PULSE TESTING OF LOUDSPEAKERS

W.J. Davies

Dept of Applied Acoustics, University of Salford, SALFORD M5 4WT

INTRODUCTION

The object of the project has been to develop an inexpensive, high-quality system to measure the frequency response of a loudspeaker in reverberent conditions.

The established method of measuring frequency response requires an anechoic chamber, which is expensive to build and inconvenient to hire. An alternative is to consider the loudspeaker as a linear system, and derive its frequency response from its impulse response. The frequency response of a linear system is defined as the Fourier Transform of the impulse response:

$$H(f) = FT[h(t)] \tag{1}$$

This approach has been used successfully before [1], where the impulse response was processed using a dedicated Fourier analyser. This is also a comparatively expensive method. In this project, a more modest piece of equipment, the BBC microcomputer, has been used.

Fig 1 explains how the pulse method works in reverberant conditions. The input pulse, $\sigma(t)$, is fed into the loudspeaker, which then outputs its impulse response, $h(t)$. A short time later, the first reflection from the walls of the measuring room occurs. Assuming that the room is big enough, this reflection interferes little with the impulse response, so it can be gated out to leave just $h(t)$.

THE MEASURING SYSTEM

A block diagram of the measuring chain is shown in Fig 2. The input pulse is amplified and fed into the loudspeaker. A microphone detects the impulse response, which is digitised by the analogue-to-digital converter (ADC) and processed by the BBC microcomputer. One advantage of using a micro is that several operations, such as gating out the room response and detecting the start of the impulse response, can easily be performed in software, rather than using electronic hardware.

The Pulse Generator

This is the first stage of the system. The ideal input pulse is a delta function. Since this is impossible to relise practically, an approximation was chosen. The approximation had to satisfy three criteria: its spectrum should be as flat as possible in the audio range, it should contain as much energy as possible to enable a good signal-to-noise ratio, and it should be easy to produce electronically.

The pulse shape chosen was a rectangular one, which satisfies the third criterion and offers a compromise between the first two. Its spectrum is the familiar sinx/x shape; the first spectral zero is at a frequency, 1/T, where T is the width of the pulse in seconds. If T is chosen so as to place this zero above the audio range, then the spectrum 'flatness' in the audio range decreases with T, while the energy content increases with T. A practical compromise of 25μs was chosen for T, resulting in the spectrum of Fig 3(a). The amplitude at 20kHz is about 2dB less than the midrange value.

<u>The Power Amplifier</u>

This is the next stage in the measuring chain of Fig 2. The main factor in its choice was that it should produce as large an output amplitude as possible. Hence, an amplifier rated at 300W into 8 Ohms, the most powerful available, was used. This will produce a maximum amplitude of 70 Volts into 8 Ohms. Because the input pulse is so short, however, the output slew rate of the amplifier distorts it, turning it into a triangular shape. The spectrum of this shape is shown in Fig 3(b). The amplitude at 20kHz is reduced by another 2dB compared with that of the original pulse. This was felt to be acceptable in view of the increased energy of the pulse.

<u>The Analogue-to-Digital Convertor</u>

This is a 12-bit system (offering 72dB dynamic range), with a sampling rate of 40kHz. It was built by a fellow student. The 40kHz sampling rate necessitates an anti-aliasing filter with a cut-off at 20kHz. Hence, almost the full audio bandwidth, up to at least 19kHz, can be sampled.

THE COMPUTER PROGRAMS

The software part of the measuring system performs three functions: it samples the impulse response, Fourier Transforms it, and plots a graph of the frequency response. Because of the limited memory available in the BBC micro, these tasks are carried out by three separate programs, driven by a 'master menu' program.

<u>The Sampling Program</u>

This has three main parts: a trigger routine, to detect the start of the impulse response; a sampling routine; and a storage routine. For reasons of speed, the first two routines are written in Assembly language - they match the ADC in running at 40kHz.

Because of the importance of signal-to-noise ratio, it was decided to incorporate signal averaging into the system, whereby the impulse response of a given loudspeaker is sampled N times, and the samples averaged. The resulting improvement in S/N ratio for random background noise is given by equation (2).

$$\text{Increase in S/N ratio} = (N)^{½}\ \text{dB} \qquad (2)$$

In order that as much of the computer's memory as possible be free for sample storage, the averaging is carried out in the next program. Thus, the sampling program stores the N samples on floppy disc, where they form a permanent description of the loudspeaker which can be processed at any time

In the future without having to repeat the original measurement.

The Calculating Program
This program is the second in the software chain. It averages the samples, performs a Fast Fourier Transform (FFT) using an algorithm stored in a ROM chip fitted to the computer, and then calculates the points for the frequency response graph. These are then stored on disc, ready for the next program.

It is a requirement of the FFT algorithm [2] that the size of its data field be an integer power of two. Hence a routine is provided which finds the nearest allowable size of data array which is bigger than the sample size, and fills the remaining space with zeros. This has the effect of making the frequency response graph look smoother, though no new information is added.

The Graph-Plotting Program
Like the sampling program, this too has been kept as short as possible - this time so that a high-resolution graphics mode may be used for the graph. The graph points are read straight off the disc and onto the screen. A hard copy option is provided.

PRACTICAL LIMITATIONS

Low Frequency Limit
This is a function of the dimensions of the measuring set-up, as shown in Fig 4. The microphone and loudspeaker are set up along the principle axis of the room, spaced d metres apart, midway between the floor and ceiling. The height, h metres, is assumed to be the smallest room dimension. There are two factors to consider.

The first is that the software gate must shut before the first room reflection occurs. This gives an expression for the maximum sampling time. The lowest frequency identifiable in the sampled impulse response is then the reciprocal of this time. The second factor is that the microphone should ideally be in the far field of the loudspeaker; that is, at least one wavelength away at the lowest frequency to be measured. Combining these two criteria gives

$$f_{min} = \frac{595}{h} \tag{3}$$

As a guide, if h is 3 metres, then the lowest frequency identifiable is 200Hz. If the 'far field' requirement is ignored, and d fixed at 1m, then this drops to 159Hz for the same value of h.

High Frequency Limit
The frequency response of the system at the upper end is a convolution of the roll-off's of the pulse spectrum, the microphone response and the anti-aliasing filter. This sets it to about 18kHz.

EXPERIMENTAL RESULTS

The system has been tested by obtaining both single-shot and averaged responses down to 200Hz. These have been compared with the response of the same loudspeakers to a swept sinewave in an anechoic chamber. The system has been found to work fairly well within the limitations mentioned, although background noise has proved to be a particular problem. Obtaining averaged results has been difficult, because the software trigger routine is not completely reliable in triggering at the same point of the impulse response each time. The system is also rather slow at present, due to the continual transfer of data and programs to and from disc: it takes over five minutes to obtain the spectrum of nine averaged 5ms samples.

These faults could be improved by using an electronic trigger signal from the pulse generator, suitably delayed, and by using a microcomputer with a larger memory, to speed up data storage.

CONCLUSION

A simple microcomputer-based system for obtaining the frequency response of loudspeakers in reverberant conditions has been developed . It works fairly well within the limitations specified, although more development work is necessary to produce acceptable results.

APPENDIX 1

Equipment Details
Custom-built pulse generator
Crown DC300A power amplifier
Bruel and Kjaer half-inch microphone, type 4133
Bruel and Kjaer microphone amplifier, type 2608
Custom-built 12-bit analogue-to-digital convertor
BBC microcomputer, model B

REFERENCES

[1] J.R. Berman and L.R. Fincham, 'The Application of Digital Techniques to the Measurement of Loudspeakers', J.A.S.A., Vol 25, no 6, 370-384, (1977).

[2] J.C. Davies and P.G. Craven, 'Fast Fourier Transform for the BBC Micro', Structured Software, 23 Redcar Drive, Eastham, Wirral, Merseyside, (1984).

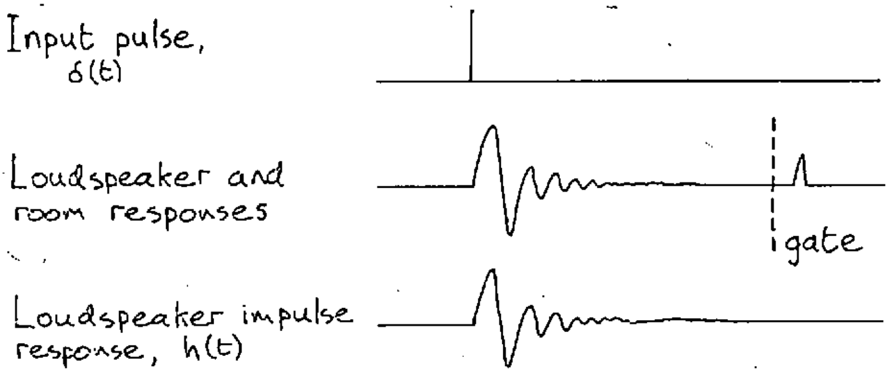

Fig.1 Gating out the Room Response

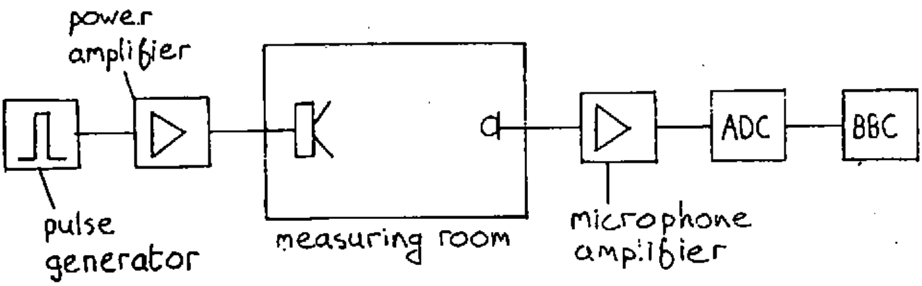

Fig. 2 Block Diagram of Measuring Chain

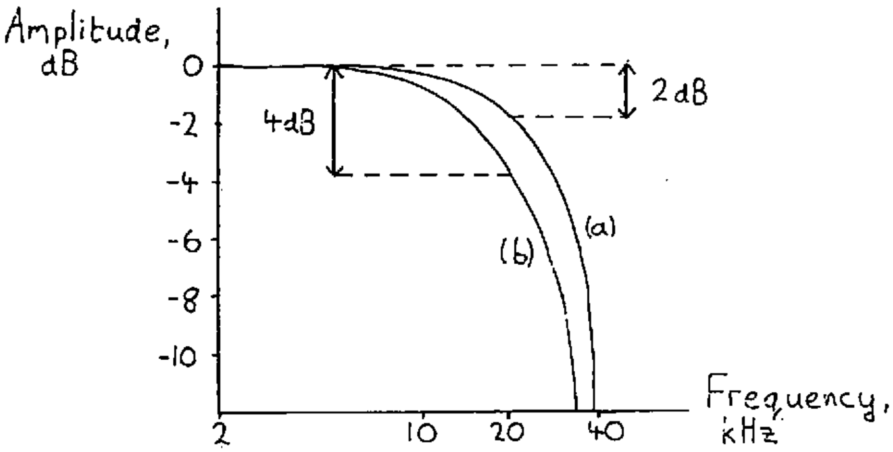

Fig. 3 Spectrum of (a) initial & (b) amplified pulse

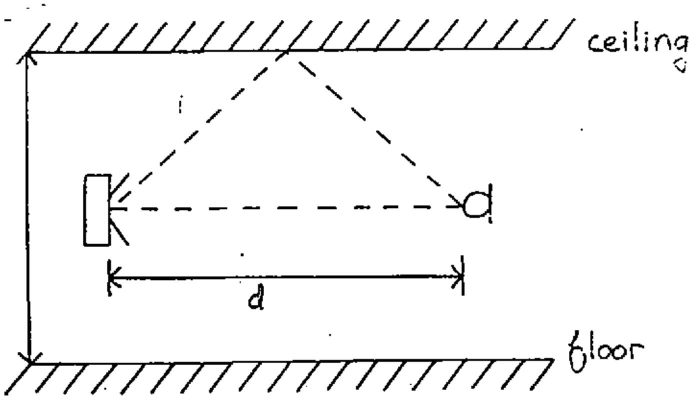

Fig. 4 Optimum Measuring Arrangement