Acoustic Measurement

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G.R.A.S. Sound and Vibration

Sound is normally defined as vibration of a solid, liquid, or gaseous medium in the frequency range of the human ear, i.e., between about 20 Hz and 20 kHz. Here, the definition is further limited and only vibrations in liquids and gaseous media are considered. In contrast to solid media, a liquid or gaseous medium cannot transmit shear forces, so sound waves are always longitudinal waves, in which the particles moves in the direction of propagation of the wave. The wave propagation in gaseous and liquid media can be described by the three variables: the pressure p, the particle velocity u, and the density ρ. The relation between these is described by the wave equation [1], and this can be derived from three basic equations: the Euler equation (this is essentially Newton's second law applied to a fluid), the Continuity equation, and the State equation. Although the wave equation in principle can be used to describe and calculate all sound waves in all situations, it will in practice often be impossible to perform the necessary calculations. In some special cases, it is possible to get analytical results directly from the wave equation and these cases are therefore of special interest. The cases most often encountered in acoustics is the *free field,* the *diffuse* (or reverberant) *field,* and the *closed coupler.* The free field is, in principle, an infinite, empty (except for the medium and the source) space, with no reflections. Here, the waves are allowed to radiate freely in all directions without reflections. In practice, the free field is implemented in an anechoic chambers, where all walls have been made nearly 100% absorptive. The diffuse field is obtained in a reverberation room where all walls have been made, in principle, 100% reflective. At the same time, the walls are made nonparallel and the result is a sound field with sound waves in all directions. The closed coupler is a small chamber, with dimensions small compared to the wavelength of the sound. A special case of this is the standing wave tube. This is a tube with a diameter smaller than the wavelength and with a sound source in one end. With a suitable loudspeaker as a source, the wave propagation in the tube can be assumed to be one-dimensional. This simplifies the mathematical description so that it is possible to calculate the sound field.

In practice, almost the only parameter measured directly in acoustics is the sound pressure, and all other parameters like sound power, particle velocity, reverberation time, directivity, etc. are derived from pressure measurements. These are performed with measurement microphones in gaseous media and hydrophones in liquid media. The measurement microphones are all of the condenser type to ensure precision, long-term stability, and sensitivity. Hydrophones are usually made with a rubber coating over a sensitive element of piezoelectric material.

The traditional frequency range from 20 Hz to 20 kHz for acoustic measurements is selected because this is the range audible to the human ear. Sound waves exist outside this range in the form of infrasound (below 20 Hz) and ultrasound (above 20 kHz). As the basic equations (the wave equation) and measurement principles are the same for both infrasound and ultrasound, many of the principles from the frequency range from 20 Hz to 20 kHz can be extended to these ranges.

27.1 The Wave Equation

Sound wave propagation cannot take place in a vacuum, but is always associated with some kind of medium. For simplicity, assume that this medium is air, although the same equations are valid also for all gaseous and fluid media. In this medium, the concept of an air particle can be introduced. An *air particle* is a small volume of air in which the acoustical parameters like pressure, density, etc. can be considered constant. On the other hand, the air volume must be large enough to include a very large number of air molecules, so that the air volume can be considered to be a continuous medium and not a collection of molecules. The *Euler equation* for such an air particle is given by:

$$-\text{grad } p = \rho_0 \left(\frac{\partial \vec{v}}{\partial t} \right) \tag{27.1}$$

where p is the pressure, ρ_0 is the density, and v is the particle velocity. This equation can be considered as Newton's second law ($F = ma$) applied to a fluid. Here, the gradient of the pressure equals the force F acting on the air particle, the density equals the mass m, and the time differentiated particle velocity $\partial v/\partial t$ equals the acceleration.

The second equation necessary to derive the wave equation is the *continuity equation.* This simply states that if you have a small volume of air and you bring in some extra air, the density (or the mass) will increase. Mathematically, this can be formulated as:

$$\text{div } \vec{v} = -\frac{1}{\rho_0} \cdot \frac{\partial \rho}{\partial t} \tag{27.2}$$

where c is the sound velocity. The sound velocity depends on the composition of the air and the temperature. For normal air at 0°C, the velocity is 314 m s^{-1} while at 20°C, the velocity is 340 m s^{-1}.

The third equation is the *state equation,* which relates pressure changes to changes in the density, that is, if a small volume of air is compressed, the density will increase:

$$\frac{\partial \rho}{\partial t} = \frac{1}{c^2} \times \frac{\partial p}{\partial t} \tag{27.3}$$

Now we have three equations relating the three variables: pressure, particle velocity, and density. By eliminating the particle velocity and the density, we obtain one differential equation for the sound pressure:

$$\nabla^2 p = \frac{1}{c^2} \frac{\partial^2 p}{\partial t^2} \tag{27.4}$$

This is the *wave equation* for acoustic waves in gaseous and fluid media. In principle, this allows one to calculate the sound pressure anywhere in a sound field, if some suitable boundary conditions are given. In practice, however, it is only possible to find solutions in a few simple cases.

27.2 Plane Sound Waves

In a free space, at great distance from the sound source, sound waves are approximately plane waves. This means that the wave equation only depends on one of the coordinates in the wave equation. If the direction of propagation of the wave fronts is in the x-direction, the solution to the wave equation reduces to:

$$p = A\cos\left[\left(\omega t - k\right)\left(x + \varphi_a\right)\right] \tag{27.5}$$

where ω is the frequency. Similarly, the particle velocity in the free field is given by:

$$v = \frac{A}{\rho c}\cos\left[\left(\omega t - k\right)\left(x + \varphi_a\right)\right] = \frac{p}{\rho c} \tag{27.6}$$

This means that in a plane wave, the particle velocity is equal to the pressure divided by the constant ρc and the pressure and particle velocity are in phase. The constant ρc is the *acoustic impedance;* and for air at 20°C, the density is 1.29 kg m^{-3} and the sound velocity is 340 m s^{-1}, giving an acoustic impedance of 438.6 kg m^{-2} s^{-1}.

The plane wave transmits energy in the direction of propagation. The power transmitted per unit area is the intensity in the direction of the propagation (in many older textbooks, terms like "the intensity of the sound" were mistakenly used for the magnitude of the sound pressure). In general, the intensity is given by the product of the sound pressure and the sound velocity; thus, in the case of the plane wave, the intensity (I) can be calculated from Equations 27.5 and 27.6:

$$I = vp = \frac{p^2}{\rho c} \tag{27.7}$$

Thus, in the plane wave, the intensity transmitted by the wave can be calculated from the sound pressure and, as the sound power is the intensity per unit area, the sound power can be calculated by multiplying the intensity by the area.

27.3 Spherical Waves

Another simple solution to the wave equation can be found for the radiation from a point source into free space. The point source is an infinitely small sphere whose surface is pulsating radially. In practice, for small sound sources (i.e., where the dimensions of the sound source is small compared to the wavelength of the sound), the point source is a good approximation for the real physical source that makes the spherical wave solution of special interest.

The wave equation in Equation 27.5 is transformed into the spherical coordinates r, θ, and ψ. As the point source radiates equally in all directions, the solution depends only on the distances r from the center of the point source:

$$p = \frac{P_0}{r}\cos\left(\omega t - kr\right) \tag{27.8}$$

It can be seen that the sound pressure is inversely proportional to the distance from the sound source. The particle velocity can be divided into a near-field contribution v_n and a far-field contribution v_f:

$$v_f = \frac{P_0}{r\rho c}\cos(\omega t - kr) \tag{27.9}$$

$$v_n = \frac{P_0}{\omega\rho r^2}\sin(\omega t - kr) \tag{27.10}$$

The far-field contribution in Equation 27.9 can be seen to be in-phase with the pressure in Equation 27.8 and also the particle velocity is inversely proportional to the distance r from the point source. The near-field contribution is inversely proportional to the square of the distance to the source and therefore dies away rapidly as the distance to the source increases.

As in the case of the plane wave, the intensity in the spherical wave is the product of the pressure and particle velocity. For the near-field contribution, one obtains:

$$I = v_n p = \frac{P_0}{\omega\rho r^2}\sin(\omega t - kr)\frac{P_0}{r}\cos(\omega t - kr) = 0 \tag{27.11}$$

That is, the near-field part of the particle velocity does not contribute to the radiated power as the particle velocity is 90° out of phase with the pressure.

The far-field contribution is given by:

$$I = v_f p = \frac{P_0}{r\rho c}\cos(\omega t - kr)\frac{P_0}{r}\cos(\omega t - kr) = \frac{P_0^2}{r^2\rho c}\cos^2(\omega t - kr) \tag{27.12}$$

It can be seen that the intensity decreases with the square of the distance to the source and by combining Equations 27.8 and 27.12, one obtains:

$$I = \frac{P_0^2}{r^2\rho c}\cos^2(\omega t - kr) = \frac{p^2}{\rho c} \tag{27.13}$$

which is identical to Equation 27.7 for the plane wave. Thus, as for the plane progressive wave, the intensity in the spherical wave can be calculated from the pressure.

27.4 Acoustic Measurements

As can be seen from the wave equation, the full acoustic field can in principle be described from only pressure measurements. This means that all other acoustic parameters can be derived from pressure measurements and, in practice, pressure is often the only parameter measured. There have been a few attempts to make transducers for particle velocity measurements based on, for example, transmission of ultrasonic waves; but the absolute dominating transducers for acoustic measurements are the condenser-type microphones, Figure 27.1. These have proven to be superior with respect to temperature stability, long-term stability, and insensitivity to rough handling. While measurement microphones are designed and produced to ensure well-defined and accurate measurements, a wide range of other microphones are available for other purposes. These can, for example, be for incorporation in telephones, where price is a very decisive factor, or for studio recordings, where a subjective evaluation is more important than the objective performance.

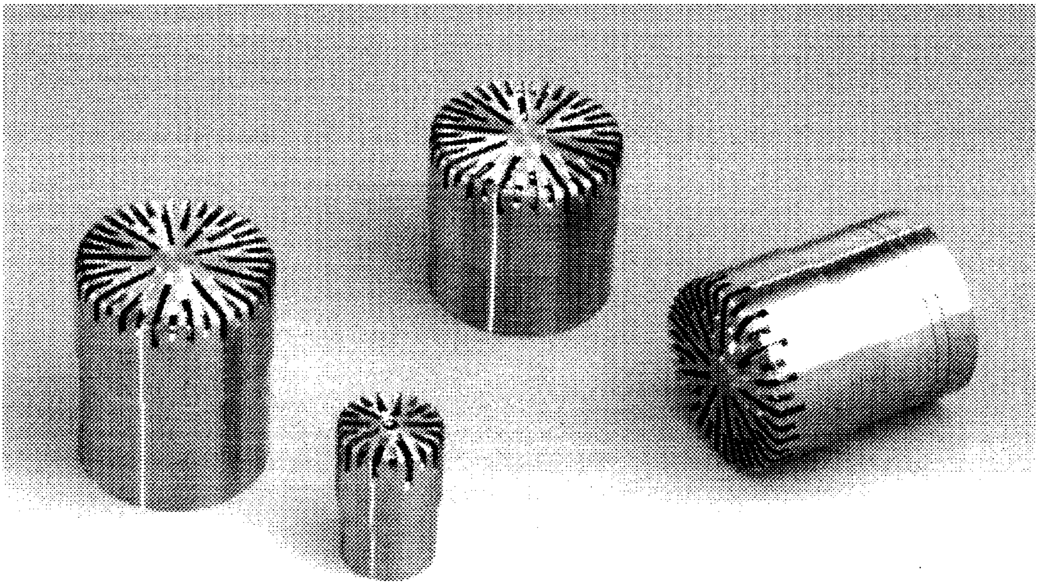

FIGURE 27.1 Measurement microphones: ½″ and ¼″.

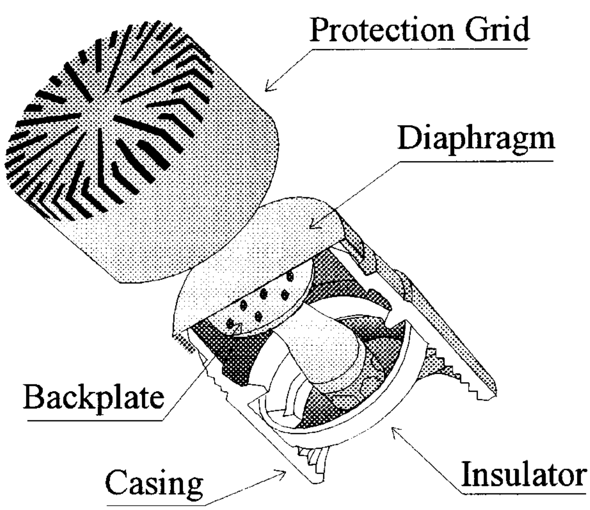

FIGURE 27.2 Basic elements of a measurement microphone.

Condenser Microphones

The condenser microphone consists basically of five elements: protection grid, microphone casing, diaphragm, backplate, and insulator; see Figure 27.2. The diaphragm and the backplate form the parallel plates of an air capacitor. This capacitor is polarized with a charge from an external voltage supply (externally polarized type) or by an electric charge injected directly into an insulating material on the backplate (prepolarized type). When the sound pressure in the sound field fluctuates, the distances between the diaphragm and the backplate will change, and consequently change the capacitance of the diaphragm/backplate capacitor. As the charge on the capacitor is kept constant, the change in capacitance will generate an output voltage on the output terminal of the microphone. The acoustical performance of a microphone is determined by the physical dimensions such as diaphragm area, the distance between the diaphragm and the backplate, the stiffness and mass of the suspended diaphragm, and the internal volume of the microphone casing. These factors will determine the frequency range of the microphone, the sensitivity, and the dynamic range. The sensitivity of the microphone is described as the output voltage of the microphone for a given sound pressure excitation, and is in itself of little interest for the operation of the microphone, except for calibration purposes. However, the sensitivity of the microphone (together with the electric impedance of the cartridge) also determines the lowest sound pressure level that can be measured with the microphone. For example, with a microphone with a sensitivity of 2.5 mV Pa^{-1}, the lowest level that can be measured is around 40 dB (re. 20 μPa), while a microphone with a sensitivity of 50 mV Pa^{-1} can measure levels down to approximately 15 dB (re. 20 μPa).

The size of the microphone is the first parameter determining the sensitivity of the microphone. In general, the larger the diaphragm diameter, the more sensitive the microphone will be. There are, however, limits to how sensitive the microphone can be made by simply making it larger. The polarization voltage between the diaphragm and the backplate will attract the diaphragm and deflect this toward the backplate. As the size of the microphone is increased, the deflection will increase and eventually the diaphragm will be deflected so much that it will touch the backplate. To avoid this, the distance between the diaphragm and the backplate can be increased or the polarization voltage can be decreased. Both of these actions will, however, decrease the sensitivity, so that the optimum size of a practical measurement microphone for use up to 20 kHz is very close to ½″ (12.6 mm).

As the size of the microphone is decreased, the useful frequency range of the microphone is increased. The frequency range, which can be obtained, is determined in part by the size of the microphone. At high frequencies, when the wavelength of the sound waves becomes much smaller than the diameter of the diaphragm, the diaphragm will stop behaving like a rigid piston (the diaphragm "breaks up" — this is not a destructive phenomenon). Different parts of the diaphragm will start to move with different magnitude and phase, and the frequency response of the microphone will change. To avoid this, the upper limiting frequency is placed so that the sensitivity of the microphone drops off before the diaphragm starts to break up. This gives, for a typical 0.5 in. microphone, an upper limiting frequency in the range from 20 kHz to 40 kHz, depending on the diaphragm tension. If the diaphragm is tensioned so that it becomes more stiff, the resonance frequency of the diaphragm will be higher; on the other hand, the sensitivity of the microphone will be reduced as the diaphragm deflection by a certain sound pressure level decreases.

The frequency response of the microphone is determined by the diaphragm tension, the diaphragm mass, and the acoustical damping in the airgap between the diaphragm and the backplate. This system can be represented by the mechanical analogy of a simple mass–spring–damper system as in Figure 27.3. The mass in the analogy represents the mass of the diaphragm and the spring represents the tension in the diaphragm. Thus, if the diaphragm is tensioned to become stiffer, the corresponding spring will become stiffer. The damping element in the analogy represents the acoustical damping between the diaphragm and the backplate. This can be adjusted by, for example, drilling holes in the backplate. This will make it easier for the air to move away from the airgap when the diaphragm is deflected, and therefore decrease damping.

The frequency response of the simple mechanical model of the microphone is given in Figure 27.4, together with the influence of the different parameters. At low frequencies (below the resonant frequency), the response of the microphone is determined by the diaphragm tension, and as described above, the sensitivity will increase if the tension is decreased. The resonant frequency is determined by the diaphragm tension and the diaphragm mass, with an increased tension giving an increased resonant frequency, and an increased mass giving a decreased resonant frequency. The response around the

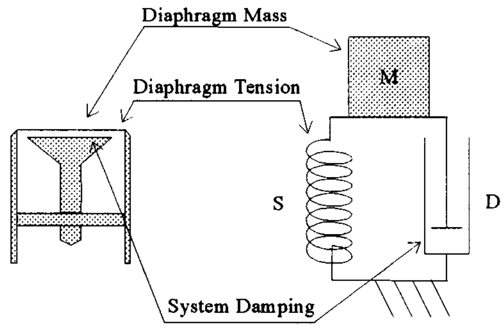

FIGURE 27.3 Mechanical analogy of a microphone.

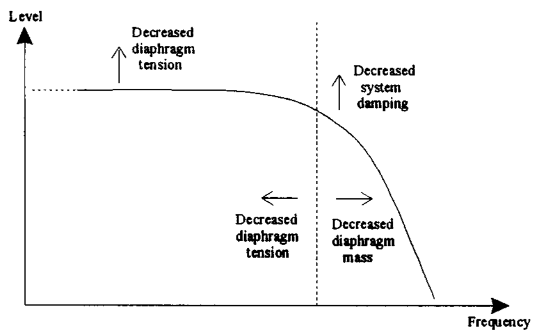

FIGURE 27.4 Influence of microphone parameters on frequency response.

resonant frequency is determined by the acoustical damping, where an increase in the damping will decrease the response.

Although the material selection and assembling techniques have changed during the last few years, the basic types of microphones remain unchanged. The basic types are the free-field microphones, the pressure microphones, and the random incidence microphones. They have been constructed with different frequency characteristics, corresponding to the different requirements.

The *pressure microphone* is meant to measure the actual sound pressure as it exists on the diaphragm. A typical application could be the measurement of the sound pressure in a closed coupler or as in Figure 27.5, the measurement of the sound pressure at a boundary. In this case, the microphone forms part of the wall and measures the sound pressure on the wall itself. The frequency response of this microphone should be flat in a frequency range as wide as possible, taking into account that the sensitivity will decrease as the frequency range is increased. The acoustical damping in the airgap between the diaphragm and the backplate is adjusted so that the frequency response is flat up to and a little beyond the resonant frequency.

The *free-field microphone* is designed to essentially measure the sound pressure as it existed before the microphone was introduced into the sound field. At higher frequencies, the presence of the microphone itself in the sound field will change the sound pressure. In general, the sound pressure around the microphone cartridge will increase due to reflections and diffraction. The free-field microphone is designed so that the frequency characteristics compensate for this pressure increase. The resulting output of the free-field microphone is a signal proportional to the sound pressure as it existed before the microphone was introduced into the sound field. The free-field microphone should always be pointed toward the sound source (0° incidence), as in Figure 27.6. In this situation, the presence of the microphone diaphragm in the sound field will result in a pressure increase in front of the diaphragm, see

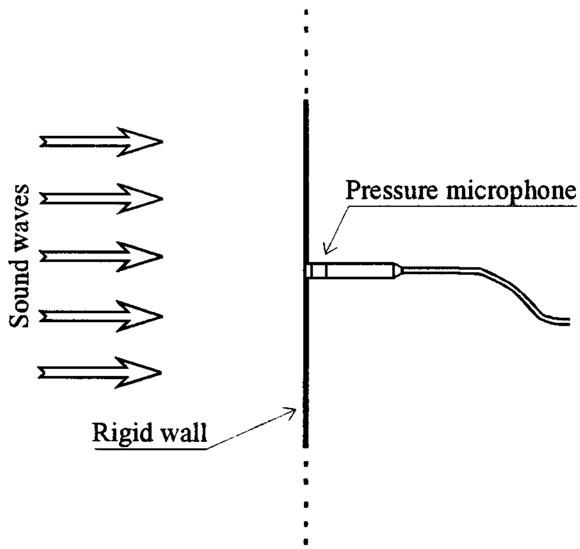

FIGURE 27.5 Application of pressure microphones.

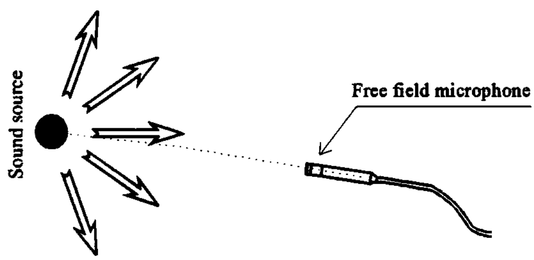

FIGURE 27.6 Application of a free-field microphone.

Figure 27.7(*a*), depending on the wavelength of the sound waves and the microphone diameter. For a typical ½″ microphone, the maximum pressure increase will occur at 26.9 kHz, where the wavelength of the sound ($\lambda = 342\ \text{ms}^{-1}/26.9\ \text{kHz} \approx 12.7\ \text{mm} \approx 0.5$ in.) coincides with the diameter of the microphone. The microphone is then designed so that the sensitivity of the microphone decreases by the same amount as the acoustical pressure increases in front of the diaphragm. This is obtained by increasing the internal acoustical damping in the microphone cartridge, to obtain a frequency response as in Figure 27.7(*b*). The result is an output from the microphone, Figure 27.7(*c*), which is proportional to the sound pressure as it existed before the microphone was introduced into the sound field. The curve in Figure 27.7(*a*) is also called the "free-field correction curve" for the microphone, as this is the curve that must be added to the frequency response of the microphone cartridge to obtain the acoustical characteristic of the microphone in the free field.

The free-field microphone is required in principle, to be pointed toward the sound source and that the sound waves travel in essentially one direction. In some cases, (e.g., when measuring in a reverberation room or other highly reflecting surroundings), the sound waves will not have a well-defined propagation direction, but will arrive at the microphone from all directions simultaneously. The sound waves arriving at the microphone from the front will cause a pressure increase, as described for the free-field microphone, while the waves arriving from the back of the microphone will be decrease to a certain extent due to the shadowing effects of the microphone cartridge. The combined influence of the waves coming from different directions therefore depends on the distribution of sound waves from different directions. For

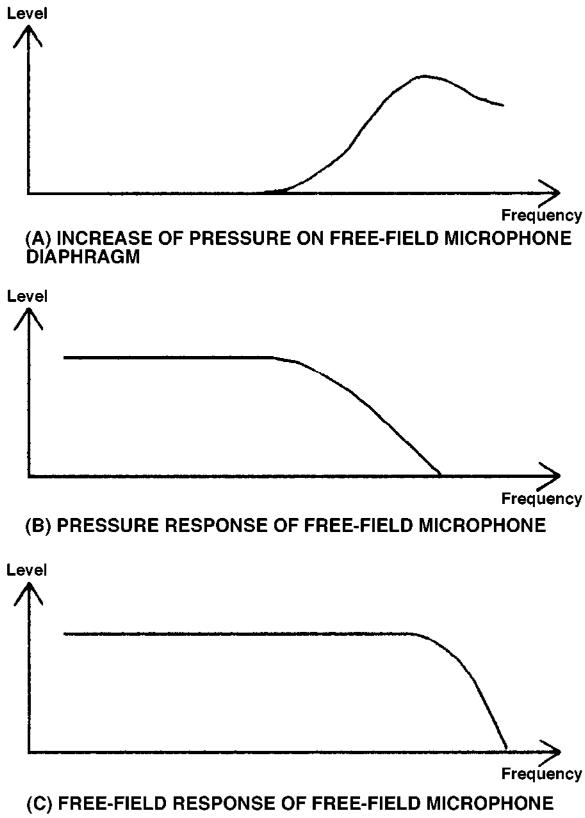

FIGURE 27.7 Frequency response of free-field microphone: (*a*) pressure increase in front of diaphragm; (*b*) microphone pressure response; (*c*) resulting microphone output.

measurement microphones, a standard distribution has been defined, based on statistical considerations, resulting in a standardized random incidence microphone.

As mentioned previously, measurement microphones can be either of the externally polarized type or the prepolarized type. The externally polarized types are by far the most stable and accurate microphones and should be preferred for precision measurements. The prepolarized microphones are, however, preferred in some cases, in that they do not require the external polarization voltage source. This is typically the case when the microphone will be used on small hand-held devices like sound level meters, where a power supply for polarization voltage would add excessively to cost, weight, and battery consumption. Still, it should be realized that prepolarized microphones in general are much less stable to environmental changes than externally polarized microphones.

27.5 Sound Pressure Level Measurements

The human ear basically hears the sound pressure, but the sensitivity varies with the frequency. The human ear is most sensitive to sound in the frequency range from 1 kHz to 5 kHz, while the sensitivity drops at higher and lower frequencies. This has led to the development of several frequency weighting functions, which attempt to replicate the sensitivity of the human ear. Also, the response of the human ear to time-varying signals and impulses has led to the development of instruments with well-defined time weighting functions. The resulting measurement instrumentation is the *sound level meter,* as defined in for example by the IEC International Standard 651, "Sound Level Meters" [2]. The standard defines four classes of sound level meters for different accuracy's (Table 27.1). Type 0 is the most accurate,

TABLE 27.1 IEC 651 Sound Level Meter Requirements

Frequency (Hz)	Type 0 (dB)	Type 1 (dB)	Type 2 (dB)	Type 3 (dB)
10	+2; –∞	+3; –∞	+5; –∞	+5; –∞
12.5	+2; –∞	+3; –∞	+5; –∞	+5; –∞
16	+2; –∞	+3; –∞	+5; –∞	+5; –∞
20	±2	±3	±3	+5; –∞
25	±1.5	±2	±3	+5; –∞
31.5	±1	±1.5	±3	±4
40	±1	±1.5	±2	±4
50	±1	±1.5	±2	±3
63	±1	±1.5	±2	±3
80	±1	±1.5	±2	±3
100	±0.7	±1	±1.5	±3
125	±0.7	±1	±1.5	±2
160	±0.7	±1	±1.5	±2
200	±0.7	±1	±1.5	±2
250	±0.7	±1	±1.5	±2
315	±0.7	±1	±1.5	±2
400	±0.7	±1	±1.5	±2
500	±0.7	±1	±1.5	±2
630	±0.7	±1	±1.5	±2
800	±0.7	±1	±1.5	±2
1000	±0.7	±1	±1.5	±2
1250	±0.7	±1	±1.5	±2.5
1600	±0.7	±1	±2	±3
2000	±0.7	±1	±2	±3
2500	±0.7	±1	±2.5	±4
3125	±0.7	±1	±2.5	±4.5
4000	±0.7	±1	±3	±5
5000	±1	±1.5	±3.5	±6
6300	+1; –1.5	+1.5; –2	±4.5	±6
8000	+1; –2	+1.5; –3	±5	±6
10000	+2; –3	+2; –4	+5; –∞	+6; –∞
12500	+2; –3	+3; –6	+5; –∞	+6; –∞
16000	+2; –3	+3; –∞	+5; –∞	+6; –∞
16000	+2; –3	+3; –∞	+5; –∞	+6; –∞

intended for precision laboratory measurements, while Type 1 is most widely used for general-purpose measurements, see Figure 27.8. Type 2 is used where low price is of importance, while Type 3 is not used in practice because of the wide tolerances, making the results too unreliable. The output of the sound level meter is, in principle, assumed to be an approximate measure of the impression perceived by the human ear.

The sound level meter can be functionally divided into four parts: microphone and preamplifier, A-weighting filter, rms detector and display (Figure 27.9). The microphone should ensure the correct measurement of the sound pressure within the frequency range for the given class. Also, the standard gives requirements for the directionality of the microphone. The frequency response of the instrument, including the weighting filter, is given for sound waves arriving at the microphone along the reference direction. For sound waves arriving from other directions, the standard allows wider tolerances at higher frequencies, taking into account the inevitable reflections and diffraction occurring at higher frequencies.

The preamplifier converts the high-impedance output signal from the microphone to a low-impedance signal, but has in itself no or even negative voltage amplification. The signal from the preamplifier is then passed through an A-weighting filter. This is a standardized filter which, in principle, resembles the sensitivity of the human ear, so that a measurement utilizing this filter will give a result which correlates with the subjective response of an average listener. The filter, with the filter characteristic as in

FIGURE 27.8 Modern Type 1 sound level meter with built-in frequency analyzer.

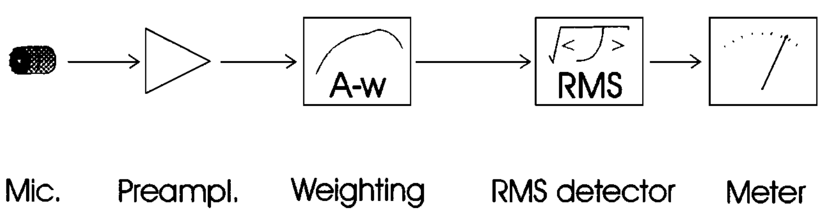

FIGURE 27.9 Functional parts of a sound level meter.

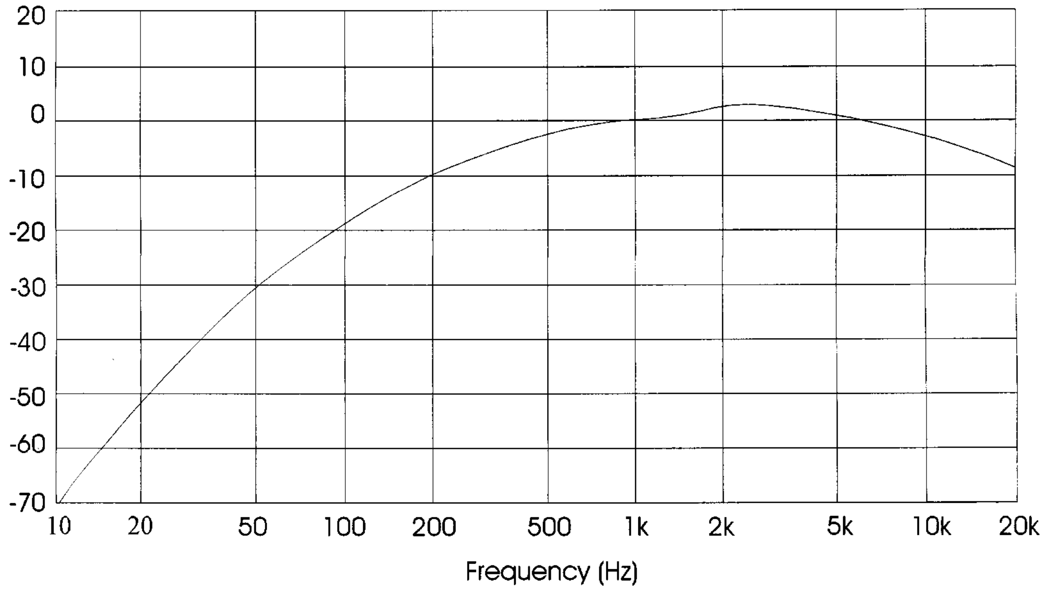

FIGURE 27.10 A-weighting curve.

Figure 27.10, attenuates low and high frequencies and slightly amplifies frequencies in the mid-frequency range from 1 kHz to 5 kHz. There are a number of other weighting curves, denoted B-weighting, C-weighting and D-weighting, which may give better correlation with subjective responses in special cases, such as for very high or very low levels, or for aircraft noise.

The signal from the A-weighting filter is subsequently passed through an exponential rms detector, with a time constant of either 125 ms ("fast") or 1 s ("slow"). These time constants simulate the behavior of the human ear when subjected to time-varying signals. Especially when the duration of the sound stimuli to the human ear becomes shorts (e.g., around 200 ms), the sound is subjectively judged as being lower compared to the same sound heard continuously. The same effect is obtained using the "fast" averaging time. This will however give a higher statistical uncertainty on the level estimate than when using the time constant "slow," so this should be chosen if the sound signal is continuously. Other standards describe special sound level meters such as integrating sound level meters or impulse sound level meters intended for special purposes.

27.6 Frequency Analyzers

While the sound level meter gives a single reading for the sound level in the frequency range from 20 Hz to 20 kHz, it is often desirable to have more detailed information about the frequency content of the signal. Two types of frequency analyzers are commonly used in acoustic measurements: FFT analyzers and real-time filter analyzers. The *FFT analyzers,* with very high frequency resolutions, can give a wealth of frequency information and make it possible to separate closely space harmonics (e.g., from a gear box). In contrast to this, the *real-time filter analyzers,* Figure 27.11, uses a much broader frequency resolution, usually in 1/3-octave bands. The frequency analysis is performed by a bank of filters (nowadays mostly digital filters) with well-defined frequency and time responses. The filter responses have been internationally standardized, with the center frequencies and passbands as shown in Table 27.2. In the frequency range from 20 Hz to 20 kHz, the 1/3-octave filterbank consists of 31 filters, simultaneously measuring the input signal. The resulting 1/3-octave spectrum resembles the subjective response of the human ear.

FIGURE 27.11 Real-time frequency analyzer for acoustic measurements.

27.7 Pressure-Based Measurements

The result of sound pressure measurements will be influenced by many factors: source, source operating conditions, surroundings, measurement position, etc. Depending on the goal of the measurement, these parameters can be controlled in different manners. If the goal of the measurement is to quantify the noise exposure to an operator's ear in a noisy environment, it is important that the microphone is in the same position as the operator's ear would normally be in, and that the environment is equal to the normal operating environment. If, on the other hand, the task is to describe the sound source as a noise-emitting machine, it is important to minimize the influence of the environment on the measurement result.

If the aim is to describe the measuring object as a noise source, it is customary to state the radiated sound power for the source. This is a global parameter quantifying the total noise radiation from the source, and to a certain extent independent of the environment. The sound power can be measured in a number of different ways: in a free field, in a reverberation room, using a substitution technique, or using sound intensity technique. A free field is a sound field in which the sound is radiated freely in all directions, with no restricting walls or reflections. This is most often obtained in a semi-anechoic chamber, where all walls and ceiling have been covered by nearly 100% absorptive material, with only the floor made of reflecting material. When the sound source is placed in the semi-anechoic chamber, the emitted sound waves will radiate freely away from the source and, in the far field, the waves can be considered to be plane waves or spherical waves. Therefore, the sound intensity can be calculated from pressure measurements using Equation 27.7.

TABLE 27.2 1/3-Octave Analysis Frequencies

Nominal center frequency (Hz)	Exact center frequency (Hz)	Passband (Hz)
20	19.95	17.8–22.4
25	25.12	22.4–28.2
31.5	31.62	28.2–35.5
40	39.81	35.5–44.7
50	50.12	44.7–56.2
63	63.1	56.2–70.8
80	79.43	70.8–89.1
100	100.0	89.1–112
125	125.89	112–141
160	158.49	141–178
200	199.53	178–224
250	251.19	224–282
315	316.23	282–355
400	398.11	355–447
500	501.19	447–562
630	630.96	562–708
800	794.33	708–891
1000	1000.0	891–1120
1250	1258.9	1120–1410
1600	1584.9	1410–1780
2000	1995.3	1780–2240
2500	2511.9	2240–2820
3150	3162.3	2820–3550
4000	3981.1	3550–4470
5000	5011.9	4470–5620
6300	6309.6	5620–7080
8000	7943.3	7080–8910
10000	10000.0	8910–11200
12500	12589.3	11200–14100
16000	15848.9	14100–17800
20000	19952.6	17800–22400

As real sound sources seldom radiate equally in all directions, a number of measurements around the test object are averaged. ISO Standard 3745 "Acoustics—Determination of sound power levels of noise sources—Precision method for anechoic and semi-anechoic rooms" [3] specifies an array of microphone positions on a hemisphere over the test object, as in Figure 27.12, with the coordinates as in Table 27.3. As all points are associated with the same area, and as the sound power is intensity times the area, the total radiated sound power can be calculated as:

$$P = A \sum I_n = \frac{2\pi r}{\rho c} \sum p_n^2 \tag{27.14}$$

where A is the area of the test hemisphere with radius r, and p_n is the pressure measured in point number n.

27.8 Sound Intensity Measurements

The calculation in Equation 27.14 of the sound power from sound pressure measurements is based on Equation 27.7. This equation, which gives the intensity based on a pressure measurement, is however only valid in a free field, in the direction of propagation. In general, in the presence of background noise or with reflections from walls, etc., it is not possible to calculate the sound intensity from a single pressure

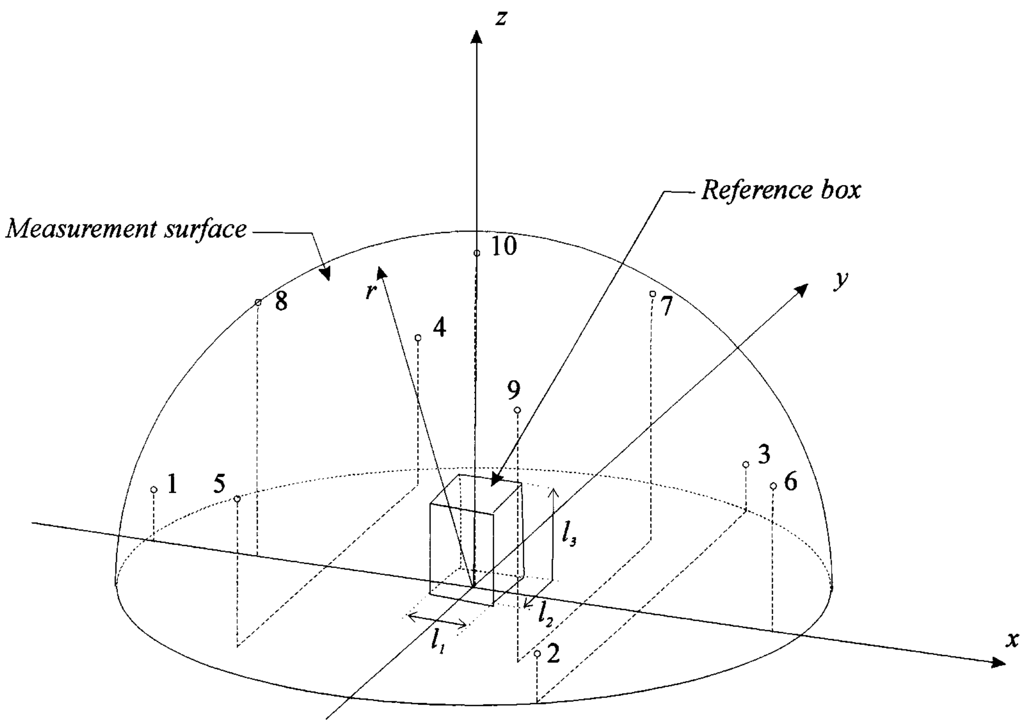

FIGURE 27.12 Measurement points for sound power determination.

TABLE 27.3. Coordinates of Measurement Points for Hemisphere with Radius *r*

Measurement point no.	*x*/*r*	*y*/*r*	*z*/*r*
1	–0.99	0	0.15
2	0.5	–0.86	0.15
3	0.5	0.86	0.15
4	–0.45	0.77	0.45
5	–0.45	–0.77	0.45
6	0.89	0	0.45
7	0.33	0.57	0.75
8	–0.66	0	0.75
9	0.33	–0.57	0.75
10	0	0	1.0

measurement. In these cases, it is however possible to measure directly the sound intensity with a two-microphone intensity probe, Figure 27.13.

Sound intensity I is the product of the pressure and the particle velocity:

$$I = pv \tag{27.15}$$

While the pressure p is a scalar and independent of the direction, the particle velocity is a vector quantity and directionally dependent. When the particle velocity is stated as in Equation 27.15, it is implicit that the velocity is in a certain direction and that the resulting intensity is calculated in the same direction.

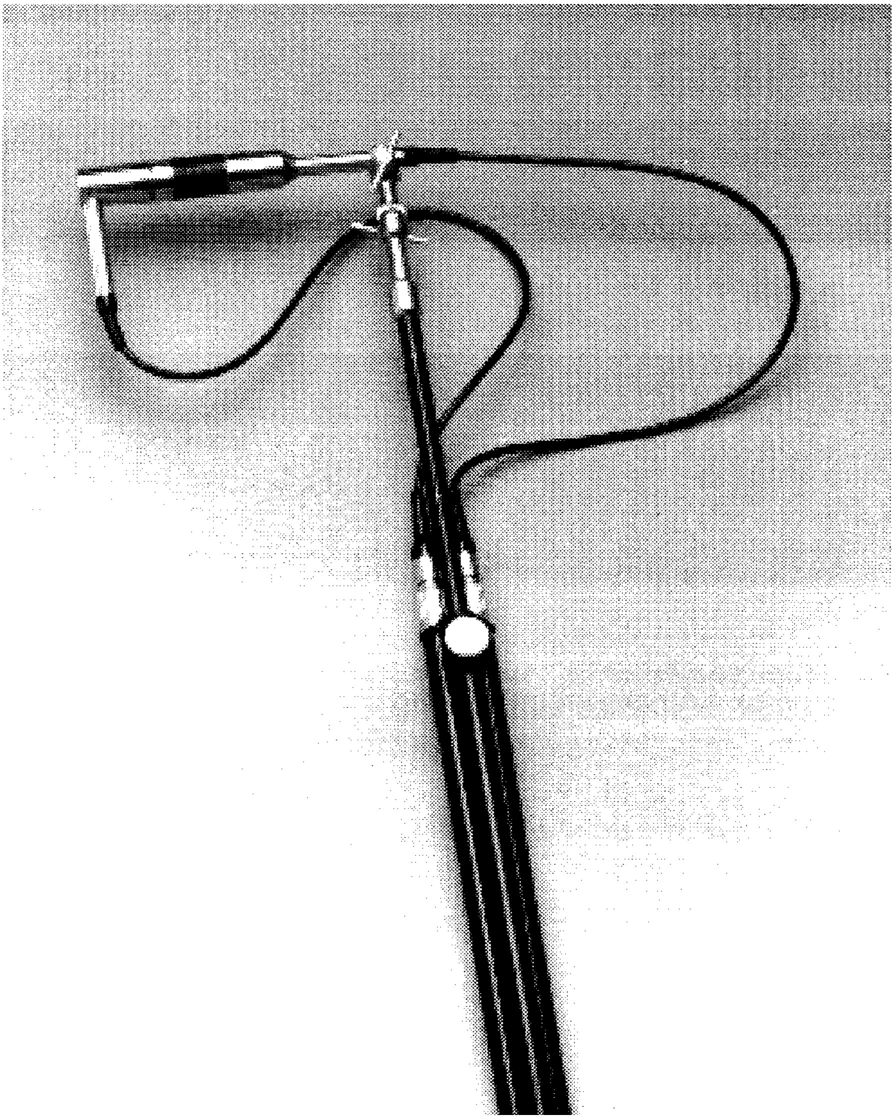

FIGURE 27.13 Two-microphone sound intensity probe.

For example, the particle velocity v in the direction of propagation, Figure 27.14(a), gives the intensity radiation away from the point source, while the particle velocity perpendicular to the propagation direction, Figure 27.14(b), is zero. The intensity calculated from Equation 27.15 will therefore be zero in the direction perpendicular to the propagation direction even though the sound pressure is the same. This means that the sound energy flows away radially from the point source and no energy is flowing tangentially.

The measurement of the sound intensity according to Equation 27.15 requires the measurement of the sound pressure and the particle velocity. With the two-microphone intensity probe, the pressure in a position in between the two microphones is calculated as the mean pressure measured by the two microphones:

$$p = \frac{p_1 + p_2}{2} \tag{27.16}$$

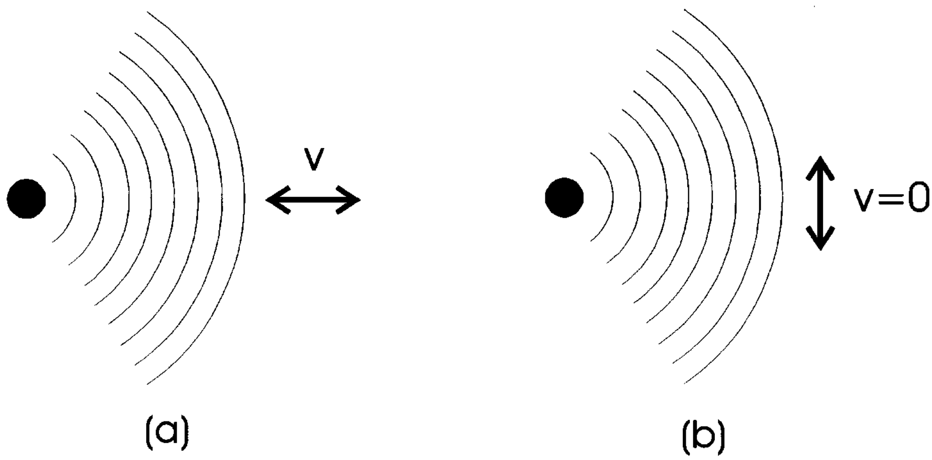

FIGURE 27.14 Particle velocity (*a*) along direction of propagation, and (*b*) perpendicular to direction of propagation.

The air particle velocity *v*, in the direction of the intensity probe, can be calculated from the pressure differences between the two microphone measurements:

$$v = \int \frac{(p_2 - p_1)}{\rho \Delta r} \partial \tau \tag{27.17}$$

where ρ is the density of the air and Δr is the distance between the microphones. The intensity I is then obtained by multiplying the pressure and the velocity:

$$I = pv = \frac{p_1 + p_2}{2} \int \frac{(p_2 - p_1)}{\rho \Delta r} \partial \tau \tag{27.18}$$

The intensity measurement technique is a powerful tool to localize acoustical noise source and to determine the sound power radiated from a sound source, even in the presence of other strong sound sources.

27.9 Near-Field Acoustic Holography Measurements

The term *acoustic holography* comes from the analogy to optical holographs. It is well known how holography, as opposed to a normal photo, enables one to reconstruct the full image of an object. This is obtained by "recording" information about both the magnitude and the phase of the light, while a normal photo only "records" the magnitude of the light. Similarly, with *acoustic holography,* both the magnitude and the phase of the sound field are measured over a plane surface. These measurements result in a complete description of the sound field where both magnitude and phase are known at all points. It is then possible to calculate acoustic quantities, including sound intensity distribution, particle velocity, sound power, radiation pattern, etc.

The basic assumption behind *near-field acoustic holography* (NAH) is that the sound field can be decomposed into two simple wave types: plane waves and evanescent waves. The *plane waves* describe the part of the sound field that is propagated away from the near field toward the far field, and the *evanescent waves* describe the complicated sound field existing in the near field. Any sound field can be described as a combination of plane waves and evanescent waves with different magnitude and directions.

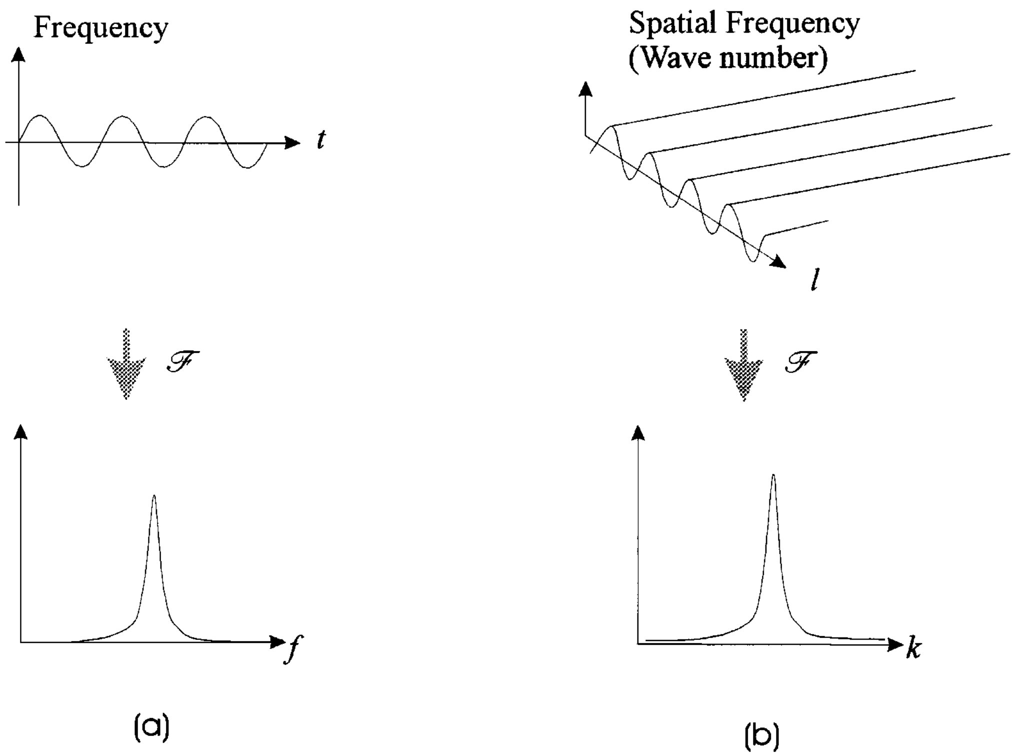

FIGURE 27.15 Temporal and spatial frequency of plane wave.

The magnitude and direction of the individual waves can be described by their spatial frequencies or wave numbers. For a simple plane wave propagating in a certain direction, this can be described in terms of its temporal frequency as well as by its spatial frequency. The temporal frequency, Figure 27.15(*a*), is obtained by looking at the pressure changes with time at a certain point in the sound field. This gives the temporal frequency in hertz or radians per second. Similarly, the spatial frequency, Figure 27.15(*b*), is obtained by looking at the pressure changes at a certain time. At that instant in time, the pressure will be different in different positions in space. If one moves in a certain direction in space, one will see a certain change in the pressure, corresponding to a spatial frequency, measured with the unit cycles per meter or radians per meter. As the temporal frequency gives information about how often the pressure changes with time at a certain point, the spatial frequency gives information about how often the pressure changes with position at a certain time. In the example of Figure 27.15(*b*), the propagation direction of the plane wave was identical to the direction of the axis along which the spatial frequency was measured. In this case, shown again in Figure 27.16(*a*), the relationship between the spatial frequency k_0 (i.e., the wave number) and the temporal frequency f is given by the speed of sound c:

$$k_0 = \frac{2\pi f}{c} = \frac{\omega}{c} = \frac{2\pi}{\lambda} \tag{27.19}$$

where λ is the wavelength. If, however, the axis along which the spatial frequency is measured is not the same as the propagation direction, see the example in Figure 27.16(*b*), this simple relationship is not valid. In this case, although the temporal frequency is the same as in Figure 27.16(*a*), the spatial frequency is lower. For one particular temporal frequency, the spatial frequencies will thus give information about the propagation directions. Therefore, if the sound field is made up of several plane waves with the same temporal frequency, but with different propagation directions, this will be shown in the spatial spectrum

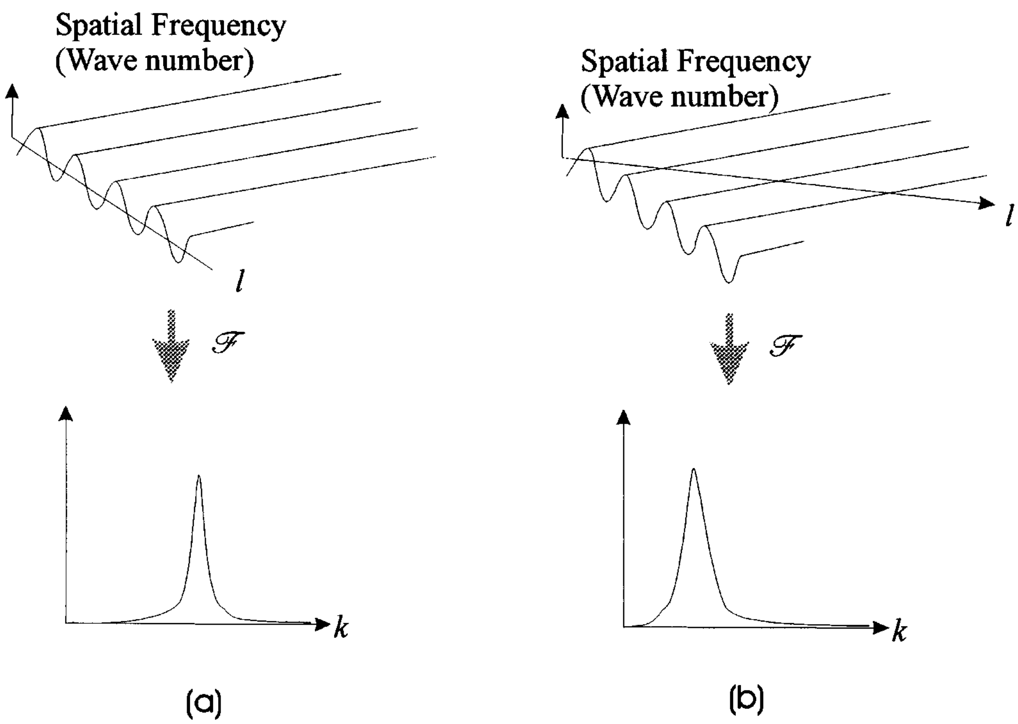

FIGURE 27.16 Spatial frequencies of waves propagating in different directions.

as several spatial frequency components. If, for example, the sound field is a sum of two waves, Figure 27.17, where one wave is traveling along the axis of measurement and the other at an angle of 45° relative to the first wave, the spatial spectrum will contain two spatial frequencies. One spatial frequency will be *k*, corresponding to a wave in the direction along the axis, and the other frequency will be k^* cos (45°). Thus far, the spatial frequencies have been defined along a single axis corresponding to a one-dimensional Fourier transformation. In the NAH technique, the sound field is sampled not only along a single axis, but over a plane. Therefore, a two-dimensional Fourier transformation is used instead. This gives as a result a two-dimensional spatial frequency spectrum, but otherwise the information is the same as before: namely, information about the direction and magnitude of the simple wave types.

The sound field from a point source cannot be explained by simple plane waves such as those in Figures 27.16 and 27.17, as the amplitude decreases with the distance from the origin. The plane waves retain the same magnitude over the full plane. Thus, to described the near-field phenomenon, one must introduce evanescent waves. In the one-dimensional Fourier spectrum, the evanescent waves can be identified as spatial frequencies higher than $k_0 = 2\pi f/c$. Similarly, in the two-dimensional spatial frequency spectrum, the evanescent waves can be identified as having spatial frequencies or wavenumbers higher than k_0.

The individual spatial frequencies in the two-dimensional spatial frequency spectrum correspond to simple plane waves or evanescent waves in the scan plane (i.e., the measurement plane). For each of these simple wave types, it is easy to calculate the pressure in other planes, see Figure 27.18. For the plane waves, a simple phase shift of the wave is required to calculate the result in a new plane. For the evanescent waves, the changes in amplitude must be taken into account; but in principle, this is also a simple transfer function applied to the two-dimensional spatial frequency spectrum. In this way, the two-dimensional spatial frequency spectrum in a new plane can be calculated from the original data by applying simple transfer function operations. The new two-dimensional spatial frequency spectrum is then an inverse Fourier transform (in two dimensions) to get the sound field in the new plane.

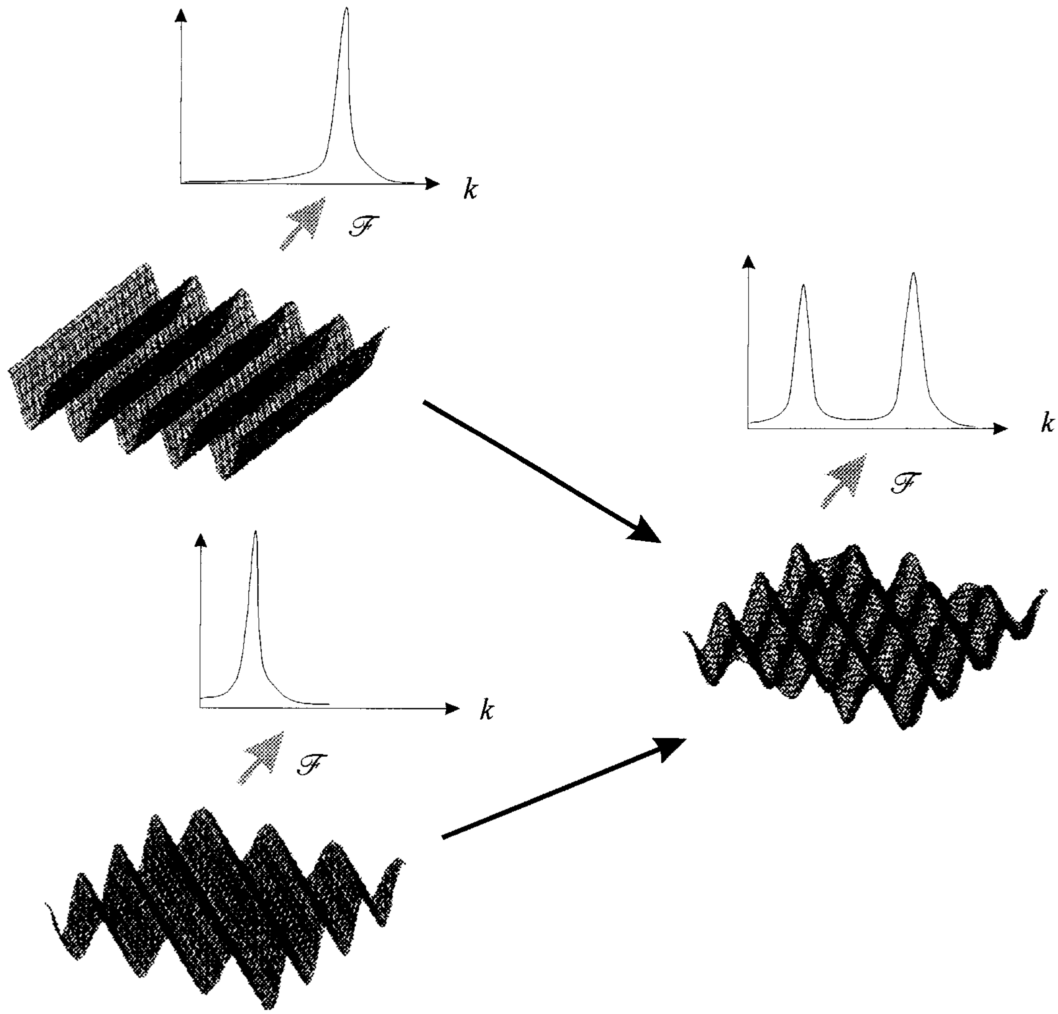

FIGURE 27.17 Spatial spectrum of two waves propagating in different directions.

The overall principle of near-field acoustic holography can be simplified as in Figure 27.19. The sound field is scanned in a plane close to the measuring object. This gives an array of temporal spectra, one for each scan position. Looking at one temporal frequency at a time, one takes out the information from each of the spectra corresponding to the actual frequency of interest. This generates a new array with information about only one temporal frequency. A Fourier transform (in two dimensions) is then applied to the array to generate a two-dimensional spatial frequency spectrum. This can then be transformed to new planes using simple transfer function operations. When the two-dimensional spatial frequency spectrum in the new plane has been calculated, an inverse Fourier transform is used to obtain the new pressure distribution in the new plane. In principle, the NAH technique requires that all cross-spectra between all the scan positions are given; that is, in each of the scan positions, all the cross-spectra to all other scan positions must be determined. A simple scan of a sound field with 2540 scan positions, defining $N = 1000$ scan positions, would result in $\frac{1}{2}N(N + 1) = 500{,}500$ cross-spectra. Instead of measuring all these cross-spectra, the system uses a set of reference transducers to reduce the amount of cross-spectra. The number of necessary reference transducers to give a complete description of the sound field without measuring the full amount of data is determined by the complexity of the sound field. A measurement

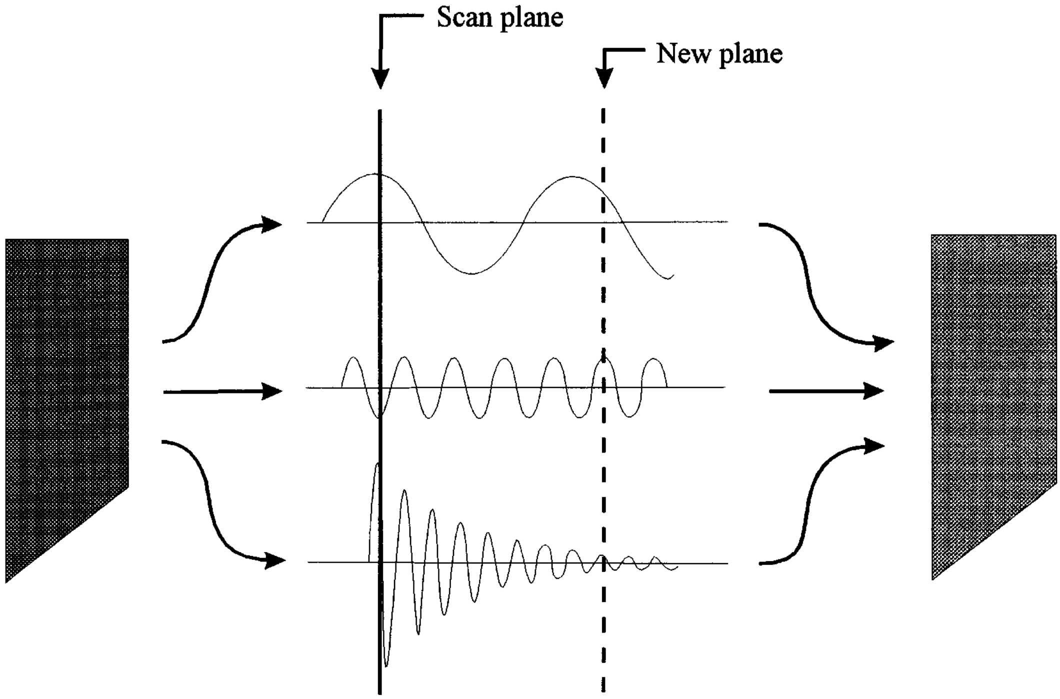

FIGURE 27.18 Transformation of simple wave types in a spatial spectrum, from one measurement plane to another plane.

with, for example, four reference transducers and 2540 scan positions will then be reduced to $4N = 4000$ cross-spectrum measurements.

27.10 Calibration

In order to make accurate and reliable measurements, the microphone and connected instruments must be properly calibrated. The calibration of measurement microphones can be divided into two parts: a level calibration and a frequency response calibration. The *level calibration* establishes the output signal of the microphone for a given acoustic input signal at a given frequency, while the *frequency response* gives the output at other frequencies relative to the level calibration frequency. The level calibration can be performed by a number of different methods with different accuracies.

The most accurate method is the *reciprocity calibration method*. This method utilizes the fact that a condenser microphone is a reciprocal transducer; that is, it can be used as a microphone (to convert an acoustical signal to a voltage signal) and as a loudspeaker (to convert a voltage signal into an acoustical signal). By measuring the relationship between three test microphones driven as both transmitters and receivers, one obtains a set of three equations with the three microphone sensitivities as the unknowns. By solving these three equations, one obtains the sensitivity of the three microphones. The reciprocity calibration method is very accurate but rather tedious and requires well-controlled environmental conditions and is therefore seldom used in practical situations.

The comparison or substitution methods are essentially identical in that they are based on measuring the differences between the test microphone and a reference microphone with known sensitivity. In this case, the reference microphone is often calibrated at an accredited national acoustical laboratory like NIST, NPL, or PTB, whereby the traceability is ensured. In the substitution method, the acoustical output of a sound source is measured with the reference microphone. Afterward, the reference microphone is replaced with a test microphone and the output is measured again. Provided that the sound source has

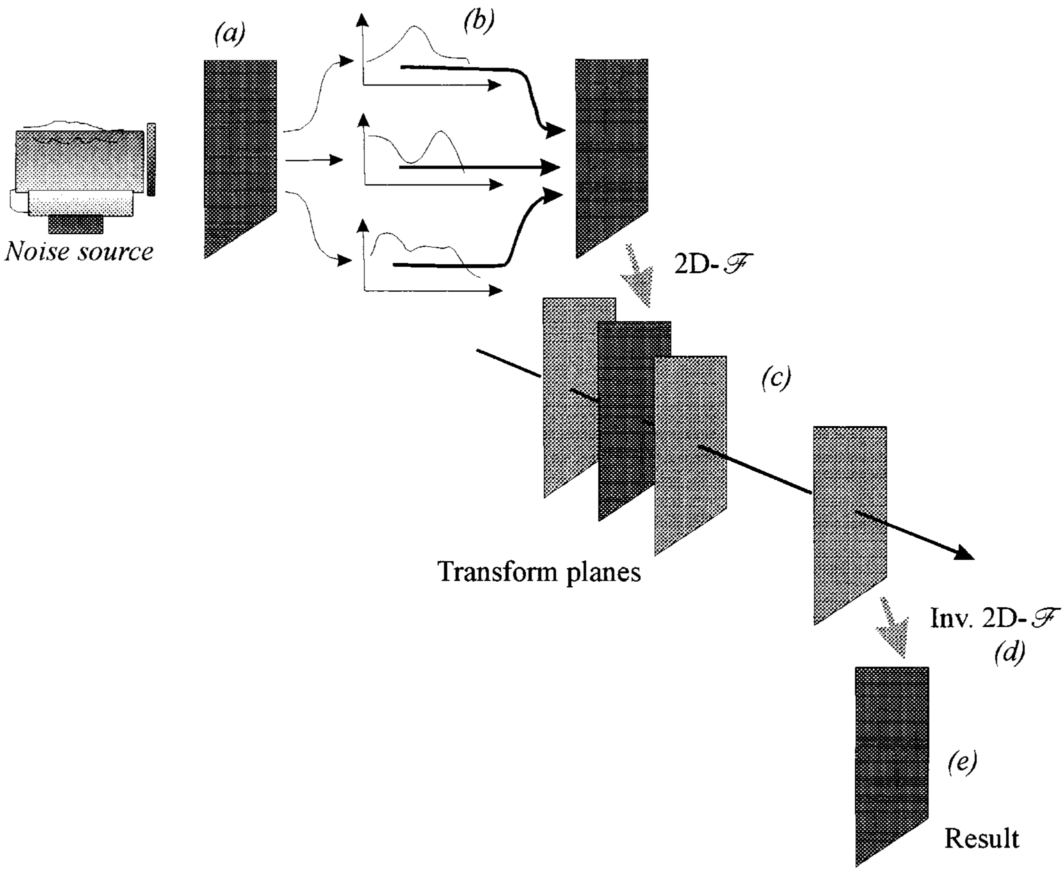

FIGURE 27.19 Overall principle of NAH: (*a*) measurement of cross-spectra in the scan plane; (*b*) calculation for one temporal frequency at a time; (*c*) 2-D spatial Fourier transformation; (*d*) transformation of simple wave types; (*e*) inverse 2-D transformation; (*f*) to obtain the sound field in the new plane.

been stable, the sensitivity of the test microphone can then be calculated. The comparison method is similar to the substitution method, except that the reference microphone and the test microphone are subjected to the same sound pressure simultaneously and therefore the requirements to the sound source stability are less important.

An often-used method for microphone calibration is the *pistonphone method.* The pistonphone, Figure 27.20, is a very stable sound source, which produces a well-defined sound pressure level inside a closed coupler. It works by volume displacements, Figure 27.21, with a well-defined velocity, usually at 250 Hz. As the piston is moving in and out, the volume of the closed coupler is changed and this will result in pressure variations. The actual pressure level obtained in the pistonphone depends on the volume of the coupler, the volume displacement of the pistons, the barometric pressure, and—to a lesser degree—on other factors such as humidity, heat dissipation, etc. As the pistonphone is based on a relatively simple mechanical system, it is very reliable and easy to use in practice, with an accuracy around 0.1 dB. Also, the pistonphone is often used as the stable sound source for calibrations using comparisons or substitution methods.

A *sound pressure calibrator* is basically a small self-contained comparison calibration device. The test microphone is inserted into a small, closed volume and a small loudspeaker produces a single frequency signal, usually at 1 kHz. The output level of the loudspeaker is controlled in a feedback system with a signal from a reference microphone. Provided that the reference microphone and the feedback gain are

FIGURE 27.20 Pistonphone for microphone calibration.

stable, the sound level at the test microphone will be well-defined and the sensitivity can be determined. The sound level calibrators are normally not used to make accurate microphone calibrations, but rather to make field checks of the integrity of a complete measurement system.

The frequency response of a microphone is most often determined by the *electrostatic actuator method.* A conducting grid is placed close to and parallel to the microphone diaphragm. An electric field is established between the actuator and the diaphragm by applying 800 V dc to the actuator. A test signal of 50 to 150 V ac is superimposed on the dc signal, and the electrostatic forces will push and pull the

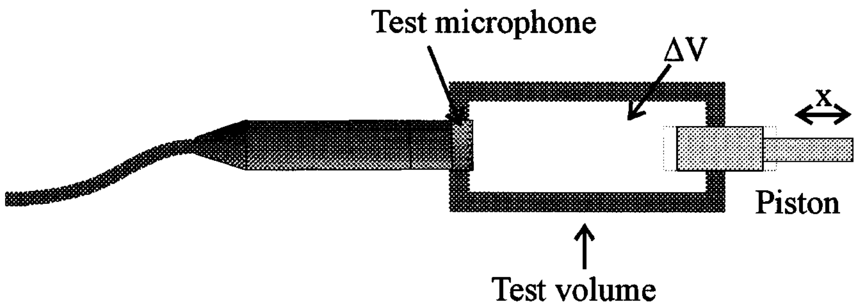

FIGURE 27.21 Principle of a pistonphone.

diaphragm, similar to a sound pressure of 1 to 10 Pa. By sweeping the test signal through the frequencies of interest, the pressure response of the test microphone can be recorded. The electrostatic actuator technique is widely used as a convenient and accurate test method, both during production and final calibration of measurement microphones.

Available instrumentation and manufacturers are given in Tables 27.4 and 27.5.

TABLE 27.4

Instrumentation	Types available	Approx. price	Manufacturers
Meas. microphones	½″ Free field ½″ Pressure ¼″ Free field ¼″ Pressure	\$750–\$825	GRAS Sound & Vibration ACO Pacific B&K The Modal Shop Larson Davies
Preamplifiers	½″ and 0.25 in.	\$600–\$850	GRAS Sound & Vibration ACO Pacific B&K The Modal Shop Larson Davies
Sound level meters	Simple type 1 SLM	\$800–\$2000	Rion CEL B&K
Sound level meters	Advanced SLM with freq. analysis and data storage	\$2000–\$10,000	Rion CEL Larson Davies B&K
Frequency analyzers	Real-time frequency analyzers/FFT	\$5000–\$50,000	Hewlett Packard Norsonic Data Physics 01dB
Near-field acoustical holography	Complete system with 16–64 channel acquisition and postprocessing	\$100,000–\$200,000	LMS B&K

TABLE 27.5 Companies That Makes Acoustical Measurement Instruments

G.R.A.S. Sound & Vibration Skelstedet 10B 2950 Vedbaek Denmark Tel: +45 45 66 40 46	Larson Davies Inc. 1681 West 820 North Provo, UT 84601 Tel: (801) 375 0177
LMS International Interleuvenlaan 68 B-3001 Leuven Belgium Tel: +32 16 384 571	Brüel & Kjær Spectris Technologies Inc. 2364 Park Central Blvd. Decatur, GA 30035-3987 Tel: (800) 332 2040
Hewlett-Packard Co. P.O. Box 95052-8059 Santa Clara, CA 95052 Tel: (206) 335 2000	Rion Scantek, Inc. 916 Gist Avenue Silver Springs, MD 20910 Tel: (301) 495 7738
Norsonic AS P.O. Box 24 N-3420 Lierskogen Norway Tel: +47 32 85 20 80	ACO Pacific, Inc. 2604 Read Avenue Belmont, CA 94002 Tel: (415) 595 8588
The Modal Shop Inc. 1776 Mentor Avenue, Suite 170 Cincinnati, OH 45212-3521 Tel: (513) 351 9919	CEL Instruments 1 Westchester Drive Milford, NH 03055 Tel: (800) 366 2966
01dB 111 rue du 1er Mars F69100 Villeurbanne France Tel: +33 4 78 53 96 96	

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