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FROM SOUND LEVEL METER TO SOUND METER

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0 ABSTRACT

The Sound Level Meter is an instrument capable of estimating statistics of the pressure signals detected by its microphone. The application of computer technologies to acoustic instrumentation has expanded the range of statistical operations that can be applied, leading to today's sophisticated handheld acoustic analysers. This paper describes a further evolution from that starting point to a new family of instruments capable of extracting information from the sound. Using techniques similar to speech recognition and machine health prognostics, examples are presented which demonstrate the ability of a realtime instrument to listen to sounds and extract information describing the sound or the sound's source. These examples range from the estimation of loudness, through detection of reverberation time, to the classification of flow induced noise within pipes. A further evolution of the instrument is predicted, in which the information extracted from the acoustic emissions of a process is exploited by a process control system. This would, for example, allow information within the noise emissions from an industrial process to be used to optimise process performance.

1.0 INTRODUCTION

Acoustic instrumentation is based upon the fundamental "building block" depicted in Figure 1, in which the output of a pressure microphone is connected to a pre-amplification stage. The amplifier has variable gain to facilitate range changes and in order that the electroacoustic gain of the system can be adjusted for calibration purposes. The output of the amplifier may be conditioned by a circuit (H_1 in Figure 1) which may, for example, provide filtering to apply standard weightings to the instrumented sound pressure.

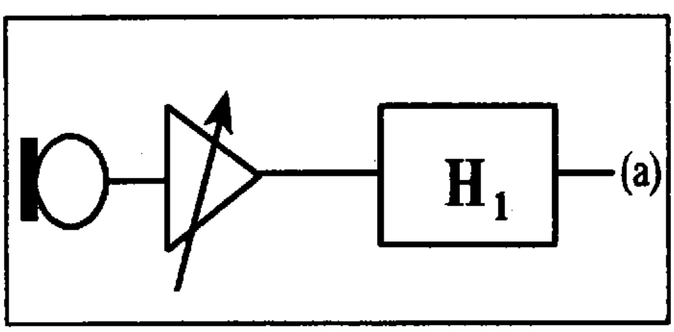

Figure 1 Input Block of Sound Meter

The measurement of Sound Pressure Level requires that a system having the functionality suggested in Figure 2 is connected to the basic input block described above. Connecting node (a) of Figure 1 to node (b) of Figure 2 results in a Sound Level Meter, as a signal proportional to the (weighted) sound pressure level detected at the microphone is available at node (d).

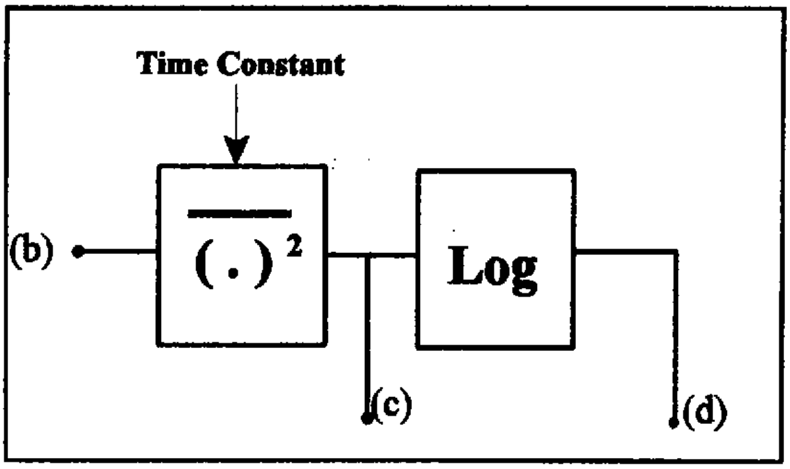

Figure 2 RMS detector and Logarithmic converter for Sound Pressure Level Measurement

With the advent of high performance, low power consumption digital signal processing, the system of Figure 2 may now be realised in digital electronics [1]. Whilst such a digital instrument will not necessarily offer superior performance to a sound level meter realised in analog electronics, it will offer some significant advantages in manufacturability. There is, however, an aspect of the digital sound level meter which a conventional instrument cannot offer; programmability. This results in limitless flexibility and configurability.

If a sound level meter is constructed using digital techniques, the hardware required to perform the functions of Figure 2 may be redirected to a completely different task by a simple change of software. A digital sound level meter may apply its processing power to ends other than the estimation of sound pressure level. Thus the sound level meter becomes a general purpose instrument capable of measuring other than sound pressure level - the sound level meter becomes a sound meter (the name sound meter is clumsy - others have suggested *acoustimeter* [2], *sonometer* and *audiometer* [3]).

It is the purpose of this paper to demonstrate by example some "sound meters", capable of extracting a range of different pieces of information from the detected acoustic pressure. All of the examples cited are developed upon the fundamental building block of Figure 1 and are feasible in the context of the computational load that can be bome by DSP devices in current digital SLM's.

2.0 FREQUENCY ANALYSIS AND LOUDNESS ESTIMATION

Analysis of the frequency components of an acoustic pressure has been a subject of considerable importance since the earliest days of acoustic instrumentation (see, e.g., [2]). The frequency components of the pressure detected by the microphone of Figure 1 could be identified by a (fast) Fourier Transform of the signal at node (a). Such narrowband analysis is possible on a portable instrument. Alternatively, the input node (e) to a parallel bank of digital bandpass filters (and a bank of the RMS detectors / log converters of Figure 2) can be connected to node (a) to provide parallel constant percentage frequency band analysis, as shown in Figure 3 [4]. Although this hardly constitutes a novel instrument, the vector of band pressure levels, at nodes (f_i), can be used as input to further computational blocks.

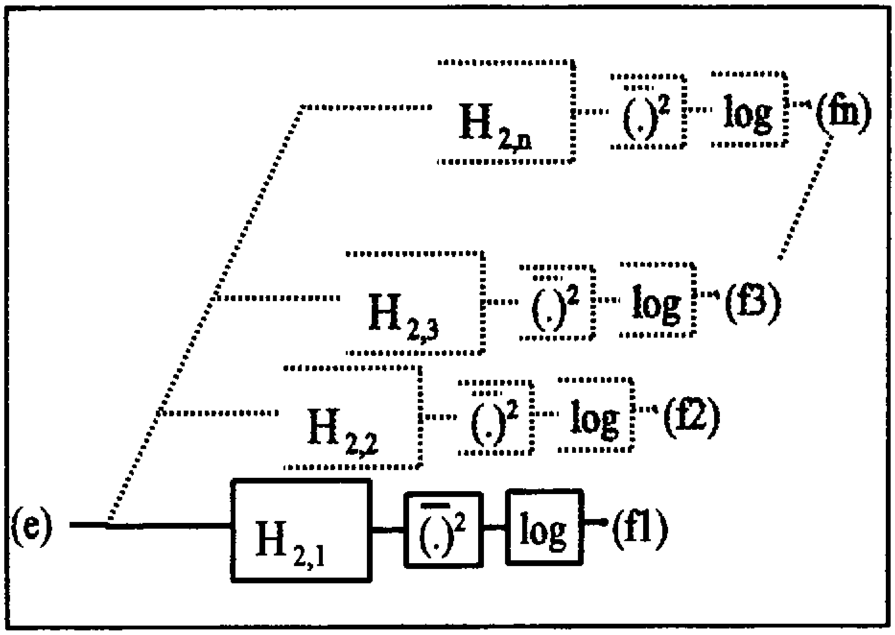

Figure 3 Band Spectral Analysis using a parallel filter bank

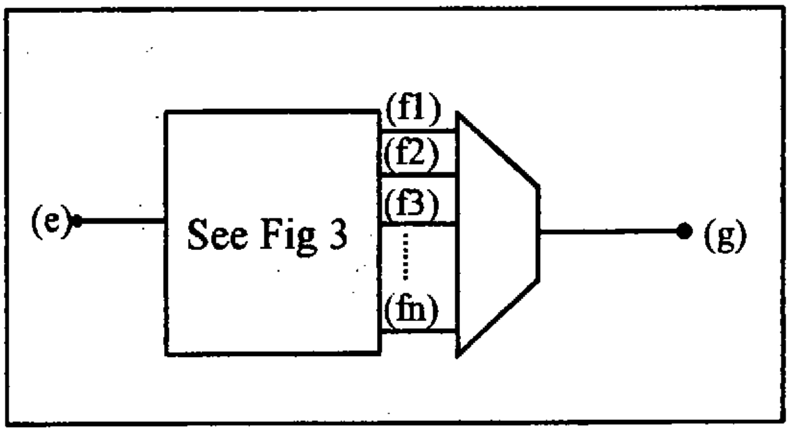

Figure 4 Computing a single valued function of band spectral levels - e.g. Loudness

As an example, the estimation of loudness in phons is a complicated non-linear function of the 1/3rd octave band pressure levels [5]. Connecting node (e) of the system shown in Figure 4 to the output node (a) of the input block (Fig 1) results in a loudness meter.

Figure 5 shows the performance of a loudness estimation in which the loudness is estimated (using the ISO 532 B procedure) from the band spectral levels between 25 Hz and 12.5 kHz using an Artificial Neural Network, trained to implement the non-linear relationship between band spectral levels and loudness. The network has been trained by exposure to a population of randomly generated 1/3rd octave band spectra. The scatter visible in Figure 5 is a reflection of the estimation errors made by the imperfectly trained neural network used in this proof-of-concept investigation. The errors could be reduced without limit to an acceptably low level by further training of the network (and, possibly, a modification of the network's architecture).

Once trained, the estimation of loudness using the neural network is a simple process requiring a fraction of the computational load of a direct application of the ISO 532 algorithm. This would allow the instrument to provide a continuous real-time estimate of loudness.

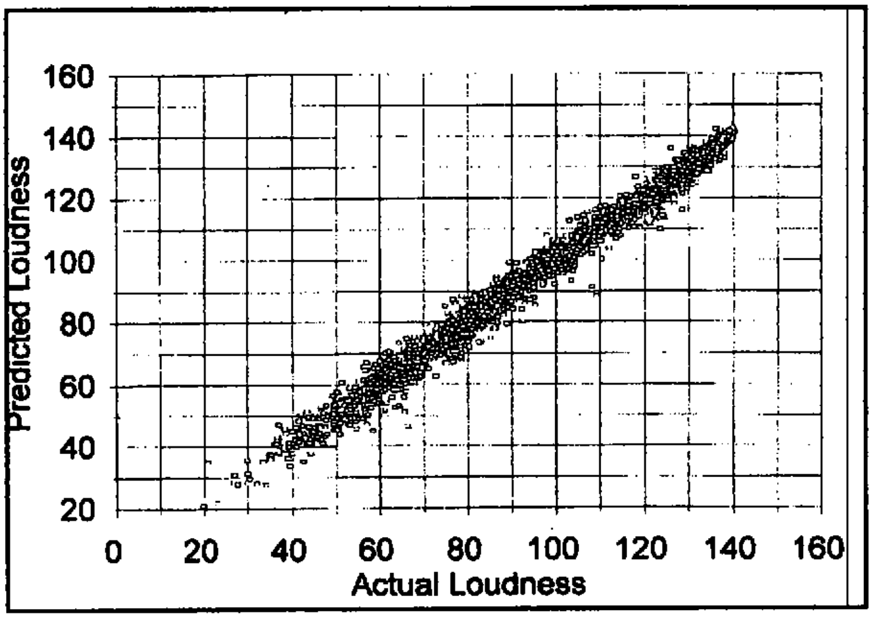

Figure 5 Performance of a Loudness Estimator Implemented by an Artificial Neural Network

3.0 ESTIMATING REVERBERATION TIME

Sound pressure levels or loudness levels express a short term time average of the pressure amplitude (or a function of it) - the time dependance is averaged out. However, as all listeners are aware, the time envelope of a sound contains much information, such that time averaging may itself destroy this temporal information content of a sound. It is possible to produce a range of instruments which explicitly seek temporal patterns in a detected acoustic pressure. In section 4, such an instrument is described which looks for temporal patterns over such a short timescale that it may equivalently be thought of as performing mixed time/frequency analysis. In the present section, an instrument which extracts temporal patterns over a longer timescale is described.

The envelope of an acoustic pressure is that low frequency waveform that amplitude modulates a pressure, such that the level (the short term RMS pressure) may vary as a function of time. Such an envelope waveform could be used, for example, to describe a vehicle pass-by; the envelope is zero valued until the vehicle becomes close enough to the observer to be detectable - then rises to a peak value - then falls away to zero as the vehicle moves away from the listener. An instrument capable of observing such envelopes is formed by connecting the signal at node (a) of Figure 1 to node (h) of Figure 6. The delay line in Figure 6 is tapped at regular time intervals, with the output of the taps fed to RMS detectors. The outputs of these detectors are applied to a network which produces a single value descriptor of the envelope (as sampled by the RMS detector outputs) at node (i).

Figure 7 shows the application of the instrument described above to the detection of Reverberation Times. The envelope of the reverberant pressure excited by changing stimulus levels in a bounded acoustic space has exponential growth and decay phases both of which have time constant defined by the reverberation time. The estimates of Figure 7 were produced by an artificial neural network observing the sampled envelope of simulated signals. These test signals were generated by convolving random pulsed excitation patterns with impulse responses of spaces with a range of reverberation times. The development of this system is the subject of continuing research (see acknowledgements).

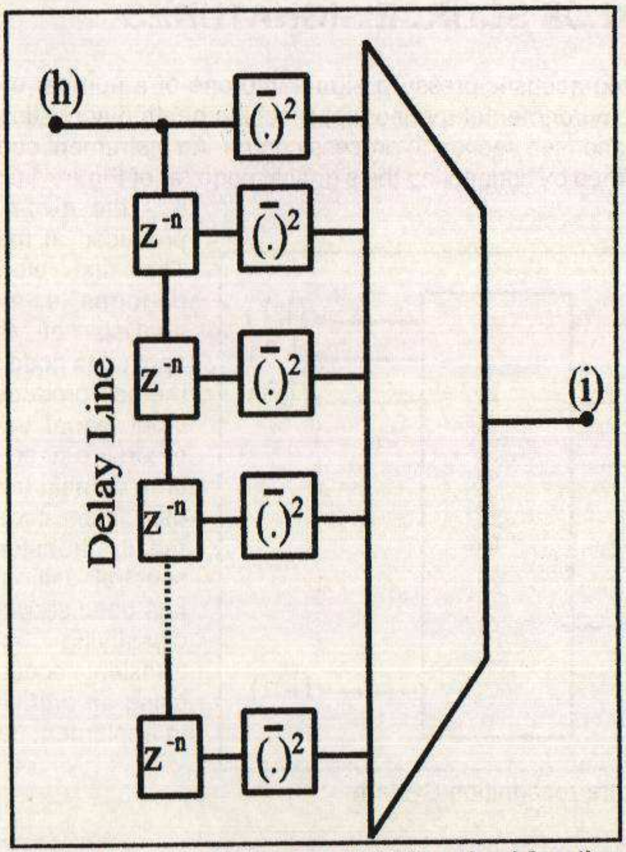

Figure 6 Computing a single valued function of signal envelope - e.g. Reverberation Time

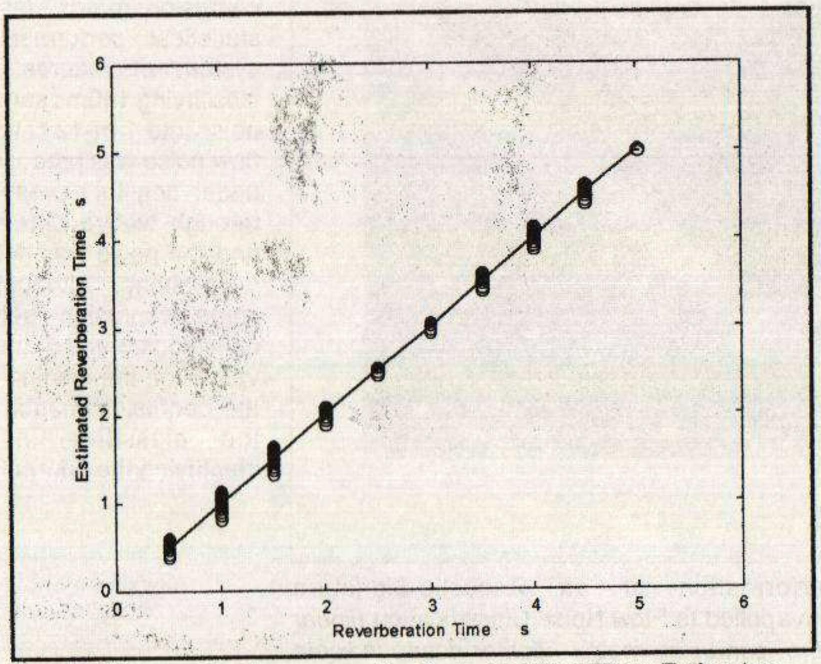

Figure 7 Performance of a Reverberation Time Estimator

4.0 RECOGNITION OF SOURCE SIGNATURES

The classification of a detected acoustic pressure signal into one of a number of categories has a number of potential uses in environmental monitoring, machine heath diagnostics and prognostics, manufacturing quality control and (see section 5) process control. An instrument capable of achieving such classification may be formed by connecting the signal at node (a) of Figure 1 to node (j) of Figure 8 - the 1-of-n classification is produced at the output node k_n. The first block of Figure 8 performs a maximum entropy analysis of the input signal applied to node (j). This statistical method produces a model of the input signal using a finite order recursive filter; it is essentially a time domain technique (although the filter coefficients can be readily transformed to give a spectral estimate). This system has been shown to be capable of classifying both steady and transient acoustic signals [6,7] using an artificial neural network to implement the classifier.

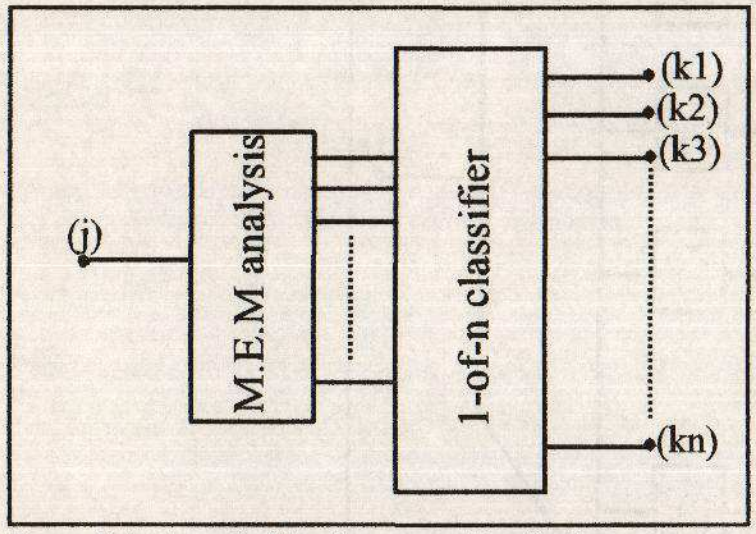

Figure 8 An Acoustic Signature recognition System

As a further example, Figure 9 is a graphical representation of a confusion matrix describing the statistical performance of the system of Figures 1 & 8 in classifying 100ms samples of flow noise into 1-of-12 categories. The flow noise was produced by water under constant pressure flowing through twelve different orifices and the noise was detected by a hydrophone. Although there is some confusion between flow conditions 3 and 5, the maximum values on the leading diagonal of the confusion matrix reveal that the classifier is correctly identifying the flow noise samples

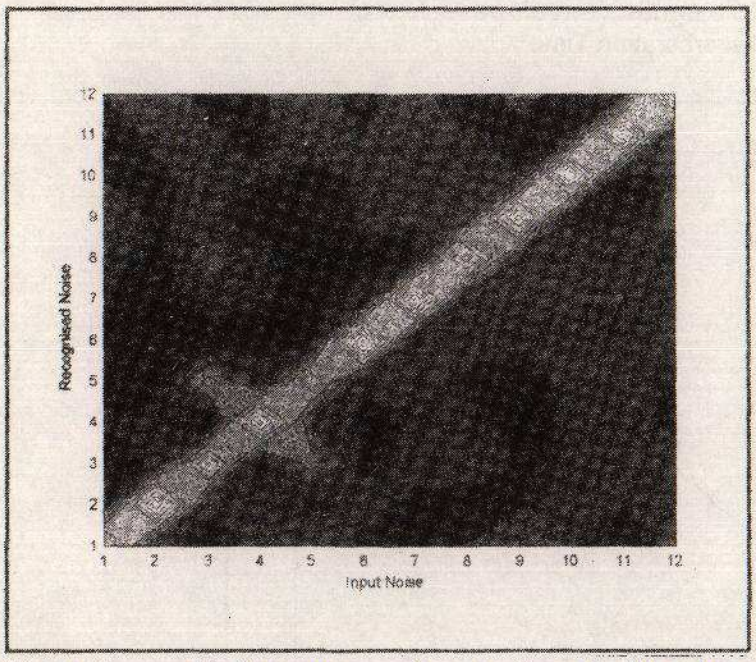

Figure 9 Performance of an Acoustic Signature Recognition System applied to Flow Noise Classification *(linear grayscale plot of the confusion matrix - lighter points indicate higher confidence of recognition)*

5.0 CLOSED LOOP CONTROL APPLICATIONS

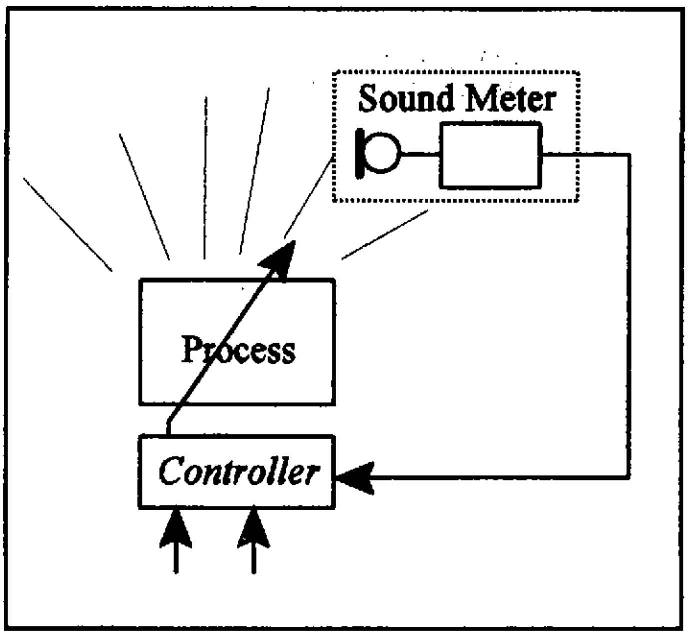

Figure 10 The Sound Meter in a Process Control Application

A principal motivation for the "sound meters" suggested in previous sections is the use of the derived information in process control applications, as shown in Figure 9.

Many processes produce noise which (to greater or lesser extent) carries information describing the state of the process. If this information can be extracted from the radiated noise, it may be used as one input to the process control apparatus. A number of physical and simulated applications of this idea are the subject of current research, including, for example, the automation of some of the parameters of an electric welding robot.

6.0 CONCLUDING REMARKS

Acoustic Signals convey much more information than can be described by sound pressure levels (or even band pressure levels) alone. In order that this information can be instrumented and used, it is necessary to construct new items of acoustic instrumentation. Given the power and flexibility of contemporary electronics, such instruments are simple to build. The challenge remains in the education of manufacturers and users alike to recognise the utility of these methods and to identify relevant applications.

Acknowledgements

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