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MICROPROCESSOR CONTROLLED EQUIPMENT FOR THE MEASUREMENT OF REVERBERATION TIME

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INTRODUCTION

Within the BBC the measurement of the reverberation time as a function of frequency is carried out in very nearly every new or newly refurbished area in which the acoustic quality is important. Also, in the study of the acoustic properties of materials, many measurements are made of the changes in the reverberation time in a reverberation chamber.

The original and probably still the most widely used method of measuring the reverberation time consists of exciting the room with a source of sound energy for a time long enough to allow the sound pressure level in the room to achieve equilibrium. The sound source is then removed and, from a recording of the decay of the sound pressure level as a function of time, the average rate of decay and hence the reverberation time can be derived. The measurement is confined to the wanted frequency band either by a bandpass filter or by using an excitation signal with a controlled spectrum. This procedure is repeated several times at each frequency band and for a number of different microphone positions. It is a laborious process which, if carried out manually would consume a large amount of effort.

THE EARLY DEVELOPMENT OF INSTRUMENTATION

In the period around 1947, several attempts were made to automate or at least to reduce the labour required for the measurement of reverberation time[1,2,3]. The most successful of these attempts culminated in the design of a set of equipment which, with some minor modifications, has been in constant use in the BBC for over 30 years[4,5]. A functional block diagram of this equipment is shown in Fig. 1. The essential feature of this equipment was that it converted the amplified microphone signal to a signal representing the logarithm of the sound pressure level which was then displayed as a function of time on a cathode-ray tube with a long-persistence phosphor. A moveable graticule, calibrated in reverberation time, could be manually aligned with the average slope of the trace as assessed by the operator, and the reverberation time read directly from the scale. Although the instability of the valve circuits made frequent recalibration necessary, this equipment significantly reduced the labour required by eliminating the recording and subsequent analysis of the individual decays. It is, however, becoming unserviceable and by modern standards it is also inconvenient, bulky and heavy.

THE DEVELOPMENT OF NEW INSTRUMENTATION

The experience of many years with the older equipment showed that its main features were satisfactory. In particular, the display of the individual decays in the form of sound pressure level on a logarithmic (decibel) scale as a function of time was valuable for the identification of acoustic problems.

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It would have been feasible and beneficial to redesign the equipment using modern components to reduce its weight and size. However, the procedure for the measurement of reverberation time is well defined and is ideally suited to automatic control by a microprocessor.

Once a microprocessor is included in the system, many additional features can be incorporated at very little extra cost. An annotated display, the averaging of several decays to reduce the random variations inherent in such measurements and the automatic computation of the reverberation time are useful and would significantly reduce the operator effort required. One additional cost incurred if the microprocessor is to process the microphone output signal instead of merely controlling the sequence of operations is the requirement for an analogue-to-digital convertor (a.d.c.). In principle, this may be placed anywhere between the microphone and the microprocessor. In practice, the instrument's total complexity and cost and the ease with which the component parts may be used for other purposes depend on the position of the a.d.c. If it is positioned before the conversion of the microphone signal to sound pressure level then it must be capable of converting wide-band audio signals over a large dynamic range. Also, all of the subsequent signal processing would be digital which is also comparatively expensive. However, if the a.d.c. is positioned after the conversion of the signal to the logarithm of the sound pressure level, then it need only be comparatively slow and have a limited dynamic range. The microphone signal processing would then be entirely analogue which, although subject to drift, is both simple and cheap. Careful circuit design can greatly reduce the effects of drift. The individual components are also more suitable for other uses.

A block diagram of the new microprocessor-controlled equipment is shown in Fig. 2. It closely resembles that of the earlier equipment except that the individual components are controlled by the microprocessor rather than manually.

DESCRIPTION OF THE NEW INSTRUMENT

Three of the component units of the new instrument were designed using analogue signal processing circuits under the control of the microprocessor. The first of these was the excitation generator which provides the input signal for the loudspeaker. It consists of a wide-band noise source, a 1/3rd octave bandwidth switchable filter and an output level control. The microphone amplifier unit contains an amplifier with a gain switchable in 1.5dB steps in the range 40-100 dB. A switchable, octave-bandwidth filter was included to improve the signal-to-noise ratio of the measurement. In the third unit, the conversion of the amplified and filtered microphone signal to a voltage representing sound pressure level is carried out by a precision rectifier and low-pass filter. No r.m.s. convertor is required because, for such band-limited signals, the mean of the rectified signal is proportional to the true r.m.s. to a degree of accuracy sufficient for the measurement of reverberation time. This voltage is converted to a digital representation by simultaneously starting a digital counter and a "ramp" waveform. The counter is stopped when the ramp is equal to the voltage which is to be converted. By making the ramp waveform exponential a direct conversion to a logarithmic (i.e. dB) scale was achieved. The output is in the form of an 8-bit binary word representing an input range of 64dB, quantised in ¼dB steps. The timing circuits, based on crystal oscillators, are so stable that recalibration is unnecessary.

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A microprocessor system is used to control these units, to provide the input to a display unit and to interpret the operator inputs from a keyboard. A GPIB interface to IEC 648 and IEEE 488 standards was also provided to enable remote operation as part of a larger acoustics measuring system.

The final instrument measured reverberation time in any of 24 1/3rd octave wide frequency bands from 50Hz to 10kHz and over a range of reverberation times from 0.12 to 15 seconds.

OPERATING SOFTWARE

The main functions of the operating software are to provide the control of the other units, the generation of the displayed information and the interpretation of the control inputs. The averaging of several successive traces at one measurement frequency was added to these basic functions and the display arranged to show this averaged trace as well as each successive decay trace. This facility improves the display and greatly eases the extraction of the average rate of decay. A computation algorithm was also included to extract the average rate of decay and to calculate and display the measured reverberation time at each frequency. Fig. 3 shows the appearance of a typical display.

Because the processing of the data is based on a microprocessor, changes can easily be made to the software to calculate other acoustic parameters such as EDT and early-to-late energy ratios. Alternatively, because the data representing the decay of sound pressure level can be transmitted via the GPIB to a master controller, other calculations can be carried out externally.

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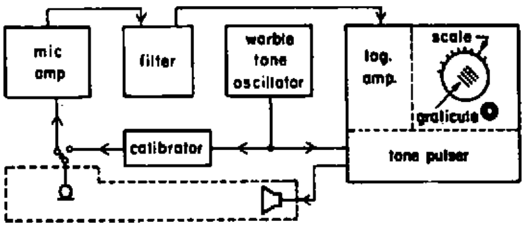

Fig. 1 - Block diagram of earlier equipment

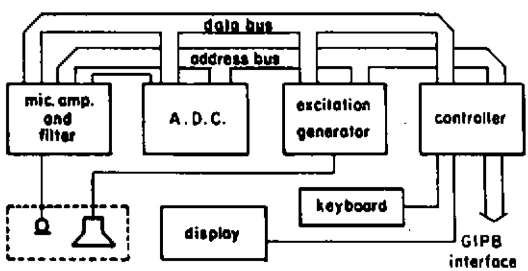

Fig. 2 - Block diagram of microprocessor-controlled reverberation time measuring equipment

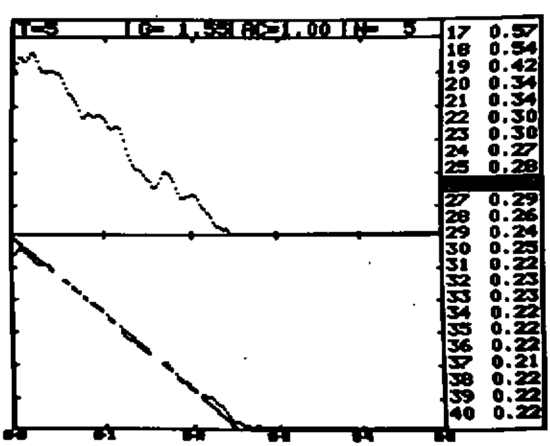

Fig. 3 - Appearance of display after several traces at one frequency