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AN INTENSITY METER ON A CHIP

C.C. Boucher and S.J. Elliott

Institute of Sound and Vibration Research, The University of Southampton, Southampton SO9 5NH, England.

1. INTRODUCTION

The two microphone technique of acoustic intensity measurement is now well accepted [1,2]. There are two principal methods of obtaining the intensity from the microphone signals:

1. The direct method [3]
2. The FFT method [3,4]

Sound intensity is defined as the time averaged product of pressure and velocity

$$\tilde{I} = \overline{p.u}$$

Using the two microphone, finite separation method this is approximated to

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

where Δr is the microphone separation and p_A and p_B are the two microphone signals.

In the frequency domain this is expressed as

$$\tilde{I} = -\frac{1}{\omega\rho\Delta r}\,\mathrm{Im}\,G_{AB}$$

where $\mathrm{Im}G_{AB}$ is the imaginary part of the cross-spectrum between the two microphone signals.

This paper is concerned with the programming of one of the new generation of high speed DSP microprocessors to perform the various functions needed in intensity measurement. Initial results for the direct method are presented.

2. DIGITAL IMPLEMENTATION

Figure 1 illustrates the stages necessary in implementing the two microphone direct method digitally. The microphone signals are amplified, passed through anti-aliasing filters and then analogue to digital converters. Hence the analogue path (and its associated matching errors) is reduced to a minimum. Once the signals are digitised, the sum and difference operations become trivial, but more importantly and unlike the analogue case, the multiplication also becomes trivial with the current generation of DSP chips. The filters (A-weight and octave) are implemented digitally and are thus perfectly matched. This eliminates an important source of potential error in intensity meters.

The integrator in the pressure difference path can cause instability due to d.c. drift, even when implemented digitally [2]. This would cause a practical, finite precision, implementation to saturate if any d.c. offsets were present. This instability is caused by a pole at d.c. (0 Hz). If, however, the filters in Fig. 1 are arranged so that they have a <u>zero</u> at d.c. this compensates for the unstable pole and a stable system results. If these filters are also moved so that they lie after the sum and difference operations, Figure 2, the integrator's transfer function can be combined with that of the filters, and only a single digital filter need be implemented in each path.

3. FILTER DESIGN

From B.S. 5969 (1981), which gives a pole-zero specification, the ideal analogue transfer function for an A-weight filter is

$$H_A(s) = \frac{s^4}{(s + \sigma_1)^2(s + \sigma_2)(s + \sigma_3)}$$

where σ_n are the specified pole positions; $\sigma_1 = 129.434$ rads^{-1}; $\sigma_2 = 676.699$ rads^{-1} and $\sigma_3 = 4636.362$ rads^{-1}.

Using the bilinear transform to give a digital approximation to this analogue filter gives the z transform of the digital filter's response as:

$$H_A(z) = \frac{G_A(1 - z^{-1})^4}{(1 - \frac{2 - \sigma_1'T}{2 + \sigma_1'T} z^{-1})^2(1 - \frac{2 - \sigma_2'T}{2 + \sigma_2'T} z^{-1})(1 - \frac{2 - \sigma_3'T}{2 + \sigma_3'T} z^{-1})}$$

where σ_n' are the prewarped pole frequencies, G_A is the gain of the filter and T is the sampling period. To maintain high accuracy this was implemented, as described below, with a sampling frequency of 20 kHz ($T = 5 \times 10^{-5}$ sec), as a series of four first order sections. The measured frequency response of the resulting system is shown in Figure 3. Results are within tolerance for a type 0 instrument above 20 Hz.

The transfer function of an ideal, analogue, integrator is given by

$$I(s) = \sigma/s$$

If this is combined with the transfer function for the A-weighting filter then the net response is

$$H_A(s)I(s) = H_{AI}(s) = \frac{\sigma s^3}{(s + \sigma_1)^2(s + \sigma_2)(s + \sigma_3)}$$

Bilinearly transforming this analogue filter gives:

$$H_{AI}(z) = \frac{G_{AI}(1 - z^{-1})^3(1 + z^{-1})}{(1 - \frac{2 - \sigma_1'T}{2 + \sigma_1'T} z^{-1})^2(1 - \frac{2 - \sigma_2'T}{2 + \sigma_2'T} z^{-1})(1 - \frac{2 - \sigma_3'T}{2 + \sigma_3'T} z^{-1})}$$

Again this was implemented as four first order sections and the measured frequency response is shown in Figure 4.

By taking the difference between the results in Figure 4 and Figure 3, the integrator's response can be calculated (Figures 5a and 5b). Figure 6 shows the warping of the high frequencies caused by the bilinear transform [2]. This was reduced to an acceptable level (= 1 dB at the maximum frequency of interest, 4 kHz), by selecting a higher sampling frequency than strictly required according to the Nyquist criterion.

4. IMPLEMENTATION AND RESULTS

The intensity meter was implemented on an expansion board for an IBM compatible PC made by Loughborough Sound Images (LSI) which uses the Texas Instruments TMS 32020 DSP chip. A two channel multiplexer was also built to obtain the two microphone signals (Figure 7).

The A-weighted intensity meter including time averaging for sound pressure level and sound velocity level (as well as intensity) uses approximately 60% of the processing power of the TMS 32020 at a sampling frequency of 20 kHz.

The PC is used to simultaneously display sound intensity, pressure and velocity levels.

As an initial test the microphones were connected to both the digital and an analogue intensity meter. Measurements were taken at varioius points around a large vacuum cleaner and are shown below.

Analogue		Digital	
Intensity (dBA)	SPL (dBA)	Intensity (dBA)	SPL (dBA)
84	86.5	84.8	86.5
92	93	93.3	92.2
81	85	81.8	85.0
80	84	81.0	83.6
76.5	83	76.4	83.1
83	85	84.0	85.0
87	88.5	88.0	88.1
90.5	91	91.5	90.3

5. FUTURE WORK

The digital implementation of octave filters should be possible and provided they have a zero at d.c. a similar technique to that used above can be applied to implement a stable integrator.

A linear setting could be realized by making the filters high pass (above the lowest frequency of interest) and ensuring that they also have a zero at d.c.

From timings of FFT's on the TMS 32020 it should also be possible to implement the FFT intensity method and use the PC to display the resulting spectrum over this frequency range.

REFERENCES

1. S. GADE 1982 Brüel and Kjaer Technical Review No. 3-1982. Sound Intensity (Theory).
2. S. GADE 1982 Brüel and Kjaer Technical Review No. 4-1982. Sound Intensity (Instrumentation and Applications).
3. F.J. FAHY 1977 Noise Control Engineering, 155-162. A technique for measuring sound intensity with a sound level meter.
4. J.Y. CHUNG 1978 Journal of the Acoustical Society of America 64, 1613-1616. Cross spectral method of measuring acoustic intensity without error caused by phase mismatch.

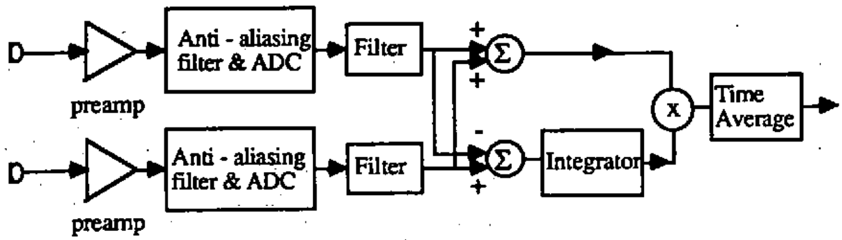

Figure 1 : A Digital, Two Microphone, Direct Method Intensity Meter.

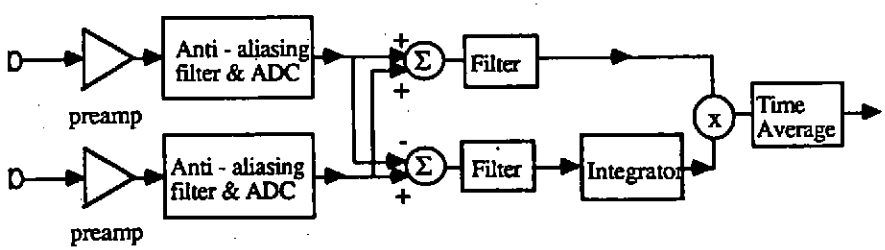

Figure 2 : A Digital, Two Microphone, Direct Method Intensity Meter.

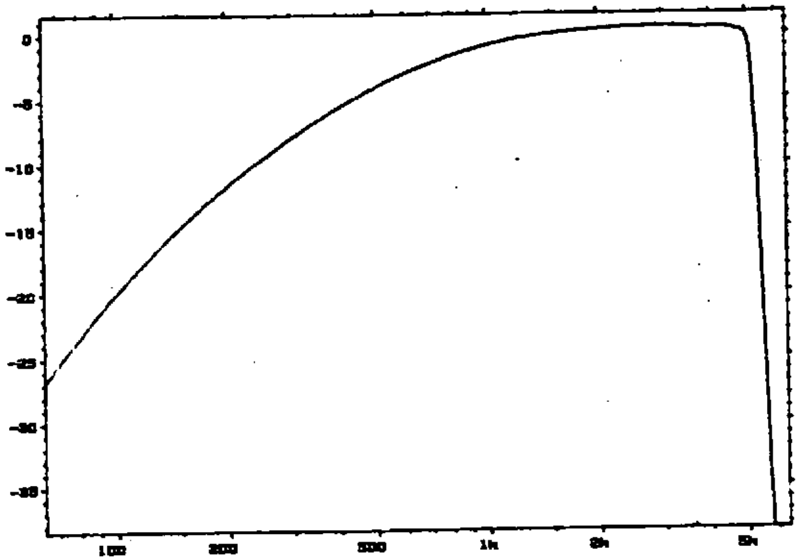

Figure 3. Measured frequency response of A-weighting filter.

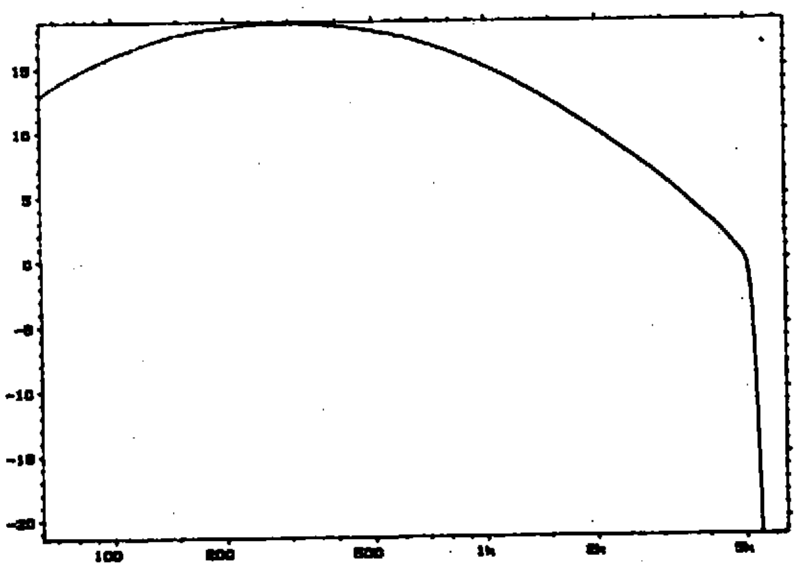

Figure 4. Measured frequency response of A-weighting filter and integrator.

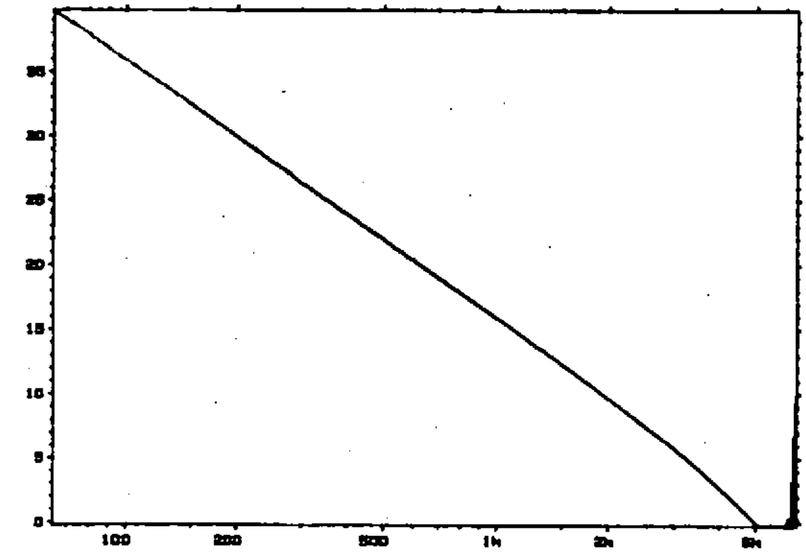

Figure 5a. Frequency response of integrator.

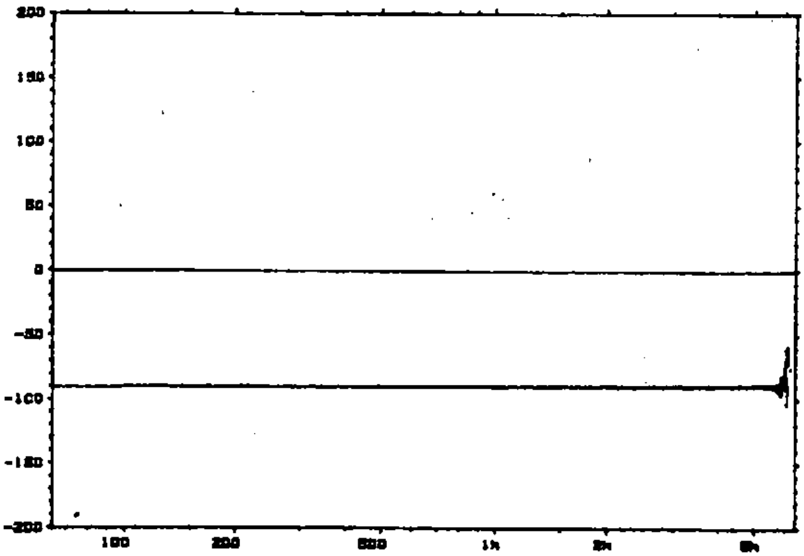

Figure 5b. Phase response of integrator.

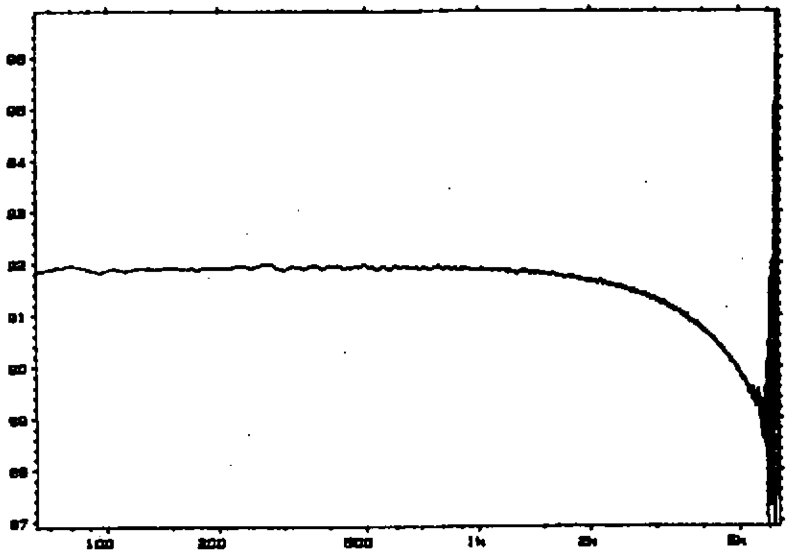

Figure 6. Frequency response error of integrator.

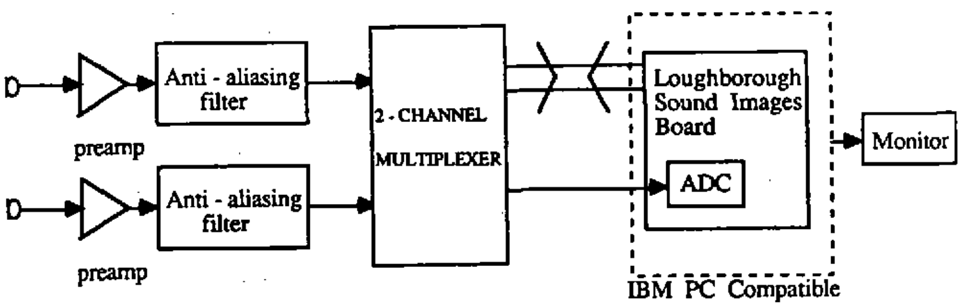

Figure 7 : Implementation of a Digital Intensity Meter.