

Kiyoshi Nagakura

Railway Technical Research Institute

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Summary

This chapter describes some measuring methods for the identification and ranking of noise source that are of benefit in noise control projects. Sound intensity measurement and directional measuring devices such as the mirror–microphone system and microphone array are introduced and their principles and applications are described.

15.1 Sound Intensity Measurement

Every noise control project starts with the identification and ranking of the noise sources. Several methods have been proposed for the purpose and have proved to be useful and widely utilized. In this chapter, sound intensity measurement and directional measuring devices such as the mirror–microphone system and microphone array are introduced and their principles and applications are described. Other useful measurements, such as acoustic holography method [1,2] and spatial transformation of sound fields [3], are described in the literature.

15.1.1 Theoretical Background

Sound intensity is a measure of the magnitude and direction of the flow of sound energy. The instantaneous intensity vector, $\mathbf{I}(t)$, is given by the product of the instantaneous sound pressure, $p(t)$, and the corresponding particle velocity, $\mathbf{u}(t)$, that is, $\mathbf{I}(t) = p(t)\mathbf{u}(t)$.

In practice, the time-averaged intensity, $\bar{\mathbf{I}}$, is more important, and is given by the equation:

$$\bar{\mathbf{I}} = \lim_{T \rightarrow \infty} \frac{1}{T} \int_{-T/2}^{T/2} p(t)\mathbf{u}(t)dt \quad (15.1)$$

The intensity vector denotes the net rate of flow of energy per unit area (watts/m^2). Thus, the acoustic power, W , of the source located in a closed surface, S , is given by the integral of the intensity passing through the surface, S , as

$$W = \iint_S \bar{\mathbf{I}} \cdot d\mathbf{S} \quad (15.2)$$

Equation 15.2 indicates that the measurement of sound intensity over a surface enclosing a source enables the estimation of its sound power, which shows the usefulness of the sound intensity concept.

15.1.2 Measurement Method

The principle of intensity measurement systems in commercial production employs two closely spaced pressure microphones [4,5], as shown in Figure 15.1.

The particle velocity, $u_r(t)$, in a particular direction, r , can be approximated by integrating over time the difference of sound pressures at two points separated by a distance Δr in that direction:

$$u_r(t) = -\frac{1}{\rho_0} \int_{-\infty}^t \frac{p_2(\tau) - p_1(\tau)}{\Delta r} d\tau \quad (15.3)$$

where p_1 and p_2 are the sound pressure signals from the two microphones. The sound pressure at the center of two microphones is approximated by

$$p(t) = \frac{p_1(t) + p_2(t)}{2} \quad (15.4)$$

Thus, the intensity in the direction r can be calculated as

$$I_r(t) = -\frac{1}{2\rho_0\Delta r} [p_1(t) + p_2(t)] \int_{-\infty}^t [p_2(\tau) - p_1(\tau)] d\tau \quad (15.5)$$

Some commercial intensity analyzers use Equation 15.5 to measure the intensity. Another type of analyzer uses the equation in the frequency domain:

$$I_r(\omega) = -\frac{\text{Im}[G_{12}]}{\omega\rho_0\Delta r} \quad (15.6)$$

where G_{12} is the cross spectrum between the two microphone signals. Equation 15.6 makes it possible to calculate sound intensity with a dual-channel fast fourier transform (FFT) analyzer.

15.1.3 Errors in Measurement of Sound Intensity

The principal systematic error of the two-microphone method is due to the approximation of the pressure gradient by a finite pressure difference. When the incident sound is a plane wave, the ratio of the measured intensity, $\hat{I}_r$, and the true intensity, I_r , is given by

$$\hat{I}_r/I_r = \frac{\sin(k\Delta r \cos \theta)}{k\Delta r \cos \theta} \quad (15.7)$$

where the angle θ is as defined in Figure 15.1 and k is the wave number. Equation 15.7 indicates that the upper frequency limit is inversely proportional to the distance between the microphones.

Another serious error is caused by the phase mismatch between the two measurement channels. In the calculation of intensity from Equation 15.5, the phase difference, φ , between the two microphone signals, p_1 and p_2 , is very important. Hence, the phase mismatch between the two measurement channels, $\Delta\varphi$, must be much smaller than φ . Since φ increases with frequency, this error is serious in lower frequencies. Other possible errors, such as in the sensitivity of microphones and random errors associated with a given finite averaging time, are usually less serious.

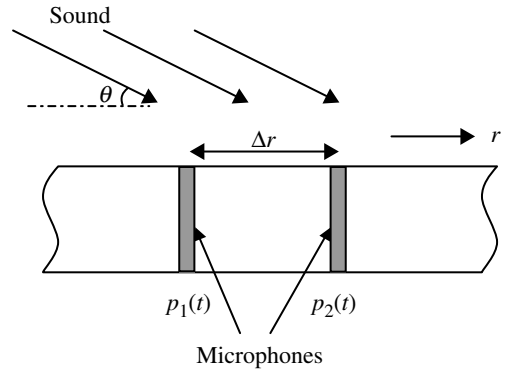


FIGURE 15.1 Microphone arrangement used to measure sound intensity.

15.1.4 Applications

One important application of sound intensity measurement is the determination of the sound power level using Equation 15.2. Furthermore, measurement of the intensity in the very near field of a source surface makes it possible to identify and rank the noise-sources. Plots of the sound intensity measured on a surface near a sound source are useful for investigating noise source distributions. Figure 15.2 shows sound intensity of noise from a wheel of a railway car. An intensity probe is located in the vicinity of the wheel and the normal component of sound intensity is measured by traversing the probe on a plane 100 mm away from the side surface of the wheel. These figures show a free vibration behavior of the wheel at each frequency; the wheel vibrates with one nodal diameter at 700 Hz and with three nodal diameters at 1150 Hz. Visualization by intensity vectors also gives valuable information about a noise source. Figure 15.3 shows the sound intensity vectors at each octave band measured in the vicinity of a railway car running at 120 km/h. These results suggest that the main radiator of rolling noise is the rail at the 500 Hz to 1 kHz band and the wheels at the 2 to 4 kHz band.

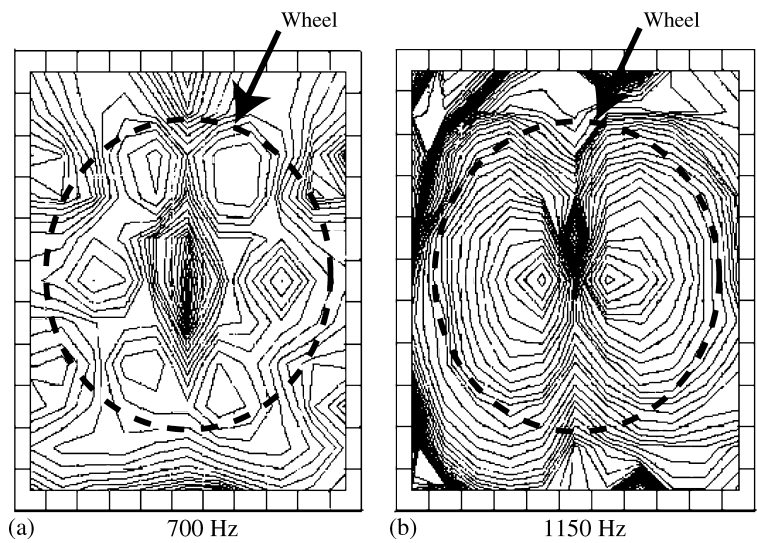


FIGURE 15.2 Measurements of the sound intensity radiated by a wheel of a railway car (1 dB contour).

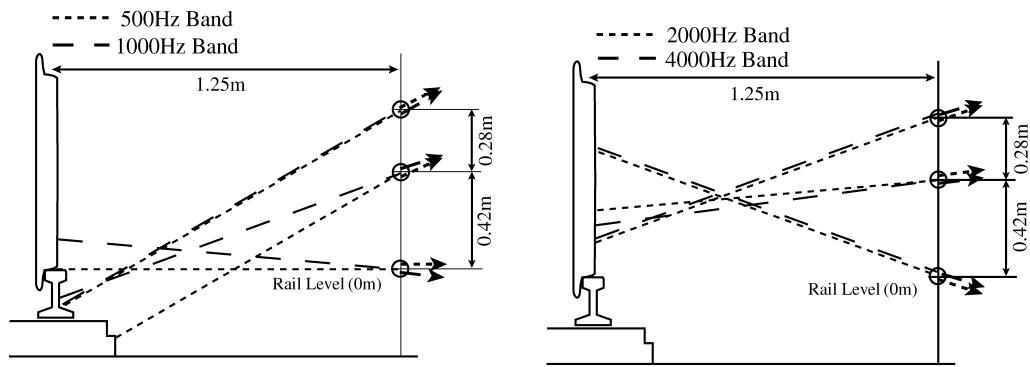


FIGURE 15.3 Sound intensity vectors measured in the vicinity of a railway car running at 120 km/h.

15.2 Mirror–Microphone System

15.2.1 Principle of Measurement

A mirror–microphone system consists of a reflector of elliptic or parabolic shape and an omnidirectional microphone located at its focus [6,7]. Figure 15.4 shows the layout of a reflector of elliptic shape, an omnidirectional microphone, and a noise source. Here, S and S' denote the front and back surfaces of the mirror, respectively; $P(\mathbf{r})$ denotes the pressure field on this configuration; $P_i(\mathbf{r})$ denotes the pressure field of free space; $\mathbf{r}_m$ is the position of the microphone; $\mathbf{r}$ is a point on the mirror surface. The normal, $\mathbf{n}_0$, directs toward the medium.

Using Green's theorem, the pressure at the microphone position $P(\mathbf{r}_m)$ is obtained by

$$P(\mathbf{r}_m) = P_i(\mathbf{r}_m) + \iint_{(s+s')} P(\mathbf{r}) \frac{\partial}{\partial \mathbf{n}_0} \times \left[\frac{e^{ikR_m}}{4\pi R_m} \right] d^2\mathbf{r} \quad (15.8)$$

where $k = 2\pi f/c_0$ is the wave number, f is the frequency of sound, c_0 is the speed of sound, and $R_m = |\mathbf{r} - \mathbf{r}_m|$ is the distance between the microphone and the mirror surface. If the wavelength is sufficiently smaller than the diameter of the reflector, the pressure field $P(\mathbf{r})$ is approximated by $2P_i(\mathbf{r})$ on the front surface, S , and by zero on the back surface, S' . In such a frequency range, the incident field term $P_i(\mathbf{r}_m)$ can be ignored. With these approximations, assuming that the noise source is a monopole type point source located at a position, $\mathbf{r}_s$, Equation 15.8 reduces to

$$P(\mathbf{r}_m) = -\frac{m(f)}{8\pi^2} \iint_s \frac{e^{ik(R_m+R_s)}}{R_m R_s} \left(ik - \frac{1}{R_m} \right) \times \cos \theta(\mathbf{r}) d^2\mathbf{r} \quad (15.9)$$

Here, $m(f)$ is the amplitude of the mass-flux rate of the source, $R_s = |\mathbf{r} - \mathbf{r}_s|$ is the distance between the sound source and the mirror surface and the angle $\theta(\mathbf{r})$ is defined in Figure 15.4. When the noise source is located at the far focus of the mirror, the sound pass length $R_m + R_s$ is constant with respect to $\mathbf{r}$, and a strong signal is obtained. As the noise source is moved away in the direction perpendicular to the mirror axis, the variance of the sound pass length, $R_m + R_s$, due to the position $\mathbf{r}$ increases, and thus the microphone signal drops off due to interference (see Figure 15.5, which we call the “directivity pattern”). The ratio of the peak level to the free field level at the microphone, G , is referred to as the “gain factor.” The spatial resolution of the mirror is characterized by the displacement of the mirror position, w , at which the microphone signal drops off by a given relative amount, such as 3 dB. The quantities G and w

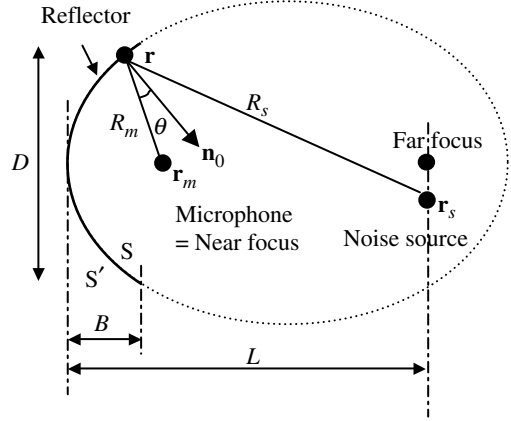


FIGURE 15.4 Layout of a reflector, microphone, and noise source.

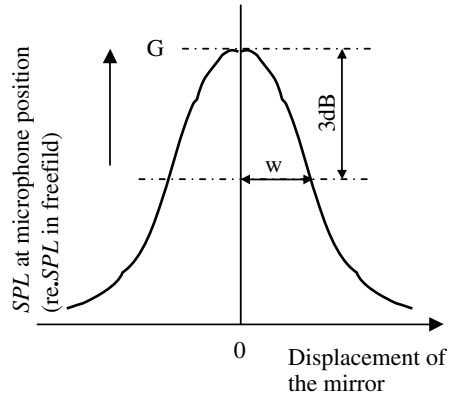


FIGURE 15.5 Directivity pattern of a mirror–microphone system.

can be related to the mirror geometries in Figure 15.4 by

$$G \approx 10 \log(CD^4/\lambda^2 B^2) \quad (C = \text{const.}) \tag{15.10}$$

$$w \propto \lambda L/D \tag{15.11}$$

The gain factor, G , increases with frequency at the rate of 6 dB per octave, and the spatial resolution, w , is inversely proportional to the frequency. The lower frequency limit is decided by the size of the mirror. On the other hand, there is no higher frequency limit, except for the capacity of an omnidirectional microphone itself. Thus, measurements with the mirror–microphone system are more suited to a scaled model test.

15.2.2 Applications

The mirror–microphone system has proved useful for identification of a noise source because of its directional property [8–10]. A scan of the source region produces a noise source map. It has an advantage in that the measurement is possible at a far field and it needs only one sensor, but has a disadvantage in that the measuring process is a time-consuming task.

Figure 15.6 shows an example of source maps of aerodynamic noise generated by a one-fifth scale high-speed train model, obtained from measurements by a mirror–microphone system, in a wind tunnel test. The surface of the car model is divided into several noise-source areas and the noise-source distribution in each area is measured by traversing the mirror–microphone system over the surface. The diameter and focal distance of the reflector are 1.7 and 3 m, respectively. Detailed maps of noise-source strength are obtained, which show that aerodynamic noise from high-speed trains is generated in relatively localized areas, namely, the local surface structures. The mirror–microphone system can be used for the measurement of the source distribution of a moving noise source. Figure 15.7 gives a time

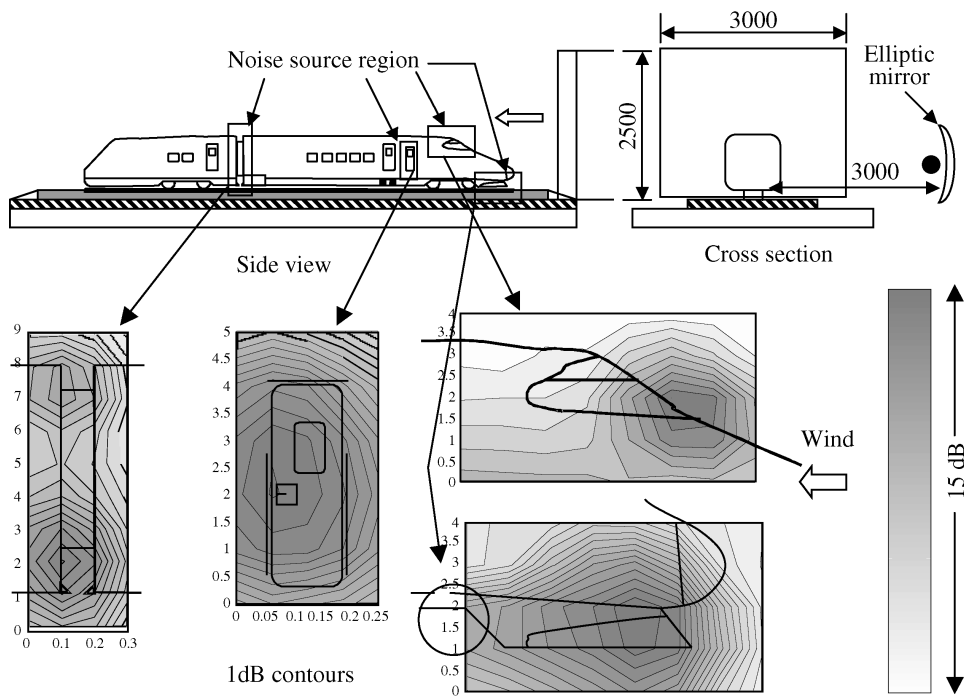


FIGURE 15.6 Noise-source distribution of a one-fifth scale Shinkansen car model in a wind tunnel test measured with an elliptic mirror–microphone system.

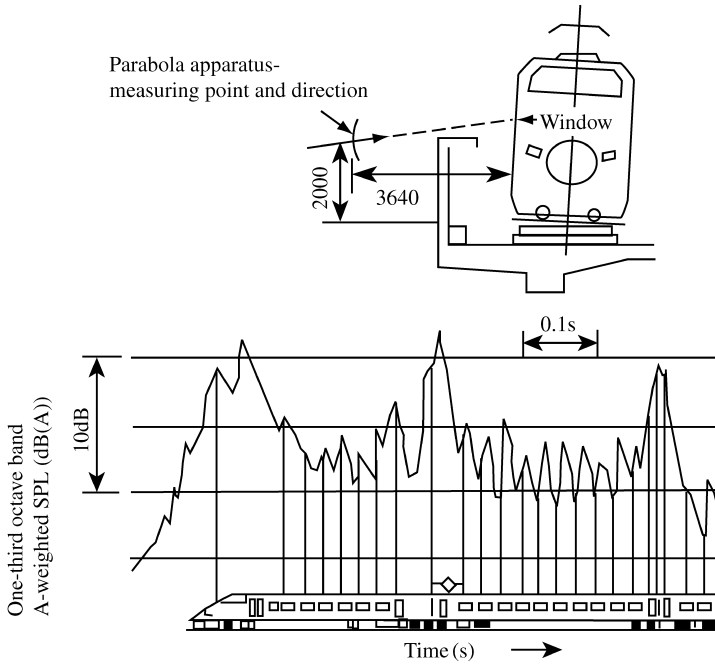


FIGURE 15.7 Time history of the A-weighted one-third octave band ($f_0 = 8$ kHz) sound pressure level measured with a parabolic mirror–microphone system ($D = 1$ m, train speed = 274 km/h).

history of noise from a high-speed train measured with a parabolic mirror–microphone system, the diameter of which is 1 m. Peaks of the time history correspond to pantographs, doors, gaps between cars and the step-up of windows, which shows that they are main noise sources.

15.3 Microphone Array

15.3.1 Principle of Microphone Array

A microphone array [11] consists of several microphones distributed spatially to measure an acoustic field. The time signals from each microphone are added, accounting for the time delay between sound sources and microphones, and a directional output signal can be obtained as a result. The algorithm is called “beamforming.” Now, consider M omnidirectional microphones distributed in a far field of noise sources. The output signal of the array focused to a particular location in the source region, $\mathbf{r}$, and $z(\mathbf{r}, t)$, is calculated as a sum of delayed and weighted signals of each microphone:

$$z(\mathbf{r}, t) = \sum_{m=1}^M w_m p_m(t - \Delta_m) \quad (15.12)$$

Here, $p_m(t)$ is the signal from the m th microphone, w_m is a weighting factor, and Δ_m is a time delay applied to signal of the m th microphone, as given by

$$\Delta_m = \frac{r_o - r_m}{c_0} \quad (15.13)$$

where r_o and r_m are the distances from the focus point to the reference point o and the m th microphone, respectively. When the focus location coincides with the source location, a strong signal is obtained (see Figure 15.8). If this process is repeated for various focus locations, $\mathbf{r}$, on the source surface, then a noise-source map can be obtained.

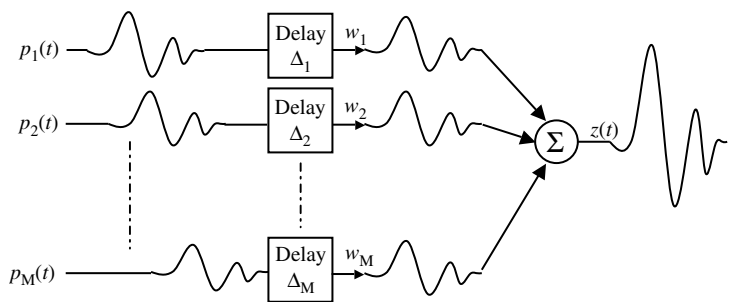


FIGURE 15.8 Principle of a microphone array. Individual time delays are chosen such that signals arriving from a given point will be added up coherently.

15.3.2 Array’s Directivity Pattern

The performance of a microphone array is characterized by the spatial resolution and signal-to-noise ratio. For simplicity, consider a linear array of $M = 2N + 1$ microphones spaced equally by d . When a harmonic plane wave is propagating with an incident angle θ , and weighting factors all equal $1/M$, the ratio of the output signal of the array to that of the center microphone is computed using

$$W(\theta) = \frac{1}{M} \frac{\sin((M/2)kd \sin \theta)}{\sin((1/2)kd \sin \theta)} \tag{15.14}$$

where k is the wave number. Figure 15.9 shows the directivity patterns for different values of the product kd based on Equation 15.14. The highest peak appears at $\theta = 0$, which we call a “main lobe,” and lower peaks also appear at some locations that are separate from a true source direction, which we call “side lobes.” The width of the main lobe decides the performance of the array to separate two closely lying sources (which we call spatial resolution), and the ratio of main lobe to side lobe decides the signal-to-noise ratio of the array. The spatial resolution improves as kd increases, that is, in proportion to the ratio of the array length to the wavelength. However, when $kd = 2\pi$, a peak of the same strength as the true source appears due to a spatial aliasing at $\theta = 90^\circ$, which occurs when $d > \lambda/2$, where λ is the wavelength. Thus, the acoustic frequency, f , is restricted by $f < c_0/2d$, to avoid aliasing.

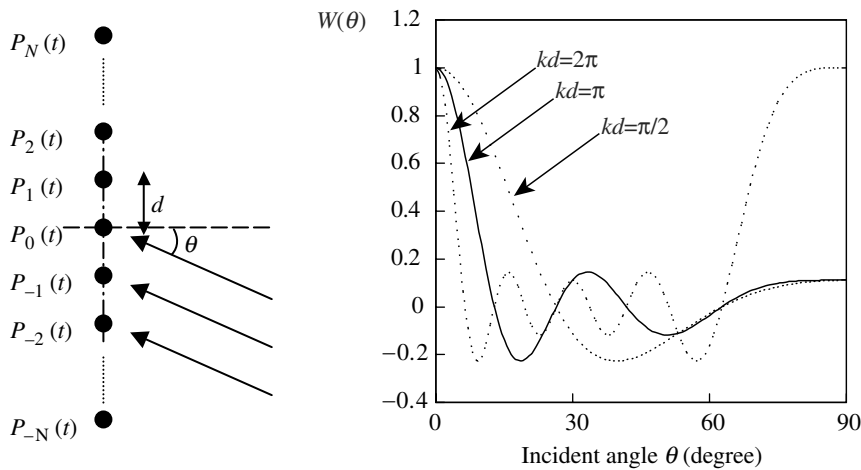


FIGURE 15.9 Directivity patterns of a linear array for different values of the product kd ($M = 2N + 1 = 9$).

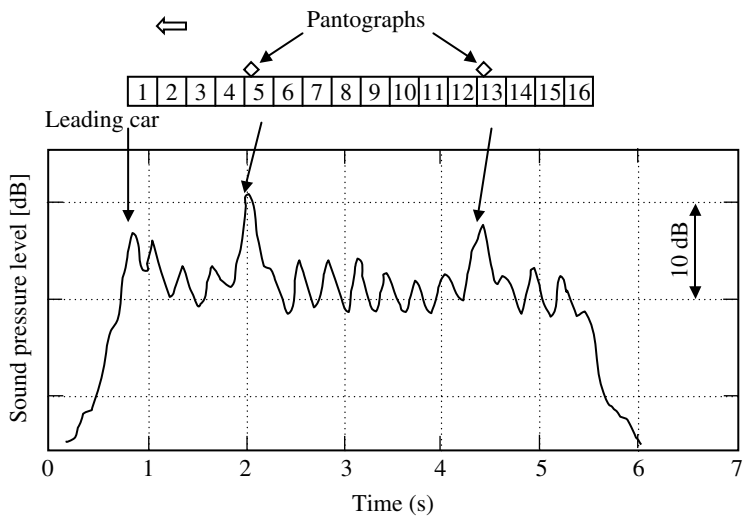


FIGURE 15.10 Time history of sound pressure level for a passing train measured with a linear microphone array located at a point 25 m away from a track (train velocity = 285 km/h).

In the above case of the linear array, the directivity exists only in the direction of the array (one-dimensional). If microphones are arranged in a two-dimensional plane, a two-dimensional directivity can be obtained. Recently, many microphone arrangements have been proposed that obtain better spatial resolution and to reduce side lobes [12–15].

15.3.3 Applications

Microphone arrays have been used for identification of the noise source in various situations, for example, in wind tunnel tests. Many actual examples can be found in published literature [10,16–18]. The measurement with a microphone array has the advantage of much shorter measuring time than that of a mirror–microphone system. Furthermore, the lower frequency limit is not so serious because the size of the apparatus can be easily extended.

Another fundamental example is given now. Nine microphones are arranged equally spaced by $d = \lambda/2$ for each one-third octave band, and their signals are summed without time delay. In this case, the array is focused to a fixed direction, perpendicular to the array axis. Figure 15.10 shows a time history of noise generated by a high-speed train, measured with a linear microphone array located at a point 25 m away from the track. It is found that pantographs, the leading car, and gaps between cars are the main noise sources in this example.

References

1. Ferris, H.G., Computation of far field radiation patterns by use of a general integral solution to the time independent scalar wave equation, *J. Acoust. Soc. Am.*, 41, 1967.
2. Maynard, J.D., Nearfield acoustic holography: theory of generalized holography and the development of NAH, *J. Acoust. Soc. Am.*, 78, 1985.
3. Ginn, K. and Hald, J., The effect of bandwidth on spatial transformation of sound field measurements, *Inter-Noise*, 87, 1987.
4. Fahy, F.J., Measurement of acoustic intensity using the cross-spectral density of two microphone signals, *J. Acoust. Soc. Am.*, 62, 1977.

5. Chung, J.Y., Cross-spectral method of measuring acoustic intensity without error caused by instrument phase mismatch, *J. Acoust. Soc. Am.*, 64, 1978.
6. Grosche, F.R., Stiewitt, H., and Binder, B., On aero-acoustic measurements in wind tunnels by means of a highly directional microphone system, *Paper AIAA-76-535*, 1976.
7. Sen, R., Interpretation of acoustic source maps made with an elliptic-mirror directional microphone system, *Paper AIAA-96-1712*, 1996.
8. Blackner, A.M. and Davis, C.M., Airframe noise source identification using elliptical mirror measurement techniques, *Inter-Noise 95*, 1995.
9. Dobrzynski, W., Airframe noise studies on wings with deployed high-lift devices, *Paper AIAA-98-2337*, 1998.
10. Dobrzynski, W., Research into landing gear airframe noise reduction, *Paper AIAA-2002-2409*, 2002.
11. Johnson, D.H. and Dudgeon, D.E. 1993. *Array Signal Processing*, Prentice Hall, Englewood Cliffs, NJ.
12. Elias, G., Source localization with a two-dimensional focused array: optimal signal processing for a cross-shaped array, *Inter-Noise 95*, 1995.
13. Dougherty, R.P. and Stoker, R.W., Sidelobe suppression for phased array aeroacoustic measurements, *Paper AIAA-98-2242*, 1998.
14. Nordborg, A., Optimum array microphone configuration, *Inter-Noise 2000*, 2000.
15. Hald, J. and Christensen, J.J., A class of optimal broad band phased array geometries designed for easy construction, *Inter-Noise 2002*, 2002.
16. Piet, J.F. and Elias, G., Airframe noise source localization using a microphone array, *Paper AIAA-97-1643*, 1997.
17. Hayes, J.A., Airframe noise characteristics of a 4.7% scale DC-10 model, *Paper AIAA-97-1594*, 1997.
18. Stoker, R.W., Underbrink, J.R., and Neubert, G.R., Investigation of airframe noise in pressurized wind tunnels, *Paper AIAA-2001-2107*, 2001.