

## Abstract

The three sections in this Lecture Note give the most fundamental explanations of the concepts behind the use of simple frequency and level measurements of sounds.

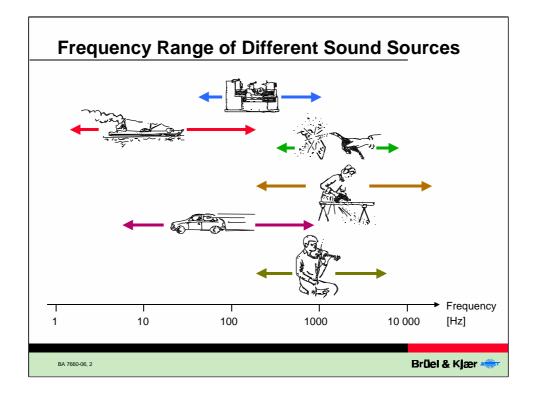
The basic instrument for sound analysis is a Sound Level Meter.

Copyright© 1998 Brüel & Kjær Sound and Vibration Measurement A/S All Rights Reserved

## LECTURE NOTE

English BA 7669-11



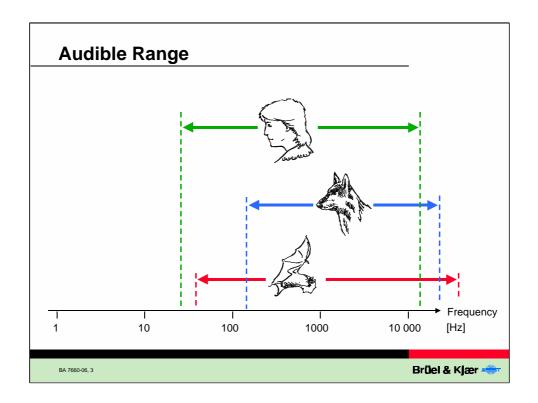


The frequency span of the sounds that typically surround human beings vary considerably.

Normally, young human beings can detect sounds ranging from 20 to 20000 Hz.

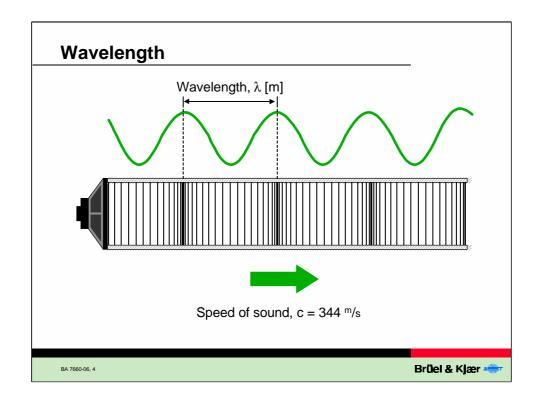
However, infrasounds in the range from 1 to 20 Hz and ultrasounds between 20000 to 40000 Hz can affect other human senses and cause discomfort.

Note that none of the illustrated sound examples cover the entire frequency range. That is why knowledge of frequency range and the need for frequency analysis is important.



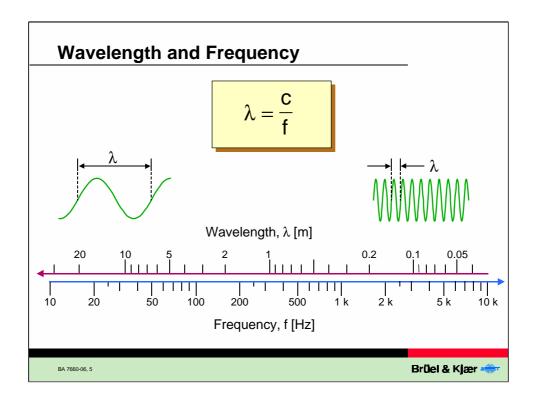
As can be seen, the range of perception of sound for humans - at a maximum for young people - goes from 20 to 20000 Hz.

With age, the human perception of the highest frequencies decreases gradually. When exposed to excessive noise levels, hearing can be damaged, causing reduced sensitivity for hearing low sound levels. The damage can also be restricted to distinct frequency ranges.



A sound signal from a loudspeaker mounted at one end of a tube will produce a sound wave that propagates forward at a speed of 344 m/s.

If the signal is a single sine signal the sound wave will consist of a number of pressure maxima and minima all separated by one wavelength.

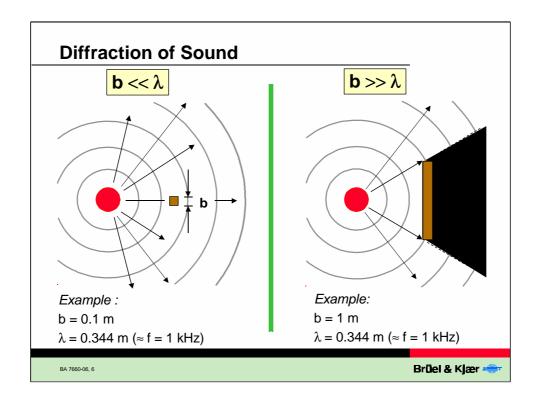


The wavelength, the speed of sound and the frequency are related according to the formula shown.

It is useful to have a rough feeling for which wavelength corresponds to a given frequency.

At 1 kHz the wavelength is close to 34 cm or one foot.

At 20 Hz it is close to 17 m, and only 1.7 cm at 20 kHz.

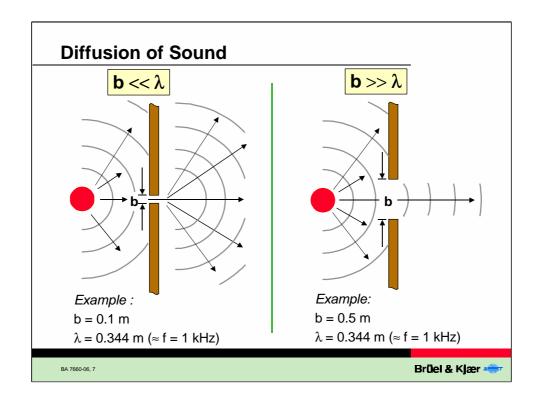


Objects placed in a sound field may cause diffraction.

But the size of the obstruction should be compared to the wavelength of the sound field to estimate the amount of diffraction.

If the obstruction is smaller than the wavelength, the obstruction is negible.

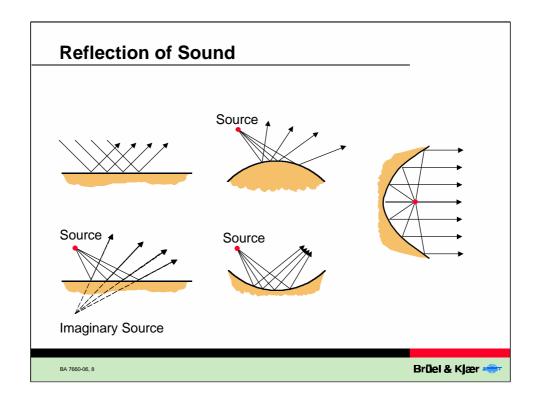
If the obstruction is larger than the wavelength, the effect is noticeable as a shadowing effect.



Diffusion occurs when sound passes through holes in e.g. a wall.

If the holes are small compared to the wavelength of the sound, the sound passing will re-radiate in an omnidirectional pattern similar to the original sound source.

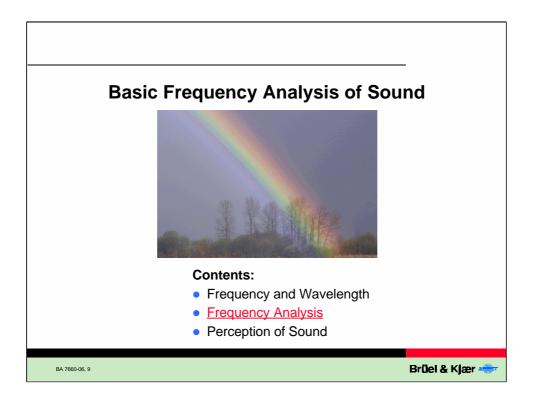
When the hole has larger dimensions than the wavelength of the sound, the sound will pass through with negligible disturbance.

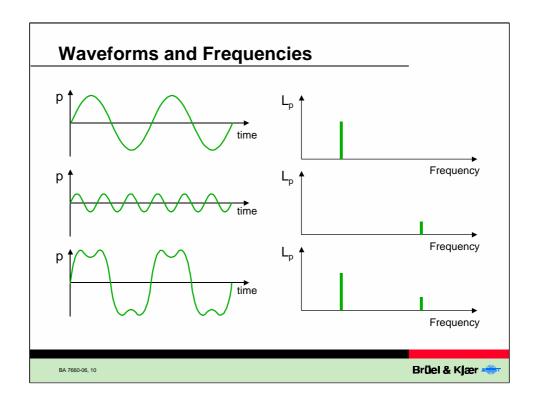


When sound hits obstructions large in size compared to it's wavelength, reflections take place.

If the obstruction has very little absorption, all the reflected sound will have equal energy compared to the incoming sound. This is one of the important design principles used when constructing reverberant rooms.

If almost all reflected energy is lost due to high absorption in the reflecting surfaces, the situation is close to what is found in an anechoic room.



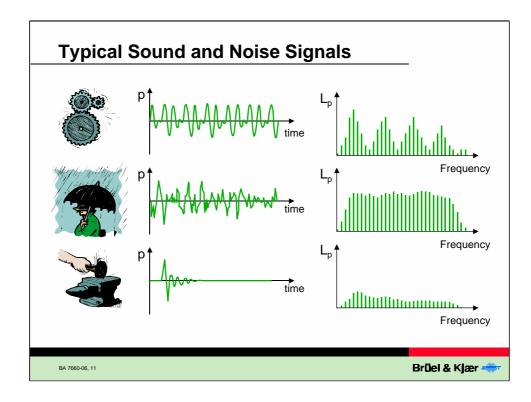


Three examples of the relationship between the waveform of a signal in the time domain compared to its spectrum in the frequency domain.

In the top figure a sine wave of large amplitude and wavelength is showing up as a single frequency with a high level at a low frequency.

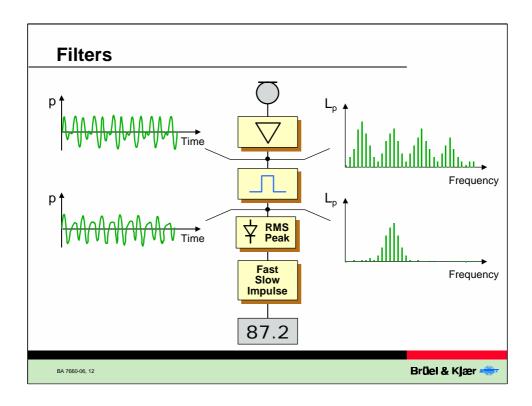
In the middle figure a low amplitude signal with small wavelength is seen to show up in the frequency domain as a high frequency with a low level.

At the bottom figure it is shown how a sum of the two signals above also in the frequency domain shows up as a sum.



Most natural sound signals are complex in shape. The primary result of a frequency analysis is to show that the signal is composed of a number of discrete frequencies at individual levels present simultaneously.

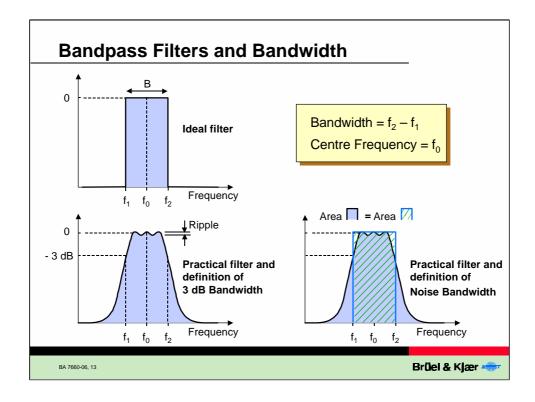
The number of discrete frequencies displayed is a function of the accuracy of the frequency analysis which normally can be defined by the user.



To analyze a sound signal, frequency filters or a bank of filters are used. If the bandwidths of these filters are small a highly accuracy analysis is achieved.

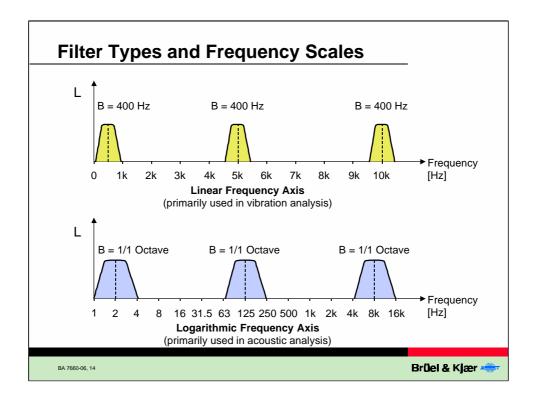
The signal flow chart shown illustrates the elements in a simple sound level meter.

On top is a microphone for signal pick up. Then a gain amplification stage followed by a single frequency filter - here shown as an ideal filter. In the following we will look at real filters. After filtering follows a rectifier with the standardised time constants Fast, Slow and Impulse and the signal level is finally converted to dB and shown on the display.



Ideal filters are only a mathematical abstraction. In real life, filters do not have a flat top and and vertical sides. The departure from the idealised flat top is described as an amount of ripple. The bandwidth of the filter is described as the difference between the frequencies where the level has dropped 3 dB in level corresponding to 0.707 in absolute measures.

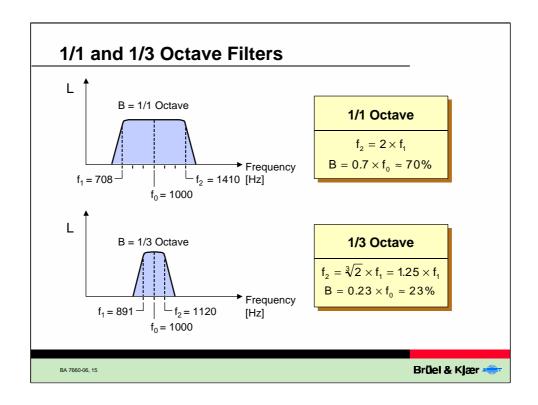
It is useful to define a Noise Bandwidth for a filter. This corresponds to an ideal filter of the same level as the real filter, but with its bandwidth (Noise Bandwidth) set to leave the two filters with the same "area".



The two most used filter banks are:

Filters that have the same bandwidth e.g. 400 Hz and displayed using a linear frequency scale. This is what normally is a result of a (FFT) Fast Fourier Transform analysis. Constant bandwidth filters are mainly used in connection with analysis of vibration signals.

Filters which all have the same constant *percentage* bandwidth (CPB filters) e.g.1/1 octave, are normally displayed on a logarithmic frequency scale. Sometimes these filters are also called relative bandwidth filters. Analysis with CPB filters (and logarithmic scales) is almost always used in connection with acoustic measurements, because it gives a fairly close approximation to how the human ear responds.

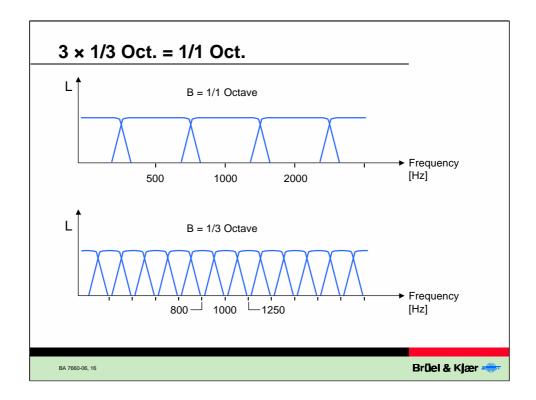


The widest octave filter used has a bandwidth of 1 octave. However, many subdivisions into smaller bandwidths are often used.

The filters are often labeled as "Constant Percentage Bandwidth" filters. A 1/1 octave filter has a bandwidth of close to 70% of its centre frequency.

The most popular filters are perhaps those with 1/3 octave bandwidths. One advantage is that this bandwidth at frequencies above 500 Hz corresponds well to the frequency selectivity of the human auditory system.

Filter bandwidths down to 1/96 octave have been realised.

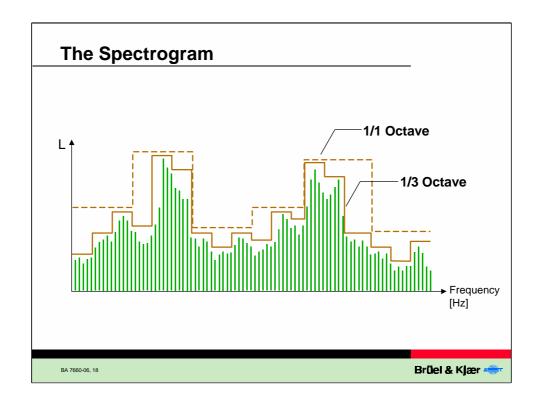


One advantage of constant percentage bandwidth filters are that e.g. two neighbouring filters combine to one filter with flat top, but with double the width.

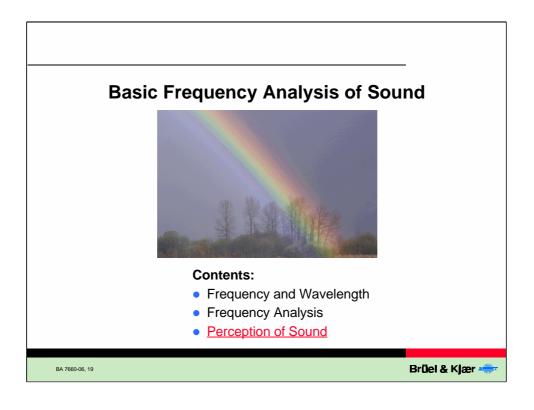
Three 1/3 octave filters combine to equal one 1/1 octave filter.

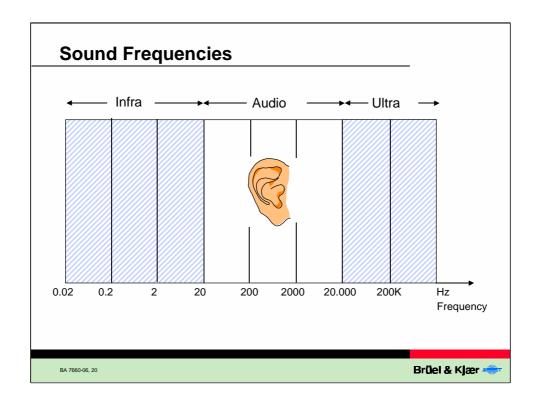
Band No.	Nominal Centre	Third-octave	Octave
	Frequency Hz	Passband Hz	Passband Hz
1 2 3 4 5 6	1.25 1.6 2 2.5 3.15 4	$1.12 - 1.41 \\ 1.41 - 1.78 \\ 1.78 - 2.24 \\ 2.24 - 2.82 \\ 2.82 - 3.55 \\ 3.55 - 4.47$	1.41 – 2.82 2.82 – 5.62
27 28 29 30 31 32	500 630 800 1000 1250 1600	447 - 562 562 - 708 708 - 891 891 - 1120 1120 - 1410 1410 - 1780	355 – 708 780 – 1410
40	10 K	8910 – 11200	11.2 – 22.4 K
41	1.25 K	11.2 – 14.1	
42	16 K	14.1 – 17.8 K	
43	20 K	17.8 – 22.4 K	

List of Band No., Nominal Centre Frequency, Third-octave Passband and Octave Passband.

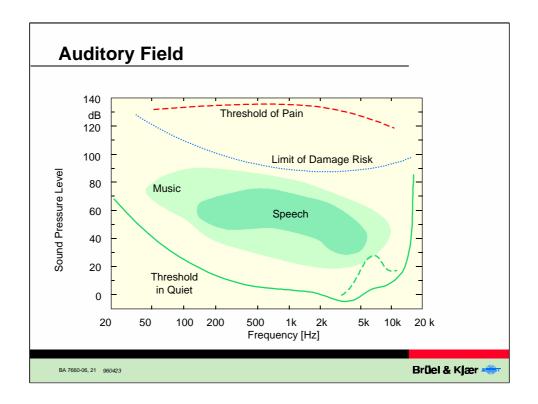


A detailed signal with many frequency components show up with a filter shape as the dotted curve when subjected to an octave analysis. The solid curve shows the increased resolution with more details when a 1/3 octave analysis is used.





Only the sounds in the frequency range from 20 to 20000 Hz can be perceived by the human ear and auditory system.

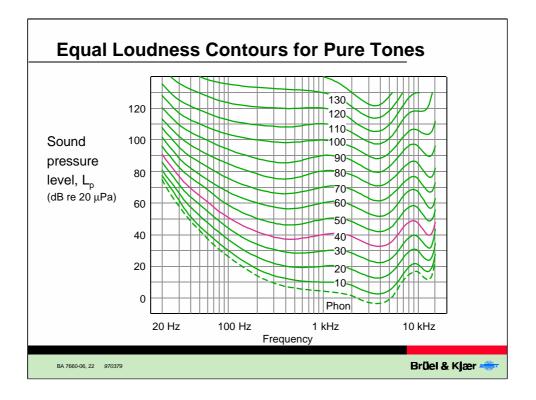


This display of the auditory field illustrates the limits of the human auditory system.

The solid line denotes, as a lower limit, the threshold in quiet for a pure tone to be just audible.

The upper dashed line represents the threshold of pain. However if the Limit of Damage Risk is exceeded for a longer time, permanent hearing loss may occur. This could lead to an increase in the threshold of hearing as illustrated by the dashed curve in the lower right-hand corner.

Normal speech and music have levels in the shaded areas, while higher levels require electronic amplification.



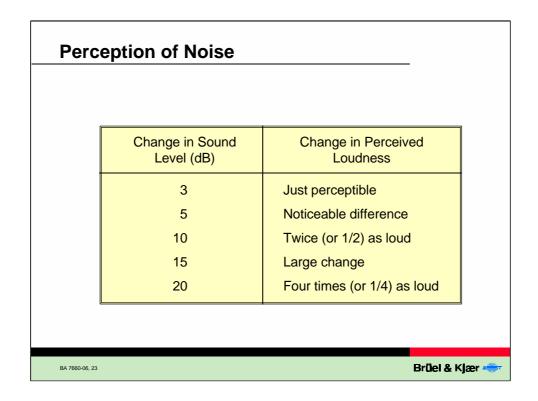
Here are shown the normal equal loudness contours for pure tones. The dashed curve indicates the normal binaural minimum audible field.

Note the very non-linear characteristics of the human perception. Almost 80 dB more SPL is needed at 20 Hz to give the same perceived loudness as at 3-4 kHz.

This observation together with frequency masking - limitations in the ears capability to discriminate closely spaced frequencies at low sound levels in the presence of higher sounds - is the foundation for the calculation of the loudness of stationary signals.

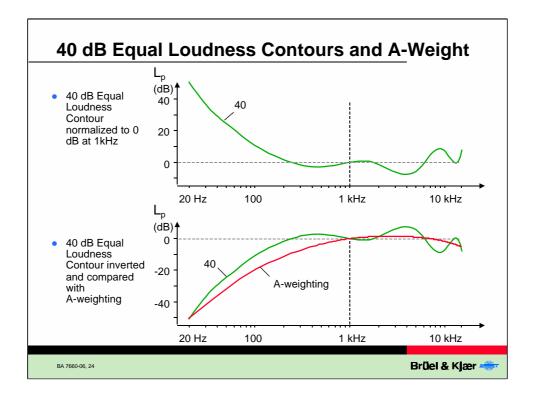
Loudness of non-stationary signals also needs to take the temporal masking of the human perception into account.

A correct calculation of these loudness values is crucial for all the following metrics calculations such as Sharpness, Fluctuation Strength and Roughness.



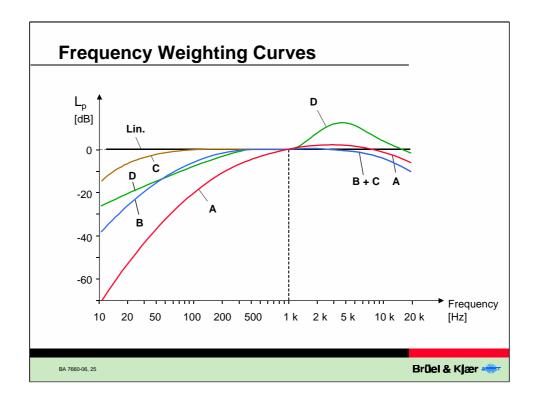
This table illustrates how much of a level change in dB there is needed to give different changes in perceived Loudness.

This data applies to frequencies around 1 kHz. At higher and lower frequencies, much larger changes in sound level are needed for similar changes in perceived loudness.



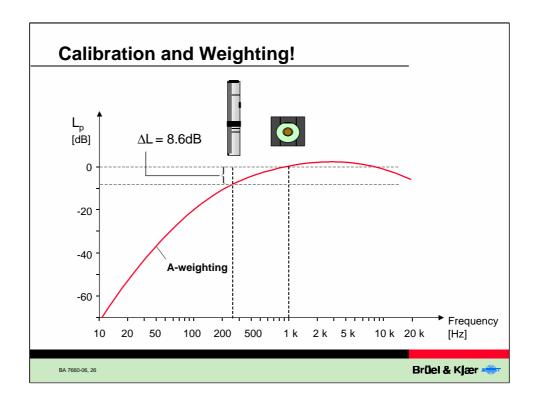
Top figure is the equal loudness contour for 40 dB at 1 kHz.

The popular A-weighting at the bottom is shown in comparison to the inverted 40 dB equal loudness contour.



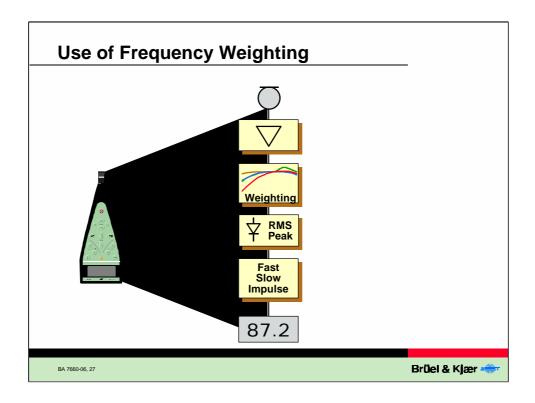
The A-weighting, B-weighting and C-weighting curves follow approximately the 40, 70 and 100 dB equal loudness curves respectively.

D-weighting follows a special curve which gives extra emphasis to the frequencies in the range 1 kHz to 10 kHz. This is normally used for aircraft noise measurements.

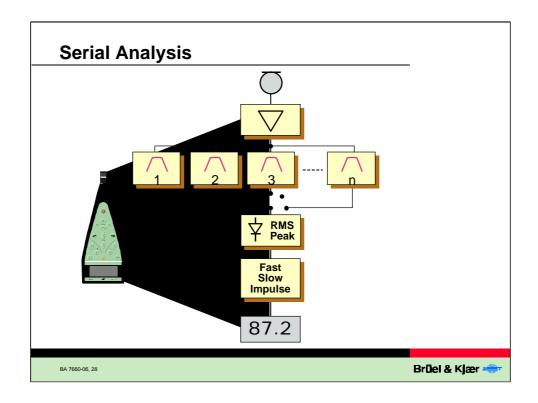


Please be careful when calibrating a system with a weighting filter switched in.

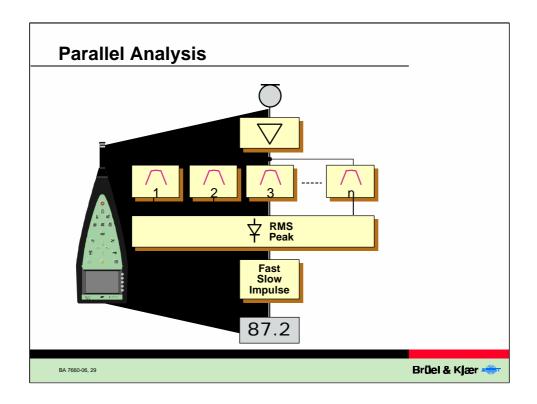
Using a pistonphone which has a calibration frequency of 250 Hz you get 8.6 dB less signal reading than using a sound level calibrator with a test frequency of 1000 Hz if an A weighting filter is switched in.



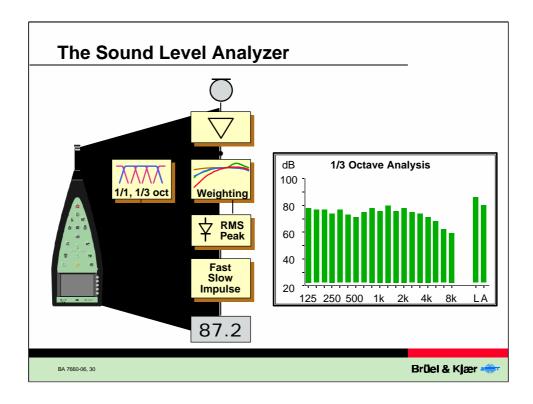
All sound level meters have built in A-weighting, and some have also others like B and C-weighting.



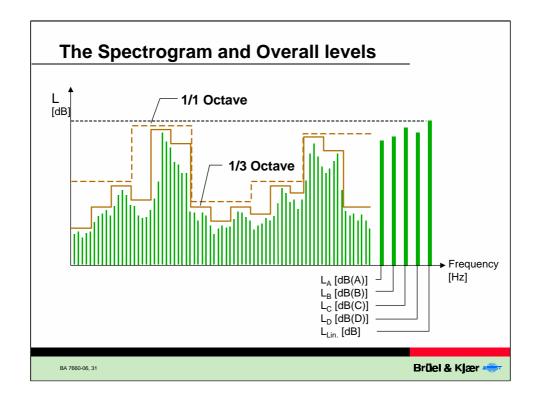
For more detailed frequency analysis sound level meters can have a serial filter bank of 1/1 octave and maybe also 1/3 octave bandwidths. As only one filter is active at a time the analysis time is long and requires the sound field to be stationary. The advantage is a lower price.



A sound level meter with a parallel filter bank is the most expensive, but the fastest in operation and does not require the signal to be stationary.



The most advanced sound level meters act more like an ordinary sound analyzer having both a parallel filter bank and a selection of weighting filters.



The advanced sound level meters often feature comprehensive displays showing all the analysis results from both the parallel analysis filters and the weighting curves simultaneously.

