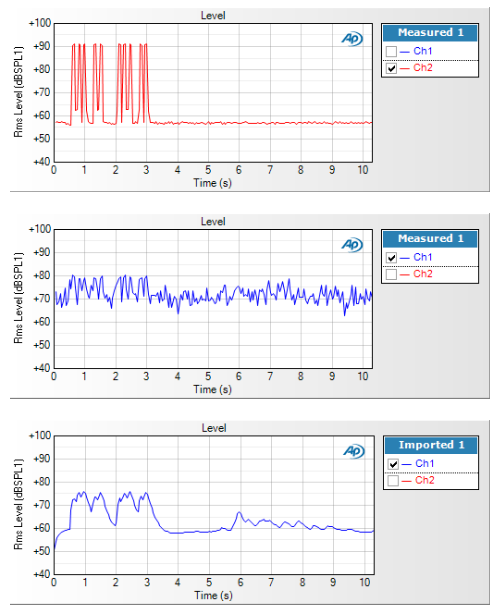
Audio precision

- [ ] 2700 Series
- [x] APx585 family
- [x] APx525 family
- [x] APx515

Measuring Sound Pressure Level with APx

By Joe Begin



Introduction

The level of acoustic noise or sound in the environment is typically measured with a handheld sound level meter. This Technote explains some of the theory behind such measurements, and how to make them instead using an Audio Precision APx500 Series analyzer. Using the analyzer adds the capability to perform spectrum analysis, and to visualize the character of the sound signal over time. The latter is especially valuable when measuring non-steady-state sounds, such as a pulsating alarm.

To use the APx Sound Level Meter Utility associated with this Technote, you will need an APx500 Series audio analyzer and a measurement microphone, such as the MMK-2 available from Audio Precision.

Sound Pressure Measurement

A sound level meter (SLM) is an electronic instrument used to measure sound pressure levels, and contains a measurement microphone, preamplifier, level detectors, one or more frequency weighting filters, and a display. Typically, an SLM is a self-contained hand-held instrument. However, an SLM could also be comprised of separate components in one or more enclosures that work together as a system.

Standards

The international standard governing sound level meters is IEC 61672, “Electroacoustics – Sound Level Meters,” with Part 1 covering specifications. The US equivalent of IEC 61672-1 is ANSI S1.4, “Specification for Sound Level Meters.”

Sound Pressure Level

Sound levels are typically expressed in decibels (dB) relative to a reference sound pressure level of 2.0 x 10^{-5} Pascal (or 20 µPa, where µ is the SI prefix for "micro" [x 10^{-6}]). The reference sound level of 20 µPa, which is agreed to by international standards, corresponds approximately to the threshold of hearing in healthy human subjects, when the sound is at frequencies in the middle of the audio range.

A-weighting

Sound levels are usually measured with a frequency weighting filter known as *A-weighting*. This filter provides significant attenuation at low and high frequencies, with a slight gain in the mid-frequency portion of the audible spectrum, as shown in Figure 1.

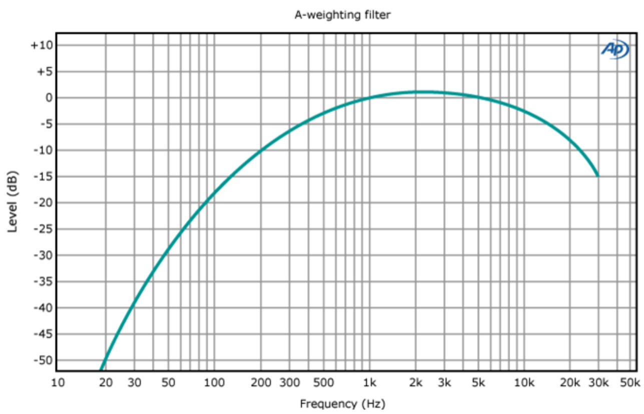


Figure 1. A-weighting filter frequency response.

A-weighting is used in an attempt to provide sound level metrics which correlate with perceived loudness in humans, who are most sensitive to sounds in the mid-frequency range. When measured with A-weighting, sound pressure levels in decibels are usually denoted as dB(A) or simply dBA.

Historically, A-weighting originated as a result of pioneering work by Fletcher and Munson in 1933[1, 2], on how variations in the level and frequency of sound affect perceived loudness. Based on experiments with pure tones presented to subjects through headphones, they produced a set of equal-loudness curves (Figure 2). Three years later, these curves were used in the first American standard for sound level meters developed by the Acoustical Society of America. The curve labeled 40 phon in Figure 2 formed the basis of the A-weighting curve, which was later adopted by the International Standards Organization and is still in use today. The phon is a unit of perceived loudness level for pure tones.

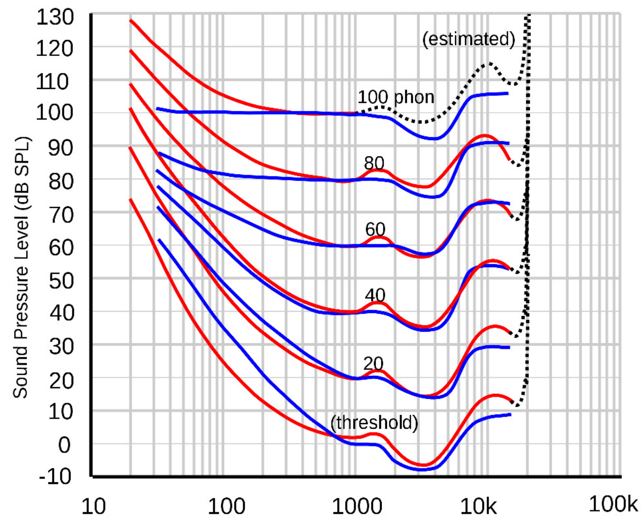


Figure 2. Equal-loudness contours from Fletcher–Munson (blue) and the later ISO 226:2003 revision (red).

During the 50+ years since its widespread adoption, various researchers have pointed out problems associated with using A-weighted sound pressure level measurements as the basis of studies in noise control, prevention of noise-induced hearing loss, and noise nuisance. Nevertheless, A-weighted sound pressure level is almost universally used in noise regulations around the world. As a result, A-weighting is a mandatory requirement for sound level meters in IEC 60261-1, whereas other frequency weightings are optional.

Temporal variation

An important distinction for this discussion is the one between steady and non-steady sounds. Steady sounds are those which do not vary in level or spectral content with time. Examples include pure tones, or a white noise signal at a constant level. Non-steady sounds are those that vary in level or frequency content, or both—for example, a cymbal crashing, music, speech, or a pulsing alarm signal.

To measure non-steady sound levels, sound level meters use one or more different types of averaging. A conventional sound level meter uses what is referred to as *exponential time weighting*. This is essentially a moving average process, where samples of the RMS level are

1 Fletcher, H. and Munson, W.A. Loudness, its definition, measurement and calculation. Journal of the Acoustical Society of America 5, 82-108 (1933).
2 The Fletcher-Munson curves, which were redetermined by Robinson and Dadson in 1956, eventually became the basis of international standard ISO 226.

weighted exponentially, such that the most recent sample has a greater influence on the indicated level than previous samples. Exponential averaging is equivalent to a simple RC filter. Standard sound level meters include exponential time weighting with two RC time constants: Fast (F) time weighting has a time constant of 125 milliseconds, and Slow (S) time weighting has a time constant of 1 second.

The time constant represents the time that it takes for the step response of the system to reach 37% of its final indicated value. For example, when measuring a steady signal with exponential time weighting, if the signal is suddenly switched off, the indicated level will decay exponentially and approach zero asymptotically. With F time weighting, the indicated level would decrease to 37% of the original level in 125 milliseconds. With S time weighting, the same decay would take 1 second. Sound pressure levels measured with exponential weighting are often designated L_F or L_S, depending on the time constant used, and L_{AF} or L_{AS} when measured with an A-weighting filter.

Another type of averaging used in some sound level meters is *integrating-averaging*. In this case, the meter integrates the data over the measurement time to determine a quantity called the *time-average sound level* in IEC 61672. This is commonly known as the *equivalent continuous sound level* (L_{eq}) and is designated as $L_{eq(t)}$. The "t" in parentheses represents the averaging time, which in practice could vary from a fraction of a second to hours or even days. L_{eq} measurements have the advantage that short duration measurements can be combined to determine the L_{eq} for a longer averaging time. For example, if you measure and record $L_{eq(1s)}$ continuously for a 24-hour period, you can combine these short term measurements to determine the $L_{eq(24hr)}$. When an A-weighting filter is used to measure L_{eq}, the results are specified as $L_{Aeq(t)}$.

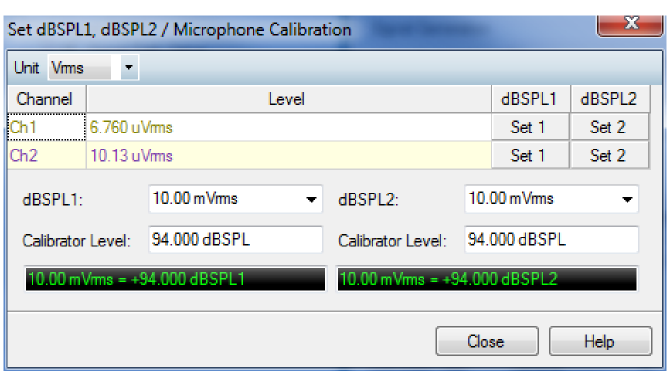


Figure 3. Microphone calibration.

Measuring sound pressure level in APx500

The units of sound pressure level, dB re 20 µPa, are sometimes referred to simply as dBSPL. In APx500, there are two independent reference levels (dBSPL1 and dBSPL2) that can be used to specify the sensitivity of measurement microphones, so that measured results can be displayed directly in dBSPL.

The *Microphone Calibration* window (Figure 3) can be accessed either from the *Set dBSPL* button in the *Reference Levels* measurement, or from the *Mic Calibration* button in the *Acoustic Response* measurement. You can either enter the sensitivity of the microphone in mV/Pa as printed on its calibration certificate (94 dBSPL is equal to 1 Pa), or use a microphone calibrator to set the dBSPL reference directly. The APx500 context sensitive help explains how to do this more fully.

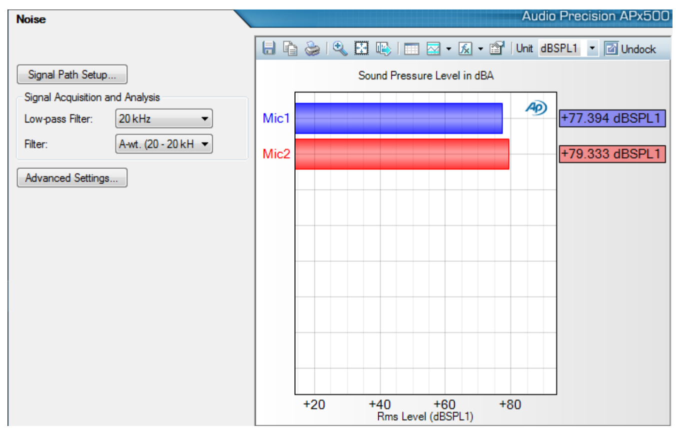


Figure 4. APx Noise measurement configured to measure A-weighted dBSPL.

For steady sounds, the APx500 *Noise* measurement (Figure 4) can be used to measure sound pressure level directly in dBSPL, once the microphone sensitivity has been entered as described above. To measure A-weighted sound pressure levels, simply set the *Filter* control to **A-wt (20 – 20 kHz)**. If you wish, you can change the graph title so that it is obvious that measured noise levels are A-weighted.

When troubleshooting noise problems, the ability to measure the noise spectrum can be a powerful analysis tool. Figure 5 shows an FFT noise spectrum measured with the APx500 *Signal Analyzer* measurement while the microphone was positioned close to the ventilation port of a PC with a noisy fan. APx500 offers additional tools in the form of *Derived Measurement Results*, which can be used for such things as fractional octave smoothing, finding minimum and maximum levels, and more.

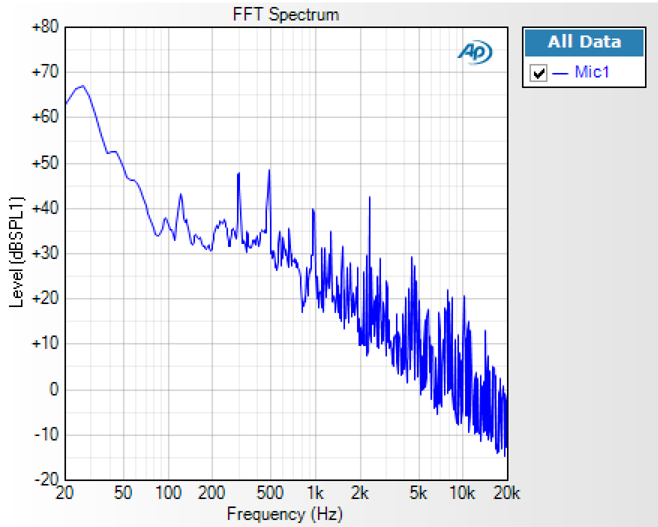


Figure 5. FFT spectrum of an acoustic noise signal.

It should be noted that virtually all measurements in APx500 are derived from FFT analysis of measured waveforms. In the case of the *Noise* measurement, the system acquires a waveform for a specified acquisition time of up to 5 seconds, performs an FFT of the entire acquisition, applies the selected filters in the frequency domain, and sums the FFT bins to determine the RMS level. Hence, the result is actually $L_{eq(t)}$, where "t" is the length of the acquisition.

While FFT-based measurements are preferable for typical audio applications, where the audio analyzer's generator is used to stimulate a device under test, this approach can be problematic for measurement of non-steady sound levels. This is because for non-stationary or transient signals, results of the FFT analysis depend strongly on the portion of the varying signal that is acquired and analyzed. As a result, there is no way to accurately reproduce with FFT analysis a time domain measurement like L_{AF} of a time-varying signal.

The APx Sound Level Meter Utility

The APx Sound Level Meter Utility allows an APx500 Series audio analyzer to measure sound pressure levels of non-steady noise sources (Figure 6). The utility provides A-weighted filtering of the signal in the time domain, and exponential time weighting with time constants of F (125 ms) and S (1 s), yielding metrics that would typically be measured with a sound level meter. A custom time constant is also available, in which case you can specify an arbitrary value for the time constant. If unweighted noise levels are preferred, the filter can be set to **None**.

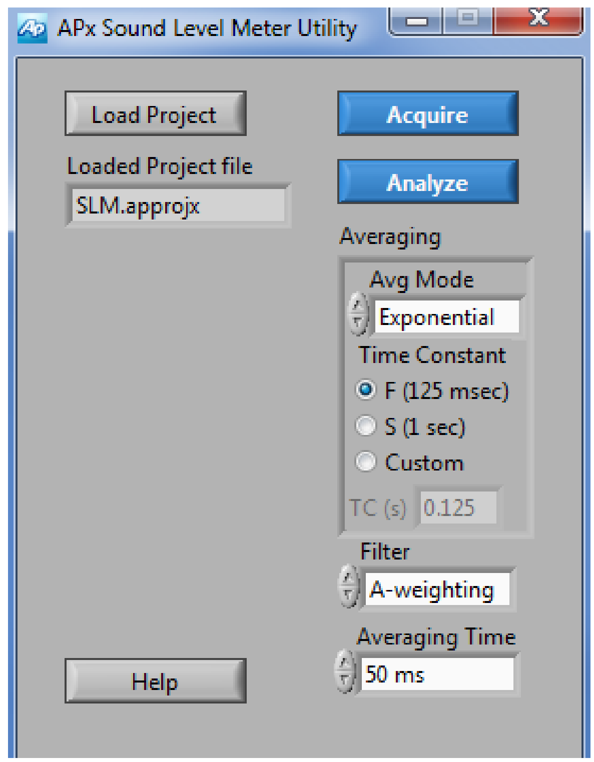


Figure 6. The APx Sound Level Meter Utility.

In addition to exponential time weighting, a linear averaging mode is also available. In this mode, the utility calculates the RMS level versus time, using the *Averaging Time* that is set. The calculated sound pressure level is then the short duration equivalent continuous sound level $L_{eq(t)}$, where *t* = *Averaging Time*.

The A-weighting filter design is based on a MATLAB® script published by Christophe Couvreur[3], implemented as a digital IIR filter. At a sample rate of 48 kHz, the filter meets the requirements for Class 1 accuracy specified in IEC 61672-1 (Figure 7). Currently, 48 kHz is the only sample rate supported by the utility.

3 http://www.mathworks.com/matlabcentral/fileexchange/69-octave

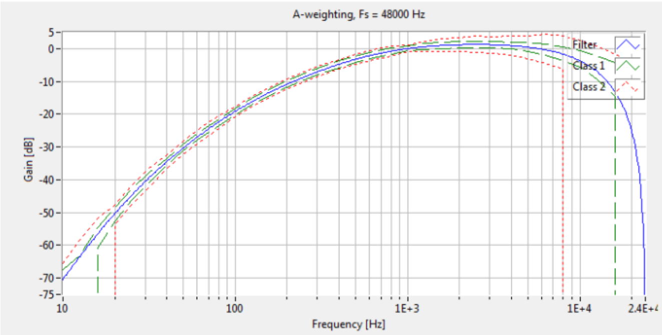


Figure 7. A-weighting filter implementation in the utility compared to IEC Class 1 and Class 2 limits.

Using the utility

The utility works with the APx500 *Measurement Recorder* and analyzes the .wav file it records. To use it, set up the audio analyzer as you normally would to conduct a measurement with *Measurement Recorder*. Be sure to check **Save to File** and to specify the saved file settings using the *File Settings* button. When ready, click the **Acquire** button in the utility to start the acquisition.

Figure 8. The utility's analysis window.

When the acquisition is complete, click the **Analyze** button. This will cause the utility to get the raw waveform data for each measurement channel from the recorded .wav file(s) and insert it into the analysis window (Figure 8), where it is processed according to the selected *Averaging*, *Filter*, and *Averaging Time* control settings. Clicking the **OK** button in the bottom right corner of the analysis window will import the data back into *Measurement Recorder* and display it on the *Level versus Time* graph, where it is selected as the currently displayed data set.

In the analysis window, the raw waveform, filtered waveform, and averaged data for the selected channel are displayed, and you can select the averaging mode, time constant, filter, and averaging time. There are also controls to select which channel is displayed, whether the averaged data is displayed in dB or linear units, and the dB reference. By default, the dB reference is set to match the dBSPL1 reference in APx500, but a control to change that independently is provided. The utility also positions the cursor in the *Averaged Data* display at the maximum value.

Demonstration waveforms

A default APx project file is included with this utility for demonstration purposes. It includes attached waveforms that are played by the arbitrary waveform generator and then routed back into the analyzer. Therefore, the *Signal Path* is set to **Analog Unbalanced** for output and **Loopback** for input (if using an APx585 or 586, connect cables from *Analog Unbalanced Outputs* 1 & 2 to *Analog Unbalanced Inputs* 1 & 2). In normal usage of the utility, a measurement microphone is connected to the *Analog Unbalanced Inputs*, and the *Analog Unbalanced Outputs* are unused. Any APx project file that includes the *Measurement Recorder* measurement can be configured and used with the utility.

The waveform shown in the analysis window in Figure 8 is one of the demonstration waveforms included with the project. This waveform contains two sine waves: each two seconds long with a level of -20 dBFS. There are two seconds of silence before, after, and between the two sine waves. The first sine wave has a frequency of 100 Hz and the second has a frequency of 1 kHz. Two features in the analysis window are worth noting:

1. The level of the first waveform is lower than the second in the *Filtered Data* and *Averaged Waveform* displays. In fact, it is exactly 19.1 dB lower. This difference corresponds to the specified gain of the A-weighting filter (-19.1 dB at 100 Hz and 0 dB at 1 kHz).

2. The rise and decay of the *Averaged Data* display is indicative of exponential time weighting. Note the constant slope of the *Averaged Data* curve after the signal is instantly switched to zero level. The slope of

this line corresponds to the selected time constant of 125 milliseconds. The signal decays to 37% of its original value (a change of about -8.7 dB) in 125 ms, yielding a slope of -69.5 dB/s.

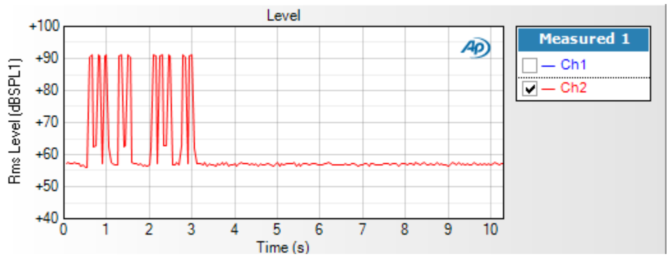


Figure 9. Pulsating alarm SPL measured at speaker terminals.

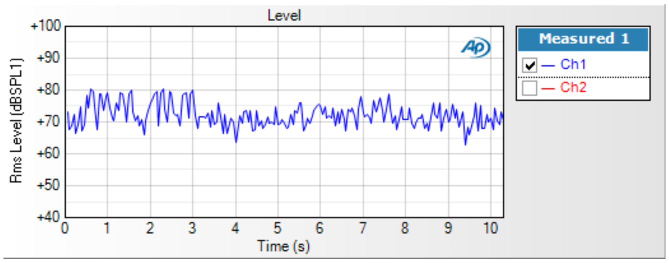


Figure 10. Pulsating alarm SPL as picked up by the measurement microphone.

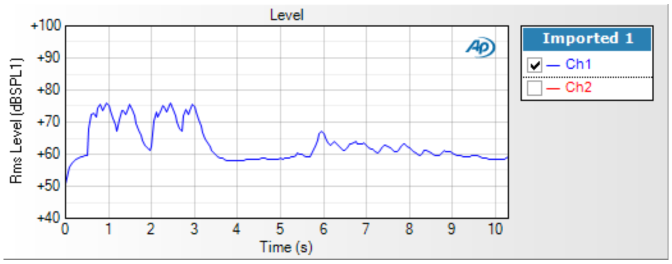


Figure 11. Pulsating alarm SPL from Figure 10, after being A-weighted and exponentially averaged by the utility.

A second waveform included in the project file is a recording of an alarm signal, which consists of ten bursts of a complex tone approximately 100 ms long and spaced 80 – 250 ms apart. Figure 9 shows channel 2, which is the voltage measured across the speaker terminals.

Channel 1, shown in Figure 10, is the sound pressure level measured with a microphone positioned near the device's loudspeaker. When measured with *Measurement Recorder*, which does not include an A-weighting filter, the alarm, which occurs in the first 3 seconds, is barely distinguishable from the background noise. However, after the utility has applied A-weighting and averaging, and has inserted the data back into the *level versus time* graph (Figure 11), the tone bursts are clearly visible.

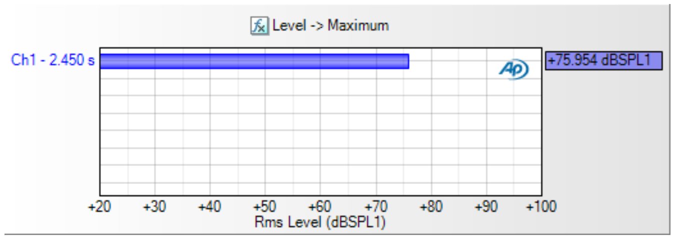


Figure 12. L_{AFmax} determined using APx500's Derived Measurement Results.

Once the data is back in the APx500 graph, you can take advantages of features such as annotation, exporting images or data, and printing. *Derived Measurement Results* also add a host of additional analysis features. For example, the curve in Figure 11 represents L_{AF} (the A-weighted sound pressure level measured with F time weighting) versus time. In Figure 12, a *Maximum* derived result has been created for this data set. This shows at a glance that L_{AFmax} is 75.9 dBA and it occurred at 2.45 seconds after the start of the measurement.

Related Downloads:

Technote 113: Measuring Sound Pressure Level with APx (this Technote):
http://ap.com/display/file/539

APx Sound Level Meter Utility (includes this Technote 113 and the sample APx project file):
http://ap.com/display/file/540

Related Resources:

IEC Standards (IEC 61672, Electroacoustics – Sound Level Meters (part 1: specifications):
http://webstore.iec.ch/webstore/webstore.nsf

ANSI Standards (S1.4, Specification for Sound Level Meters)
http://webstore.ansi.org

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