Identification of Moving Acoustic Sources from Pass-by Noise

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Abstract: A time domain technique based on the finite difference method and regularization is developed for the recovery of the stationary source characteristics from the distorted signal measured at a fixed point. The sources are assumed to be point sources with stationary frequency characteristics, and move along a line at a constant subsonic speed. A source signal recovery method is developed using multiple time origins determined by continuous wavelet transform.

1. INTRODUCTION

Moving noise sources, such as the automobiles and trains, show the noise characteristics that change during their operation. If one try to measure such noise with microphones attached to the moving source, he may encounter a strong wind noise induced by the microphone itself, which masks the sound signal from the source. On the other hand if the sound from the moving source is to be measured at a fixed observation point, i.e. the pass-by noise measurement, the signal is greatly distorted by the motion of the source, hence what is measured is not the source characteristics but only the pass-by characteristics. In a previous work we tried to restore the source characteristics of a point noise source moving at a constant velocity from its pass-by noise. However when the noise sources consist of several discrete point sources, as the case of the train, the time origins for each sources must be determined separately. In this study a method is proposed for the determination of the time origins.

2. SOUND Emitted BY TWO MOVING SOURCES AT A CONSTANT VELOCITY

Let's consider the sound sources moving along a straight path. The acoustic field emitted from a point source of strength \( q(t) \) moving at a constant velocity can be expressed by a non-stationary wave equation as [1]

\[
\nabla^2 p - \frac{1}{c^2} \frac{\partial^2 p}{\partial t^2} = \frac{\partial}{\partial t} \sum_{j=1}^{N} q_j(t) \delta(x-x_j-Vt) \delta(y) \delta(z)
\]

(1)

The acoustic pressure \( p_j(t) \) from a point source of strength \( q_j(t) \) moving at a constant velocity, which is measured at the observer position, can be represented as a differential equation of \( q(t) \) as[1,2]

\[
p_j = \frac{q_j'[t-R_j/c]}{4\pi R_j (1-M \cos \theta_j)^2} + \frac{q_j[t-R_j/c]V(\cos \theta_j - M)}{4\pi R_j^2 (1-M \cos \theta_j)^3},
\]

(2)

where \( R_j \)'s and \( \theta_j \)'s are as given in reference [1].

3. RESTORATION OF THE SOURCE SPECTRUM

In the previous work of present authors for a single source solving the eq.(3) as a differential equation with respect to \( q(t) \) using finite difference method (FDM) restored \( q(t) \). The solution method requires the information about the parameters such as the velocity and the time when each of the sources approaches nearest to the observation point, defined as the time origin. Tikhonov regularization technique is applied to suppress the numerical instability near time origin. The velocity of the sources can be estimated from the slope of the phase difference between the signals of two microphones [2]. In case of a single source the time origin can be easily determined for a harmonic source since the instantaneous frequency at the time origin is the mean of two asymptotic values of \( f(1-M) \) and \( f(1+M) \). For general source a dominant harmonic component can be traced using Kalman filter. However for multiple sources it is quite difficult to find the time origin of each source. In order to determine the time origins a sort of continuous wavelet transform is applied to the acoustic pressure signal. The signal from a virtual harmonic source moving at the same velocity is used as the wavelet as
\[ \phi(\omega, \tau) = \int_{-\infty}^{\infty} p(t) g^*(t; \omega, \tau) dt \]  

where \( g(t; \omega, \tau) \) is the acoustic field from a harmonic source located at \( Vt \) behind the reference position.

4. NUMERICAL SIMULATION AND RESULTS.

In numerical simulation the pass-by sound signal is generated from source signals using the equation (3). The source signal restoration process is applied to the simulated pass-by signal and the result is compared with the original source characteristics.

Fig.1 shows a time signal of pass-by sound from a moving object with two separate sources, each of which has two harmonic components. Fig.2 (a) and (b) show its time-frequency map and the integration of \( \phi(\omega, \tau) \) with respect to \( \omega \) respectively. In this figure the locations of two sources are clearly distinguished.

![FIGURE 1. Pass-by sound signal from an object containing two point sources with harmonic components.](image1)

![FIGURE 2. (a) Time-frequency map of the pass-by sound signal which shows the harmonic components of each source. (b) Integration of the time-frequency map with respect to the frequency shows two sources at different locations.](image2)

As discussed in reference [2], there is no apparent changes in the magnitude spectra though the relative estimation error in time origin, given as the ratio of the error with respect to the \( r/c \), the shortest propagation time from the line of motion to the observation point, reaches up to 50%. However the phase spectrum, which is significant in the source position estimation problem, may be affected by the time origin error.

5. CONCLUSION

The source characteristics of multiple stationary noise sources running at a constant speed is estimated from the signal measured at a fixed position. To solve the inhomogeneous equation with respect to the source signals, a time domain estimation scheme is proposed using the finite difference method. A continuous wavelet transform is used in order to determine the time origins of the sources. The original source characteristics are well restored in the numerical simulation even for the low signal to noise ratios.

REFERENCES