COMPARISON OF SOUND POWER MEASUREMENTS USING BOTH DIRECT AND INDIRECT INTENSITY TECHNIQUES

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1 Introduction

The determination of sound power of machines is one of the major industrial applications of sound intensity techniques. Using the technique of two closely spaced microphones, there are basically two methods of calculating the sound intensity; the **direct**, or time-domain, method and the **indirect**, or frequency-domain, method. This paper provides an overview of the two methods, along with their advantages and disadvantages, and presents the results of a sound power determination of an electric drill using two direct intensity analysers and one indirect intensity analyser

2. Intensity Techniques

2.1 The Direct Intensity Method. Sound intensity is defined as the product of the instantaneous acoustic pressure and the instantaneous acoustic particle velocity. By using two closely spaced microphones, the pressure is given as the mean of the two microphone signals, and the particle velocity is calculated from the pressure gradient between the two microphones. The intensity is then given by:

$$I = \frac{-1}{2\rho\Delta r} [P(b) + P(a)] \int [P(b) - P(a)]\, dt \quad \ldots\ldots (1)$$

where ρ = density of air
Δr = microphone spacing
and $P(b)$ and $P(a)$ are the pressures in microphones **a** and **b** respectively

Equation (1) can be realised directly in an analyser by using suitably scaled sum-and-difference amplifiers and an integrator, and a typical configuration is shown in Figure 1. Note that in this example, filters have been incorporated into the signal paths to yield frequency information.

The main attraction of this method is that the calculation may be performed in true real-time, allowing analysis of transient phenomena to be made. If a parallel bank of filters is incorporated, for example, 1/3-octave digital filters, then the frequency analysis may also be performed in real-time without loss of data. This finds many additional applications in the study of reciprocating machines, and other impulsive sources[1]

In practice, the frequency analysis is performed with constant percentage bandwidth (CPB) filters, either in octaves or fractional octaves. This is convenient to most acousticians, as it offers data reduction, and results which may be compared to a large existing database. Existing sound power standards (using pressure measurements[2]) also specify the use of CPB presentation, and it is likely that future sound power standards using intensity will follow suit. An example of a sound intensity analysis using CPB is shown in Figure 2. The hatched columns in the spectrum indicate intensity incident from the rear of the probe.

2.2 The Indirect Intensity Method. The frequency domain realisation of Equation (1) can be shown[3] to give the sound intensity as:

$$I = \frac{-1}{\rho\omega\Delta r} \quad \mathrm{Imag} \quad G_{ab} \qquad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (2)$$

where G_{ab} is the complex cross spectrum of the two microphone signals,

and ω = angular frequency in radian/sec,
ρ = density
Δr = microphone spacing

The practical realisation of this is to use a cross-spectrum analyser, the most universal of which is the dual channel Fast Fourier Transform (FFT) analyser. All that is required is to calculate the imaginary part of the cross-spectrum, and scale it according to equation (2). Some commercial analysers now have this scaling built in as part of the display processing (Figure 3).

N.B. Equation (2) can also be realised in the time domain by use of the Hilbert Transform and is often a feature of CPB analysers[4].

The advantage of the indirect technique is the high resolution offered by FFT analysis, making it ideal as a research tool,e.g. for diagnosing tonal noise sources.

An example of an intensity spectrum calculated from FFT is shown in Figure 4.

However, for sound power determination, CPB presentation will be essential, so a conversion must be performed from the FFT spectrum (constant bandwidth on a *linear* frequency scale) to a CPB spectrum (constant *percentage* bandwidth on a *logarithmic* frequency scale).

This process is known as synthesis and is discussed in the next section.

The major disadvantage of FFT analysis is that the analysis is not necessarily performed in real-time, being limited by the calculation speed of the processor. This implies that in many cases, data may be lost, or suppressed, imposing a requirement for time invariance, or stationarity, on the measured intensity. This could be a problem when impulsive sources are being characterised.

3 Synthesis

3.1 Synthesis of CPB spectra from FFT spectra. In essence, the conversion of narrow band data to CPB format is very simple and is normally performed by the measurement processor in basic analysers (hence reducing still further the real-time rate) or by the display processor in split-processor analysers without influencing the data acquisition. The basic difficulty, however, is the incompatibility between the filter types of the two analysis methods. The bandwidth of a CPB filter is a percentage of its centre frequency; for example, the width of the 100 Hz 1/3 octave filter is 23 Hz, and at 1000 Hz, the width becomes 230 Hz. In other words, the bandwidth increases with centre frequency. If displayed on a logarithmic frequency scale, these two filters will appear the same "width".

With FFT analysis, the bandwidth depends upon the length of the time buffer,the full-scale frequency, the transform size, and the weighting function used in the time domain. For a 2048 point transform up to 25.6 kHz, and with Hanning weighting, the effective filter bandwidth will be 48 Hz. Typically, 800 filters will be calculated, spaced at 32 Hz intervals. In this case, the filter bandwidth will stay constant, regardless of centre frequency.

In order to calculate the CPB spectrum, the FFT may be displayed on a logarithmic axis (Figure 5) and a "mask" of CPB filters superimposed. The power of all the FFT lines within each mask is then summed and the result displayed. This procedure runs into problems at low frequencies where there are too few FFT lines available, so synthesis is often restricted to the top two decades of the frequency span (e.g. 256 Hz to 25.6 kHz). The resulting "filter" shape depends upon the centre frequency and the time weighting, and two examples are shown in Figure 6[5] compared with the limits for the ANSI standard. Note that while these filters comply in most respects with the standard, they are only valid for broad-band signals, and serious errors could occur with deterministic signals. Note also that the filter shapes differ from the established Butterworth or Chebychev filter characteristics around which the standards were written, which makes comparison with the established database of measurements difficult if not impossible

3.2 Synthesis and Real-Time Frequency. Another consideration of FFT analysis is "real-time frequency". This is defined as the highest selectable full-scale frequency at which analysis may be performed, without input data being lost. This criterion is based upon the blockwise approach of FFT analysis, whereby the processor first samples data, then calculates and displays the FFT before returning to sample more data. If the calculation time exceeds the length of the next data block, then the analyser is no longer working in real-time. So as the data block gets shorter (i.e. the full-scale frequency increases), eventually, data will be lost, defining an upper frequency limit to true real-time analysis.

Depending upon the analyser architecture, dual channel operation will also slow down the processor, reducing the real-time frequency further. Also, when a time weighting function is specified, this will increase processing time again, in order that the data may be multiplied by the appropriate table of values. All these variables allow analyser manufacturers to qualify the way in which their particular analyser is specified, and very often, real-time frequency will be specified with no time weighting, and in single channel operation, hence giving the impression of high real-time speed.

COMPARISON OF SOUND POWER MEASUREMENTS

However, a typical specification might be 5 kHz real-time rate for dual channel operation and Hanning weighting. Note that the same analyser could be quoted at *17 kHz* in single channel and flat weighting! The implication of this is that if frequencies above 5 kHz are to be measured, then real-time operation cannot be achieved and the source must be assumed to be stationary with time. Another problem comes with the use of Hanning weighting. This signal weighting is used for analysis of continuous signals to avoid leakage errors in the frequency domain. The weighting effectively suppresses the data in the first and last quarters of the time record and emphasises the middle. This means that to ensure that all data receives equal emphasis, consecutive time records must be overlapped by at least 66%. The result of this is that the real-time frequency will be reduced by a factor of three, so with the example, 1.6 kHz is the highest range that may be selected to ensure real-time operation.

Synthesis, along with the intensity scaling of Equation (2), will further decrease the real-time rate, unless performed by a secondary processor, and also, to achieve sufficient resolution at low frequencies, a multiple pass analysis may have to be performed, one at 25.6 kHz, one at 2560 Hz and one at 256 Hz to give four decades of data comparable to real time CPB analysis. This effectively precludes the synthesis of CPB spectra on anything other than very stationary signals. To illustrate the errors that may be seen, three commercial analysers were used, each having different synthesis techniques,and the same signal(a 10ms toneburst at 2kHz repeated at 1s intervals) was analysed to give the 1/3 octave spectra. Note that the analysers were used in single channel mode (not intensity). The results may be seen in Figure 7.

It can immediately be seen that large errors of typically 6-8 dB occur (20dB in one case!) even though the signal was not a particularly demanding one with respect to stationarity. The results are compared with an analysis performed using a 1/3 octave digital filter analyser operating in true real-time.

4. Sound Power Determination using CPB and FFT techniques.

4.1 Measurement procedure. Using both methods, the sound power of a small domestic power drill was determined.Three different analysers were used, two using the direct method, one using the indirect.

The first analyser is a dedicated 1/1 ,1/3 or 1/12 octave real-time digital filter analyser, the second a 1/1 octave serial portable analyser and the third is an 800 line dual channel FFT analyser with a real-time frequency of 1.6 kHz when using Hanning with 75% overlap.

To compute sound power from sound intensity, a surface is taken to enclose the power source, and the intensity measured at several points normal to the surface.

The sound power is then calculated from:

$$W = \sum_{i=0}^{n} I_i . \delta S \quad \text{.................(3)}$$

where I_i = intensity flowing through surface element δS

For the drill, a cylindrical measurement surface was taken (65cm diameter X 180cm high), and the drill suspended over a reflecting plane.A total of 26 measurement points were taken over this surface. The environment was acoustically quite dead with a reactivity of less than 3 dB.
A two microphone probe was used with a spacing of 12mm, and to minimise discrepancies between systems, the same probe was used with all three analysers, switching inputs with a multiplexer.
The same effective averaging time was taken for each analyser, and in the case of the FFT, synthesis was performed in a single pass, yielding two decades af data. The synthesis was done using a separate post-processor to maintain the same real-time rate.
Initially, 1/3 octave synthesis was done, then to aid comparison with the smaller octave analyser, the results were converted into octaves.

4.2 Results. The results are shown in Table 1 for the three analysis systems, and in graphical form in Figure 8.

dB re: 1.0 E -12 watts

Octave Hz	3360	2032	4433
250	56.7	56.5	<60
500	68.8	68.3	65.5
1000	79.4	79.1	78.8
2000	81.7	81.4	82.9
4000	87.0	86.8	87.6
"A" weighted	89.6	89.4	89.3 +

+calculated from octaves

3360 = digital CPB analyser
2032 = dual channel FFT analyser
4433 = serial octave CPB analyser

Table 1

A comparison between the 1/3 octave CPB results and the synthesized FFT results is shown in Table 2.
The results show a very good level of agreement, in both octaves and 1/3 octaves, with the calculated "A"-weighted values also

being within 0.3 dB. With the portable CPB analyser, the sound power was also determined using an "A"-weighting filter which gave the result of 89.8 dB.
As the source was very broad banded and stationary in nature, with only one pure tone apparent at 5032 Hz (gearbox whine), it should be expected that the synthesis of 1/3 octaves from the FFT data is valid and the agreement with the digital filters is extremely good.

dB re: 1.0E-12 watts

1/3 oct.	3360	2032
200	47.6	49.4
250	50.9	50.2
315	54.7	54.5
400	61.7	62.1
500	63.4	63.6
630	65.6	65.6
800	69.4	69.7
1000	72.5	72.9
1250	77.4	77.7
1600	77.0	77.3
2000	76.3	76.2
2500	76.7	77.1
3150	79.1	79.3
4000	78.8	78.8
5000	85.1	85.3

Table 2

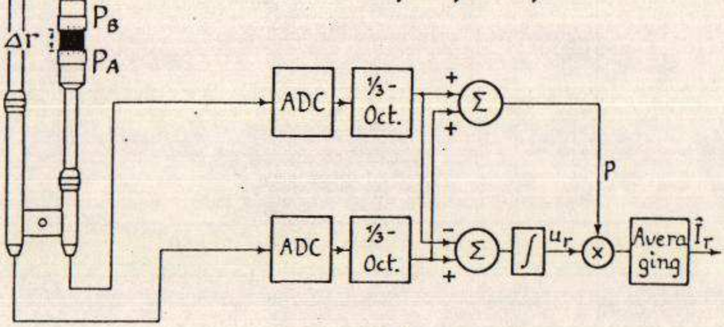
5. Conclusions

The principles of sound intensity measurement using direct and indirect techniques has been discussed, along with the concepts of synthesizing fractional octave filters from narrow band data. Shortcomings with respect to real-time capability and effective filter shape have been highlighted, but to demonstrate the validity of the technique, a measurement of sound power has been performed on a broad-banded source which is stationary in time. The agreement between the two methods has been shown to be extremely good, but the need for care when using FFT synthesis cannot be overstated.

6. References

[1] "Gated Sound Intensity Measurements on a Diesel Engine", Bruel & Kjaer Application Note BO 0203-11
[2] British Standard BS4196:1981(1986) "Sound Power Levels of Noise Sources", British Standards Institution
[3] Bruel & Kjaer Technical Review TR3-82 Appendix D
[4] Bruel & Kjaer Product Data Sheet Type 2134/WH1493
[5] Instruction Manual Type 2313/BZ7006, Bruel & Kjaer

$$I_r = \overline{p \cdot u_r}$$



$$\hat{I}_r = \frac{-1}{2\rho\Delta r}\overline{(p_A + p_B)\int(p_B - p_A)\,dt}$$

Figure 1

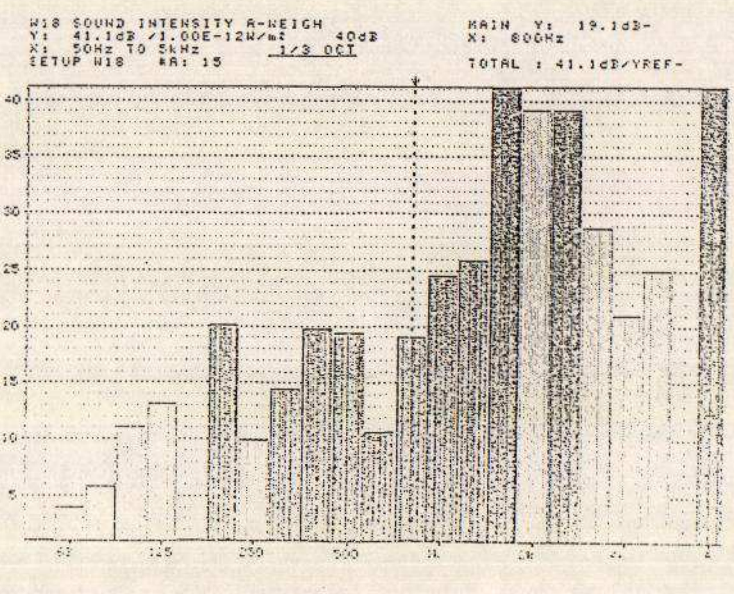


Figure 2

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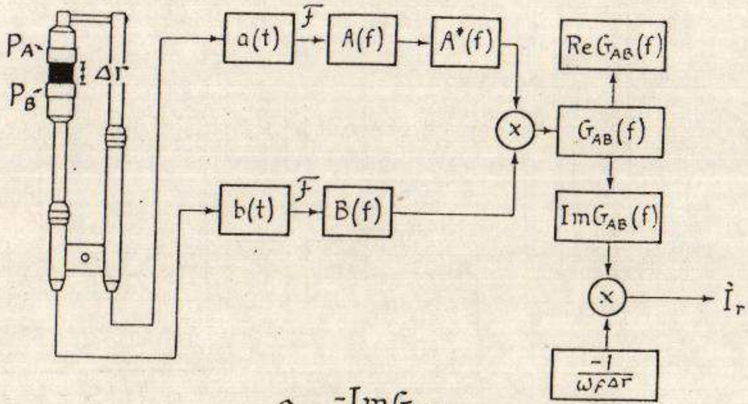


$$\hat{I}_r = \frac{-\mathrm{Im}\,G_{AB}}{\omega\rho\Delta r}$$

Figure 3

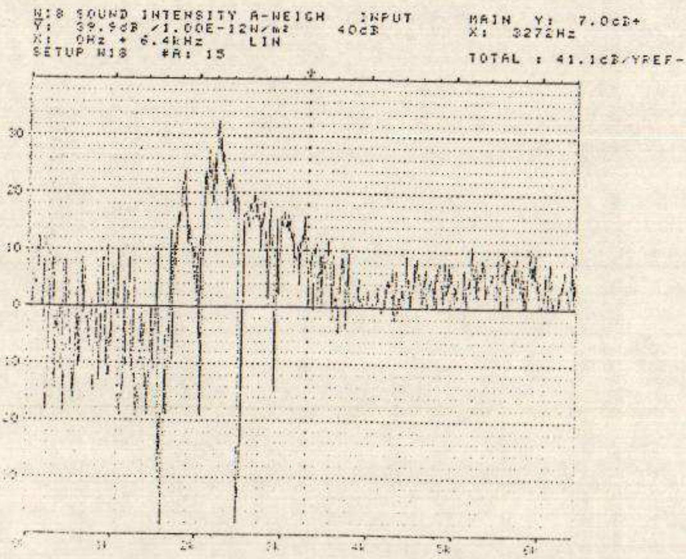


Figure 4

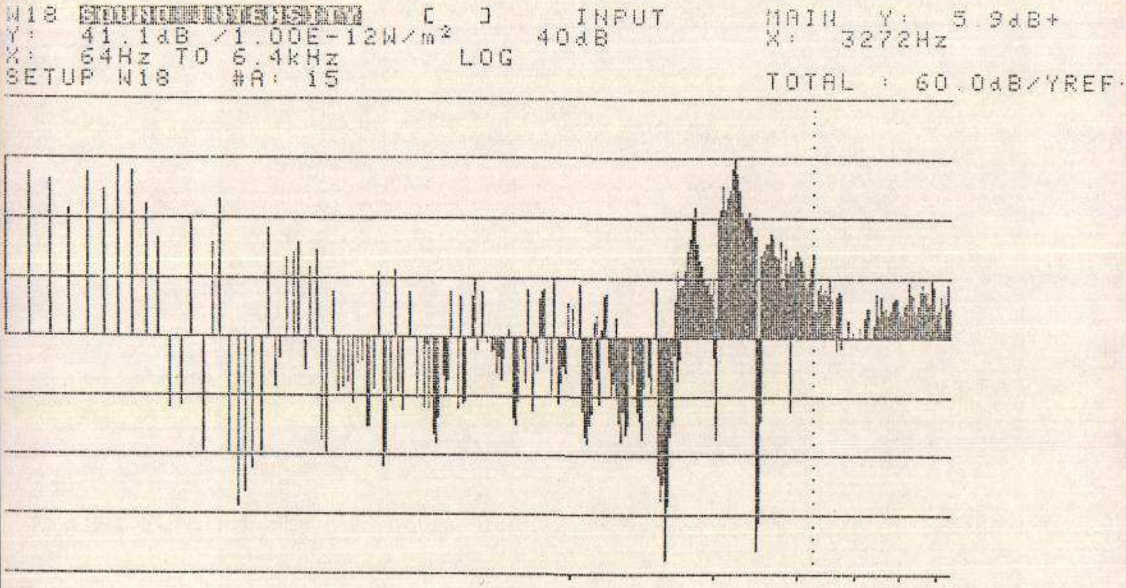
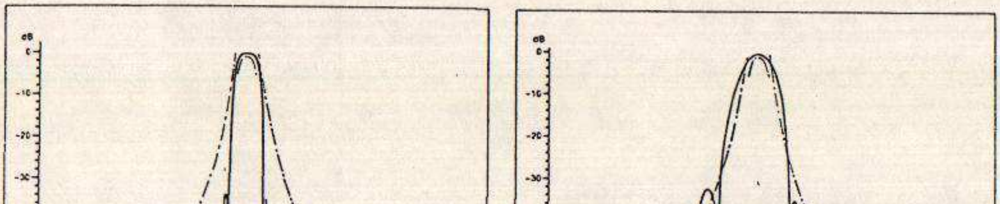


Figure 5



1/3 Octave Filter Response, Center Frequency = 630 Hz

1/3 Octave Filter Response, Center Frequency = 200 Hz

Figure 6

COMPARISON OF SOUND POWER MEASUREMENTS

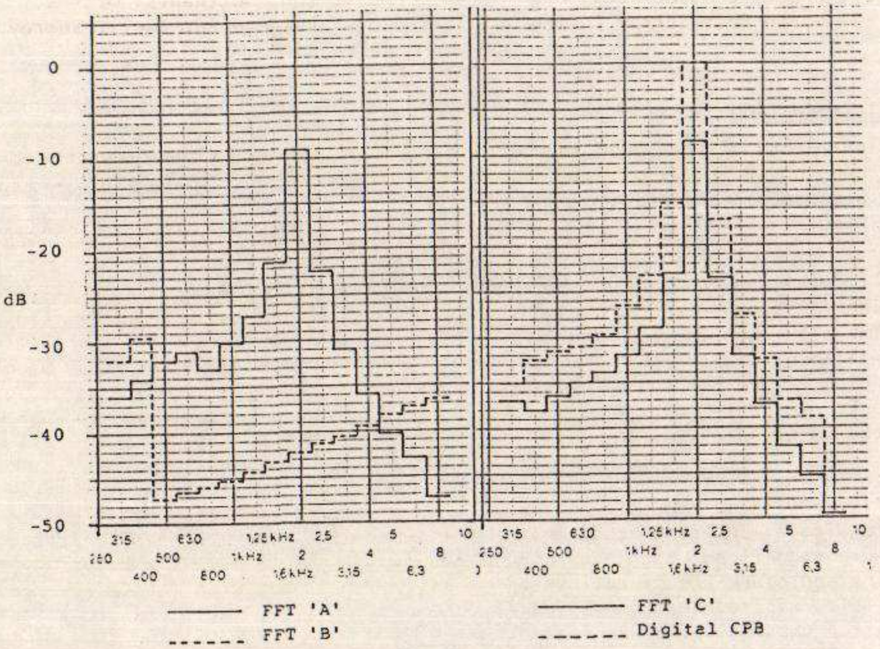


Figure 7

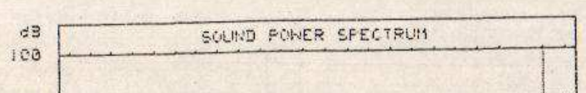


Figure 8