4 ACOUSTIC MEASUREMENTS AND NOISE METRICS

4.1 MICROPHONES

Both microphones and loudspeakers are transducers—electromechanical devices for converting sound waves into electrical signals and vice versa. Microphones sense small changes in sound pressure through motion of a thin diaphragm. Cone loudspeakers create changes in pressure through the motion of a diaphragm driven by a coil of wire, immersed in a magnetic field. Since both microphones and loudspeakers operate in a similar manner, microphones can be used as loudspeakers and loudspeakers as microphones. Even the human eardrum can act as a loudspeaker.

The most common types of microphones in use are: 1) dynamic, 2) condenser, 3) electret, 4) ceramic, and 5) ribbon. All microphones consist of a diaphragm, which moves back and forth in response to changes in pressure or velocity brought about by a sound wave, and electronic components that convert the movement into an electric signal. Microphones are characterized by a sensitivity, which is the open circuit output voltage produced by a given pressure, expressed in decibels re 1 V/Pa. A one-inch diameter instrumentation microphone might produce 54 mV for an rms pressure of 1 Pa, yielding a sensitivity of $20 \log [(54\,\text{mV}) / (1\,\text{Pa})]\,[(1\,\text{Pa}) / (1\,\text{V})] = -\ 25\,\text{dB}$. Note that 1 Pa is the sound pressure that corresponds to the 94 dB sound pressure level generated by standard pistonphone calibrators.

A dynamic microphone, illustrated in Fig. 4.1, operates on the same principal as a loudspeaker. A diaphragm moves in response to the changes in sound pressure and is mechanically connected to a coil of wire that is positioned in a magnetic field. The induced current, produced by the motion of the coil, is the microphone's output signal. Both the diaphragm and the coil must be very light to produce adequate high-frequency response. Most dynamic microphones produce a very low output voltage; however, since the electrical output impedance is low, the microphone can be located relatively far away from the preamplifier. Dynamic microphones are rugged and are primarily used in sound reinforcement applications, where low fidelity is good enough. One manufacturer of dynamic microphones used to demonstrate its product's toughness by using the side of it to pound a nail into a block of wood.

A condenser microphone, in Fig. 4.2, consists of a thin stretched stainless-steel diaphragm that is separated from a back plate by a narrow air gap. The two parallel plates

FIGURE 4.1 Dynamic (Moving Coil) Microphone (Rossing, 1990)

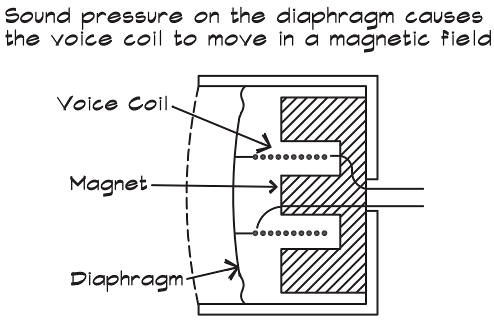
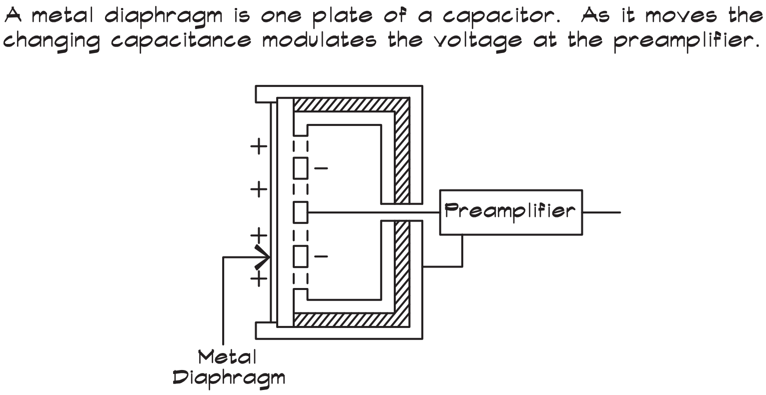


FIGURE 4.2 Condenser Microphone (Rossing, 1990)



become a capacitor when a DC polarizing voltage, typically 150 to 200 V, is applied. Motion of the diaphragm generates an electrical signal by varying the capacitance and thus the voltage between the plates. These microphones are very sensitive and accurate and have excellent frequency response characteristics. They are less rugged than dynamics and require a source of the polarizing voltage.

An electret microphone, in Fig. 4.3, is another form of condenser, which is sometimes called an electret condenser. It includes a thin polymeric diaphragm, where the polarizing voltage is not externally applied but is built into the polymer so that it is permanent. Otherwise the microphone operates in much the same way as the condenser does.

The ceramic microphone, in Fig. 4.4, has a diaphragm that is mechanically coupled to a piezoelectric material. A piezoelectric generates a voltage when strained. Many such materials exist such as lead zirconate titanate, called PZT, barium titanate, and rochelle salt. These microphones are more rugged than the capacitive types, are less sensitive, and do not require an external polarization voltage.

A ribbon microphone, sometimes referred to as a velocity microphone, works by suspending a thin metallic foil in a magnetic field. Figure 4.5 shows an example. The conducting ribbon is light enough that it responds to the particle velocity rather than the pressure. Since the ribbon is open to the back and shielded on the sides by the magnet, these microphones have a bidirectional polarity pattern. Ribbon microphones are very sensitive to moving air

FIGURE 4.3 Electret Condenser Microphone (Rossing, 1990)

A thin metallized plastic diaphragm is tightly stretched across a perforated backing plate. The holes in the back plate couple to an air cavity.

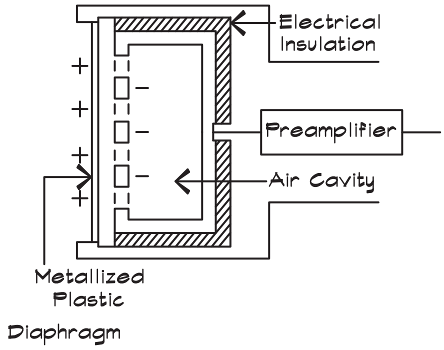


FIGURE 4.4 Ceramic Microphone (Rossing, 1990)

Sound pressure on the diaphragm causes deformation of the crystal, generating an electrical signal.

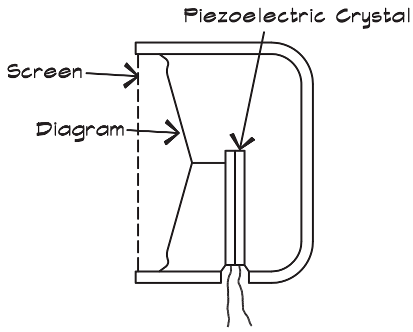


FIGURE 4.5 Ribbon Microphone (Rossing, 1990)

A lightweight ribbon diaphragm moves in a magnetic field, thus generating an electrical signal.

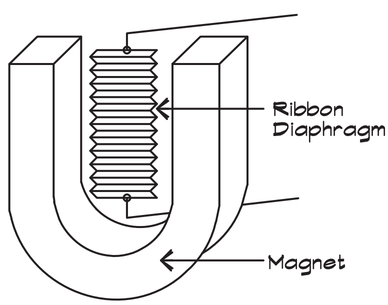
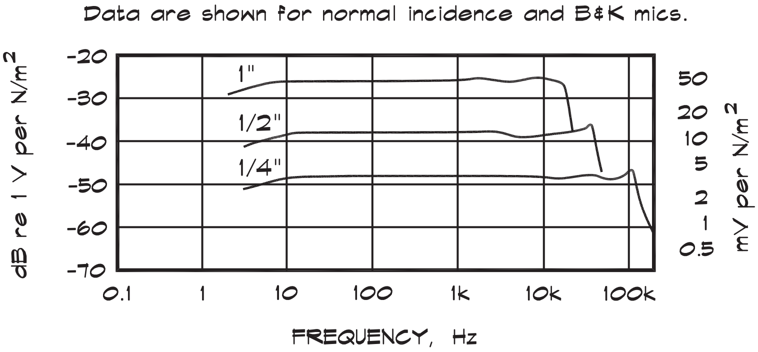


FIGURE 4.6 Sensitivity of Condenser Microphones (Hassall and Zaveri, 1979)



currents as well as high sound pressure levels. An unsuspecting acoustician, seeking to determine the characteristics of a reverberation chamber, once fired a blank pistol in a room full of ribbon microphones, quickly converting them into expensive paperweights. Due to the fragility of this type of microphone, its use is limited to the studio.

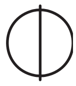
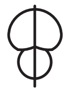
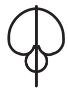
Frequency Response

Instrumentation microphones, so called because they can be calibrated using a pistonphone calibrator, are cylindrical and come in nominal sizes: one-inch (actually 0.936 in or 23.8 mm), half-inch (12.7 mm), and quarter-inch (6.5 mm) diameters. The size of a microphone affects its performance. Small microphones can measure sounds at higher frequencies and generally are less directional and less sensitive since they have a lower surface area. A one-inch instrumentation microphone, for example, might be able to measure levels as low as 0 dBA, while having an upper frequency limit of 10 kHz. A half-inch microphone might be good to 10 dBA and 30 kHz, and a quarter-inch microphone typically can measure down to 20 dBA and as high as 70 kHz. Examples of their response curves are given in Fig. 4.6.

Directional Microphones

Microphones, like sound sources, can have a response that varies with angle, which is represented by a polar diagram with angles measured relative to the normal to the diaphragm. Ideally, instrumentation microphones are nondirectional; however, at high frequencies there is some self shielding and loss of sensitivity, which is often greatest at a 120° to 150° angle of incidence. The polar diagrams for several types of microphones are shown in Table 4.1. Directional microphones are not used for precision measurements, but are quite useful for recording and sound-reinforcement systems. When the microphone capsule is smaller than a quarter wavelength, it is not directional; however, directivity can be built in by manipulating the construction of the housing. Figure 4.7 illustrates the design of a cardioid housing. By leaving an opening at the rear, sound coming from the rear arrives at the front and back of the diaphragm at the same time, thus canceling. Sound arriving from the front takes some additional time to reach the rear of the microphone diaphragm. By carefully attenuating selected frequencies traveling along certain paths the sound entering the rear cavities can be delayed so that it arrives close to 180° out of phase and does not cancel out the frontal sound.

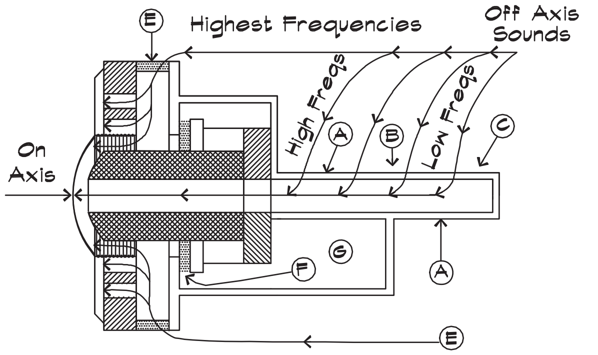
TABLE 4.1 Directional Characteristics of Microphones (Shure Inc., 2002)

	Omni-Directional	Bidirectional	Cardioid	Hypercardioid	Super-Cardioid
Polar Response Pattern					
Polar Equation	1	$\cos\theta$	$1/2(1+\cos\theta)$	$1/4(1+\cos\theta)$	$0.37+0.63\cos\theta$
Pickup Arc 3 dB Down	360°	90°	131°	105°	115°
Pickup Arc 6 dB Down	360°	120°	180°	141°	156°
Relative Output At 90° (dB)	0	$-\infty$	−6	−12	−8.6
Relative Output At 180° (dB)	0	0	$-\infty$	−6	−11.7
Angle at Which Output = 0	--	90°	180°	110°	126°
Random Energy Efficiency	0 dB	0.333 −4.8 dB	0.333 −4.8 dB	0.250* −6.0 dB	0.268** −5.7 dB
Distance Factor	1	1.7	1.7	2	1.9

* Minimum random energy efficiency for a first-order cardioid.
** Maximum front to total random energy efficiency for a first-order cardioid.

FIGURE 4.7 Cross-Section of an Electrovoice Variable-D Cardioid (Burroughs, 1974)

The letters indicate paths available for rear entry.



Highly directional microphones can be made using a series of openings in a tube, or a group of different length tubes, leading to the diaphragm. These so-called shotgun microphones work because sounds arriving on axis and entering through the holes combine in the tube in the proper phase relationship. Sounds arriving from the side and traveling down the tube combine with a random phase relationship that attenuates the signal at the diaphragm.

Directional microphones are very important in sound reinforcement systems. They selectively amplify sound coming from one direction, ideally from the user, and attenuate sound from other directions. This reduces feedback and allows a greater system gain. Properly designed directional microphones should have a consistent directivity pattern over a range of frequencies, otherwise they would color the off-axis sound. The more directional a microphone, the greater the coloration and the greater the directional lobing. Sometimes highly directional microphones can generate more system feedback than cardioid microphones, due to the influence of off-axis lobing patterns. In general, the less directional the microphone the more natural sounding it is.

Sound Field Considerations

Microphone directivity sometimes influences the method of making measurements, even with instrumentation microphones. Typical microphones have their greatest sensitivity for sound incident on the diaphragm at 0°, called normal incidence. When the sound is traveling in a direction that is parallel to the plane of the diaphragm, at 90° to the normal, it is called grazing incidence. Most microphones have an angle for which their response is the flattest, usually 0° or 90°, but sometimes it can be another angle.

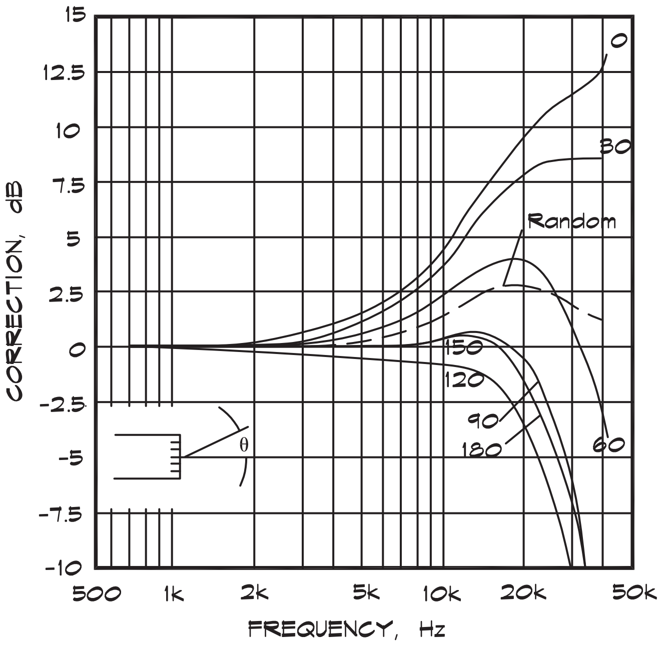
Microphones are described by their preferred type of sound field; for example, free field, random incidence, or pressure field. All microphones respond to pressure, but their sensitivity can be adjusted to produce the flattest response for a given angle of incidence or type of sound field. A free field is characterized by direct, unimpeded propagation of the wave from the source to the receiver. A diffuse or random field is one where the sound arrives from every direction with equal probability, and in a pressure field the sound pressure has the same magnitude throughout the space.

For a half-inch instrumentation microphone, below 5000 Hz all orientations produce a frequency response that is flat to within 2 dB. If a measurement is being made in a free field above 5000 Hz, the microphone should be oriented so that its flattest response direction is used, but this may vary with frequency, as can be seen in Fig. 4.8. Different standards organizations make different recommendations for proper free-field measurements (Fig. 4.9). IEC standards specify that the meter be switched to frontal mode and be oriented for normal incidence. ANSI standards require the selection of the random mode and an orientation of 70° to 80° to the source. For moving sources the microphone should be oriented for grazing incidence so that the directivity does not change with the motion of the source. This is achieved by angling the microphone upward.

When measurements are being done indoors, the random correction should be selected. Measuring with a free-field microphone in a diffuse field or with a random-incidence microphone in a free field yields only small inaccuracies, usually at high frequencies. The most accurate results will be obtained by using the setting appropriate to the type of sound field, but the differences are generally small.

FIGURE 4.8 Free Field Correction Curves for a Microphone (Bruel and Kjaer, 1986)

Half inch microphone fitted with a protective grid. This response is added to the on-axis response of the microphone as shown in Fig. 4.6.



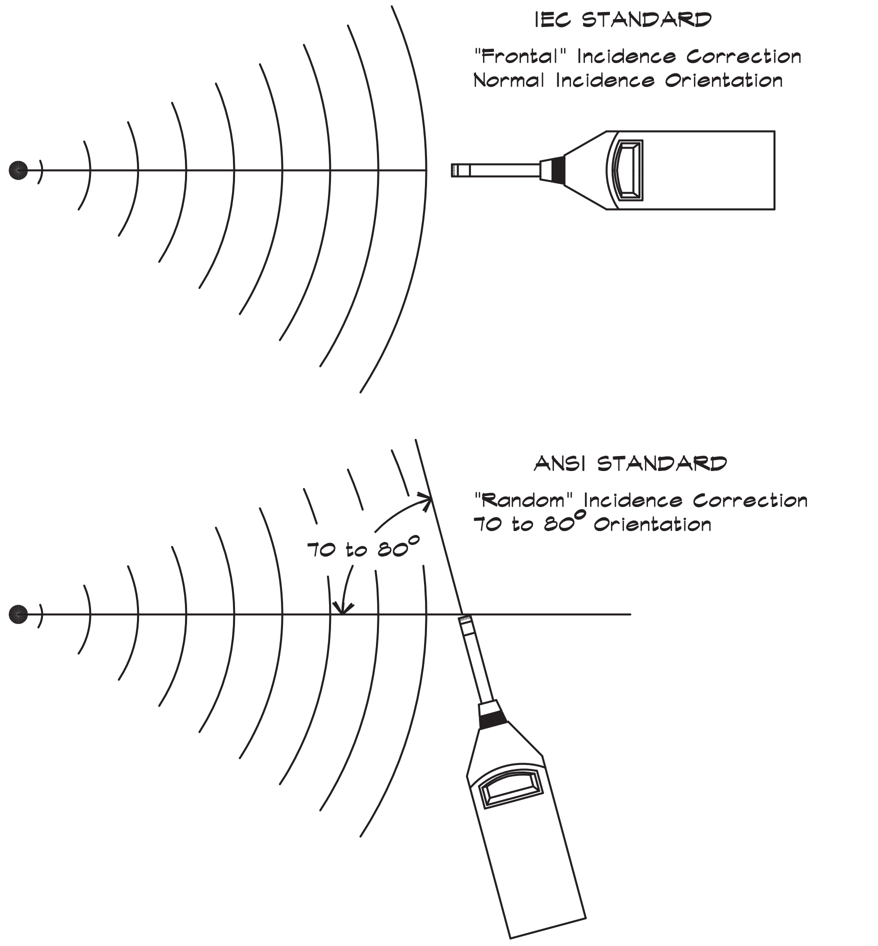
4.2 SOUND LEVEL METERS

The sound level meter, such as that shown in Fig. 4.10, is the fundamental acoustical instrument. Meters are battery powered and have become increasingly sophisticated, frequently containing internal processing, which automates many of the measurement functions. The individual controls vary from meter to meter; however, in general, there is a commonality of features. The basic controls allow for a selection of time weightings—fast, slow, and impulse—each of which represents a different ballistic time constant. Several frequency weightings are available: linear (unweighted), A-weighted, C-weighted, and a band limited linear scale. Frequency bandwidths may be selected from all pass, octave, and third-octave bands. There is a range selection that determines the highest and lowest levels measurable by the meter. Depending on the meter, there may be various types of automatic processing.

The internal parts of a meter include a microphone, preamplifier, various filters, a range control, time averager, and level indicator. The filters sometimes are contained in a separate module that may be attached to the meter, or are an integral part of the meter itself. On most hand-held sound level meters the filter selection is made manually. Where a group of filters operate simultaneously and display a number of levels on a bar graph in real time, the meter is called a spectrum analyzer or real-time analyzer.

Sound level meters are classified into three different groups by accuracy. Each class has a slightly different tolerance allowed in its precision. These standards are defined by the

FIGURE 4.9 Free Field Sound Measurements (Bruel and Kjaer, 1986)



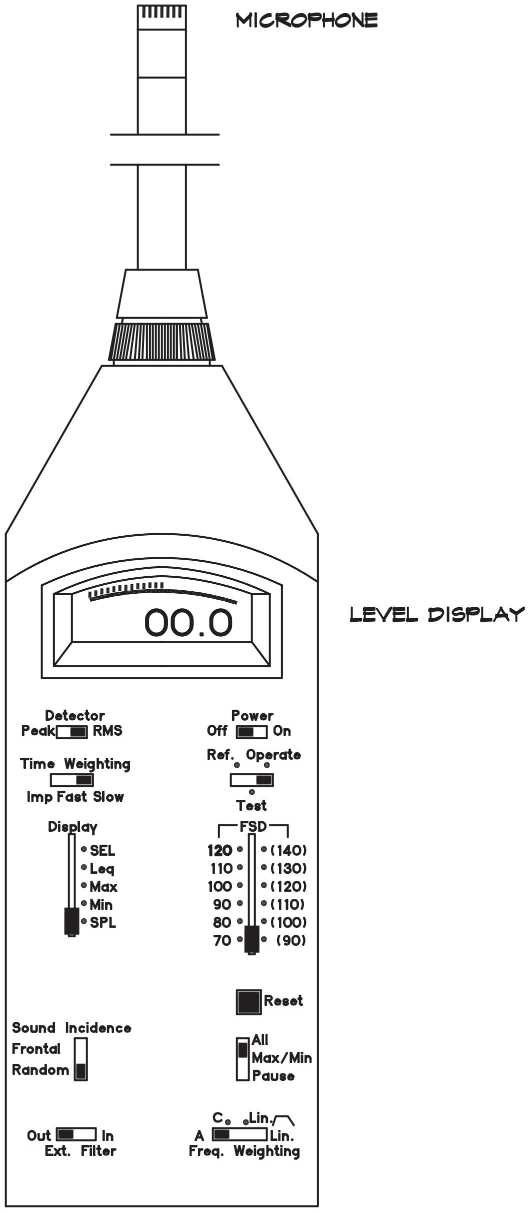
American National Standard Specification for Sound Level Meters, ANSI S1.4-1983.

Class 0	Laboratory	±0.2 dB 22.4 – 11200 Hz
Class 1	Precision	±0.5 dB 22.4 – 11200 Hz
Class 2	General Purpose	±0.5 dB 63.0 – 2000 Hz
		±1.0 dB 22.4 – 11200 Hz

Meter Calibration

Sound level meters should be calibrated before use, using a pistonphone calibrator placed over the microphone. These calibrators generate a steady tone, usually at 1000 Hz, by means

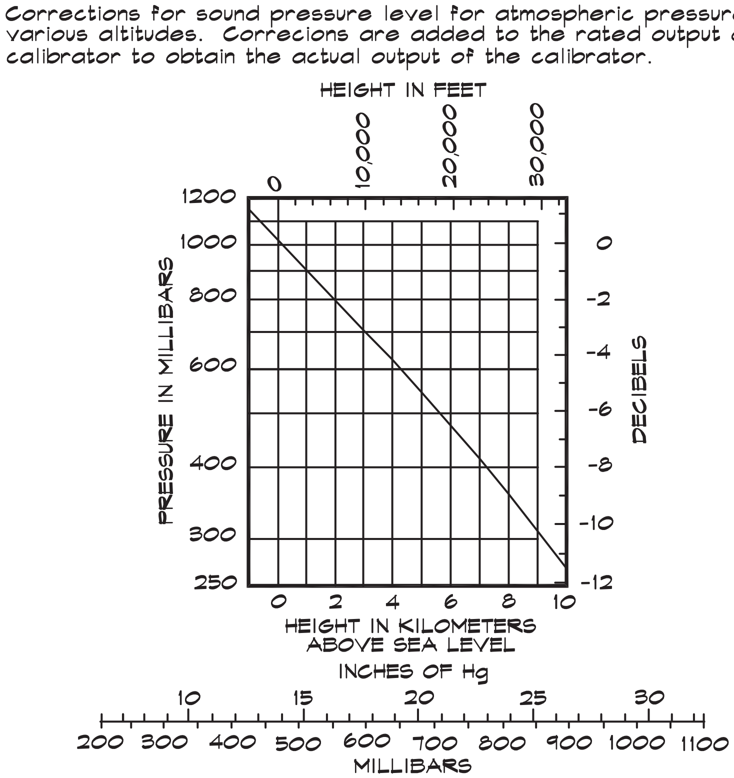
FIGURE 4.10 Sound Level Meter



of an oscillating piston in one end of a small cavity. The calibrator produces a nominal 94 dB, or with some calibrators a 114 dB, pure tone signal. The meter is adjusted to the proper level using a screw adjustment.

Pistonphone calibrators produce changes in volume in the cavity, which can be translated into changes in pressure using an equation of state. Most calibrators are set to produce the reference level at normal atmospheric pressure of 1013 millibars (1.01×10^5 Pa). Since atmospheric pressure varies, there is a correction given in Fig. 4.11 that must be applied according to altitude. This is the same correction as the term $10\ \log(\rho_0 c_0/400)$ in Eq. 2.67, including a density that changes with altitude.

FIGURE 4.11 Sound Level Meter Calibration Corrections (Peterson and Gross, 1974)



Calibrators themselves should be calibrated periodically against a microphone of known sensitivity. Since microphones are used to calibrate calibrators and vice versa, we encounter a classic chicken and egg conundrum; that is, how do we calibrated the original reference? The original microphone must be calibrated using another microphone in what is called a reciprocity calibration. The microphones used are identical and both transducers are used as loudspeakers and microphones in this technique. Refer to Kinsler et al. (1982) for further details.

Meter Ballistics

Early sound level meters were equipped with a d'Arsinval galvanometer, which responds to a voltage and indicates the sound level with a needle pointer. These early meters were very sensitive and tended to chatter or move back and forth rapidly. Electrical damping was added, which slowed the needle's response and made it more readable. The choice of the damping resistor in the indicator circuit, along with the capacitance of the microphone, set the exponential time constant of the circuit. Three response speeds are now used—*slow, fast*, and *impulse*. The slow setting has a time constant of 1000 ms (1 second), while for fast response it is 125 ms. A time constant has a precise mathematical meaning in engineering. In one time constant the value rises to $(1 - 1/e)$ or falls to $1/e = 1/2.718$ of its steady value. If a sound is instantaneously raised to a certain level the meter will rise to within 2 dB of the actual level in one time constant. Standard practice is to use 200 ms tone bursts at 1000 Hz to test a meter's response, since real sine waves have a finite rise time. The fast meter response must read within 2 dB of the steady level, and the slow meter response must be between 3 to 5 dB of the steady level (ANSI S4.1).

The rise time for fast and slow response is about the same as the fall time, so for integrating sound level meters, which measure L_{eq} levels, either fast or slow response gives about the same result. Some metrics, such as the CNEL level in California, require a particular response time, in this case, the slow response. For general use, the fast response is preferred. Impulse response is only employed to measure impact noise and other rapidly rising waveforms. The impulse time constant for a rising signal is 35 ms and for a falling signal is 1500 ms. Thus the meter holds the reading near its maximum level.

Meter Range

Sound level meters have an adjustable scale that allows the range of measurable levels to be set. If the range is set too low, then when a high level event occurs the meter will overload and not yield an accurate reading. If the range is too high, the indicated level will not fall below a certain value, and quiet events will not be measured accurately. Most meters have an overload indicator that signals the user to change the range. The range should be set as low as possible without tripping the overload indicator.

Detectors

There are two types of detector circuits found on most meters, peak, and rms (root mean square). Peak circuits sense the maximum amplitude present in the waveform. Mean-square detectors measure the time average of the square of the wave. Since the energy in the wave is proportional to the mean-square value, the rms detector is the most commonly used setting. Peak amplitudes are often of interest in vibration measurements. Peak-hold circuits, which capture the highest level during the measurement period, are utilized in the measurement of special sources such as sonic booms, where the wave shapes are not sinusoidal.

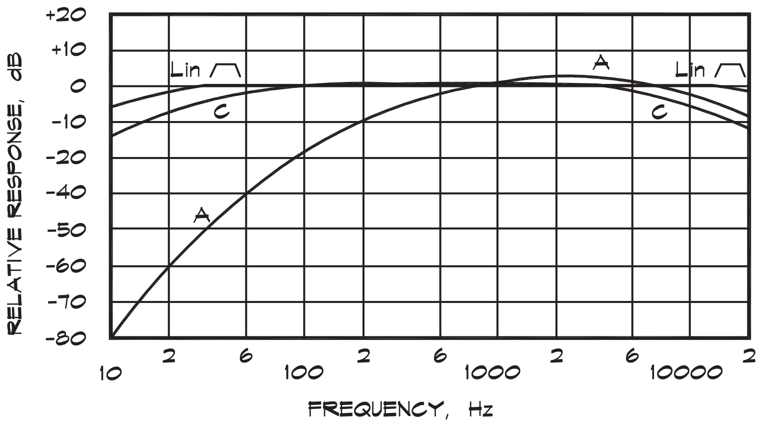
Filters

Sound meters come equipped with various selectable filters. The simplest is the linear filter, which passes sounds within the overall band limits of the instrument, for example 5 Hz to 100 kHz. This is not of particular interest in architectural acoustics, since it includes sounds that are well beyond our hearing capability. A second selection, the band-limited linear setting, includes a bandpass filter between 20 Hz and 20 kHz, and is quite useful for recording, since it blocks out low-frequency sounds that would otherwise overload a tape recorder. The characteristics of this filter along with the A and C weighting networks are shown in Fig. 4.12. Octave and third-octave bandwidth filters are also available. The standard frequency ranges have been given in Table 2.1. Filters may be cascaded, for example both octave band and A-weighting may be applied, yielding an A-weighted octave-band level. It is preferable to use the linear or band-limited linear settings when narrow-band filtering is done. This yields a consistent measurement methodology that does not require undue bookkeeping.

4.3 FIELD MEASUREMENTS

Field measurements are a critical part of architectural and environmental acoustics. Even with the simplest sources, care must be taken to follow proper procedure. A meter appropriate to the task must be selected. For environmental survey work a meter, tripod, calibrator, windscreen

FIGURE 4.12 A, C, and Lin Weighting Characteristics (Bruel and Kjaer, 1979)



(to reduce wind generated noise), logbook, distance measuring device (tape or rolling ruler), and watch are the standard kit. A small screwdriver is used to set the calibration. Spare batteries are a good idea. If they are left in the original packaging they can be distinguished from used ones. A camera is handy to record any unusual features of the site.

Headphones sometimes are included for listening to the sound being measured through the meter. They are essential for tape recording. Sometimes extraneous noise occurs that is not audible except through headphones. An example is arcing of the microphone, which can be caused by high humidity. Arcing produces a spurious popping sound that affects the data. Thus headphones are recommended when the relative humidity exceeds 90%.

For all measurements a record should be kept, noting the following information where it is relevant:

1) Location
2) Source description
3) Pertinent source details (e.g., manufacturer, model, operating point conditions)
4) Date and time
5) Engineer
6) Source dimensions and the radiating surfaces
7) Distance and direction to the source or a description of the measurement location
8) Meter settings
9) Background noise levels
10) Any unusual conditions
11) Time history
12) Measured data

Sources, which are outdoors and well away from reflecting surfaces, are the most straightforward. If the source is a piece of mechanical equipment the measurement position is selected based on the number of locations necessary to characterize the directivity of the source. For estimation of far-field levels from near-field measurements, data should be taken no closer than the largest dimension of the source, unless the area of the source is taken into account, by using Eq. 2.91.

The measurement distance for source characterization in a free field should be greater than a wavelength. For frequencies of 100 Hz the minimum distance is about 11 ft (3.4 m), while for 50 Hz the distance is about 22 ft (1.7 m). The danger of taking measurements too close is the possibility of including energy from only a portion of the source. If the source includes several separate pieces of equipment, the overall level will not be accurately represented if measurements are made too close to one individual component. Sometimes sound waves close to a source are not planar or are nonpropagating. Low-frequency emissions from large transformers are a good example of this type. Often low-frequency measurements require multiple samples and the microphone locations should be at least $\frac{\lambda}{4}$ apart.

Some sources are simply too large to conveniently get away from them. A good example is a refinery or a power plant. In such cases noise levels should be taken at regular distance intervals around the source and the results logged, according to where they were taken. Measurement locations should be spaced so that there is no more than a few decibels difference from one location to the next.

Measurements that are made to characterize a source rather than a location should be taken well away from reflecting surfaces. A minimum distance of $\frac{\lambda}{4}$ is recommended. If octave-band measurements are being taken and the 63 Hz band is of interest, then a distance of 4 to 5 feet is appropriate. Measurements will include reflections from the ground or other reflecting surfaces. Reflections from the observer can cause high-frequency comb filtering (Fig. 4.13), so the common practice is to hold the meter so that the microphone is extended away from the body or to support the instrument on a tripod.

An accurate measurement for source characterization is also difficult if the source receiver distance is too great. Even if the line-of-sight path is unimpeded, wind, atmospheric turbulence, ground cover, and air attenuation all play an important role in determining the measured noise level given off by a fixed source. At distances greater than 60 m (200 ft), noise level measurements can be dependent on wind velocity and direction. At distances greater than 150 m (500 ft), sound levels can be greatly influenced, even on a calm day, by ground cover, atmospheric turbulence, and air attenuation. At greater distances, thermal inversion layers can also be a major contributor. For all these reasons it is difficult to perform characterization measurements at large distances (say $>$ 60 m or 200 ft) from the source. Such measurements may be representative of a noise environment at a particular location under the measurement conditions, but may not be sufficiently accurate to characterize the source.

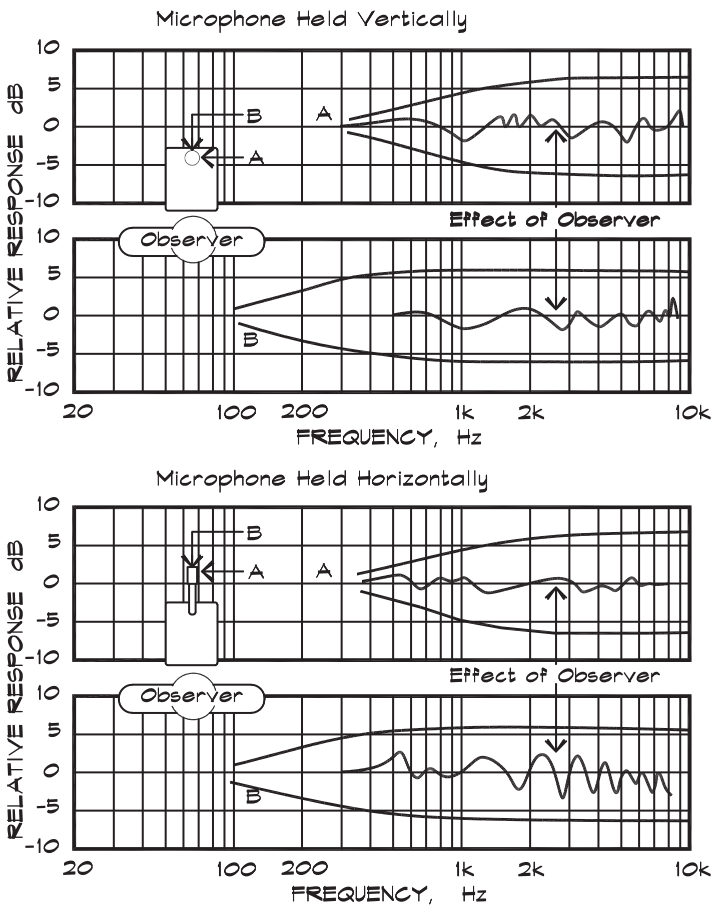
Background Noise

If there is a significant background (ambient) level present it should be measured. For a steady background it is best to turn off the source to be measured and note the ambient separately in all frequency bands of interest. The actual source-generated level then can be calculated from

$$L_{Source} = 10 \log \left[10^{0.1 L_{Tot}} - 10^{0.1 L_{Amb}} \right] \tag{4.1}$$

where L_{Source} = source sound pressure level (dB)
L_{Tot} = total combined source + ambient sound pressure level (dB)
L_{Amb} = ambient sound pressure level (dB)

FIGURE 4.13 Effect on Frequency Response as a Result of the Microphone Position (Petersen and Gross, 1974)



If the sound source cannot be turned off, it may be possible to measure the ambient noise level at a location that is similar to the location of interest but is away from the influence of the source. Locations may be available in shielded areas or, if the ambient noise is due to a roadway, at another site that is the same distance from the roadway.

When the background noise is variable and the source is steady, it is often easiest to measure the minimum combined level at a time when the ambient is quiescent. This gives an accurate source level if the ambient is sufficiently low. When the ambient is quiet, usually 10 dB below the source, its contribution can be ignored. With a varying ambient, if the source can be turned off, the minimum ambient can be recorded and then the minimum combined level measured. This gives a good value for the source level after adjustment using Eq. 4.1 as long as the minimum ambient levels are repeatable.

If the ambient is relatively steady and close to the source level, it can be measured separately using an averaging meter on the L_{eq} setting. The combined level then is measured in the same way and the source level calculated as before. This technique is also useful if the source, or background level, varies periodically, as it might with a pump motor or multiple sources such as fans or pumps, which produce beats. In taking data of this type, it is important to average over several beat cycles so that variations are properly taken into account.

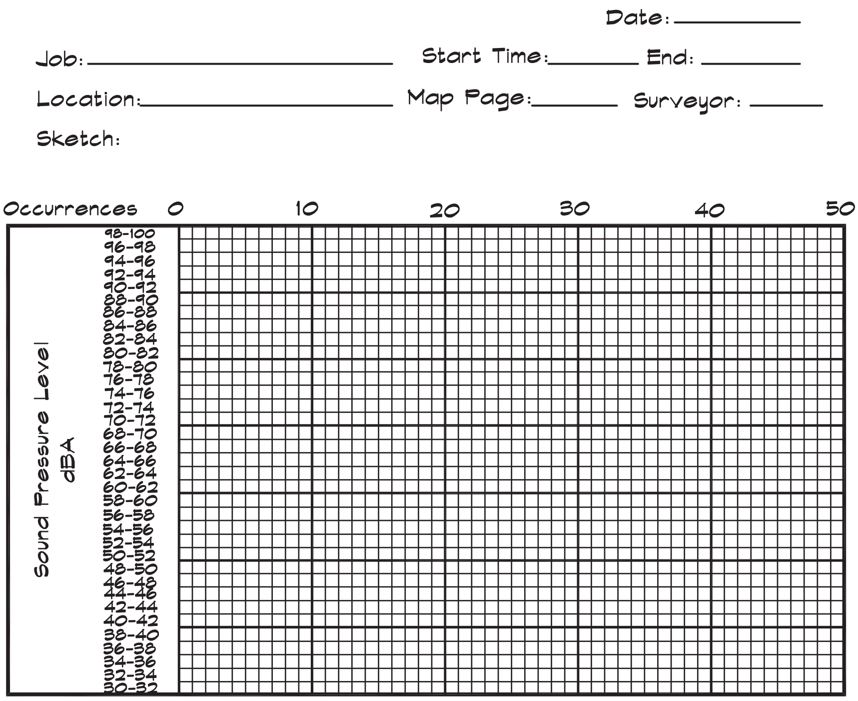
When the source level is less than the ambient, accurate measurements are difficult unless both the source and ambient levels are very steady. Even in these cases long averaging times are required to get good results. If the source is steady and the ambient varies, the minimum level gives the most accurate source level.

Time-Varying Sources

When traffic or other time-varying sources are to be measured, certain additional steps are useful. Although integrating meters are highly accurate, the nature of their output (i.e., one number) is sometimes not ideal, particularly when the data must be presented to a nontechnical audience. In these cases a log sheet such as that shown in Fig. 4.14 is helpful. In taking the data the meter is read at regular intervals, usually 5 or 10 seconds apart, and a notation is made on the log of the level that the meter shows at the interval mark. A representative number of samples are taken as determined either by the metric or the time period. One advantage to this methodology lies in the ability of the user to analyze the sampled data and extract more than one metric from the record. It also allows the engineer to ignore spurious signals such as barking dogs or aircraft flyovers that may not be relevant to the data being collected. Recording data, either on tape or in a recording sound level meter for later analysis, is another way of accomplishing the same goal. Data can be regularly sampled, and average levels calculated over a fixed time period and saved internally on a storage device for later analysis.

When a single moving source is to be measured, data are taken at a standard distance, say 15 m (50 ft), under prescribed conditions of velocity or acceleration. Data may be analyzed internally within the meter, or captured on a digital or analog recording device, or displayed

FIGURE 4.14 Noise Survey Log



graphically on a strip chart. When a recording is made, the calibration should flow through to all devices downstream of the meter. A tone is introduced using a pistonphone calibrator and is recorded along with the data. The meter range may then be adjusted by a known amount to accommodate the actual range of the data. A record should be made in a log or on the strip chart or verbally on tape noting the change in scale.

Both analog and digital recording devices can overload when signal levels exceed their dynamic range. When digital devices run out of headroom the resultant sound is most unpleasant. Analog tape recorders overload by producing a nonlinear or compressed version of the actual signal. If a two-channel device is available, the data may be recorded simultaneously on both channels at different level settings. This technique allows the data having the greater signal-to-noise ratio to be used, while retaining a margin of safety on the attenuated channel in case of overload.

Diurnal (24-Hour) Traffic Measurements

If a diurnal noise metric such as an L_{dn} or CNEL is to be measured, the ideal methodology is to position monitoring equipment at the location of interest for the entire 24-hour period. Often this is not practical due to the security, financial, or technical difficulties involved. In such cases a good estimate of the actual metric can be obtained by short-term monitoring if the hour by hour distribution of traffic is known or can be approximated. Measured distributions (Wyle, 1971) are given for urban traffic in Fig. 4.15 and for highway traffic in Fig. 4.16. The interesting feature about these data is that although they were taken 10 years apart they are almost identical. This implies that average diurnal traffic patterns are relatively stable.

If the reference L_{eq} level is known for the passage of one vehicle then the L_{eq} for N_h identical vehicles over the same time period is

$$L_{eq} = L_{ref} + 10 \log N_h \tag{4.2}$$

FIGURE 4.15 Typical Hourly Distribution of Total Daily Urban Vehicle Traffic (Wyle Laboratories, 1971)

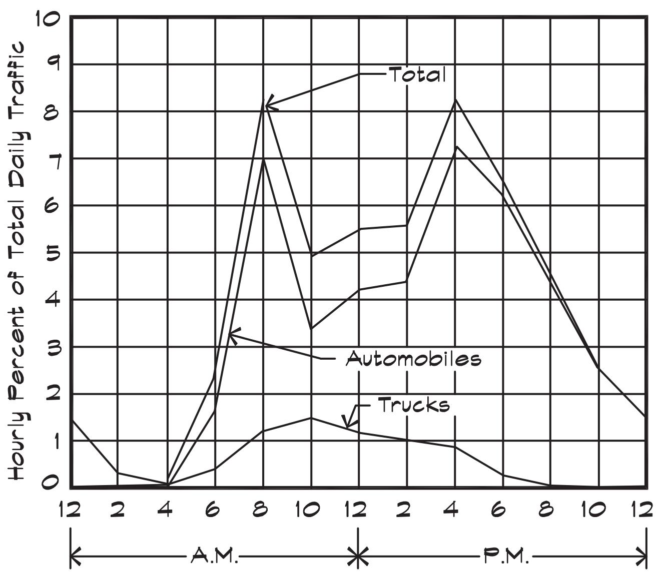
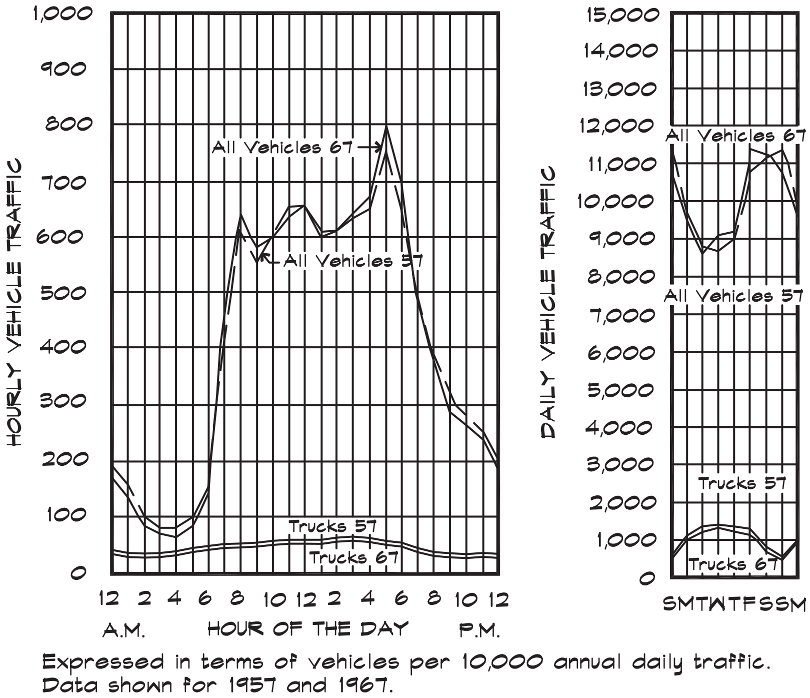


FIGURE 4.16 Hourly and Daily Variations in Intercity Highway Traffic in California (Wyle Laboratories, 1971)



where L_{eq} = equivalent sound level during the time period of interest (dBA)
L_{ref} = equivalent sound level for one vehicle passage during the time of interest (dBA)
N_h = number of like vehicles passing the measurement point during the time period of interest (usually one hour)

Assume that we can obtain the L_{eq} level for a given hour by direct measurement at a site. This can be accomplished by measuring over an hour period or by sampling the noise over a shorter time period and by assuming that the sample is representative of the hour period. Once the data have been obtained for the known hour, they can be adjusted for the time of day in which they were measured using standard distributions such as those in Figs. 4.15 and 4.16 or the actual site-specific traffic distribution, if it is known.

A traffic calculation uses a weighted hourly number of vehicles passing a point that yields the L_{dn} or CNEL level if inserted into Eq. 4.2. Thus

$$L_{dn} = L_{ref} + 10 \log N_{dn\,ave} \tag{4.3}$$

where L_{dn} = day night noise level (dBA)
L_{ref} = equivalent sound level for one vehicle passing by during an hour period (dBA)
$N_{dn\,ave}$ = weighted average number of like vehicles passing the measurement point during an equivalent hour

The day-night average number can be calculated from the distributions for urban and highway conditions using Eq. 3.7 for L_{dn} or Eq. 3.8 for CNEL.

$$N_{dn\,ave} = \left\{ \frac{1}{24} \left[\sum_{i=8}^{22} N_i + (10) \sum_{i=23}^{7} N_i \right] \right\} \tag{4.4}$$

where N_i = number of vehicles passing the measurement point during the i th hour

Finally by subtracting Eq. 4.2 and 4.3 we can obtain the difference in decibels between an L_{eq} level in any particular hour and the day-night level over a 24-hour period for a known traffic distribution.

$$L_{dn} \cong L_{eq}(h) + C(h) \tag{4.5}$$

where L_{dn} = day - night noise level (dBA)

$L_{eq}(h)$ = equivalent sound level for a given hour, h (dBA)

$C(h) = 10 \log \frac{N_h}{N_{dn\,ave}}$

= correction (dB) for the hour, h, based on the appropriate traffic distribution

The result is given in Table 4.2 for the Wyle urban and highway distributions for L_{dn}. The CNEL for these distributions is about 0.5 dB higher. If the traffic pattern at a particular site differs from those given here and is known, a similar calculation can be done for the specific distribution.

Included in these approximations is the assumption that the traffic speed and other factors that affect traffic noise, such as truck percentage, remain nearly the same over a 24-hour period. On crowded city streets this may not be the case. If traffic is free-flowing during the measurement period this method gives a conservative (high) estimate of the L_{dn} level. If traffic is slowed due to congestion, the noise levels will not be representative of a free-flowing condition.

If readings are taken during congested periods, the method will underestimate the actual 24-hour levels. If traffic slows significantly during rush hour, measurements made during off-peak periods, when traffic is flowing freely, will yield a result that is somewhat higher than the actual L_{dn} value.

The distribution of truck traffic over the day does not exactly track the automobile distribution. A similar calculation can be undertaken that includes truck percentages, with a knowledge of the difference between the reference level for trucks and cars. Naturally this introduces additional complexity. Based on 24-hour measurements, the method has been found to yield levels within one or two dB of the actual values, even without inclusion of a separate truck percentage distribution.

4.4 BROADBAND NOISE METRICS

At first glance the number and variety of acoustic metrics is overwhelming. In no other science are there as many different fundamental ways of measuring and characterizing the

TABLE 4.2 **Approximate Conversion from Leq to Ldn or CNEL (Based on the traffic distributions shown in Figs. 4.15 and 4.16)**

Hour	Highway Distribution		Urban Vehicle Distribution	
	$CNEL - L_{eq}$ (dB)	$L_{dn} - L_{eq}$ (dB)	$CNEL - L_{eq}$ (dB)	$L_{dn} - L_{eq}$ (dB)
1	8.2	7.7	10.9	10.4
2	10.4	9.9	15.1	14.6
3	11.2	10.7	16.9	16.4
4	11.6	11.1	19.9	19.4
5	10.6	10.1	9.5	9.0
6	8.2	7.7	6.5	6.0
7	3.6	3.1	2.6	2.1
8	1.6	1.1	0.8	0.3
9	2.0	1.5	1.7	1.2
10	1.9	1.4	2.9	2.4
11	1.6	1.1	2.7	2.3
12	1.5	1.0	2.6	2.2
13	1.9	1.4	2.5	2.1
14	1.8	1.3	2.4	1.9
15	1.6	1.1	1.5	1.0
16	1.4	0.9	0.8	0.3
17	0.7	0.2	1.3	0.8
18	1.3	0.8	1.8	1.3
19	2.7	2.2	2.7	2.2
20	4.0	3.5	3.6	3.1
21	5.0	4.5	4.3	3.7
22	5.3	4.8	5.6	5.1
23	5.9	5.4	6.9	6.4
24	7.1	6.6	7.8	7.4

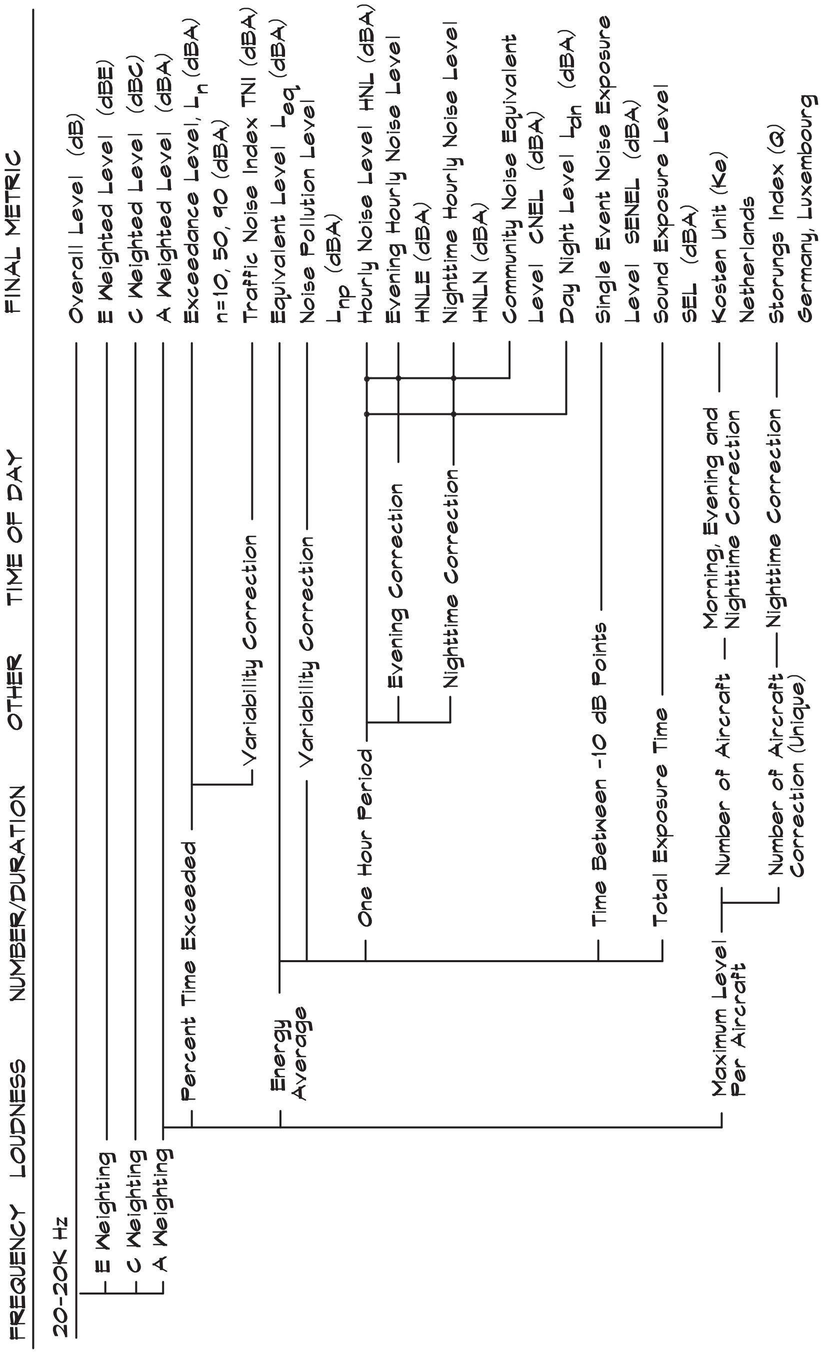
basic parameters. In physics the kilogram, meter, and second do not change. In electronics the volt, ohm, and ampere are stable and well defined. In environmental acoustics, however, different countries, states, cities, and counties often use different measurement schemes, which may not be directly convertible from one to another.

Even though the absolute number of metrics is large, the number of types of corrections applied to the measured data is rather modest. For example, a frequency correction for the loudness of a sound is included in most sound metrics but there are a number of ways to account for it, including A-weighting, NC curves, noys, and so on. The fundamental types of corrections include bandwidth, loudness, source number or duration, time of day, variability, onset, and pure tone content. The way each is included in a particular metric varies, but several are usually included in some fashion.

Bandwidth Corrections

The first correction category is the bandwidth of the measurement. Generally this is either wide band (i.e., 20 to 20 kHz) or band limited to octave or third-octave bandwidths. Narrow-band or chirped (swept) filters are also employed but the other corrections are seldom

FIGURE 4.17 Broadband Noise Metrics



applied to these measurements. Several metrics based on wide-band measurements are shown in Fig. 4.17. The loudness corrections in these measurements are applied by means of electronic filters, such as the A-weighting network, which are included in the meter itself. Subsequent corrections can be applied internally by the meter or can be added by a separate calculation.

Duration Corrections

One of the earliest metrics for describing traffic generated noise was the L_{10} (pronounced *ell-ten*) level. An L_n level is defined as the A-weighted sound level exceeded n % of the time during the measurement period. The L_{10} level is close to the maximum level occurring during a time period and its use reflects the fact that the highest levels are the most annoying. L_{10} levels are measured by using a histogram sampling technique, either manually or internally within the meter. If a histogram of measurements is made and there are 100 total samples, the L_{10} level is determined by counting 10 (10% of the total) measurements down from the highest level.

In a similar fashion the other excedance levels of interest can be determined. The L_{50} level or median is sometimes used. The L_{90} level is frequently used to characterize the residual background. L_n levels are expressed as whole numbers. From the statistical distribution of noise levels that can be characterized as normally distributed, certain relationships can be developed relating excedance levels to L_{eq} levels. For example, the energy average level, expressed in terms of the mean value (Barry and Reagan, 1978), is

$$L_{eq} = L_{50} + 0.115\,\sigma^2 \tag{4.6}$$

where L_{eq} = equivalent sound level (dB)
L_{50} = mean value sound level (dB)
σ = standard deviation of the sound levels (dB)

For a normal distribution, the L_{50} level and the L_{10} level are related

$$L_{10} = L_{50} + 1.28\,\sigma \tag{4.7}$$

The relationship between L_{10} and L_{eq} can be obtained

$$L_{eq} = L_{10} - 1.28\,\sigma + 0.115\,\sigma^2 \tag{4.8}$$

where L_{eq} = equivalent sound level (dB)
L_{10} = sound level exceded 10% of the time (dB)
σ = standard deviation of the sound levels (dB)

The standard deviation of highway traffic noise is usually 2 to 5 dB, so the L_{10} level is higher than the L_{eq} level. For traffic noise, the L_{eq} level is about equal to the L_{20} level. Not all outdoor noise distributions are normal, so these equations should be used carefully as general estimates of the actual values.

Variability Corrections

Metrics have been developed that include a term for the variability of the sound, the theory being that the more variable the sound distribution, the more annoying it is. The noise pollution level is one of these and is used to characterize community noise impacts. It is defined as

$$L_{NP} = L_{eq} + 2.56\,\sigma \tag{4.9}$$

where L_{NP} = noise pollution level (dBA)
L_{eq} = equivalent sound level (dBA)
σ = standard deviation of the sound levels (dBA)

Note that the noise pollution level uses A-weighting.

The traffic noise index (TNI) is another metric that includes a term for the variability of the noise environment. In this metric the variability is characterized in terms of the difference between the L_{10} and the L_{90} levels. The traffic noise index is given by

$$TNI = 4(L_{10} - L_{90}) + L_{90} - 30(dBA) \tag{4.10}$$

where TNI = traffic noise index (dBA)
L_{10} = level exceeded 10% of the time (dBA)
L_{90} = level exceeded 90% of the time (dBA)

Both the noise pollution level and the traffic noise index were developed for use in characterizing traffic noise and are not as accurate in predicting human reaction to other environmental noise sources.

Sound Exposure Levels

Metrics that utilize the format of energy times time are called exposure levels and are expressed in decibels with a reference period time of one second. There is considerable usefulness in such metrics in that they contain all the energy that occurs during a given event packed into a period one second long. The sound exposure level (SEL) is one such metric and is defined as

$$SEL = 10 \log \left[\sum_{i=1}^{N} 10^{0.1 L_i} \right] \tag{4.11}$$

where SEL = sound exposure level (dBA)
L_i = sound level for a given one - second time period (dBA)
N = number of seconds during the measurement period

The SEL can be measured directly by many sound level meters. The meter can be set to display the SEL, which is internally computed, following the initiation of the measurement, by pushing the meter reset button. The L_{eq} can be calculated from the SEL for a given time period T

$$L_{eq} = SEL - 10 \log(T) \tag{4.12}$$

where SEL = sound exposure level (dBA)
L_{eq} = equivalent sound level for a given time period (dBA)
T = time (s)

When there are several events, the L_{eq} level can be calculated from the SEL levels for each event. The SEL levels are combined using Eq. 2.62 and the L_{eq} level is calculated using Eq. 4.12. If both the L_{eq} and the SEL are measured simultaneously, the measurement time period can also be calculated using Eq. 4.12.

Single Event Noise Exposure Level

The single event noise exposure level (SENEL) is similar to the SEL in that it sums the energy times the time associated with an event. Originally, it was developed to measure the noise energy of the flyby of a single aircraft. In such measurements it is sometimes difficult to tell when to begin and when to stop the readings. If the data are recorded on a strip chart or tape recorder it is unclear at what point on either side of the peak to stop adding up the energy. To short cut the process the SENEL was developed. This metric is the exposure level contained in the top 10 dB of a single event sound level record. The duration of the event in a SENEL is the time between the two points at which the level falls 10 dB below the maximum. Figure 4.18 shows the L_{eq} for a triangular sound pattern. The SEL or SENEL may be calculated from these L_{eq} levels by using an equation similar to 4.12, where the time period is equal to the pulse duration τ. Once the SENEL is known, the L_{eq} can be calculated for any period of time containing the event.

$$L_{eq} = \text{SENEL} - 10 \log(T) \tag{4.13}$$

where SENEL = single event noise exposure level (dBA)
L_{eq} = equivalent sound level for a given time period (dBA)
T = time period for which the L_{eq} is to be calculated (s)

Note that it is necessary that the time period T in both Eqs. 4.12 and 4.13 be equal to or greater than the time period over which the SEL or SENEL was measured; otherwise, the event is not accurately represented. As with SEL, if several events occur within a given time period, then the individual SENEL levels may be combined using Eq. 2.62. An equivalent level can be calculated using Eq. 4.13 from the combined SENEL level.

4.5 BAND LIMITED NOISE METRICS

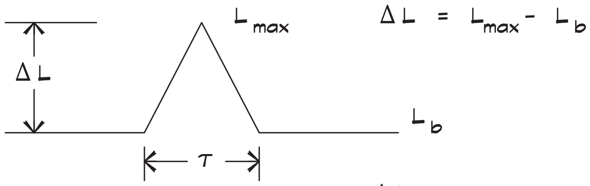
Techniques used for measurements employing octave band or other bandwidth filters vary little from those described for measuring broadband levels. Care must be taken in measuring low-frequency sounds so that the appropriate spacing between the source, reflecting surface, and the measurement location is observed. Sufficient sampling time is also a factor with low-frequency measurements because some low-frequency sources produce beat frequencies, which may be on the order of 1 Hz or less and may vary slowly over time.

Figure 4.19 shows a summary of the types of metrics obtained from octave-band measurements. As with the broadband systems there are a number of different metrics; however, the number of correction categories is relatively small. A loudness can be measured using electronic filters such as the A-weighting network. The A-weighted octave-band spectrum is useful as an aid in the determination of the frequency band making the most significant contribution to the overall A-weighted noise level. If most of the A-weighted energy is contained in one frequency band, then noise control efforts should be concentrated there.

A simple unweighted octave-band level is the basis for a number of metrics that determine the loudness by a direct comparison of the measured data to a standard curve of values. Several standards have been developed over the years, having to do principally with heating, ventilating, and air conditioning (HVAC) noise. The NC and RC curves are described in Chapt. 3.

FIGURE 4.18 L_{eq} Levels for Various Time Patterns (US EPA, 1973)

For triangular shaped patterns - time period = τ



$$L_{eq} = L_b + 10 \log \frac{10}{2.3\,\Delta L}\left(10^{\frac{\Delta L}{10}} - 1\right)$$

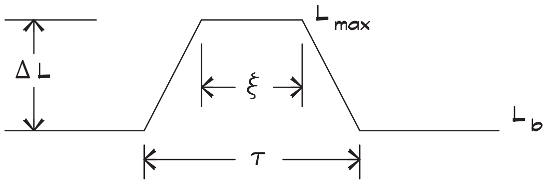
When ΔL is greater than 10 dB the following approximation is accurate.

$$L_{eq} = L_{max} - 10 \log \frac{2.3\,\Delta L}{10}$$

When there are a series of n identical triangular time patterns as shown above occurring within an interval T then

$$L_{eq} = L_b + 10 \log \left[1 + \frac{n\,\tau}{T}\left\{\frac{\left(10^{\frac{\Delta L}{10}} - 1\right)}{2.3} - \frac{\Delta L}{10}\right\}\right]$$

For trapezoidial shaped patterns - time period = τ



$$L_{eq} = 10 \log \frac{1}{\frac{(\tau - \xi)\,\Delta L}{10} + \frac{\xi}{2}}\left[10^{\frac{L_b}{10}}\,\frac{(\tau - \xi)}{2.3}\left(10^{\frac{\Delta L}{10}} - 1\right) + 10^{\frac{L_{max}}{10}}\,\frac{\xi}{2}\right]$$

When ΔL is greater than 10 dB and ξ is small compared to τ.

$$L_{eq} = L_{max} - 10 \log \frac{2.3\,\Delta L}{10} + 10 \log \xi$$

When there is a series of n identical trapezodial time patterns as shown above occurring within an interval T then

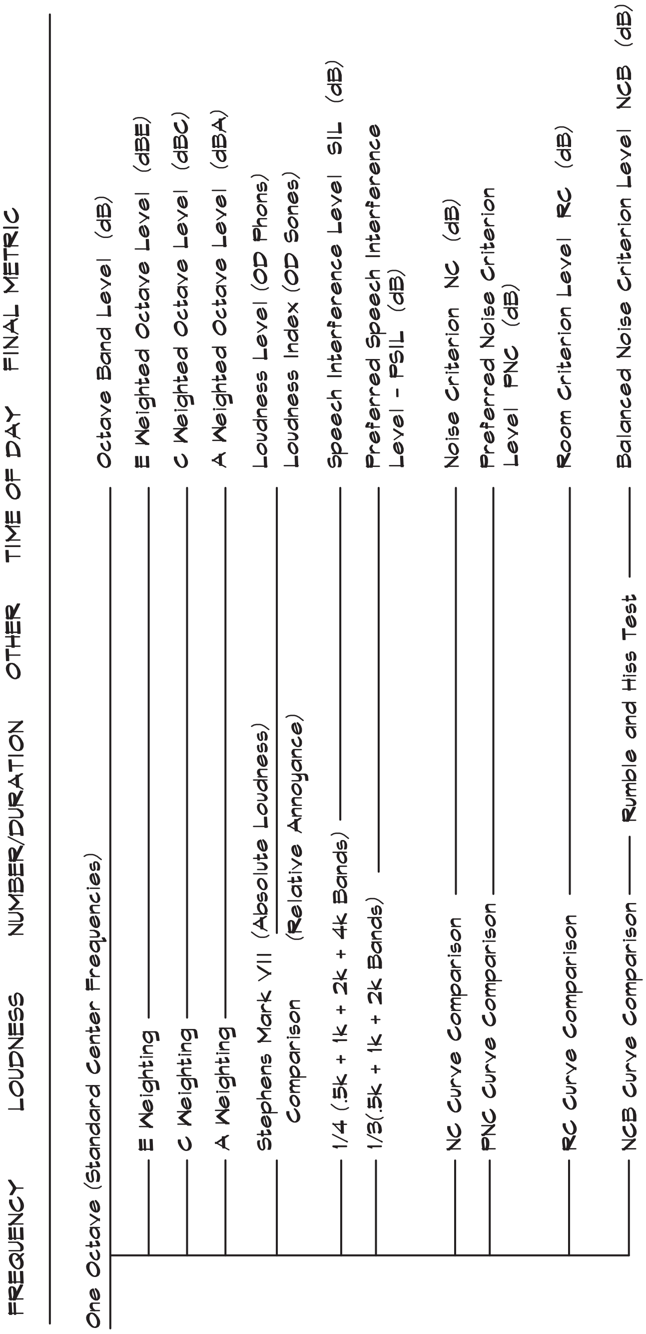
$$L_{eq} = L_{max} + 10 \log \frac{n\,\tau}{2.3\,T} + 10 \log n\,\xi$$

For a series of triangular pulses the last term above can be omitted.

Preferred Noise Criterion (PNC) Curves

PNC curves were introduced by Beranek in 1971 and are a revision of his earlier (1957) NC curves. PNC curves altered the high- and low-frequency octave values somewhat. The difference between the two has not been sufficient to result in the wide acceptance of the PNC version. The PNC curves are shown in Fig. 4.20. The use of the PNC curve is similar to that of the NC curve in that the PNC level is determined using the method of tangency.

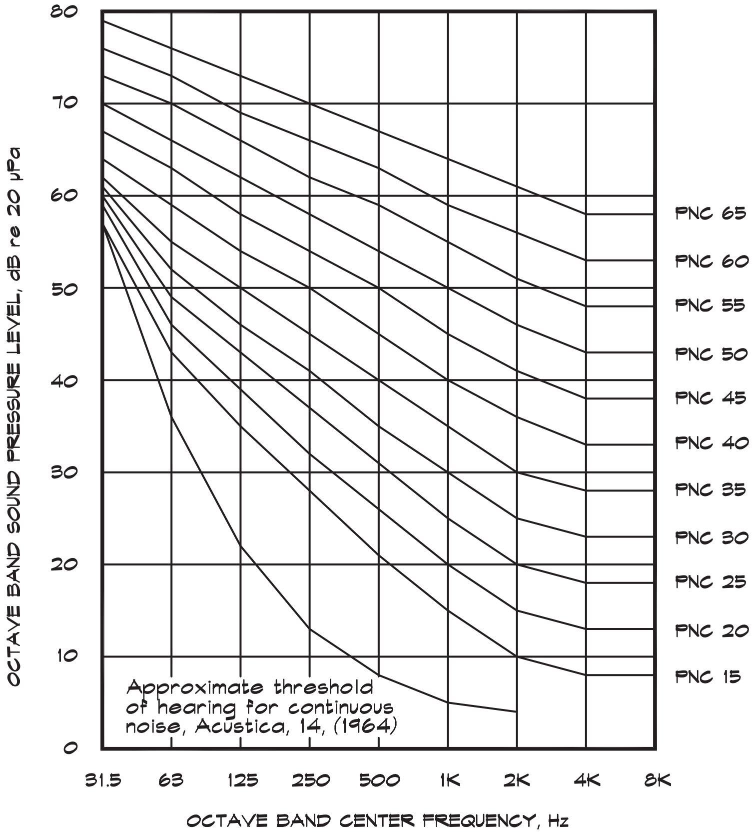
FIGURE 4.19 Octave Band Noise Metrics



Balanced Noise Criterion (NCB) Curves (Beranek, 1989)

In 1989, Beranek introduced another version of his 1957 NC curves, which he suggested for application to unoccupied rooms. These NCB levels, given in Fig. 4.21, are similar to the NC curves; however, the frequency range extends to 16 Hz and the metric calls for the calculation of the speech interference level (SIL) from the noise spectrum. This is rounded

FIGURE 4.20 Preferred Noise Criterion (Beranek, 1971)



to the nearest dB and compared with the NCB curve designation, which is also characterized by its SIL. If the measured SIL is equal to or below the curve designation, then the noise level meets the NCB criterion for speech interference.

Next comes a test for rumble or low-frequency energy. To check for this condition, 3 dB is added to the measured SIL and the NCB curve corresponding to this level is overlaid on the measured data. Where there are excedances of the new curve in the octave bands below 1 k Hz, they must be reduced to the elevated curve levels to comply with the standard.

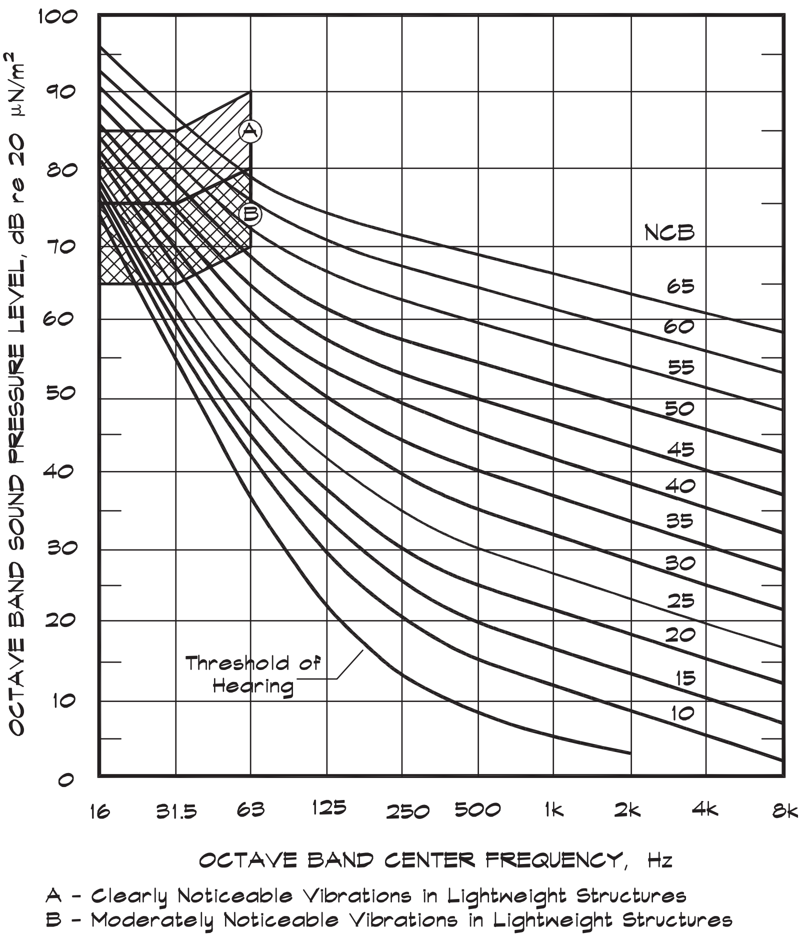
Finally there is the NCB test for hiss or high-frequency annoyance. An NCB curve is selected that provides the best fit to the measured data in the 125–500 Hz bands. Then this curve is plotted against the measured data. If the measured data exceed it in any of the three bands above 1 k Hz, then they must be reduced to meet the hiss criterion.

In occupied spaces Beranek calls for the measurement or estimation of noise levels due to normal work activities, which are to be combined with the unoccupied (HVAC) levels before a comparison to the NCB curve is made.

Other Octave-Band Metrics

Other systems exist for the determination of loudness based on measured octave-band data. They are based on empirical tests of relative or absolute comparisons presented to listeners in much the same way that the Fletcher-Munson experiments were done.

FIGURE 4.21 Balanced Noise Criterion Curves (NCB) (Beranek, 1989)



Robinson and Whittle (1962) constructed relative loudness curves in a very similar way. Stevens (1972) developed a series of systems for the calculation of loudness from octave band and other narrower bandwidth data. These systems rarely are encountered in architectural acoustics.

Octave-Band Calculations

It is frequently necessary to obtain an overall A-weighted level from unweighted octave-band data. The calculation is done by first adding the corrections for A-weighting, given in Table 3.1, to the level in each octave, and then by combining the A-weighted octave-band levels together using Eq. 2.62.

Occasionally it is necessary to generate an octave-band spectrum to match a given A-weighted level. This is straightforward if the spectrum shape of the sound source can be obtained. For example, let us assume that it is known that recorded music has a given octave-band spectrum and that this spectrum generates an overall A-weighted sound pressure level of 70 dBA. If we wish to obtain the octave-band spectrum of music that will yield an overall A-weighted level of some other level, for example 80 dBA, it is only necessary to add the difference between 80 and 70 to each octave-band level. It is assumed that the spectrum shape does not change with level for this source. It is useful to prepare normalized spectra for standard sources, which, when added to the overall A-weighted sound level, will yield an unweighted octave-band level having the same overall value.

If there are two sources present at the same time and we know the octave-band spectrum levels of each source independently, the spectrum for the two sources combined is obtained by applying Eq. 2.62 to the pairs of levels in each octave.

Third-Octave Bandwidth Metrics

Third-octave band metrics are similar to octave-band levels—they are simply a thinner slice of the same pie. They can be combined into groups of three centered around the octave-band center frequencies using Eq. 2.62 to obtain octave-band levels.

A summary of various third-octave and narrow band metrics is shown in Fig. 4.22. As with the octave-band metrics there are different versions of loudness and annoyance comparisons. One of these, the perceived noise level (PNdB) developed by Kryter (1970), has been used as the basis for several of the standard metrics for characterizing aircraft noise.

Aircraft Noise Rating Systems

Aircraft noise ratings vary principally in the methodologies they use for adjusting for the number of aircraft, the addition of pure tone corrections, and the inclusion of nighttime penalties. An excellent review of aircraft metrics was prepared by (Schuller et al., 1995). He summarizes the descriptors using the equation

$$\text{Level} = A \log \left(\sum_{\mathrm{i}=1}^{\mathrm{N}} \mathrm{n_i}\ \mathrm{w_i}\ 10^{\mathrm{L_i}/B} \right) - C \tag{4.14}$$

where A, B, C = constants
i = aircraft type category index
N = total number of aircraft type categories
$\mathrm{n_i}$ = number of noise events for aircraft category i per 24 - hour day
$\mathrm{w_i}$ = penalty (or weighting) factor for aircraft operation i
$\mathrm{L_i}$ = single event noise level for aircraft category i

The parameters used in Eq. 4.14 for various environmental metrics are given in Table 4.3.

Most of the metrics used for aircraft correlate well with the simpler $\mathrm{L_{dn}}$ level, which is the most commonly used system in the United States. For estimation purposes the following formulas may be used:

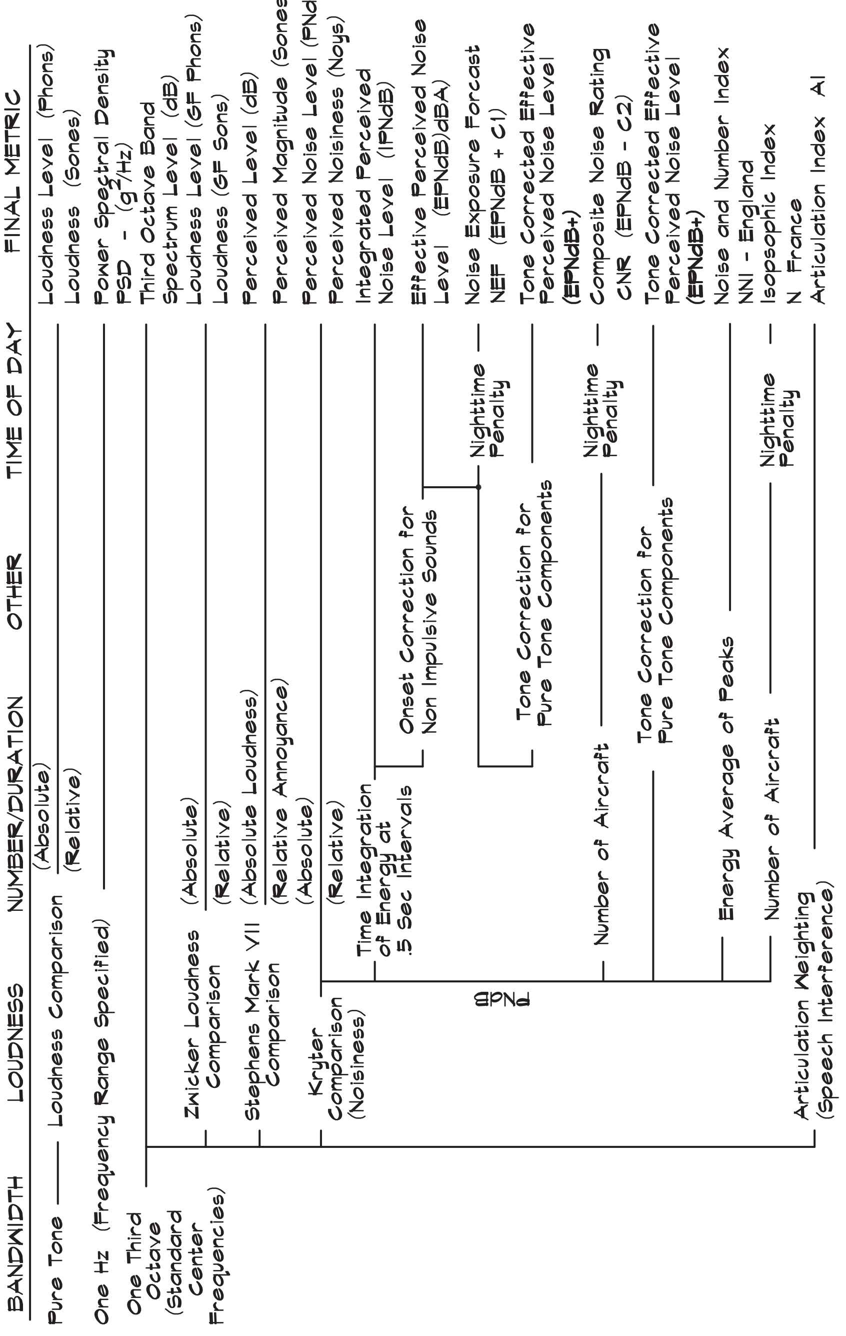
$$\mathrm{L_{dn}} \cong \mathrm{CNEL} \tag{4.15}$$

$$\mathrm{L_{dn}} \cong \mathrm{NEF} + 35(\pm 3) \tag{4.16}$$

$$\mathrm{L_{dn}} \cong \mathrm{CNR} - 35(\pm 3) \tag{4.17}$$

Similar relationships can be derived for the other metrics currently in use.

FIGURE 4.22 Narrow Band Noise Metrics



Narrow-Band Analysis

The analysis of sound in frequency bands of one-third octave and less is often useful for the detailed analysis of room acoustics and vibration. Instruments used for this type of measurement in real time are called spectrum analyzers or real-time analyzers (RTA). Two types of meters are most frequently encountered, those having a constant percentage bandwidth

TABLE 4.3 Parameters Used in Equation 4.14 (Schuller et al., 1995)

Metric	Constants A	B	C	Day Interval (Hours)	W_i	Morning, Evening Interval (Hours)	W_i	Night Interval (Hours)	W_i	L_i (dB)
Ke	20	15	105.8	08–18	1	06–08, 18–23	2–8	23–06	10	L_{ASmx}
L_n	10	10	44	06–23	0			23–06	1	L_{AE}
L_d	10	10	47.6[1]	07–23	1			23–07	0	L_{AE}
Q	13.3	13.3	65.7[2]	06–22	1			22–06	5[3]	L_{ASmx}
IP	10	10	49.4	07–22	1			22–07	10	L_{AE}
L_{dn}	10	10	44	06–23	0			23–06	1	L_{AE}
CNEL	10	10	44	07–19	1	19–22	3	22–07	10	L_{AE}
L_{24h}	10	10	44	00–24	1					L_{AE}
NNI	10	10	$80 - 5 \log \sum_{i=1}^{N} n_i w_i$	06–18	1			18–06	0	L_{pnmx}
NEF	10	10	88	07–22	1			22–07	16.7	L_{epn}

Single Event Noise Level Descriptors
L_{AE} = A-weighted sound exposure level
L_{ASmx} = Maximum S (slow) A-weighted sound level
L_{pnmx} = Maximum perceived noise level
L_{epn} = Effective perceived noise level

1) 7 hour night from 00.00 to 06.00 and 23.00 to 24.00 hours on a given day
2) 16 hour daytime period from 07.00 to 23.00 hours on a given day
3) Seperate calculations are specified for day and night. Values shown here are for calculations with emphasis on the contributions from nightime flight operations, Q_n. The weighting penalty includes a multiplication by the duration, in seconds, beween the first and last times that the instantaneous A-weighted sound level is within 10 dB of the maximum A-weighted sound level.

filter such as octave, third-octave, twelfth-octave, and so on, and those having a constant bandwidth such as 1 Hz. The latter type is used primarily in a laboratory while the former are the more common field instruments.

Instruments have filters of one of two types: analog and digital. An ideal bandpass filter is a device that passes all the electrical signals within its bandwidth and totally rejects all other signals. Analog filters are made of passive (resistors, inductors, and capacitors) or active (operational amplifiers) that approximate this ideal behavior. A meter having a group of such filters, operating in parallel, with each center frequency separated from the next by one-third octave, constitutes a real-time analyzer. These devices are robust and responsive. Some offer internal processing, such as energy averaging, and others feature only a freeze-and-save capability. Internal averaging is preferred since it is difficult to catch a varying signal at a point where all frequencies of interest are simultaneously at an average value.

A second type of system utilizes a mathematical filter, sometimes referred to as a digital filter. Digital filters can be constructed with the same characteristics as their analog counterparts. In these instruments the electrical signal is sampled periodically and the resulting string of numbers analyzed mathematically. One such procedure is the Fourier analysis (Joseph Fourier, 1768–1830), whereby a periodic signal is decomposed into its various harmonic components. Fourier's mathematical theorem states that any periodic waveform can be constructed from the sum of a specific sinusoidal wave called the fundamental, and a series of harmonics of the fundamental, multiplied by suitably selected constants. A graph of the amplitude versus frequency of these components is the spectrum of the original signal. A signal that has been digitally sampled can be sorted into its component frequencies using a mathematical process called the fast Fourier transform (FFT). Using similar techniques, filters can be constructed mathematically and applied to the digital number stream. The advantages of the digital filter are its flexibility, its low-frequency resolution, and its low cost. Disadvantages are its high-frequency limitations (eventually we cannot sample and calculate fast enough) and the features available on a given instrument.

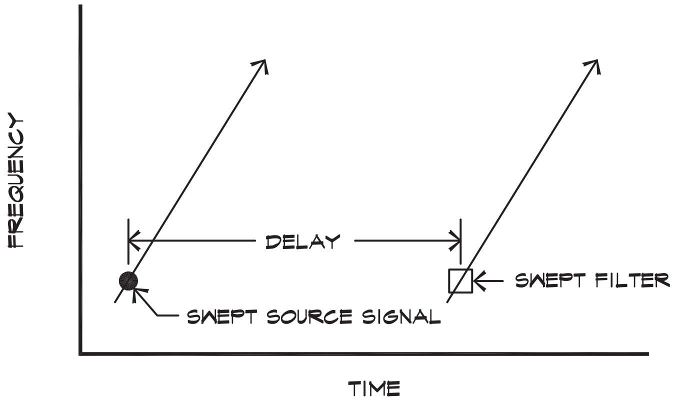
4.6 SPECIALIZED MEASUREMENT TECHNIQUES

Time-Delay Spectrometry

It is desirable to exclude noise intrusions from the measurements of a given signal. Anechoic chambers have been used for this purpose, since they are constructed not only to reduce sound from outside sources, but also to minimize the sound that is reflected off the walls and other surfaces of a room. A measurement technique originally developed by Richard Heyser (1931–1987) called time-delay spectrometry (TDS) can be used to make isolated measurements, even in a reverberant environment. The technique is based on the idea that when a sound source emits a signal it arrives at the receiving microphone after a given time. All the reflected sounds associated with the original signal arrive at some later time since they traveled along longer paths. If the measurement is made during a narrow time interval centered about the arrival time of the direct sound, later sounds are excluded and a nearly anechoic result can be achieved.

This is accomplished by converting the time delay into a frequency change. Figure 4.23 illustrates the principle. A loudspeaker is fed a sinusoidal signal that is chirped, or swept upward in frequency, at a fixed rate. At the receiver a narrow-band filter also is swept upward at the same rate. If the timing is correct the signal at a given frequency will arrive precisely when its filter window arrives. This technique is the same as that used by a quarterback

Figure 4.23 Time Delay Spectrometry



to throw a pass to a moving receiver. The ball (signal) and the receiver (filter window) must arrive at the same point at the same time for a reception. Passes that are delayed by reflections (off the ground or defensive linemen) do not arrive at the proper time and thus are not received.

Time-delay spectrometry can be used to measure the spectral response curve of a loudspeaker. The narrow-band analysis in Fig. 4.24 illustrates the detailed variations found in a typical loudspeaker. To obtain third-octave or octave-band data, the narrow-band energy data must be summed together over the appropriate frequency range. This process tends to smooth out the ripples in the curve and yields a more charitable portrait of the frequency response.

Energy-Time Curves

If a loudspeaker system is excited electronically with an impulsive signal, the signal received by a microphone can be plotted with time. This type of plot is called an energy-time curve (ETC) and contains useful information about the loudspeaker system as well as the room it is in. Turning first to loudspeakers, ETC plots are used to align transducers so that the signals from different components arrive at the listener at the same time. An example is shown in Fig. 4.25. Alignment is critical since a time delay is equivalent to a phase shift, which can produce a cancellation at the crossover frequency between transducers. Note that crossover points can be either electronic or spatial.

When two loudspeakers are misaligned, the ETC plot shows two distinct spikes. If this misalignment is sufficiently large, the result is a lack of clarity. When the two are aligned the overall level increases by 6 dB (due to an in-phase pressure doubling) and the peaks coincide in time. Loudspeaker alignment can be accomplished either by physical arrangement or by electronically delaying the signal transmitted to the forward transducer or both.

ETC plots can also reveal important information about reflections in rooms. A long-delayed reflection from the rear wall of a room, if sufficiently loud, can be disturbing to the perception of speech. Sometimes it is difficult in practice to identify the exact path that is causing the problem, particularly when multiple reflections are involved. An ETC plot can reveal the delay time of a given reflection and aid in the identification of the problem path. Patches of absorption can then be placed on the appropriate surfaces and the ETC measurements repeated for confirmation.

FIGURE 4.24 TDS Loudspeaker Measurements (Community, 1991)

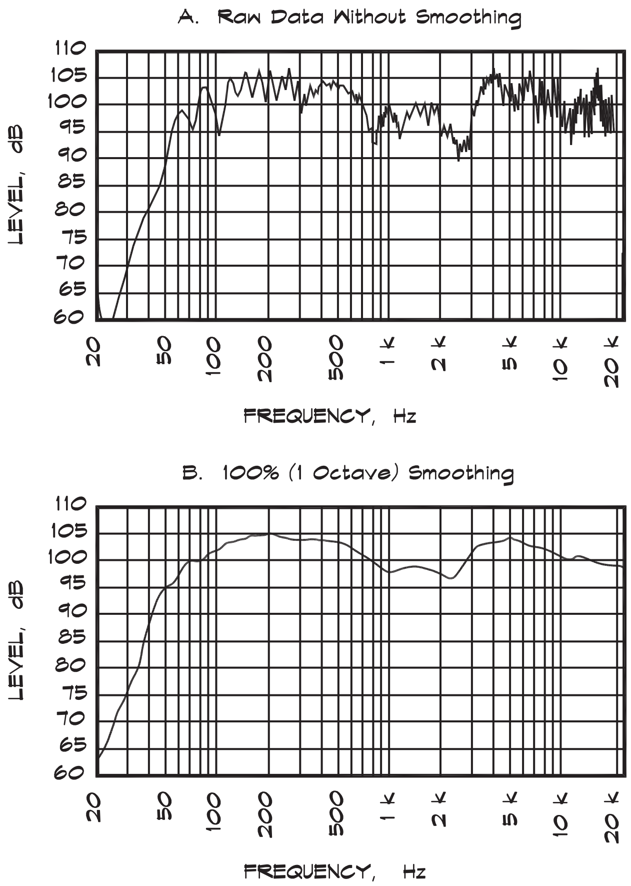
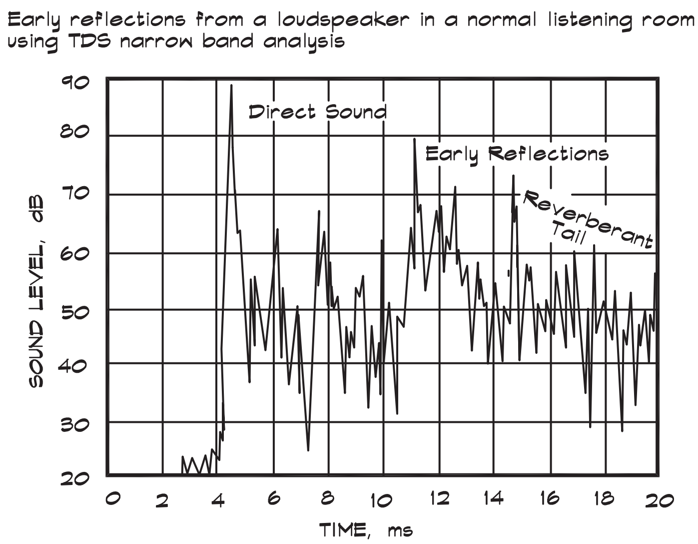


FIGURE 4.25 Energy Time Curves–ETC (Biering and Pedersen, 1983)



Sound Intensity Measurements

Direct measurement of the sound intensity has become possible through recent developments in commercial instrumentation. The intensity in a plane wave is defined as

$$I = p\,u \tag{4.18}$$

where I = maximum acoustic intensity (W / m^2)
p = acoustic pressure (Pa)
u = acoustic particle velocity (m / s)

The pressure is easily measured; however, direct measurement of the particle velocity is difficult. Instead the pressure can be measured using two closely spaced microphones, shown in Fig. 4.26, from which the change in pressure or pressure gradient can be obtained.

The reasoning is based on Newton's second law (F = m a) in one direction

$$\frac{d\,p}{d\,x} = -\rho_0 \frac{d\,u}{d\,t} \tag{4.19}$$

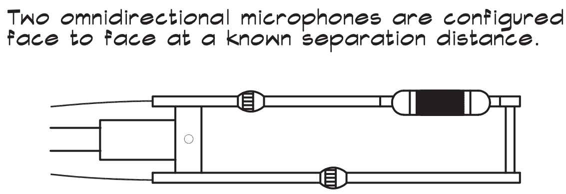
where ρ_0 = density of the bulk fluid (kg / m^3)
$d\,p$ = acoustic pressure change over a small distance $d\,x$ (Pa)
$d\,u$ = acoustic particle velocity change in time $d\,t$ (m / s)

The minus sign is there to indicate in which direction the slice accelerates. This equation is a well-known fluid dynamic relationship called Euler's equation. It relates the difference in pressure across a slice of fluid to an acceleration in the fluid slice that is proportional to its mass. The intensity is calculated by solving Eq. 4.19 for the particle velocity by integration

$$u = -\frac{1}{\rho_0\,\Delta\,x} \int (p_a - p_b) \tag{4.20}$$

where ρ_0 = density of the bulk fluid (kg / m^3)
p = acoustic pressure measured at two points a and b which are $\Delta\,x$ apart (Pa)
u = acoustic particle velocity (m / s)

FIGURE 4.26 Microphones Used in Intensity Measurements (Gade, 1982)



The intensity is then obtained by multiplying the pressure and the particle velocity.

$$I(\theta) = p\,u(\theta) = -\frac{p_a + p_b}{2\,\rho_0\,\Delta x}\int (p_a - p_b) \tag{4.21}$$

where $I(\theta)$ = acoustic intensity in a given direction (W/m^2)
p = acoustic pressure, which when measured at two points a and b that are Δx apart, is designated with a subscript (Pa)
$u(\theta)$ = acoustic particle velocity in a given direction (m/s)
ρ_0 = density of the bulk fluid (kg/m^3)

Since the intensity is a vector, its magnitude depends on the direction in which the microphones are oriented. Using this feature the intensity probe can be used for source location and strength.

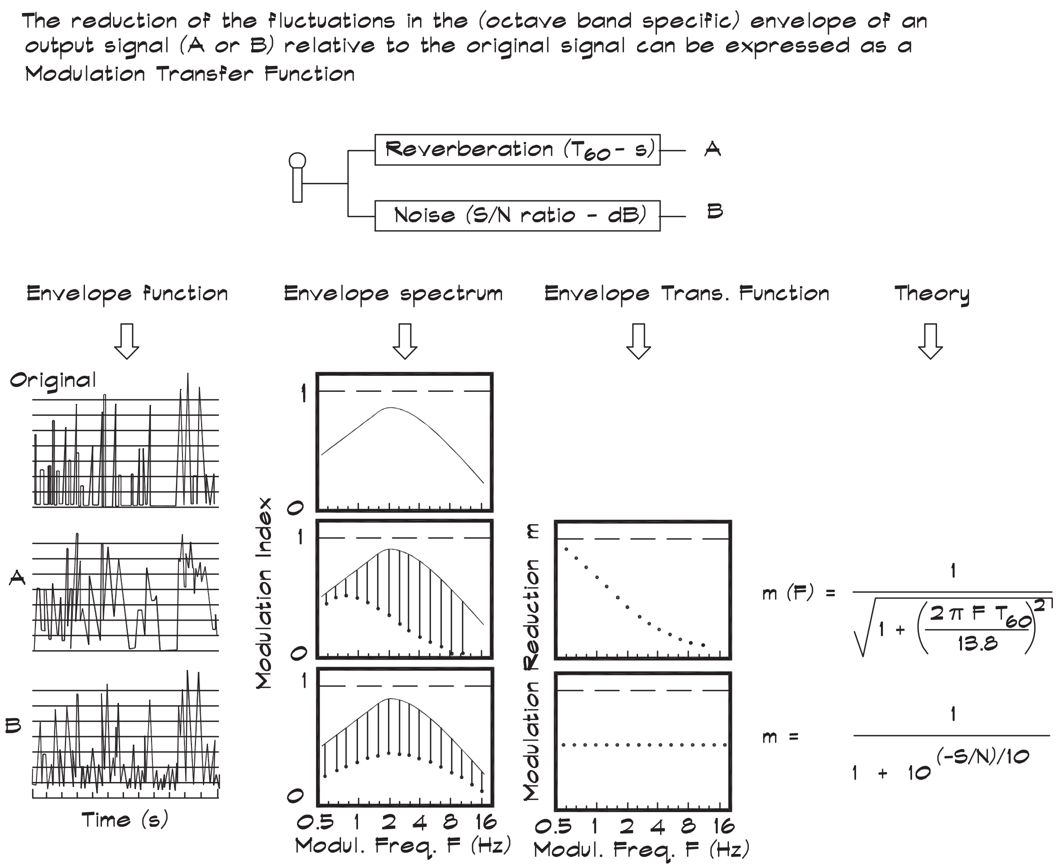
Modulation Transfer Function and RASTI

The intelligibility of speech has traditionally been measured by conducting tests, using various word lists, in rooms with human listeners. Although this methodology is the basis of most of our systems for predicting the intelligibility, it is highly desirable to have an electronic method of directly measuring these quantities. Human speech patterns are complex, and simple sinusoidal signals do not accurately mimic their behavior. Two Dutch scientists, Houtgast and Steeneken (1973), developed a measurement system, called the modulation transfer function (MTF), which replicates many of the properties of human speech. The concept is illustrated in Fig. 4.27. The idea behind MTF is that speech consists of modulated bands of noise. Our vocal cords vibrate to produce a band of noise, whereas our mouths modulate it at various frequencies to form words. To recreate this pattern, we start with an octave-wide band of noise and modulate it with a low-frequency tone. Mathematically this means that the carrier is multiplied by a sinusoidal function having a peak-to-peak amplitude of one. The result is a source signal that looks like the one on the left side of Fig. 4.28. For an accurate measurement the test signal level must be set to that of an average speaker and positioned where his mouth would be.

When this signal is transmitted to a listener, it is altered by the environment to some degree and can result in reduced speech intelligibility. The distortion mechanisms include background and reverberant noise, which raise the bottom of the signal above zero, and reflections, which add back a delayed and perhaps distorted copy of the signal. A typical receiver signal, shown on the right side of Fig. 4.28, is less modulated than the original, where the degree of modulation is defined by the depth of the modulation envelope. The reduction in modulation is characterized by a modulation reduction factor, $m(f_m)$, which is a function of the modulation frequency f_m. The modulation reduction factor varies from 0 for no reduction to 1 for 100% modulation reduction. Curves can be measured of the behavior of m versus f_m as shown on the bottom of Fig. 4.28.

When background noise is the principal source of the distortion, the effect on modulation reduction appears in terms of a signal-to-noise ratio, which is independent of modulation frequency. The noise raises all levels at the receiver within the carrier band and thus reduces modulation equally. When the distortion is produced by reverberation, the modulation reduction has the form of a low-pass filter with the faster fluctuations more sensitive to the effects of reverberation. This effect is characterized by the product of the modulation frequency and the

FIGURE 4.27 Basis of the Modulation Transfer Function (Houtgast and Steeneken, 1985)



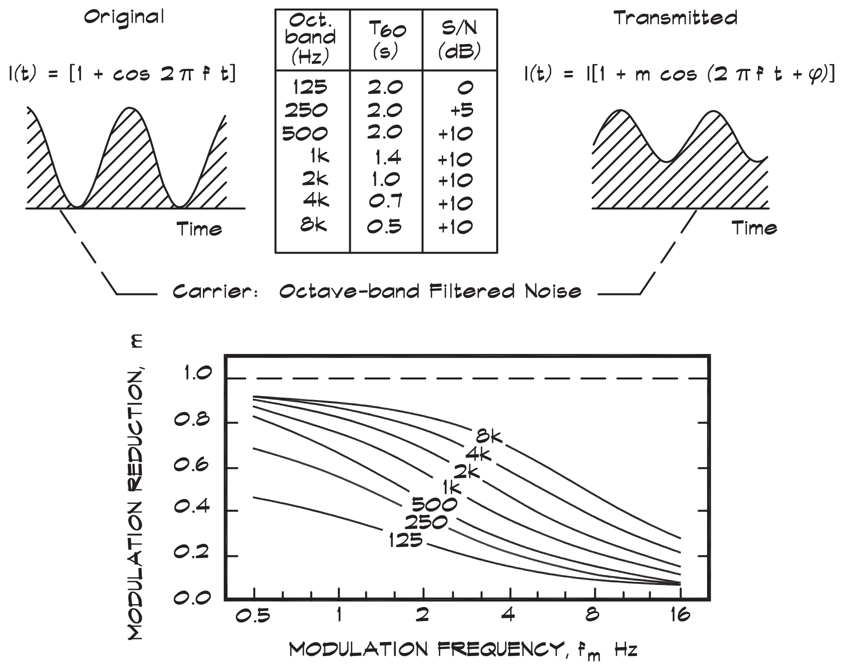
room reverberation time. The overall modulation reduction factor is given mathematically as the product of these two effects for an unamplified signal

$$m(f_m) = \frac{1}{\sqrt{1+\left[2\,\pi\, f_m \dfrac{T_{60}}{13.8}\right]^2}}\;\frac{1}{1+10^{\left(-0.1 L_{SN}\right)}} \tag{4.22}$$

where $m(f_m)$ = modulation reduction factor
L_{SN} = signal to noise level (dB)
f_m = modulation frequency (Hz)
T_{60} = room reverberation time (s)

The modulation frequency f_m ranges in value from 0.63 Hz to 12.5 Hz in third-octave intervals. The input-output analysis for a given system is done at 7 octave bands and 14 modulation frequencies, for a total of 98 separate values of m.

FIGURE 4.28 Modulation Transfer Function (Houtgast and Steeneken, 1985)



Oct. band (Hz)	T60 (s)	S/N (dB)
125	2.0	0
250	2.0	+5
500	2.0	+10
1k	1.4	+10
2k	1.0	+10
4k	0.7	+10
8k	0.5	+10

Illustration of how an MTF analysis is performed by using an octave-band filtered noise carrier, 100% intensity modulated, for each modulation frequency sucessively. This leads to a family of MTF curves. Each curve is calculated for the data given in the table.

Speech Transmission Index

With the MTF we have a quantity that mimics the behavior of speech, and can be physically measured with a properly constructed instrument. The missing link is the relationship between MTF and speech intelligibility. This is given in Fig. 4.29 by a speech transmission index (STI), which is similar to an articulation index or a percentage loss of consonants, in that it is a direct measure of speech intelligibility. All three are numerical schemes used to quantify the intelligibility of speech. Fig. 4.30 shows the relation between STI and Alcons, and Fig. 4.31 shows the similarity of STI to AI.

Steeneken and Houtgast (1980, 1985) developed an algorithm for transforming a set of m values into a speech transmission index (STI) by means of an apparent signal-to-noise ratio expressed as a level. This level is the signal-to-noise ratio that would have produced the modulation reduction factor, had all the distortion been caused by noise intrusion, irrespective of the actual cause of the distortion.

$$L_{SNapp} = 10 \log \frac{m}{1 - m} \tag{4.23}$$

where L_{SNapp} = apparent signal to noise ratio (dB)
m = modulation reduction factor

A weighted average of the 98 apparent signal-to-noise ratios yields the STI after applying a normalization such that

FIGURE 4.29 Typical Relations between the STI and Intelligibility Scores for the Various Types of Tests (Houtgast et al., 1985)

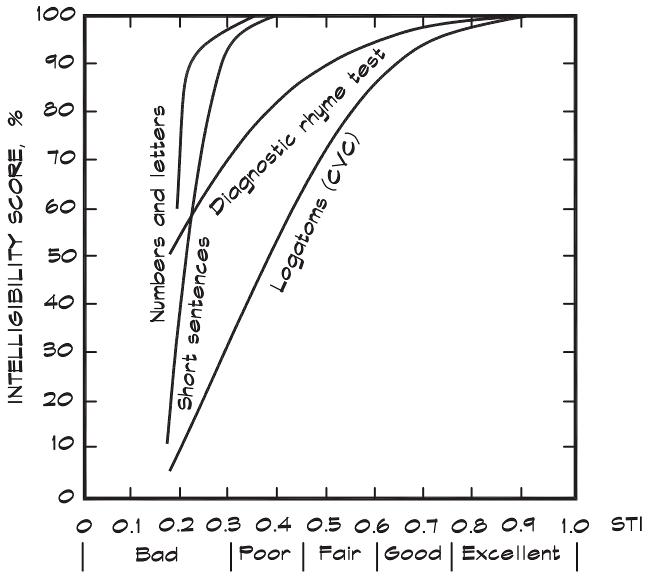


FIGURE 4.30 A Comparison of Articulation Index and Speech Transmission Index (Houtgast et al., 1980)

In this comparison the interference is only due to speech-shaped noise. The intelligibility rating is based on CVC (consonant, vowel, consonant) nonsense syllables embedded in a neutral carrier sentence.

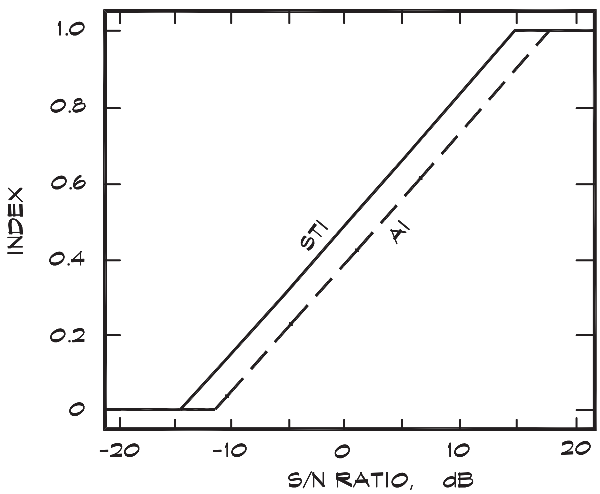
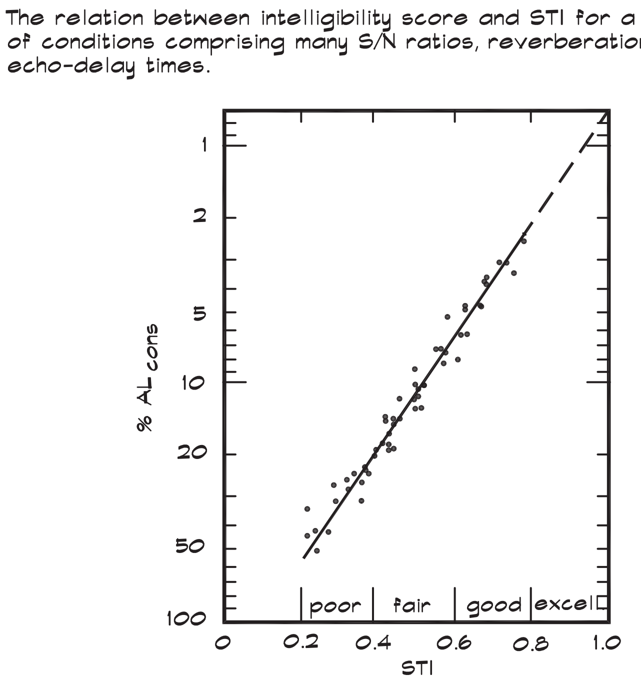


FIGURE 4.31 Relation between STI and Intelligibility Score (Houtgast et al., 1980)



STI = 1.0 when $L_{SNapp} \geq 15$ dB for all 98 data points

STI = 0.0 when $L_{SNapp} \leq -15$ dB for all 98 data points

and

$$\overline{L_{SNapp}} = \sum_{i=1}^{7} w_i \left(L_{SNapp}\right)_i \tag{4.24}$$

where $\overline{L_{SNapp}}$ = average apparent signal-to-noise ratio (dB)
w_i = weighting for octave bands from 125 Hz to 8 k Hz
= 0.13, 0.14, 0.11, 0.12, 0.19, 0.17, and 0.14

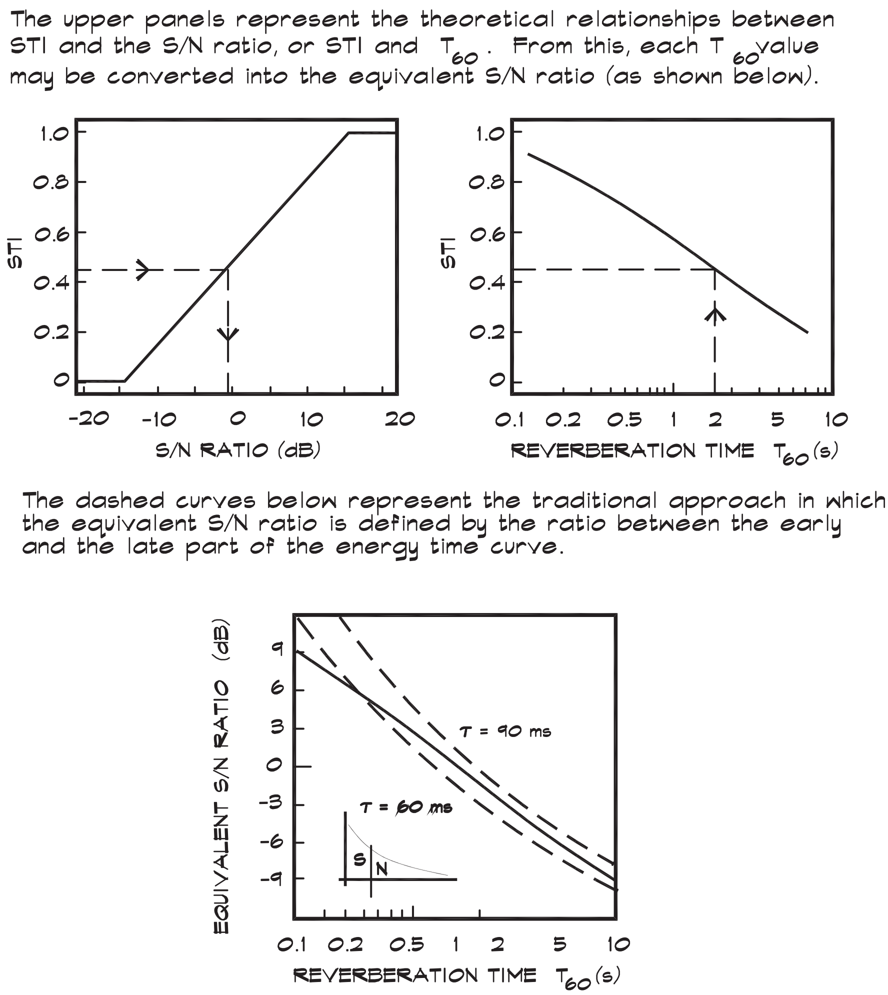
then

$$STI = \left[\overline{L_{SNapp}} + 15\right] / 30 \tag{4.25}$$

Figure 4.32 shows the relationship between STI and the signal-to-noise ratio as well as the reverberation time.

The bottom part of Fig. 4.32 represents an alternative way of interpreting the effect of reverberation. The early part of the reverberant tail is considered helpful to the understanding of speech, whereas the end is considered detrimental. The boundary between the two regions occurs in the neighborhood of 70 to 80 ms. Though it may seem that this is rather long, in that a single 65 ms delay can be detected as an echo, it should be remembered that in normal rooms the listener hears a series of reflections and thus the Haas region is extended somewhat (Fig. 3.29).

FIGURE 4.32 Relationship between Modulation Transfer Function and the Speech Transmission Index (Houtgast and Steeneken, 1985)

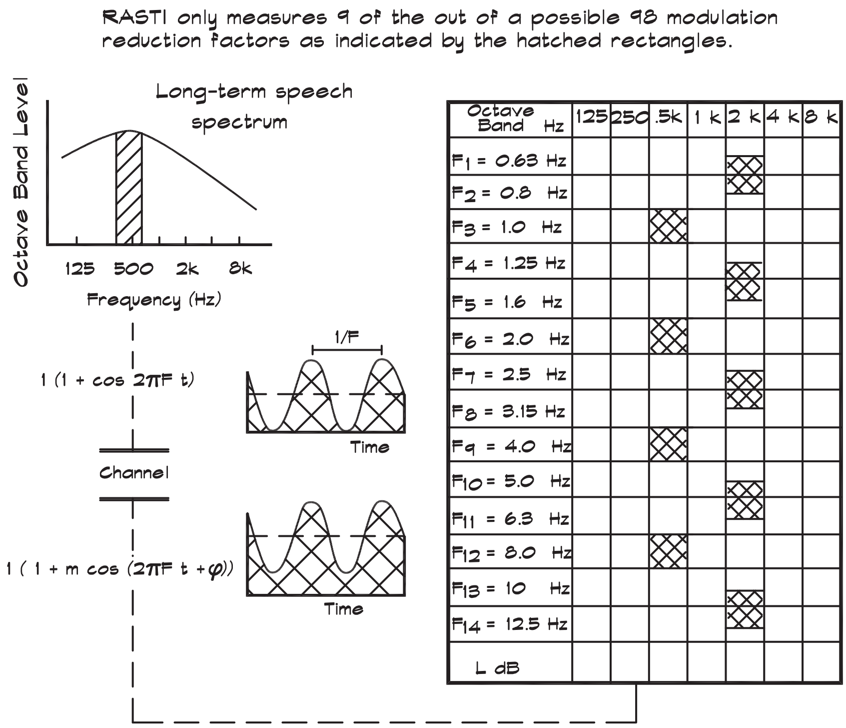


The research done by Houtgast and Steeneken established a way of measuring speech and intelligibility using an electronically generated test signal rather than a group of human subjects. Their calculation method is useful in evaluating rooms for an omnidirectional source, but does not include consideration of loudspeaker directivity, so necessary to the design of reinforcement systems. Once the method has been established as equivalent to other measures of intelligibility without amplification, the measurement system can be used to evaluate installed sound systems.

RASTI

RASTI or RApid STI is an approximation of the full STI taken by doing a measurement of nine of the 98 m values marked on the graph shown in Fig. 4.33. Two octave bands 500 and 2000 Hz are sampled. At 500 Hz, values of m are measured at four modulation frequencies, 1, 2, 4, and 8 Hz. At 2000 Hz, five modulation frequencies are measured, 0.7, 1.4, 2.8, 5.6, and 11.2 Hz. An apparent signal-to-noise ratio is calculated from the measured m values in each band and truncated so as to fall within the range of ±15 dB. The L_{SNapp} values are

Figure 4.33 The RASTI Analysis System (Houtgast and Steeneken, 1985)



averaged and a RASTI value is calculated

$$\text{RASTI} = \left[\overline{\text{L}_{\text{SNapp}}} + 15\right] / 30 \tag{4.26}$$

where RASTI = rapid STI measurement index
$\overline{\text{L}_{\text{SNapp}}}$ = average apparent signal to noise ratio (dB)

In practice RASTI measurements can be made to evaluate the intelligibility of speech both for an unamplified talker as well as for an amplified sound system. The RASTI source is positioned at the talker location. If there is a microphone, the source is set in front of it so that the public address system can be tested. The receiver microphone is located at various points throughout an auditorium to determine the RASTI rating.

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