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A COMPARISON OF SOME ESSEX ALGORITHMS FOR MAJOR INDUSTRIAL APPLICATIONS.

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INTRODUCTION

Most "active" systems employ some form of signal processing or filtering to achieve the correct cancellation waveform. Processing is needed to modify the original noise waveform to compensate for the distortion which is added to it by the loudspeaker and by the acoustic environment. It is fundamental that some time is taken to acquire the signal sample and to process it. This processing time puts constraints on the physical configurations to which active cancellation can be successfully applied.

Prior to the mid-1970s, two main configurations were possible:

a) The direct feedback configuration, consisting of a microphone, amplifier and loudspeaker in a simple feedback loop. (Ref.1) This configuration produces local cancellation at the loudspeaker.
b) The duct configuration, consisting of an upstream microphone, signal processing filter, and downstream loudspeaker(s).(Ref 2)

In the mid-70s a further technique with wide-ranging applications was added to the list, as a result of work at Essex. (Ref 3) This technique, for cancelling repeating noise or vibrations of any form, inside or outside a duct, uses the time between repeat cycles to modify the signals. It assumes the following repeat cycle will have a similar waveform to the current repeat cycle. This assumption is valid for a wide range of low frequency problems associated with machinery, vehicles, vessels and aircraft.

GENERAL FEATURES OF THE PERIODIC TECHNIQUE

Figure 1a illustrates how the waveform synthesiser module operates. On its own, it would only be suitable for a time invariant waveform, in which case the timing information could be derived from a clock. (The adaption algorithms shown in b, c, and d of Figure 1 are discussed later.)

However, if the repeat rate of the noise waveform varies, there will be drift between the cancellation waveform and the noise waveform, and some means of locking the two together is needed. A suitable sensor close to a toothed wheel on the offending machine can provide tens or hundreds of pulses per repeat cycle. The cancellation system can then produce a sample of the cancellation waveform every time one of these pulses is received. The cancellation waveform is then continuously adapted to take account of changes in the acoustic environment. This is accomplished by plug-in algorithms, the choice of which depends both on the transducers and on the acoustic environment.

Pre-set Adaption. The algorithms are currently programmed to start with a blank cancellation waveform; it would also be possible to infer an approximate initial waveform from knowledge of the machine's operating conditions, and thus achieve a degree of cancellation at switch-on.

Noise Immunity. Greater degrees of cancellation can be achieved in the presence of unsynchronised background noise if the residual waveforms are (synchronously) averaged over a number of repeat cycles. Thus, there is a trade-off between the degree of cancellation and the adaption time required for cancelling stationary waveforms, and there is an optimum compromise when a particular rate of change of waveform must be accommodated despite some given level of background noise.

Interacting Systems. In many cases, several cancellation systems operating in close proximity (e.g. a set of vibration mounts) will only interact to a limited degree, and will produce a stable cancellation condition. With particularly severe interaction, it is necessary to take precautions such as allowing only one system to adapt at a time, or even characterising the interaction between the systems.

TYPES OF ADAPTION ALGORITHM

The algorithm should be selected with due consideration for both the transducers and the acoustic environment, since the stability and degree of cancellation depend on these features as well as on the algorithm. The features of the three major classes of adaption algorithm are illustrated in Figure 2, and an example of each is given in Figures 1b, c and d, as follows:

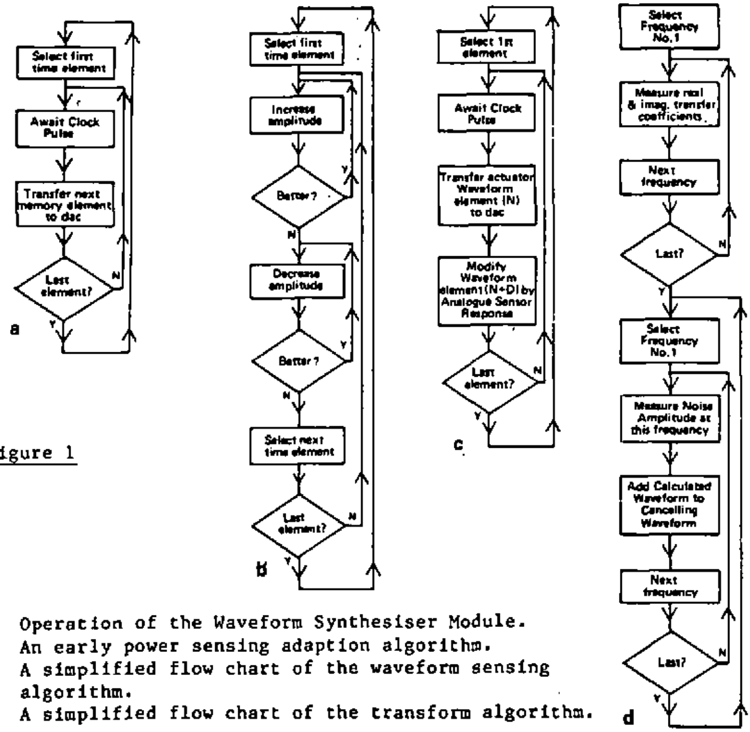

Figure 1

a. Operation of the Waveform Synthesiser Module.
b. An early power sensing adaption algorithm.
c. A simplified flow chart of the waveform sensing algorithm.
d. A simplified flow chart of the transform algorithm.

POWER SENSING. (Figure 1b) The amplitude of each time element (or other waveform parameter) is adjusted incrementally in turn and the adjustment is retained or abandoned depending on whether the noise power has reduced or not.
WAVEFORM SENSING. (Figure 1c) Each residual waveform element is used to adjust, in real time, the amplitude of its corresponding actuator waveform element. This method is functionally similar to the direct feedback method of Ref 1, but with compensation for the transport delay from actuator to residual sensor.
TRANSFORM METHOD. (Figure 1d) The transfer function between the actuator and the residual sensor is characterised in terms of, for example, Fourier or Walsh spectra. This is then used to deduce the actuator signal required to cancel the measured noise waveform.

Algorithm:	Power Sensing	Waveform Sensing	Transform Method
Assumption:	Little Unsynchronised Interference	Frequency Independent Loop Delay	Reasonable Amplitude Linearity
Efficiency:	Reasonable (20dB)	Excellent (50dB)	Good (35dB)
Interference Immunity:	Poor	Excellent	Good
Frequency Range:	Limited to ≈8 harmonics	Unlimited	Unlimited
Currently Achieved Adaption Times:	30 Seconds	Less than 1 second	1-10 Seconds
Actuator Requirement:	Can be non-linear	Phase Linearity	Amplitude Linearity
Examples of Applications:	Novel Actuators.	Electronic Signal Enhancement. Headphones. Engine Inlets.	In-cab Systems. Vibrating Structures. Transformers.

Figure 2

Practical Experience of the Three Main Classes of Algorithm

CONCLUSION

By taking advantage of inherent features of the noise and the environment, cancellation of noise and vibration has been achieved in circumstances that were not previously possible.

For active methods to be used to best advantage, an informed judgement must be made on:

(1) Optimising the mix of active and passive measures;
(2) Choosing the most relevant active technique (see Introduction); and
(3) Determining the algorithm most suited to the acoustic features of the problem.

REFERENCES

[1] H.F. Olson & E.G. May, 'Electronic Sound Absorber', J.Acoust.Soc.Am. 25, 6, 1130-1136 (1953)

[2] H.G. Leventhall & Kh. Eghtesadi, 'Active Attenuation of Noise: Dipole and Monopole Systems', Proc.Inter-Noise 79, 175-180.

[3] G.B.B. Chaplin, R.A. Smith & R.G. Bearcroft, 'Active Attenuation of Recurring Sounds', UK Patent 19717/76.