BBC

ENGINEERING DIVISION

MONOGRAPH

NUMBER 80: DECEMBER 1969

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M. E. B. MOFFAT, M.A., D.Phil., C.Eng., M.I.E.E., A.Inst.P.

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BRITISH BROADCASTING CORPORATION

PRICE FIVE SHILLINGS

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ANNOUNCEMENT

Future of the BBC Engineering Monograph Series

The BBC has now been publishing the Engineering Monograph series for fourteen years. These monographs have proved a valuable medium for making the results of work done in the BBC Engineering Division available for engineers engaged in the field of broadcasting and telecommunications generally, but it has become increasingly clear that their scope has been limited by the very fact – which is implicit in the name given to the series – that each issue has been restricted to a single paper conforming approximately to a standard length.

BBC Engineering Division have, therefore, decided that this will be the last issue of the monographs in their present form, and that each issue will in future include a number of articles, papers, or announcements covering a wide range of BBC engineering developments in both television and radio. The title will be changed to 'BBC Engineering' and there will be an average of four issues per year. It is hoped that, in addition to papers of a type similar to those which have been appearing in the monographs, some of the contributions in 'BBC Engineering' will appeal to readers who are interested in broadcast engineering developments generally, although not professionally engaged in this field.

The first issue will be in January 1970, and the total amount of material published each year will be at least equal to the average contents of a year's issues of monographs.

The annual subscription to 'BBC Engineering' will be 30s. (£1·50) or $4.00 post free but, unless BBC Publications are notified to the contrary, existing subscriptions to the monographs will be automatically transferred to 'BBC Engineering' until expiry without extra charge. Single copies will cost 8s. (40p) or $1.00 post free. Orders can be placed with newsagents and booksellers or BBC Publications, 35 Marylebone High Street, London W1M 4AA.

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AN AUTOMATIC METHOD FOR THE MEASUREMENT OF REVERBERATION TIME

SUMMARY

The automatic measurement of reverberation time includes the making of a special recording which is characteristic of the studio or other room tested; equipment subsequently selects the required signals from the recording, converts them to digital form and punches a paper tape describing relevant characteristics of the signals. Reverberation times may then be calculated using suitable computer programmes. The advantages over conventional methods of measurement include a reduction in the number of man-hours required, more objective results and the storage of studio sound-decay data on an inexpensive medium.

1. Introduction

For many years the reverberation times of BBC studios have been measured by observing the logarithm of the decaying sound-pressure by means of an oscilloscope.[1–4] The method requires special test equipment to be transported to the studio under test. Bursts of 'warble-tone' are radiated by a loudspeaker into the studio and the output of a microphone in a known position is fed into a logarithmic amplifier and thence to an oscilloscope whose time-base runs in synchronism with the warble-tone bursts. After each excitation burst the logarithm of the sound-pressure decay is displayed on the screen as a downward-sloping irregular line. A graticule is rotated manually so as to bring a set of parallel lines engraved on it into alignment with the mean slope of the trace, and the reverberation time is read from a scale attached to the graticule. With most satisfactory rooms the line has a definite and recognisable mean slope, but with many others this is not so; disparate measurements are obtained at different times and with different observers.

While this method had many advantages, it often entailed lengthy travelling time, a serious disadvantage if a visit to a studio was made merely to carry out a routine check. Moreover, it provided no permanent record of the decay curves, and any peculiarities had to be laboriously noted at the time of measurement. When a number of rooms were to be measured at one site it often proved difficult to gain access to all the rooms consecutively in order to complete the job quickly; as a result measurements sometimes had to be made in haste, at fewer microphone positions than were really desirable.

For these reasons, a method was devised using a tape recording which consisted of a series of one-second bursts of warble-tone with centre-frequencies at half-octave intervals; this was sent to the studio at which measurements were required and the local engineers replayed the tape through a loudspeaker in the studio, re-recorded the sound at a number of microphone positions in succession and returned the recording for analysis. The second recording, which contained the original test-signal modified by the studio reverberation, was analysed by filtering into octave bands and tracing the rectified envelope of the decaying signal using a graphic high-speed level-recorder. Although travelling time was saved by this procedure, the total time of analysis was generally longer than the time taken by on-site analysis with an oscilloscope.

To speed up the analysis a test tape and analysing apparatus have been devised whereby the recorded signals may be processed automatically by computer, after digitising and conversion to punched paper tape form; the results are printed out as a table of reverberation times. As it would be wasteful to transfer the whole recorded signal to punched paper tape – only that part corresponding to the decaying signal is required – the timing of the signals recorded on the test tape is known accurately. The analysing equipment is triggered at the onset of the excitation signal and the paper-tape punch starts after a period equal to the known duration of the excitation signal.

2. The Test Tape and its Use

2.1 *The Test Tape*

The test tape comprises recordings of one-third octave noise bands with centre frequencies spaced one-third of an octave apart; the signal, for each value of centre frequency, consists of a group of three one-second bursts of noise, each separated by half-second periods of silence. After an interval of $3\frac{1}{2}$ seconds another group follows, with a different centre-frequency, and so on (Fig. 1(*a*)); the noise bursts are centred at 23 frequencies ranging from 50 Hz to 8 kHz. This sequence of groups is repeated nine times to provide for tests at up to ten microphone positions.

2.2 *The Test Procedure*

The test tape is replayed at $7\frac{1}{2}$ in/s (191 mm/s) through a loudspeaker in the studio under test and the studio response is recorded at a number of different microphone positions in the studio. Five microphone positions are normally used in a small talks studio and ten in a large music or television studio.

The engineer recording the studio response uses the first two noise bursts of each group to monitor the recorded levels, adjusting the recorder gain so that the third noise-burst fully modulates the tape. In order that timing errors do not lead to serious inaccuracies in the measurement of short reverberation times, it is necessary that the speeds of the replay and recording machines do not differ by more than 0·2%; this is ensured in practice by using full 732 m (2400 ft) reels of tape on professional machines, a separate reel of tape being used for each studio and, as ten sequences of noise burst groups occupy no more than 34 minutes, only the outer halves of the tape spools are used. Fig. 1 (*b*

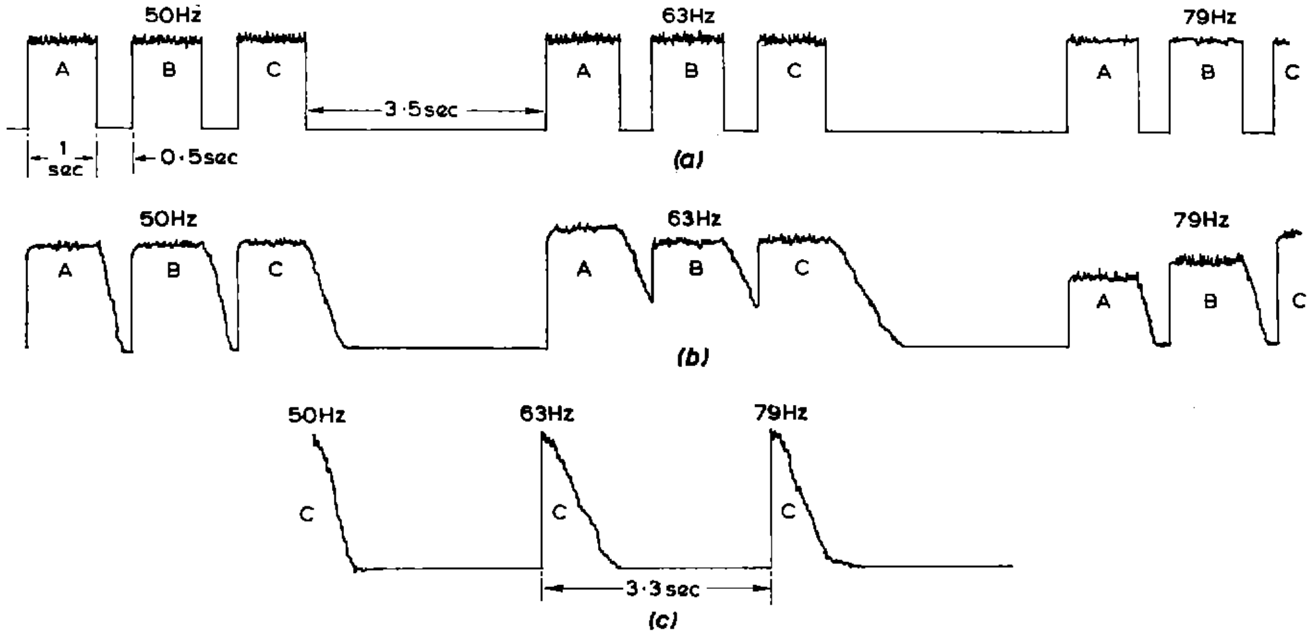

Fig. 1 – Signal waveforms (logarithm of rectified envelopes)

(*a*) Signal on test tape. (*b*) Signal on returned studio recording. (*c*) Signal digitised and punched on paper tape.

shows the form of the recording, in which the original noise bursts are modified by reverberation in the studio.

Details of the test procedure are given in Appendix I.

3. Digitising and Punching

Fig. 2 shows a block schematic diagram of the apparatus used for punching the paper tape from the returned recording and Fig. 3 shows the timing of pulses at important points in the apparatus. The recording is played through the $\frac{1}{3}$-octave filter-set of a Brüel and Kjaer Spectrometer Type 2112 and then passes through the 'set-output amplifier,'* an amplifier which adjusts its gain to suit the high initial level of an input signal (which may be between -35dBm and $+16$dBm) so that the maximum level of the output signal is fixed. At any instant in time after the initial gain setting, provided no signals of higher level are received, the gain remains constant until reset at the end of a decay. The output of this amplifier is then passed to a logarithmic amplifier and rectifier[3] and an analogue-to-digital converter which provides quantised samples of the logarithm of the signal envelope every 10 ms. The sample amplitudes are recorded on punched paper tape having eight tracks and are quantised in units of approximately $\frac{1}{4}$dB, allowing the 50dB range of the logarithmic amplifier to be accommodated comfortably within the 2^8 distinguishable levels available.

The output of the set of filters is also connected to a timing chain which controls the tape punch. It utilises a counter-timer unit (Racal Type SA535) from which pulses are derived for operating various logic elements. In Fig. 2, the numbers involving decimal fractions represent the relative onset times, in seconds, of pulses derived from the counter-timer. The integers 23 and 27 refer to pulses obtained at counts 23 and 27 on the serial binary counter; $\overline{23}$ refers to any count other than 23.

When the leading edge of the first noise burst of a group reaches the pulse former a short pulse is generated which passes through the input gate (open, because the counter is at 0·00 and there is a GO condition initially) and starts the counter-timer. The pulse passes through gate 1 and sets signal indicator 1. At 1·48 s, just before the leading edge of burst B is due, the input gate and gate 2 open. At about 1·50s a pulse passes through and sets signal indicator 2. Similarly burst C sets signal indicator 3 at about 3·00s. As all three signal indicators are set, the GO condition is maintained and the counter-timer continues. At 3·04 s the set-output amplifier is reset. At 4·00s, when the decay of burst C commences, the punch starts and the serial binary counter registers a count of one. At 7·30s the punch stops and the filter is automatically switched to the next higher $\frac{1}{3}$-octave.

Because the binary counter has not yet counted 23 pulses on the punch-start line, the counter-timer is reset to zero at 7·36s and the system is ready to receive the first of the next three noise bursts due 0·14s later.

If the counter-timer is wrongly started because of a spurious signal on the tape recording (clicks, doors closing, footsteps, coughing, etc.), punching is inhibited because unless further onsets of signal occur at the appropriate times, one or both of signal indicators 2 and 3 is not set, a NO-GO condition is established at 3·05s, a buzzer sounds, the magnetic tape transport is automatically stopped and the counter-timer reset to zero. It is then

* Designed by K. F. L. Lansdowne.

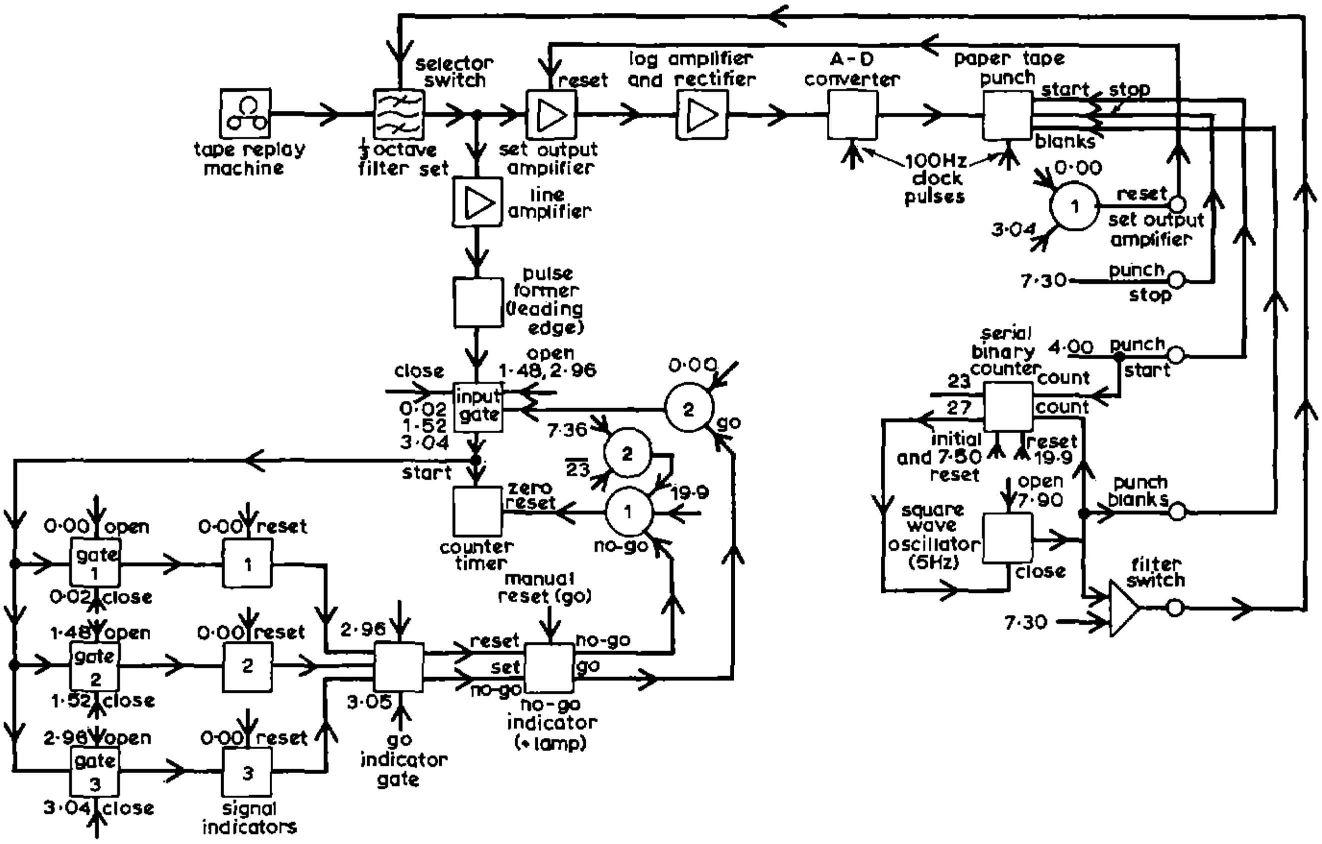

Fig. 2 – Block diagram of automatic reverberation time measurement apparatus

necessary to rewind the tape recording manually, depress momentarily the manual reset (GO) button and replay.

Sometimes a NO-GO condition arises even though there may be no spurious signals present. This will occur if the studio replay and record machines differ in speeds by more than about 1·3%. The NO-GO condition can be overridden by holding the manual reset button depressed between 2·96 s and 3·05 s, but the fact must be noted down since it implies that the punch will either punch at least four samples of excitation signal or omit at least four samples of the start of the decay. The resulting error in computation of reverberation time could be appreciable for rapid decays.

After 23 decays have been punched, the counter-timer is not reset at 7·36s but continues. At 7·50s the binary counter is reset to zero and the gate to the 5 Hz square-wave oscillator is opened at 7·90s. The pulses from the oscillator switch the filter selector through the 27 positions from 8 kHz round to 50 Hz again. At the same time, 27 marker blanks are punched to indicate the end of the run at the current microphone position. At 19·9s the counter timer is reset to zero and the system is ready to receive another set of signals corresponding to another microphone position.

The information on the reel of eight-track paper tape consists of samples of decays and subsequent noise as shown in Fig. 1(*c*).

The Elliott 803B computer in Research Department is an early model which cannot economically be converted to accept eight-track data tapes. It is therefore necessary to transcode the eight-track tape to five-track tape, each character on the former being represented by a pair of characters on the latter.[5]

Brief instructions for operating the reverberation measurement equipment are given in Appendix II.

4. Computer Programmes

4.1 *General*

Several programmes for use with the above data have been developed. The principal object has been the measurement of reverberation time, but programmes have also been developed for plotting out selected decay curves using a teleprinter. Attempts have also been made to produce programmes which assess the degree of sound diffusion in a room.

All the programmes are in Elliott 803, A103 Autocode.

4.2 *Reverberation Time Computation*

Of the programmes developed for reverberation time measurement, only the most recent, called RT.IV.DS., will be described.

Before running the data in the computer with RT.IV.DS. an initial data tape is punched, which contains:

(*a*) a label to appear on the printout, identifying the room concerned and showing the initial data, and

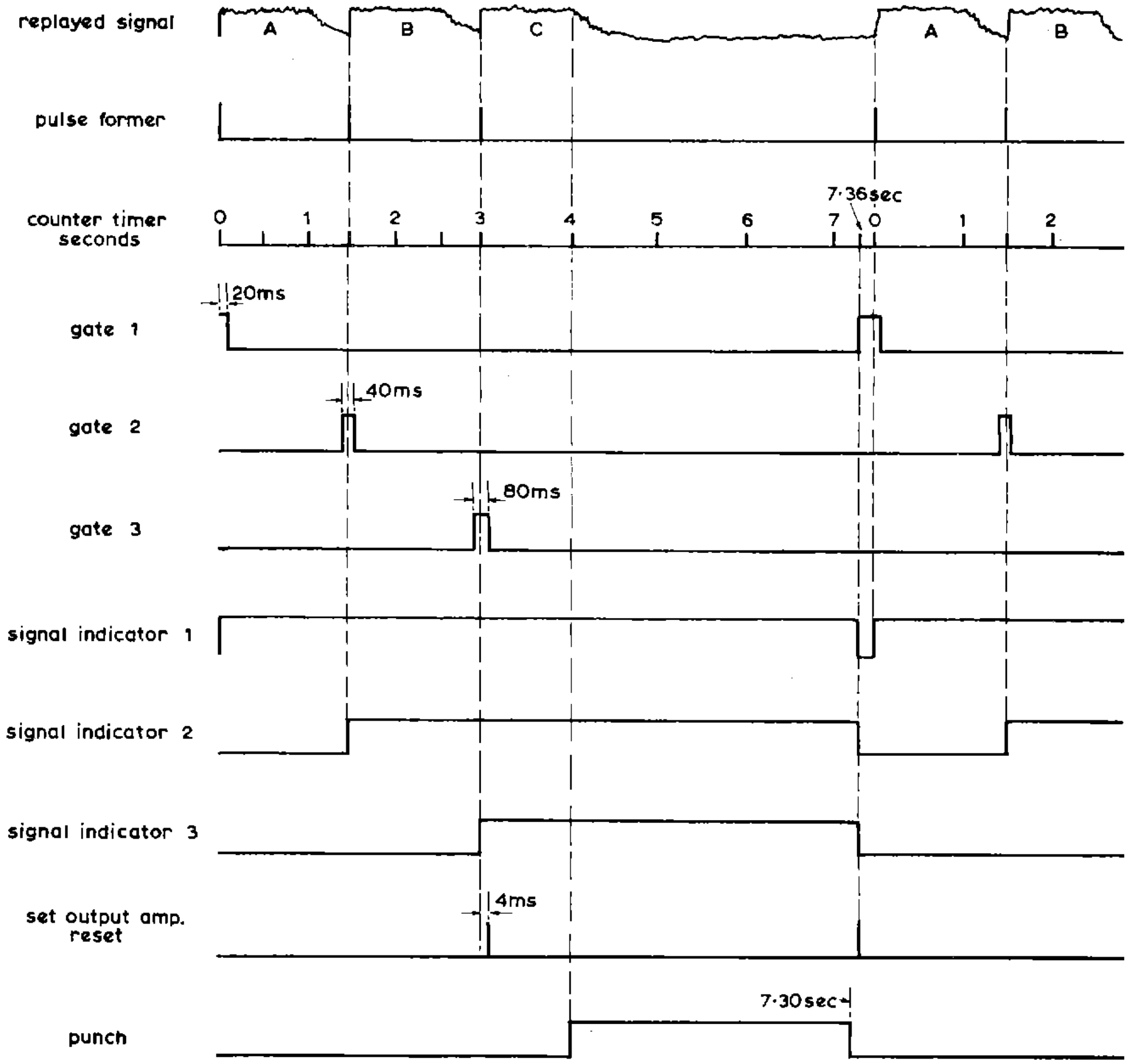

Fig. 3 – Logic timing diagram of automatic reverberation time measurement apparatus

(*b*) the initial data, viz:

P2, the number of microphone positions

D1, the mean slope (negated) of the logarithmic amplifier characteristic in ADC (output) units/dB,* normally 3·50 ADC units/dB

L1, one ADC unit more than the ADC output corresponding to the minimum acceptable signal at the start of a decay

N1, the maximum duration of the analysis in tens of milliseconds

D6, minimum acceptable signal/noise ratio in dB, up to 125Hz

D7, minimum acceptable signal/noise ratio in dB, 125Hz and above

S9, the deviation of the magnetic tape speed from the nominal value, expressed as a percentage.

L1 is normally set to 128 so that acceptable decays may be recognised, visually if necessary, by the absence of a hole in track eight at the beginning of a decay. Note that a high ADC reading corresponds to a low sound level and vice versa. The logarithmic amplifier is usually adjusted so that 0 dBm corresponds to 64 ADC units. The set-output amplifier ensures that the peak level of the excitation signal corresponds to about +4dBm (50 ADC units). Since the decay does not always commence at peak level, and because punching may start a little late the punched part of the decay often starts at between 70 and 100 ADC units. L1 must not be set too high, otherwise some noise samples (usually 200–250 ADC units) may be misinterpreted as decays.

* The eight most significant binary digits at the ADC output are used here. The unit value is given by a voltage of 0·040V at the ADC input.

N1 is set sufficiently great to ensure that all of the specified range of decay curve is analysed. For example, for a small talks studio and 30dB range of analysis, N1 is set to 60 (0·6s). For the large reverberation room at Research Department it is set to 300 (3s). D6 and D7 are usually set to 30, but some noisy recordings may necessitate a lower figure. Two ranges are provided since the signal-to-noise ratios below 125 Hz are sometimes much less than those at higher frequencies.

The initial data tape is read into the computer, followed by the transcoded decay data tape. The computer decodes each pair of five-track characters into ADC units and discards them until it encounters a pair corresponding to a number of ADC units less than L1. This marks the commencement of the first decay and the computer reads and stores N1 + 10 pairs of characters, i.e. it stores all of the decay and some of the following noise. The point on the curve which is 6dB above the mean level of samples N1 + 1 to N1 + 10 is then determined and the slope of a line of best-fit is then computed up to this point, multiplied by the appropriate factor and the resulting reverberation time stored. Further pairs are then read and discarded until a pair with ADC units less than L1 is again reached, marking the start of the second decay. As with the first decay, the process is repeated every time a pair corresponding to the start of a decay is read.

If the signal-to-noise ratio of any decay is less than D6 then the reverberation time for this decay is stored as zero, a physically impossible value indicating an unacceptable signal-to-noise ratio.

One counter in the programme counts the decays and changes D6 to D7 above 125 Hz and back to D6 again after 8 kHz in readiness for the next 23 decays at a new microphone position.

After all the decays have been counted, determined by the number of microphone positions specified, the mean and standard deviation of all the non-zero reverberation times at each frequency are calculated. They are printed out together with the individual reverberation times as a table, an example of which is shown in Fig. 4.

For ten microphone positions, the running time of the programme is 20 to 40 minutes depending on the magnitudes of the reverberation times.

4.3 *Decay Curve Plotting*

A programme to plot selected decay curves has been found to be extremely useful in developing reverberation time and related programmes. It has been used to determine the reason for absurd values of reverberation time, leading to improvements in the programme. It also has occasional use, when unexpected reverberation time results are questioned.

The initial data for this programme is similar to that for the reverberation time programmes, except that S9, the deviation of the tape speed, is omitted. In its place, the number of decay curves required is punched (up to five are allowed) followed by the ordinal numbers of the particular decay curves selected.

In running the programme the transcoded tape is read through and decay curves counted until a selected one is reached. At this point the computer reads in pairs of transcoded characters until D6 (or D7) decibels of the decay curve have been read in. The output punch then produces a tape which, when fed into a Friden Flexowriter teleprinter, prints a line-by-line plot of the decay curve plus ten further samples. Fig. 5 is an example of a plot of a decay curve.

REV TIME :MARK IVDS(NOISE SAMPLED). MODIFIED RANGE.

FREQUENCY REVERBERATION TIME (SECS.)

KR680117/1 NOTTINGHAM CUBICLE TO STUDIO 2. (LOCAL RADIO).
PROCESSED 18.1.68.
5 3.55 129 80 30 30 0.0

HZ	1	2	3	4	5	MEAN	S.D.
50	0.46	0.42	0.00	0.45	0.00	0.44	0.02
63	0.39	0.00	0.40	0.42	0.55	0.44	0.07
79	0.55	0.40	0.28	0.20	0.52	0.39	0.15
99	0.27	0.16	0.37	0.20	0.17	0.23	0.03
125	0.36	0.29	0.36	0.36	0.31	0.33	0.04
157	0.41	0.44	0.37	0.42	0.52	0.43	0.06
198	0.42	0.38	0.31	0.38	0.44	0.39	0.05
250	0.38	0.39	0.36	0.42	0.44	0.40	0.03
315	0.22	0.42	0.36	0.38	0.42	0.36	0.09
397	0.37	0.36	0.27	0.32	0.30	0.33	0.04
500	0.32	0.37	0.39	0.26	0.36	0.34	0.05
630	0.30	0.33	0.25	0.37	0.31	0.31	0.04
794	0.27	0.33	0.27	0.29	0.33	0.30	0.03
1000	0.33	0.37	0.31	0.39	0.31	0.34	0.03
1260	0.28	0.34	0.30	0.32	0.31	0.31	0.02
1587	0.32	0.30	0.26	0.30	0.35	0.31	0.03
2000	0.32	0.31	0.34	0.35	0.32	0.33	0.02
2520	0.32	0.31	0.33	0.35	0.34	0.33	0.02
3175	0.34	0.35	0.35	0.33	0.34	0.34	0.01
4000	0.34	0.31	0.33	0.38	0.35	0.34	0.03
5040	0.35	0.34	0.36	0.38	0.38	0.36	0.02
6350	0.35	0.38	0.36	0.36	0.37	0.36	0.01
8000	0.29	0.35	0.00	0.37	0.00	0.33	0.04

Fig. 4 – Example of printout from RT.IV.DS.

4.4 *S-Factor Determination*

Randall and Ward[6] showed that the ratio of the initial and final slopes of a decay curve split into two halves could be used, under certain conditions, to assess the degree of sound diffusion in a room.

It would clearly be a great advantage if information about the diffusion of a room could be obtained from the data tapes prepared for reverberation time measurements. To some extent, this has already been done since, as Randall and Ward also showed, the variation of reverberation time with position (the 'S.D.' column in a reverberation time printout – Fig. 4) is also an indicator of diffusion. However, it is an unreliable parameter, especially where less than ten microphone positions are employed.

A number of programmes have been developed in an attempt to measure the ratio of initial to final slopes (S-factor) but all have failed to discriminate consistently between different states of diffusion of a room. Examination of the decay curve plots showed that the main difficulty lies in the precise distinction between what one

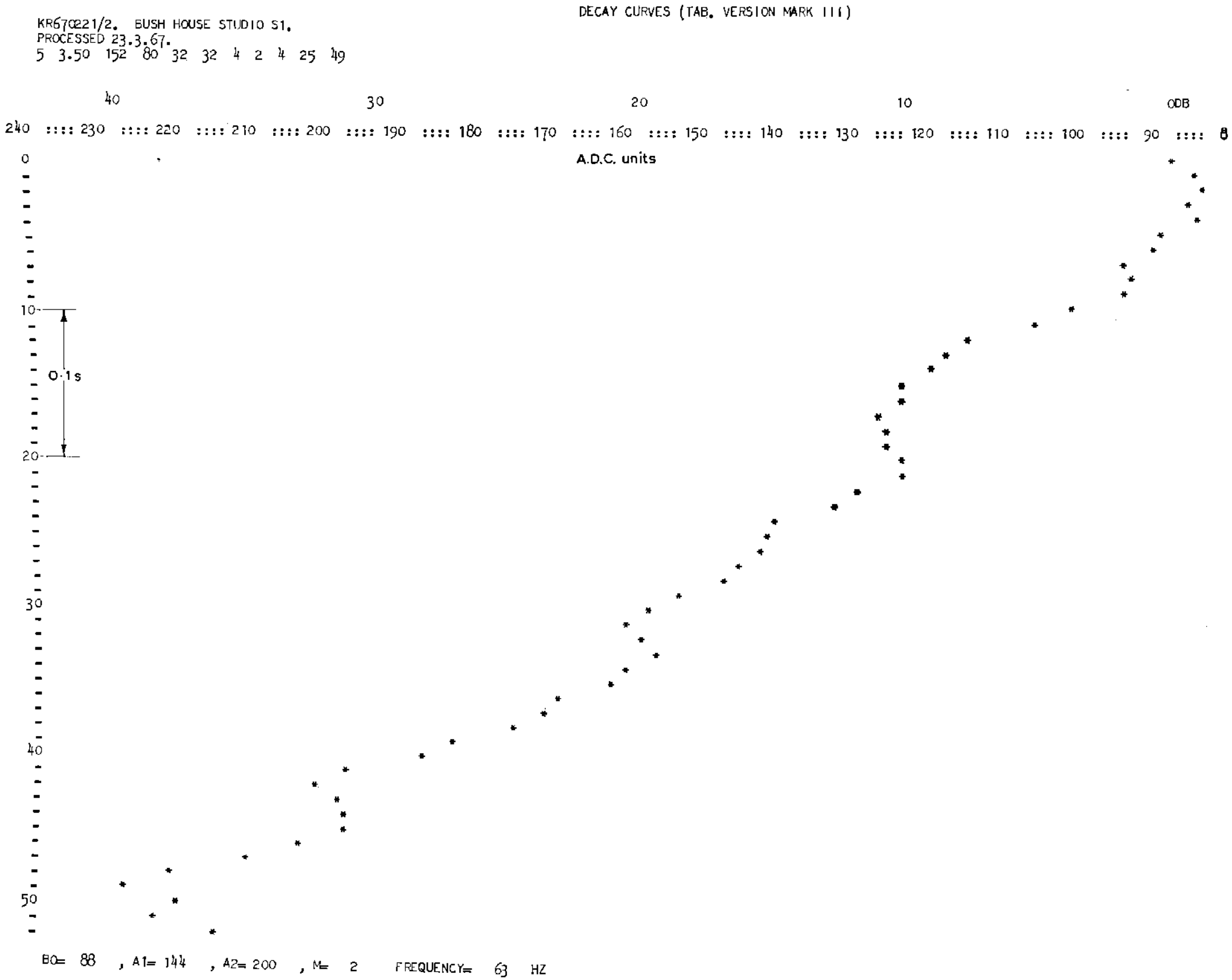

Fig. 5 – Example of plot of decay curve

intuitively feels to be the 'top half' and 'bottom half' of a decay curve.

Nevertheless, measurement of the variation of slope, invoking other criteria, is producing promising results and is the subject of another report.[7]

5. Discussion of Results

5.1 *Range and Accuracy*

The smallest reverberation time which can be measured is determined by the decay times of the filters at low frequencies and of the logarithmic amplifier/rectifier at the higher frequencies. By processing the test-tapes as though they contained reverberant signals, the lower limits were found to vary from 0·30s at 50Hz to 0·20s at 125Hz and 0·10s above 250Hz. If a 30dB range is specified, the interval between successive signals restricts the maximum reverberation time to about 6s.

The maximum error in reverberation time arises through tape speed errors of 1·3% in conjunction with a rapid decay. Such a signal would be accepted by the system, although it could mean that four samples of excitation signal would be punched before the decay, these samples contributing to the reverberation time calculation. The result of such a situation would be that a true reverberation time of 0·10s would be represented as 0·15s when computed over a range of about 30dB; a true reverberation time of 0·30s would be represented as 0·33s.

In practice, such extreme errors occur only rarely since there is a provision in the programme which discards the first three samples of a decay if they do not correspond to a falling signal. Furthermore, it is rarely necessary to measure very short reverberation times with great accuracy.

B.S.1568:1960 states that the maximum permissible deviation of the mean tape speed from the nominal value shall be ±0·5% for professional machines. Nevertheless, measurements on a number of machines in Research Department showed that speed errors of up to 2% could occur near the end of a 2400ft (732m) reel of tape, presumably due to the higher back tension of the reservoir spool. This is why the recording is made on the outer half only of a full 2400ft (732m) reel of tape.

Speed errors of the order of 0·5% would result in a reverberation time error of about 2% for a decay of 0·3s reverberation time. For reverberation times above 0·3s percentage errors in reverberation time approach the percentage errors in tape speed.

The accuracy of the method has been checked by punching a number of artificial decays of definite slope using a teleprinter.

The results of measurements on real rooms have generally agreed with those of conventional methods. In the early development of the method, when unexpected mean reverberation times were obtained, the tape recording was played into an oscilloscope as used for on-site measurements and the reverberation time checked by eye. In every case, the check measurement agreed with the computer measurement.

5.2 *Comparison with Conventional Results*

One noteworthy feature of reverberation time/frequency characteristics obtained by computer is that they are often much less smooth than those obtained by conventional methods. At low frequencies, the difference can be partly ascribed to the greater resolution of the third-octave bandwidth noise signal compared with the half-octave bandwidth noise signal used earlier, but at medium and high frequencies the spacing of room-modes on the frequency scale is so small that signals of either bandwidth would excite a very large number of modes. The most likely explanation lies in the objective and unprejudiced way in which the computer deals with each decay curve.

However dispassionate the observer tries to be, it is probably impossible for him to remain totally unbiased when subjectively assessing the slope of a decay curve. Firstly, if a decay trace is such that a range of slopes could be ascribed to it, the observer will probably tend to give it a value biased towards that of the previous decay curve read. Secondly, the reason for 'unexpected' or unusual reverberation times will be sought in the studio at the time of measurement, i.e. an attempt will be made to rationalise the unexpected value, and valid grounds may be found for ignoring it.

Fig. 6 shows a computer determination of the reverberation time/frequency characteristic of a studio at Radio Leicester. The peak around 400Hz was investigated and found to be due to the resonance of an undamped sheet metal object in the room. Had the measurements been made by conventional methods, the observer would have located the offending object and prevented it from vibrating. If this had proved impracticable he would probably have rejected that part of the decay curve attributable to the resonating object as being unrepresentative of the room. Thus neither method suppresses the revelation of important acoustical defects but the results are presented differently.

As is to be expected with any method of remote testing, the results must be interpreted with special care. It is interesting to note that the data for the Leicester studio was run with an S-factor programme which showed the presence of marked double-sloped decays in the 400Hz region, a feature associated with the presence of resonating objects.

5.3 *Practical Defects*

5.3.1 *Signal-to-noise Ratio*

The tapes received from some rooms have suffered from poor signal-to-noise ratio at low frequencies, which has led to triggering difficulties for the automatic processing equipment. The reason has usually been either the use of too small a loudspeaker or poor insulation against extraneous noise.

After repeated use, the test tapes suffer attenuation of high-frequency signals. This is probably due to residual magnetisation of the heads or parts of the tape transport of the machines on which they are played.

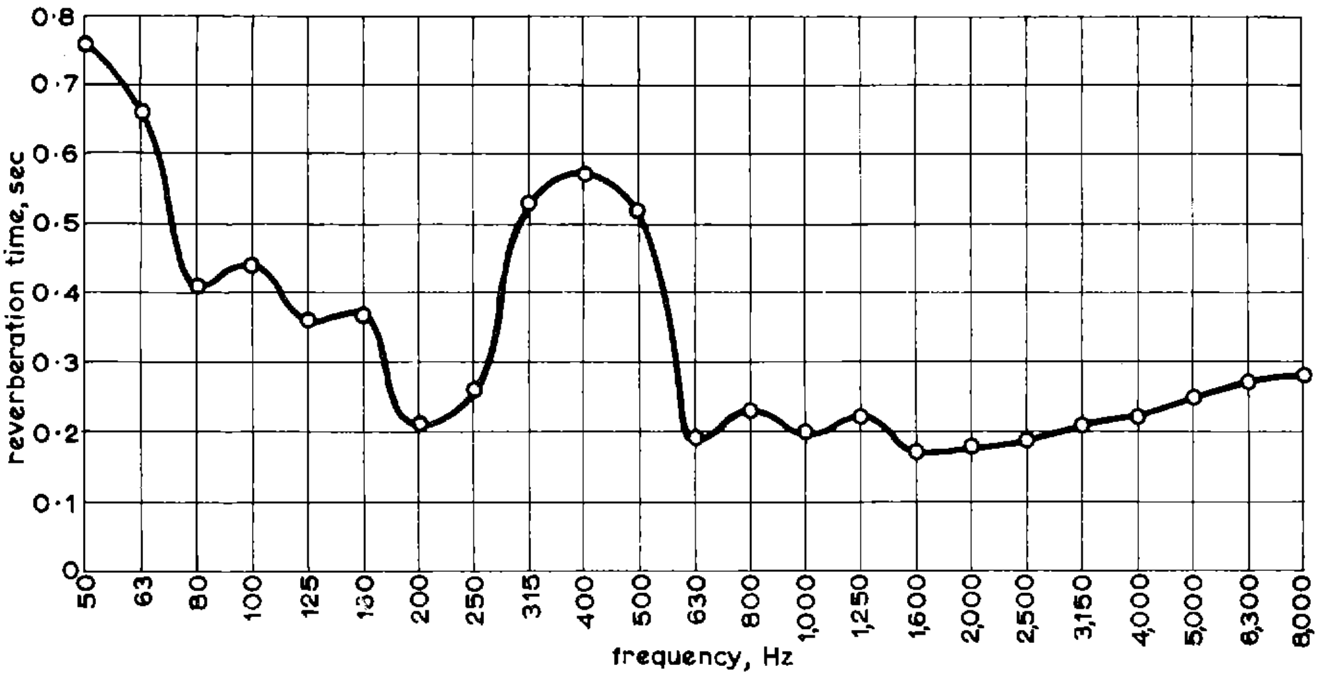

Fig. 6 – Reverberation time/frequency curve of Studio 1, Leicester

5.3.2 *Print-through*

On early test tapes, print-through became a serious problem. The current test-recordings are made on a type of tape with low print-through characteristics. Since the effects of print-through cannot be removed by filtering, this defect, coupled with the attenuation at high frequencies, limits the useful life of test tapes.

5.3.3 *Reliability*

The reliability of the apparatus is now generally good. The transcoding process, which was rather slow and unreliable, has been speeded up by a factor of six after the acquisition of a faster and more reliable paper-tape reader.

6. Discussion

Up to the time of writing, the reverberation-time characteristics of over 100 rooms or room conditions have been measured by the automatic method. Apart from the saving in man-hours, the basis of a library of paper tapes has been produced so that any tape which may later prove to be of interest may be processed to yield information about the properties of rooms, e.g. curvature of the decay curves, spatial distribution of reverberation times, and irregularities in the decay curves.

Even when no travelling time is saved, the man-hours required can generally be reduced in applications such as the measurement of acoustical absorption coefficients in the reverberation rooms of Research Department. In this application, the test-tape is played into the room and the paper tape punched immediately without the use of an intermediate recording.

A further development will be the modification of the equipment to carry out the method of reverberation time measurement recently proposed by Schroeder.[8] In this method, the decaying signal is squared and integrated over reversed time. Schroeder claims that the resultant curve, unlike the conventional decay curve, is independent of the precise point at which the excitation signal is cut off. Smoother decays are thus produced from which irrelevant fluctuations have been removed and which therefore lend themselves to more accurate analysis.

7. References

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APPENDIX I

PROCEDURE FOR ACOUSTIC TESTS – INSTRUCTIONS TO OPERATORS
ACOUSTIC TESTS USING PRE-RECORDED TAPES FOR COMPUTER PROCESSING – SERIES IV

A. Reverberation Time

Equipment Required

Reverberation test tape (Research Department) Series IV.
Replay machine, $7\frac{1}{2}$ in/s (191 mm/s)
Record machine, $7\frac{1}{2}$ in/s (191 mm/s)
Loudspeaker amplifier
High-output loudspeaker (e.g. LSU/10, LS5/1, LS3/1)
Microphone amplifier with P.P.M.
Microphone, preferably omni-directional.

Reverberation Test Tape

Recorded at $7\frac{1}{2}$in/s (191mm/s) full track. Signal consists of:

(i) Three one-second bursts of $\frac{1}{3}$-octave bandwidth noise, repeated at 23 centre frequencies ranging from 50 Hz to 8 kHz in steps of $\frac{1}{3}$-octave upwards.

(ii) –(x) Signal (i) repeated a further nine times.
Signal bands are separated by yellow marker tapes, excepting signals (v) and (vi) which are separated by a blue marker tape.

Note that there are no line-up tones or announcements. The tape after signal (x) has no useful recorded signal but it is retained because otherwise the variation in tape speed due to different diameters of reservoir and take-up spools might be too great for accurate data processing.

Procedure

To carry out the test, it is necessary to replay the test tape over the loudspeaker and to record the sound in the room simultaneously. In some studios, simultaneous record and replay using the studio microphone channels and the studio loudspeaker will be frustrated by the anti-howl back circuits.

In Type B equipment, Marks II and III, this difficulty may be overcome by replaying through the ACOUSTIC GRAM channel.

(i) Place the loudspeaker near one of the corners of the room.

(ii) If the ventilation equipment is audible, switch it off. Low background noise is essential for a successful test.

(iii) Play signal (i) over the loudspeaker, setting the loudspeaker channel gain to the maximum possible without seriously overloading the loudspeaker. Some distortion is permissible in order to achieve the maximum signal-to-noise ratio. With most loudspeakers, the low frequencies will overload first so it saves time to carry out the initial trial gain adjustments at low frequencies only, checking finally with the whole of signal (i).

(iv) Rewind the test tape to the beginning. Place a 2400 ft (732 m) spool of tape on the recording machine. Do not use a smaller length of tape since with some machines this may lead to unacceptable variations in tape speed.

(v) Place the microphone at moderate height towards the centre of the room. Play signal (i) and record the signal from the microphone, keeping the recorded level high, but avoiding excessive overload. The first of the triple noise-bursts at each frequency may be used to check the level, the gain being altered, if necessary, before the third burst starts.

(vi) When the first yellow marker tape is reached, stop the replay machine and then the record machine. Move the microphone to a different position and height ensuring that it is further than six feet from the loudspeaker. Start the record machine and then the replay machine, recording the signals for the new microphone position as before. (It is essential to follow the correct sequence in switching the tape machines on and off, otherwise there may be insufficient time allowed for the automatic data processing equipment to reset between successive microphone positions.)

(vii) Repeat instruction (vi) with further microphone positions randomly disposed over the working area of the room. Carry on until recordings have been made with the number of microphone positions noted on the form* attached to these instructions.

(viii) If there are further rooms to be tested, use the beginning of a 2400 ft (732 m) reel for each recording.

Return of Tapes to Studio after Processing

If you would like us to return your tape recordings after processing, please indicate on the form attached to these instructions.

* Not reproduced.

APPENDIX II

Summarised Procedure for Automatic Processing of Reverberation Test Tapes

1. Connect up the signal chain as shown in the block diagram (Fig. 7), paying due attention to the various signal level recommendations and restrictions.
2. Replay the test tape. The equipment should then cause the decay after every third excitation signal to be sampled, digitised and punched on the paper tape 100 times per second for 3·30s after passing through the appropriate $\frac{1}{3}$-octave filter. After the 23 frequency bands between 50Hz and 8kHz have been dealt with, the filters will automatically reset to 50Hz and the timing unit will wait for 12·6s before accepting the first signal at the next microphone position. There is no limit to the number of microphone positions which can be handled sequentially by the equipment.
3. Signal Faults. If the signal does not lie within the timing or amplitude limits specified elsewhere for the test tapes then the signal will not be punched and the red NO-GO lamp will come on, and there will be a hold-up in the processing just after the beginning of the third excitation in the faulty group.

   After rewinding the appropriate length of test tape, and pressing the reset button on the timing unit, a second attempt may be made to process the faulty signal. If there is again a hold-up then the signal can be either rejected or digitised and punched by repeating the process with the RESET button on the timing unit held down throughout the duration of the three excitations in the group.
4. If, for any reason, it is decided to abandon processing in the middle of a frequency run, ensure that the bistable circuits are correctly set before recommencing by switching off the −48V power supply and then the −10V power supply for a few seconds. Switch on in the reverse order.

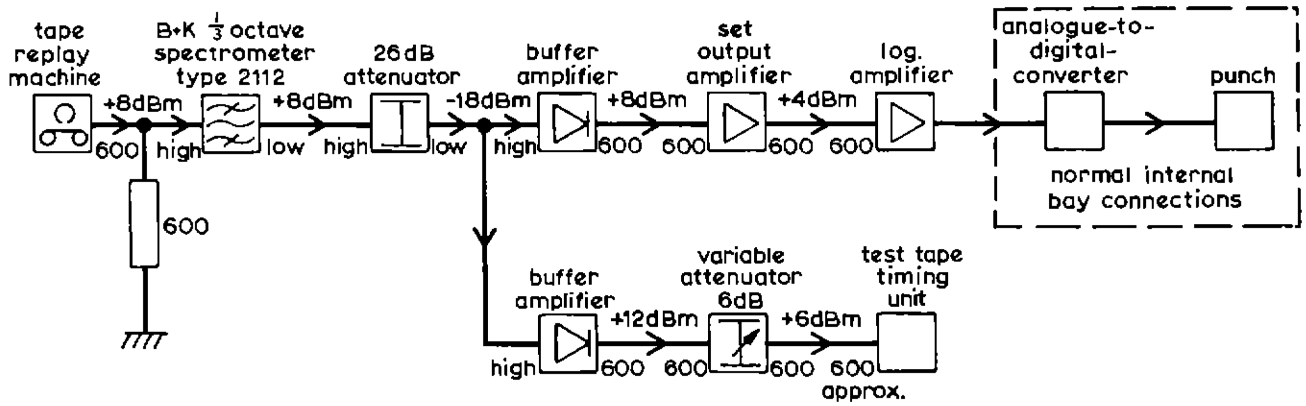

Fig. 7 – Signal chain and recommended signal levels

APPENDIX III

Computer Programme RT.IV.DS.

Fig. 8 shows a simplified flow diagram of programme RT.IV.DS.

A copy of the programme appears at the end of this appendix.

From 72),* the initial data are read in and the computer waits. The transcoded tape is then loaded into the reader ensuring that the reader light illuminates the character immediately preceding the first non-blank character, before the READER button is depressed. The programme then sets M, K1, J2 and N to their initial values.

M is the decay curve counter.

$K1 = 0$ before the reverberation time of the first decay is calculated, and is set to 1 thereafter.

J2 is the decay flag and is set to 1 from the start of a decay to the time its reverberation time is calculated; at all other times it is set to zero.

After the WAIT (150)) is cleared, a machine code block (73)) INPUTs the transcoded tape, decodes a pair of characters and sets the variable L to their value. Further pairs are INPUT until an L less than L1 is encountered (start of decay). N is then reset to zero and the decay flag J2 set to 1. L is converted to floating-point form (75)) and the value placed in BO. The allowable noise level is the sum of BO (the ADC reading at the start of a decay) and D3 (the minimum acceptable signal-to-noise ratio in ADC units).

Further characters are INPUT at 73) but now $J2 = 1$ so, after being counted, the samples are immediately converted into floating-point form (B1, B2, B3 . . . etc.).

When $B(N1 + 11)$ is reached, the arithmetic mean of $B(N1 + 1)$ to $B(N1 + 10)$ is determined (60)). Twenty ADC units are subtracted, leaving B(1000) approximately 6 dB above the average noise level following the decay. If the signal-to-noise ratio is less than expected then $B1000 + 20 \ngtr A2$ and both TM (reverberation time) and CM (range of computation) are set to zero.

If the signal-to-noise ratio is acceptable the values of BO to $B(N1 + 1)$ are examined in turn until one exceeding B1000 is found (77)). Only the samples up to this one are used in the computation of reverberation time.

Next, a test has to be applied which is a consequence of the unreliability of the reader used in the transcoding operation. Occasionally a single transcoding error occurred such that a sample of noise was converted to a level high enough for it to be misinterpreted as the commencement of a decay. It was rare for the next three samples to be mis-transcoded in a similar way, so it should be possible to check the authenticity of the 'decay' by examining whether or not there are less than five samples constituting the decay. This is done after 69). If $N < 5$, the fact is printed out, together with the values of BO to B4, and the computer waits. The computer operator then examines the values of BO to B4. A sequence such as

48 225 227 226 225

is clearly an error since the numbers do not increase steadily. A very rare but possible sequence such as

122 144 166 188 190

may either be a genuine decay of reverberation time approximately 0·10 s and very poor signal-to-noise ratio or the result of a click on the tape replay channel. The action to be taken in such an event is too complicated to discuss here.

Assuming that the printout indicates a transcoding error, the WAIT is cleared and all the remaining stored samples of BI are examined for the start of a decay. If there is none, further characters are read in from 73). If a decay sample is found, BO is reset to the sample value and B1 to $B(N1 + 11)$ are reset, in turn, to the succeeding sample values.

The mean noise level is then calculated at 60), as before, the samples to be used in the reverberation time computation are determined at 77) and 69) is encountered again.

Assuming that there are more than four samples to the decay, the reverberation time is then computed under sub-routine 78). This sub-routine computes the slope of the line of best-fit through the samples, such that the sum of the squares of the deviations of the BI from the line is a minimum. Note that samples BO to B2 are discarded if they do not correspond to a monotonic decrease in level. This is a safeguard against the inclusion of the last part of an excitation signal as part of the beginning of a decay.

The decay counter, M, is then increased by one and the reverberation time TM and range of computation CM are stored. Sub-routine 87), which changes the minimum acceptable signal-to-noise ratio after 125 Hz and 8 kHz, is carried out. Further characters are INPUT at 73) until the last decay has been computed. The means and standard deviations of the reverberation times are then computed and printed out.

The transcoded data may be punched in parts with no special precautions as the programme causes a WAIT to occur if a five-track blank is encountered.

Should the transcoded tape be loaded incorrectly, such that wrong pairs are read in, this is usually revealed by a series of 'N less than 5 at M = n' statements being printed up as though there were a large number of transcoding errors. The only safe course then is to start afresh.

Programme RT.IV.DS is the development of a number of earlier programmes. It was arrived at by editing these earlier programmes and no doubt contains some minor logical redundancies as a consequence. It is thought that the time spent in seeking out, correcting, and testing these redundancies is likely to be greater than the resultant saving in computer time.

* In this description, programme reference numbers are shown thus: n) where n is an integer.

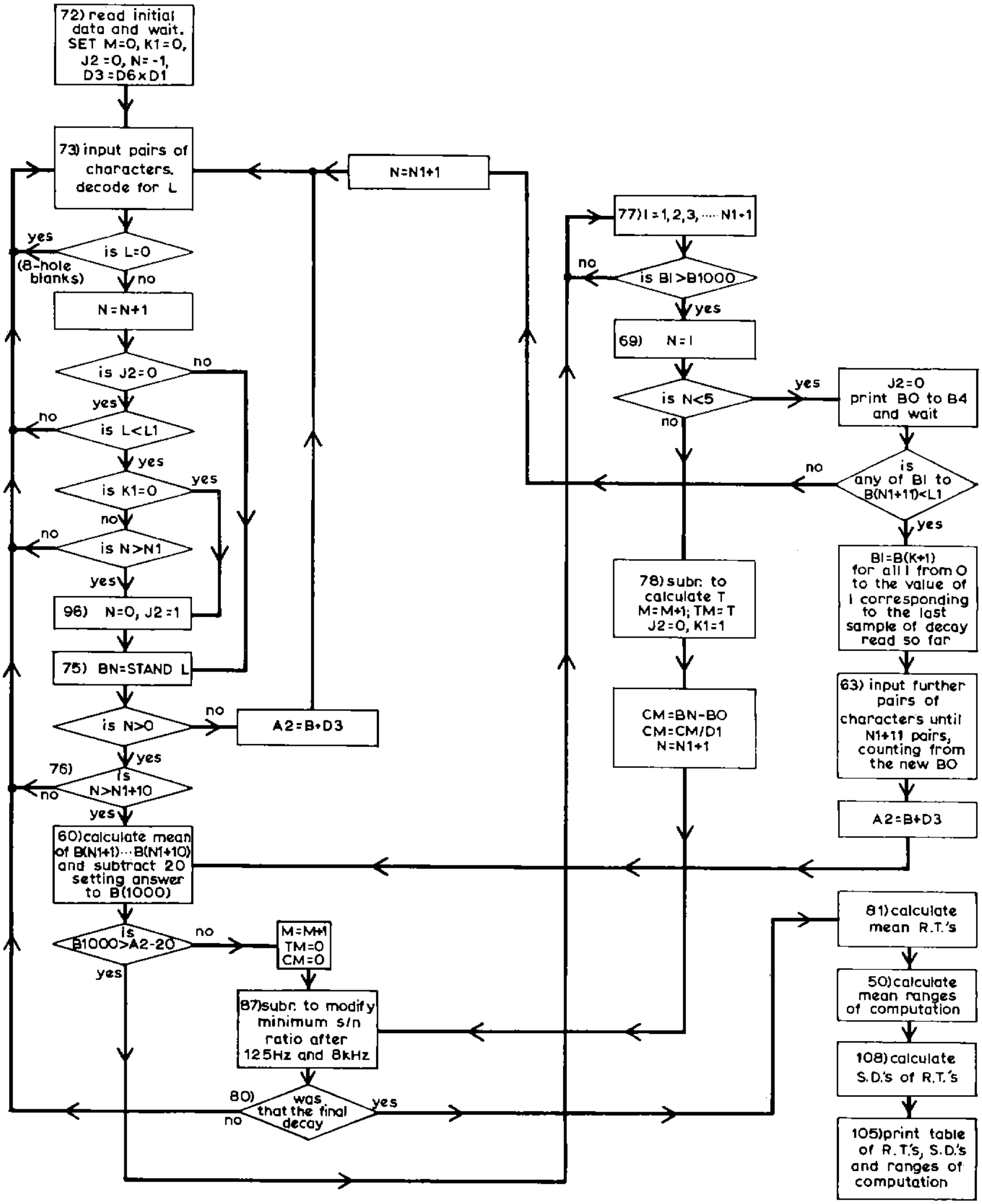

Fig. 8 – Simplified flow diagram of programme RT.IV.DS.

```
:: REV TIME: MARK IV.D.S.(NOISE SAMPLED). MODIFIED RANGE.
SETV A(5)B(1000)C(300)D(10)F(5)S(10)T(300)Z(5)
SETS I(5)J(5)K(5)L(5)M(5)N(5)P(10)
SETF INT SQRT

SETR 150

70)LINES 10
71)TITLE
        REV TIME :MARK IVDS(NOISE SAMPLED). MODIFIED RANGE.

  FREQUENCY              REVERBERATION TIME (SECS.)


72)READ P2::    NUMBER OF MIC. POSITIONS
READ D1::   MEAN SLOPE OF LOG AMP CHARACTERISTIC IN UNITS/DB
READ L1::   ALLOWED MAXIMUM ADC READING AT START OF DECAY
READ N1::   MAXIMUM DURATION OF ANALYSIS IN  CENTISECS.
READ D6:: MINIMUM S/N RATIO ACCEPTABLE IN DB UP TO 125HZ.
READ D7::                   DITTO                  ABOVE 125 HZ
D5=0.60*D1
READ S9::   DEVIATION OF TAPE SPEED FROM NOMINAL VALUE, %
T299=100+S9
T299=T299/100
D5=D5*T299
89)M=0
P=23*P2
K1=0
D3=D6*D1
90)J2=0
N=-1
150)WAIT
73)@060:710
42150):035,
554:16L
710:035,
04L:420,
20L:0015
)
N=N+1
JUMP UNLESS J2=0@75
JUMP UNLESS L$L1@73
JUMP IF K1=0@96
JUMP UNLESS N%N1@73
96)N=0
J2=1
75)BN=STAND L
JUMP IF N%0@76
A2=B+D3
JUMP @73
76)JUMP UNLESS N%N1+10@73
60)B1000=0
P7=N1+1
VARY I=P7:1:10
B1000=B1000+BI
68)REPEAT I
B1000=B1000/10
B1000=B1000-20


CHECK B1000
JUMP IF B1000%A2-20@77
M=M+1
TM=0
CM=0
CHECK TM
J2=0
K1=1
SUBR 87
JUMP @80
77)VARY I=0:1:P7
JUMP IF BI%B1000@69
67)REPEAT I
69)N=I
JUMP UNLESS N$5@149
J2=0

LINE
TITLE N LESS THAN 5@M =
PRINT M,3
TITLE   B0 TO B4:
CYCLE I=0:1:4
PRINT BI,3:0
REPEAT I
WAIT
B999=STAND L1
P9=N1+11
CYCLE K=N:1:P9
JUMP IF BK$B999@66
65)REPEAT K
66)P9=P9-K
CYCLE I=0:1:P9
BI=B(K+I)
64)REPEAT I
N=N1+1
JUMP UNLESS B0$B999@73
N=P9
63)JUMP @62
61)WAIT
62)@060:710
4261):035,
554:16L
710:035,
04L:420,
20L:0015
)
```

```
N=N+1
J2=1
BN=STAND L
JUMP UNLESS N%N1+10@62
A2=B+D3
JUMP @60
149)SUBR 78
K1=1
J2=0
M=M+1
TM=T
CM=BN-B
CM=CM/D1
N=N1+1
SUBR 87
80)JUMP UNLESS M=P@73
81)CYCLE I=1:1:23
P3=P-23
I3=I+P3
T(P+I)=0
P4=P2
CYCLE K=I:23:I3
T(P+I)=T(P+I)+TK
JUMP UNLESS TK=0@83
P4=P4-1
83)REPEAT K
JUMP IF P4=0@112
Z=STAND P4
T(P+I)=T(P+I)/Z
JUMP @84
112)SUBR 110
84)REPEAT I

50)CYCLE I=1:1:23
P3=P-23
I3=I+P3
C(P+I)=0
P4=P2
CYCLE K=I:23:I3
C(P+I)=C(P+I)+CK
JUMP UNLESS CK=0@51
P4=P4-1
51)REPEAT K
JUMP IF P4=0@52
Z=STAND P4
C(P+I)=C(P+I)/Z
52)REPEAT I

108)CYCLE I=1:1:23
I3=I+P3
B(P+I)=0
P4=P2-1
CYCLE K=I:23:I3
BK=T(P+I)-TK
JUMP UNLESS TK=0@113
BK=0
113)BK=BK*BK
B(P+I)=B(P+I)+BK
JUMP UNLESS TK=0@107
P4=P4-1
107)REPEAT K
JUMP IF P4=0@109
Z=STAND P4
B(P+I)=B(P+I)/Z
B(P+I)=SQRT B(P+I)
JUMP @111
109)SUBR 110
111)REPEAT I

LINE
105)TITLE
   HZ


SPACES 4
CYCLE I=1:1:P2
SPACES 3
PRINT I,2
101)REPEAT I
106)TITLE           MEAN      S.D.
LINES 2
F=49.6063/1.259921
CYCLE I=1:1:23
LINES 2
SPACES 1
F=F*1.259921
PRINT F,4:0
SPACES 2
P4=P2-1
CYCLE K=0:1:P4
PRINT T(I+23K),2:2
102)REPEAT K
SPACES 3
PRINT T(P+I),2:2
SPACES 1
PRINT B(P+I),2:2
85)REPEAT I
LINES 4
TITLE                      RANGE OF COMPUTATION, DB

   HZ
CYCLE I=1:1:P2
SPACES 3
PRINT I,2
REPEAT I
TITLE          MEAN
```

```
LINES 2
F=49.6063/1.259921
CYCLE I=1:1:23
LINES 2
SPACES 1
F=F•1.259921
PRINT F,4:0
SPACES 2
P4=P2-1
CYCLE K=0:1:P4
PRINT C(I+23K),2:2
REPEAT K
SPACES 5
PRINT C(P+I),2:0
REPEAT I
WAIT
JUMP @70
78)S1=0
S2=0
S3=0
S4=0
N2=N-1


JUMP IF B2$B3@91
N4=3
JUMP @94
91)JUMP IF B1$B2@92
N4=2
JUMP @94
92)JUMP IF B$11@93
N4=1
JUMP @94
93)N4=0
94)CYCLE I=N4:1:N2
Z=STAND I
S1=S1+Z
S2=S2+BI
S5=Z•B1
S3=S3+S5
Z1=Z•Z
S4=S4+Z1
82)REPEAT I
Z4=STAND N4
Z=Z+1
Z=Z-Z4
S6=S1•S2
S6=S6/Z
S6=S3-S6
S7=S1•S1
S7=S7/Z
S7=S4-S7
T=S6/S7
T=D5/T
CHECK T
EXIT

87)VARY I=5:23:P2
JUMP IF M=1@88
103)REPEAT I
VARY I=23:23:P2
JUMP IF M=1@95
104)REPEAT I
EXIT
88)D3=D7•D7
EXIT
95)D3=D6•D7
100)EXIT

110)LINE
TITLE Z=0@ T
I4=P+I
PRINT I4,3
EXIT

START 70
```

Published by the British Broadcasting Corporation, 35 Marylebone High Street, London, W1M 4AA. Printed in England by The Broadwater Press Ltd, Welwyn Garden City, Herts. SBN: 563 09285 8