



Sound Power Measurements

Application Note 1230

$$L_{WA} = 10 \log \sum_{i=1}^n 10^{L_{WAi}/10} + 10 \log \frac{S}{S_0}$$

$$I = - \left(\frac{P_1 + P_2}{2\rho\Delta r} \right) \int_0^t (P_2 - P_1) dt$$

Preface

Sound power level measurements are gaining recognition worldwide as a means of characterizing a product's acoustic signature. The usefulness of sound power is well known among noise control experts. Unlike sound pressure measurements, sound power tests are independent of measurement environment, making possible direct noise level comparisons between different products. Given a 1/3-octave spectra of sound power, noise control engineers can determine the resulting sound pressure level in an enclosure or space and choose the quantity and type of acoustic treatment required. With many international import regulations requiring conformance to noise power test standards such as ISO 7779, the measurement has become increasingly important.

Growing interest in sound power level is not restricted to noise control professionals. Appliance and office equipment manufacturers—and their customers—are paying attention too. For competitive reasons, many companies now specify sound power levels in product documentation. Their assumption is that consumers will select a quieter appliance.

A final reason for the growing popularity of sound power measurements is the sharp drop in costs associated with making the measurement. Instrumentation is less expensive and new test techniques are eliminating the need for costly anechoic or reverberant chambers.



Contents

Preface	1
Introduction	3
What is Acoustic Sound Power?	3
How is Sound Power Measured?	3
Using this Application Note	4
Evaluating Sound Power Using Sound Pressure Measurements	4
Basic Assumptions	4
Measuring Sound Pressure	5
Test Environment	7
Measurement Surface	8
Microphone Location and Orientation	8
Instrumentation Configuration and Setup	8
Test Procedure	10
Test Report	11
Evaluating Sound Power Using Sound Intensity Measurements	12
Eliminating the Need for Chambers	12
What is Sound Intensity?	12
Measuring Power in the Presence of Noise	13
How is Sound Intensity Measured?	13
Off-Axis Measurement Error	20
Sound Power Test Procedures using Sound Intensity	21
Choosing Between Pressure and Intensity Techniques	25
Comparing Sound Power Test Techniques	26
Test Case 1	26
Test Case 2	28
Comparison of Test Results	30
Appendix A:	
Acoustic Terminology	31
Appendix B:	
Sound Intensity Approximation Errors	33
Comparing True Intensity to Measured Intensity	37
Appendix C:	
Theoretical Analysis of Sound Intensity Measurements	37
References	40

Introduction

The trend in international trade is toward open boundaries and freer competition. This means product quality and performance will have a more direct impact on success or failure in the world market. Acoustic noise standards and regulations will play a significant role in this movement for two reasons: ergonomics in the work environment are becoming increasingly regulated and are providing a new basis for competitive comparison; and because export and import licenses will require compliance with specific acoustic measurement standards. Sound power level will be the preferred measurement for such standards, as opposed to sound pressure level, because power level is consistent, comparable, and more useful for noise control engineering.

Sound pressure level (SPL) measurements generally provide the best indicator of human hearing response, but SPL measurements depend on the acoustic environment. Unless the test room is perfectly absorptive (an anechoic chamber) or perfectly reflective (a reverberant chamber) any measurement of SPL will be dependent on the characteristics of the room. Wall and floor absorption, room geometry and volume become factors that influence SPL measurements due to reflections, standing waves, and absorption. For example, in an anechoic chamber one might measure 75 dB at a distance of 1 meter from the air inlet of a small fan. In a room with several hard walls, however, the reading may be 79 dB at 1 meter from the inlet. This difference makes comparison between different fans less reliable. An alternative technique would be to measure all fans at 1 meter from their inlet in a qualified anechoic chamber. The problem with this specification is that it does not account for potential sources of noise that are emitted in other directions. In contrast, such directivity is accounted for when measuring total sound power.

Sound power is a useful technique for noise abatement because it is possible to compute SPL at a specified distance from an object given its total sound power—and therefore calculate the amount of sound absorption needed to limit the noise level.

What is Acoustic Sound Power?

Acoustic sound power is the acoustic energy per unit time emitted by a device. Acoustic power levels of many devices are quite small; for example a coffee grinder may emit only 0.005 W of acoustic power. Other devices emit enormous amounts of acoustic energy. A rocket at takeoff can emit 100 MW of acoustic power, much of which is low-frequency noise that propagates for miles.

Sound power has units of "Watts," but is usually referred to in decibel units, with a reference of 1 pW (10^{-12} Watts).

$$L_W = 10 \log \left(\frac{W}{W_0} \right),$$

where $W_0 = 10^{-12}$ Watt and W = total acoustic power.

ISO 9296 requires documentation of acoustic power ratings in units of "Bels," where 1 Bel is equal to 10 dB reference: 1 pW.

$$\text{Bels} = \log \left(\frac{W}{W_0} \right)$$

How is Sound Power Measured?

Acoustic sound power is not measured directly but is derived from the measurement of sound intensity on a surface enclosing the test article as shown in figure 2. The units of sound intensity are W/m^2 , so total sound power can be calculated by multiplying the average sound intensity over the entire closed surface by the surface area:

$$W_{\text{total}} = \bar{I} A_s,$$

where $\bar{I}$ is the average intensity over the enclosed surface, and A_s is the total area of the enclosed surface.

Two basic techniques are used to measure sound intensity. The first technique uses a sound intensity probe and a sound intensity analyzer to measure sound intensity directly. The second technique uses a microphone and a 1/3-octave analyzer to measure sound pressure level in a special acoustic environment, such as an anechoic chamber. Measurement procedures for each of the two techniques are quite similar:

Evaluating Sound Power Using Sound Pressure Measurements

- 1) Select a facility that matches the requirements of the test—for example, an anechoic chamber with a hard, acoustically reflecting floor for pressure measurements, or a conference room for sound intensity measurements.
- 2) Place the device on an acoustically-rigid stand.
- 3) Define a measurement surface, usually box-shaped, surrounding the device and terminated by the reflecting plane (floor). Choose measurement positions.
- 4) Calibrate microphones and do field checks to confirm the acceptability of the measurement environment.
- 5) Measure pressure or intensity at each position.
- 6) Verify that the number of measurement positions is sufficient.
- 7) Compute total sound power.
- 8) Document results.

Using this Application Note

This application note examines two techniques for measuring sound power: pressure measurement in a free field over a reflecting plane, and the sound intensity method. Test procedures for each technique are presented with a detailed look at instrumentation and facility requirements. To help compare the two techniques, two measurement surveys of the sound power of a laser printer are documented. One survey uses pressure measurements in a free field over a reflecting plane, and the other uses sound intensity in a conference room.

Selecting the best technique for measuring sound power levels requires trade-offs between capital costs of instrumentation, facilities cost, direct test cost and time, and measurement accuracy. In the concluding section of this document each measurement technique is assessed relative to these factors so you can choose the test technique you prefer.

Using sound pressure measurements to determine sound power levels is a well-established technique. ISO 3744 and 3745 are two international standards that define procedures for testing noise sources using free-field environments. ISO 3744 has fewer constraints on the test environment and is used for sound power measurements with engineering-grade accuracy, and ISO 3745 is a precision technique with tight constraints on the semi-anechoic environment. Each test standard requires careful attention to the test environment or chamber to ensure that the assumption of proportionality between pressure and intensity remain valid within a specified accuracy.

The basic test procedures covered in ISO 3744 and ISO 3745 are covered here, these techniques are the same as most test standards for sound power measurements. Any differences in requirements are usually dependent on the objective of the standard. For example, ISO 7779 is a sound power measurement standard for office machines. It is based on ISO 3744 and 3745, but in addition to sound power it also requires specific sound pressure level measurements for operator and bystander positions as well as impulsive and tonal measurements. ISO 7779 requires these additional measurements because office machines can exhibit impulsive and tonal noise which can be ergonomically objectionable.

Basic Assumptions

In a free field environment where plane progressive waves are freely propagating, the relationship between intensity and pressure is:

$$I = \frac{P_{rms}^2}{\rho c}$$

where ρc = the characteristic impedance of air
= 415 N/m² at 20°C and barometric pressure 1013 mbar.

Coincidentally, the intensity level and sound pressure level in a free field are numerically almost equal. This can be shown as follows:

$$L_I = 10 \log \frac{I}{I_0}, I_0 = 10^{-12} \frac{W}{m^2}$$

$$L_P = 10 \log \frac{P^2}{P_0^2}, P_0 = 20 \times 10^{-6} P_a$$

$L_P - L_I$ = Pressure - Intensity Level Difference,

$$L_P - L_I = 10 \log \frac{P^2}{(20 \times 10^{-6})^2} - 10 \log \left(\frac{P^2 / 415}{10^{-12}} \right) = 0.16 .$$

The 0.16 dB difference varies with the impedance of air ρc , which in turn depends on air temperature and pressure. At $\rho c = 400$, $L_P = L_I$.

The quantity $L_P - L_I$ is called the Pressure-Intensity Index. It is used frequently in sound intensity measurements because it describes the sound field. A low P-I index means the environment is active, or nearly free-field. A high P-I index indicates that much of the pressure is due to the existence of a reactive sound field.* When using the free-field assumption for evaluating sound power via sound pressure level measurements, it is critical that the acoustic environment have a P-I index as close to zero as possible.

* Active and reactive sound fields are covered in greater detail in the sound intensity section.

Measuring Sound Pressure

Sound pressure level is the simplest and, by far, the most prevalent measurement in acoustics today. The condenser microphone has been used for 50 years to convert pressure fluctuations from sound waves into electrical signals to be amplified and analyzed. Acoustic instrumentation continues to become more powerful and less expensive, because of the incorporation of analog-to-digital converters and digital signal processing integrated circuits in mass produced consumer electronics.

The Mechanics of Sound Waves

Sound is created by the small mechanical displacement of air molecules. The piston in figure 1 creates sound waves in the tube when it vibrates. When the piston moves in the direction of the open end of the tube, the airspace in front of it becomes compressed. This compression produces an axial pressure gradient between the compressed region and the undisturbed region downstream. The pressure gradient causes air molecules to accelerate. The motion of this air mass causes the wave of compression to move

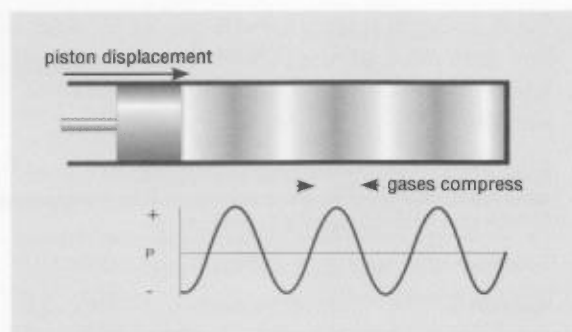


Figure 1: Piston displacement causes compression waves.

down the tube. When the piston then moves in the opposite direction, the airspace in front of the piston becomes rarefied. This pattern of compression and rarefaction produces sound. These minute pressure fluctuations may be audible if the frequency of fluctuation is between 20 Hz and 20 kHz and the pressure amplitude is at least 20 μ Pa, or 2×10^{-10} atmospheres.

Converting Sound Waves to an Electrical Signal: Condenser Microphones

Condenser microphones are the most common transducer used for acoustic measurements. They generally have high sensitivity, so that low sound pressure levels can be measured accurately. They also exhibit linear behavior over a wide dynamic range. Another characteristic of condenser microphones is excellent flatness. When used in their specified frequency range, a typical condenser microphone exhibits a flat response to within ± 2 dB. Most importantly, high quality condenser microphones remain accurate over a broad range of environmental conditions because of their low sensitivity to temperature, pressure, and humidity.

Table 1 shows typical specifications for a 1/2-inch free field microphone. (Note this microphone size represents excellent compromise in terms of sensitivity, frequency range, and size.)

Table 1

1/2-inch Free Field Microphone Specifications

Frequency Response	4 Hz to 20 kHz (+/- 2dB)
Sensitivity	50 mV/Pa
Capacitance	18 pF
Upper SPL Limit	145 dB
Lower SPL Limit	15 dB
Resonance Frequency	14 kHz
Polarization Voltage	200 V
Temperature Sensitivity	0.01 dB/°C
Aging	0.01 dB/yr
Static Pressure Sensitivity	0.001 dB/mbar
Diameter	0.5 inch/12.7mm

Instrumentation for Measuring Sound Pressure

Instrumentation for evaluating sound power levels using sound pressure must make measurements with sufficient accuracy. Requirements are typically as follows:

- Real-time 1/3-octave analysis meeting ANSI S1.11-1986. Accuracy in the 1/3-octave filter standard is specified for each individual filter. The standard defines amplitude accuracy, pass band ripple tolerance, 1/3-octave filter shape, tolerance on equivalent sound power, and other important factors governing the design of 1/3-octave filters. An analyzer that meets the requirements of a Type 1-D filter provides consistent, accurate results for sound power measurements.
- Digital or analog A-weighting filters that meet ANSI S1.4 or IEC 651. A-weighting filter networks are required to measure A-weighted sound power, L_{WA} .

Many standards also specify sound pressure level measurements at the operator or bystander positions. Often these levels require special filtering and detection circuits. For example, the ISO 7779 *Acoustics—Measurement of airborne noise emitted by computer and business equipment* requires the following measurements:

Table 2: ISO 7779 Measurements

Measurement	Band Levels	A-weighted Overall Levels
Sound Power	as specified	required
SPL	as specified	required
Impulsive SPL	as specified	as specified
Discrete Tones (FFT)	narrowband	

Note: "As specified" means that the specifying entity can include these measurements in a specification.

In the past, each of these measurements were made with a different analog analyzer. A precision impulse sound level meter was used for the overall levels, a real-time 1/3-octave analyzer was used for the 1/3-octave SPL, and an FFT spectrum analyzer was used for the tonal measurements. Recently, several real-time analyzers have emerged with the ability to measure all required parameters by simply selecting different measurement modes on one analyzer. This is possible with the use of real-time digital signal processing hardware and algorithms.

Most modern digital analyzers use a similar process. The incoming signal is digitized using analog-to-digital converters. For real-time 1/3-octave filtering, the sampled signal is sent through digital Butterworth filters, and each 1/3-octave band is integrated using linear or exponential averaging routines. For impulse detection, the digital data is sent through a custom digital impulse detector (having the necessary fast rise time constant of 35 ms and the slow decay time constant of 1.5 s). For overall levels, the energy in the digital samples is integrated over a specified bandwidth. Finally for FFT analysis, the digital data stream is fed into FFT processors—in blocks of samples—with the specified overlap and window applied.

Selection of the analysis mode is usually done via the front panel. But if the instrument has an on-board processor or is controllable over the HP-IB (IEEE-488 bus) or via RS-232, the selection of the measurement mode can be handled automatically. In this case the entire procedure for a sound power evaluation can be programmed to start with the push of a single button.

Test Environment

To ensure that the basic assumptions for measuring sound power using pressure measurements are met (per ISO 3745), tests are conducted in a semi-anechoic chamber. A semi-anechoic chamber (sometimes called a *hemi-anechoic* chamber) is an anechoic chamber with a hard, reflecting floor. The chamber must be free from reflecting objects and surfaces (except for the reflecting plane).

Test Chamber Size

The test chamber must be large enough to allow the measurement surface to be placed outside of near-field effects and in the free field. Because the size of the near-field environment depends on acoustic wavelength, the size of the measurement surface and the volume and shape of the test chamber is usually determined by the low frequency measurement requirement since the lower frequencies have the longest wavelengths.

One conservative rule-of-thumb for determining the measurement distance (the distance from the noise source to the measurement surface) is that the far field begins at a distance equal to twice the maximum dimension of the noise source. For example, a printer with a maximum dimension of 50 cm should have a measurement distance of 1 m.

The walls of the test chamber must not be too close to the measurement surface. This distance should be at least $\lambda/4$, where λ is the wavelength of sound corresponding to the center frequency of the lowest 1/3-octave band required. So, for accurate measurements down to the 100 Hz 1/3-octave band, the minimum distance from the measurement surface to the acoustic treatment in the chamber should be:

Near Field

In the immediate proximity of a noise source—such as a vibrating surface—acoustic energy is composed of both active and reactive components. The active component represents energy flow similar to that which exists in a free field. The reactive component represents the energy in a reverberant room or standing wave tube. The reactive component can actually dominate the active energy flow close to the source, but dissipates quickly because of the lack of reflecting surfaces, such as those at the end of the wave tube or the walls of a reverberant chamber.

So for accurate measurement of active energy flow using sound pressure level, it is important to measure SPL at some distance away from the vibrating surface. This ensures that the reactive component has died out and the active portion can be measured correctly.

The Pressure-Intensity Index is a good indicator of bad measurements in the near field. This is because the reactive pressure contributes to the overall pressure reading, thus making the P-I index noticeably too high.

$$\text{Distance} = \frac{\lambda}{4} = \frac{c}{4f_{CF_{min}}} = \frac{344 \text{ m/s}}{4(100) \text{ 1/s}} = 0.86 \text{ m}.$$

For acceptable results in the 100 Hz frequency band, the measurement surface should be at least 0.86 m from the acoustic treatment.

Absorption of the Test Chamber

Since the test chamber is semi-anechoic, representing a free field over a reflecting plane, the ideal test chamber walls and ceiling would have an absorption coefficient of 100% over the frequency range of interest and the floor would have an absorption coefficient of 0%. Practical requirements specify wall and ceiling absorption to be greater than 99% and floor absorption to be less than 6%, when measured in a plane-wave impedance tube.

Test chamber walls are normally treated with wedges of absorptive material, with a small air space behind them. The total recommended depth of treatment including the air space is $\lambda/4$, where λ is again the wavelength of sound corresponding to the center frequency of the lowest 1/3-octave band. For the 100 Hz 1/3-octave band, this corresponds to a depth of material and airspace of 0.86 m.

Unwanted Reflections

Reflections can occur from any exposed object such as vents, piping, ductwork, and cables. To reduce measurement error, all reflecting surfaces should be placed outside the chamber walls.

Practical Considerations

Other factors play an important role in the design of anechoic test chambers:

- air conditioning and ventilation noise
- noise and vibration from local traffic
- background noise from within the facility

These factors may reduce accuracy or constrain the usable frequency range of inadequately designed chambers.

Measurement Surface

The standard measurement surface for sound power evaluation is box-shaped, called a *rectangular parallelepiped*, terminated by the hard reflecting floor. As mentioned previously, the measurement surface should be located outside of the near field, yet as close to the source as possible. The distance between the noise source and the measurement surface is called the measurement distance. It is typically 1 m. Microphones are located on the measurement surface at regularly spaced positions as shown in figure 2a. For small noise sources it may be more convenient to use a hemispherical surface similar to figure 2b.

Microphone Location and Orientation

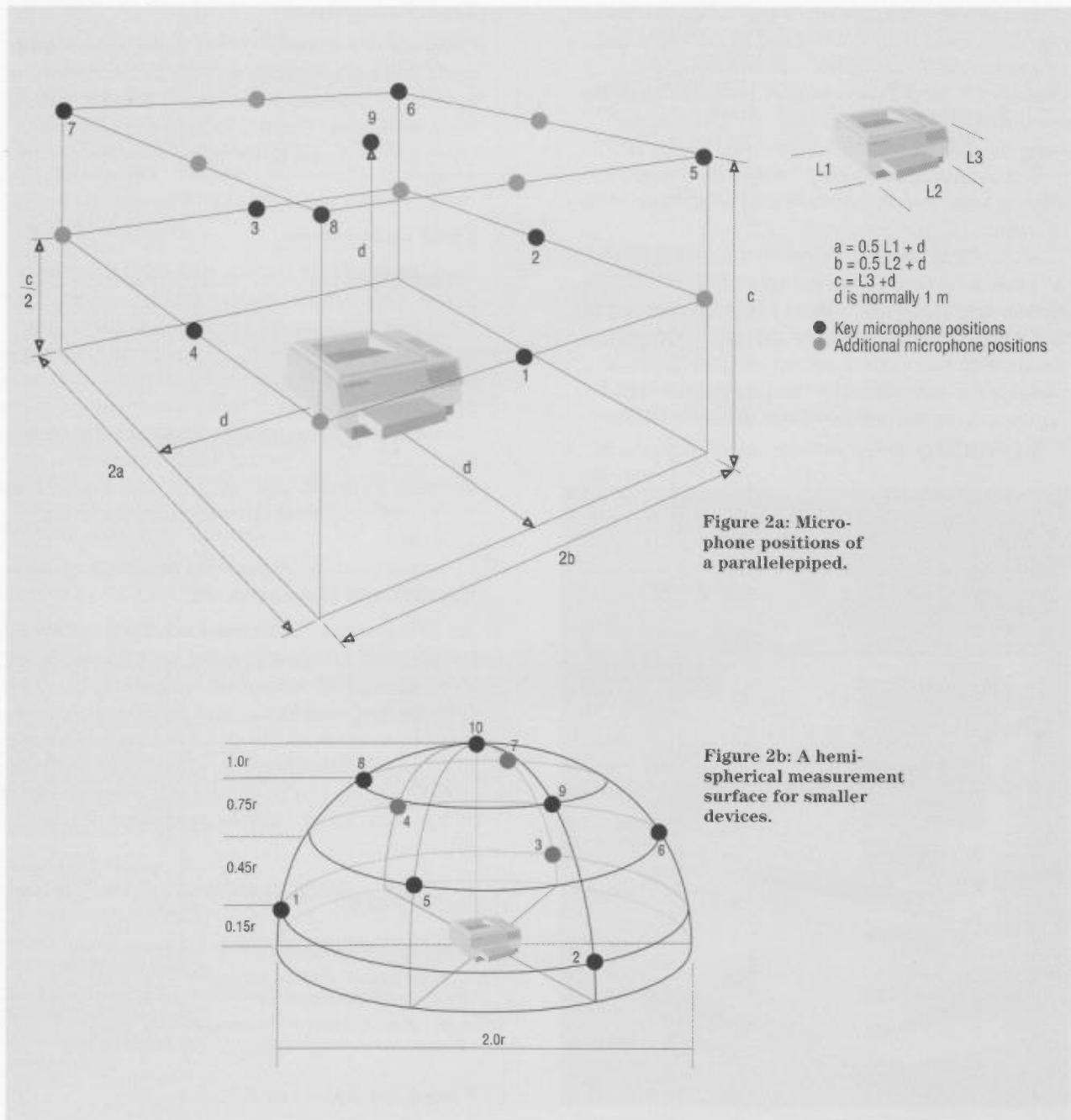
A minimum of nine microphone positions are required for the parallelepiped measurement surface as shown in figure 2a. Additional microphone locations may be required if the noise source is long or if the noise is highly directional.

There are two tests for determining if more microphone locations are needed. First, the difference between the highest and lowest measured SPL must be less than the number of microphone locations. For nine microphones, the range of measured sound pressure levels must be less than 9 dB. Secondly, if the noise source is long such that the parallelepiped has a side that is more than twice as long as the measurement distance, the additional microphone locations (as indicated in figure 2a) must be measured.

Free field microphones are recommended for SPL measurements in a semi-anechoic chamber because it is a free field environment. To obtain the flattest frequency response, the microphones should be pointed at the geometric center of the noise source.

Instrumentation Configuration and Setup

A sound power test for ISO 7779 requires sound pressure measurements at nine locations. In addition, measurements are required at the operator location and four bystander locations.



This means that an array of 10 to 13 microphone positions are required to satisfy ISO 7779.

Several instrumentation configurations can be used to measure 13 microphone locations. If test time is critical, it is possible to measure all 13 positions simultaneously using a multichannel analyzer. This configuration requires 13 microphones and preamps, 7 two-channel microphone

power supplies, and at least thirteen channels of real-time analysis. This system would be the most expensive, but in certain circumstances it could be economical. One benefit of this test setup is that all measurements are taken simultaneously, so far less variation due to operating conditions could be expected. This may be important for devices with transient or intermittent acoustic signatures.

If time is less critical, a more typical configuration could be used. This consists of 13 microphones and preamps, 7 microphone power supplies, a multiplexer, and a two-channel real-time analyzer as shown in figure 3. Through HP-IB programming, this system could make complete sound power evaluations without human intervention, although the microphones would be measured two at a time.

A less expensive system configuration is 2 microphones and preamps, a two-channel power supply, and a two-channel real-time analyzer. Measurements with this system are much more tedious than with the previous two because the microphone pair must be moved and oriented before every measurement.



Figure 3: The HP 35665A is a two-channel real-time analyzer that can be programmed to control a multiplexer for automated measurements.

Test Procedure

Once the test chamber is prepared, the measurement surface specified, and the instrumentation system configured, it is possible to start making measurements. The following test sequence is typical for a sound power measurement per ISO 7779.

Field Calibration

Each microphone system, including the microphone, preamplifier, power supply, and analyzer input channel must be calibrated before every series of measurements. The calibration must be performed at one or more frequencies with an accuracy of ± 0.5 dB. The sensitivity of the microphone system is set using a piston phone or sound level calibrator. A pistonphone is typically accurate to better than ± 0.3 dB, and a sound level calibrator is accurate to within the required ± 0.5 dB.

Background Measurement

All SPL measurements must be corrected for background noise, unless the background noise is more than 10 dB below the measured SPL. Therefore the first step in a sound pressure measurement is to measure SPL at all microphone locations—including operator or bystander locations—with the noise source turned off. This background SPL is subtracted from the measured SPL as follows:

$$L_p = 10 \log(10^{0.1L_C} - 10^{0.1L_B}),$$

where L_p is the corrected sound pressure level, and

L_C is the measured sound pressure level with the noise source turned on, and

L_B is the background sound pressure level with the noise source turned off.

Check for Range of SPL

After the background noise check, the noise source is turned on and the SPL is measured at each microphone location to determine if the range of SPL is acceptable. If the range of sound pressure values (i.e. the difference in dB between the highest and lowest levels) is less than the number of microphone positions on the measurement surface, then the locations are acceptable.

Operational Measurement

The next step is the measurement of SPL used in the actual sound power calculation. With the equipment operating, the sound pressure level is integrated over an interval that includes at least three whole cycles of operation. The minimum integration time is usually 8 seconds, however is can easily extend to 32 seconds for long operational cycle times.

The averaging should be a linear integration of sound pressure, often referred to as Leq. A-weighting should be applied to the overall measurement—however, the 1/3-octave band data should be unweighted.

Measuring at Operator Positions

Pressure level measurements at the operator or bystander position should be made next with the following detection schemes:

- A-weighted linear overall SPL and unweighted 1/3-octave band level SPL
- Overall A-weighted impulse
- Unweighted narrowband FFT

Computation of Corrected SPL and Total Sound Power

SPLs at all locations are corrected for background levels. The overall surface-averaged SPL is then calculated as follows:

$$\overline{L_{pA}} = 10 \log \left(\frac{1}{N} \sum_{i=1}^N 10^{0.1L_{pAi}} \right),$$

where N is the number of microphone locations on the measurement surface,

L_{pAi} is the corrected, A-weighted sound pressure level at each location, and

$\overline{L_{pA}}$ is the A-weighted surface-averaged sound pressure level.

The total A-weighted sound power level is calculated from the following equation:

$$L_{WA} = \overline{L_{pA}} + 10 \log \left(\frac{S}{S_0} \right),$$

where S is the total area of the measurement surface, in m^2 , and

S_0 is the reference area, which is $1 m^2$.

This equation only applies if the standard measurement surface is used (where each measurement location represents the same surface area). If some other microphone spacing is used (where each microphone represents different surface areas), then the equation is as follows:

$$\overline{L_{WA}} = 10 \log \left(\sum_{i=1}^N \left(10^{0.1L_{pAi}} \times S_i \right) \right),$$

where S_i is the area covered by the i^{th} microphone location.

Test Report

A test report for a sound power survey should contain the following information:

- Name and model number of equipment being tested
- A-weighted sound power level, $\overline{L_{WA}}$, in dB, reference: 1 pW for idling and operating modes
- Sound power levels, $\overline{L_W}$, in dB, reference: 1 pW, in 1/3-octave bands for idling and operating modes
- Description of the operating conditions
- Description of the measurement surface, measurement distance, surface area, microphone location and microphone orientation
- Background corrections
- Date, time, and location of test
- Name of the person doing the testing

Evaluating Sound Power Using Sound Intensity Measurements

The need for sound power measurements is well known and accepted in the noise control engineering community. Over the past several decades, however, many vendors and their customers have been slow to adopt sound power testing as a requirement because of the prohibitive cost of test facilities. Building and equipping even a modest test chamber could cost a small company a year's profit. Fortunately, there is a measurement technology that can eliminate the need for testing in chambers—sound intensity. Measuring sound intensity directly not only eliminates the need to build a semi-anechoic chamber, but can also relax other measurement constraints such as the presence of external noise or measurements in the near field.

If using sound intensity has so many advantages, why hasn't it been used more often? Previously, sound intensity test equipment was very expensive and cumbersome. And although the cost of a sound intensity system was less than the cost of an anechoic chamber, it was more expensive than companies were willing to pay for custom, dedicated electronic instrumentation. Modern real-time analyzers that can measure sound intensity are now more portable and far less expensive than in the past, so the price of instrumentation for sound intensity is no longer a constraint.

But the most important development with sound intensity techniques has been acceptance by the engineering community. In the past, many domestic and international standards specified the use of sound pressure level to evaluate sound power, not sound intensity. Now standards are being adopted, such as ANSI S12.12-1992 and ECMA-160, that offer guidance and credibility to manufacturers for the use of sound intensity. This opens the floodgates for the general acceptance of its use in industry.

This section covers sound power measurements using the sound intensity method. A description of sound intensity measurements and constraints is presented, so that the advantages of sound intensity can be better understood. Finally, a description of recommended test procedures associated with this relatively new technique is given.

Eliminating the Need for Chambers

Previously in this application note, the concept of determining sound intensity by measuring sound pressure level in a free field was presented. It was explained that a precise relationship between sound intensity and sound pressure exists in a free field as follows:

$$I = \frac{\overline{p^2}}{\rho c}$$

Since the accuracy of measuring sound power with sound pressure depends on this free field constraint, it is necessary to carefully control the acoustic environment. Because of this, expensive, dedicated semi-anechoic chambers are necessary for pressure measurements.

Sound intensity reduces the need for the free field constraint because intensity is measured directly with a very relaxed assumption about the acoustic environment.

What is Sound Intensity?

Sound intensity is the rate of flow of acoustic energy in a given direction. Since it is a vector quantity, it can be used to assess the net power emanating from a surface. Total sound power can be computed by adding the net power contribution from each portion of the measurement surface, assuming that the measurement surface surrounds the noise source. This is also how net power is computed when using pressure measurements in a semi-anechoic chamber, as in the previous section.

Intensity is actually composed of two components: *active* and *reactive* intensity. Active intensity is the focus of this section of the application note—it represents the net flow of energy and is caused by pressure gradients. Reactive intensity, on the other hand, is the acoustic energy that is stored in an acoustic medium but does not cause energy to flow. Such an environment exists in a reverberant room or a wave tube. Because of the hard walls and high reflection coefficients, the sound pressure can be quite high even though very little acoustic energy is absorbed by the walls or transmitted from the room.

Analytically, active intensity is differentiated from reactive intensity by the phase of the particle velocity relative to pressure. In an active environment, pressure and velocity are in phase, so there is power being transmitted and therefore a net flow of energy. In a reactive environment, pressure and particle velocity are out of phase, so no power is being transmitted. When pressure is highest in a *reactive* environment, velocity is zero. In an *active* environment, when pressure is highest, so is velocity.

Measuring Power in the Presence of Noise

A big advantage of using sound intensity instead of pressure to measure total sound power is that steady external noise sources do not contaminate the intensity measurement. This is because intensity is a vector quantity, as shown in figure 5. When the contribution to an external noise source is positive on one surface, it will be negative on the opposite side. This results in a net power of zero from the surface.

The external noise source must be steady, or errors occur when the net power is summed. This is because the intensity due to noise into one side of the measurement surface might not be equal to the intensity out of the other side—if measurements are done at different times, or the external noise source is intermittent or different between the two measurements.

Most common noise sources are steady. Examples are air conditioning noise or the hum from lights or transformers. Other sources of steady noise can be cooling fans on instrumentation or the whine of a disk drive.

Note that sound power measurements using sound pressure could never be done in the presence of noise because pressure is a scalar quantity—that is magnitude without direction—so sound pressure due to external noise could not be differentiated from the device under test. Therefore, the noise would contaminate the measurement.

How is Sound Intensity Measured?

In contrast to pressure (measured directly with a microphone), sound intensity must be calculated in real-time from measurements of pressure and particle velocity. The transducer that measures pressure and particle velocity simultaneously is called a sound intensity probe as shown in figure 4. The electronic instrument that computes sound intensity in real-time is called a sound intensity analyzer. Some newer 1/3-octave real-time analyzers are capable of computing sound intensity in real-time. In this section, we'll explain how sound intensity probes and analyzers work.



Figure 4: Newer 1/3-octave real-time analyzers with intensity probes can measure both pressure and particle velocity at the same time.

Intensity from Pressure and Velocity

Intensity is equal to the net power per unit area. Since power is equal to force times velocity, in a given direction, the following relationship exists:

$$I = \frac{W}{A} = \frac{F \times v}{A} = \frac{F}{A} \times v = P \times v,$$

where I is intensity, W is power, A is area, F is force, P is pressure, and v is velocity.

Because velocity is a vector quantity and pressure is a scalar quantity, intensity becomes a vector. So the general relationship at a point in space is as follows:

$$\vec{I} = P \times \vec{v}$$

The velocity vector, and therefore the intensity vector, has three components in space: (I_x , I_y , I_z). Typically a probe will measure sound intensity only along its axis, as shown in figure 5, so the measured intensity is the dot product of the actual intensity vector and the axis direction vector:

$$I_{\text{measured}} = (I_x, I_y, I_z) \cdot \vec{A}_{\text{probe}}$$

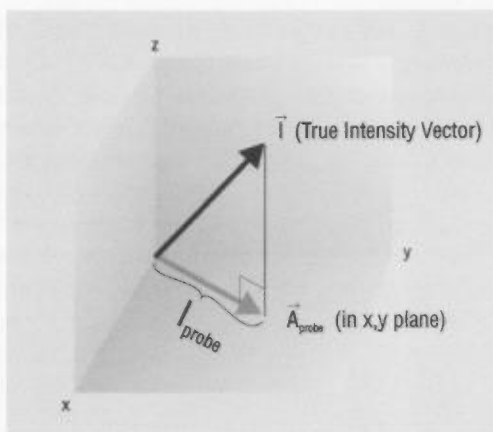


Figure 5: The intensity vector from the measurement surface is the x component of the vector emanating from the source.

When evaluating sound power, the intensity component from a measurement surface is important, since this is the component that determines the total sound power of the device under test. It is therefore necessary when measuring sound power with an intensity probe to keep the probe's axis normal to the measurement surface. This can be difficult if the measurement surface is curved or irregular.

Measuring Velocity in One Direction

The biggest technical challenge in sound intensity measurements is the measurement of velocity. Currently, the best method for measuring velocity is the pressure difference technique. From equations of motion, a relationship between velocity and pressure difference can be found so that velocity can be computed from the pressure

difference between two microphones. If V_p is the component of velocity along the probe axis, then intensity along the probe axis equals velocity times pressure:

$$I_{\text{probe}} = P \times v_{\text{probe}}$$

Measuring intensity can be accomplished with two microphones by computing the velocity term from their pressure difference and the pressure term from their average pressure.

Converting Pressure Difference to Velocity

From meteorology we know that winds blow when there is a pressure difference between two locations. Similarly in acoustics, a pressure difference causes acceleration of air molecules. Newton's Second Law, $F = m \times a$, governs both phenomena. For compressible fluid flow, Newton's Second Law relates pressure, density, and rate of change of velocity (acceleration) as follows:

$$-\frac{\partial P}{\partial x}(x, t) = \rho_0 \frac{\partial v}{\partial t}(x, t),$$

where ρ_0 is the air density,

$\frac{\partial P}{\partial x}(x, t)$ is the pressure gradient,

$\frac{\partial v}{\partial t}(x, t)$ is the time rate of change of velocity (accel.).

Restating this equation in terms of rate of change of velocity,

$$\frac{\partial v}{\partial t}(x, t) = -\frac{1}{\rho_0} \frac{\partial P}{\partial x}(x, t).$$

For a sound intensity probe in figure 7, the pressure gradient, $\frac{\partial P}{\partial x}(x, t)$, is approximated by the following equation:

$$\frac{\partial P}{\partial x}(x, t) \cong \frac{P_2 - P_1}{\Delta r},$$

where Δr is the microphone spacing.

Also since velocity is the time integral of acceleration, then

$$v = \int_0^t \frac{\partial v}{\partial t}(x, t) dt,$$

so for the sound intensity probe,

$$v = -\frac{1}{\rho_0} \int_0^t \frac{P_2 - P_1}{\Delta r} dt.$$

This equation represents the measurement of acoustic velocity at a point in space midway between the microphone pair. The equation for pressure at precisely the same point is much more straightforward:

$$P = \frac{P_1 + P_2}{2}.$$

The equation for sound intensity measured along the probe axis can then be computed as follows:

$$I_{\text{probe}} = P \times v_{\text{probe}} = \left\{ \frac{P_2 + P_1}{2} \right\} \times \frac{-1}{\rho_0} \int_0^t \frac{(P_2 - P_1)}{\Delta r} dt$$

$$I_{\text{probe}} = -\frac{P_2 + P_1}{2\rho_0\Delta r} \int_0^t (P_2 - P_1) dt.$$

This is the algorithm used in a real-time analyzer to compute instantaneous intensity from two simultaneous measurements of pressure. Since intensity is needed in each 1/3-octave band of interest, each pressure signal is passed through a bank of digital 1/3-octave bandpass filters before computation. Figure 6 depicts the processing method to compute sound intensity. The sum and integrated difference of bandpass filtered pressure values are combined and averaged as shown.

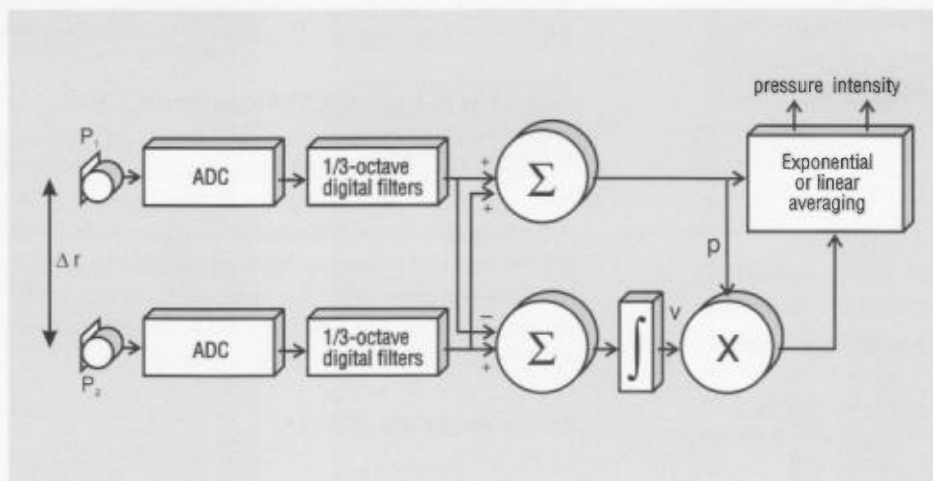


Figure 6: Depicts signals from two pressure microphones which are combined to compute intensity in 1/3-octave bands.

The Two-Microphone Sound Intensity Probe

The transducer most used for sound intensity measurements is a dual microphone pressure-pressure sound intensity probe. It uses two phase-matched microphones mounted on a long handle. The microphone diaphragms usually face each other separated by a solid spacer as illustrated in figure 9. Other configurations can be used successfully (such as two parallel microphones), but the face-to-face arrangement using a solid spacer between them gives the best performance.

The microphones of a sound intensity probe must be able to measure intensity in the frequency range of interest. In addition, the probe's mechanical components should not affect the sound field. To meet the accuracy specifications required in standards such as IEC 1043, a probe will often use high-quality, phase-matched, free-field condenser microphones. For frequency ranges between 50 Hz to 6300 Hz, a 1/2-inch microphone pair is used. For measurements up to 10 kHz, a smaller 1/4-inch microphone pair may be required.

To provide optimum dynamic measurement capability, a sound intensity probe must be capable of using spacers of different sizes. Common spacer sizes range from 6 mm to 50 mm. Larger spacers are required for lower-frequency measurements.

As shown in the next section, the quality of the microphone phase matching and the selection of the spacer between microphones play a key role in defining the range of frequencies and acoustic environments that yield accurate measurements.

Sources of Measurement Error

Making accurate sound intensity measurements requires knowledge of the limitations of the sound intensity probe and analyzer. Because sound intensity requires two parameters to be measured simultaneously—pressure and velocity—sound intensity is more difficult to measure than sound pressure alone. High-quality probes and analyzers that are closely phase-matched can help, but even so the limitations of the system should be known to ensure accurate measurements.

High-Frequency Errors

There are many possible sources of measurement error at high frequencies:

- Physical effects of microphones, protection grid, spacer and probe body
- Amplitude and phase match of the system
- Effective spacer distance variation
- Straight line approximation error

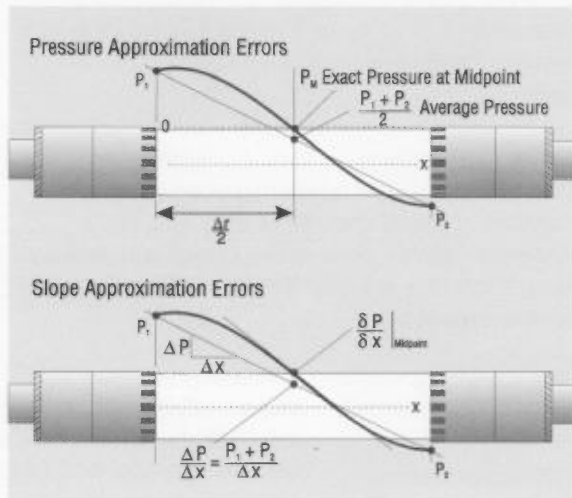


Figure 7: Finite difference approximation errors affect both mean pressure and velocity calculations.

At high frequencies, these sources of error are dominated by the straight line approximation error—assuming that a high quality probe and analyzer are used. The straight line approximation actually introduces two errors at high frequencies; incorrect pressure and incorrect velocity. The combined effect is analyzed in appendix B by analyzing a mathematically-known acoustic wave.

Intensity at a probe's midpoint, derived from pressure measurements at two microphones, is compared with the exact intensity at the midpoint derived from the exact pressure equation.

As derived in appendix B, the approximation error depends upon the size of the spacer between the microphones and the wavelength:

$$e_{\text{probe}} = 10 \log \left(\frac{\sin \left(2\pi \frac{\Delta r}{\lambda} \right)}{2\pi \frac{\Delta r}{\lambda}} \right)$$

This error formula is used to determine the permissible frequency ranges for a specified microphone spacing. The following chart shows the approximation error plotted against frequency for microphone spacing of 6mm, 12mm, and 50mm.

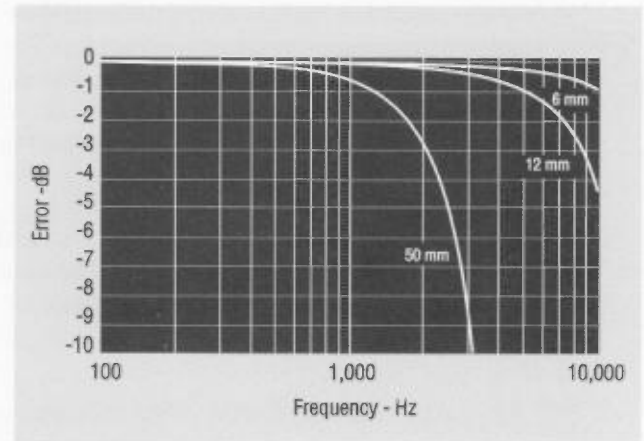


Figure 8: Spacer size can greatly impact approximation errors as frequency changes.

For an approximation error of 1 dB or less, measurements are acceptable at or below the frequencies listed in table 3 on page 22. As a rule of thumb, the error is 1 dB or less when the ratio of spacing to wavelength is less than about 0.18.

Low-Frequency Errors

In the last section, we showed that intensity can be derived from two pressure readings (P_1 and P_2) measured with a sound intensity probe. This measurement was prone to high-frequency errors due to finite difference approximations.

At low frequencies, another source of error occurs that has direct impact on the ability to measure velocity accurately—phase mismatch. Phase mismatch exists to some extent in all sound intensity analyzers and probes. However, for reasons shown below, the low-frequency phase match must be carefully specified for successful measurements.

At low frequencies, where the wavelength is long relative to the length of the spacer, the phase difference between P_1 and P_2 will be small. In fact, at very low frequencies, the difference is so small it can be masked by the phase error of the probe and analyzer. This problem can be compounded by the presence of reactive intensity, where the sound pressure level can be 20 dB or more above the actual sound intensity level. How are intensity, pressure, frequency and spacer size related? And how does phase accuracy affect intensity measurement accuracy?

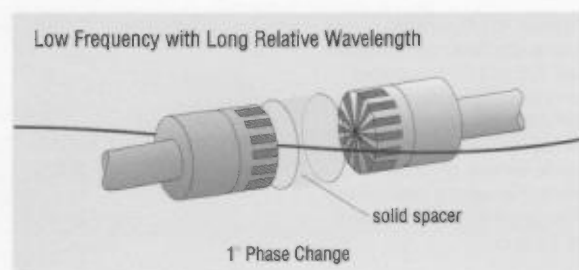


Figure 9: Phase change across the spacer is small when the spacer size is short relative to the wavelength; mainly at low frequencies.

To understand these relationships and potential pitfalls, the parameters associated with sound intensity measurements must first be analyzed. From appendix C, the relationship between intensity, pressure, phase difference, frequency, and microphone spacing is given as:

$$I = \frac{P_{1rms} P_{2rms} \sin(\phi)}{2\pi f \rho_0 \Delta r}$$

where I is intensity,

P_{1rms} and P_{2rms} are the pressures at microphones 1 and 2,

ϕ is the measured phase difference between channel 1 and channel 2 at frequency f .

ρ_0 is the density, and

Δr is the spacer size.

At low frequencies, the pressures at each microphone are almost identical, so the rms pressure can be used instead of the individual pressures:

$$I = \frac{P_{rms}^2 \sin(\phi)}{2\pi f \rho_0 \Delta r}$$

The ratio of sound pressure level to intensity is called the pressure-intensity index and is a good indicator of the acoustic environment. Measurements with high P-I indices are more difficult because the phase change across the probe spacer becomes smaller when the pressure-intensity index increases. This can be shown by rearranging the terms:

$$\sin(\phi) = \frac{2\pi f \rho_0 \Delta r}{P_{rms}^2 / I}$$

Since sound pressure level and sound intensity level are defined as follows:

$$L_p = SPL = 10 \log \frac{P_{rms}^2}{P_0^2},$$

$$\text{where } P_0 = 20 \mu Pa \text{ and } L_I = 10 \log \frac{I}{I_0},$$

$$\text{where } I_0 = 10^{-12} \text{ Watt.}$$

Then the pressure-intensity index can be defined as:

$$L_{PI} = L_p - L_I = 10 \log \left(\frac{P_{rms}^2}{P_0^2} \frac{I}{I_0} \right) = 10 \log \left(\frac{P_{rms}^2}{I} \frac{I_0}{P_0^2} \right)$$

$$\frac{P_{rms}^2}{I} = \frac{P_0^2}{I_0} 10^{L_{PI}/10}.$$

The equation for phase angle can then be stated in terms of:

$$\sin(\phi) = \frac{2\pi \rho_0 I_0}{P_0^2} \frac{f \Delta r}{10^{L_{PI}/10}}.$$

This shows that the measured phase angle is proportional to frequency and spacer size but inversely proportional to the P-I Index. In other words, in situations of high pressure relative to intensity (as in a reactive sound field, such as a reverberant room or duct), the measured phase difference is small. This is depicted in figure 10 for the curve with a P-I Index above 10 dB. Note that the phase change is small even at higher frequencies.

Problems Measuring Small Phase Changes

Figure 10 shows that when the P-I Index is high, especially over 10 dB, the phase change over the probe spacer is small. This would not be a problem if the sound intensity probe and analyzer had phase accuracy that was much smaller than any measured phase change. Unfortunately, this is not the case. The best intensity analyzers have cross-channel phase accuracies of $\pm 0.02^\circ$ and the best sound intensity probes are matched to within 0.05° . In a moment, we'll show how these phase accuracies are the limiting factor in measuring intensity at low frequencies (where the spacer is small relative to the wavelength) or in reactive environments (where the P-I Index is high).

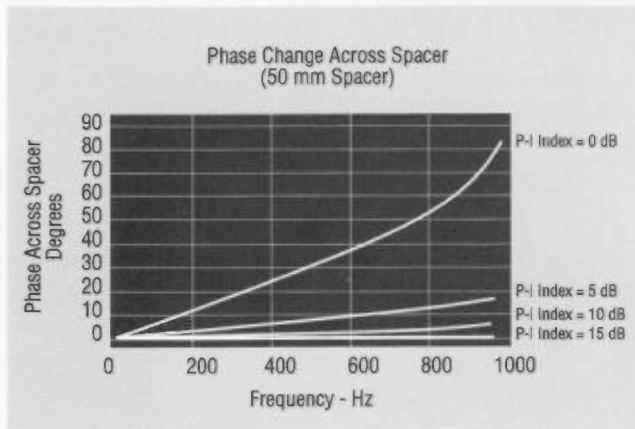


Figure 10: Phase change across a 50 mm spacer.

Residual Intensity

When a sound intensity probe is placed in a completely reactive environment where both microphones are subjected to the same pressure with the same phase, the intensity analyzer should indicate zero intensity. But a zero intensity reading is not possible, due to some residual phase mismatch between the microphones in the probe. The sound intensity analyzer indicates some residual intensity that reflects the phase mismatch of the probe and analyzer, not the sound field. This

remnant is called residual intensity. The level of residual intensity increases when the reactive pressure increases, so it is convenient to refer to this term as the residual pressure-intensity index, or L_{RPI} . This term defines the frequency range over which accurate measurements can be made. A typical residual pressure-intensity index for a probe and analyzer is shown in figure 11.

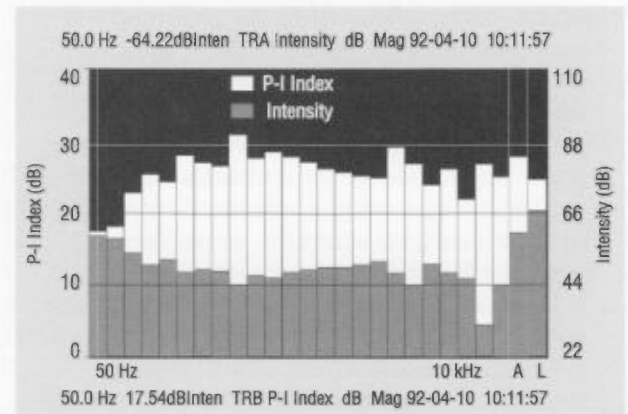


Figure 11: Residual intensity and corresponding P-I index when the two microphones of an HP 35230A intensity probe are exposed to the same sound pressure within a cavity calibrator. The microphone spacing is 12 mm.

Phase Accuracy Specifications of Intensity Analyzers

IEC 1043 is an international standard that specifies accuracy requirements for sound intensity probes and analyzers. Instead of specifying phase accuracy directly it defines a minimum residual pressure-intensity index versus frequency, as shown in figure 12.

Since most sound intensity analyzers specify phase accuracy, not a residual pressure-intensity index, it is helpful to convert one specification to the other. Assume that the phase angle of an intensity measurement has an error equal to $\pm\theta_e$, where $\pm\theta_e$ is the phase accuracy of the measurement system. It can be shown that θ_e is directly related to L_{RPI} . Since residual pressure is measured when the phase angle between the two microphones is actually zero, then the indicated intensity is all residual:

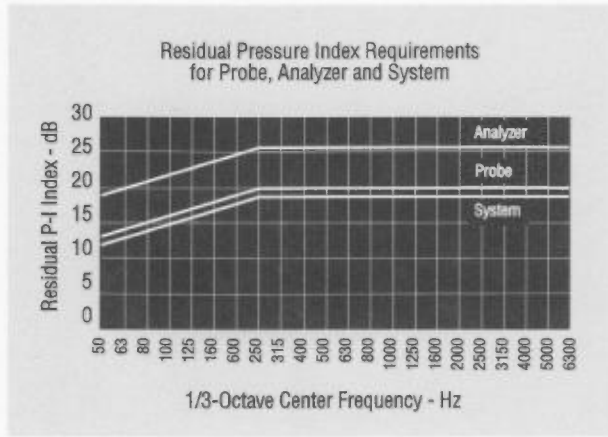


Figure 12: IEC 1043 Class 1 requirements of L_{RPI} for the probe, analyzer, and combined system, for a 25 mm microphone spacer.

$$I = \frac{P_{rms}^2 \sin(\phi \pm \theta_e)}{2\pi f \rho_0 \Delta r} \text{ and when } \phi = 0,$$

$$I_{residual} = \frac{P_{rms}^2 \sin(\theta_e)}{2\pi f \rho_0 \Delta r},$$

and the residual pressure intensity index based on θ_e is:

$$L_{RPI} = 10 \log \left(\frac{P_{rms}^2 / P_0^2}{I_{residual} / I_0} \right) = 10 \log \left(\frac{2\pi \rho_0 I_0}{P_0^2} \frac{f \Delta r}{\sin(\theta_e)} \right).$$

Typical values of θ_e for a system at 100 Hz might be $\pm 0.15^\circ$, so with a 12 mm spacer, the residual pressure-intensity index L_{RPI} would be:

$$L_{RPI} = 10 \log \left(\frac{2\pi \rho_0 I_0}{P_0^2} \frac{f \Delta r}{\sin(\theta_e)} \right) = 10 \log \left(\frac{2\pi (1.29) (10^{-12}) 100 (12 / 1000)}{(20 \cdot 10^{-6})^2 \sin(0.15^\circ)} \right) = 9.7 \text{ dB}.$$

Intensity Measurement Error from Phase Mismatch

Knowing the residual P-I index of a sound intensity measurement system means that the dynamic capability of a measurement system is known. Intensity smaller than the residual intensity cannot be differentiated from analyzer phase mismatch and therefore cannot be measured. The residual intensity is dependent on the pressure, so in reactive environments where pressure is high, the residual intensity is also high. This sets a lower limit on accurate measurements of intensity and thus constrains the acceptable range of P-I indices for accurate measurements.

To determine the acceptable P-I range based on the knowledge of L_{RPI} , the intensity error due to phase mismatch will be calculated. Since the intensity error is the difference between actual intensity and intensity measured with an imperfect probe and analyzer, the error is a function of L_{PI} and L_{RPI} .

For small phase angles, where $\sin \phi = \phi$, the relationship between the P-I Index and the phase angle is:

$$\theta_e = \frac{2\pi \rho_0 I_0}{P_0^2} \frac{f \Delta r}{10^{L_{RPI}/10}}$$

$$\phi = \frac{2\pi \rho_0 I_0}{P_0^2} \frac{f \Delta r}{10^{L_{PI}/10}}$$

$$e_{\theta} = 10 \log \left(\frac{I_{\text{measured}}}{I_{\text{actual}}} \right) = 10 \log \left(\frac{\frac{P_{\text{rms}}^2 \sin(\phi \pm \theta_e)}{2\pi f \rho_0 \Delta r}}{\frac{P_{\text{rms}}^2 \sin(\phi)}{2\pi f \rho_0 \Delta r}} \right) = 10 \log \left(\frac{\phi \pm \theta_e}{\phi} \right)$$

$$e_{\theta} = 10 \log \left(1 \pm \frac{\theta_e}{\phi} \right) = 10 \log \left(1 \pm \frac{\frac{2\pi \rho_0 I_0}{P_0^2} \frac{f \Delta r}{10^{L_{\text{RPI}}/10}}}{\frac{2\pi \rho_0 I_0}{P_0^2} \frac{f \Delta r}{10^{L_{\text{PI}}/10}}} \right) = 10 \log \left(1 \pm \frac{10^{L_{\text{PI}}/10}}{10^{L_{\text{RPI}}/10}} \right)$$

$$e_{\theta} = 10 \log \left(1 \pm 10^{[L_{\text{PI}} - L_{\text{RPI}}]/10} \right)$$

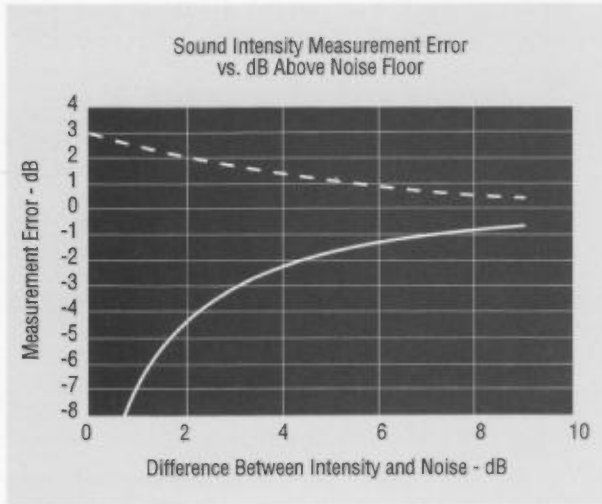


Figure 13: The chart shows the expected sound intensity measurement error as a function of the difference between L_{PI} and L_{RPI} .

From the diagram in figure 13, we see that as long as intensity is 7 dB greater than the residual noise, less than a 1 dB measurement error can be expected. We can state this in terms of relative magnitude of phase shifts:

since $e_{\theta} = 10 \log \left(1 \pm \frac{\theta_e}{\phi} \right)$, then

$$\frac{\theta_e}{\phi} = 1 - 10^{e_{\theta}/10} = 1 - 10^{-1/10} = \frac{1}{4.86}.$$

So accurate intensity measurements require the phase change across the spacer to be at least 4.86 times the cross-channel phase accuracy of the system.

Off-Axis Measurement Error

The last source of intensity measurement error to be analyzed is that due to the probe pointing in the wrong direction. A sound intensity probe designed in accordance with IEC 1043 has an off-axis response characteristic that is similar to figure 13. This follows the cosine relationship:

$$\frac{I_{\theta}}{I_0} = 10 \log [\cos \theta]$$

When measuring sound power, the probe must be held relatively perpendicular to the measurement surface to prevent errors. If the probe axis is tilted 10° , the measured intensity will be wrong by only 0.07 dB. This is not a significant concern. However, a bigger error can occur from intensity that is perpendicular to the measurement surface. This component of intensity should not contribute to the net intensity or sound power calculation. With a 10° slant, however, the coefficient goes from 0 at $\theta=90^\circ$ to 0.17 at $\theta=80^\circ$. This could cause an unwanted contribution from an off-axis source of noise. Although the null at 90° can cause errors in a sound power measurement, this characteristic is valuable when using intensity to precisely locate a sound source.

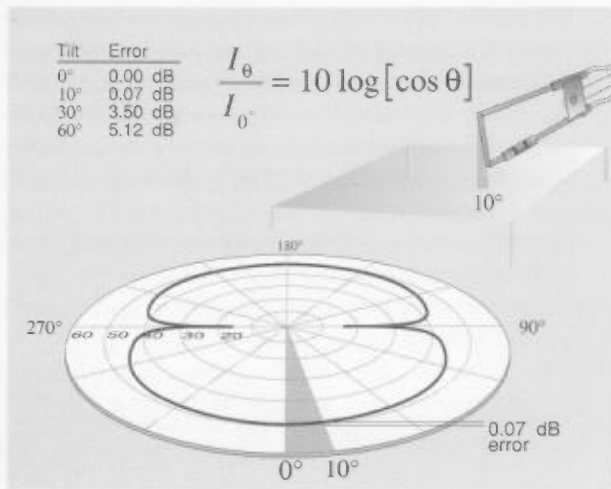


Figure 14: If the probe is not held perpendicular to the scanning surface an error (i.e. 0.07 dB at 10° tilt) will be introduced.

Sound Power Test Procedures using Sound Intensity

Successful sound power measurements can be made using sound intensity methods as long as the test engineer matches the sound intensity probe and analyzer capability with the test environment. If measurements are to be made in the presence of noise, the noise must be steady and the phase accuracy of the measurement system must be sufficient.

The following test procedures, combined with good engineering judgement based on the analysis of potential errors in the previous section, will yield accurate sound power measurements.

Select the Appropriate Frequency Range

To determine the sound power of a device, many international standards require measurements in 1/3-octave bands from 100 Hz to 6300 Hz. Some contracting agencies may extend or restrict this frequency range to tailor requirements to a specific group of equipment. For example, the sound power level of very large machinery such as hydroelectric generators may contain significant energy below 100 Hz, so the required frequency range for sound power testing may be from 50 Hz to 4000 Hz. Likewise, a small turbine may contain

significant noise power above 6300 Hz, so the specified range may be 100 Hz to 10 kHz. Knowledge of the required frequency range is critical for choosing the right test environment and the appropriate probe and analyzer.

If the frequency range is extended on the lower side, the dynamic capability of the probe and analyzer will be compromised. This was shown earlier in the section covering low frequency errors. At lower frequencies, the phase change across the microphone spacer is smaller, so the phase accuracy of the system can affect the ability to measure. This situation becomes worse when the acoustic environment is reactive. So if frequencies are to be measured below

100 Hz, care should be taken to use a very high-quality sound intensity probe and analyzer. Additionally, large spacers should be used and a test room should be as near to free field as possible (where L_{pt} is less than 5 dB).

Intensity measurements beyond 6300 Hz may require the use of a special sound intensity probe, with a flat, phase-matched frequency response to 10 kHz. The sound intensity analyzer must also have the capability to measure intensity in real-time to 10 kHz.

Configuring the Sound Intensity Probe

Once the frequency range of the test is selected, the required spacers for the probe can be selected. Spacer size depends on the frequency range, acoustic environment (L_{pt}), and the total phase mismatch of the probe and analyzer. Often two or more spacers are required for each test. A large spacer is used at low frequencies and a small spacer is used at high frequencies. Generally if more than one spacer is used, the frequency range for each spacer overlaps by three or more 1/3-octave bands. The measurement results from each spacer are then combined into an aggregate spectrum using the band level from the measurement with the lowest error.

The highest required test frequency determines how small a spacer must be for a given approximation error. Table 3 shows the results of the previous analysis of approximation error. For an approximation error of less than 1 dB, each of the listed spacer sizes has a maximum test frequency. For example, if testing is required up to the 6300 Hz one-third octave band, a 10 mm or smaller spacer must be used for an error of less than 1 dB.

Table 3: Maximum 1/3-Octave Measurement Band vs. Spacer Size

Spacer Size	Maximum Band for 1 dB Error
6 mm	10000 Hz
10 mm	6300 Hz
12 mm	5000 Hz
50 mm	1250 Hz

The lowest required test frequency determines how large a spacer must be. Previously, we saw that the low frequency capability depends on several factors:

- Spacer size
- Pressure-intensity index (L_{PI}) of the acoustic environment
- Phase accuracy of the probe and analyzer
- Desired accuracy

We also saw that low-frequency measurement error is a function of the actual phase shift of pressure across the probe spacer relative to the phase accuracy of the system. This can be shown as follows:

$$e_\theta = 10 \log \left(1 \pm \frac{\theta_e}{\phi} \right),$$

so if e_θ is to be less than ± 1 dB, then the ratio $\frac{\theta_e}{\phi}$ can be determined:

$$\frac{\theta_e}{\phi} = 1 - 10^{e_\theta/10} = 1 - 10^{-1/10} = 0.206 = \frac{1}{4.86}.$$

The result is that the phase change across a spacer must be at least 4.86 times the phase accuracy of the system for measurement errors of less than 1 dB. From this ratio it is possible to determine the minimum spacer size for a given accuracy (1dB), acoustic environment, and frequency.

From above, $\phi = 4.86\theta_e$. In the section on low-frequency errors, it was shown that

$$\phi = \frac{2\pi\rho_0 I_0}{P_0^2} \frac{f\Delta r}{10^{L_{PI}/10}}$$

for small phase angles. Also, since accuracy is required down to the lower corner frequency of a particular 1/3-octave band, which is a factor of $2^{-1/6}$ from the center frequency, the minimum spacer size can be calculated as follows:

$$\Delta r = \frac{4.86\theta_e (2\pi/360^\circ) P_0^2 10^{L_{PI}/10}}{2\pi\rho_0 I_0 \left(2^{-1/6} f \right)} = 4.7 \left(\frac{\theta_e 10^{L_{PI}/10}}{f} \right),$$

where L_{PI} is the pressure-intensity index (or acoustic environment),

f is the center frequency of the 1/3-octave band of interest, and

θ_e is the system phase error, including probe and analyzer.

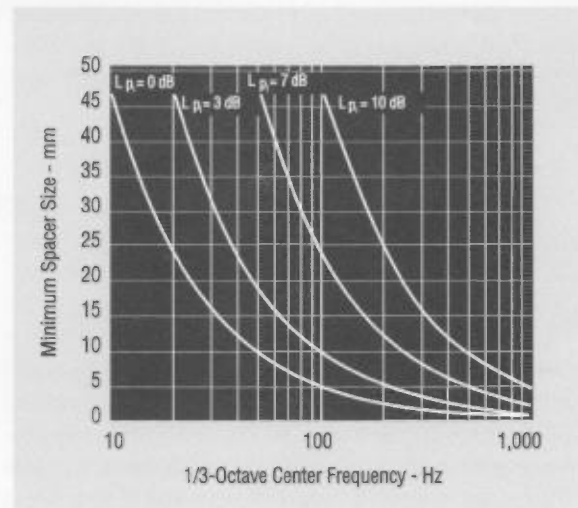


Figure 15: Minimum spacer size for various acoustic environments assuming a system phase accuracy of 0.1°.

Measuring sound power from 100 Hz to 6300 Hz may require a single spacer or several spacers, depending on the phase accuracy of the measuring system and the acoustic environment. At 100 Hz, for a Class 1 system as defined by IEC 1043, the combined low-frequency phase accuracy for the probe and analyzer is 0.08° . A 50 mm spacer would allow testing even in reactive acoustic environments where the P-I index is as much as 10 dB. Unfortunately, this 50 mm spacer is useful only up to 1250 Hz, where a smaller spacer would be required to reduce the approximation error. At 6300 Hz, a 10 mm or smaller spacer would be required. A 10 mm spacer can also be used to measure down to 100 Hz if the acoustic environment is mostly active, where L_{pi} is less than 3 dB. A quiet conference room may meet this criteria.

In conclusion, it may be possible to complete an entire sound power survey using a sound intensity probe configured with a single spacer. However, if the acoustic environment is relatively reactive at low frequencies, two spacers are required.

Test Environment

Even though anechoic chambers are not required for sound intensity measurements, care should be taken to avoid test environments that violate the assumptions of sound power calculated from intensity. For example, noise external to the device under test is acceptable unless it is intermittent. Also, if intensity is being measured in a reactive environment such as a shop floor with hard walls and ceiling, the dynamic capability of the instrumentation must be compared to the measured pressure-intensity index using the following procedure:

1. Measure pressure-intensity index, L_{pi} , averaged over the measurement surface with the device turned on.
2. Determine L_{RPI} , the residual pressure-intensity index for the probe and analyzer measurement system, using the manufacturer's data or by measuring the pressure-intensity index in a perfectly reactive environment, such as a cavity calibrator.
3. Check that the difference between L_{pi} and L_{RPI} is at least 7 dB for all frequencies of interest for this test environment.

Outdoor measurements of intensity have been difficult in the past because of the size and weight of the test instrumentation. With the advent of portable sound intensity analyzers, it is now possible to test devices outdoors. This can eliminate problems with reactive environments because the acoustic environment outdoors can be nearly free-field. Background noise can be a problem, however. Aircraft flyovers or intermittent local traffic can cause obvious errors. Outdoor measurements can also be affected by wind noise. Generally, wind screens should be used. Measurements are acceptable if wind-induced noise is more than 10 dB below the device noise itself.

Measurement Surface

The selection of the measurement surface for sound power evaluation using sound intensity measurements is very important. The measurement surface should completely enclose the device. A reflective floor is acceptable. However, absorptive surfaces such as carpets or turf should be avoided. Intensity measurements should not be so far away from the device that the level is too low to measure accurately.

Intensity measurements can be made in the near field, very close to the device, if the sound intensity probe manufacturer states that near-field measurements meet accuracy specifications. Often, phase-matched omnidirectional microphones are used, and these are vented out the side or back of the microphone cartridge. These vents can lead to low-frequency errors in near-field measurements even if the microphones are well matched. Other phase-matched microphones with special venting chambers have reduced this problem and thus can be used in the near field.

Even though the choice of the closed measurement surface is arbitrary, the use of regular surfaces is recommended because of the ease of measurement. Rectangular boxes (parallel-pipeds), hemispheres, and conformal shapes are common. Each surface should be broken down into segments, each of which having roughly the same area and same distance from the device.

Field Calibration

Each microphone of the sound intensity probe should be calibrated at a single frequency and amplitude before and after every series of measurements. Typically, the sensitivity of each channel of the intensity analyzer is individually set using a pistonphone or sound level calibrator. A record is made of these calibration measurements for the test report.

Field Check

An excellent method of confirming the performance of the sound intensity measurement system is to measure the intensity of the device in a typical measurement location, then measure again after turning the probe around 180° (but keeping the probe's midpoint on the measurement surface). The intensity level of the peak 1/3-octave band between 200 and 5000 Hz should not change by more than 1 dB, and the intensity should be the opposite sign.

Measuring Intensity on the Surface

Generally two techniques are used to measure intensity at each surface segment—fixed point and surface scanning. Fixed-point measurements require a test to ensure that the number of points measured is sufficient. This is because a device that emits noise in a highly-directional fashion may require a finer mesh than a small, omnidirectional source. Some devices or test setups may lend themselves to the fixed-point technique better than others.

For surface scanning, the probe is slowly scanned over the measurement surface while the intensity analyzer is doing a linear average. This works best if the analyzer uses gated averaging, so that it averages only when a button on the probe is pressed. In this way, a measurement surface can be completely scanned while the operator concentrates on holding the probe perpendicular to the surface and moving the probe at a slow, steady pace. When the scan is complete, the operator releases the button and the average is complete. In this case the operator would not have to pace the scan rate with the average time selected on the intensity analyzer.

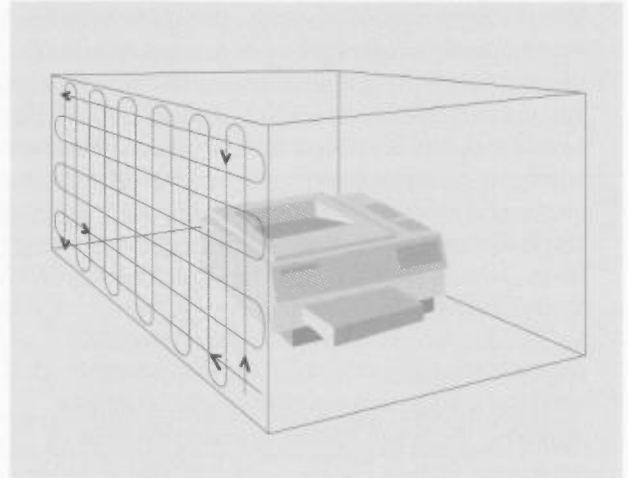


Figure 16: Scanning technique with gated averaging simplifies intensity measurement, shortening test time.

Acceptable scanning speeds are between 0.1 and 3 m/s, but a common speed used is roughly 0.5 m/s. In this way a 1 meter square surface can be scanned (as shown in figure 16) in about 32 seconds. Total test time for a small device in a room over a tiled floor would be 160 seconds.

Computation of Surface Sound Intensity Level

The sound intensity averaged over the entire measurement surface can be calculated as follows:

$$\bar{I} = \frac{\sum_{i=1}^N I_i S_i}{\sum_{i=1}^N S_i},$$

where N is the number of measurement segments,
 S_i is the surface area, and
 I_i is the component of intensity perpendicular to S_i .

The surface sound intensity level can be computed from:

$$\bar{L}_I = 10 \log \left(\frac{|\bar{I}|}{I_0} \right),$$

where $\bar{L}_I$ is the surface-averaged intensity level,
 $|\bar{I}|$ is the absolute value of the surface sound intensity, and
 I_0 is the reference intensity, which is 1 pW/m².

Calculation of Surface Sound Power Level

The total sound power level can be calculated from $\bar{L}_I$ and the total surface area, S , as follows:

$$L_W = \bar{L}_I + 10 \log \left(\frac{S}{S_0} \right).$$

Note that all measurement levels can be A-weighted and should reflect this in the nomenclature, such as L_{WA} instead of L_W .

Test Report

A test report for a noise power survey should contain the following information:

- Name, model number, and dimensions of the device being tested
- Description of the mounting and operating conditions
- Description of the measurement environment, including physical layout of the test area and acoustic properties of surrounding surfaces, surface sound pressure levels, temperature, pressure, and wind speed and direction (if applicable)
- List of model numbers, serial numbers, manufacturers of all instrumentation—including bandwidth of analyzer, frequency response, probe geometry, and last system performance verification
- Description of test procedures, including field calibration, field checks, sampling technique, scan speed and scan pattern, measurement surface size, shape, and location
- A-weighted sound power level, L_{WA} , in dB, reference: 1 pW for idling and operating modes
- Sound power levels, L_W , in dB, reference: 1 pW, in 1/3-octave bands for idling and operating modes
- Date, time, and location of test
- Name of the person doing the testing

Choosing Between Pressure and Intensity Techniques

Quantifying sound power has become much easier because of the advances in sound intensity. Now instead of bringing a test device into a central laboratory with an expensive, high quality anechoic chamber, companies can measure sound power remotely—at the source—in many diverse environments. The advent of the portable, relatively low-cost sound intensity analyzer has been a significant factor in this trend. Acceptance in the engineering community of sound intensity measurement techniques has also been a key factor for growth in the use of this technique.

This is not to say that sound pressure level measurements are a thing of the past. Pressure measurements are still the most ubiquitous method of measuring sound power. In a properly designed anechoic facility, results can be accurate and reliable. Its major disadvantage is facility cost.

Choosing between pressure and intensity test techniques is becoming a fairly straight forward proposition:

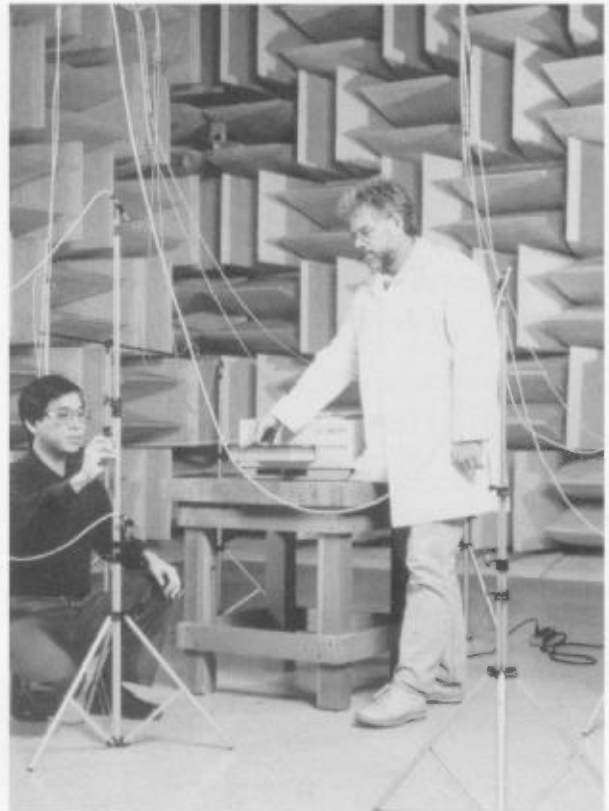
- Can the device be moved into a test facility? If not, use intensity and measure sound power in-situ.
- Does your company need to test the device in more than one location? If so, use intensity and test the device in each location without having to build several acoustic test facilities.
- Can your company afford an expensive, dedicated test chamber? If not, use intensity and schedule test time in any conference room.
- If your company can afford or already has an anechoic chamber, it would be beneficial to understand the acoustic environment using intensity techniques anyway. So even those who are using pressure could use intensity to help qualify their chamber.

Comparing Sound Power Test Techniques

Two methods of determining sound power levels are described in this application note: sound pressure measurements in a semi-anechoic chamber and sound intensity measurements in a quiet conference room. It was shown that sound pressure measurements constrained the acoustic environment severely, but sound power levels were easy to measure with a microphone and a real-time, 1/3-octave analyzer. On the other hand, it was found that ordinary conference rooms could be used for sound intensity measurements, but severe constraints on the sound intensity probe and analyzer (phase matching) were required. Because the prices of high-quality sound intensity probes and analyzers have become quite reasonable, while the cost of building and maintaining anechoic facilities is high, the sound intensity method is becoming the preferred technique.

Theoretically, the two methods should result in the same sound power level if proper guidelines are followed. The following two example tests measure the sound power level of two different Hewlett-Packard LaserJet printers using sound pressure level measurements in a semi-anechoic chamber and sound intensity measurements in a quiet conference room. Results from each example test will be compared.

Test Case 1



Purpose

Determine the sound power level of an HP LaserJet printer using the procedures in ISO 7779, Section 6, which calls for sound pressure level measurements in a semi-anechoic chamber.

Procedure

Frequency Range of Interest

According to ISO 7779, the frequency range of interest normally extends from 100 Hz to 10 kHz, however, the standard suggests that for computers and business equipment a frequency range of 200 Hz to 5 kHz is more typical. For this example, the sound power will be measured from 125 Hz to 5 kHz in 1/3 octave bands.

Test Environment

- The chamber meets the test room qualification requirements of ISO 3745 Annex A for semi-anechoic rooms as specified in ISO 7779.
- The environmental correction factor, K, as determined in ISO 3745 was negligible in the frequency range of interest.

- Background noise checks were conducted to ensure that background levels were at least 6 dB and preferably 10 dB below the sound pressure level to be measured in each frequency band of interest. Actual background levels were more than 15 dB below the measured level, so no background corrections were required as per ISO 7779.
- Test conditions were within prescribed limits: temperature 15°C to 30°C, barometric pressure 86 kPa to 106 kPa, and relative humidity 40% to 70%.

Instrumentation

- The HP 35665A 1/3-octave analyzer was used for measuring sound pressure level. It exceeds Type 1 requirements of IEC 804 for frequency response, accuracy, linear integration and A-weighting.
- Bruel & Kjaer free field Type 4165 microphones which meet the Type 1 accuracy and stability requirements of IEC 651 were used for these measurements.
- The entire measurement system was checked with an acoustic calibrator that is accurate to within ± 0.5 dB.

Equipment Setup and Operation

The printer was mounted on an acoustically rigid table located in the middle of the chamber floor. During the acoustic test, printing operation was continuous.

Measurement Surface and Microphone Positions

The measurement surface was parallelepiped with a total surface area of 19 m² and at a measurement distance of 1 m. The nine microphones were oriented toward the center of the printer.

Averaging Time

According to ISO 7779, the averaging time must be at least 8 seconds. If the equipment under test runs in cycles, at least 3 operational cycles must be averaged. For the printer in this test, an averaging time of 32 seconds was chosen. The printer was turned off and a background measurement was taken at each of the 9 microphone locations. The printer was then turned on and operated, and the measurements were repeated.

Results and Calculations

Sound pressure level spectra were acquired at the nine microphone positions. The surface-averaged band sound pressure level is calculated from the following equation:

$$\overline{L_P} = 10 \log \left(\frac{1}{n} \sum_{i=1}^N 10^{0.1 L_{Pi}} \right), \text{ where}$$

L_{Pi} is the 1/3-octave band sound pressure level at the i^{th} microphone. To determine the 1/3-octave band sound power level, the following equation is used from ISO 7779:

$$L_W = \overline{L_P} + 10 \log \left(\frac{S}{S_0} \right) = \overline{L_P} + 10 \log \left(\frac{19}{1} \right) = \overline{L_P} + 12.8$$

S is the measurement surface area, equal to 19 m², and S_0 is the reference area of 1 m².

Figure 17 shows the sound power spectrum for the printer. The overall A-weighted sound power level calculated from 125 Hz to 5000 Hz is 56.3 dB reference: 1 pW.

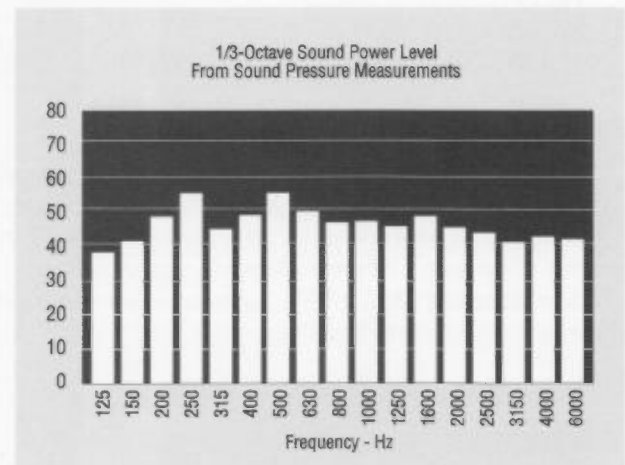


Figure 17: Sound power spectrum for an HP LaserJet printer in a semi-anechoic chamber.

Test Case 2



Purpose

The purpose of this test is to determine the sound power level of a HP LaserJet printer using sound intensity measurement techniques as described in the European Computer Manufacturers Association Standard ECMA-160. A quiet conference room was used as a test chamber.

Procedure

Frequency Range of Interest

ECMA-160 *requires* sound power measurement from 100 Hz to 8000 Hz unless it can be shown that the total A-weighted contribution from unmeasured bands is more than 10 dB below the overall sound power level. Measuring the 100-Hz band would have required another measurement pass using a 50-mm spacer because a 12-mm spacer would not have enough phase change across it. Also measuring out to 8000 Hz would have required smaller, 1/4-inch microphones and yet another measurement pass because the accuracy of the probe used was specified only to 5000 Hz. Fortunately, the sound pressure measurements in test case 1 proved that the energy in the 100 Hz

band combined with the energy in the 6300 Hz and 8000 Hz 1/3-octave bands were more than 10 dB below the overall level. The frequency range of interest was therefore chosen as 125 Hz to 5000 Hz.

Test Environment

A quiet conference room was chosen where the background noise was steady. ECMA-160 requires the field indicator F , or surface averaged pressure-intensity index, L_{PI} , to be less than 5 dB in all frequencies of interest. This limits the amount of reactive intensity allowed in the acoustic environment. A plot of field indicator F vs. frequency for the conference room is shown in Figure 18.

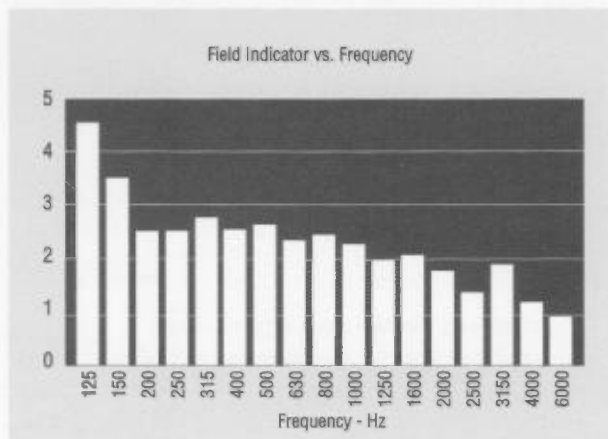


Figure 18: Field indicator in conference room shows an acceptable acoustic environment.

As the plot shows, the conference room provided an acceptable measurement environment for all frequencies between 125 Hz and 5000 Hz.

Instrumentation

- The HP 3569A portable 1/3-octave sound intensity analyzer was used for measurements. It meets or exceeds the requirements of IEC 1043 for Class 1 processors.
- A Bruel and Kjaer Type 3545 sound intensity probe was fitted with a 12-mm spacer for these measurements. It meets or exceeds the requirements of IEC 1043 for Class 1 probes.
- The frequency span of interest was 125 Hz to 5000 Hz as per ECMA-160, Section 8.6. ECMA-160 does not require conformance to IEC 1043, but instead

specifies a minimum L_{RPI} of 15 dB for the sound intensity analyzer and probe. Because Class 1 performance requires L_{RPI} to be 15 dB or better above 100 Hz, the requirements of ECMA-160 are met by this probe/analyzer combination.

The combination of a field indicator F of less than 5 dB (good environment) with an L_{RPI} greater than 15 dB (good instrument) ensured that the intensity measurements were accurate to within 0.5 dB as shown by figure 12.

Equipment Setup and Operation

The printer was placed on the floor. During the acoustic test, printing operation was continuous.

Measurement Surface and Scan Rate

The measurement surface was a small parallelepiped with a total surface area of 2.4 m² as shown in Table 4. The measurement distance to each surface was 0.35 m. The sound intensity probe was oriented normal to the measurement surface and scanned at a rate of 0.5 m/s. Two scans per surface segment were measured, with the second scan at right angles to the first.

ECMA-160 specifies that the minimum averaging time must be proportional to the bandwidth of the lowest 1/3-octave band as follows:

$$T = \frac{800}{BW},$$

$$BW = \left(2^{1/6} - 2^{-1/6} \right) (CF) = 0.232 (125) = 29 \text{ Hz},$$

where BW is the bandwidth and CF is the center frequency of the lowest 1/3-octave.

$$T = \frac{800}{BW} = \frac{800}{29} = 28 \text{ seconds}.$$

Table 4: Surface Measurements

Surface	Area - m ²	Total Average Time - T
1	0.793	65
2	0.383	46
3	0.400	39
4	0.383	40
5	0.400	45
Total Area 2.36		

Calibration and Field Checks

- The sound pressure level of each microphone of the sound intensity probe was checked with a calibrator accurate to within ± 0.5 dB.
- A field check of sound intensity was done by comparing the A-weighted sound intensity at the measurement surface to the level measured when the probe was rotated 180°, with the center of the probe in the same location. The difference in levels was 0.6 dB, which is within the 0.9 dB level of acceptability as specified by ECMA-160.

Background Sound Power Level

The printer was turned off and the measurement surface scanned. The apparent sound power of the printer was 25.9 dB reference: 1pW. ECMA-160 specifies that this environment is acceptable for measuring sound power from devices generating 10 dB greater than this level, or sound power greater than 35.9 dB.

Results

The HP 3569A measures and stores the gated average for each measurement surface. A table entry of surface area shown in Table 4 is used to compute sound power directly from the sound intensity data. A marker function in the HP 3569A automatically computes the surface averaged 1/3-octave sound power as shown in Figure 19. This spectrum represents the net sound power level emanating from the printer in each 1/3-octave band.

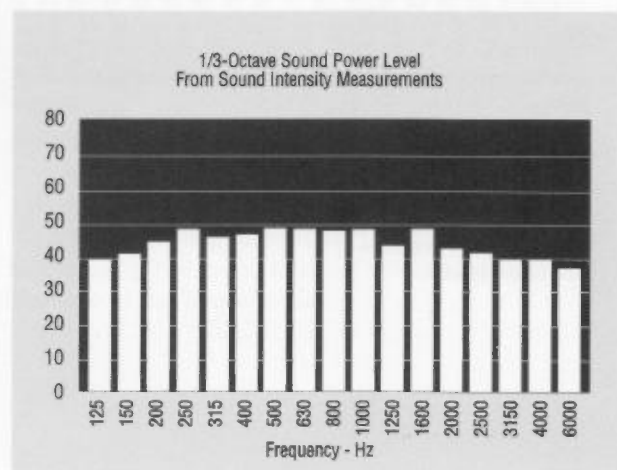


Figure 19: Surface averaged sound power is determined using a marker function on the HP 3569A.

The total A-weighted sound power level from 125 Hz to 5000 Hz was indicated on the HP 3569A to be 55.3 dB reference: 1 pW.

Comparison of Test Results

Test case 1 produced an A-weighted sound power level, L_{WA} , of 56.3 dB reference: 1 pW using sound pressure measurements in a semi-anechoic chamber. Test case 2 produced an A-weighted sound power level of 55.3 dB reference: 1 pW using sound intensity scanning. This 1 dB difference between the two tests is reasonable even if the same printer was used for both measurements because the standard deviation on the overall level for both techniques is 1.5 dB.

A different printer was used in each example test, and figure 20 shows that the two printers indeed have a unique acoustic signature. The printer used

in the semi-anechoic chamber had an audible fan tone at 250 Hz that showed up clearly in the data. This tone increased the overall A-weighted sound power level of the printer. In fact if the data is modified so that the level of the tone at 250 Hz and its harmonic at 500 Hz are closer to the surrounding bands, the overall A-weighted sound power drops to 55.3 dB reference: 1 pW, the same level measured using the intensity scanning technique.

The two measurement techniques yielded very similar results in the 1/3-octave bands outside of the tone. Both sets of data would provide useful information to a noise control engineer for development of materials for sound absorption or installation techniques that would minimize noise.

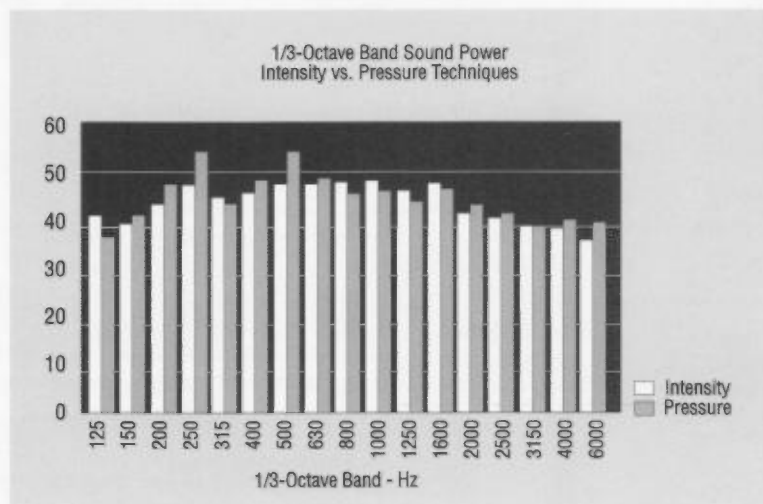


Figure 20: Comparison of sound power levels using pressure and intensity measurement methods, for two LaserJet printers of the same model.

Appendix A: Acoustic Terminology

Active and Reactive Intensity

Sound fields in general are composed of two components of intensity: active and reactive. Two components exist because particle velocity in a sound field has two components relative to instantaneous pressure: velocity in-phase with pressure fluctuations and velocity out-of-phase with pressure fluctuations. The product of pressure and in-phase velocity is called *active intensity*, or just *intensity*. Active intensity measures the net flow of acoustic energy. Intensity of a plane wave in a free field is purely active.

Reactive intensity measures potential energy in a wavefront. A standing wave in a tube has no net intensity or energy flow, but does contain potential energy in the form of reactive intensity.

An acoustic environment that has a significant component of reactive intensity is called a *reactive environment*. Examples of reactive environments are reverberation chambers, wave tubes, gymnasiums, and shower stalls.

The presence of reactive intensity makes active intensity more difficult to measure because of the reduced phase change across the sound intensity probe's spacer and the increased noise due to any phase mismatch in the probe and analyzer.

Free Field

Free field conditions exist when an acoustic wave travels unimpeded by boundaries, and is thus free from reflection and diffraction. Examples of free field conditions are the path between an aircraft and the ground, or an open field free from obstructions. Free field conditions can be frequency-dependent. An anechoic chamber may simulate free field conditions above 100 Hz, but below that frequency the walls may reflect too much sound energy to be considered free field.

Diffuse Field

A *diffuse field* exists in a reverberant chamber or reflective room, where all noise sources can be detected from all angles. Machine shops can produce diffuse sound fields because of the high reflection coefficients of the walls, floors and ceilings.

Near Field

Close to a noise source, a sound field is composed of both reactive and active sound intensity. In this region, called the near field, the active intensity may be dominated by the reactive component. Farther from the noise source, the reactive component dissipates quickly unless reflecting surfaces are present. The near field is limited to about one wavelength from a noise source.

Far Field

Far field is the region beyond near field effects. Sound pressure measurements used for sound power calculation must be made in the far field because one microphone, by itself, cannot differentiate between active and reactive sound. Measurements in the near field would over estimate the sound power of a noise source.

Pressure-Intensity Index - L_{PI}

L_{PI} measures the difference between pressure and intensity. It is an excellent indication of the acoustic environment, because in a free field where pressure equals intensity, L_{PI} equals 0 dB. In very reactive environments, intensity may be 15 to 20 dB less than pressure. An environment with a high L_{PI} indicates that intensity will be difficult to measure.

Residual Pressure Intensity Index - L_{RPI}

When intensity is measured in a completely reactive environment, such as in a probe calibration chamber or wave tube where the pressure at each microphone is the same and therefore no particle velocity is present, the analyzer should indicate zero intensity. Because of slight phase mismatch, however, the probe and analyzer will indicate a residual intensity. L_{RPI} is equal to the pressure minus the residual intensity component measured in a completely reactive environment. Residual intensity depends on frequency, the phase accuracy of the measurement system, as well as the size of the probe's spacer.

Dynamic Capability

The dynamic capability of a measurement system determines the limits of acoustic environments (range of L_{PI}) which would produce accurate intensity measurements. It has been shown that if L_{PI} is at least 7 dB less than L_{RPI} in all frequency bands of interest, then the measurement accuracy will be better than ± 1 dB. For example, ECMA-160 specifies a minimum L_{RPI} for the analyzer and probe to be 15 dB. At low frequencies this is a very stringent specification which requires phase accuracy better than $\pm 0.02^\circ$. ECMA-160 also constrains the measurement environment to have a surface-averaged L_{PI} less than 5 dB. The required dynamic capability of a ECMA-160 measurement system is therefore 10 dB or better. This produces a measurement accuracy of ± 0.5 dB.

Measurement Surface

When measuring sound power level, the equipment under test is enclosed by a measurement surface. The measurement surface can be a parallelepiped (rectangular box), a hemisphere, or a conformal shape. The objective when measuring sound power is to accurately determine the intensity emanating from each measurement surface.

Measurement Distance

Measurement distance is the distance from the equipment under test to the nearest measurement surface. A typical measurement distance for sound pressure measurements in a semi-anechoic chamber is 1 m. This distance allows pressure measurements to be made in the far field for all but the lowest frequencies. A sound intensity measurement distance is typically from 0.1 m to 1 m, although intensity measurements can vary depending on the acoustic environment and the geometry of the equipment under test.

Appendix B: Sound Intensity Approximation Errors

Using two microphones separated by a spacer to determine sound intensity is a reliable, accurate measurement method as long as its limitations are understood. This appendix will investigate the potential errors that can result from using the two-microphone approximation technique.

Previously, the relationship between pressure and particle velocity was shown: $\vec{I} = P \times \vec{v}$.

From this relationship the equation for intensity was developed for the two microphone sound intensity probe:

$$I_{\text{probe}} = -\frac{P_2 + P_1}{2\rho_0 \Delta r} \int_0^t (P_2 - P_1) dt.$$

This equation uses two approximations. The first approximation estimates the pressure at the midpoint of the probe as the average value of the pressures measured at each microphone:

$$P = \frac{P_1 + P_2}{2}.$$

The second approximation estimates the pressure gradient at the midpoint of the probe as the slope of pressure across the two microphones:

$$\frac{\partial P}{\partial x}(x, t) \cong \frac{P_2 - P_1}{\Delta r}.$$

Both linear approximations produce errors when the acoustic signal changes too much from one microphone to the other, when a straight line no longer represents the signal accurately. To analyze the resulting error, a pure tone in a free field will be "measured" theoretically by a probe that is perfectly phase-matched. This result will be compared to the theoretically derived intensity at the probe's midpoint. The ratio of these two values represents the approximation error from using a two-microphone sound intensity probe.

Note that perfectly matched microphones are used so that the error due to the straight line approximation can be decoupled from errors due to phase mismatch.

True Intensity of a Free Field Tone

A pure tone in a free field can be represented by the following equation:

$$P(x, t) = P_0 \sin\left(\frac{2\pi}{\lambda}x - 2\pi ft\right)$$

Since pressure is known exactly over x and t , then the true value of average intensity can also be computed exactly. From Newton's Second Law, the particle velocity can be determined from the pressure gradient as follows:

$$\frac{\partial v}{\partial t}(x, t) = -\frac{1}{\rho_0} \frac{\partial P}{\partial x}(x, t), \text{ and}$$

$$\frac{\partial P}{\partial x}(x, t) = P_0 \frac{2\pi}{\lambda} \cos\left(\frac{2\pi}{\lambda}x - 2\pi ft\right), \text{ so}$$

$$\frac{\partial v}{\partial t}(x, t) = -P_0 \frac{2\pi}{\lambda \rho_0} \cos\left(\frac{2\pi}{\lambda}x - 2\pi ft\right).$$

Velocity as a function of time can be derived by integrating acceleration:

$$v(x, t) = \int_0^t \frac{\partial v(x, t)}{\partial t} dt = -P_0 \frac{2\pi}{\lambda \rho_0} \int_0^t \cos\left(\frac{2\pi}{\lambda}x - 2\pi ft\right) dt = -P_0 \frac{2\pi}{\lambda \rho_0} \frac{-1}{2\pi f} \left[\sin\left(\frac{2\pi}{\lambda}x - 2\pi ft\right) \right]_0^t$$

$$v(x, t) = \frac{P_0}{\rho_0 c} \left\{ \sin\left(\frac{2\pi}{\lambda}x - 2\pi ft\right) - \sin\left(\frac{2\pi}{\lambda}x\right) \right\}$$

Computing the theoretical sound intensity in the middle of the probe can be accomplished from the equation for intensity:

$$I = P(x, t)v(x, t) = P_0 \sin\left(\frac{2\pi}{\lambda}x - 2\pi ft\right) \frac{P_0}{\rho_0 c} \times$$

$$\left\{ \sin\left(\frac{2\pi}{\lambda}x - 2\pi ft\right) - \sin\left(\frac{2\pi}{\lambda}x\right) \right\}$$

$$I = \frac{P_0^2}{\rho_0 c} \sin\left(\frac{2\pi}{\lambda}x - 2\pi ft\right) \left\{ \sin\left(\frac{2\pi}{\lambda}x - 2\pi ft\right) - \sin\left(\frac{2\pi}{\lambda}x\right) \right\}$$

At the midpoint of the probe, where $x = \Delta r/2$, instantaneous intensity can be written exactly as:

$$I_m = \frac{P_0^2}{\rho_0 c} \left\{ \sin^2 \left(\frac{\pi \Delta r}{\lambda} - 2\pi f t \right) - \sin \left(\frac{\pi \Delta r}{\lambda} - 2\pi f t \right) \sin \left(\frac{\pi \Delta r}{\lambda} \right) \right\}$$

which can be stated as:

$$I_m = \frac{P_0^2}{\rho_0 c} \left\{ \frac{1}{2} - \frac{1}{2} \cos \left(2 \left(\frac{\pi \Delta r}{\lambda} - 2\pi f t \right) \right) - \sin \left(\frac{\pi \Delta r}{\lambda} - 2\pi f t \right) \sin \left(\frac{\pi \Delta r}{\lambda} \right) \right\}.$$

The average intensity over time, T, can be calculated as follows:

$$\bar{I}_m = \frac{1}{T} \int_0^T \frac{P_0^2}{\rho_0 c} \left\{ \frac{1}{2} - \frac{1}{2} \cos \left(2 \left(\frac{\pi \Delta r}{\lambda} - 2\pi f t \right) \right) - \sin \left(\frac{\pi \Delta r}{\lambda} - 2\pi f t \right) \sin \left(\frac{\pi \Delta r}{\lambda} \right) \right\} dt$$

$$\bar{I}_m = \frac{1}{T} \frac{P_0^2}{\rho_0 c} \left[\frac{t}{2} + \frac{1}{8\pi f} \sin \left(2 \left(\frac{\pi \Delta r}{\lambda} - 2\pi f t \right) \right) - \frac{1}{2\pi f} \sin \left(\frac{\pi \Delta r}{\lambda} \right) \cos \left(\frac{\pi \Delta r}{\lambda} - 2\pi f t \right) \right]_0^T$$

$$\bar{I}_m = \frac{1}{T} \frac{P_0^2}{\rho_0 c} \left\{ \frac{T}{2} + \frac{1}{8\pi f} \sin \left(2 \left(\frac{\pi \Delta r}{\lambda} - 2\pi f T \right) \right) - \frac{1}{2\pi f} \sin \left(\frac{\pi \Delta r}{\lambda} \right) \cos \left(\frac{\pi \Delta r}{\lambda} - 2\pi f T \right) - \frac{1}{8\pi f} \sin \left(2 \left(\frac{\pi \Delta r}{\lambda} \right) \right) + \frac{1}{2\pi f} \sin \left(\frac{\pi \Delta r}{\lambda} \right) \cos \left(\frac{\pi \Delta r}{\lambda} \right) \right\}$$

When integrating over an integral number of cycles,

$T = 1/f$, so $\cos(\alpha + 2\pi f T) = \cos(\alpha)$, and $\sin(\alpha + 2\pi f T) = \sin(\alpha)$.

Under this assumption the equation simplifies to:

$$\bar{I}_m = \frac{1}{T} \frac{P_0^2}{\rho_0 c} \frac{1}{4\pi f} \left\{ 2\pi f T + \frac{1}{2} \sin \left(2 \left(\frac{\pi \Delta r}{\lambda} \right) \right) - \sin \left(2 \left(\frac{\pi \Delta r}{\lambda} \right) \right) - \frac{1}{2} \sin \left(2 \left(\frac{\pi \Delta r}{\lambda} \right) \right) + \sin \left(2 \left(\frac{\pi \Delta r}{\lambda} \right) \right) \right\}$$

which simplifies to:

$$\bar{I}_m = \frac{1}{T} \frac{P_0^2}{\rho_0 c} \frac{1}{4\pi f} \{ 2\pi f T \} = \frac{P_0^2}{2\rho_0 c}$$

and since $\frac{P_0^2}{2} = P_{rms}^2$,

$$\bar{I}_m = \frac{P_{rms}^2}{\rho_0 c}.$$

This is the theoretical value of the average intensity of a free-field plane wave calculated at a probe's midpoint.

Approximation of Intensity of a Free-Field Tone using Measurements at Two Microphones

Previously the equation used to calculate intensity from two microphone measurements was derived:

$$I_{\text{probe}} = -\frac{P_2 + P_1}{2\rho_0\Delta r} \int_0^t (P_2 - P_1) dt$$

Using the exact expression for pressure,

$$P(x, t) = P_0 \sin\left(\frac{2\pi}{\lambda}x - 2\pi ft\right),$$

the equation for instantaneous intensity measured by the probe can be derived. Since P_1 is measured at $x = 0$ and P_2 is measured at $x = \Delta r$, the following equations apply:

$$P_1 = P_0 \sin(-2\pi ft)$$

$$P_2 = P_0 \sin\left(2\pi \frac{\Delta r}{\lambda} - 2\pi ft\right)$$

$$I_{\text{probe}} = -\frac{P_0 \sin\left(2\pi \frac{\Delta r}{\lambda} - 2\pi ft\right) + P_0 \sin(-2\pi ft)}{2\rho_0\Delta r} \int_0^t \left\{ P_0 \sin\left(2\pi \frac{\Delta r}{\lambda} - 2\pi ft\right) - P_0 \sin(-2\pi ft) \right\} dt$$

$$I_{\text{probe}} = -\frac{P_0^2}{4\pi f\rho_0\Delta r} \left\{ \sin\left(2\pi \frac{\Delta r}{\lambda} - 2\pi ft\right) + \sin(-2\pi ft) \right\} \left[\cos\left(2\pi \frac{\Delta r}{\lambda} - 2\pi ft\right) - \cos(-2\pi ft) \right]_0^t$$

$$I_{\text{probe}} = -\frac{P_0^2}{4\pi f\rho_0\Delta r} \left\{ \sin\left(2\pi \frac{\Delta r}{\lambda} - 2\pi ft\right) + \sin(-2\pi ft) \right\} \left(\cos\left(2\pi \frac{\Delta r}{\lambda} - 2\pi ft\right) - \cos(-2\pi ft) - \cos\left(2\pi \frac{\Delta r}{\lambda}\right) + 1 \right).$$

This long equation gives the instantaneous intensity measured by the probe. To get the average intensity, it is integrated from $t=0$ to $t=T$.

The above equation can be broken down into eight terms and integrated separately:

$$I_{\text{probe}} = -\frac{P_0^2}{4\pi f\rho_0\Delta r} (T_1 + T_2 + T_3 + T_4 + T_5 + T_6 + T_7 + T_8),$$

where the terms are as follows:

$$T_1 = \sin\left(2\pi \frac{\Delta r}{\lambda} - 2\pi ft\right) \cos\left(2\pi \frac{\Delta r}{\lambda} - 2\pi ft\right) = \frac{1}{2} \sin\left(2\left(2\pi \frac{\Delta r}{\lambda} - 2\pi ft\right)\right)$$

$$T_2 = -\sin\left(2\pi \frac{\Delta r}{\lambda} - 2\pi ft\right) \cos(-2\pi ft) = \frac{1}{2} \cos\left(2\pi \frac{\Delta r}{\lambda}\right) \sin(2(2\pi ft)) - \frac{1}{2} \sin\left(2\pi \frac{\Delta r}{\lambda}\right) - \frac{1}{2} \sin\left(2\pi \frac{\Delta r}{\lambda}\right) \cos(2(2\pi ft))$$

$$T_3 = -\sin\left(2\pi \frac{\Delta r}{\lambda} - 2\pi ft\right) \cos\left(2\pi \frac{\Delta r}{\lambda}\right)$$

$$T_4 = \sin\left(2\pi \frac{\Delta r}{\lambda} - 2\pi ft\right)$$

$$T_5 = \sin(-2\pi ft) \cos\left(2\pi \frac{\Delta r}{\lambda} - 2\pi ft\right) = \frac{1}{2} \sin\left(2\pi \frac{\Delta r}{\lambda}\right) \cos 2(-2\pi ft) + \frac{1}{2} \cos\left(2\pi \frac{\Delta r}{\lambda}\right) \sin(2(-2\pi ft)) - \frac{1}{2} \sin\left(2\pi \frac{\Delta r}{\lambda}\right)$$

$$T_6 = -\sin(-2\pi ft) \cos(-2\pi ft) = \frac{1}{2} \sin(2(2\pi ft))$$

$$T_7 = -\sin(-2\pi ft) \cos\left(2\pi \frac{\Delta r}{\lambda}\right)$$

$$T_8 = \sin(-2\pi ft).$$

Evaluating the integral over an integral number of cycles so that $T = 1/f$, each term contributes the following when integrated:

$$\int_0^T T_1 dt = \int_0^T \frac{1}{2} \sin\left(2\left(2\pi \frac{\Delta r}{\lambda} - 2\pi ft\right)\right) dt = \frac{1}{8\pi f} \left\{ \cos\left(2\left(2\pi \frac{\Delta r}{\lambda} - 2\pi fT\right)\right) - \cos\left(2\left(2\pi \frac{\Delta r}{\lambda}\right)\right) \right\} = 0$$

$$\int_0^T T_2 dt = \frac{1}{2} \int_0^T \left\{ \cos\left(2\pi \frac{\Delta r}{\lambda}\right) \sin(2(2\pi ft)) - \sin\left(2\pi \frac{\Delta r}{\lambda}\right) - \sin\left(2\pi \frac{\Delta r}{\lambda}\right) \cos(2(2\pi ft)) \right\} dt = \frac{-T}{2} \sin\left(2\pi \frac{\Delta r}{\lambda}\right)$$

$$\int_0^T T_2 dt = \frac{1}{2} \int_0^T \left\{ \cos\left(2\pi \frac{\Delta r}{\lambda}\right) \sin(2(2\pi ft)) - \sin\left(2\pi \frac{\Delta r}{\lambda}\right) - \sin\left(2\pi \frac{\Delta r}{\lambda}\right) \cos(2(2\pi ft)) \right\} dt = \frac{-T}{2} \sin\left(2\pi \frac{\Delta r}{\lambda}\right)$$

$$\int_0^T T_4 dt = \int_0^T \sin\left(2\pi \frac{\Delta r}{\lambda} - 2\pi ft\right) dt = 0$$

$$\int_0^T T_5 dt = \frac{1}{2} \int_0^T \left\{ \sin\left(2\pi \frac{\Delta r}{\lambda}\right) \cos(2(-2\pi ft)) + \cos\left(2\pi \frac{\Delta r}{\lambda}\right) \sin(2(-2\pi ft)) - \sin\left(2\pi \frac{\Delta r}{\lambda}\right) \right\} dt = -\frac{T}{2} \sin\left(2\pi \frac{\Delta r}{\lambda}\right)$$

$$\int_0^T T_6 dt = \int_0^T \frac{1}{2} \sin(2(2\pi ft)) dt = 0$$

$$\int_0^T T_7 dt = -\int_0^T \sin(-2\pi ft) \cos\left(2\pi \frac{\Delta r}{\lambda}\right) dt = 0$$

$$\int_0^T T_8 dt = \int_0^T \sin(-2\pi ft) dt = 0.$$

The two-microphone approximation of the time averaged intensity at the probe is then equal to:

$$I_{\text{probe}} = \frac{P_0^2}{4\pi f \rho_0 \Delta r} \sin\left(2\pi \frac{\Delta r}{\lambda}\right) = \frac{P_{\text{rms}}^2}{2\pi f \rho_0 \Delta r} \sin\left(2\pi \frac{\Delta r}{\lambda}\right).$$

Because $\lambda f = c$, this can be re-stated as:

$$I_{\text{probe}} = \frac{P_{\text{rms}}^2}{\rho_0 c} \frac{\sin\left(2\pi \frac{\Delta r}{\lambda}\right)}{2\pi \frac{\Delta r}{\lambda}}.$$

Comparing True Intensity to Measured Intensity

The ratio of true intensity I_m to the measured intensity I_{probe} , indicates the extent of the approximation error:

$$\frac{I_{\text{probe}}}{I_m} = \frac{\frac{P_{\text{rms}}^2 \sin\left(2\pi \frac{\Delta r}{\lambda}\right)}{\rho_0 c}}{\frac{P_{\text{rms}}^2}{\rho_0 c}} = \frac{\sin\left(2\pi \frac{\Delta r}{\lambda}\right)}{2\pi \frac{\Delta r}{\lambda}},$$

and the error in dB is equal to:

$$e_{\text{probe}} = 10 \log\left(\frac{I_{\text{probe}}}{I_m}\right) = 10 \log\left(\frac{\sin\left(2\pi \frac{\Delta r}{\lambda}\right)}{2\pi \frac{\Delta r}{\lambda}}\right).$$

This can be described graphically by showing the error in dB versus the spacer size to wavelength ratio:

$$\frac{\Delta r}{\lambda}.$$

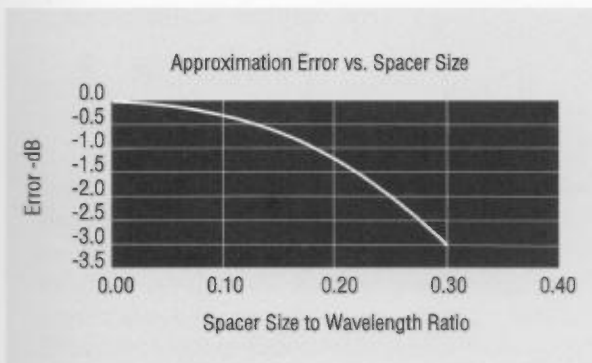


Figure B1: Intensity bias in dB caused by finite difference approximation error.

From this chart we see that the intensity approximation always underestimates the true intensity level. It is also possible to determine the maximum spacer to wavelength ratio that yields an error of less than 1 or 2 dB. For a 1 dB maximum error, the maximum spacer to wavelength ratio is 0.18. For a 2 dB maximum error, it is permissible to use a 0.25 ratio.

Appendix C: Theoretical Analysis of Sound Intensity Measurements

Intensity in an acoustic wave is equal to pressure times velocity: $\vec{I} = P \times \vec{v}$. As discussed previously, a two-microphone sound intensity probe can be used to calculate each component of this relationship:

$$P = \frac{P_1 + P_2}{2}, \text{ and}$$

$$v_{\text{probe axis}} = \int_0^t \frac{\partial v}{\partial t}(x, t) dt = -\frac{1}{\rho_0} \int_0^t \frac{P_2 - P_1}{\Delta r} dt.$$

The instantaneous intensity along the axis of a sound intensity probe can therefore be described as follows:

$$I = -\left(\frac{P_2 + P_1}{2}\right) \frac{1}{\rho_0} \int \frac{(P_2 - P_1)}{\Delta r} dt.$$

This equation can be used to analyze the relationship between measurement parameters, including the probe configuration, the acoustic environment, and the phase difference between channels. Knowing these relationships can help test engineers make intelligent trade-offs when selecting the test room, probe, analyzer, and in setting up the measurements.

The sound intensity measurement system measures pressure at the two microphones that can be represented as follows:

$$P_1 = P_{1_0} \sin(2\pi ft) \text{ and } P_2 = P_{2_0} \sin(2\pi ft - \phi),$$

where P_1 and P_2 are the peak pressures at frequency f for each microphone, t is the time, and

ϕ is the measured phase difference between channel 1 and channel 2 at frequency f .

Using the equation above for instantaneous intensity, the average intensity over time, T , can be expressed in terms of P_1 and P_2 :

$$I = -\frac{1}{T} \int_0^T \left[\frac{P_{2_0} \sin(2\pi ft - \phi) + P_{1_0} \sin(2\pi ft)}{2} \frac{1}{\rho_0} \int_0^t \frac{P_{2_0} \sin(2\pi ft - \phi) - P_{1_0} \sin(2\pi ft)}{\Delta r} dt \right] dt.$$

Notice that velocity is calculated by integrating the pressure difference in the inside integral from time 0 to t. Evaluating this inside integral first, the equation becomes:

$$I = -\frac{1}{4\pi f \rho_0 \Delta r T} \int_0^T \left[\left\{ P_{2_0} \sin(2\pi ft - \phi) + P_{1_0} \sin(2\pi ft) \right\} \left(P_{1_0} \cos(2\pi ft) - P_{1_0} - P_{2_0} \cos(2\pi ft - \phi) + P_{2_0} \cos(-\phi) \right) \right] dt$$

multiplying terms yields eight new terms:

$$I = -\frac{1}{4\pi f \rho_0 \Delta r T} \int_0^T [T_1 + T_2 + T_3 + T_4 + T_5 + T_6 + T_7 + T_8] dt,$$

where the terms are as follows:

$$T_1 = P_{2_0} \sin(2\pi ft - \phi) P_{1_0} \cos(2\pi ft) = P_{1_0} P_{2_0} \left(\frac{1}{2} \sin(4\pi ft) \cos \phi - \frac{1}{2} (1 + \cos(4\pi ft)) \sin \phi \right)$$

$$T_2 = -P_{2_0} \sin(2\pi ft - \phi) P_{1_0}$$

$$T_3 = -P_{2_0}^2 \sin(2\pi ft - \phi) \cos(2\pi ft - \phi) = -P_{2_0}^2 \frac{1}{2} \sin 2(2\pi ft - \phi)$$

$$T_4 = P_{2_0}^2 \sin(2\pi ft - \phi) \cos(\phi)$$

$$T_5 = P_{1_0}^2 \sin(2\pi ft) \cos(2\pi ft) = P_{1_0}^2 \frac{1}{2} \sin(4\pi ft)$$

$$T_6 = -P_{1_0}^2 \sin(2\pi ft)$$

$$T_7 = -P_{1_0} \sin(2\pi ft) P_{2_0} \cos(2\pi ft - \phi) = -\frac{1}{2} P_{1_0} P_{2_0} \left\{ \sin(4\pi ft) \cos(-\phi) + (1 - \cos(4\pi ft)) \sin \phi \right\}$$

$$T_8 = P_{1_0} \sin(2\pi ft) P_{2_0} \cos(\phi)$$

The following equalities were used:

$$\cos \alpha \sin \alpha = \frac{1}{2} \sin 2\alpha$$

$$\cos^2 \alpha = \frac{1}{2} (1 + \cos 2\alpha)$$

$$\sin^2 \alpha = \frac{1}{2} (1 - \cos 2\alpha)$$

$$\sin^2 \alpha = \frac{1}{2} (1 - \cos 2\alpha), \text{ and also}$$

$$\sin(\alpha + \beta) = \sin \alpha \cos \beta + \cos \alpha \sin \beta, \text{ each term becomes:}$$

Integrating each term over an integral number of cycles yields the following results:

$$\int_0^T T_1 dt = P_{1_0} P_{2_0} \left\{ \frac{-\cos \phi}{8\pi f} (\cos 4\pi f T - 1) - \frac{\sin \phi}{2} \left(T + \frac{\sin 4\pi f T}{4\pi f} \right) \right\} = -P_{1_0} P_{2_0} \frac{T \sin \phi}{2}$$

Combining results from all of the terms, the measured intensity can be described as follows:

$$\int_0^T T_2 dt = P_{1_0} P_{2_0} \frac{1}{2\pi f} \{(\cos 2\pi f T - \phi) - \cos(-\phi)\} = 0$$

$$\int_0^T T_3 dt = P_{2_0}^2 \frac{1}{8\pi f} \{\cos(4\pi f T - 2\phi) - \cos(-2\phi)\} = 0$$

$$\int_0^T T_4 dt = P_{2_0}^2 \frac{1}{2\pi f} \cos \phi \{(\cos 2\pi f T - \phi) - \cos(-\phi)\} = 0$$

$$\int_0^T T_5 dt = -P_{1_0}^2 \frac{1}{8\pi f} (\cos 4\pi f T - 1) = 0$$

$$\int_0^T T_6 dt = P_{1_0}^2 \frac{1}{2\pi f} (\cos 2\pi f T - 1) = 0$$

$$\int_0^T T_7 dt = -P_{1_0} P_{2_0} \left\{ \frac{-\cos \phi}{8\pi f} (\cos 4\pi f T - 1) + \frac{\sin \phi}{2} \left(T - \frac{\sin 4\pi f T}{4\pi f} \right) \right\} = -P_{1_0} P_{2_0} \frac{T \sin \phi}{2}$$

$$\int_0^T T_8 dt = P_{1_0} P_{2_0} \frac{\cos(-\phi)}{2\pi f} (\cos 2\pi f T - 1) = 0$$

$$I = -\frac{P_{1_0} P_{2_0}}{4\pi f \rho_0 \Delta r T} \left(-\frac{\sin \phi}{2} T - \frac{\sin \phi}{2} T \right) = \frac{P_{1_0} P_{2_0} \sin \phi}{4\pi f \rho_0 \Delta r}$$

Using more familiar rms pressure terms, the equation can be restated as:

$$I = \frac{P_{1_{rms}} P_{2_{rms}} \sin(\phi)}{2\pi f \rho_0 \Delta r}$$

Using a two-microphone sound intensity probe, we've shown that intensity is proportional to the pressure at each microphone and the phase angle between the two measurements at the frequency of interest, and inversely proportional to the frequency and microphone spacing. This relationship exposes the importance of cross-channel phase accuracy of the instrument and probe, particularly for measurements at low frequencies. At low frequencies the spacer is small relative to the wavelength. Thus the pressure difference across the two microphones can be small relative to the phase accuracy of the analyzer and the phase match of the microphones.

References

Harris, Cyril, *Handbook of Acoustical Measurements and Noise Control*, 3rd Ed., McGraw-Hill, 1991

Towne, Dudley, *Wave Phenomena*, Addison-Wesley, 1967

Rasmussen, Gunnar, "Intensity—its Measurement and Uses," *Sound and Vibration*, March 1989

Gade, Svend, "Validity of Intensity Measurements in Partially Diffuse Sound Field," *Bruel & Kjaer Technical Review* No. 4-1985

ANSI S1.11-1986, "Specifications for Octave-Band and Fractional Octave-Band Analog and Digital Filters."

ANSI S12.12-1992, "Engineering Method for the Determination of Sound Power Levels of Noise Sources Using Sound Intensity."

ISO 7779:1988, "Acoustics - Measurements of Airborne Noise Emitted by Computer or Business Equipment."

ISO 3744:1981, "Acoustics - Determination of Sound Power Levels of Noise Sources - Engineering Methods for Free-Field Conditions over a Reflecting Plane."

ISO 3745:1977, "Acoustics - Determination of Sound Power Levels of Noise Sources - Precision Methods for Anechoic and Semi-Anechoic Rooms."

ECMA-160, "Determination of Sound Power Levels of Computer and Business Equipment Using Sound Intensity Measurements; Scanning Method in Controlled Rooms," December 1991.



For more information, call your local HP sales office listed in your telephone directory or an HP regional office listed below for the location of your nearest sales office.

United States:

Hewlett-Packard Company
4 Choke Cherry Road
Rockville, MD 20850
(301) 670 4300

Hewlett-Packard Company
5201 Tollview Drive
Rolling Meadows, IL 60008
(708) 255 9800

Hewlett-Packard Company
1421 S. Manhattan Ave
Fullerton, CA 92631
(213) 337-8035

Hewlett-Packard Company
2000 South Park Place
Atlanta, GA 30339
(404) 980-7351

Canada:

Hewlett-Packard Ltd.
6877 Goreway Drive
Mississauga, Ontario L4V 1M8
(416) 678 9430

Europe:

Hewlett-Packard
European Marketing Centre
P.O. Box 999
1180 AZ Amstelveen
The Netherlands
(31) 20 547 9999

Japan:

Yokogawa-Hewlett-Packard Ltd.
3-29-21 Takaido Higashi
Suginami-ku
Tokyo 168, Japan
(813) 3335 8192

Latin America:

Latin American Region Headquarters
Monte Pelvoux No. 111
Lomas de Chapultepec
11000 Mexico, D.F.
(525) 202 0155

Australia/New Zealand:

Hewlett-Packard Australia Ltd.
31-41 Joseph Street
Blackburn, Victoria 3130
Australia (A.C.N. 004 394 763)
(03) 895 2895

Far East:

Hewlett-Packard Asia Ltd.
22/F EIE Tower, Bond Centre
89 Queensway, Central
Hong Kong
(852) 848 7070

Data subject to change.
Copyright © 1992 Hewlett-Packard Co.
Printed in Singapore 7/92
5091-4426E