APPENDIX

# D Glossary

This appendix contains technical definitions of key acoustical and vibration terms commonly used with Larson Davis instruments. The reader is referred to American National Standards Institute document S1.1-1994 for additional definitions. Specific use of the terms defined are in the main body of the text.

**Allowed Exposure Time ($T_i$)**

It is the allowed time of exposure to sound of a constant A-weighted sound level given a chosen Criterion Level, Criterion Duration, and Exchange Rate. The equation for it is

$$T_i = \frac{T_c}{2^{(L_{avg} - L_c)/Q}} = \frac{T_c}{10^{(L_{avg} - L_c)/q}}$$

where $L_c$ is the Criterion Level, $T_c$ is the Criterion Duration, Q is the Exchange Rate, K is the Exchange Rate Factor and $L_{avg}$ is the Average Sound Level.

E*xample*: If $L_c$ = 90, $T_c$ = 8, Q = 3 and $L_i$ = 95 then

$$= \frac{8}{10^{(95-90)/10}} = \frac{8}{2^{(95-90)/3}} = 5.656 = \text{5 hours and 39 minu}$$

This means that if a person is in this area for 5 hours and 39 minutes he will have accumulated a Noise Dose of 100%.
*Standard*: ANSI S12.19

**Average Sound Level ($L_{avg}$)**

It is the logarithmic average of the sound during a Measurement Duration (specific time period), using the chosen Exchange Rate Factor. Exposure to this sound level over the period would result in the same noise dose and the actual (unsteady) sound levels. If the Measurement Duration is the same as the Criterion Duration, then $L_{avg}=L_{TWA(LC)}$

$$L_{avg} = qLog_{10}\left(\frac{1}{T}\int_{T_1}^{T_2} 10^{(L_p(t))/q}dt\right)$$

where the Measurement Duration (specified time period) is $T=T_2-T_1$ and q is the Exchange Rate Factor. Only sound levels above the Threshold Level are included in the integral. *Standard*: ANSI S12.19

**Community Noise Equivalent Level (CNEL, $L_{den}$)**

A rating of community noise exposure to all sources of sound that differentiates between daytime, evening and nighttime noise exposure. The equation for it is

$$L_{den} = 10\log_{10}\left\{\frac{1}{24}\left[\sum_{0000}^{0700} 10^{(L_i+10)/10} + \sum_{0700}^{1900} 10^{L_i/10} + \sum_{1900}^{2200} 10^{(L_i+5)/10} + \sum_{2200}^{2400} 10^{(L_i+10)/10}\right]\right\}$$

The continuous equivalent sound level is generally calculated on an hourly basis and is shown in the equation as L. The levels for the hourly periods from midnight to 7 a.m. have 10 added to them to represent less tolerance for noise during sleeping hours. The same occurs from 10 p.m. to midnight. The levels for the hourly periods between 7 p.m. and 10 p.m. have 5 added to them to represent a lessened tolerance for noise during evening activities. They are energy summed and converted to an average noise exposure rating.

**Criterion Duration ($T_c$)**

It is the time required for a constant sound level equal to the Criterion Level to produce a Noise Dose of 100%. Criterion Duration is typically 8 hours.
*Example*: If the Criterion Level = 90 dB and the Criterion Duration is 8 hours, then a sound level of 90 dB for 8 hours, will produce a 100% Noise Dose. See Noise Dose. *Standard*: ANSI S12.19

**Criterion Sound Exposure (CSE)**

The product of the Criterion Duration and the mean square sound pressure associated with the Criterion Sound Level when adjusted for the Exchange Rate. It is expressed in Pascals-squared seconds when the exchange rate is 3 dB. where q is the Exchange Rate Factor. See Exchange Rate.

$$CSE = T_c 10^{L_c/q}$$

*Standard*: ANSI S1.25

**Criterion Sound Level ($L_c$)**

It is the sound level which if continually applied for the Criterion Duration will produce a Noise Dose of 100%. The current OSHA Criterion Level is 90 dB.
*Standard*: ANSI S12.19

**Daily Personal Noise Exposure ($L_{EP,d}$)**

It is the level of a constant sound over the Criterion Duration that contains the same sound energy as the actual, unsteady sound over a specific period. The period is generally shorter, so the sound energy is spread out over the Criterion Duration period.
*Example*: If the Criterion Duration = 8 hours and the specific period is 4 hours and the average level during the 4 hours is 86 dB, then the $L_{EP,d}$ = 83 dB.

**Day-Night Average Sound Level (DNL, $L_{dn}$)**

A rating of community noise exposure to all sources of sound that differentiates between daytime and nighttime noise exposure. The equation for it is

$$L_{den} = 10\log_{10}\left\{\frac{1}{24}\left[\sum_{0000}^{0700} 10^{(L_i + 10)/10} + \sum_{0700}^{1900} 10^{L_i/10} + \sum_{1900}^{2200} 10^{(L_i + 5)/10} + \sum_{2200}^{2400} 10^{(L_i + 10)/10}\right]\right\}$$

$$L_{dn} = 10Log_{10}\left\{\frac{1}{24}\left[\sum_{0000}^{0700} 10^{(L_i + 10)/10} + \sum_{0700}^{2200} 10^{L_i/10} + \sum_{2200}^{2400} 10^{(L_i + 10)/10}\right]\right\}$$

The continuous equivalent sound level (See definition) is generally calculated on an hourly basis and is shown in the equation as L.

The values for the hourly periods from midnight to 7 a.m. have 10 added to them to represent less tolerance for noise during sleeping hours. The same occurs from 10 p.m. to midnight. They are energy summed and converted to an average noise exposure rating.

**Decibel (dB)**

A logarithmic form of any measured physical quantity and commonly used in the measurement of sound and vibration. Whenever the word *level* is used, this logarithmic form is implied. The decibel provides us with the possibility of representing a large span of signal levels in a simple manner as opposed to using the basic unit Pascal for e.g. acoustic measurements.

It is not possible to directly add or subtract physical quantities when expressed in decibel form since the addition of logarithmic values correspond to multiplication of the original quantity.

The word *level* is normally attached to a physical quantity when expressed in decibels; for example, *Lp* represents the sound pressure level.

The difference between the sound pressure for silence versus loud sounds is a factor of 1,000,000:1 or more, and it is very unpractical to use these large numbers. Therefore, a measure that would relate to “the number of zeros” would help, for example, 100,000 would be equal to 50 and 1000 would be equal to 30 and so on. This is the basic principal of the dB measure.

All dB values are unit free and therefore, the dB value is not the value of the quantity itself, but the ratio of that quantity to an actual reference quantity used. Thus, for every level in decibels there must be a well defined reference quantity. Sound versus vibration uses different references, but the dB principal is the same. When the quantity equals the reference quantity the level is zero. To keep dB values above zero, the reference is generally set to be the lowest value of the quantity that we can imagine or normally wish to use. Before explaining the calculation of dB values, it is useful to remember the following rules of thumb when dB values are used for sound levels:

- Doubling of the Sound Pressure = 6 dB

- Doubling of the Sound Power = 3 dB

- Doubling of the Perceived Sound Level = (approx) 10 dB

Note: The latter is frequency and level dependent, but the value “10 dB” is a good rule of thumb, especially around 1 kHz.

Table 1 shows the actual value of a specific item, such as sound power, for which the sound level is calculated. First, the sound power value is divided with the reference used and then the ten-based logarithm is applied. This value is then multiplied by 10 to create the decibel value (see equation D-1 below).

For every 10 decibels, a unit called Bel is created. The decibel stands for: *deci* for “one tenth” and *bel* for “Bel” (compare decimeter). The relationship between Bel and decibel is thus: 1 Bel = 10 decibels. It is not possible to directly add or subtract decibel values, since addition of logarithmic values correspond to multiplication of the original quantity.

(eq. D-1)

**Table 1**

| Power form, squared units | | Level form |
|---|---|---|
| **Ration of Value to Reference** | Exponential Form of Ratio | 10•Exponent |
| 1 | $10^0$ | 0 |
| 10 | $10^1$ | 10 |
| 100 | $10^2$ | 20 |
| 200 | $10^{2.3}$ | 23 |
| 1,000 | $10^3$ | 30 |
| 10,000 | $10^4$ | 40 |
| 100,000 | $10^5$ | 50 |
| 1000,000 | $10^6$ | 60 |

Each time the sound *pressure* level increases by 6 dB, the corresponding sound *pressure* value is doubled and thus multiplied by 2. Each time the sound *power* level increases by 3 dB, the sound *power* value is multiplied by 2. Thus, it is important to notice that a doubling of the *sound power* is equal to 3 dB, and a doubling of the sound pressure is equal to 6 dB, since a doubling of the sound pressure will result in a quadruple increase of the sound power. The advantage with using dB is simply that they remain the same even if we use sound pressure or sound power. Compare this to the use of voltage and power units in electrical engineering, units being related by $P \sim V^2$. In table 2 an illustration is made of values calculated on sound pressure, non-squared units.

The original definition of decibel was intended for power-like quantities, such as sound power. If we consider sound pressure levels instead (usually denoted *P* in acoustics), the equation will be the same, since the "two" in the squared units will move from within the bracket and become a *20 log* instead of a *10 log* and thus compensate for using linear or quadratic units. Please note that it is not allowed to use *20 log* for squared units, since that expression assumes that we use linear units, like sound pressure in acoustics or voltage in electrical engineering. This is illustrated in equation D-1 below:

$$dB = 10Log_{10}\left[\frac{P^2}{{P_0}^2}\right] = 20Log\left[\frac{P}{P_0}\right] \qquad ;p_0 = 20\mu Pa$$

Table 2 illustrates how a a tenfold increase of the sound pressure will result in an increase in 20 dB steps, while sound power increases in 10 dB steps. See the linear form (Table 2) and compare with equation D-1. In conclusion, dB values are always the same, independent of using sound power or sound pressure as the base unit. A 6 dB increase implies four times the sound power or two times the sound pressure.

| Table 2 | | |
|---|---|---|
| **Linear form, non-squared units** | | Level form |
| **Ration of Value to Reference** | Exponential Form of Ratio | 20•Exponent |
| 1 | $10^0$ | 0 |
| 10 | $10^1$ | 20 |
| 100 | $10^2$ | 40 |
| 200 | $10^{2.3}$ | 46 |
| 1,000 | $10^3$ | 60 |
| 10,000 | $10^4$ | 80 |
| 100,000 | $10^5$ | 100 |
| 1000,000 | $10^6$ | 120 |

**Department of Defense Level ($L_{DOD}$)**

The Average Sound Level calculated in accordance with Department of Defense Exchange Rate and Threshold Level. See Average Sound Level

**Dose**

(See Noise Dose)

**Detector**

The part of a sound level meter that converts the actual fluctuating sound or vibration signal from the microphone to one that indicates its amplitude. It first squares the signal, then averages it in accordance with the time-weighting characteristic, and then takes the square root. This results in an amplitude described as rms (root-mean-square).

**Eight Hour Time-Weighted Average Sound Level ($L_{TWA(8)}$)**

It is the constant sound level that would expose a person to the same Noise Dose as the actual (unsteady) sound levels. The equation for it is

$$L_{TWA(8)} = L_c + qLog_{10}\left(\frac{D}{100}\right)$$

NOTE: This definition applies only for a Criterion Duration of 8 hours.
*Standard*: ANSI S12.19

**Energy Equivalent Sound Level ($L_{eq}$)**

The level of a constant sound over a specific time period that has the same sound energy as the actual (unsteady) sound over the same period.

$$L_{eq} = 10Log_{10}\left[\frac{\int_{T_1}^{T_2} p^2(t)dt}{p_o^2 T}\right]$$

where p is the sound pressure and the Measurement Duration (specific time period) $T=T_2-T_1$. See Sound Exposure Level.

**Exchange Rate (Q), Exchange Rate Factor (q), Exposure Factor (k)**

It is defined in ANSI S1.25 as “the change in sound level corresponding to a doubling or halving of the duration of a sound level while a constant percentage of criterion exposure is maintained.” The rate and the factors are given

in the table below.
*Standard*: ANSI S12.19

| Exchange Rate, Q | Exchange Rate Factor, q | Exposure Factor, k |
|---|---|---|
| 3.01 | 10 | 1 |
| 4 | 13.333 | .75 |
| 5 | 16.667 | .60 |
| 6.02 | 20 | .50 |

**Far Field**

There are two types of far fields: the *acousti*c far field and the *geometric* far field.

*Acoustic Far Field*: The distance from a source of sound is greater than an acoustic wavelength. In the far field, the effect of the type of sound source is negligible. Since the wavelength varies with frequency (See the definition of Wavelength), the distance will vary with frequency. To be in the far field for all frequencies measured, the lowest frequency should be chosen for determining the distance. For example, if the lowest frequency is 20 Hz, the wavelength at normal temperatures is near 56 ft. (17 m); at 1000 Hz, the wavelength is near 1.1 ft. (1/3 m). See the definition of Acoustic Near Field for the advantages of in the acoustic far field.

*Geometric Far Field*: The distance from a source of sound is greater than the largest dimension of the sound source. In the far field, the effect of source geometry is negligible. Sound sources often have a variety of specific sources within them, such as exhaust and intake noise. When in the far field, the sources have all merged into one, so that measurements made even further away will be no different. See the definition of Geometric Near Field for the advantages of being in the geometric far field.

**Free Field**

A sound field that is *free* of reflections. This does not mean that the sound is all coming from one direction as is often assumed, since the source of sound may be spatially extensive. See the definitions of near and far fields for more detail. This definition is often used in conjunction with reverberant field.

**Frequency (Hz, rad/sec)**

The rate at which an oscillating signal completes a complete cycle by returning to the original value. It can be expressed in cycles per second and the value has the unit symbol Hz (Hertz) added and the letter f is used for a universal descriptor. It can also be expressed in radians per second, which has no symbol, and the greek letter ω is used for a universal descriptor. The two expressions are related through the expression $\omega=2\Pi f$.

**Frequency Band Pass Filter**

The part of certain sound level meters that divides the frequency spectrum on the sound or vibration into a part that is unchanged and a part that is filtered out. It can be composed of one or more of the following types:

*Low Pass*: A frequency filter that permits signals to pass through that have frequencies below a certain fixed frequency, called a *cutoff frequency*. It is used to discriminate against higher frequencies.

*High Pass*: A frequency filter that permits signals to pass through that have frequencies above a certain fixed frequency, called a *cutoff frequency*. It is used to discriminate against lower frequencies.

*Bandpass*: A frequency filter that permits signals to pass through that have frequencies above a certain fixed frequency, called a lower cutoff frequency, and below a certain fixed frequency, called an *upper cutoff frequency*. The difference between the two cutoff frequencies is called the *bandwidth*. It is used to discriminate against both lower and higher frequencies so it passes only a band of frequencies.

*Octave band*: A bandpass frequency filter that permits signals to pass through that have a bandwidth based on octaves. An *octave* is a doubling of frequency so the upper cutoff frequency is twice the lower cutoff frequency. This filter is often further subdivided in 1/3 and 1/12 octaves (3 and 12 bands per octave) for finer frequency resolution. Instruments with these filters have a sufficient number of them to cover the usual range of frequencies encountered in sound and vibration measurements.The frequency chosen to describe the band is that of the center frequency. Note table in Frequency Filter - Frequency Weighting.

**Frequency Filter - Weighted**

A special frequency filter that adjusts the amplitude of all parts of the frequency spectrum of the sound or vibration unlike band pass filters. It can be composed of one or more of the following types:

*A-Weighting*: A filter that adjusts the levels of a frequency spectrum in the same way the human ear does when exposed to low levels of sound. This weighting is most often used for evaluation of environmental sounds. See table below.
*B-Weighting*: A filter that adjusts the levels of a frequency spectrum in the same way the human ear does when exposed to higher levels of sound. This weighting is seldom used. See table below.
*C-Weighting*: A filter that adjusts the levels of a frequency spectrum in the same way the human ear does when exposed to high levels of sound. This weighting is most often used for evaluation of equipment sounds. See table below.
*Flat-Weighting*: A filter that does not adjust the levels of a frequency spectrum. It is usually an alternative selection for the frequency-weighting selection.

| Center Frequencies, Hz | | Weighting Network Frequency Response | | |
|---|---|---|---|---|
| 1/3 Octave | 1 Octave | A | B | C |
| 20 | | -50.4 | -24.2 | -6.2 |
| 25 | | -44.7 | -20.4 | -4.4 |
| 31.5 | 31.5 | -39.4 | -17.1 | -3.0 |
| 40 | | -34.6 | -14.2 | -2.0 |
| 50 | | -30.2 | -11.6 | -1.3 |
| 63 | 63 | -26.2 | -9.3 | -0.8 |
| 80 | | -22.5 | -7.4 | -0.5 |
| 100 | | -19.1 | -5.6 | -0.3 |
| 125 | 125 | -16.1 | -4.2 | -0.2 |
| 160 | | -13.4 | -3.0 | -0.1 |
| 200 | | -10.9 | -2.0 | 0 |

| Center Frequencies, Hz | | Weighting Network Frequency Response | | |
|---|---|---|---|---|
| 1/3 Octave | 1 Octave | A | B | C |
| 250 | 250 | -8.6 | -1.3 | 0 |
| 315 | | -6.6 | -0.8 | 0 |
| 400 | | -4.8 | -0.5 | 0 |
| 500 | 500 | -3.2 | -0.3 | 0 |
| 630 | | -1.9 | -0.1 | 0 |
| 800 | | -0.8 | 0 | 0 |
| 1000 | 1000 | 0 | 0 | 0 |
| 1250 | | 0.6 | 0 | 0 |
| 1600 | | 1.0 | 0 | -0.1 |
| 2000 | 2000 | 1.2 | -0.1 | -0.2 |
| 2500 | | 1.3 | -0.2 | -0.3 |
| 3150 | | 1.2 | -0.4 | -0.5 |
| 4000 | 4000 | 1.0 | -0.7 | -0.8 |
| 5000 | | 0.5 | -1.2 | -1.3 |
| 6300 | | -0.1 | -1.9 | -2.0 |
| 8000 | 8000 | -1.1 | -2.9 | -3.0 |
| 10000 | | -2.5 | -4.3 | -4.4 |
| 12500 | | -4.3 | -6.1 | -6.2 |
| 16000 | 16000 | -6.6 | -8.4 | -8.5 |
| 20000 | | -9.3 | -11.1 | -11.2 |

**$L_{eq}$**

See “Energy Equivalent Sound Level”, “Sound Level”, Energy Average”, and “Time Weighted Average”

**Level (dB)**

A descriptor of a measured physical quantity, typically used in sound and vibration measurements. It is attached to the name of the physical quantity to denote that it is a logarithmic measure of the quantity and not the quantity itself. The word *decibel* is often added after the number to express the same thing. When frequency weighting is used the annotation is often expressed as dB(A) or dB(B).

**Measurement Duration (T)**

The time period of measurement. It applies to hearing damage risk and is generally expressed in hours.
*Standard*: ANSI S12.19

**Microphone Guidelines**

***Microphone - Types***: A device for detecting the presence of sound. Most often it converts the changing pressure associated with sound into an electrical voltage that duplicates the changes. It can be composed of one of the following types:

*Capacitor* (Condenser): A microphone that uses the motion of a thin diaphragm caused by the sound to change the capacitance of an electrical circuit and thereby to create a signal. For high sensitivity, this device has a voltage applied across the diaphragm from an internal source.
*Electret:* A microphone that uses the motion of a thin diaphragm caused by the sound to change the capacitance of an electrical circuit and thereby to create a signal. The voltage across the diaphragm is caused by the charge embedded in the electret material so no internal source is needed.

**Microphone - Uses**: The frequency response of microphones can be adjusted to be used in specific applications. Among those used are:

*Frontal incidence (Free Field):* The microphone has been adjusted to have an essentially flat frequency response when in a space relatively free of reflections and when pointed at the source of the sound.
*Random incidence:* The microphone has been adjusted to have an essentially flat frequency response for sound waves impinging on the microphone from all directions.
*Pressure:* The microphone has not been adjusted to have an essentially flat frequency response for sound waves impinging on the microphone from all directions.

**What a microphone measures**: *A microphone detects more than just sound*. The motion of a microphone diaphragm is in response to a force acting on it. The force can be caused by a number of sources only one of which are we interested: sound. Non-sound forces are: (1) direct physical contact such as that with a finger or a raindrop; (2) those caused by the movement of air over the diaphragm such as environmental wind or blowing; (3) those caused by vibration of the microphone housing; and (4) those caused by strong electrostatic fields.
*Rules*:
1. Do not permit any solid or liquid to touch the microphone diaphragm. Keep a protective grid over the diaphragm.
2. Do not blow on a microphone and use a wind screen over the microphone to reduce the effect of wind noise.
3. Mount microphones so their body is not subject to vibration, particularly in direction at right angles to the plane of the diaphragm.

4. Keep microphones away from strong electrical fields.

*A microphone measures forces not pressures*. We would like the microphone to measure sound pressure (force per unit area) instead of sound force. If the pressure is applied uniformly over the microphone diaphragm a simple constant (the diaphragm area) relates the two, but if the pressure varies across the diaphragm the relationship is more complex. For example, if a negative pressure is applied on one-half the diaphragm and an equal positive pressure is applied to the other half, the net force is zero and essentially no motion of the diaphragm occurs. This occurs at high frequencies and for specific orientations of the microphone.

*Rules*:

1. Do not use a microphone at frequencies higher than specified by the manufacturer; to increase the frequency response choose smaller microphones.
2. Choose a microphone for *free field* or *random incidence* to minimize the influence of orientation.

*A microphone influences the sound being measured*. The microphone measures very small forces, low level sound can run about one-billionth of a PSI! Every measurement instrument changes the thing being measured, and for very small forces that effect can be significant. When sound impinges directly on a microphone the incident wave must be reflected since it cannot pass through the microphone. This results in the extra force required to reflect the sound and a microphone output that is higher than would exist if the microphone were not there. This is more important at high frequencies and when the microphone is facing the sound source.

*Rules*:

1. Do not use a microphone at frequencies higher than specified by the manufacturer; to increase the frequency response choose smaller microphones.
2. Choose a microphone for *free field* or *random incidence* to minimize the influence of orientation.

*A microphone measures what is there from any direction*: Most measurements are intended to measure the sound level of a specific source, but most microphones are not directional so they measure whatever is there, regardless of source.

*Rules*:

1. When making hand-held measurements, keep your body at right angles to the direction of the sound you are interested in and hold the meter as far from your body as

possible. Use a tripod whenever possible.
2. Measure the influence of other sources by measuring the background sound level without the source of interest. You may have to correct for the background.

**Near Field**

There are two types of near fields: the *acoustic near field* and the *geometric near field*.

*Acoustic Near Field*: The distance from a source of sound is less than an acoustic wavelength. In the near field, the effect of the type of sound source is significant. Since the wavelength varies with frequency (See the definition of Wavelength), the distance will vary with frequency. The most common example of a near field is driving an automobile with an open window. As you move your ear to the plane of the window, the sound pressure level builds up rapidly (wind noise) since most of the pressure changes are to move the air and very little of it compresses the air to create sound. Persons not far way, can hardly hear what you hear. The acoustic near field is characterized by pressures that do not create sound that can be measured in the far field. Therefore measurements made here are not useful in predicting the sound levels far way or the sound power of the source.

*Geometric Near Field*: The distance from a source of sound is less than the largest dimension of the sound source. In the near field, effect of source geometry is significant. Sound sources often have a variety of specific sources within them, such as exhaust and intake noise. When in the near field, the sound of a weaker, but close, source can be louder than that of a more distant, but stronger, source. Therefore measurements made here can be used to separate the various sources of sound, but are not useful in predicting the sound levels and sound spectrum far from the source.

**Noise**

Typically it is *unwanted* sound. This word adds the response of humans to the physical phenomenon of sound. The descriptor should be used only when negative effects on people are known to occur. Unfortunately, this word is used also to describe sounds with no tonal content (random):

*Ambient:* The all encompassing sound at a given location caused by all sources of sound. It is generally random, but need not be.

*Background:* The all encompassing sound at a given location caused by all sources of sound, but excluding the source to be measured. It is essentially the sound that interferes with a measurement.
*Pink:* It is a random sound that maintains constant energy per octave. Pink light is similar to pink noise in that it has a higher level at the lower frequencies (red end of the spectrum).
*White:* It is a random sound that contains equal energy at each frequency. In this respect, it is similar to white light.

**Noise Dose (D)**

It is the percentage of time a person is exposed to noise that is potentially damaging to hearing. Zero represents no exposure and 100 or more represents complete exposure. It is calculated by dividing the actual time of exposure by the allowed time of exposure. The allowed time of exposure is determined by the Criterion Duration and by the sound level (the higher the level, the shorter the allowed time). The sound levels must be measured with A-frequency weighting and slow exponential time weighting. See Projected Noise Dose.

$$D = \frac{100T}{T_c}10^{(L_i - L_c)/q}$$

where

T is Measurement Duration

$T_c$ is Criteria Time

$L_i$ is TWA

$L_c$ is Criteria Level

q is exchange rate factor; see page D-8 "Exchange Rate (Q), Exchange Rate Factor (q), Exposure Factor (k)"

*Standard*: ANSI S12.19

**Noise Exposure** (See Sound Exposure)

**OSHA Level ($L_{OSHA}$)** The Average Sound Level calculated in accordance with the Occupational Safety and Health Administration Exchange Rate and Threshold Level.

**Preamplifier** A part of the sound level meter that matches a particular model of microphone to the meter. It must be chosen in conjunction with a microphone and a cable that connects them.

**Projected Noise Dose** It is the Noise Dose expected if the current rate of noise exposure continues for the full Criterion Duration period.

**Single Event Noise Exposure Level (SENEL, $L_{AX}$)** The total sound energy over a specific period. It is a special form of the Sound Exposure Level where the time period is defined as the start and end times of a noise event such as an aircraft or automobile passby.

**Sound** The rapid oscillatory compressional changes in a medium (solid, liquid or gas) that propagate to distant points. It is characterized by changes in density, pressure, motion, and temperature as well as other physical quantities. Not all rapid changes in the medium are sound (wind noise) since they do not propagate.
The auditory sensation evoked by the oscillatory changes.

*Difference between sound and noise:* Sound is the physical phenomenon associated with acoustic (small) pressure waves. Use of the word *sound* provides a neutral description of some acoustic event. Generally, noise is defined as unwanted sound. It can also be defined as sound that causes adverse effects on people such as hearing loss or annoyance. It can also be defined as the sound made by other people. In every case, noise involves the judgment of someone and puts noise in the realm of psychology not physics.
*Rules*:
1. Use word *sound* to describe measurements to remove the emotional overtones associated with the word *noise*. Some sound metrics use noise in their name and it is proper to use the name as it is.

**Sound Exposure (SE)**

It is the total sound energy of the actual sound during a specific time period. It is expressed in Pascals-squared seconds.

$$SE = \int_{T_1}^{T_2} p_A^2(t)dt$$

where $p_A$ is the sound pressure and $T_2$ - $T_1$ is the Measurement Duration (specific time period).
When applied to hearing damage potential, the equation is changed to

$$SE = \int_{T_1}^{T_2} [p_A^2(t)]^k dt$$

where k is the Exposure Factor. See Exchange Rate.
*Standard*: ANSI S1.25

**Sound Exposure Level (SEL, $L_{ET}$)**

The total sound energy in a specific time period. The equation for it is

$$SEL = 10Log_{10}\left[\frac{\int_{T_1}^{T_2} p^2(t)dt}{p_0^2 T}\right]$$

The sound pressure is squared and integrated over a specific period of time ($T_2$-$T_1$) this is called the sound exposure and has the units Pascal squared- seconds or Pascal squared-hours. P0 is the reference pressure of 20 µPa and T is the reference time of 1 second. It is then put into logarithmic form. It is important to note that it is not an average since the reference time is not the same as the integration time.

**Sound Pressure**

The physical characteristic of sound that can be detected by microphones. Not all pressure signals detected by a microphone are sound (e.g., wind noise). It is the amplitude of the oscillating sound pressure and is measured in Pascals (Pa), Newtons per square meter, which is a metric equivalent of pounds per square inch. To measure sound, the oscillating pressure must be separated from the steady (barometric) pressure with a detector. The detector takes out the steady pressure so only the oscillating pressure remains. It then squares the pressure, takes the time average, and then takes the square root (this is called rms for root-mean square). There are several ways this can be done.

*Moving Average*: The averaging process is continually accepting new data so it is similar to an exponential moving average. The equation for it is

$$p_{rms} = \sqrt{\frac{1}{T}\int_{t_s}^{t} p^2(\xi)e^{-(t-\xi)/T}d\xi}$$

The sound pressure is squared and multiplied by a exponential decay factor so that when the time of integration is near the current time (t) it is essentially undiminished. For times older (less) than the current time, the value is diminished and so becomes less important. The rate at which older data are made less influential is expressed by the constant T. The larger is it the slower the decay factor reduces and the slower the response of the system to rapid changes. These are standardized into three values called Time Weighting. See the values below.

*Fixed Average:* The averaging process is over a fixed time period. The equation for it is

$$p_{rms} = \sqrt{\frac{1}{(T_2 - T_1)} \int_{T_1}^{T_2} p^2(t)dt}$$

The sound pressure is squared and averaged over a fixed time period. Unlike the moving average, the sound pressures in all time intervals are equally weighted.

**Sound Pressure Level (SPL, $L_p$)**

The logarithmic form of sound pressure. It is also expressed by attachment of the word decibel to the number. The logarithm is taken of the ratio of the actual sound pressure to a reference sound pressure which is 20 MicroPascals (µ Pa). There are various descriptors attached to this level depending on how the actual sound pressure is processed in the meter:

*Instantaneous*: The time varying reading on a meter face on in a meter output due to changes in the sound pressure. The reading will depend on the time-weighting applied.

The fundamental relationship between the two is logarithmic

$$L_p = 20\log_{10}\left[\frac{p_{rms}}{p_0}\right] \qquad p_{rms} = p_0 10^{L_p/20}$$

where $p_0$ is the reference sound pressure of 20 µPa. The square of the sound pressure is a power-like quantity that can be expressed in the original form of the level definition

$$L_p = 10\log_{10}\left[\frac{p_{rms}^2}{p_0^2}\right] \qquad p_{rms}^2 = p_0^2 10^{L_p/10}$$

Sound Pressure Level can be converted to sound pressure as follows. If the sound pressure is 1 Pascal, then the sound pressure level is

$$L_p = 20\log_{10}\left[\frac{1}{20 \bullet 10^{-6}}\right] = 20\log_{10}[50000] = 20[4.699] = 94.0dB$$

Calibrators often use a level of 94 dB so they generate a sound pressure of 1 Pascal.

If the sound pressure level = 76.3 dB, then the sound pressure is

$$Pa = 20 \bullet 10^{-6} \bullet 10^{76.3/20} = 20 \bullet 10^{3.815 - 6} = 20 \bullet 10^{-2.185} = 20[0.0065] = 0.13$$

*Energy Average ($L_{eq}$)*: The value of a steady sound measured over a fixed time period that has the same sound energy as the actual time varying sound over the same period. This descriptor is widely used. It is a fixed average (See Sound Pressure).

*Impulse*: The value of an impulsive sound. The reading will depend on the time-weighting applied.

*Unweighted Peak*: The peak value of a sound with a meter that has flat frequency weighting and a peak detector.

*Weighted Peak:* The peak value of a sound with a meter that has a frequency weighting other than flat and a peak detector.

**Sound Power(W)**

The sound power emitted by a sound source. It is measured in Watts.

**Sound Power Level (PWL, $L_W$)**

The logarithmic form of sound power. It is also expressed by attachment of the word decibel to the number. The logarithm is taken of the ratio of the actual sound power to a reference sound power, which is 1 pico-watt. Sound power level cannot be measured directly, but can only be deduced through measurements of sound intensity or sound pressure around the source. The equation for it is

$$L_w = 10\log_{10}\left[\frac{W}{W_0}\right] \qquad W = W_0 10^{L_w/10}$$

**Sound Speed, (c,)**

The speed at which sound waves propagate. It is measured in meters per second. It should not be confused with sound or particle velocity which relates to the physical motion of the medium itself.

$$c = 20.05\sqrt{\text{degC} + 273} \qquad \text{m/sec}$$

$$c = 49.03\sqrt{\text{degF} + 460} \qquad \text{ft/sec}$$

**Spectrum (Frequency Spectrum)**

The amplitude of sound or vibration at various frequencies. It is given by a set of numbers that describe the amplitude at each frequency or band of frequencies. It is often prefixed with a descriptor that identifies it such as sound pressure spectrum. It is generally expressed as a spectrum level.

**Threshold Sound Level (Lt)**

The A-weighted sound level below which the sound produces little or no Noise Dose accumulation and may be disregarded. It is used for hearing damage risk assessment.
*Standard*: ANSI S1.25

**Time Weighted Average Sound Level (TWA, $L_{TWA(TC)}$)**

It is the level of a constant sound over the Criterion Duration, that would expose a person to the same Noise Dose as the actual (unsteady) sound over the same period. If the Exchange Rate is 3 dB then the TWA is equal to the $L_{eq}$.

where $T_c$=$T_2$-$T_1$ and K is the Exchange Rate Factor. It is used for hearing damage risk assessment.
*Standard*: ANSI S12.19

$$L_{TWA(TC)} = K\log_{10}\left(\frac{1}{T}\int_{T_1}^{T_2} 10^{(L_p(t))/K} dt\right)$$

**Time Weighting**

The response speed of the detector in a sound level meter. There are several speeds used.

*Slow*: The time constant is 1 second (1000 ms). This is the slowest and is commonly used in environmental noise measurements.
*Fast*: The time constant is 1/8 second (125 ms). This is a less commonly used weighting but will detect changes in sound level more rapidly.
*Impulse*: The time constant is 35ms for the rise and 1.5 seconds (1500 ms) for the decay. The reason for the double constant is to allow the very short signal to be captured and displayed.

**Vibration**

The oscillatory movement of a mechanical system (generally taken to be solid). It is used as a broad descriptor of oscillations.

**Wavelength (l)**

The distance between peaks of a propagating wave with a well defined frequency. It is related to the frequency through the following equation

$$\lambda = \frac{c}{f}$$

where $c$ is the sound speed and $f$ is the frequency in Hz. It has the dimensions of length.

**Wavenumber (k)**

A number that is related to the wavelength of sound and is used to compare the size of objects relative to the wavelength or the time delay in sound propagation. It is related to wavelength through the following equation

$$k = \frac{2\pi}{\lambda} = \frac{2\pi f}{c} = \frac{\omega}{c}$$

where $\lambda$ is the wavelength, $c$ is the sound speed, $f$ is the frequency in Hz, and $\omega$ is the radian frequency. It has the dimensions of inverse length.