRODUCTION

In general, active-range estimation is useful to the far field between sound source and an observer, on the other hand, the passive method is efficient for the target in the near field. A lot of efforts have been expended in the development of passive ranging systems for underwater environments since the past several years [1]–[5]. The basic estimator consists of an input nonlinear signal processing which takes the passive time-delay measurements and converts them into a linear set of source-range measurements. The weak point of this transformation is that the additive measurement error is a non-Gaussian, nonstationary, random process. This includes wavefront curvature, the triangulation method, and the range-estimation technique using single focused beamformer. The beamformer method has several merits more than the other methods in low-SNR environments.

In triangulation, the intersection of the bearing lines from these two arrays provides information on the source location. The triangulation procedure is robust because the signal need only remain coherent over an individual array aperture. With wavefront-curvature processing, three sensor groups are employed to estimate the two delays from the forward sensor group to the mid-sensor group and from the mid-sensor group to the after sensor group. The wavefront-curvature processing requires that the signal remain coherent from one sensor group to the next. If, in the triple-aperture array system, we envision the system as being two groups, one composed of the forward group and mid-group and one composed of the mid-group and after group, then, wavefront-curvature can be viewed as triangulation [2]. When multitargets are relatively in the same direction or close to each other, the single focused beamformer estimator is not able to find each source for the ambiguity. From this result, the estimator perceived the multitargets to be one target, and in detecting the multitargets, it is practically difficult for the geometrical architecture to detect the targets at the same time.

We assume that the true range is from the center of the linear array to the acoustic source. It means that there are two groups including several or many subarrays to make their own beam. These beamformers of the subarrays make their own beams in equal increments to the equal range in the known direction of the target, step by step. As a result of this, the maximum value can be determined by measuring the power of summed output of the each beamformer. The proposed technique can estimate the ranges of multitargets and resolve the targets which are close or in the same direction, where the single beamformer cannot resolve. So, it is possible to know the relative position of the targets, according to the bearings and ranges. Performance of the proposed passive-range estimation is compared with a method using time-delay estimation, and the range estimation error is analyzed in the area of interest.

II. GENERAL PASSIVE RANGE ESTIMATION METHODS

Hassab [4] has considered the case where all the three sensors are on board the observer, but the physical dimensions of a submarine put a stringent limitation on the distances separating the sensors. Fig. 1 explains the geometry utilized in the time-delay analysis. “A” and “K” are the two sets of on-board sensors and “F” is the towed array of sensors. $l_1$ is the separation between the towed array, and sensor $K$, and is much larger than $l_2$, the separation between the on-board sensors. $R_F$, $R_K$, and $R_A$ are the three direct paths of the sonar signals from the source to these sensors. This gives the two values of time-delay measurements. The difference in propagation times between paths $R_F$ and $R$ is...
Fig. 2. Triangulation method.

\[ \tau_1 = \frac{(R_F - R_o)}{c}, \text{ the difference in propagation times between} \]

paths \( R \) and \( R_A \) is \( \tau_2 = \frac{(R - R_A)}{c} \) where \( c \) is the velocity of sound under the sea. Target range and bearing are measured with respect to sensor “K” and so “R” is the actual range, and \( \theta \) the actual bearing angle of the target from the observer. We now derive the relation between the sonar measurements \( \tau_1 \) and \( \tau_2 \) and target range \( R \) and bearing \( \theta \). Using the law of cosines on triangles \( TKK \) for \( \tau_1 \) and \( TKA \) for \( \tau_2 \) in Fig. 1, and defining \( \tau_1 = \frac{(R_F - R)}{c} \), \( \tau_2 = \frac{(R - R_A)}{c} \). We find that for \( \tau_1 \) and \( \tau_2 \) as

\[
\begin{align*}
\tau_1 &= \frac{-R + (R^2 + R_1^2 - 2RR_1 \sin \theta)^{0.5}}{c} \quad \text{(1a)} \\
\tau_2 &= \frac{R - (R^2 + R_2^2 - 2RR_2 \sin \theta)^{0.5}}{c} \quad \text{(1b)}
\end{align*}
\]

Equation (1a) and (1b) are two nonlinear algebraic expressions for \( \tau_1 \) and \( \tau_2 \) in terms of \( R \) and \( \theta \). We now develop the nonlinear signal processor necessary to get separate expressions for the range \( R \) and bearing \( \theta \) in terms of the time delays \( \tau_1 \) and \( \tau_2 \) which eventually become the set of input measured parameters. So, from (1a) and (1b) we can find the next expression for range \( R \) using the Taylor series expansion and neglecting higher order terms

\[
R = \frac{l_1l_2(l_1 + l_2) \cos^2 \theta}{2c(l_2\tau_1 - l_1\tau_2)}. \quad \text{(2)}
\]

The triangulation method can be derived from the law of sine. From Fig. 2, we have two expressions for \( R_2 \). Multiplying those two expressions together and taking the square root, yields

\[
R_2 = \left( \frac{R_{21}}{R_{62}} \right)^{0.5} \cdot \left( \frac{\cos B_{22}}{\cos B_2} \right)^{0.5}. \quad \text{(3)}
\]

If we use the law of sine to find \( R_{21} \) and \( R_{32} \) and substitute their values in (3), we have, for long range sources, where \( \cos B_{32} \approx \cos B_2 \approx \cos B_{21} \)

\[
R_2 \approx \frac{l_1 + l_2 + \cos B_2}{\cos(B_{32} - B_{21})}. \quad \text{(4)}
\]

With both wavefront curvature and triangulation, we are measuring the difference in bearings between the forward and after array pairs in order to determine range.

III. PASSIVE RANGE ESTIMATION USING DUAL FOCUSED BEAMFORMERS

From Section II, the wavefront-curvature method and triangulation method are fundamentally similar in using time-delay estimation and bearing information.

The common rule of thumb for the approximate distance at which the far-field approximation begins to be valid is \( R = 2L^2/\lambda \), where \( R \) is a distance from an arbitrary array origin, \( L \) is the largest array dimension and \( \lambda \) is an operating wavelength \([8], [9]\). Because the whole area in this paper is included in near field, we use the focused beamformer. We explain the process of the beamformer in Section IV, and analyze and compare the simulation results.

The array consists two subarrays (left and right) from the center of the array as Fig. 3. Because we assume that we already know the direction of a target, we make a virtual line from the array to the direction of a target and divided the line into \( D \) steps \((1 \text{ step} = l)\). Passive ranging is not necessary for the nearest area from array. If the variable “\( k \)” is the interval from array to initial searching point, then, the area to form the beam is from \( k \) to \((D - 1)l + k\) and the beam is formed at each step. The output powers of left and right beamformers at the same step are as follows:

\[
E_L(d) = G_L(d) \cdot P(d) \quad d = 1, \ldots, D \quad \text{(5a)}
\]
\[
E_R(d) = G_R(d) \cdot P(d) \quad d = 1, \ldots, D \quad \text{(5b)}
\]

where \( G(\bullet) \) is a beamformer gain and \( d \) denotes a step. \( P(d) \) is an output power of sound source. When the beam is resolved as much as the number of steps that included at the direction, one direction can consist many steps, that is, the direction of the sound source at a main-lobe has some steps whose number depends on the range. The longer the range is, the more the number is. And the closer the direction is to end-fire of the array, the more the number is.
The estimated range \( E_{LR} \) is calculated as follows by summing the pattern of direction of sound source:

\[
E_{LR}(d) = E_L(\theta_\alpha(d_L)) + E_R(\theta_\beta(d_R)), \quad 1 \leq d_L \text{ or } d_R \leq D
\]  

(6)

where \( \theta_\alpha \) and \( \theta_\beta \) are the direction of maximum power from left and right beamformers, \( d_L \) and \( d_R \) consist several steps included in \( \theta \) according to the direction and the range. So, the more far the beamforming is, the closer \( d_L \) and \( d_R \) are. \( E_L(\theta_\alpha) \) and \( E_R(\theta_\beta) \) are the output of left and right beamformer at maximum output power. Because the pattern of \( E_{LR} \) actually includes many steps in one direction (1°), an interpolation is used to get a proper function on a regular one. The last pattern given by the interpolation using Lagrange’s formula [10] is

\[
E_T(d) = E_{LR}(d)L_0(d) + E_{LR}(d)L_4(d) + \cdots + E_{LR}(d)L_\eta(d)
\]  

(7)

where \( L_\eta(d) \) is a \( \eta \)th interpolation polynomial equation [10]. After the interpolation, the estimated range derived by a maximum value of the last pattern is

\[
E_T(d_m) = \max\{E_T(d)\}.
\]  

(8)

From (8), the step having the maximum power value, \( d_m \) is derived. It causes the next and final step (9) for the estimated range from the center of the array to the sound source

\[
R_{\text{est}} = k + d_m \cdot l
\]  

(9)

where \( k \) is set in advance and \( l \) is the distance of a step.

IV. SIMULATION AND RESULTS

The number of hydrophones is 100 and the spacing between hydrophones is 3.75 m, so, the length of the array is 371.25 m. The frequency of sound source is 200 Hz. In this paper, we used the simplified multipath underwater channel model which assumes uniform depth and constant sound velocity [7]. Fig. 4 compares the estimated range in detecting the sound source with a wavefront-curvature method and the proposed dual focused beamformer technique. The mark ‘+’ for wavefront curvature, ‘o’ for dual beamformers and ‘*’ for true range are used to compare. It is assumed that a sound source is about 1.8 km distance and the direction is various from 0° to 60° at 15° intervals. From the results, we are sure that the proposed method is more stable than the previous method in detecting a target. Fig. 5 shows the output of single and dual focused beamformer. To get this result, we used a 1) single; 2) dual focused beamformer for two targets which are 1.47 km (source 1) and 3.22 km (source 2) at 43°. The patterns resulted from the summation of left and right beamformer outputs. When two sources exist in the same direction, the single beamformer can detect only one source that keeps maximum array output power. The result shows that resolution of the proposed method is higher than the single beamformer. This is the effect of the difference of two beamformers’ steering direction. Fig. 6 shows ambiguous area in target detection (AATD) at the area of interest, according to direction and range. In the figure, the center point for abscissa corresponds to the array center. AATD is \(-3 \) dB width from maximum power point in Fig. 5. This is affected by the source direction and range, and means the probability of ranging error. We can see that AATD increases at far field or end fire region in Fig. 6. The reason is that the beamformer has the broad beamwidth at far field or end fire region.
V. CONCLUSIONS

In this paper, a passive-range estimation method using dual focused beamformer was proposed and compared with previous wavefront-curvature method. The proposed method remarkably decreased the range-estimation error under a multisource environment and was analyzed by a comparison with the single beamformer.

To minimize an ambiguous area in the target-range estimation, more research about various beamformer architecture and multitarget experiments is needed. In future work, we will apply the distorted array like water-pulley model and the estimated model using the Kalman filter for a real situation.

REFERENCES


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