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NON DIGITAL PROCESSING OF THE IMPULSE RESPONSE

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Introduction

The advent of very fast A/D conversion and large memories has made possible the capture and processing of the impulse response of rooms by digital computers. However, there are some ingenious analogue techniques which it seems worth recording for the benefit of those without access to large computing facilities. Three such techniques are described in this paper. The first concerns the direct realisation of the Schroeder Integrated Impulse Response theorem. A simple relation for the prediction of clarity which derives from it is also described. The second technique exploits the relation between the time domain (impulse response) and frequency domain (transmission function) descriptions of the acoustic behaviour of a room. The fourth section presents the application to acoustic scale models of a technique described by Kuttruff for the derivation of the autocorrelation function.

Integrated Impulse Response

A theorem published by Schroeder in 1965 (1) relates the average of an infinite number of random noise decays to an integral of the impulse response:

$$\langle s^2(t)\rangle = N\int_t^{T+t} p^2(t)\,dt \qquad \ldots\ldots\ldots\ldots\ldots 1$$

This may be regarded as a generalisation of Cambell's theorem in the field of electronic noise. The terms represent the following: s(t) is the response at time t to a random noise signal started at -T and stopped at t = 0; r(t) is the impulse response of the room following impulse exitation at time t = 0. The brackets indicate an average over many decays and the squaring of the terms indicates power summation. Fig. 1. is a graphic representation of the theorem. Random noise may be considered as a series of impulses; at any time t the total (noise) response during the decay is made up of the sum of the energies left over from the series of impulses which make up the random noise. The random interval between impulses in any one noise process allows us to ignore phase effects, so that only energy ($p^2(t)$) needs to be summed. It will be seen from the diagram that summation along the staggered row of impulse responses is equivalent to forward integration along a single impulse.

To realise this theorem by digital techniques would require digitisation and storage of the complete impulse response - perhaps an 8K, 16 bit store. However, at least three techniques are available which use analogue integration. The first requires a bi-directional tape recorder. The impulse response is recorded, then replayed in the reverse direction. The integral can then be implemented with the limits reversed so that after passing the signal through a squaring circuit it can be integrated continuously. The output of the integrator represents the 'average noise decay' curve with reversed time direction.

A second technique passes the impulse response directly to a squaring and integrating circuit. The direct output gives a 'growth' curve representing the build up of reverberant sound due to a random noise source. This signal is passed to a simple summing circuit and to a unit which delays the signal by a few hundred milliseconds, then inverts it before passing it to the summer. As a result the output of the summer gives first a curve representing the average random noise growth, followed by the decay curve.

The third technique is a variation of this; the impulse is fed to the room once and this first response is led to a simple integrator. The feedback capacitor is then reversed and the room excited a second time. As the second response is integrated the net output is reduced to zero. Fig. 2. shows the voltage traces obtained in each of these cases.

Clarity Relationship

A idealised form for the impulse response of a room can be written:

$$p^2(t) = \sum_{n=1}^{n=\infty} k^2 a_n^2 \, \delta(t - t_n) \qquad \text{............ 2}$$

Here the Dirac delta function $\delta(t)$ is everywhere zero except at $t = 0$ (where it takes the value 1), a_n is the amplitude in an echogram of the nth reflection, t_n is its arrival time and k is a constant relating sound pressure to amplitude. According to the work of Haas, Neise and Theile the clarity fraction is given by:

$$C = \int_0^{50ms} p^2(t)\,dt \Big/ \int_0^{\infty} p^2(t)\,dt \qquad \text{............ 3}$$

If the idealised response from equation 2 is substituted for the numerator and the Schroeder theorem used to relate the denominator to the reverberant (steady state) level, it can be shown after a little algebra that:

$$C = \frac{V}{100\pi r^2 T}\left[1 + \left(\frac{a_2}{a_1}\right)^2 + \left(\frac{a_3}{a_1}\right)^2 + \text{-----} \left(\frac{a_{last}}{a_1}\right)^2\right] \ldots \qquad 4$$

where a_{last} is the amplitude of the last reflection to arrive within 50 milliseconds, V is the volume of the room, T its reverberation time and r the distance (metres) from the source to the point at which the clarity is to be calculated. The amplitude of the direct sound is a_1; the amplitudes of the reflections (a_n) can be measured in an existing hall or calculated for a project, and hence the clarity determined.

Impulse Response Derived From Transmission Function

It is well known that the transmission (system) function of a room is related to its impulse response by:

$$H(i\omega) = \int_{-\infty}^{+\infty} p(t) \exp(-i\omega t)\,dt \qquad \text{.. .... 5}$$

and this is often used to generate transmission functions from impulse responses. However, for the ideal impulse response of 2 above this becomes:

$$H(i\omega) = \sum_n a_n \cos \omega t_n + i \sum_n a_n \sin \omega t_n \qquad \text{.......... 6}$$

Fig. 3 provides a graphical interpretation of this relationship: the amplitude and phase of the frequency response of the room is made up of the vector superposition of a series of terms each representing one of the paths giving rise to the terms of the ideal impulse response. If the input signal to the room is represented by:

$$A\cos\omega t$$

and the signal received by a microphone at the point of interest is

$$B\cos(\omega t + \phi)$$

an analogue multiplier can be used to form the product:

$$A\cos\omega t . B\cos(\omega t + \phi)$$

which equals:

$$\frac{AB}{2}\left[\cos(2\omega t + \phi) + \cos\phi\right]$$

A low pass filter may be used to remove the term containing $2\omega t$, leaving a signal that varies only as ω changes. If A is kept constant and ω changed at constant speed the output circuit delivers a slowly varying periodic signal (AB cos ϕ) the power spectrum of which (with linear frequency scale) mimics the impulse response. This will be apparent from Fig. 4 which shows the superposition of the contributions of the various paths in the Argand diagram. As ω increases the constituent vectors rotate at rates proportional to the time delays t_n, so that the angular 'frequencies' of each component of the resulting fluctuation as ω increases are determined by these t_n and the 'amplitude' of the components is determined by the a_n.

The transformation may be simply realised experimentally by recording the output fluctuation generated by a slowly swept frequency on to a tape loop and relaying this loop with large speed multiplication to a conventional spectrum analyser. The resulting spectrum then represents the original impulse response.

The Autocorrelation Function

Kuttruff has provided an interesting method for generating the autocorrelation function of the transmission path of a room from the impulse response (Ref.2). An impulse is delivered to a loudspeaker in a room; the resulting impulse response is then picked up by a microphone and recorded. This recording is then replayed with reversed time through the same loudspeaker; the signal picked up by the microphone is now the autocorrelogram of the transmission path in real time. The results of applying this technique in a scale model will be reported.

References

1) M.R.Schroeder, J.Acoust.Soc.Amer. v.37 p409-412 (1965)
2) H.Kuttruff, "Room Acoustics", pub. Applied Science 1973 p210-214

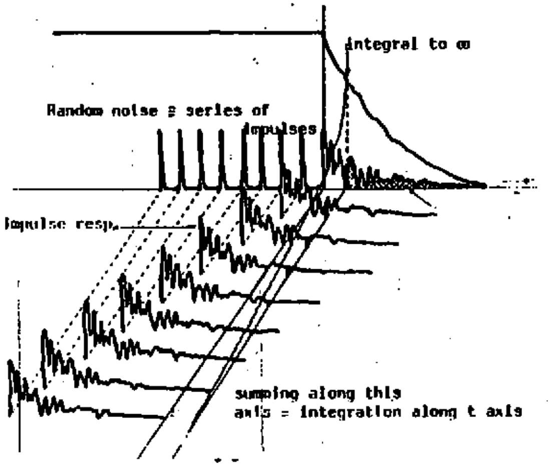


Figure 1

Figure 2

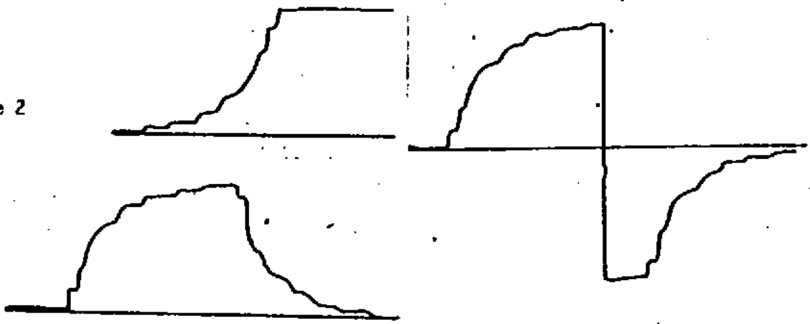
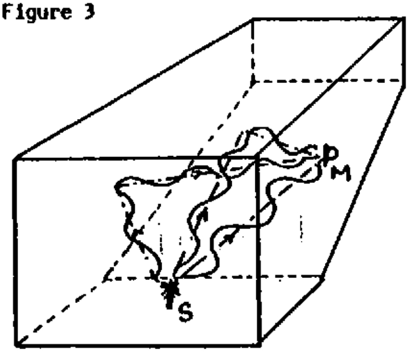


Figure 3



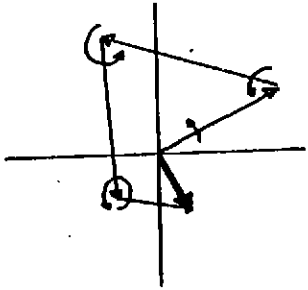


Figure 4

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THE APPLICATION OF DIGITAL TECHNIQUES TO THE DESIGN AND TESTING OF LOUDSPEAKERS

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The evaluation of a loudspeaker's transfer function or frequency response (amplitude and phase) via digital processing of its measured impulse response is now an established technique. Work first started on this method nearly ten years ago, early in 1971, and interim reports were given in 1973 (1), 1975 (2), 1977 (3) and 1979 (4).

The purpose of this paper is to provide an up to date assessment of this method and its usefulness for both laboratory and production testing of loudspeakers.

Impulse Response Measurement

This method assumes that a loudspeaker may be treated as a linear time invariant system and as such may be represented either in the time domain by its impulse response, h(t), or in the frequency domain by its frequency response or transfer function, H(f). h(t) and H(f) are related mathematically through the Fourier transform and once either is known, the system output for any arbitrary input may be predicted.

The direct digital measurement of a loudspeaker's impulse response may be obtained using the general arrangement shown below in Figure 1.

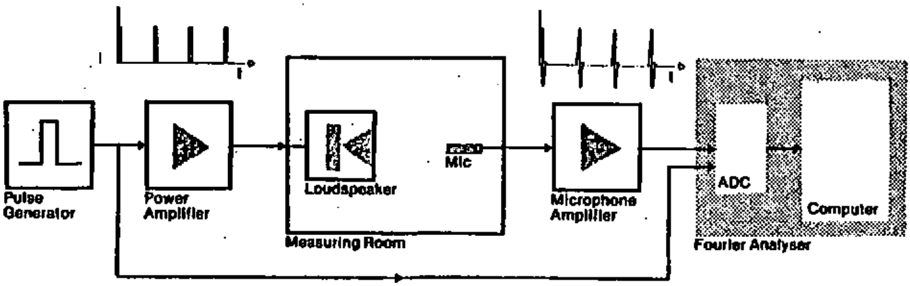

FIGURE 1

The loudspeaker under test is placed near the centre of the test room and excited with a short electrical pulse (Figure 2a). The resulting acoustical waveform is picked up by a measuring microphone (Figure 2b), digitized by an analogue-to-digital converter (ADC), and stored in a digital processor. The room need not be anechoic but should be large enough so that the response of the drive unit has died down to a negligible level before the first room reflections arrive at the microphone at time t_F (Fig 2b). Due to the low energy content of the test pulse, signal averaging is usually employed to obtain a satisfactory signal-to-noise ratio (Fig 2c). This is done by repeating the test pulse at equal time intervals just greater than the reverberation time (RT) of the test room, and adding the resulting responses so that the required signal is enhanced with respect to the random background noise. It is important to note that signal averaging is ineffective when the background noise contains either periodic components, such as line frequency, or impulsive interference, and where these are present special care must be taken to eliminate them if accurate results are to be achieved.

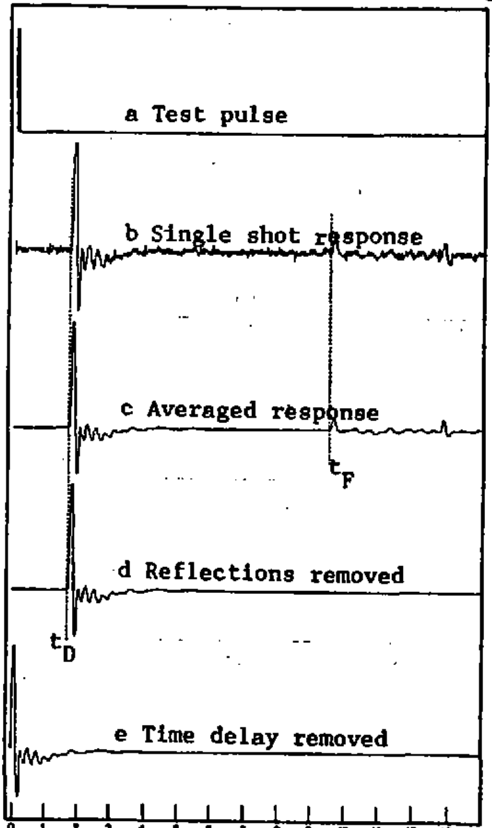

FIGURE 2 - IMPULSE RESPONSE MEASURING SEQUENCE

The measured drive unit impulse response is completely isolated from that of the room in which it has been measured by setting to zero all values of the sampled waveform which occur from the start of the first room reflection (Figure 2d). The loudspeaker microphone time delay t_D is removed by re-defining the time origin (Figure 2e) and the free field frequency response is then calculated from the impulse response using the fast Fourier transform to provide both the amplitude and the phase/frequency characteristics (Figure 3).

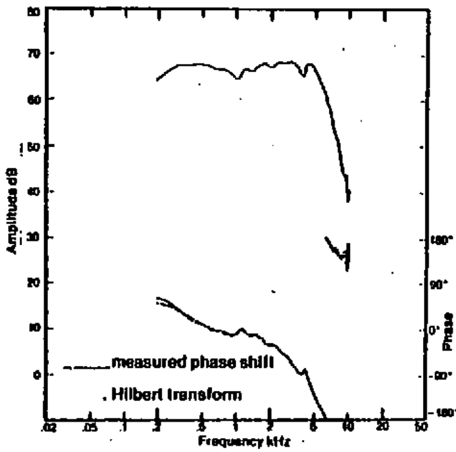

FIGURE 3 - FREQUENCY RESPONSE OF 110mm DRIVE UNIT

For loudspeakers exhibiting "minimum-phase behaviour", the phase shift between the electrical input and the corresponding acoustical output is no greater than that inherent in the amplitude frequency response characteristic. For such devices, the amplitude and phase characteristics are uniquely related, and one may be calculated from the other using the Hilbert transform. The phase characteristic derived from the Hilbert transform of the amplitude characteristic is shown superimposed upon the measured phase response in Figure 3.

The excellent agreement between the measured and calculated phase characteristics indicates that this loudspeaker drive unit is of "minimum-phase shift" type and as such its transient behaviour is uniquely defined by the shape of its amplitude/ frequency response characteristic.

It has been found that most moving-coil loudspeaker drive units are of the "minimum-phase shift" type, which leads to considerable simplifications for certain applications of the technique, since only the amplitude/frequency response characteristic need be considered. This has been particularly beneficial for production testing for it is now possible to measure drive units on a standard baffle and through subsequent processing of the amplitude frequency characteristics only, to sort and grade drive units for sensitivity and frequency response shape prior to assembly into pairs of loudspeaker systems intended for stereophonic reproduction (4).

Conclusions

Experience of the method for both R & D and production testing applications has shown it to be versatile, highly repeatable and to have much greater accuracy than previously obtainable with traditional analogue methods. The absolute accuracy of loudspeaker measurements is now limited more by the effect of ambient conditions e.g. temperature and humidity, upon the long term stability of the drive unit under test, than by the measuring method itself.

Further work still needs to be carried out to establish improved methods for minimising the effects of time-record truncation and drive unit non-linearity upon the calculated frequency response.

References

(1) L R Fincham and R V Leedham "Loudspeaker Evaluation Using Digital Fourier Analysis", presented to British Section of the Audio Engineering Society at the IEE, London Feb 1973.

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(3) J M Berman and L R Fincham "The Application of Digital Techniques to the Measurement of Loudspeakers" Journal of the Audio Engineering Society June 1977, Vol 25 No 6.

(4) L R Fincham "Production Testing of Loudspeakers Using Digital Techniques" Journal of the Audio Engineering Society, December 1979, Vol 27 No 12.