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Tutorial on Sound Level Meters: Practical Tips on Use

Judith Rochat
Cross-Spectrum Acoustics
1055 E. Colorado Blvd, Suite 500
Pasadena
CA 91106
jrochat@csacoustics.com

Herbert Singleton
Cross-Spectrum Acoustics
25A Granby Rd
East Longmeadow
MA 01028
hsingleton@csacoustics.com

ABSTRACT

This portion of the Tutorial on Sound Level Meters discusses practical tips for deploying a sound level meter and preparing for measurements. The practical tips include best practices for calibrating a sound level meter system, both in terms of keeping it field-ready and for use in the field prior to and after data collection. The use of wind screens—when it's necessary or helpful—is also discussed. Other practical tips include how to orient your sound level meter/microphone in relation to the sound source of interest and holding or tripod-mounting your system. Also, sound level meter settings both before and during field deployment are discussed, along with consideration of background noise.

1 INTRODUCTION

This paper discusses practical tips on use of sound level meters; it is the third of a series of five papers: Tutorial on Sound Level Meters. There may be some overlap in material contained in the five papers in order for each paper to be a standalone reference. Practical Tips on Use covers five main topics, seen as the sections that follow:

- Calibration best practices;
- When to use a windscreen;
- Deploying a sound level meter (SLM);
- Choosing settings; and
- Background noise.

2 CALIBRATION BEST PRACTICES

2.1 Terminology

The term "calibration" is used herein in three ways:

1) (LAB) at manufacturer-recommended intervals, to send your sound level meter to a certified laboratory to have it calibrated (checked for conformance);
2) (SET-CAL) to perform an in-situ, amplitude sensitivity *adjustment* of an acoustic measurement system using a reference-level sound pressure signal (by attaching a sound calibrator to the microphone) to establish and, if necessary, change the level sensitivity such that the system is properly indicating the absolute sound pressure level (SPL); and

3) (CHECK-CAL) perform an in-situ, *measurement* of a reference-level signal (by attaching a sound calibrator to the microphone) to validate or confirm that the system sensitivity is indicating absolute sound pressure levels properly. Such CHECK-CALs should be performed immediately after performing a SET-CAL, as well as immediately before and after collecting noise measurement data. (Measurement data should not be considered to be valid unless bracketed by valid CHECK-CALs.)

When performing either a SET-CAL or a CHECK-CAL, all components of the measurement system that are part of the signal path between the microphone and the "readout" (basically anything used to present a noise level value, such as a screen, stored value in a file, recorded signal, etc.) should be included during the calibrations. For field calibration purposes, an acoustic instrumentation system can include: microphone, preamplifier, sound level meter, interconnecting cables, signal-conditioning components such as filters or gain amplification stages, and audio recorders.

2.1 Laboratory

To maintain -sound level meter performance (and subsequently, accuracy), it is typical for agencies and standards to require periodic laboratory calibration (LAB). A calibration laboratory can measure, check, confirm, and provide documentation for performance testing, and should provide a dated and signed calibration certificate. In some cases – especially when a calibration laboratory is operated by the instrument manufacturer/distributor, the lab may be able to adjust internal settings that affect acoustical performance. In such cases, a tamper-resistant seal of some kind may be applied to the internals or the housing of the instrument to indicate this status. Some agencies or standards require an annual calibration interval to keep the unit field-ready. Current thinking is that the instrument manufacturer should provide documentation of recommended intervals between laboratory calibrations, and these intervals may be based on the history of calibration adjustments made with a particular instrument.[1] For now, agency/standard requirements need to be adhered to unless otherwise arranged or documented. However, look to the future for potential changes; one source of guidance that may lead to future changes in required calibration intervals is ILAC-G24-OIML-D10.[2]

2.2 Measurements

An acoustic calibrator provides the means of checking the entire acoustic instrumentation system's sensitivity by producing a known sound pressure level (referred to as the calibrator's reference level) at a known frequency, typically 94 dB or 114 dB SPL (referenced to 20 microPascals for sound measurements in air) at a frequency of 1 kHz, or 124 dB at 250 Hz. Note that calibrations made at 250 Hz may require careful testing and switching of frequency-weighting networks. For example, the A-weighting curve applies an adjustment of approximately -8.6 dB at 250 Hz relative to the effect at 1 kHz. Also note that the reference level generated by the calibrator may depend on the type of microphone being tested. For example, while the Larson Davis CAL200 calibrator nominally provides a reference pressure level of 94.0 dB, the actual level generated when testing free-field microphones is 93.88 dB. The manufacturer's documentation or accredited lab-tested levels should be consulted to determine the actual reference pressure level for a given microphone size and type.

Pistonphones are a type of calibrator that use a mechanical piston to generate the reference pressure, rather than the electromagnetic loudspeaker found in typical acoustic calibrators. Pistonphones can provide a highly precise calibration level but note that the generated level will

vary depending on ambient air pressure. Pistonphones often provide adjustment values for various ambient pressure levels that should be applied to the level shown on the sound level meter. Pistonphones are not as common as acoustic calibrators but are sometimes used for measurements at high or low altitudes where ambient pressure may affect sound levels.

In practice, a calibrator is first used during a measurement deployment to establish the sensitivity of the system as it is set up in the field (SET-CAL), then again—to confirm the system sensitivity—at the beginning and end of each measurement session and before and after any changes are made to the system configuration or components (CHECK-CAL).[3] Note that if you need to re-establish the sensitivity (SET-CAL) after data collection, it should not be done until a CHECK-CAL has been performed and the calibration level documented; *caution:* some sound level meters will discard any stored data when you do a SET-CAL calibration (for such instruments, measurement protocols should be established that include downloading of measured noise levels to external storage prior to re-establishing system sensitivity by performing another SET-CAL calibration).

When using a calibrator that can provide multiple output levels (e.g., 94 or 114 dB) the choice of output level should be selected with optimization of the dynamic range of the SLM/system in mind. This topic is beyond the scope of this paper, but considerations for such optimization include: the range of sound levels expected during data collection, the dynamic range of your system, and the dynamic range of supplemental instrumentation (e.g., audio recorder). For some systems, the applied calibrator output level needs to be entered and stored in the sound level meter settings, so the system properly adjusts to the known input. You may notice the user-entered calibration level is not set properly if the displayed readout sound level values are unusually higher or lower than expected during a calibration check (CHECK-CAL) or data collection. For example, if the setting in the sound level meter was entered as 114 dB, and you actually used the 94 dB setting on the calibrator, the system will interpret 94 dB to be 114 dB; as a result, your readings during data collection will be 20 dB higher than what they really are. Hopefully, you will notice this error when you take a calibration reading (CHECK-CAL) when checking the system; in this case you can just recalibrate the system to the proper level (SET-CAL) prior to another system check (CHECK-CAL) and data collection. If you notice such erroneous level readings during or after any noise data collection, you may be able to "correct" the measured levels by adjusting them appropriately (in the example case, you would need to subtract 20 dB from measured values).

When placing a calibrator on the microphone, the calibrator must be properly seated (fully inserted in the coupler); please refer to Figure 1. Gently place the calibrator on the microphone (avoid slamming it onto the microphone so as not to disturb the microphone diaphragm). While placing and removing the calibrator, turn the calibrator slightly clockwise to prevent the microphone from unscrewing (and it's best to gently check that the microphone is securely screwed onto the preamplifier after it has been removed from the calibrator coupler). Before calibrating, ensure that the calibrator is turned on; you can see this in standby mode or by simply running the sound level meter (the level should read close to what the input level is – unless it was set improperly in its previous use!).

Figure 1: *Sound level meter with calibrator seated on microphone.*

3 WHEN TO USE A WINDSCREEN

Wind can impinge the microphone diaphragm and introduce contaminated readings, even through a proper windscreen. Measurements should not be taken when wind exceeds 11 mph (17 kms or 5 m/s), regardless of direction (or avoid inclusion and use of any measured noise data collected during such periods in the data analysis phase). When applying A-weighting and the sound levels of interest are lower than 40 dBA, even lower-speed winds can generate false noise. It is best to make measurements when winds are calm.[3] An exception to this guidance would be if the purpose of the study is to determine effects of wind on sound propagation, in which case the wind limit should still not exceed 11 mph (17 kms or 5 m/s).

A typical windscreen is a porous sphere placed around a microphone to reduce the effects of wind-generated noise on the microphone diaphragm. The windscreen (preferably new) should be clean, dry, and in good condition. The screen should be stored in its original box or equivalent safe container to avoid compression, dirt, and exposure to the elements.

Windscreens are available in a number of sizes, typically 90-mm (most common), 63 mm, and 180 mm. Larger windscreens are used in quieter environments where wind contamination may be especially problematic. Usually, the insertion effect on the measured sound level due to the presence of a windscreen into an acoustic instrumentation system can be neglected (exception: when spectral analysis is critical). However, larger windscreens will generate a larger insertion effect than smaller windscreens. The windscreen manufacturer may be able to provide a table of corrections based on frequency and sound incidence angle. Alternatively, testing via high-

precision measurements in an anechoic chamber may be performed if a windscreen manufacturer does not provide necessary correction data.[3]

Windscreens should be used in all outdoor measurements to minimize contamination from wind. When attaching/applying the windscreen to a microphone, it should be placed around the microphone gently, with the microphone inserted until there is slight resistance. This places the microphone diaphragm in the proper location within the foam sphere. Although a practitioner hopes to avoid this, a windscreen can also help to protect a microphone in the event that a sound level meter/tripod topples. Note that after any such impact, the microphone should be visually inspected for damage or physical contamination of the protective grid. Best practice—especially for pre-polarized (electret) condenser mics, which are particularly susceptible to physical shock—is to eliminate that microphone from use for that measurement day. Such shocks can temporarily alter the acoustical sensitivity of the microphone, but fortunately such changes are typically temporary and allowing the diaphragm to "settle" overnight is often successful.

4 DEPLOYING A SOUND LEVEL METER

4.1 Microphone Orientation

The type of microphone and its orientation in the sound field can influence the accuracy of the measurements.[4] Microphone sensitivity vs. frequency also can vary by angle of incidence between the sound waves and the microphone diaphragm. Ideally, the microphone should have a frequency response for which the sensitivity is uniform (or "flat") over the frequency range of interest.[4] Manufacturers of high quality microphones can provide response curves and tables of microphone performance and corrections at different orientations. A correction for microphone frequency response often may be neglected unless predominantly high-frequency noise is being measured.

Two typical microphone orientations are "normal" and "grazing" incidence:

Normal incidence (0° incidence, relative to the cylindrical axis of the microphone, see Figure 2) occurs when sound waves impinge at an angle perpendicular ("normal") to the microphone diaphragm. Normal incidence is best used for situations involving point-source measurements, where the sound being measured is coming from a stationary, single, known direction (e.g., an idling automobile or a power generator).[3]

Grazing incidence (90° incidence, see Figure 2) occurs when sound waves impinge at an angle parallel to the plane of the microphone diaphragm. This orientation is preferred for moving—or line-source—measurements (e.g., operating transportation sources) because, when aligned properly, the microphone can present a constant incidence angle to any source located within the plane of the microphone diaphragm.[3]

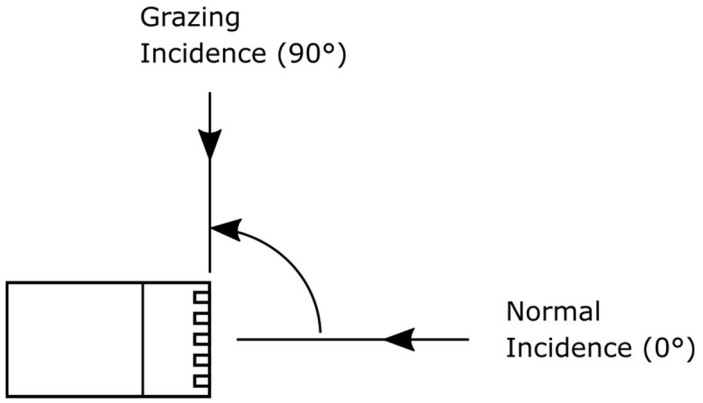

Figure 2: *Microphone incidence (FHWA Noise Measurement Handbook[3]).*

Another common orientation is at an angle 70-80° from normal incidence to the source. Such orientation may be most accurate for a random-incidence microphone and stationary source.

4.2 Handheld Operations versus Tripod-Mounting of a Sound Level Meter

Although convenient for short-term noise measurements, manually holding a sound level meter is considered less optimal than tripod-mounting for several reasons;

1. Tripods are physically stabile, and can prevent mechanical noise from handling being introduced during a measurement;
2. The exact location of the microphone can be determined and recorded;
3. The body of the operator can sometimes interfere with the sound field and affect the measured data during handheld operation. To minimize such effects, the operator should hold the instrument at an arm's length and to one side, although this can still influence the sound level up to about 1 dB at several frequencies.[4]

Preferably, the microphone/preamplifier should be mounted on a tripod with the sound level meter connected by cable at a distance/position minimizing effects of reflections from the sound level meter body or case in which it is stored. The sound level meter itself can be mounted on a tripod; the influence of the instrument body can usually be ignored.[4]

5 CHOOSING SETTINGS

Prior to collecting data, it is important to select the proper settings in your sound level meter. Noise measurement standards, policy, practice guidance, and agency specifications often stipulate at least some of the applicable settings. In most cases, settings can be stored as presets in your sound level meter. The filename can be related to the type of measurement or the name of a particular measurement program. The settings can also be listed on a "cheat sheet" to be used in the field to either select or confirm the applicable settings. Having such a sheet or sound level meter preset file is important for helping to document settings applied to a particular program or for future application to similar measurements.

The primary settings that are of concern for typical noise measurements are listed below. The choice of settings depends on the purpose of the noise measurements and applicable guidance/standards/specifications/practice. Prior to any field (data collection) measurements, practitioners should consult the manual for their sound level meter to determine how to apply the key settings shown below in addition to other settings. The sound level meter should be tested not

only for data collection but also for data download and processing. This is also a good time to test all supporting equipment (e.g., calibrator, cables, tripod), as well as battery life.

Note: For some sound level meters, settings can be applied separately for logging and reporting. Make sure appropriate settings are applied to each per your sound level meter documentation and intended outcome.

Metric

Typical metrics applied to noise measurements include:

- time-averaged equivalent sound levels over a specified duration (L_{eqT}, this metric is the quantity measured directly by an integrating-averaging sound level meter; this is also simply referred to as L_{eq} with the duration specified separately);
- time-averaged sound levels over a 24-hour period penalized for nighttime noise (day-night average sound level, L_{dn} or DNL) or penalized for evening and nighttime noise (community noise equivalent level, L_{den} or CNEL);
- maximum or minimum sound levels (L_{max} or L_{min});
- peak sound level (L_{pk}, can be higher than the maximum sound level depending on the other settings and the source noise);
- or percentile sound levels (e.g., L_{90} or L_{10} for the sound level exceeded 90% or 10% of a measurement period).

While some metrics are not required for a typical noise measurement study, they may be helpful during data analysis. For example, if you need to know the average sound level over a time period, it is also useful to know the minimum and maximum sound levels (may help to determine how loud single events are or if data are contaminated or how quiet an area is without a targeted noise source), and/or statistical sound levels (L_{90}, for example, is a good indicator of the background noise level, and the L_{10} - L_{eq} difference provides clues about the amount of variation in levels during the sample period[3]). Often, metrics like L_{dn} and L_{den} are constructed from field measurements taken simply as L_{eq} values.

For a particular type of measurement, the metric to apply is typically specified, although sometimes it must be inferred from context. Settings for the metric types are typically found under the SLM, Measurement, Logging, Statistics, or Ln category of settings. Some sound level meters allow output of spectral data (e.g., 1/3-octave band data) in addition to broadband sound levels. Sound level meters with this capability have a setting to include spectral data in your output.

Frequency Weighting

The most common frequency weightings are A, C, and Z. Applying weightings allows for a better measure of human response.

- For noise control engineering, the most commonly applied weighting is A weighting, which accounts for changes in level sensitivity as a function of frequency. When unspecified, particularly in noise policies, A weighting should be applied.
- C weighting can be applied to louder sounds, when humans become more sensitive to lower frequencies. Sometimes, the difference of C-weighted sound levels and A-weighted sound levels is used to describe the relative contribution of low frequency noise to the broadband sound level.
- Z weighting is actually not a weighting; it simply means that no human sensitivity weighting is applied. Z weighting may be appropriate for equipment testing, or similar, when a sound

level is desired but the effect on humans does not apply. Z-weighted sound levels can also be used as a baseline, then A, C, or other weightings are applied during data processing.

Exponential Time Weighting

The choice of exponential time weighting depends on the variability of the sound being measured and may be specified in an applicable measurement guidance or standard. The choices discussed here are Slow and Fast (which are most common) and Impulse. The difference between the exponential time weightings is the running time average duration, which affects recorded/viewed signal fluctuations. The greater the time constant, the more the most recent sound level is affected by preceding sounds. Be aware that the maximum and minimum sound levels are dependent on the exponential time weighting.

- Slow exponential time weighting applies a 1-second constant. The sound level fluctuates slowly. Sounds that occur as much as 4 seconds before the time of observation can contribute to an observed sound level.[4] Slow response is typically used for measurements of sound sources that vary slowly as a function of time, such as aircraft or steady highway traffic[3] or factory noise emissions.
- Fast exponential time weighting applies a 0.125-second constant. The sound level fluctuates quickly. Sounds that occur 1 second earlier than the time of observation must be greater than at the time of observation to contribute significantly. Fast response is typically used for measurement of individual events, such as vehicle pass-bys[3] or wind turbine amplitude modulation. It should be noted that for the same non-steady sound, the maximum *fast* sound level is generally greater than the corresponding maximum *slow* sound level.[4]
- Impulse exponential time weighting applies non-uniform running time-averaging. It applies a 0.035-second constant for sounds that increase with time and a 1.5-second constant for sounds that decrease with time. Impulse response could be used for measurements of impulsive sounds. However, measurements require special care for proper interpretation, especially for measurements of high-level, short-duration repetitive sounds that occur at intervals less than 1 second.[4]

Some SLMs allow for linear averaging over time (as opposed to applying the exponential time weightings listed above), where notionally the only sound energy included is that within the specified averaging period (see Time Interval below). This is fundamentally the same as an L_{eq} for the time interval. In addition to the time weightings listed above, a Peak setting may be allowed, which is based on the instantaneous sound pressure.

Manufacturers' documentation must be consulted to help in understanding what exactly is being calculated for particular settings. In addition, linear and exponential time weighting settings should be tested in the lab or office to understand and verify the sound levels being displayed/reported.

Time Interval

Time interval settings can refer to two parameters: 1) time interval over which the squared instantaneous sound pressure is integrated to compute the time-averaged sound pressure (referred to here as Averaging Time); and 2) time interval for which data are being reported (referred to here as Reporting Time). Depending on the SLM, the nomenclature for Averaging Time and Reporting Time varies and can and have different implications.

Averaging Time can be found under labels such as Time History Period, Integration Interval, Logging Period, and Sampling Period. Reporting Time can be found under labels such as Interval

Period, Numeric Results time setting, Report Period, and Measurement Duration. Refer to the SLM manufacturer's documentation to understand which setting refers to Averaging Time and which refers to Reporting Time. Then test the SLM in the lab/office to understand what you are seeing on the screen and what is contained in the files generated. Be aware that captured sound levels may not be what you think they are. As an example, if you chose an Averaging Time of one minute and a Reporting Time of 15 minutes, what you see on the screen when you stop the measurements may be the sound level for the full 15 minutes or it may be only for the last minute of data. For data files, you will need to determine which files provide you with the desired output. Averaging Time data should provide you with L_{eq} values (for example, representing each minute separately) that can be combined to generate other metrics, such as 15-minute L_{eq} values or L_{dn} values. Reporting Time data may only show you the running average, such that the level reported for a particular minute, for example, is influenced by sound levels preceding it; referencing SLM documentation and conducting tests are necessary to know how values are being reported.

Dynamic Range

Although some sound level meters have a nominal linear operating range of 90 dB or more, others have a more limited range and provide options for range control. This range control may be adjusted automatically by the meter based on average sound levels or may need to be set manually by the operator prior to operation. The range needs to be adequate for expected sound levels. Manufacturer documentation should state the dynamic range, and the measurement range should be adjusted appropriately as required.

Additional Note

The meter settings for the parameters listed above should be checked prior to every field measurement program to ensure that meter settings were not unknowingly changed. For example, a sound level meter used by a team may typically be used for A-weighted measurements but may have been set to C-weighting for a particular task. If this meter were used again without changing back to A-weighting, the resulting data may be unusable (unless saved files allow for spectral weighting adjustments during data processing). Many modern sound meters allow the simultaneous collection of A-, C- and Z- weighted data along with many other metrics. It is often advisable to collect as many data permutations as possible to permit the determination of any arbitrary metric after the fact.

6 BACKGROUND NOISE

When conducting noise measurements, it is important to consider background noise, the all-encompassing sound of a given environment, excluding contributions from the sound source or sources of interest[3] (unless the source of interest is the background noise itself). There are two primary cases for measuring background noise: 1) to know how much the background noise may be contributing to noise levels measured for a targeted noise source (e.g, train, computer fan, etc.); and 2) to establish existing noise for the purpose of determining acceptable thresholds/limits for noise increases due to a proposed new noise source.

The total measured sound level is a combination of all noise sources. If you are targeting a particular noise source, the total is a combination of that source noise and background noise. If the combined measured sound level is 10 dB more than the background noise, then the background contribution to the total measured level is minimal, less than 0.5 dB. If the combined level does not exceed the background noise by 5 dB or more, then the source-only level cannot be accurately

determined. If the combined level exceeds background between 5 and 10 dB, then the total broadband sound level can be adjusted applying the following equation:

$$L_{adj} = 10\,log_{10}\left[10^{(0.1L_c)} - 10^{(0.1L_b)}\right] \quad (dB)$$

where: L_{adj} is the background-adjusted measured level;

L_c is the total measured level with source and background combined; and

L_b is the background level alone.[3]

Examples of adjusting combined noise to determine the source noise are provided in the FHWA guidance[3] and ANSI S12.9 Part 4.[4] It should be noted that the decibel adjustment guidance is based on broadband noise sources. It may be possible to measure a source at particular frequencies that are well above those same frequencies in the background noise, even when the broadband sound levels for background and combined are within a few decibels. This requires your SLM functioning as a spectrum analyzer in order to make the determination.

Sometimes it is necessary to know background (existing) noise levels to determine thresholds or limits of acceptance for a specification or policy. In these cases, it is important to conduct measurements at times that are representative of the environment, and remove any noise sources from the data that may have been intrusive or not representative. A noise engineer will often need to make a judgement during measurements as to whether or not an extraneous noise source is intrusive (potentially contaminating or influencing your noise source of interest). Start and stop times of intrusive noise should be noted for later examination and potential removal of the intrusive noise event or a time block containing the event (e.g., 1 minute) from the data collected.

Note that it is important to know the low end of the dynamic range of your measurement system when conducting background noise measurements in a very quiet environment.

7 CONCLUSIONS

Practical tips on use of sound level meters are described in this paper. The topics covered are: calibration best practices, when to use a windscreen, deploying a sound level meter (microphone orientation and holding versus tripod-mounting), choosing settings, and consideration of background noise. More information on metrics, choosing a sound level meter, using a sound level meter for standardized measurements, and using a sound level meter for conducting a noise survey can be found in the series of papers: Tutorial on Sound Level Meters.

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