

NOISE SOURCE IDENTIFICATION - ACOUSTIC INTENSITY TECHNIQUE

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ABSTRACT

The determination of the sound power of audible noises and their location are usually complicated by room effects, high ambient conditions and frequently the presence of other sources. The two-microphone acoustic intensity approach was found to be a promising technique for analyzing sources of noise where conventional methods are not particularly practical. A procedure is described for measuring directly acoustic intensity levels using a simple "calibrator" and commercial available dual channel analyzer.

RESUME

En général, la détermination de la puissance de bruits audibles ainsi que leur localisation se complique par des effets du local, par des bruits de fond hauts et souvent par la présence d'autres sources. La technique de mesure de l'intensité sonore à l'aide de deux microphones est trouvée prometteuse pour l'analyse des sources des bruits, là où les méthodes conventionnelles ne sont pas particulièrement pratiques. Une procédure est décrite pour mesurer directement l'intensité sonore en utilisant un "calibreur" simple et un analyseur à deux canaux.

1.0 INTRODUCTION

A principal objective in noise control engineering should be identification of the sources of noise and reduction at the source, rather than obstructing the transmission of the noise using enclosures, ear plugs, etc. This approach usually provides cost effective solutions to noise problems while maintaining the efficiency and reliability of the equipment involved.

In many industrial plants, the noise radiated by a particular piece of equipment is often difficult to isolate because of operational constraints. Similarly in many machines, the noise radiated by a given component is frequently difficult to identify. Conventional methods of noise analysis are sometimes impractical due to reflections from nearby surfaces, closely spaced sources, or high levels of background noise/1/.

Recent advancements in instrumentations, such as in the two-channel Fast Fourier Transform (FFT) analyzer, together with the development of a new mathematical analysis for acoustic intensity has made it possible to determine sound power directly and relatively simply using two closely spaced microphones/2,3,4/. Preliminary reports from investigators using this technique have been positive and it is anticipated that acoustic intensity measurements will become widely used in the future.

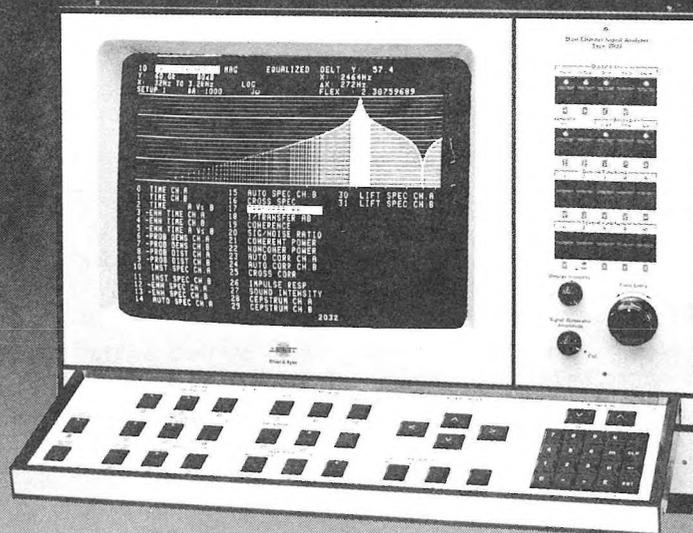
This paper describes a practical method for measuring acoustic intensity and presents some experimental results.

2.0 THEORETICAL BACKGROUND

The sound power from a source is defined as the acoustic intensity integrated over a hypothetical surface surrounding the source/5/. Intensity is a vector quantity representing the time-averaged flow of acoustic energy per unit area in watts/m². A basic expression for intensity is

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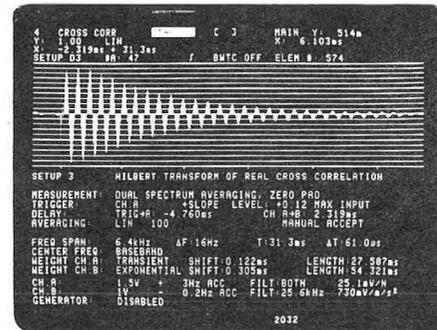
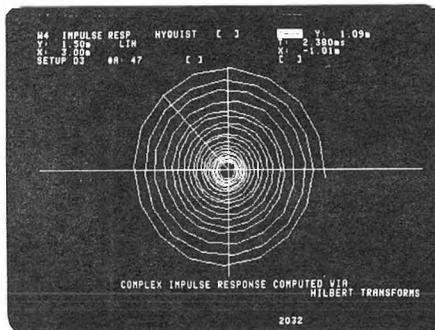
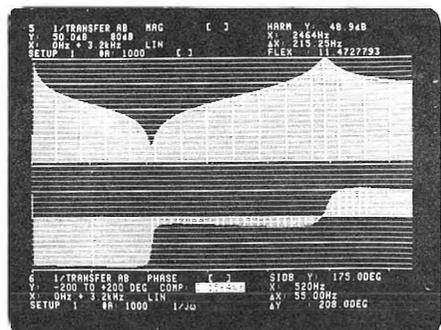
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Condensed Specifications Type 2032

FREQUENCY RANGE: 0 to 25,6 kHz

RESOLUTION: 801 lines based on a 2048 point time function in both single and dual channel operation

ZOOM: built-in with 15 selectable frequency spans from 1,5625 Hz to 25,6 kHz

REAL-TIME FREQUENCY: dual channel operation, > 5 kHz, single channel operation, > 10 kHz

SAMPLING FREQUENCY: internal, 65,536 kHz or external, 67 kHz max., 12-bit A/D conversion

NUMBER OF INPUTS: 2 fully instrumented input channels, A and B, each equipped with a direct, an accelerometer line drive input and a microphone preamplifier

CHANNEL MATCHING: maximum gain difference due to analogue anti-aliasing filters, 0,3 dB; maximum phase difference, 3° up to 20 kHz, 5° up to 25,6 kHz

INPUT VOLTAGE RANGE: minimum detectable signal level 1 μ V, maximum peak input voltage, selectable from 15 mV to 100 V in a 1,5 - 2 - 3 - 4 - 6 - 8 - 10 sequence

DYNAMIC RANGE: > 75 dB, (> 80 dB typical)

LINEARITY: \pm 0,1 dB or \pm 0,01% of full scale, whichever is greater

INPUT AUTORANGE: autoranges the input attenuators to select the optimum setting. Can be manually overridden

CALIBRATION: user-defined in engineering units, in volts or in dB referred to a user-defined reference

CALIBRATION ANNOTATION: in V/V, V/unit, V/Pa, V/m.s⁻², V/m.s⁻¹, V/m, V/N, V/g, V/in.s⁻¹, V/mil

FUNCTIONS MEASURED:

Instantaneous time function: ch. A or ch. B

Instantaneous time function: ch. A vs. ch. B

Enhanced time function: ch. A or ch. B

Enhanced time function: ch. A vs. ch. B

Probability density: ch. A, or ch. B

Probability distribution: ch. A or ch. B

Instantaneous spectrum: ch. A or ch. B

Enhanced spectrum: ch. A or ch. B

Auto spectrum: ch. A or ch. B

Cross spectrum

Transfer function: AB or 1/TRANSFER AB

Coherence

Signal-to-noise ratio

Coherent output power

Non-coherent output power

Auto correlation: ch. A or ch. B

Cross correlation

Impulse response

Sound Intensity

Cepstrum: ch. A or ch. B

Lifted spectrum: ch. A or ch. B

MEASUREMENT MODES:

Spectrum averaging: single and dual channel

Spectrum averaging zero pad: single and dual channel

Signal enhancement: dual channel

Probability density mode: dual channel

AVERAGING OF DATA: from 1 to 32767 time or frequency ensembles can be averaged in linear or exponential averaging, or, (single channel only), peak averaging

HILBERT TRANSFORM: built-in for calculating the envelopes of complex time domain functions such as correlation functions, impulse responses

DISPLAY OF DATA: as real part, imaginary part, (where relevant), magnitude, phase, Nyquist plot, and Nichols plot, on a 12" raster scan display fully annotated with the measurement and display set-ups

FURTHER ANNOTATION: 2 x two 50 character text lines for user-defined annotation

DISPLAY OF SPECTRA: linear, RMS, power, PSD, ESD

DISPLAY SCALES:

Amplitude: lin. or log.

Frequency: lin. or log.

Time: lin.

AUTOSCALE FUNCTION: automatically optimises display scales; can be manually overridden

CURSOR FUNCTIONS: main, harmonic, sideband, delta, mask, and reference

STORAGE OF MEASUREMENT SET-UPS: up to 10 user-defined measurement set-ups can be stored in non-volatile memory. Further 10 factory-defined measurement set-ups built-in

STORAGE OF DISPLAY SET-UPS: up to 10 user-defined display set-ups can be stored in non-volatile memory. Further 10 factory-defined display set-ups built-in

STORAGE OF DATA: one complete measurement including the measurement set-up can be held in memory

POST - PROCESSING: equalize function, single and double integration and differentiation of results, flexible cursor for engineering units calibration of the x-axis, removal of bow-tie correction on correlation functions, power in band, total power, power in band / total power, A-weighting

TRIGGER:

Trigger Source: free-running, internal, external, manual, or noise

Trigger Level: adjustable in 199 steps between plus and minus full scale

Trigger Slope: positive or negative

Trigger Delay: adjustable in steps of 1 sample from minus one record to 9999 s

OFFSET BETWEEN CHANNELS: adjustable in steps of 1 sample from 0 to 9999 s

TIME WEIGHTING: flat, Hanning, transient, and exponential, with user-definable position and width for transient and exponential

SELF TEST: automatic self test function built-in

SIGNAL GENERATOR: built-in signal generator with sine, impulse, random, and pseudo-random output. Random and pseudo-random output band-limited to zoom frequency span

INTERFACE: fulfils IEC 625-1, and when equipped with adaptor AO 0195, IEEE Std. 488

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OPERATING TEMPERATURE: 5° to 40°C (41°F to 104°F)

STORAGE TEMPERATURE: -25°C to + 70°C (-13°F to +158°F)

DIMENSIONS AND WEIGHT: ("A" cabinet without feet)

Height: 310,4 mm (12,2 in)

Width: 430 mm (16,9 in)

Depth: 500 mm (19,7 in)

Weight: 35 kg (77 lb)

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$$\vec{I} = p \cdot \vec{u} \quad (1)$$

where p is the sound pressure in N/m^2 and \vec{u} is the particle velocity in the direction of propagation in m/s .

In the past, Equation (1) was difficult to use because the instrumentation for making reliable measurements of u was not available/6/. However, it can be shown that for a free-progressive wave the intensity becomes/5/

$$I = \frac{p_{rms}^2}{\rho c} \quad (2)$$

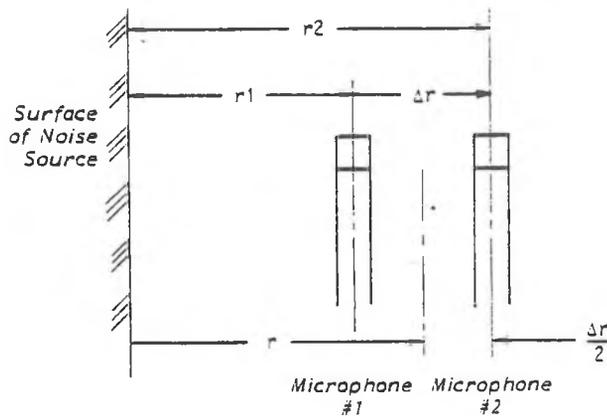
where P_{rms} is the root-mean-square of the sound pressure and ρc is the acoustic impedance (ie for air $\rho c \approx 400 N\cdot s/m^3$).

Equation (2) applies in the far-field of a source where \vec{u} and p are in phase. In this case, reference quantities have been chosen so that the level in decibels corresponding to the intensity and to the pressure are the same. That is,

$$L_I \text{ (re } 10^{-12} \text{ watt/m}^2\text{)} = L_p \text{ (re } 20 \text{ } \mu\text{Pa)}. \quad (3)$$

Therefore, in most atmospheric conditions found in practice $L_I \approx L_p$ and can be readily obtained using pressure sensitive instruments, ie microphone.

The alternative method of measuring intensity, hence sound power is based on Equation (1) and the signals from two closely spaced microphones arranged as shown below: (r , r_1 , etc are distances in meters).



The acoustic velocity and pressure can be represented, in complex form, by the following finite difference approximations:

$$u = - \frac{1}{\rho \Delta r} \int (p_2 - p_1) dt \text{ and } p = (p_1 + p_2)/2$$

where p_1 and p_2 are the pressures measured at r_1 and r_2 respectively, ρ is the density of the medium, and Δr is the spacing between microphones. Substituting these approximations into Equation (1) and converting the result into the frequency domain, using Fourier Transform, the acoustic intensity can be expressed as/3,4/

$$I(f) = \frac{\text{im}(G_{12}(f))}{\rho \Delta r \omega} \quad (4)$$

where $\text{im}(G_{12}(f))$ is the imaginary part of the cross-spectrum between the two microphone signals (rms pressure), $\omega = 2\pi f$ the circular frequency, Δr and r as before. This equation can be readily computed using commercially available two channel FFT analyzers with calculators.

One inherent limitation in this technique is associated with the spacing of the microphone relative to the wavelength of sound being analyzed. The following limits have been proposed, tentatively, for achieving an accuracy within ± 1.5 dB:

$$0.1 \leq k\Delta r \leq 1.3 \text{ and } 0 \leq \Delta r/r \leq 0.5$$

where $k = \omega/c$ is the wave number, c is the speed of sound, and Δr , r as before/7/.

Another limitation is due to the fact that Equation (4) is sensitive to phase and gain differences between the two microphone-channel systems. Although some instrument manufacturers provide special, phase matched components, in practice it is desirable to verify on-site that the complete two channel systems are acceptably matched.

One method of compensating for phase and gain differences is by switching channels while measurements are made/4/. This provides very accurate results but is relatively slow because two sets of data must be collected for every measurement. A preferred method is to use the transfer function of the two channel system for compensating for phase and gain differences/8/. There are also other methods presently being investigated/9/.

3.0 DESCRIPTION AND PROCEDURE OF A PRACTICAL SYSTEM

3.1 Introduction

Several microphones and microphone configurations, and data processing techniques are available for acoustic intensity measurements/10/. A typical arrangement is illustrated in Figure 1.

Briefly, the signals from the two microphones are fed into the analyzer which computes the cross-spectrum. The analyzer, in conjunction with a programmable calculator, is used to make the necessary calibration corrections, compensation for phase and gain, and the calculation of the acoustic intensity level in dB. A calculator can be used to make additional computations such as sound power level, octave band analysis, source order ranking, etc. Finally, the results may be recorded using a digital plotter.

In this study a simple calibration procedure was used with an FFT analyzer having complex mathematical capability. This enables the intensity level, in dB, to be obtained directly from the analyzer without an auxiliary calculator and facilitates making exploratory scanning of sound fields around mechanical equipment in-situ with the minimum of instrumentation and programming.

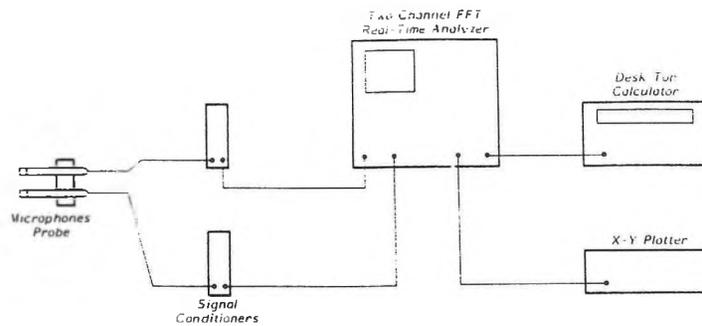


FIGURE 1
TYPICAL INSTRUMENTATION ARRANGEMENT FOR ACOUSTIC INTENSITY MEASUREMENTS

3.2 Selection of Microphone Size and Spacing

The choice of microphone size and spacing appropriate for the particular application depends on the required dynamic and frequency ranges. The dynamic ranges of typical microphones are shown in Table I.

TABLE I

Microphone Size, mm	Dynamic Range
25 mm (1 in)	15-145 dB
13 mm (1/2 in)	30-145 dB
6 mm (1/4 in)	50-150 dB
3 mm (1/8 in)	65-170 dB

In most applications involving large power equipment, either the 6 mm or the 13 mm microphones will give satisfactory results. Based on the frequency range of interest, the spacing between the two microphones and the distance from the source can be chosen. A tentative selection chart is given in Figure 2.

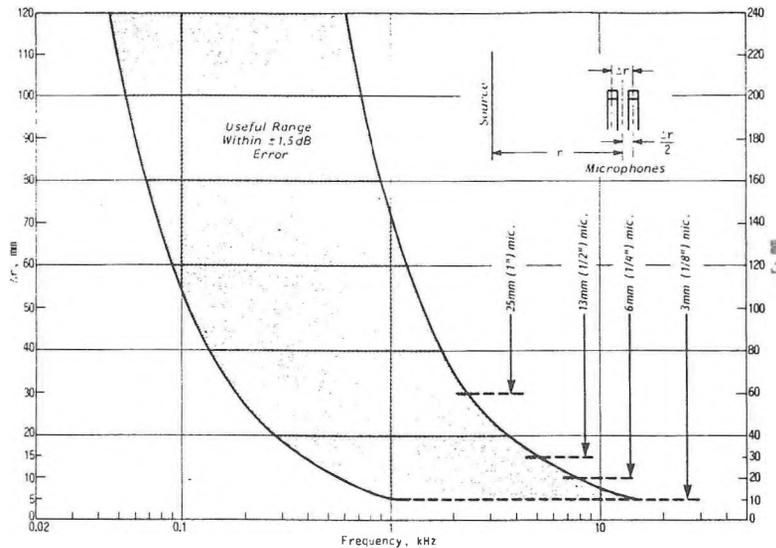


FIGURE 2
TENTATIVE CHART FOR SELECTING PARAMETERS FOR ACOUSTIC INTENSITY MEASUREMENTS

3.3 Gain Calibration

It was found convenient in this procedure to use two sound level meters as signal conditioners. These facilitated calibration in terms of acoustic intensity level and monitoring the input signals during analysis.

Using a standard calibration signal fed into one of the microphones, the sound pressure levels shown by the corresponding sound level meter and the analyzer were adjusted to read correctly. This channel, therefore, was designated for reference purposes. The second channel was then adjusted to agree with the reference channel above.

3.4 Phase Mismatch Compensation

Corrections for phase differences between the two channel systems were made using the transfer function method described in Ref 8. Accordingly, the transfer function was obtained by exposing the microphones simultaneously to random noise in a duct as illustrated in Figure 3.

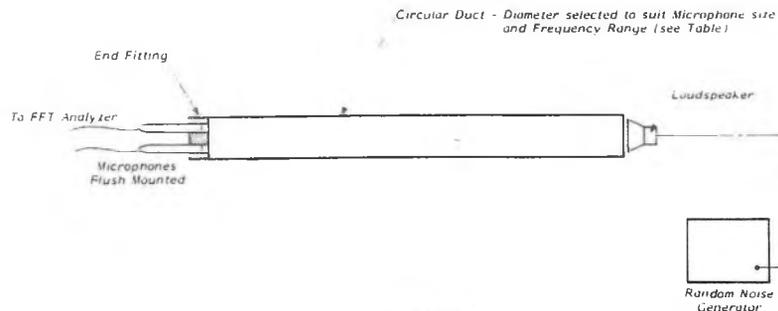


FIGURE 3

The size of microphones, of course, determines the minimum diameter of duct that can be used. This in turn determines the upper cutoff frequency of the duct where the first cross-mode occurs/11/.

It was found that for a duct having the smallest practical diameter that would fit a given pair of microphones, the cutoff frequency is greater than the maximum frequency range of this technique (shown in Figure 2). A summary of the maximum frequency ranges corresponding to four microphone sizes and the recommended duct diameter and its upper cutoff frequencies are in Table II.

TABLE II
RECOMMENDED DUCT SIZES FOR OBTAINING THE
TRANSFER FUNCTION BETWEEN MICROPHONES

MIC DIAM mm in	Upper Freq for Intensity* Hz	Recommended Duct Diam for Transfer Function mm in	Upper Cutoff Freq of Duct Hz
25 (1")	2345	63.5 (2.5")	3138
13 (1/2")	4690	40.1 (1.5")	4878
6 (1/4")	7035	19.1 (0.75")	10460
3 (1/8")	14069	12.7 (0.5")	15690

*Based on $0.1 \leq k \Delta r \leq 1.3$

Phase mismatch compensation was made, therefore, by dividing the cross-spectrum measurement by the transfer function in their complex notations. The imaginary part of this result was then used in Equation 4 for computing acoustic intensity.

3.5 Intensity Level Computation

In order to compute the intensity level, a special duct arrangement was used as an "acoustic intensity calibrator" (Figure 4).

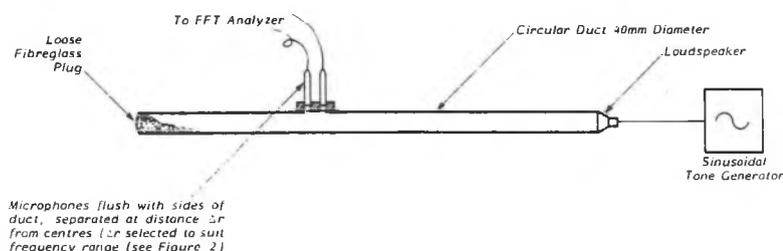


FIGURE 4
 ARRANGEMENT FOR ACOUSTIC INTENSITY CALIBRATION

The microphones were mounted, at their preselected spacing, in the side of the duct. A sinusoidal tone (1 kHz) generated by a loudspeaker at one end of the duct propagated free progressive waves

past the microphones. The other end was filled with loose fibre-glass wool to reduce reflection of sound. With the analyzer programmed, the intensity of the tone signal was obtained. Since the level of sound pressure and intensity at the microphones in the duct are essentially the same (Equation 3) the intensity level (re the standard 10^{-12} watts/m²) was read directly from the sound level meter that was designated for reference in section 3.3. The analyzer was then "calibrated" to read the same intensity level as the meter with the necessary corrections for barometric pressure and temperature.

The level of the calibrating tone was set 40 dB higher than the ambient level in order to minimize error due to sound waves incoming at the open end of the tube.

Successive acoustic intensity level measurements were made by simply collecting the time-averaged cross-spectrum and executing the programmed phase and amplitude mismatch compensation and intensity level calculations. The direction of sound flow was shown by displaying the imaginary part of the cross-spectrum.

4.0 EXPERIMENTAL RESULTS

A few experiments were performed in the laboratory in order to verify the procedure described above. The duct used for obtaining the transfer function was a plastic pipe, 40.1 mm in diameter, and 2400 mm long. The same duct was used for intensity calibration by replacing the end fitting with a section of pipe as illustrated in Figure 4.

The intensity level of a random noise source was measured using the conventional method (Equation (2)) and compared to the acoustic intensity method (Equation (4)). The test was made inside the duct described above, and in an anechoic chamber. The microphones were 13 mm in diameter, the spacing was 30 mm and the expected useful frequency range, from Figure 2, was 270 to 3600 Hz. The results within this range were in very good agreement, see Figure 5(a), (b).

Figure 6 shows the sound power level of a small loudspeaker generating random noise measured by the acoustic intensity method and by the standard anechoic room method which requires 20 measurement points (ANSI S1.35/12/). The agreement was again good over the expected useful frequency range of the probe. The time taken to obtain these results was estimated to be about 1/2 h compared to 2 h for the standard method.

The sound power level of a small loudspeaker generating random noise was obtained with and without the presence of an interfering, uncorrelated noise from jig-saw. The interfering noise level was 10 dB higher, overall, than the loudspeaker, see Figure 7. The agreement between the sound power levels of the source only (standard method) and of the source with the interfering noise (acoustic intensity method) was satisfactory - ie within 3 dB, except at frequencies where the level of the interfering noise exceeded that of the source by more than 10 dB. See Figure 8.

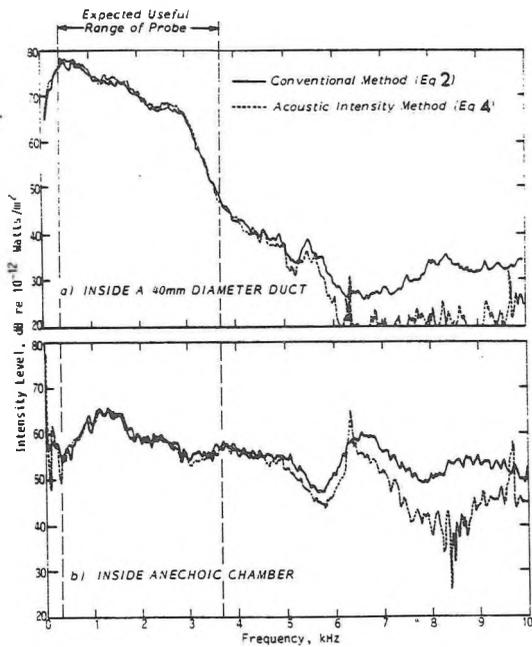


FIGURE 5
COMPARISON OF ACOUSTIC INTENSITY SPECTRUM LEVELS
USING THE CONVENTIONAL AND THE CROSS-SPECTRUM METHODS
UNDER CONTROLLED CONDITIONS

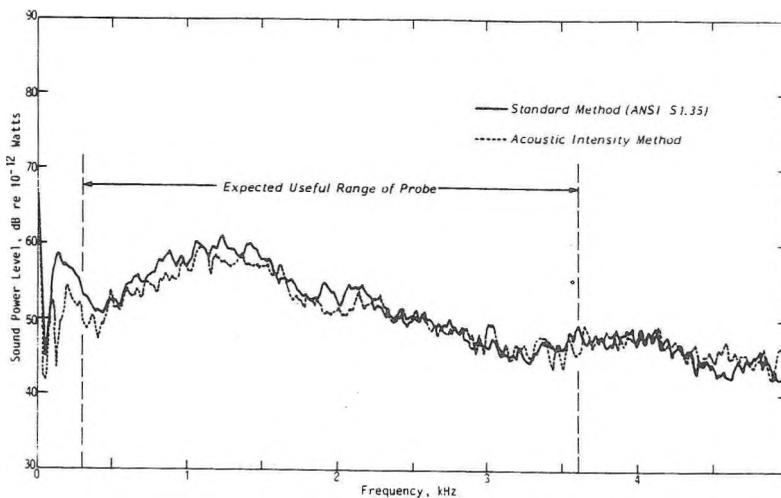


FIGURE 6
COMPARISON OF SOUND POWER LEVEL OF A NOISE SOURCE
USING STANDARD METHOD AND ACOUSTIC INTENSITY METHOD

Figure 9 illustrates the degree of correlation expected with this technique at low frequencies. Although a wider microphone spacing would expand this range further, the result here indicates very acceptable accuracy with the configuration used down to about 100 Hz.

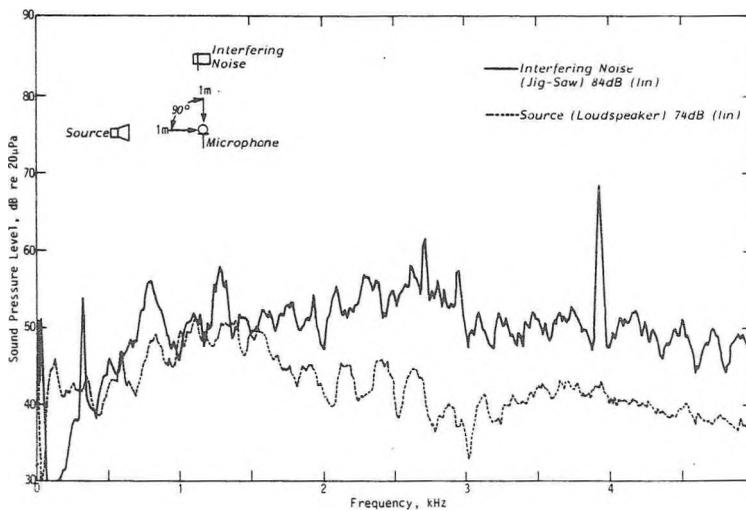


FIGURE 7
SOUND PRESSURE LEVEL OF SOURCE AND OF INTERFERING NOISE AT 1M

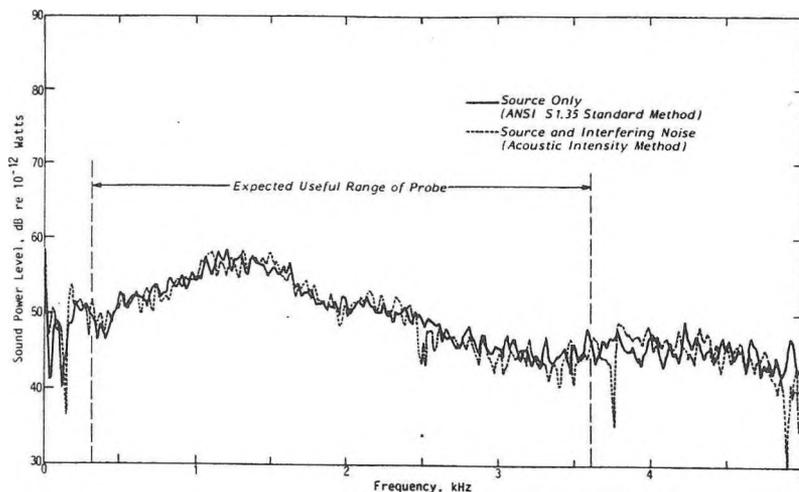


FIGURE 8
COMPARISON OF SOUND POWER LEVEL OF A NOISE SOURCE AND OF THE SOURCE
IN THE PRESENCE OF AN INTERFERING NOISE USING ACOUSTIC INTENSITY METHOD

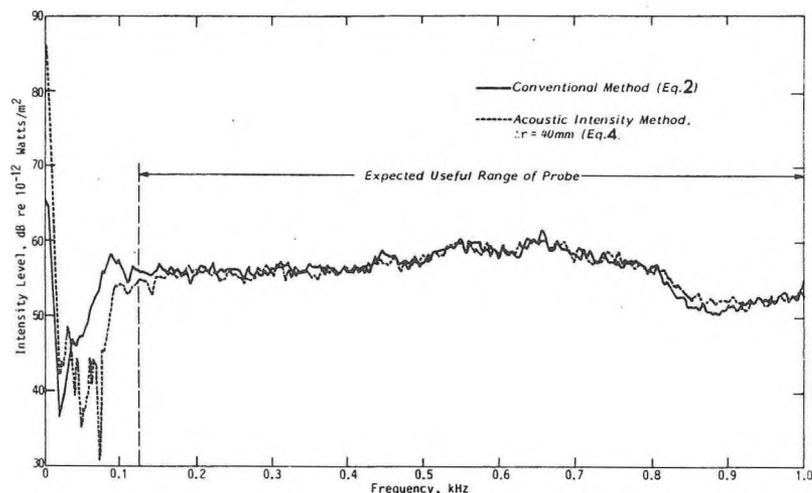


FIGURE 9
EXAMPLE OF CORRELATION OF ACOUSTIC INTENSITY METHOD AT LOW FREQUENCIES

5.0 CONCLUSIONS

A relatively simple method was developed for making acoustic intensity level measurements directly in $\text{dB re } 10^{-12} \text{ watts/m}^2$, using a special calibrator and an FFT analyzer with complex mathematics capability.

The parameters to be selected for a particular application, ie microphone size, spacing and distance from source, and their corresponding frequency and dynamic ranges, have been identified (see Table II and Figure 2).

Comparison of the acoustic intensity technique with the conventional (pressure squared) method have shown the technique under a variety of conditions to be well within practical accuracy.

The determination of the sound power of a source required about one quarter of the time compared with the conventional anechoic room method.

ACKNOWLEDGEMENTS

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