Sound Pressure Measurement

Objectives:

1. Become familiar with hardware and techniques to measure sound pressure
2. Measure the sound level of various sizes of fan modules
3. Calculate the signal-to-noise ratio of acoustic measurements
4. Investigate the noise of the fan modules as a function of size and operating speed

Background

Sound can be described as a tiny variation in air pressure, detected by some receiver. Typical sound pressure amplitudes range from a few micropascals to a few hundred Pascals that fluctuate around atmospheric pressure (~101,000 Pascals). This variation in air pressure travels as a longitudinal wave, i.e. a compression wave, which propagates throughout the medium (air) at a velocity called the sound speed. In air at 20 °C, the speed of sound is about 344 m/s. In general, the tiny variation of pressure oscillates very rapidly in time around the atmospheric pressure. In the audio-range, these oscillations occur between 20 and 20,000 times per second. In other words, audible sound is made up of frequencies between 20 and 20,000 Hz. The frequency components in this wave are perceived as different tones and pitches by the human ear. 20 Hz corresponds to a very low pitch while 20,000 Hz corresponds to a very high pitch. Most noises are composed of many different frequencies.

The Brüel and Kjær Type 2236 Precision Integrating Sound Level Meter, as shown in Figure 1, can be used to measure the magnitude of sound pressure levels (SPL's). Common reasons that sound pressure level is measured include:

1. The evaluation of noise emanating from a product to determine how the product could be made less annoying to customers. For example, automotive companies perform these measurements to determine the sound that a car engine produces when it is running. The Department of Transportation takes SPL readings to measure noise coming from interstate highways.
2. The measurement of noise to evaluate if an environment is damaging to human ears. For example, a factory must evaluate the noise level that the workers are exposed to while working. This involves comparison of the levels measured to published standards by regulating agencies such as OSHA (Federal Occupational Safety and Health Administration).

One of the most important components of a sound level meter is the microphone Most of the high quality sound level meters have a condenser microphone. This type of microphone is constructed with a thin metal foil stretched like a drumhead over a frame. It is always protected by a guard because it is fragile. When the slight changes in pressure, caused by a sound wave, reach this diaphragm, the diaphragm moves relative to a rigid metal backplate. The capacitance (the ability of two closely spaced metal plates to store energy in an electric field) changes with the movement of the diaphragm. This change in capacitance is converted to a milli-volt signal, which goes on to a preamplifier. The preamplifier in a microphone-sound measurement system is always positioned as close as possible to the microphone to reduce the magnitude of the electrical noise in the signal. The signal then goes to other amplifier stages and then to an analogto-digital converter where it is quantified digitally. The sound signal can then be filtered, measured and stored by the data acquisition systems introduced earlier in this manual

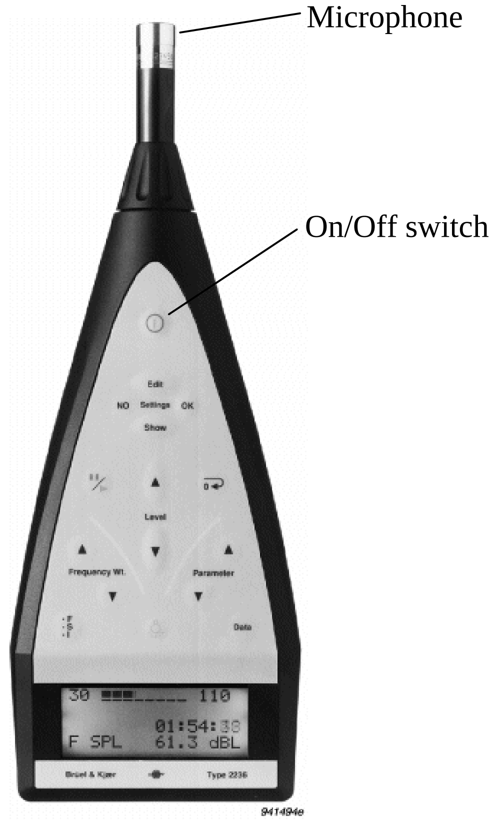


Figure 1. Bruel & Kjær 2236 Precision Integrating Sound Level Meter. The location of the microphone and on/off button are shown. The LCD display at the bottom shows the sound level reading (Bruel & Kjær, 2003).

A condenser microphone can be of two different types. A free-field-response microphone is made to measure sound which is coming from one direction only. This would be the case if the measurement was being made outside or in a place where the sound did not reflect off any walls or other objects, such as an anechoic chamber. An anechoic (*an* – means ‘no’, *echoic* – echo) chamber is a room that has been prepared to minimize sound reflections off its walls so that a free-field microphone can be used. A random-incidence microphone is made to measure sound equally from all directions. This type of microphone must be used when reflections or multiple sound sources produce an environment where there is more than one path from the source(s) to the microphone. Sometimes a random-incidence microphone is used to measure sound which is coming from one direction only. In this case, the microphone axis should be turned at 70° to 80° from the line between the source and the microphone to minimize the error from using the wrong type of microphone. The measurements will have reduced accuracy if the proper type of microphone is not used.

The Brüel and Kjær Type 2236 Sound Level Meter comes with a type 4188 free-field response microphone. As an additional feature, the sound level meter comes with an adapter that makes the free-field microphone have a similar response to a random-incidence microphone. This is to avoid the cost of having two separate microphones for each measurement situation.

Sound pressure level is the measurement of sound strength on a logarithmic scale (this is the base ten logarithm, not the natural logarithm, “Ln”). The measurement was derived at Bell Labs in the 1930’s as a way of comparing the power of a sound level to a reference value. The unit of a “Bell” was first defined there. Because a “Bell” is a small value, sound level meters read in deci-Bells (dB), or more commonly spelled, “decibels”. The following equations define the sound pressure level of a certain noise.

$$L = \text{SoundPressureLevel} = 10 \log(\text{Signal Power/Reference Power}) \qquad (1)$$

Because the power of the acoustic signal is related to the square of the RMS pressure, this equation can be rearranged and written as:

$$L = 20 \log(P_{\text{rms}}/P_{\text{reference}}) \qquad (2)$$

where P represents pressure. The reference value for sound in air, $P_{\text{reference}}$, is 20 micropascals. The RMS pressure is the square root of the integral over a certain time period of the square of the pressure divided by the time period.

$$P_{\text{rms}} = \left[\frac{1}{T} \int_0^T p^2(t)dt \right]^{1/2} \qquad (3)$$

An integrating sound level meter measures p(t) with a microphone and computes P_{rms} (over a set duration T) and finally displays the sound pressure level in dB.

Because the sound pressure level is measured on a logarithmic scale, the sound pressure level resulting from two combined sources is not the algebraic sum of the two separately measured levels. One must convert the individual measurements back to a linear value and then add them together before converting back to a logarithmic scale. If three sources with individually measured sound pressure levels of $L_1, L_2,$ and L_3 are combined, the total sound pressure level will be L_T as shown in equation 4:

$$L_T = 10 \log \left[10^{\frac{L_1}{10}} + 10^{\frac{L_2}{10}} + 10^{\frac{L_3}{10}} \right] \qquad (4)$$

Sound level meters often have the ability to measure noise with four weighting networks. These weighting networks are labeled as dBA dBC, dBL and dB xxxHz on the Brüel and Kjær Type 2236 meter. The dBL setting measures the loudness of all frequencies with the same sensitivity. The A and C weightings allow the user to measure the lower frequencies of sound with less sensitivity than others.. The purpose of this unequal sensitivity is to model the response of the human ear. The human ear is not as sensitive to the lower frequencies in its range. When the signal is passed through a filter that has a similar frequency response as the human ear, you can obtain a measure of the loudness of the sound that corresponds to the perception of an average human listener. The A network, measured in dBA is the most common weighting used today.

In addition to weighting networks, “bandpass” filters are often used to measure the amplitude of sound within a certain range of frequencies. Although this range could be any width, sound is usually separated into octaves. An octave is an interval of frequencies where the highest frequency is twice the lowest frequency. For example, the span of frequencies between 707 Hz and 1414 Hz are one octave because 2 x 707 = 1414. A filter, which rejects all frequencies lower than 707 Hz and also rejects frequencies above 1414 Hz, is called a single octave bandpass filter. This filter would normally be designated by the frequency in the center of its range, in this case 1000 Hz. The relationship of the center frequency to the upper and lower cutoff frequencies is given in the following equations. This is only for a one-octave bandpass filter

$$f_{lower} = \left[\frac{1}{\sqrt{2}}\right] f_{center} \qquad f_{upper} = \sqrt{2} f_{center} \qquad (5)$$

Therefore, an octave bandpass filter with a center frequency of 1000 Hz, filters out all frequencies from the sound being measured except those between 707 Hz and 1414 Hz. Using several bandpass fitters, one can investigate which frequencies are contributing the most to the total SPL. The B&K 2236 meter has one octave band pass filters with center frequencies of 31.5, 63, 125, 250, 500, 1000, 2000, 4000, and 3000 Hz.

INSTRUCTIONS

I. Calibration (perform this ONLY if directed to by your instructor!) It is important to calibrate a sound level meter each time it is used, so ask your instructor if the meter needs to be calibrated.

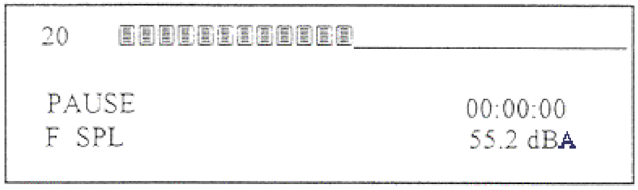
1. While the sound level meter is mounted on a tripod, carefully remove the random incidence adapter if it is installed on the microphone and put it in a safe place. Turn the sound level meter on by pressing the button, which is labeled with a red circle and a short green line.
2. Insert the microphone into the Brüel and Kjær calibrator. Make sure that the microphone is inserted until it bottoms out inside the calibrator.
3. Press the Frequency Wt. buttons to set the weighting to dBL. If the OVL letters are shown on the display, press the reset button (which has a curved line and an arrow).
4. Turn on the calibrator by pressing its On/off button.
5. The display should now show approximately the emitted value of the calibration signal.
6. Press the Show button twice to start the calibration procedure.
7. Press Edit once to continue.
8. Press OK to check the status of the present calibration. (If an error occurs, go back to step 3 and set the weighting to dBL)
9. Press OK after the screen returns with the results of its test
10. Press OK again to exit the calibration routine.
11. Turn off the calibrator and remove it from the microphone.

II. Measurement of Sound Pressure Levels. The sound level meter will be used to determine the effect of combining multiple sound sources, transmission loss, and signal-to-noise ratio (SNR). If directed by your instructor, take your measurements inside the anechoic chamber. Keep in mind for this case that the measurements can be considered as 'free-field'.

1. Set the parameters of the sound level meter to the following.

Power Button	On	Press the button with the red circle
Displayed Parameter	SPL	Press the Parameter buttons
Level	20-100 dB	Press the Level buttons
Time Weighting	F (fast)	Press the button labeled .F.S.I
Frequency Weighting	dBA	Press the Frequency Wt. button

The display should appear as.



2. Place the sound source, the fan module Number 1, in the acoustic experiment box. If the acoustic experiment box is used inside the anechoic chamber, leave the lid to the box open. Turn on the fan measure, and measure the sound level in dBA using at least three different speeds by varying the voltage level on the power supply. For the lower voltages, if the fan doesn't start by itself, try giving the blade a push to overcome the starting torque.

   A LabView VI (SPL Data Logger.vi) has been written to help automate the process of gathering data. There is a computer in the soundproof room adjacent to the anechoic chamber. Double-click the icon for the VI to start it.

3. Measure the background noise in acoustic box by quietly watching the display while no sound sources are turned on. This is the background noise level. Take a background measurements before each measurement when the fan is turned on.

4. Repeat step 1 and 2 for fan modules number 2 and 3.

5. Calculate the signal to noise ratio of the SPL levels.

<u>**III. Measurement of Fan Speed**</u> After you have taken SPL measurements, you will measure the fan speeds using a reflective optosensor and an oscilloscope. A small piece of white tape has been placed on one of the fan blades for each fan. When the optosensor is placed near the tape, a voltage pulse can be observed on the oscilloscope once for each revolution of the fan blade.

1. Ask your instructor to verify that the optosensor is wired properly.

2. Set the oscilloscope to make a frequency measurement (which will be in Hz, so you'll need to convert to RPM).

3. Measure the fan speeds at the voltages you used when making the noise measurements.

References

Bruel & Kjær 2236 Precision Integrating Sound Level Meter data sheet, http://www.bksv.com/pdf/Bp1535.pdf, 2003.

Data Sheet

Acoustics: Measurement of Sound Pressure Level

Measurement Data

Description of Fan 1: ____

Description of Fan 2: ____

Description of Fan 3: ____

Source (Fan Module #): ______

Voltage Level	Fan Speed	dBA	Background Noise	SNR

Source (Fan Module #): ______

Voltage Level	Fan Speed	dBA	Background Noise	SNR

Source (Fan Module #): ______

Voltage Level	Fan Speed	dBA	Background Noise	SNR