Name that tone

Report by Paul Hopwood AMIOA

Many acoustic investigations require the measurement of tones, which can vary from simple noise assessments such as BS EN 4142 that require adjustments added to noise rating levels if tones are present through to much more detailed analysis required in electroacoustic measurements such as BS EN 60268 or investigation of sources of noise in machinery.

The simplest form of assessments for tones requires an analyser with the ability to select a frequency range for analysis, the degree of frequency selectivity required in the instrument depends upon the task being under taken, originally it was common to find octave bank filters in tools but these days 1/3rd octave or even 1/12th octave bank filters are not uncommon in standard instruments. These days it is unlikely that the filters in instruments will be analog but rather digital with the performance of the filter banks specified in BS EN 61260 ensuring that measurements made on separate instruments are comparable.

When it comes to identifying whether a noise source is tonal the best instrument for performing this task is the human ear, however the ear is not suitable for measuring tones and whilst ISO 1996 identifies the need for subjective assessment it also specifies in detail in its annexes two separate methods for performing measurements on tones.

To perform a tonal analysis it is likely that either 1/n octave analysis will be used or narrowband FFT analysis, there are issues with both of these methods, FFTs are complex to correctly setup and are constant bandwidth whereas 1/n octave filters are constant percentage bandwidth filters.

Tones are often identified in filter banks by a peak in one band with lower levels in adjacent bands, but with 1/n octave analysis the frequency resolution decreases as the frequency of interest increases, a 1/3rd octave filter at 63Hz has a bandwidth of 14.5 Hz whilst at 16 KHz the bandwidth has grown to 3.6 KHz. Similarly if the tone falls between two of the octave filters the energy will be shared between the two filters and the indicated level will be inaccurate and is unlikely to be noticed as a tone.

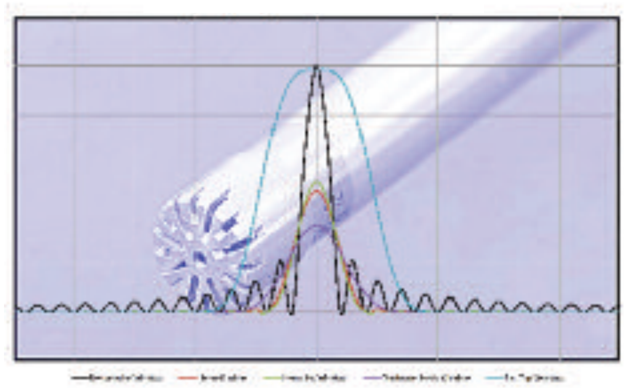
Using an FFT analysis involves a series of trade offs and potential pitfalls, to get the optimum settings for analysis prior knowledge of the noise source is normally needed which is why it is often preferable to have a recording for analysis. Fundamentally FFT analysis takes a window of time and transfers this to the frequency domain, this results in the major trade off, a high frequency resolution results in a low time resolution and vice versa, the sampling frequency of a FFT is normally fixed in hardware so control over the trade off between time & frequency resolution is governed by increasing or decreasing the FFT block length, often called the number of lines or bins.

The mathematics behind FFTs assume a periodic time signal is being analysed, in the real world most signals analysed are not periodic in the sample time windows and the discontinuities lead to spectral leakage, frequencies being created which are not present in the input signal.

The solution to spectral leakage is to use a window on the input data, multiplying or convolving the signal with a function which is zero at the end of the time window and large in the centre of the window, essentially concentrating the FFT on the data at the centre of the time window.

When a window is used the input signal is being modified, essentially it is being amplitude modulated and perfectly accurate results cannot be expected. Once again selecting the correct window is a trade off and requires knowledge of the signal that is to be analysed.

Using a hanning window gives good frequency resolution and is best for noise and periodic signals longer than the time windows but gives an inaccurate reading with transient signals which are already zero at the start and end of the time signal, for these signals a uniform rectangular window should be used. If a signals amplitude is needed to be known accurately, such as during calibration a flat-top window is used to combat the temporal variation in amplitude of the hanning window, the flat-top is accurate to within 0.1dB compared to the hanning which is accurate to within 1.5dB.



Frequency response of common FFT window functions

There are many other window functions available for special applications such as the Blackman-Harris used for resolving closely spaced frequencies with differing levels or the force window used for impact testing. Often settings are available in the FFT to overlap windows. Overlap means that instead of waiting for a new sample period we use some new data & some old data to create a measurement. This data increases the display rate, indicating the direction and the change in spectrum but it is not correct until the new sample period is reached, but the increased display rate is useful for situations where making adjustments or for RMS averaging.

Typically there are also settings for scaling the data and averaging the data, scaling allows the measurement to be treated as a RMS signal or as a peak signal, the later multiplies the calculated FFT by √2 and should therefore be used carefully. There are two main types of averaging, although they are called by numerous names, both types of averaging may also be weighted in time with linear or exponential weights. Power averaging reduces signal fluctuations giving a RMS figure for the signal plus noise and with a sufficient number of averages allows an estimate of the noise floor. Vector averaging works on the complex FFT spectrum and requires a trigger signal to make sense, with a trigger the signal is phase coherent but the noise isn't allowing an improvement in the signal to noise ratio.

There is a promising technology which has been written about in papers on acoustics that may bridge the gap between 1/n octave analysis and FFT analysis. Wavelets are widely used in other engineering fields; the images in this issue of Acoustics Bulletin have probably been stored & compressed using wavelets and have attractive properties for acoustic measurements. Wavelets use constant percentage bandwidth like 1/n octave analysis but the time & frequency tradeoff isn't consistent, at low frequencies wavelets have a higher frequency resolution with low time resolution but as the frequency is increased the frequency resolution worsens whilst the time resolution improves, more closely matching the performance of the human ear.