

Analysis of array gain degradation of near-field passive synthetic aperture method

Rui Duan^{a,b,c,*}, Kunde Yang^{a,b}, Zhixiong Lei^{a,b}

^a School of Marine Science and Technology, Northwestern Polytechnical University, Xi'an, Shaanxi 710072, China

^b Key Laboratory of Ocean Acoustics and Sensing (Northwestern Polytechnical University), Ministry of Industry and Information Technology, Xi'an, Shaanxi 710072, China

^c State Key Laboratory of Acoustics, Institute of Acoustics, Chinese Academy of Sciences, Beijing 100190, China

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ABSTRACT

A previous study established a method for localizing low-frequency acoustic sources in the near field using a passive synthetic aperture method [Lei et al., 2015]. Sound fields sampled at different positions (sub-apertures) and times are synthesized to improve the spatial resolution of conventional beamforming. However, in practice, the position errors of the sub-apertures could significantly degrade the performance of the method. In this work, the analysis is extended to measure the sensitivity of the method to the position errors of the sub-apertures using array gain degradation (AGD). First, the position errors in three directions are assumed to be independent and obey the same Gaussian distribution to present a simple expression for AGD. Then, the analysis is generalized to the case of different Gaussian distributions to obtain an expression with explicit physical meaning. The two expressions are verified by simulations and partially validated using the experimental data sampled by a planar array in a semi-anechoic chamber. Finally, the results of localizing the sources on a transformer in a 750 kV substation are revealed.

1. Introduction

Localizing low-frequency sources acoustically in the near field using a small array is an important issue in signal processing. Conventional beamforming is robust; however, its spatial resolution is poor. High-resolution methods such as MUSIC [1,2], ESPRIT [3], and cumulant [4], are sensitive to the position errors of sensors and/or the correlated interference [5,6]. The passive synthetic aperture method creates a large virtual aperture using the motion of an array. Previous studies introduced algorithms [7–11] as synthetic aperture processors, which are applied to far-field signals (plane wave model). The uniform motion of the array causing the Doppler shift is included in the received signal model.

The near-field case differs from the far-field case in two important aspects. First, a spherical wave, rather than a plane wave, is received by the array. Thus, sampling the sound field with a moving array would cause different Doppler shifts for different microphones. The moving array complicates the coherent synthesis. Secondly, the sound suffers minimally from environmental effects when it propagates within a short distance. Thus, the spatial coherence length is usually long. Based on these properties, a near-field passive synthetic aperture (NPSA)

method for arbitrary arrays is proposed and verified using the experimental data in our previous study [12]. The satisfactory performance of the method is a result of the introduction of a reference microphone. The signals recorded by the reference microphone are used to synthesize the sound fields sampled at different positions and times to improve the spatial resolution of the conventional beamforming.

According to the NPSA theory, under ideal conditions, sub-apertures can form a large synthetic aperture and thus have a high resolution. However, in practice, every recording location has position errors. The accumulation of these errors can significantly degrade the performance of an NPSA method. This study investigates the sensitivity of NPSA with respect to the position errors of the sub-apertures through array gain degradation (AGD), which is defined as the correlation coefficient between the nominal array manifold at the source location and the actual signal vector. The mismatch between the predicted and real positions of the sub-apertures reduces the correlation, resulting in a small AGD. To provide a basis for the design of the application of the NPSA method, the approximate AGD expressions are presented and the physical meanings of the parameters in these expressions are explained in this work.

Our previous work [12] demonstrated that an NPSA method

* Corresponding author at: School of Marine Science and Technology, Northwestern Polytechnical University, Xi'an, Shaanxi 710072, China.

E-mail addresses: duanrui@nwpu.edu.cn (R. Duan), ykdzym@nwpu.edu.cn (K. Yang).

improved the array resolution using the experimental data in a semi-anechoic chamber. In this paper, our latest work is presented using data from two other experiments in the same semi-anechoic chamber. More importantly, the dependence of the AGD on the parameters indicated by the expressions is validated from the experimental data. Furthermore, considering that some technologies based on acoustic sensing have been widely applied in visualizing a spatial sound field [13–17] with a single sensor or an array, this work uses NPSA to localize the acoustic sources on a transformer in a 750 kV substation.

This paper is organized as follows. The NPSA method is described in Section 2, followed by the derivation of the AGD in Section 3. Section 4 is a simulation verification of the expressions for the AGD, and Section 5 presents the experimental results. The conclusions are provided in Section 6.

2. Theory of NPSA

We begin with the background material on NPSA. This background theory is necessary for this manuscript and for the derivation of the self-contained AGD. To extend the array aperture, an array is used to sample the sound field at discrete locations. A reference microphone is held stationary during the entire data collection to enable coherent processing.

Fig. 1 displays one near-field, narrowband source observed by an arbitrary array of N microphones with positions relative to the center of the array $\mathbf{r}_n = (x_n, y_n, z_n)$ for $n = 1, 2, \dots, N$. The signal received by the n th microphone $x_n(t)$ is expressed as

$$x_n(t) = \frac{A(t - |\mathbf{r}_n - \mathbf{r}_0|/c_0)}{|\mathbf{r}_n - \mathbf{r}_0|} e^{i(\omega t - k|\mathbf{r}_n - \mathbf{r}_0| + \varphi(t - |\mathbf{r}_n - \mathbf{r}_0|/c_0))} + w_n(t) \text{ for } 1 \leq n \leq N, \quad (1)$$

where $A(t)$ and $\varphi(t)$ are the amplitude and phase of the source, respectively. Here, $|\cdot|$ denotes the vector length, $\mathbf{r}_0$ is the source location relative to the center of the array, ω is the signal angular frequency, k is the signal wavenumber, c_0 is the sound speed in the air, and $w_n(t)$ is the independent zero-mean, Gaussian random noise. The NPSA method is designed to localize low-frequency sources, which are virtually stationary; as such, the phase $\varphi(t)$ is assumed to be a slow variable. A reference microphone is introduced to sample the sound field at the same time and eliminate the influence of this variable. The received signal is

$$x_{ref}(t) = \frac{A(t - |\mathbf{r}_{ref} - \mathbf{r}_0|/c_0)}{|\mathbf{r}_{ref} - \mathbf{r}_0|} e^{i(\omega t - k|\mathbf{r}_{ref} - \mathbf{r}_0| + \varphi(t - |\mathbf{r}_{ref} - \mathbf{r}_0|/c_0))} + w_{ref}(t) \text{ for } 1 \leq n \leq N, \quad (2)$$

where $\mathbf{r}_{ref}$ is the location of the reference microphone relative to the center of the array. The noise term is ignored because the signal-to-

noise ratio (SNR) is typically high in the near-field case. Furthermore, the amplitude term in Eq. (2) is eliminated by normalization of the received signal at every microphone. After subtracting the phase of the reference microphone, Eq. (1) becomes

$$x_n(t) = e^{i(-k|\mathbf{r}_n - \mathbf{r}_0| + k|\mathbf{r}_{ref} - \mathbf{r}_0| + \varphi(t - |\mathbf{r}_n - \mathbf{r}_0|/c_0) - \varphi(t - |\mathbf{r}_{ref} - \mathbf{r}_0|/c_0))} \text{ for } 1 \leq n \leq N. \quad (3)$$

The source phase $\varphi(t)$ is a slow variable; as such, the term $\varphi(t - |\mathbf{r}_n - \mathbf{r}_0|/c_0) - \varphi(t - |\mathbf{r}_{ref} - \mathbf{r}_0|/c_0)$ is close to zero. Therefore,

$$x_n(t) \approx e^{i(-k|\mathbf{r}_n - \mathbf{r}_0| + k|\mathbf{r}_{ref} - \mathbf{r}_0|)}. \quad (4)$$

Note that $x_n(t)$ is independent of time in this equation.

Secondly, assuming that the sound field is sampled by the array at M discrete positions and only the translational motion of the array is performed over the measurement, the signals received by all the sub-apertures can be expressed according to Eq. (4) as

$$x_{n,m}(t) = e^{i(-k|\mathbf{r}_n + \mathbf{r}_m^{sub} - \mathbf{r}_0| + k|\mathbf{r}_{ref} - \mathbf{r}_0|)} \text{ for } 1 \leq n \leq N \text{ and } 1 \leq m \leq M, \quad (5)$$

where $\mathbf{r}_m^{sub}$ is the position of the center of the m th sub-aperture relative to the center of the first sub-aperture. Eq. (5) indicates that all the microphones possess a common phase term $k|\mathbf{r}_{ref} - \mathbf{r}_0|$, which has no effect on the beamforming. Therefore, all sub-apertures sample the sound field simultaneously and a large virtual aperture is formed. The reference microphone is placed near the center of the synthetic aperture to decrease the total distance between the reference microphone and the sub-apertures and thus decrease the errors due to the approximation of Eq. (4).

3. Expressions for AGD

For a stationary array, the phase errors and positioning uncertainties of the microphones are errors typically analyzed in the beamforming. In addition to these two errors, the imprecise positioning of the sub-apertures can significantly reduce the performance of the NPSA method. In this section, AGD is derived to measure the sensitivity of the NPSA to this error.

After ignoring the common phase term in Eq. (5), the nominal signal vector $\mathbf{V}$ is

$$\mathbf{V} = [x_{1,1}, x_{2,1}, \dots, x_{n,m}, \dots, x_{N,M}]^T, \quad (6)$$

where $x_{n,m} = e^{i(-k|\mathbf{r}_n + \mathbf{r}_m^{sub} - \mathbf{r}_0|)}$. The imprecise position of the sub-aperture causes the same position errors for all the microphones on this sub-aperture; hence, the actual signal vector $\mathbf{D}$ is

$$\mathbf{D} = [x_{1,1}^a, x_{2,1}^a, \dots, x_{n,m}^a, \dots, x_{N,M}^a]^T, \quad (7)$$

where $x_{n,m}^a = e^{i(-k|\mathbf{r}_n + \mathbf{r}_m^{sub} + \Delta\mathbf{r}_m - \mathbf{r}_0|)}$ and $\Delta\mathbf{r}_m = (\Delta x_m, \Delta y_m, \Delta z_m)$. The vector $\Delta\mathbf{r}_m$ is the position error of the m th sub-aperture. The AGD [18] is defined as

$$\text{AGD} = E \left(\frac{(\mathbf{V}^H \mathbf{D})(\mathbf{V}^H \mathbf{D})^H}{(\mathbf{V}^H \mathbf{V})(\mathbf{D}^H \mathbf{D})} \right), \quad (8)$$

where $E(\cdot)$ and $(\cdot)^H$ denote the expectation and conjugate transpose operators, respectively. The AGD reflects the spectral output degradation at the source location under a high SNR condition [18]. The maximum value of AGD is one, which means that the nominal signal vector is the same as the actual signal vector and the localization result is accurate. Substituting Eqs. (6) and (7) into $\mathbf{V}^H \mathbf{D}$ and defining $\mathbf{r}_{m,n} = \mathbf{r}_n + \mathbf{r}_m^{sub} - \mathbf{r}_0$, which is the vector from the source to the n th microphone of the m th sub-aperture, yields

$$\mathbf{V}^H \mathbf{D} = \sum_{m=1}^M \sum_{n=1}^N e^{i(k|\mathbf{r}_{m,n}| - k|\mathbf{r}_{m,n} + \Delta\mathbf{r}_m|)}. \quad (9)$$

According to the Taylor expansion, $|\mathbf{r}_{m,n} + \Delta\mathbf{r}_m|$ can be approximated by $|\mathbf{r}_{m,n}| + \nabla|\mathbf{r}_{m,n}| \cdot \Delta\mathbf{r}_m$, where ∇ denotes the gradient, because $|\Delta\mathbf{r}_m|$ is small. Substituting Eq. (9) in Eq. (8) and applying this approximation

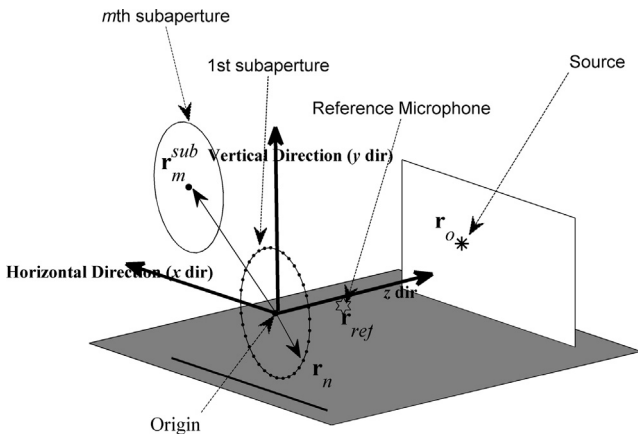


Fig. 1. Illustration of NPSA.

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