ACOUSTIC VARIABLES MEASUREMENT

The science and technology of acoustics encompass sound, ultrasound, and infrasound in all media. There are many specialties in acoustics as well as interrelations to other branches of engineering, physics, and chemistry. Consequently, the list of quantities and their corresponding units is rather extensive (1), including both fundamental (e.g., mass) and derived (e.g., force) physical quantities as well as measures of level (e.g., sound pressure level). This article concentrates on the measurement of variables that are peculiar to acoustics.

Acoustical variables have evolved from two general disciplines: wave theory and circuit theory. Because of this, many acoustical variables have their counterparts or analogs in electromagnetic theory, electrical-network theory, mechanics, and heat—a convenience that must be used with care. Naturally, as in other branches of engineering and physical science, variables applicable to acoustics only have been introduced. As an exact science, acoustics is relatively young; hence, its quantitative development has been primarily in terms of the metric system, which accounts for the almost exclusive use of the SI units.

MEASURABLE ATTRIBUTES OF SOUND

Sound pressure is the most important quantity in acoustics, probably because it is the attribute of sound measurable by all hearing mechanisms. The term sound pressure (or acoustic pressure) refers to the increment over ambient pressure, or excess pressure, produced by a disturbance of the medium, not to the total ambient pressure (2). Similarly, acoustic displacement refers to the net motion of the molecules of the medium, in response to the disturbing force and not to the random motion of particles when the medium is “at rest.” Another measurable quantity is the density, which has the usual definition of mass per unit volume. When a sound wave is propagated in the medium, the ambient value of density changes in proportion to the changes in pressure. Particle velocity and acceleration have the same meaning as in any mechanical system, their magnitude depending on the strength of the disturbance. Particle velocity is not to be confused with the speed of propagation of the acoustic wave, which depends entirely on the properties of the medium and environmental parameters, such as temperature. Since by their nature all of the above quantities are alternating, frequency (or period) is one of their most basic parameters. In modern acoustical practice, frequency is measured by a frequency counter or signal analyzer after the acoustic signal has been converted into electrical. The unit of frequency is the hertz, Hz. The quantities already listed are of a fundamental nature because other quantities can be derived from them. The derived quantities include intensity, impedance, and a variety of coefficients and ratios, some of which are too specialized and, hence, outside the scope of this work, and others are discussed in separate articles.

Sound Pressure and Sound Pressure Level

The American National Standard (3) defines instantaneous sound pressure as the difference between total pressure and the atmospheric static pressure that exists at a given point in space, at a particular instant of time, in a stated frequency band. The SI unit is the pascal, Pa, which results from a 1 newton force uniformly applied over an area of 1 m^2. In acoustical practice, the term sound pressure or effective sound pressure is the root mean square of the instantaneous sound pressures determined over a specified time interval, at a point in space. Because of the wide range of pressures attending acoustic phenomena (from micro- to kilo-pascals), it is frequently more convenient to compare a sound pressure to some reference value, rather than state its magnitude in pascals. This ratio, converted to the logarithmic scale, is called sound pressure level in dB. It is defined as follows:

$$SPL = 10\log\left(\frac{p}{p_{\text{ref}}}\right)^2 = 20\log\frac{p}{p_{\text{ref}}} \qquad (1)$$

The reference pressure, p_{ref}, must always be specified. The standard value for gases is 20 μPa; for other media, $p_{\text{ref}} = 1$ μPa, unless otherwise specified.

Particle Displacement, Velocity, and Acceleration

In the acoustics of fluids, the number of applications requiring a direct measurement of variables describing particle motion is very small. For instance, in the case of a loudspeaker it is the motion of the vibrating diaphragm rather than the displacement of the air particles that one measures. When particle displacement is required, it can usually be calculated from pressure measurements and the properties of the acoustic medium, especially since no simple instruments are available to perform this task. The opposite is true for solid media, where vibratory displacement, velocity, and acceleration are measured routinely by a variety of sensors and instrumentation. In terms of their relative usage in vibration measurements, accelerometers have considerable advantages over displacement and velocity sensors (4).

There are two measures of velocity of an acoustic medium: the particle velocity and the volume velocity. The particle velocity, defined at a point in the medium, is the velocity of a portion of the medium small enough that the variations of acoustic parameters within the volume of the portion can be considered infinitesimal. It is measured in m/s. The volume velocity is the volume of the medium displaced per unit time through a specified area. Thus, it is equal to the integral of the particle velocity over the area, and its unit is the cubic meter per second (5). Particle acceleration has the usual meaning of the rate of change of velocity, and is measured in $m \cdot s^{-2}$. In the United States, acceleration is also expressed in units of g_n, the standard acceleration of free fall, equal to 9.80665 $m \cdot s^{-2}$, and acceleration levels are often stated with reference to 1 μg. As in the case of acoustic pressure, when working with these variables, one must distinguish between instantaneous and effective root-mean-square (rms) values, which have the same meaning as for the electrical variables (6).

Measures of the Strength of an Acoustic Signal

The sound strength depends on the amount of energy carried by the wave. Two measures of this property are commonly used: the acoustic energy density flux and acoustic intensity. The energy transported by acoustic waves through a fluid medium comprises two components: the kinetic energy associated with particle motion, and the potential energy stored during compression and rarefaction of successive fluid elements. The energy density is measured in joules per cubic meter. Because the instantaneous particle velocity and acoustic pressure are functions of both position and time, the instantaneous energy density, ϵ, is not constant throughout the fluid. Its time average, $\bar{\epsilon}$, at any point in the fluid can be expressed by a simple formula only if the relationship between particle velocity, u, and pressure, p, is known. For example, in a plane simple harmonic wave,

$$p = \rho_0 c u \tag{2}$$

hence

$$\bar{\epsilon} = \frac{1}{T}\int_0^T \epsilon\, dt = \frac{1}{2}\rho_0 U^2 = \frac{1}{2}\frac{P^2}{\rho_0 c^2} \tag{3}$$

where ρ_0 is the medium density, c is the speed of wave propagation, and P and U are the amplitudes of the acoustic pressure and particle velocity respectively (7). In sound fields where such relationships do not apply, pressure and velocity have to be measured by sampling over the area or volume of interest and appropriately summed. Investigation of jet noise is a good example (8).

Acoustic intensity, I, of a sound wave is defined as the mean rate of flow of energy through a unit area normal to the direction of propagation (5). Consequently, it represents the power of a sound wave incident on a unit area of a detector. However, unlike their optical counterparts, the simple acoustical detectors do not directly respond to intensity, but rather to pressure, displacement, velocity, or acceleration. But, since energy, and hence intensity, varies as the square of these quantities, one needs only to find the factor of proportionality. For example, basic acoustical theory shows that for a plane free progressive wave the intensity, I, and acoustic pressure, p, are related by the following equation:

$$I = \frac{1}{\rho_0 c}p^2 \tag{4}$$

The unit of intensity is the watt per square meter. For the calculation of intensity levels, $10^{-12}\ \mathrm{W \cdot m^{-2}}$ is often taken as the reference intensity. Measurements of acoustic intensity and acoustic energy-density flux are the subject of a separate article.

Frequency Response Functions

Acoustic variables, individually or in combination, depend on various parameters, frequency being the most common. Since most variables by their nature are phasors, they are representable by complex quantities having a real and an imaginary part or, alternately, magnitude (also called amplitude) and phase. Magnitude information is sought more often than phase; however, the dependence of both magnitude and phase on frequency are both very important characteristics of an acoustic or electro-acoustic system. Typically the magnitude is expressed in normalized form based on some convenient reference, which must be stated. Moreover, the ratio can be expressed in decibels instead of a fraction when the quantities can be related to energy or power, as was done for sound pressure level in Eq. (1). For phase, a common reference value is the phase at a specified frequency. In Fig. 1 the theoretical amplitude and phase response of a Bruel & Kjaer microphone is compared to the measured response (9). It should also be noted that not all frequency-response functions represent ratios of the same denomination: they can, for example, be pressure-to-velocity ratios considered in the next section.

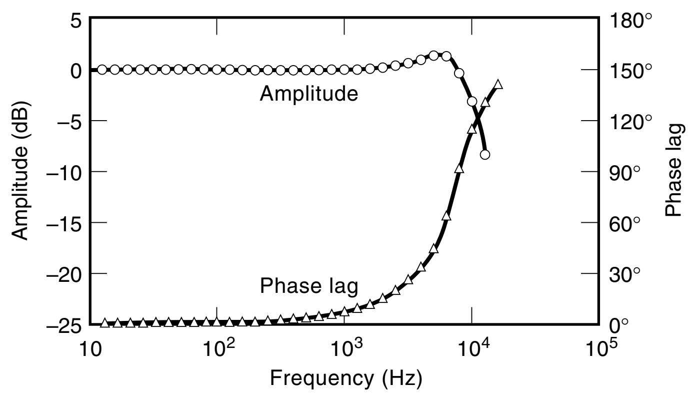


Figure 1. Example of a frequency-response function: relative amplitude and phase response of B&K type 4146 microphone. Solid lines: theoretical; symbols: experimental. See Ref. 9.

DESCRIPTIVE PARAMETERS AND FUNCTIONS

The behavior of acoustic systems is described not only by the basic variables discussed in the preceding section, but also by a collection of quantities derived from them. For a complete treatment of these parameters and functions the reader is referred to the rather extensive specialized literature, Ref. 10 being a good starting point, while in this article we consider only those deemed most important and commonly used.

Impedance

The concept of impedance was introduced into acoustics by A.G. Webster in 1919 by analogy with electrical circuits (11). Webster saw the value of extending the significance of the concept of impedance to the case of mechanical oscillations and acoustic wave propagation. The concept has been found to be of great value in both theoretical acoustics and applications. Impedance is a very useful quantity connecting the excitation and response of a system (see Fig. 2). Electrical im-

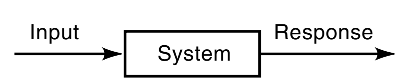


Figure 2. Block diagram of a single input/single output system for defining frequency-response functions of Table 1. The complex function of frequency, given by the output-to-input ratio, characterizes the system.

Table 1. Frequency-Response Functions in Common Use

Output→ ↓ Input	Pressure	Force	Displacement	Velocity	Acceleration
Pressure	SPL				
Force		Force ratio	Compliance	Mobility	
Displacement		Stiffness	Displ. ratio		
Velocity		Impedance		Vel. ratio	
Acceleration		Mass			Accel. ratio

pedance is defined as the ratio of the phasor representing an alternating voltage to the phasor representing a corresponding current (6). As such, it is a complex quantity having magnitude and phase angle (or, alternatively, a real and an imaginary part), is time invariant, but generally frequency dependent. Impedance, therefore, is one of the frequency-response functions listed in Table 1 which are used to describe the behavior of a linear system.

Before proceeding, it is proper to warn the reader of the dichotomy in choosing the sign of the reactance. It arises from the different representations of a traveling wave used in physics and engineering. A simple illustration readily makes this clear. Consider, for example, a plane wave of acoustic pressure, p, traveling in the direction of the x-axis. The physicist usually represents it by

$$p = P_{+}e^{-i(\omega t - kx)} + P_{-}e^{-i(\omega t + kx)} \tag{5}$$

where P_+ and P_- are the amplitudes of the forward and backward traveling waves respectively, k is the wave number, and ω is the angular frequency of the alternating pressure. The electrical engineer prefers the following notation:

$$p = P_{+}e^{j(\omega t - kx)} + P_{-}e^{j(\omega t + kx)} \tag{6}$$

The reason for these choices is that the physicist likes to think of the forward wave as traveling in the positive-x direction, while the engineer focuses on the phasor representation $P_+e^{j\omega t}$. Elementary analysis (12) readily shows that the former choice $(-i)$ leads to a negative sign for mass reactance and a positive sign for compliant reactance. In order to make the results of electrical circuit theory directly applicable to acoustics, the latter $(+j)$ choice is followed in this article.

Types of Impedance Used in Acoustics. In accord with national and international standards, impedance of acoustic systems is specified in three different ways; hence, the type of impedance selected by the user must be carefully stated. The first of these, Z_a, is called acoustic impedance and is defined as the quotient of the sound pressure, p, on a given surface, S, divided by the volume velocity, U, through the surface:

$$Z_a = \frac{p}{Su} = \frac{p}{U} \tag{7}$$

where u is the particle velocity and S is the area. Its unit is the acoustic ohm, which has the dimensions of $\mathrm{Pa \cdot s \cdot m^{-3}}$.

The second type of impedance is the specific acoustic impedance, Z_s, defined as the quotient of the sound pressure and the particle velocity:

$$Z_s = \frac{p}{u} \tag{8}$$

The unit is the rayl, equal to $\mathrm{Pa \cdot s \cdot m^{-1}}$.

The third type of impedance, called the mechanical impedance, Z_m, is defined as the quotient of the force exerted on a given area and the resulting particle velocity:

$$Z_m = \frac{pS}{u} \tag{9}$$

The unit of Z_m is the mechanical ohm, equal to $\mathrm{N \cdot s/m}$.

Each type of impedance has certain advantages in the solution of specific problems. For instance: the acoustic impedance does not vary with changes in cross section of a conduit down which a sound wave is traveling; specific acoustic impedance in a plane progressive wave is nearly the same at all points; mechanical impedance may be used directly in equivalent circuits. Moreover, like its electrical counterpart, acoustic impedance can be realized in either lumped or distributed form. When one or more dimensions of an acoustic system are not small as compared to the wavelength of sound, it is no longer possible to describe the system by lumped parameters.

Acoustic Compliance. Acoustic compliance is that parameter of an acoustical circuit which accounts for volume displacement without acceleration (5). The physical law governing the compression of a volume of acoustic medium is

$$p(t) = \frac{1}{C_a}\int U(t)\,dt \tag{10}$$

where $p(t)$ is the instantaneous pressure acting to compress the volume of the medium, $U(t)$ is the instantaneous volume velocity of the medium flowing into the volume, and C_a is the acoustic compliance. In the steady state of an acoustic pressure varying sinusoidally at an angular frequency ω,

$$P = \frac{U}{j\omega C_a} \tag{11}$$

where P and U are complex rms quantities. It is evident from Eq. (11) that acoustic compliance has the units of $\mathrm{m^5 \cdot N^{-1}}$. Note also that the acoustic impedance of a purely compliant element is

$$Z_a = -j\frac{1}{\omega C_a} \tag{12}$$

As a circuit element, acoustic compliance is a two-terminal device, but one of its terminals must always be at "ground potential," that is, one terminal is the outside of the enclosure containing the compressed medium. The reciprocal of compliance, called stiffness, is equally prevalent, the choice being a matter of convenience. Moreover, the adjective *dynamic* is sometimes used with the terms compliance and stiffness, especially in situations where the values of the static and dynamic parameters are substantially different.

Acoustic Inertia. The acoustic mass or inertance, m_a, is a quantity associated with a mass of an acoustic medium accelerated by a net force which acts to displace the medium without appreciably compressing it:

$$p(t) = m_a \frac{dU(t)}{dt} \tag{13}$$

As a consequence of this definition, the dimensions of acoustic mass are $kg \cdot m^{-4}$. In the sinusoidal steady state, the value of the parameter m_a multiplied by ω yields the magnitude of the imaginary part of the acoustic impedance resulting from the inertia or effective mass of the medium:

$$p = j\omega m_a U \tag{14}$$

As an acoustic-circuit element, acoustic mass can be realized as a tube open at both ends. The tube must have rigid walls and be short enough so that the medium in it moves as a whole without appreciable compression. Furthermore, the assumption of zero pressure at the ends requires that the tube diameter be small relative to the wavelength of sound in order to reduce its radiation impedance. The effect of the radiation impedance is to increase slightly the apparent length of the tube. One end of an acoustic mass is usually driven by a source, while the other end can be terminated in a boundless medium, a larger cavity, or a damping element.

Acoustic Resistance. Resistance may be contributed to an acoustic system by a number of different causes, but it is always associated with dissipative losses. The physical law defining acoustic resistance is

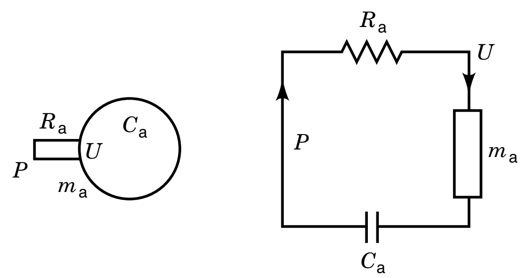
$$p(t) = R_a U(t) \tag{15}$$

Being the real part of acoustic impedance, Z_a, acoustic resistance has the same unit dimensions, $Pa \cdot s \cdot m^{-3}$, also called the acoustic ohm. In lumped form it can be realized by having the sound go through short capillary tubes or a fine-mesh screen.

The Helmholtz Resonator. A very instructive example of an acoustic system containing the three basic elements (compliance, inertance, and resistance) is provided by the Helmholtz resonator, shown in Fig. 3 alongside its electrical analog, the series RLC circuit. The lumped-parameter values can readily be calculated in terms of its dimensions and properties of the acoustic medium, as is done in Ref. 13 and other textbooks on acoustics. Thus,

$$C_a = \frac{V}{\rho_0 c^2},\ m_a = \frac{\rho_0 L}{S},\ R_a = \frac{R_r}{S^2} + R_{eq} \tag{16}$$

where R_r is the radiation resistance of the neck opening, and R_{eq} represents the effect of all damping present in the system, and all other quantities have the previously defined meaning.



(a) (b)

Figure 3. A Helmholtz resonator consisting of acoustic compliance, mass, damping, and radiation resistance (a), and a schematic representation of its electrical equivalent (b).

Acoustic Absorption

When sound traverses matter, be it solid, liquid, or gaseous, the acoustic intensity decreases with distance. This fact is a consequence of the properties of the wave itself and the medium through which it travels. In the absence of boundaries this propagation loss is attributable mainly to spreading (e.g., the so-called inverse r^2 loss), scattering, and absorption by the medium. When boundaries are present, any diminution of intensity upon reflection from them is governed by the nature of the reflecting surfaces which in addition to presenting an impedance mismatch and absorbing acoustic energy can diffuse the acoustic field by virtue of their roughness. Absorption is a consequence, primarily, of the viscosity of the medium, and is usually accounted for by a complex value of the propagation velocity. Accordingly, the wave number must be complex also. In a boundless medium, this is accounted for by an extra factor in the spreading-loss expression. Likewise, when sound is reflected from a medium which has absorption, the transmitted and reflected waves are affected: inhomogeneous waves are produced along the boundary, and total internal reflection cannot take place (14).

Absorption Coefficient. The simplest theory of absorption yields the following expression for the acoustic pressure of a wave propagating in a viscous medium:

$$p = p_0 e^{-\alpha x} e^{j(\omega t - kx)} \tag{17}$$

where α is the absorption coefficient, x is the distance traveled by the wave, ω is the angular frequency, and k is the wave number. Accordingly, one of the standard methods of measuring sound absorption coefficient is the classical impedance-tube method (15). Yet even in this simple model, the absorption coefficient is a function of not only viscosity, but also of frequency and the speed of sound. To obtain good agreement between predicted and measured values of α, other processes may have to be built into the theoretical model. For example, in air, humidity becomes important, while in sea-water, chemical relaxation measurably affects the results at low frequencies.

An alternate definition of the sound absorption coefficient is as the fraction of the sound energy incident on a surface which is absorbed or dissipated. A convenient measure of the effectiveness of any absorptive treatment is, quite naturally, the reverberation time, that is, the time interval during which the sound is audible after its source has been turned off. Quantitatively, the reverberation time is defined as the time required for the level of the sound to drop by 60 dB.

The preceding discussion considered the effect of absorption on the propagation of sound in a definite direction. There are, however, many situations where sound is not expected to travel from one specified point to another, but rather to fill a volume of space with a uniform intensity. Such a sound field is called diffuse or reverberant, and is usually produced by multiple reflections. This process lends itself to straightforward analysis, which was originally performed by Sabine and can be found in one of several variants in most books on acoustics, for example in chapter 6 of Ref. 7. The analysis is based on a few realistic assumptions and involves the room dimensions as well as properties of the absorbers. When the average absorption coefficient, $\overline{\alpha}$, for the room is small, the Sabine formula in metric units reads:

$$T = 0.161 \frac{V}{S\overline{\alpha}} \tag{18}$$

where T is the reverberation time, V is the room volume, and S is the area of the bounding surfaces. This formula can also be used to measure the average absorption coefficient of the room, and in turn the absorption coefficient of an individual absorber. Since

$$\overline{\alpha} = \frac{S_1\alpha_1 + S_2\alpha_2 + \cdots + S_n\alpha_n}{S_1 + S_2 + \cdots + S_n} \tag{19}$$

where S_i and α_i are corresponding areas and absorption coefficients, any α_i can be calculated if all the others are known and $\overline{\alpha}$ is measured. This is the basis of a standard method of measuring the absorption coefficient of materials.

DATA ACQUISITION, VALIDATION, AND ANALYSIS

Some of the most important measurement activities should take place before data are acquired. Planning for data acquisition and analysis should address several issues: the selection of transducers, evaluation of instruments to be used in the measurement system, measurement location and duration, and protocol for estimating uncertainties in the measurement. This is particularly important if a new method of measurement is contemplated or the procedure of a standard method is not adequately documented. Prior to data acquisition, the measurement system must undergo a complete calibration of all components as well as an end-to-end calibration of the system. The lack of a proper end-to-end electrical calibration is probably the most important source of errors in dynamic data measurements. Modern instruments, though fast and versatile, still require their user to be fully conversant with the different types of averaging, sampling rate, filtering, and statistical analysis that are available.

It is of paramount importance to carry out a reliable validation procedure prior to the analysis of data. Validation in this context means ensuring that the recorded signal correctly represents the measurand of the variable required for subsequent analysis. Signal clipping, excessive background noise, inadequate bandwidth of instrumentation, undesirable coupling, spurious signal pickup all make validation an essential component of the measurement procedure. Poor coherence between the input calibration signal and the output at the relevant channels of the data acquisition system can readily invalidate seemingly good results. The coherence function should have a value in excess of 0.99 at all frequencies of interest (16). If the practice is to record the measured signals on either analog or digital storage devices for later analysis, the signals can be validated prior to analysis. When data are acquired and analyzed on-line, complete validation prior to analysis may be impracticable; however, many of the desired data validation operations can be automated by knowledge-based computer programs, resulting in substantial reduction in time and labor (17).

Characterization of Measuring Instruments

In its elemental form, an instrument for measuring an acoustic variable is a transducer which converts acoustical energy into another form, usually electrical. Since both the input and output measurands are phasor quantities, it is very convenient to describe performance of the transducer by its transfer function, that is the ratio of the output to the input. This function is called the sensitivity of the transducer and, like any complex quantity, may be expressed in terms of magnitude and phase, as in Fig. 1. Each of these components exhibits a definite dependence on frequency and, to a lesser extent, on environmental parameters, such as temperature, humidity, static pressure, magnetic field, and so on. For a specific application consider first the microphone, probably the most important instrument in acoustical metrology.

Measurement of Microphone Response. When a microphone is placed in a sound field, the field will be disturbed. The disturbance results in the sound pressure on the diaphragm being different from that of the undisturbed field. This pressure deviation is accounted for by the so-called free-field correction, which depends on the dimensions of the microphone, the shape (including any protective grid), and the angle of incidence of the sound. When sound arrives from many directions, the free-field response must be averaged over all directions to yield the correction necessary to compensate for the disturbing influence of the microphone. This is known as the random-incidence correction, required in situations such as machine shops and reverberation chambers. See Fig. 17.9 of Ref. 18.

There are two methods for expressing microphone frequency response: the free-field sensitivity and the pressure sensitivity. The free-field sensitivity is defined as the ratio of the microphone's electrical output to pressure applied on the diaphragm when the applied pressure is that which would exist at the microphone location in the absence of the microphone. The pressure sensitivity of a microphone is defined as the ratio of the electrical output to pressure at the diaphragm when the pressure is uniform and in phase over the entire sensing surface of the microphone. Thus, the actual response of a microphone in a sound field can be represented by the sum of its pressure response and the applicable free-field correction. The free-field and pressure responses of a microphone are identical at low frequencies, but begin to diverge at frequencies where the sound wavelength becomes comparable to the dimensions of the microphone. This information can be presented in absolute terms or relative to the value at some reference frequency.

The free-field response data is the most applicable description of microphone sensitivity as a function of frequency when the microphone is operated in an open space, several wavelengths removed from a sound source and any reflecting boundaries. The free-field correction curves are relevant only when the sound field is diffracted or reflected by the microphone body. When the microphone forms part of a natural boundary, the pressure response for sound normally incident upon the diaphragm should be used. For example, when measuring the sound pressure acting on an aircraft wing, the microphone should ideally be mounted so that its diaphragm is flush with the surface of the wing. As long as the diaphragm's input impedance is large compared with the radiation impedance loading the diaphragm, the microphone will not disturb the field, and the pressure response will adequately represent the microphone's performance. There exist standard methods of calibrating microphones, based on the above definitions, as well as the so-called pistonphone calibrator which is easier to use, but less accurate. The pistonphone calibration is most suitable for use at frequencies where the wavelengths are very long compared with the dimensions of both the microphone and the volume of air entrapped within the pistonphone. Recently the pistonphone technique has been adapted to phase-response measurements (Chapter 9 of Ref. 18).

Regardless of the method employed the microphone response is typically displayed in two plots: amplitude vs. frequency, and phase vs. frequency, as illustrated in Fig. 1. Until the 1980s users were interested mostly in the magnitude of the microphone sensitivity, but in recent years the demand for phase response information has greatly increased, primarily because of its importance to sound intensity measurements. What is needed in practice is the relative phase response of two microphone systems, that is, the phase difference between them. Methods involving phase angles of microphone response are the subject of active research (19).

Sound-Level Meter. Although the availability of a calibrated microphone together with an amplifier and indicating meter is all that is required to construct an instrument for measuring sound pressure, such instruments have not come into common use. Instead, a related instrument with increased capabilities has evolved. It is the sound level meter, which has become the basic instrument for all sound- and noise-level measurements, particularly those made in the field. Figure 4 shows a photograph of a representative sound level meter. These instruments come in different shapes and sizes as well as signal-processing capabilities, but all comprise a microphone, an amplifier, frequency weighting networks, and an output-indicating meter, and are constructed according to national and international standards (20). The sound level meter measures the weighted sound pressure level, defined as follows:

$$SL = 10\log\frac{(p)^2_{\text{av}}}{(p)^2_{\text{ref}}} \tag{20}$$

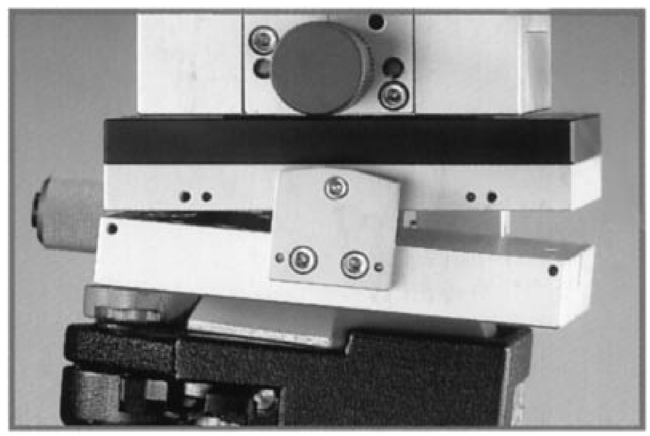
where p is the acoustic pressure being measured, and p_{ref} is a reference pressure, which must always be specified. Normally, for airborne sound $p_{\text{ref}} = 20\ \mu\text{Pa}$, and for underwater sound, $p_{\text{ref}} = 1\ \mu\text{Pa}$. Although the unit of the displayed quantity is the decibel, Eq. (20) will yield the same value as Eq. (1) only when no spectral weighting, resulting in a flat frequency response, is used. Usually, however, one of three or four (A, B, C, D) weighting characteristics is employed, which modifies the response at low frequencies, and consequently the indicated average reading. The precise dependence of these weighting networks on frequency has been standardized (20). (Originally they were intended to make the sound level meter respond approximately like the human ear to the loudness of the sounds being measured.) When reporting data in an isolated context, one should state not only the reference pressure, but also the weighting, for example: "the A-weighted SL = 43 dB."

Figure 4. An example of the basic instrument for acoustical measurements, a sound level meter (courtesy of Bruel & Kjaer Instruments, Inc.).

Displacement Transducers. In acoustics it is the vibratory displacement of a surface, rather than the particle displacement of the acoustic medium, that is commonly measured. To accomplish this function a number of successful methods have been developed, using either contacting or noncontacting probes. The method of choice appears to be some type of optical interferometry, especially since the advent of affordable good-quality lasers. An interferometer, based on one of several classical configurations, usually Michelson or Mach-Zehnder, can be custom built from off-the-shelf components, is easy to set-up, and offers the user the assurance of accuracy traceable to the wavelength of the laser light. Many systems are available commercially, and some are designed to respond to either displacement or velocity. In principle, the behavior of an interferometer does not depend on frequency, but practical instruments do have limitations on dynamic range as a function of frequency.

Velocity Transducers. Faraday's law of electromagnetic induction is the principle of the moving-coil transducer, an intrinsically velocity transducer that finds application in both microphones and vibration pickups. The voltage induced in the coil is, therefore, proportional to the relative velocity of the coil motion. An instrument that has gained great popularity in recent years is usually referred to as the laser vibrometer, marketed by manufacturers worldwide for both laboratory and field use. It operates on the Doppler principle whereby light reflected from the vibrating surface experiences a frequency shift which is proportional to the instantaneous velocity of the vibration. In the fiber-optic version the lens probe is decoupled from the interferometer system by an optical fiber probe allowing the experimenter to reach locations not accessible otherwise. Figure 5 shows one of several commercially available laser vibrometers.

(a)



(b)

Figure 5. Components of a laser Doppler vibrometer, a versatile vibration measuring instrument: (a) sensor and optical fiber probes; (b) positioning stage for x, y, z, tip, and tilt adjustment (courtesy of Polytec PI, Inc.).

Accelerometers. Transducers that directly sense particle acceleration and that are employed to measure vibrations of solid surfaces are by far more common than those which directly sense their velocity or displacement. There exist several general types of accelerometers with many specialized implementations of each type. The piezoelectric accelerometer usually is the optimum choice because of its very wide frequency and dynamic ranges. For specific information the reader is referred to publications on vibration measurements (21,22) and product literature.

Mechanical-Impedance Heads. In the measurement of mechanical impedance, a force transducer and a velocity transducer must be used simultaneously. Some manufacturers supply devices containing a force transducer and an accelerometer, in which case the output of the latter must first pass through an integrator. Like all joint measurements, mechanical impedance measurements present special problems. The force and velocity transducers must be much closer together than the shortest wavelength of interest in order for the two transducers to be considered as a single transduction system. Moreover, because of the complex nature of impedance, both amplitude and phase responses of each transducer must be accurately known. Mechanical resonances of structures other than the one whose impedance is measured can also affect the results.

Application of Correlation Techniques

Correlation methods originally were used to obtain statistical measures of randomly varying signals, for example to characterize the sound field in an enclosure. However, it soon became apparent that because of their inherent advantages, methods based on measuring auto- and cross-correlation of acoustic signals could rival many measurements traditionally using deterministic signals. The chief among these advantages is the ability to discriminate against noise, the latter being defined as the unwanted part of a signal. This is particularly important in situations of low signal-to-noise ratio. Another is the opportunity to design pseudo-random test signals which can generate the desired results more rapidly than measurements using deterministic signals. To illustrate we present two specific applications, particularly relevant to this article: measurements of the speed of sound and of transmission loss.

The feasibility of measuring the speed of sound follows from the form of the cross-correlation function of the same signal recorded at two points along the path of propagation. If the separation between the microphones is d, the maximum of the correlation function $R_{12}(\tau)$ occurs when the time delay $\tau = d/c$. Knowing the separation and noting the value of time delay, the speed of propagation, c, is readily obtained (23). In this respect the correlation method resembles the pulse method. They differ primarily in the nature of the test signal, which should be random and broadband for the correlation

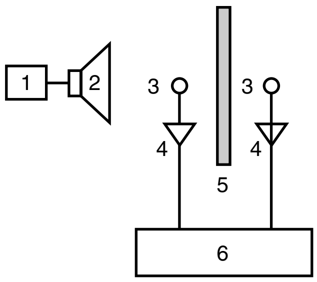


Figure 6. Equipment layout diagram for the measurement of sound transmission loss of an absorptive panel by a correlation method: (1) random noise generator and amplifier; (2) loudspeaker; (3) microphones; (4) signal conditioners; (5) test panel; (6) correlator and data recorder.

method, but for the pulse method it usually takes the form of a pulse-modulated sine wave. This method is particularly convenient for measurements in solids or liquid-filled tubes, where more than one mode may simultaneously be excited (24). As an example, consider the measurement of the speed of propagation of breathing waves in a tube with soft walls (25).

As a second example, consider the problem of measuring sound isolation and damping (23,24). The experimental setup, shown schematically in Fig. 6, consists of a loudspeaker driven by a noise generator and power amplifier, a microphone with signal conditioner, and a correlator. The microphone picks up the direct sound plus any reflected and diffracted sounds. All those signals have different arrival times and, consequently, the maxima of their correlation with the source signal are appropriately delayed. The transmission loss, TL, equals the difference between the first maxima of the correlation functions, $R_0(\tau)$ and $R_p(\tau)$, corresponding respectively to the direct paths in the absence and presence of the test panel. (The delay in the signal that has passed through the panel is caused by the sound speed in the panel material being slower than in air.)

$$TL = 20 \log \frac{R_0(\tau_1)}{R_p(\tau_1)} + 20 \log \frac{K_0}{K_p} \text{ dB} \tag{21}$$

where K_0 and K_p are the respective microphone-channel gains.

For accurate results it is important that the level of the transmitted sound greatly exceed the ambient noise level. The constraint on the sound source is that it have sufficient bandwidth.

Slight variations of the technique just described can be used to measure the reflection coefficient of a wall and the absorption coefficient of a reverberant room (24). It should also be noted that the correlation measurements here described, as well as all others, have their counterparts in the frequency domain and can be measured in terms of spectral functions.

ELECTROMAGNETIC-ACOUSTICAL ANALOGIES

There exist well-defined analogies between acoustical and electrical systems. Researchers familiar with electromagnetism and circuit theory can readily transfer much useful information to acoustics by a judicious application of a few rules and by keeping in mind the fundamental distinctions between sound and electromagnetic waves, particularly with regard to the variables in each system. For example, the most often measured quantity of an acoustic field, the sound pressure, is a scalar satisfying the Helmholtz equation, while the electromagnetic field quantities are vectors satisfying Maxwell's equations. These differences notwithstanding, the results of most boundary-value problems are mathematically analogous (26).

As an illustration consider the two-dimensional reflection problem. The acoustic pressure, p, can be shown (27) to be analogous to the tangential component of the electric field, E_y, and the normal component of particle velocity, u_z, to be analogous to the magnetic field, H_x, as evidenced by the following relationships:

$$u_z = \frac{1}{j\omega\rho} \cdot \frac{\partial p}{\partial z} \quad \text{and} \quad H_x = \frac{c}{j\omega\mu} \cdot \frac{\partial E_y}{\partial z} \tag{22}$$

Consequently, the reflection coefficients in the two cases differ only by their constant factors, which themselves have similar physical meaning, if one recalls that ρc and $\sqrt{\epsilon/\mu}$ characterize the wave impedance of the respective media.

The same analogy can generate other important results. For example, the well-known Snell's and Brewster's laws have identical forms in both systems. However, in three-dimensional problems, it is usually not possible to find acoustical counterparts to a general solution of Maxwell's equations, except for the so-called geometrical optics case, when Kirchhoff's approximation holds (27). It is because under that condition it is permissible to treat the electromagnetic-field quantities as scalars.

The analogies extend beyond the variables describing the behavior of systems in terms of field quantities. Individual elements of electromagnetic and acoustic circuits are characterized by analogous parameters that preserve the same frequency and space dependence, for example, when mechanical impedance is represented by its electrical analog. This can be especially useful in transfering results of electric-circuit theory to the design of acoustical and vibration measurements. The formulation of circuit-theory analogies follows definite rules, discussed by Olson (28), Kurtze (29), and many others. Table 2 shows some commonly employed analogies between acoustical and electrical variables and parameters.

As an illustration consider the task of calculating the transmission loss provided by a rigid wall whose area mass density is m. This basic problem of sound transmission between two media can also be solved by the application of

Table 2. Electroacoustical Analogies

Acoustical	Electrical
Sound pressure	Potential difference (voltage)
Volume velocity	Electric current
Volume displacement	Electric charge
Acoustic mass (inertance)	Inductance
Acoustic compliance	Capacitance
Acoustic resistance	Resistance
Acoustic impedance	Electric impedance

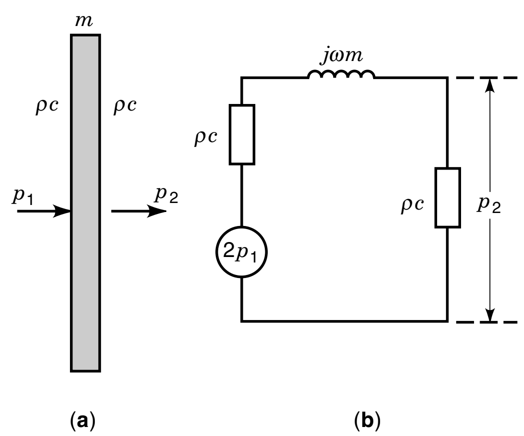


Figure 7. Derivation of an expression for the sound transmission loss by the application of Thevenin's theorem to an equivalent electrical circuit: (a) an acoustic panel of area mass density m, placed in a medium of specific acoustic impedance ρc; (b) an analogous electrical circuit.

Thevenin's theorem, resulting in the circuit of Fig. 7(b), where acoustic pressure is the analog of voltage, and particle velocity is the analog of current (30). Accordingly,

$$u = \frac{2p_1}{2\rho c + j\omega m} \quad \text{and} \quad p_2 = p_1 \frac{2\rho c}{2\rho c + j\omega m} \tag{23}$$

Hence, the transmission loss, in decibels, becomes

$$TL = 20 \log \left| \frac{p_1}{p_2} \right| = 20 \log \sqrt{1 + \left(\frac{\omega m}{2\rho c} \right)^2} = 10 \log \left[1 + \left(\frac{\omega m}{2\rho c} \right)^2 \right] \tag{24}$$

While the answer originally sought is correct, the reader is cautioned not to regard the two parts of Fig. 7 as equivalent in every respect. To understand this warning, consider, for example, the particle velocities on both sides of the partition. Clearly, they are in the same ratio as the pressures since the acoustic impedances on both sides are equal to ρc, but according to the schematic diagram the same particle velocity "flows" on either side of the wall.

It likewise follows that the matrix equations representing acoustical and electric systems are analogous. This is very fortuitous because the elegant and easily understood theory of multiports can serve as the foundation for important analytical and experimental investigations. For example, two- and three-port models of microphones and accelerometers facilitate the development and implementation of methods for calibrating them. When two microphones are coupled by an acoustic medium, the system can be represented by a four-port model, which provides the basis for a reciprocity calibration of standard microphones, elucidated in chapter 4 of Ref. 18.

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