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Signal reconstruction of moving sound sources with a fixed microphone array

Institute of Sound and Vibration Research, Hefei University of Technology, 193 Tunxi Road, Hefei 230009, People's Republic of China

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ABSTRACT

The time-domain signals of noise sources are of great importance to acoustic design of machinery. One available way to obtain source signals is to reconstruct them from the acoustic signals measured by a microphone array. However, when sources are moving, the acoustic signals sampled by a fixed microphone array are distorted in amplitude and frequency due to the Doppler effect, leading to that the reconstructed source signals may not reflect the real information of sources. This paper first presents a timewavenumber-domain method to eliminate the Doppler effect in the measured acoustic signals, in which no priori information on the number, frequencies and locations of sources is needed. Then the acoustic signals after eliminating the Doppler effect are used as the input of the passive time reversal method to reconstruct source signals. Numerical simulation and experimental results show that the proposed method can effectively eliminate the Doppler effect and accurately reconstruct the source signals even when the sources emit transient signals. It is also found that the proposed method can work stably when the sources are moving at high speeds if increasing the array aperture and in the situation of low signal-to-noise ratio.

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1. Introduction

In the acoustic design of machinery, obtaining the time-domain signals of noise sources is of great importance to help reveal the noise generation mechanism, analyze acoustic fatigue and realize active control of sources. One available way to obtain source signals is to reconstruct them from the acoustic signals measured by a microphone array. Currently, several different methods, such as time-domain nearfield acoustic holography [\[1–9\]](#page--1-0), time-domain beamforming [\[10–13\]](#page--1-0) and time reversal (TR) method [\[14–17\],](#page--1-0) have been used to realize this reconstruction. The present paper only focuses on the TR method for the source signal reconstruction due to the fact that the TR method can work stably even when the measured acoustic signals have a very low signal-to-noise ratio (SNR) [\[14\].](#page--1-0) The TR method has been employed to reconstruct the source signals in a semi-anechoic room [\[14\],](#page--1-0) in solid media [\[15,16\]](#page--1-0) and in an environment with flow [\[17\]](#page--1-0). However, in these studies, the sources are confined to be static.

For a moving noise source, the amplitude and frequency of the acoustic signal measured with a fixed microphone will be distorted due to the Doppler effect [\[18\].](#page--1-0) Garnier et al. [\[19\]](#page--1-0) pointed out that the Doppler effect allows the time-reversal mirror to record information with a bandwidth that is larger than the source bandwidth, and hence the spatial resolution of the TR focusing can be enhanced especially when the source is supersonic. But they did not study the performance of the TR method

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[⇑] Corresponding author. E-mail address: cxbi@hfut.edu.cn (C.-X. Bi).

to reconstruct the source signal. When the source is moving, the reconstructed source signal by the TR method may not reflect the real source information due to the Doppler effect. Therefore, it comes to the idea to eliminate the Doppler effect before conducting the reconstruction of source signal. Some frequency-domain methods have been proposed to eliminate the Doppler effect, such as short-time Fourier transform-based method [\[20\],](#page--1-0) disturbance-oriented dynamic signal resampling method [\[21,22\]](#page--1-0) and moving frame technique [\[23,24\].](#page--1-0) These methods eliminate the Doppler effect with different processes and have been successfully applied to different situations. But they are not suitable when the moving source emits a transient signal due to the side band overlapping $[23]$, thus limiting their application ranges. Generally for the transient signal, it is appropriate to eliminate the Doppler effect in the time domain, and some time-domain de-Dopplerization methods [\[25–31\]](#page--1-0) have been developed. These time-domain de-Dopplerization methods are similar in their principles since they involve the time-domain re-sampling based on interpolation and have been successfully applied to acoustic imaging of aircraft flyover noise and railway noise, wayside acoustic diagnosis of defective train bearings and pass-by noise radiated by moving vehicles. However, it should be noted that all of these time-domain methods must be re-iterated for each source when there exists more than one source.

This paper intends to propose a time-wavenumber-domain method to eliminate the Doppler effect in the measured acoustic signals. This method doesn't require any interpolation operation and any priori information on the number, frequencies and locations of sound sources but only requires a priori knowledge of the source velocity. After eliminating the Doppler effect, the restored acoustic signals are further used to reconstruct the time-domain source signals by the passive TR method [\[14,17\]](#page--1-0). This paper is organized as follows. In Section 2, the theories of the Doppler effect elimination method and the passive TR method are given. [Section 3](#page--1-0) presents the numerical simulations to validate the effectiveness of the proposed method, and the experimental validation is performed in [Section 4](#page--1-0). Finally, conclusions are drawn in [Section 5](#page--1-0).

2. Theory

2.1. Doppler effect elimination

Consider that a point source moving with a constant velocity ν along the positive x direction and a microphone fixed in space, as depicted in Fig. 1. Two coordinates are employed in this measurement system, one is the measurement coordinate, (x_m, y_m, z_m) , which is fixed in space, and the other is the source coordinate, (x_s, y_s, z_s) , in which the frame of reference is moving with the same velocity as the source. Pressures on these two coordinates are denoted as $p_m(x_m, y_m, z_m, t)$ and $p_s(x_s, y_s, z_s, t)$, respectively, and one can obtain [\[23\]](#page--1-0)

$$
p_m(x_m, y_m, z_m, t) = p_s(x_s, y_s, z_s, t). \tag{1}
$$

Assuming that the two coordinates are in parallel and their origins are at the same location at $t = 0$. Then, at arbitrary time t,

$$
\begin{cases}\n x_m = x_s + vt \\
y_m = y_s \\
z_m = z_s\n\end{cases} (2)
$$

Substituting Eq. (2) into Eq. (1) yields

 $p_m(x_m, y_m, z_m, t) = p_s(x_m - vt, y_m, z_m, t).$ (3)

The spatial Fourier transform of Eq. (3) is

$$
p_m(k_x, y_m, z_m, t) = \int_{-\infty}^{\infty} p_s(x_m - \nu t, y_m, z_m, t) e^{-jk_x x_m} dx_m = e^{-jk_x \nu t} \times \int_{-\infty}^{\infty} p_s(x_m - \nu t, y_m, z_m, t) e^{-jk_x (x_m - \nu t)} dx_m
$$

= $e^{-jk_x \nu t} \times p_s(k_x, y_m, z_m, t).$ (4)

Fig. 1. Coordinate systems and their relative motion.

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