7.   INSTRUMENTATION FOR NOISE MEASUREMENTS

7.1   PURPOSES OF MEASUREMENTS

There are many reasons to make noise measurements. Noise data contains **amplitude, frequency, time** or **phase** information, which allows us to:

1.   Identify and locate dominant noise sources
2.   Optimize selection of noise control devices, methods, materials
3.   Evaluate and compare noise control measures
4.   Determine compliance with noise criteria and regulations
5.   Quantify the strength (power) of a sound source
6.   Determine the acoustic qualities of a room and its suitability for various uses
       and many, many more…..

7.2   PERFORMANCE CHARACTERISTICS

The performance characteristics of sound measurement instruments are quantified by:

Frequency Response - Range of frequencies over which an instrument reproduces the correct amplitudes of the variable being measured (within acceptable limits).
       Typical Limits over a specified frequency range:
                   Microphones $\pm$ 2dB
                   Tape Recorders $\pm$ 1 or $\pm$ 3 dB
                   Loudspeakers $\pm$ 5 dB

Dynamic Range - Amplitude ratio between the maximum input level and the instrument's internal "noise floor" (or self noise). All measurements should be at least 10 dB greater than the noise floor. The typical dynamic range of meters is 60 dB, more is better.

Response Time - The time interval required for an instrument to respond to a full scale input, (limited typically by output devices like meters, plotters)

7.2   SOUND LEVEL METERS

The primary tool for noise measurement is the Sound Level Meter (SLM). The compromises with sound level meters are between accuracy, features and cost. The precision of a meter is quantified by its type (see standards IEC 651-1979, or ANSI S1.4-1983 for more details)

Type 0   Laboratory reference standard, intended entirely for calibration of other sound level meters
Type 1   Precision sound level meter, intended for laboratory use or for field use where the acoustical environment can be closely controlled. (ballpark estimate: ~$5000)
Type 2   General purpose, intended for general field use and for recording noise level data for later frequency analysis (~$500)
Type 3   Survey meter, intended for preliminary investigations such as the determination of whether noise environments are unduly bad.   (~$50, Radio Shack)

Table 7.1 Principal allowable dB tolerance limits on sound level meters (ref ANSI S1.4-1983)

Characteristic	Type 0	Type 1	Type 2
Accuracy at calibration frequency to reference sound level	±0.4 dB	±0.7 dB	±1.0 dB
Accuracy of complete instrument for random incidence sound	±0.7	±1.0	±1.5
Maximum variation of level when the incidence angle is varied by ±22.5°	±0.5 (31-2000 Hz) ±1.5 (5000-6300 Hz) ±3 (10000-12500 Hz)	±1.0 (31-2000 Hz) +2.5, -2 (5000-6300 Hz) +4, -6.5 (10000-12500 Hz)	±2.0 (31-2000 Hz) ±3.5 (5000-6300 Hz) * (10000-12500 Hz)
Maximum allowable variation of sound level for all angles of incidence	±1.0 (31-2000 Hz) ±1.5 (5000-6300 Hz) ±3 (10000-12500 Hz)	+1.5, -1(31-2000 Hz) ±4 (5000-6300 Hz) +8, -11 (10000-12500 Hz)	±3(31-2000 Hz) +5, -8 (5000-6300 Hz) * (10000-12500 Hz)

 * none specified

The most basic SLM will have an analog or digital output of A-weighted (or unweighted) sound pressure. Additional features can include octave or 1/3 octave filters, frequency weighting networks (A,C, D, Lin), time averaging, and interface to a PC for data storage and plotting.

Response Time: Sound level meters commonly have settings of:
          "Fast" -   200 msec response time (or sometimes 125 msec)
          "Slow -   1 sec response time
The Slow setting will smooth out transients and provide a steady, average value. The Fast setting is useful if you are interested in the transient events.

If a noise is an impulse (very short duration, fast rise and decay) such as an explosion, or impact, neither of these settings will give an accurate reading. Impulsive sounds measured on a fast or slow setting may be up to 30 dB less than the true peak level. More expensive meters also have a "Peak" response option. Peak measurements (of impulsive sounds) are made by storing the peak level (which might be reach in a few milliseconds) in a capacitor, then displaying the peak value of the meter.

Weighting Networks
Frequency weighting networks (implemented with electronic filters) are built into sound level meters to provide a meter response that tries to approximate the way the ear responds to the loudness of pure tones.   These weighting curves are directly derived from the Fletcher/Munson equal loudness contours. See section 4.2 for more information on weighting networks
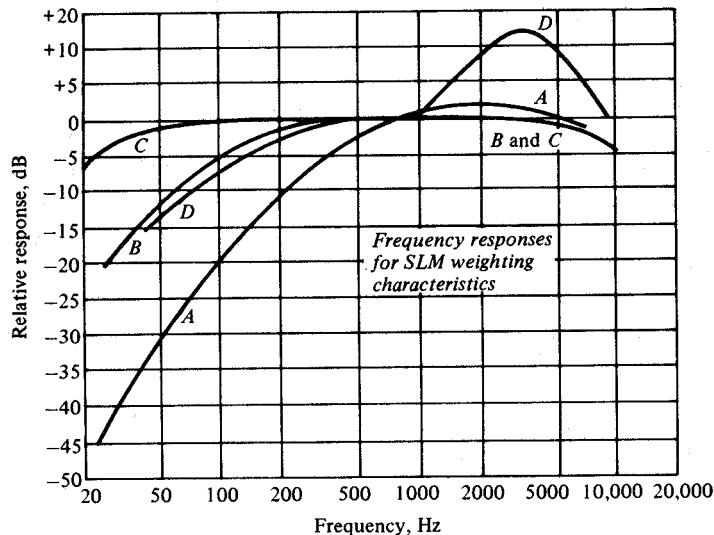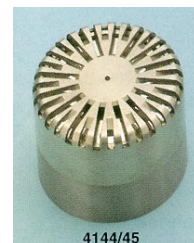
Figure 7.1   Frequency characteristics of weighting networks commonly found in sound level meters.

7.3   MICROPHONES

Microphones are available from a variety of suppliers (B&K, GenRad, Aco, etc) in a wide range of sizes (1" to 1/8" diameter). Most microphones used for precision measurements are of the

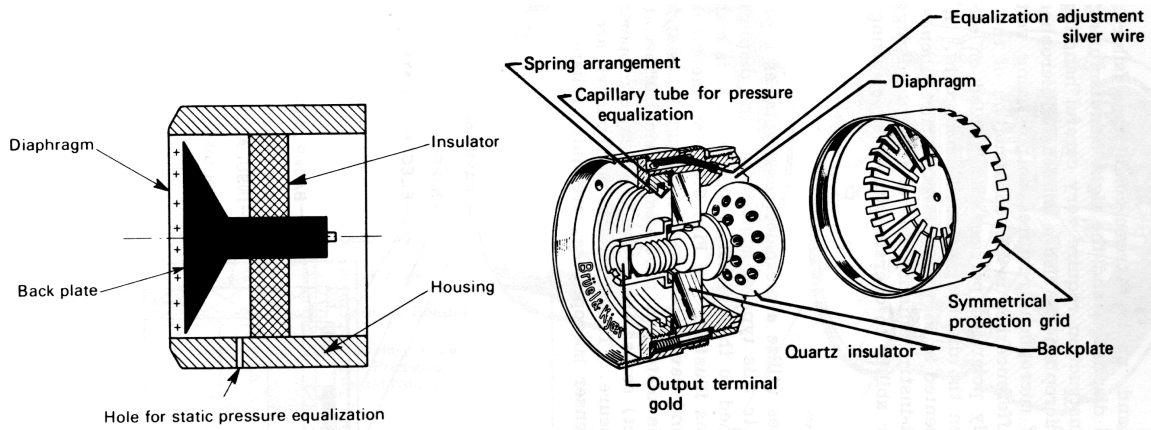condenser type. The construction of a condenser microphone is shown in Figure 7.2.

Figure 7.2 Schematic and cutaway views of a typical condenser microphone

The basic operating principle for a condenser microphone is: a thin diaphragm and the fixed back plate, separated by a thin air gap, form the two plates of a capacitor. Pressure fluctuations from incoming sound waves cause the diaphragm to vibrate, changing the air gap. This changes the capacitance, which is measured electronically and converted into a voltage by appropriate circuitry, usually contained in a separate unit called a pre-amplifier. Instrumentation grade microphones are specially designed to have negligible sensitivity to temperature and humidity, and have excellent long term stability (see Table 7.2).

Table 7.2 Specifications of general purpose B&K condenser microphones

Size	1/8″	1/4″	½″	1″
Model	4138	4135	4133	4145
Frequency response (± 2 dB)	6.5-140KHz	4-100KHz	4-40KHz	2.6-18KHz
Sensitivity (mV/Pa)	1.0	4.0	12.5	50
Temperature Coefficient (dB/°C)	-.01	-.01	-.002	-.002
Expected Long Term Stability at 20°C		>600 years/dB	>1000 years/dB	>1000 years/dB

Microphone selection depends on two primary parameters:

- **Sensitivity** - ratio of microphone output voltage to input pressure amplitude (in units of mV/Pa). In general, larger microphones have a greater sensitivity.

- **Frequency Response** - variation in sensitivity as a function of frequency (the ideal is a perfectly flat response). Frequency response is specified as a range over which the output signal deviates less than ±2 dB. Typical frequency response curves are shown in Figure 7.3. Smaller microphones have a wider frequency response. At high frequencies (when wavelength approaches the diameter of the microphone) diffraction effects occur which alter the frequency response. These effects are dependent on the incidence angle of the sound waves (see Figure 7.4).

The frequency response curve approaches flat for 90 degrees (grazing) incidence. Each microphone is supplied with calibration curves, which can be used to compensate for this diffraction effect at high frequencies (but most people don't). To minimize this error, use as small a microphone as possible.
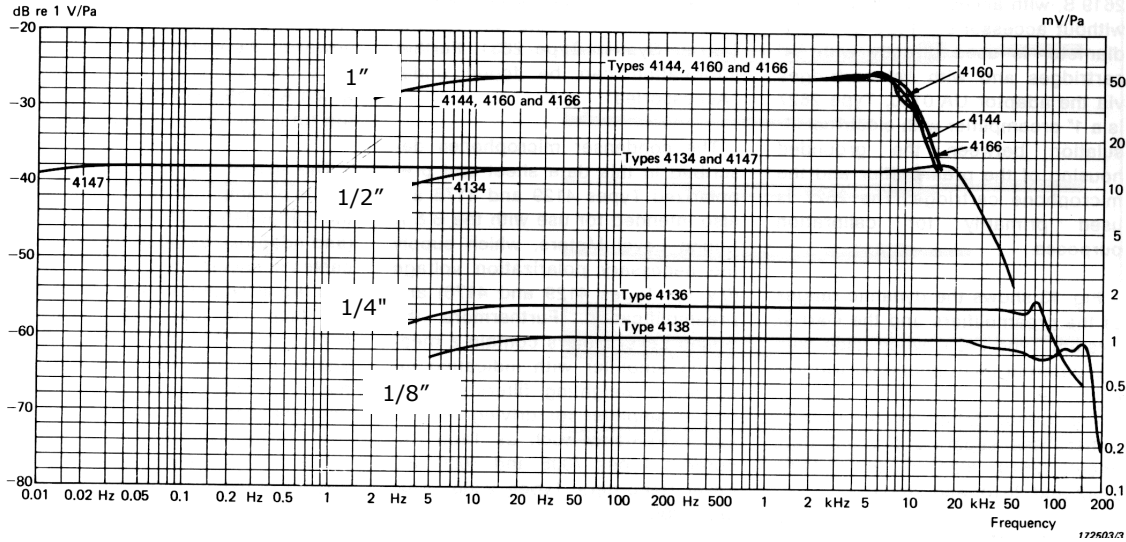

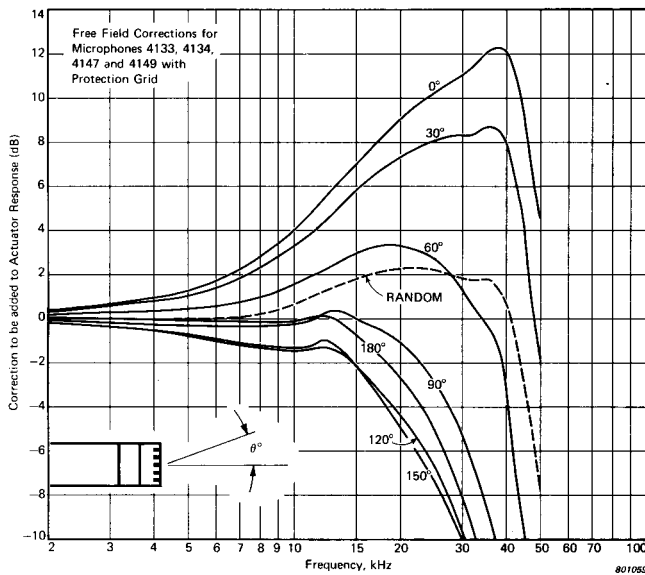Figure 7.3 Frequency response of B&K condenser microphones of various sizes using an electrostatic actuator

Figure 7.4 Directional characteristics of ½" condenser microphone

Microphone types

Pressure – designed to be used for coupler measurements, i.e. directly coupled to a test chamber

Random (diffuse field) – designed to give optimum frequency response for random incidence sound (equal probability of sound from all directions, such as in a reverberant chamber)

Free Field - designed to give optimum frequency response for sound from a particular incidence angle (usually 0 degrees)
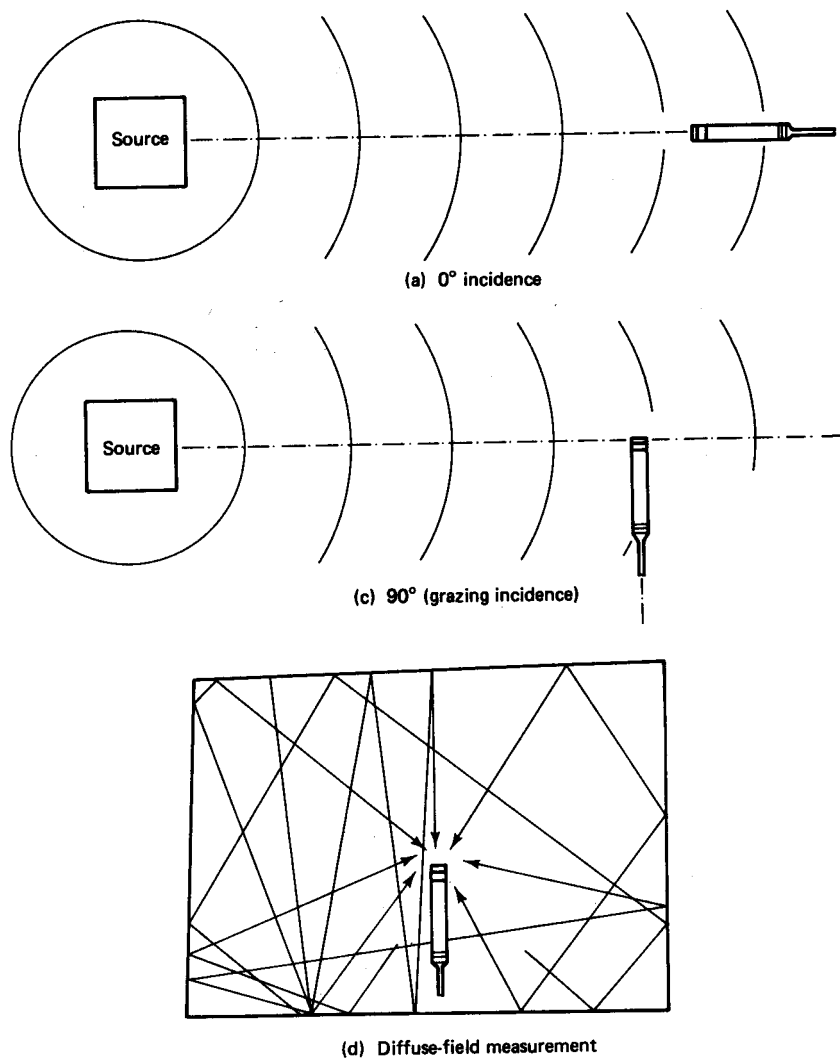

(a) 0° incidence

(c) 90° (grazing incidence)

(d) Diffuse-field measurement

Figure 7.5   Microphone orientation

7.5   FREQUENCY ANALYSIS (1/n Octave)

The most basic measurement any sound level meter can make is an overall dB level. This is a single number, which represents the sound energy over the entire frequency range of the meter. It provides no information about the frequency content of the sound. We can obtain information on the frequency content by using filters. The most common are octave band and 1/3 octave band filters. The most frequency detail is provided by FFT analysis.

Octave Band - Measures the total acoustical energy within the passband of a band pass filter. The term "octave" denotes a doubling in frequency. Hence, each octave band covers a frequency range of one octave. We refer to the octave band by its center frequency. The center frequencies of successive filters are separated by one octave. The preferred octave band center frequencies (by international standard) are: 31.5, 63, 125, 250, 500, 1000, 2000, 4000, 8000 and 16000 Hz. The shape of a typical octave filter is shown in Figure 7.4 below. The bandwidth of a filter is the width in frequency between the −3 dB points. This is an example of a constant percentage bandwidth filter. The width of octave filters progressively increases with frequency.   When plotted on a log scale, the shape of the band response is independent of frequency. The output of a percentage bandwidth filter is: dB/Bandwidth
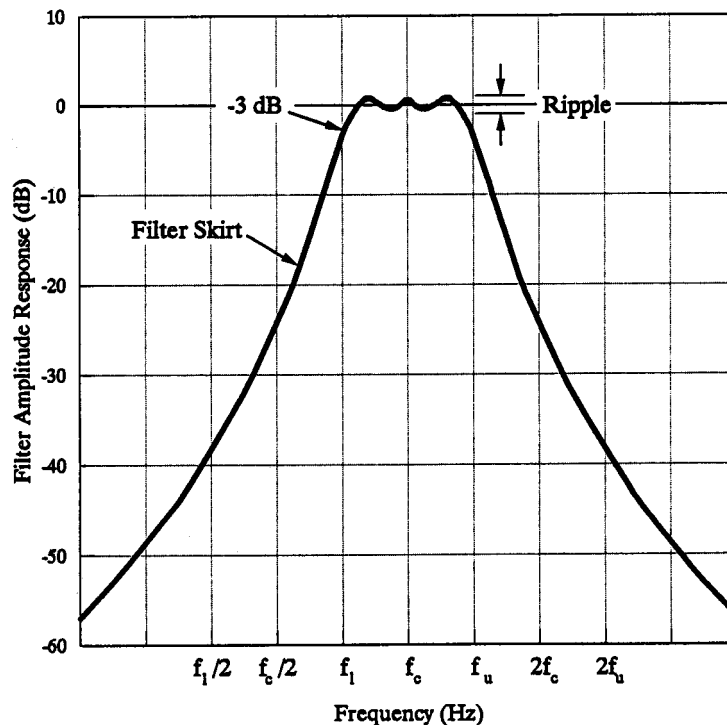

Figure 7.4   Characteristics of an octave band filter

An octave band filter is not a perfect bandpass filter (it is physically impossible to build one). There is a finite "rolloff" or skirt on each side of the band. As can be seen in Figure 7.5, adjacent filters overlap each other slightly.
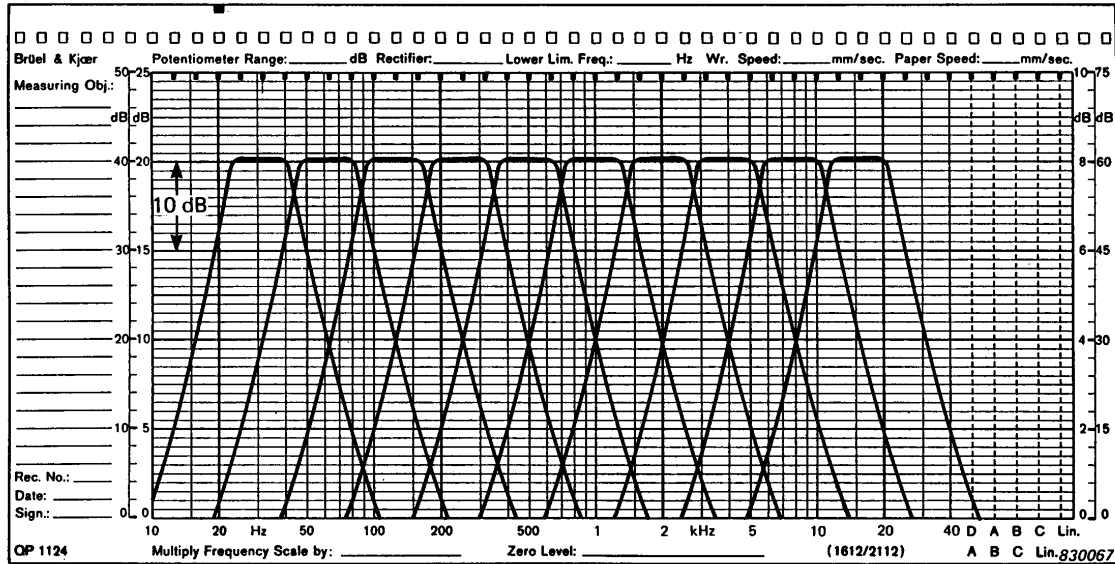

Figure 7.5 Complete filter characteristics for a typical octave band filter set

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1/1 Octave Filter Relationships:

$$f_l = 2^{-1/2f_c} \qquad f_u = 2^{1/2f_c}$$

$f_{c_{i+1}} = 2f_{c_i}$ = center frequencies of adjacent filters

f_l = lower cutoff frequency (to - 3 dB point), Hz

f_u = upper cutoff frequency (to - 3 dB point), Hz

One-third, one-tenth octave analysis - More detail is sometimes needed to obtain adequate frequency resolution, hence the need for bandwidths finer than one octave. The choice of filter bandwidth depends on the nature of the measured noise - is it broadband or does it have significant pure tones which you want to identify? Closely spaced pure tones will not be discovered by a wide bandwidth analysis.

1/3 octave - Each full octave is spanned by three 1/3 octave bands

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1/3 Octave Filter Relationships:

$$f_l = 2^{-1/6f_c} \qquad f_u = 2^{1/6f_c}$$

$f_{c_{i+1}} = 2^{1/3}f_{c_i} = 1.26f_{c_i}$ = center frequencies of adjacent filters

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1/10 octave – Each full octave is spanned by ten 1/10 octave bands (not very common anymore, due to the widespread availability of inexpensive FFT analyzers)

General 1/n Octave Filter Relationships:

$$f_l = 2^{-(1/2n)f_c} \qquad f_u = 2^{(1/2n)f_c}$$
$$f_{c_{i+1}} = 2^{1/n} f_{c_i}$$

The center frequencies and upper and lower limits of the octave and 1/3 octave filter bands are shown in Table 7.3 below.

Table 7.3   Center and cutoff frequencies (Hz) for standard full octave and 1/3 octave filters

Octave			1/3 Octave		
Lower limit	Center Freq	Upper limit	Lower limit	Center Freq	Upper limit
11 Hz	16 Hz	22 Hz	14.1 Hz	16 Hz	17.8 Hz
			17.8	20	22.4
			22.4	25	28.2
22	31.5	44	28.2	31.5	35.5
			35.5	40	44.7
			44.7	50	56.2
44	63	88	56.2	63	70.8
			70.8	80	89.1
			89.1	100	112
88	125	177	112	125	141
			141	160	178
			178	200	224
177	250	355	224	250	282
			282	315	355
			355	400	447
355	500	710	447	500	562
			562	630	708
			708	800	891
710	1000	1420	891	1000	1122
			1122	1250	1413
			1413	1600	1778
1420	2000	2840	1778	2000	2239
			2239	2500	2818
			2818	3150	3548
2840	4000	5680	3548	4000	4467
			4467	5000	5623
			5623	6300	7079
5680	8000	11360	7079	8000	8913
			8913	10000	11220
			11220	12500	14130
11360	16000	22720	14130	16000	17780
			17780	20000	22390

7.6 FFT ANALYSIS

The FFT = Fast Fourier Transform, is a narrow band, constant bandwidth analysis (the frequency resolution does not change over the frequency range)

FFT refers to the numerical algorithm used to calculate the Fourier transform in "real time" (in less time than it takes to acquire the actual data). In layperson's terms, the FFT determines the frequency content of a time signal.   The mathematical definition of a Fourier transform is:

$$X(f) = \int_{-\infty}^{+\infty} x(t)e^{-j2\pi ft}dt$$

The FFT algorithm discretizes this calculation. It requires a finite number of time data points, typically a power of 2, such as 512 (2^9) or 1024. It is a transformation from time to frequency.

TIME DOMAIN:
$x(i\Delta t)$ = amplitude at time intervals of Δt (seconds), N data samples

FFT

FREQUENCY DOMAIN:
$X(j\Delta f)$ = amplitude (complex) at frequency intervals of Δf (Hz), $\Delta f = 1/(N\Delta t)$ N/2 valid points

Figure 7.6 The Fast Fourier Transform

Some sample time data (induction noise from a 2.5L 4 cylinder engine), and its associated frequency spectrum obtained by FFT are shown in Figure 7.7.

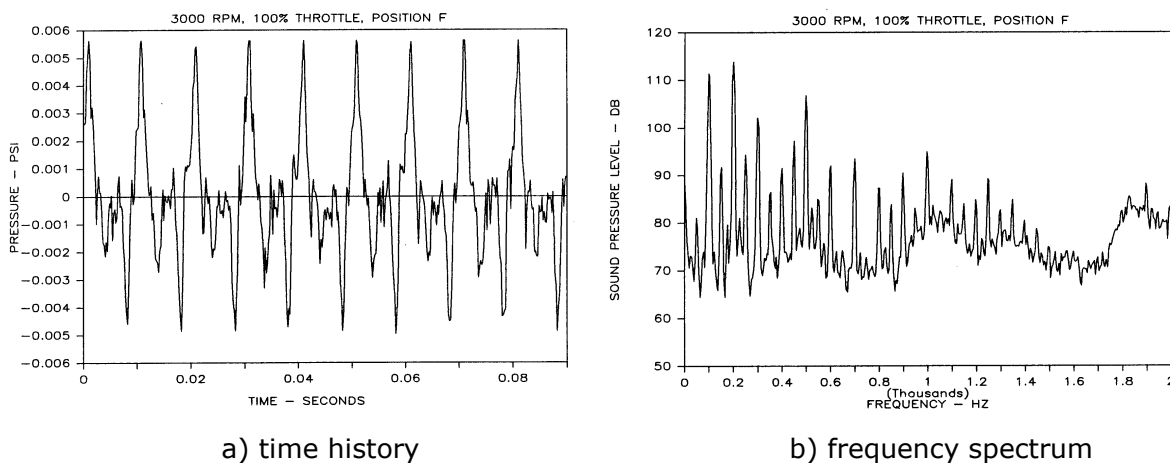a) time history   b) frequency spectrum

Figure 7.7   Induction noise data from a 4 cylinder engine running at 3000 RPM, wide open throttle

Useful things you can do with a FFT Analyzer:

- multiple channel analysis (transfer functions)
- signal averaging (in time or frequency)
- modal analysis (determine mode shapes)
- display time signals (like a digital oscilloscope)
- order tracking (for rotating equipment)
- correlation analysis
- mathematical operations (* / + -, integration, derivative)
- frequency zoom
- waterfall plots (spectral maps)
- store data to disk for later analysis and plotting
- data interface to MATLAB for additional calculations and display

Figure 7.8   Waterfall plot, showing variation in vibration spectrum with time
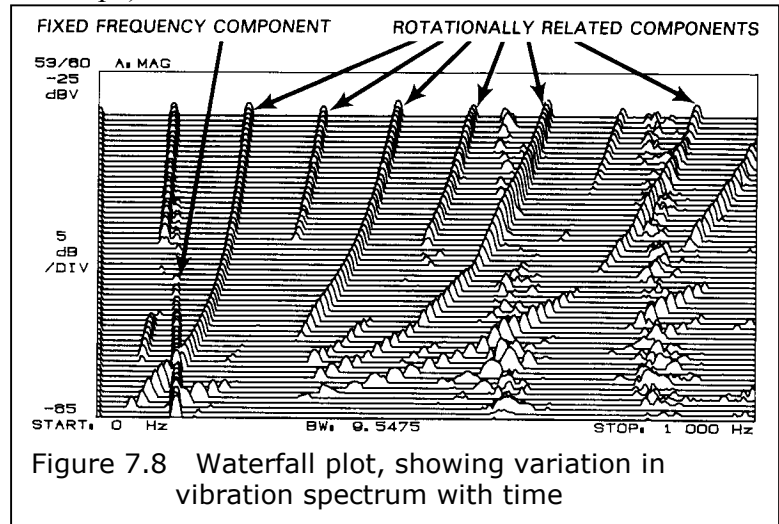
Things to watch out for:

- bad data, faulty transducers, poor signal/noise ratio
- choice of data window - use Hanning or Flat-top for steady, continuous data; Rectangular (sometimes called "Boxcar") for transient or impact data
- adequate signal levels (>10 dB over ambient, no overloads)
- sufficient frequency range to see everything of interest
- sufficient frequency resolution (only accurate to $\pm\Delta f/2$) – can be difficult to separate closely spaced peaks

7.7   COMBINATION OF TWO OR MORE FREQUENCY BANDS

A typical task is to determine the octave band level from 1/3 octave band measurements, or to calculate an overall level from individual octave or 1/3 octave bands. This is just like summing dB's from several sources as in Section 6.1.

The general expression for total pressure over the interval of interest is:

$$P_T^2 = \sum_{i=1}^{n} P_1^2 + \sum P_1^2 + P_2^2 + ...P_n^2$$

and in dB's: $10\log_{10}\left(\dfrac{P_T}{P_{ref}}\right)^2 = 10\log_{10}\sum_{i=1}^{n}\left(\dfrac{P_i}{P_{ref}}\right)^2 = 10\log_{10}\sum_{i=1}^{n}10^{L_{P_i}/10}$

The total power in an octave band is related to the individual 1/3 octave bands by:

$$P_{octave}^2 = P_1^2 + P_2^2 + P_3^2 \quad \text{(because we are adding energy, which is proportional to } P^2 \text{)}$$

$$L_P = 10\log_{10}\frac{P_{octave}^2}{P_{ref}^2} = 10\log_{10}\left[10^{L_{P_1}/10} + 10^{L_{P_2}/10} + 10^{L_{P_3}/10}\right]dB$$

Example 1:
The levels in the 400, 500 and 630 Hz 1/3 octave bands measure 72, 74, 68 dB. What is the octave band level for the 500 Hz band?
$$10\log_{10}\left(10^{7.2} + 10^{7.4} + 10^{6.8}\right) = 76dB$$

Example 2: Calculate the overall level for the following octave band measurements:
        (*Answer:* 102.6 dB)

Center Freq	dB
125	79
250	80
500	94
1000	100
2000	94
4000	94
8000	88

7.8   CONVERSION FROM ONE BANDWIDTH TO ANOTHER

There may be cases where you acquire data with one width of filter, and you later find you really needed to know the level over a different bandwidth. If you recorded all the 1/3 octave bands, it's easy to convert to full octaves, just logarithmically add the dB's as shown in the last section. However, what if you only measured one 1/3 octave band, but you desperately need to know the level over that entire octave? You have lost some information, but if you assume that the energy is uniformly distributed over the entire band (and there are no pure tones), then you can still make an estimate:
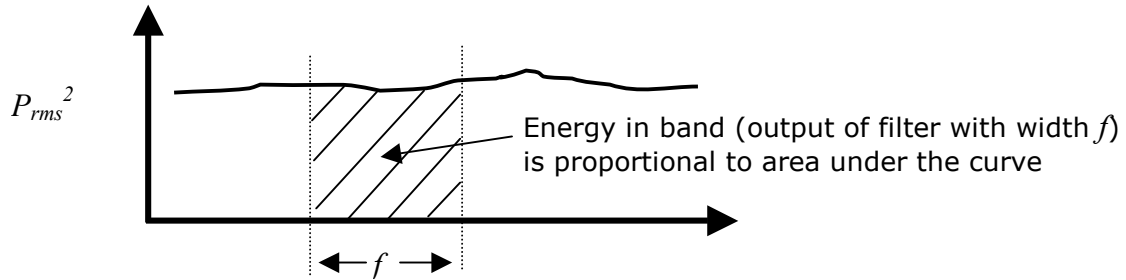

Figure 7.9 The output of a filter is determined by the amplitude and the bandwidth

First, let us define:
Spectrum level = Sound level (dB) read by an ideal analyzer with a 1 Hz bandwidth

We can relate spectrum level to levels taken with other bandwidths by:

$$P^2 = P_{SL}^2 f$$

where: P = rms pressure output of filter with bandwidth f
$\quad\quad P_{SL}$ = rms pressure in 1 Hz band

This implicitly assumes that the total energy in a given band is proportional to p^2 times the width of the band (i.e. the area under the p^2 curve). To convert a pressure measurement from one bandwidth to a different bandwidth:

$P_2^2 = P_1^2 \dfrac{f_2}{f_1}$ where: P_1 = rms pressure output over bandwidth f_1

$\quad\quad\quad\quad\quad\quad\quad\quad\quad P_2$ = rms pressure output over bandwidth f_2

and in terms of sound pressure level:

$$L_2 = L_1 + 10\log_{10}\frac{f_2}{f_1}$$

Example: the output of the 100 Hz 1/3 octave band is 58 dB, how much would be measured using the 125 Hz full octave band?

Answer: assuming that the level is uniform over the entire octave band,

$$L_2 = L_1 + 10\log_{10}\frac{f_2}{f_1} \qquad\qquad = 58 + 10\log_{10}\frac{3}{1} = 58 + 4.8 = 63 \ dB$$

7.9   MEASUREMENT OF PURE TONES WITH OCTAVE OR 1/3 OCTAVE FILTERS:

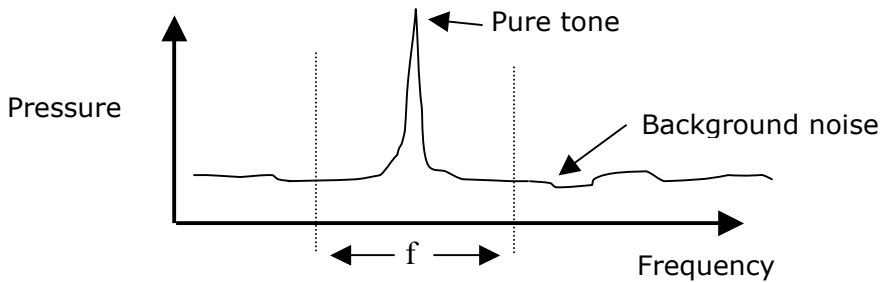If we have a prominent pure tone in addition to background noise:

Figure 7.10 A pure tone combined with background noise

The total power in the band is proportional to:

$$P_{band}^2 = \sum P^2 \ \text{ over the band}$$
$$= P_{pure\ tone}^2 + \text{ background noise } (\Sigma\ P^2)$$

Examples:
1. A pure tone of 80 dB at 120 Hz is combined with broadband noise which measures 75 dB in the 125 Hz band. What is the total SPL in the 125 Hz band? (*Answer:* 81.2 ≈ 81 dB)

2. A pure tone which measures 93 dB alone is combined with broadband noise which measures 80 dB by itself. What is the combined noise level? (Answer: 93.2 ≈ 93 dB)

Important Result: A pure tone will measure the same dB level on any bandwidth analysis, provided it is significantly higher (by at least 10 dB) than the background level.

7.10 SYNTHESIZING OCTAVE OR 1/3 OCTAVE BANDS FROM DISCRETE FFT DATA

We can use an FFT analyzer to measure and record a noise spectrum (the sound pressure amplitude at discrete, evenly spaced intervals of frequency, Δf). In some situations, it might be convenient to not have to haul around an octave band sound level meter too, so is there a way that we can use the FFT data to construct or "synthesize" the octave band data?   The answer is a qualified "yes".

Each FFT data point represents the output of a filter which is $\Delta f\,Hz$ wide. The overall energy over a frequency band larger than Δf is proportional to the area under the P_{rms}^2 curve:

$$P^2_{BAND} = \sum_{i=1}^{n} P_i^2$$

where: P_i^2 = mean square acoustic pressure of the ith FFT data point

n = number of FFT data points that fall within the bandwidth of the synthesized filter (see Table 7.2 for octave and 1/3 octave limits)

Δf = FFT frequency increment or "bin" size (Hz)

and in dB's:

$$dB_{BAND} = 10\log \sum_{i=1}^{n} P_i^2$$

Example: The noise from a portable circular saw is measured with a microphone and an FFT analyzer.   For comparison purposes, the 1/3 octave sound pressure levels were simultaneously measured using a Type I sound level meter.   The FFT spectrum of the microphone output from 0 – 5000 Hz is shown in Figure 11.   Calculate the 1/3 octave level in the 100 - 200 Hz bands.   The microphone output for a 250 Hz pistonphone calibrator is –15.2 dBV.   The FFT analysis particulars: N=800 points, Fs=12800 Hz, Δf=6.25 Hz, Hanning window.
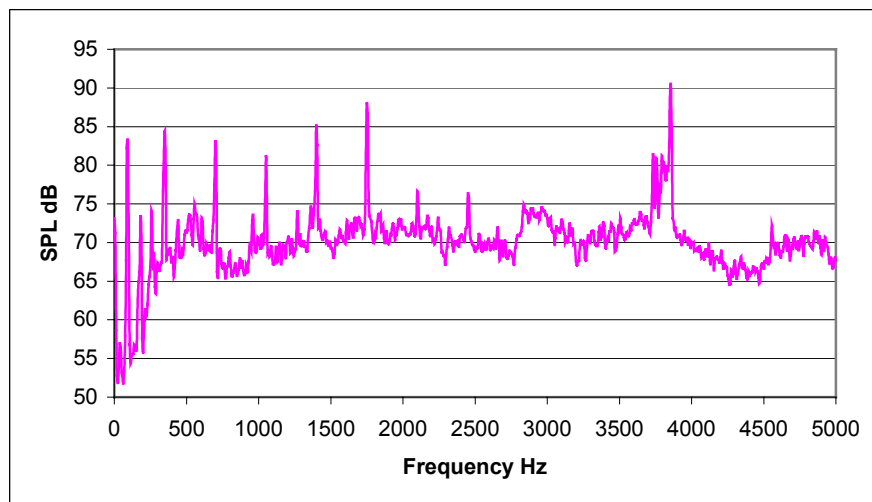

Figure 7.11 Noise spectrum for circular saw

Table 7.4 Partial FFT data for Circular Saw

Frequency (Hz)	Microphone Output- dBV	Calibrated DB SPL	Synthesized dB	Measured dB
81.25	-61.64	67.68		
87.50	-47.08	82.24		
93.75	-46.08	83.24		
100.00	-58.03	71.29	83.5	83.2
106.25	-72.53	56.79		
112.50	**-74.97**	**54.35**		
118.75	**-73.90**	**55.42**		
125.00	**-73.78**	**55.54**	**62.6**	**61.3**
131.25	**-73.73**	**55.59**		
137.50	**-72.51**	**56.81**		
143.75	-73.05	56.27		
150.00	-73.33	55.99		
156.25	-72.97	56.35		
162.50	-67.07	62.25	70.1	70.2
168.75	-64.37	64.95		
175.00	-63.12	66.20		
181.25	**-55.79**	**73.53**		
187.50	**-59.01**	**70.31**		
193.75	**-71.64**	**57.68**		
200.00	**-73.54**	**55.78**	**75.7**	**75.1**
206.25	**-70.26**	**59.06**		
212.50	**-67.87**	**61.45**		
218.75	**-68.99**	**60.33**		
225.00	-68.64	60.68		
231.25	-67.33	61.99		

The major problem with this approach is for the lower frequency bands. Since FFT's provide data at equal frequency intervals, the lowest octave bands may only encompass a few FFT points. This will degrade the accuracy of the "synthesized" band calculation.

7.11   WHITE NOISE AND PINK NOISE

White noise is defined as having the same amplitude at all frequencies (radio static, or a jet of compressed air are pretty good approximations). It is often used as a known input to a system, in order to determine the system's frequency response.

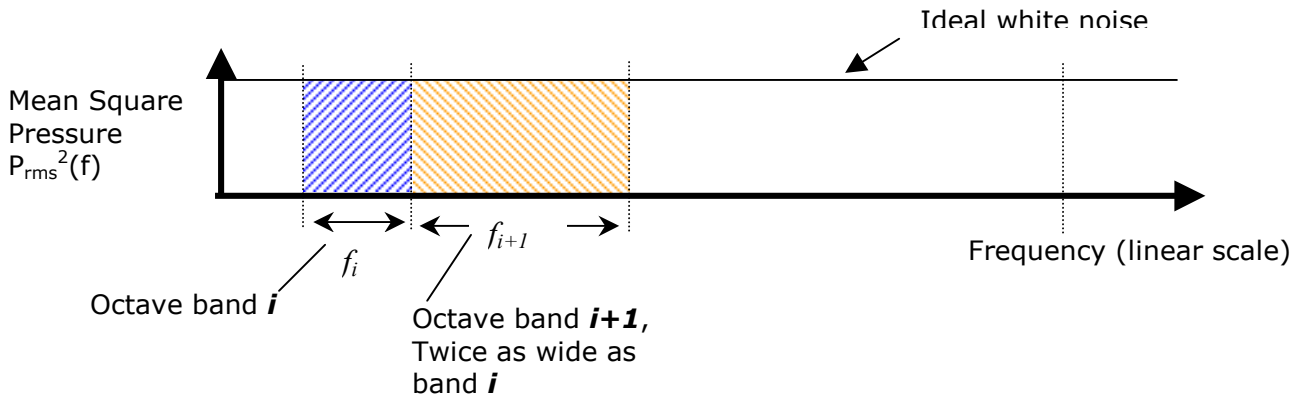What happens when white noise is measured using an octave band filter system?

Figure 7.12   White (random) noise has constant amplitude at all frequencies

The energy (and SPL) in band i is proportional to the area under the curve: $P_{rms}{}^2 f_i$

Each succeeding octave band doubles in width, therefore the total energy doubles for each succeeding band. This results in an increase in SPL of 3dB (10log2) for each successive octave band as displayed in Figure 13.
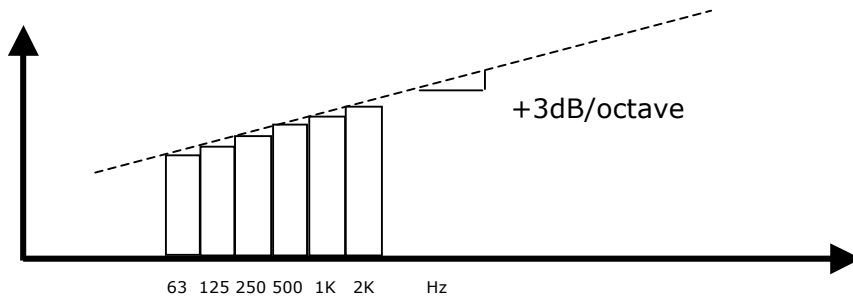

Figure 7.13   Output of octave band filters to white noise input - each successive octave band increases by 3dB

Pink noise is specifically designed to yield constant amplitude across all octave bands. On a linear scale, it decreases in amplitude as frequency increases in just the right amount (-3 dB/octave) to compensate for the increasing widths of the octave filters.

7.12 SOUND LEVEL METER OR FFT ANALYZER?

So which do you use? It depends on the application, and your budget. Here's a qualitative comparison:

Table 7.5 Comparison of Sound Level Meter to FFT Analyzer

Sound Level Meter (with octave band filters)	FFT Analyzer
minimal data (less to write down)	Produces lots of data (need to plot or record on disk)
crude frequency analysis	detailed frequency analysis, identification of noise sources
good for assessing compliance with regulations	can separate closely spaced sources
compact, portable, weather resistant	bulky, heavy, fussy
easy to learn, and use, limited features	complex, lots of features, storage, data manipulation
adequate for material selection	useful for many other purposes (vibration, transfer function analysis, etc)
relatively inexpensive $500 to $3000	$5K-15K