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ACOUSTIC REVERBERATION KIT - A NEW TOOL, TO ASSIST IN ACOUSTIC MEASUREMENTS

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1. INTRODUCTION

The IBA is responsible for the technical performance of television and radio stations and these standards are laid down in the Authority's Code of Practice[1]. The Code's provisions lay down performance requirements in various sections which cover the generic performance of broadcast equipments and also, where radio broadcasting is concerned, the acoustic performance of studios.

The principal acoustic performance parameters of interest are threefold,

a. reverberation times
b. background noise
c. isolation from general airborne noise and from impact noise.

Typically measurements of these parameters are made at a number of positions in each studio area and as each ILR station may possess some five studio areas then it is clear that a great amount of time is spent gathering and analysing acoustic data. The acoustic measurement sequence is carried out both on new stations and also after studio modifications. In addition there is an increase in the number of sound control rooms and studios at television stations requesting affirmation of their acoustic performance.

At the present time there are 48 operational ILR stations with more planned and the work involved imposes a workload on staff and current equipment which is increasingly difficult to fulfill. Currently acoustic tests have been carried out using commercially available apparatus. For reverberation times a conventional arrangement of Fig 1 has been employed.

The method employed for reverberation times particularly leads to an extremely tedious process of manually analysing some 300 pen traces for each location in a room, such as those shown in Fig 2. The storage and categorising of these traces for necessary archival purposes poses several difficulties and data retrieval is slow. Acoustic data collection is largely a matter of data logging and thus it was envisaged that some means of harnessing microprocessor power to collect the data and to file it electronically would be desirable.

This paper describes the outline of an apparatus designed to be used with a commercially available microcomputer. It has acquired the acronym ARK (Acoustic Reverberation Kit) though the equipment has uses in other acoustic fields.

2. BACKGROUND

The efforts employed in assessing reverberation times were naturally those for which some automation was sought. The inclusion of facilities to measure background noise and isolation would use the system components in a similar manner and the processing of data was seen more as a software task.

The approach to measuring reverberation times within the IBA has been to use filtered bands of pink noise and plot the decay curve using a high speed pen plotter. The main alternative described by Schroeder[2] has never been seriously considered due to the expense of the necessary processing and lack of portable real-time analyser[3]. It is also necessary to appreciate that considerable confidence and experience had been built up with many acoustic - consultants using the filtered pink noise approach. Hence, automation of the measurement technique needed to be directly related to the trusted approach of previous years.

A search of available literature revealed no suitable existing product which would have satisfied the requests of both engineers and acousticians required to operate the equipment and therefore it was decided to investigate the possibility of producing a purpose built system.

An Apple microcomputer had been acquired for use in various projects where it performed the role of a reasonably intelligent controller. Initial experiments involving construction of cards to plug in the application slots of the Apple were disappointing. Attempts to modify directly existing commercial equipment would have raised doubts concerning the integrity of the equipment.

It was decided to produce a separate unit see Fig 3, which would contain in a much smaller and lighter package, all of the required measuring facilities which at that time occupied some five separate units. A key requirement of ARK, to achieve the standards of performance obtained from our current equipment, is made possible by the development of modern electronic components.

3. THE SYSTEM

The system block diagram is given in Fig 4.

The division of the system into its functional blocks was required so that the activity of any of the system components could be controlled independently via the microcomputer keyboard. The essential bonus of partitioning the equipment in this manner is that the ARK box itself becomes a collection of relatively simple analog circuits separated from the digital complexity and programming of its controller.

The use of a commonplace microcomputer as a controller brings its advantages.

a. Relative cheapness of a twin disc microcomputer.

b. Easy programming in a high level language (BASIC) with only the short high speed serial averaging routines being in machine language.

c. Provision of an already proven disc operating system which allows the data to be stored onto floppy disc.

d. Ease of finding replacement parts should some part of the microcomputer fail. Most large towns have agents representing the major microcomputer manufacturers.

e. Ease with which the tasks performed by the equipment may be altered by competent programming. Where there is a need for modifications to the standard procedures required by the Code, a technique of menu windows has been provided.

f. Facility of using a similar microcomputer at IBA's base in order to review or further analyse the performance of studios.

Building the analog sections of ARK in a free standing box allow it to be controlled by many other types of microcomputer, other than the basic Apple microcomputers which we use, via a simple interface. Thus the cost of ARK replication is relatively low - less than £300 in component costs and the construction is such that it could readily be built in small batch quantities.

3.1 Specifications

3.1.1 Signal generation

The signal generator is based on a 28 stage pseudo random binary sequence generator which takes in excess of 13 minutes for the sequence to repeat. The digital waveform is low pass filtered before being applied to a pink noise filter thus making available both white noise (band limited to 20kHz) and pink noise. The required centre frequency and bandwidth are selected using a filter system based on the use of switched capacitor filters. Centre frequency setting is provided by a clock signal of appropriate frequency. A simple series of dividers provides the relevant frequencies from a single 4.43MHz crystal oscillator. The frequency range provided is from 10kHz through to 25Hz.

The signal returned from the filter card is controlled in level by a logarithmically scaled multiplying digital to analog converter, so that the setting of the output level is under the control of the microcomputer. Complete gating of the noise signal is provided by a series shunt transmission gate.

The output arrangements permit the twin output sockets to be driven either single ended or electronically balanced by the choice of internal connectors. The facility to drive two sets of speakers is especially advantageous when measuring the performance of inter-area isolation.

3.1.2 Signal Reception

The microphone amp design is based upon the requirements of the Bruel and Kjaer capacitor microphone system. The gain is switched by reed relays in accurately determined 10dB steps. The noise performance has been kept sufficiently low for the system noise floor to allow reliable measurement of the background noise in any studio. One of two inputs may be selected both of which accept Bruel and Kjaer connectors. These inputs are each paralleled to standard three pole jills intended for use with other microphones or line level sources. The signal is passed through a 25Hz high pass filter and is then presented to the filters card. For monitoring purposes and for driving external filters or other apparatus the preamplified signal is available on the front panel.

The filtering action is primarily performed by switched capacitor filters as used in the generation side of ARK. However the response time of a filter is related to its bandwidth and in the unusual case of reverberation analysis using 1/3 octave bandwidth filters the response time could readily exceed that expected for the reverberation time in a small studio at low frequencies. Thus for the eight 1/3 octave spaced bands from 40Hz through to 200Hz the preamplified signal is routed instead through an analog filter. This filter comprises two stagger tuned bandpass filters based on a twin integrator loop. The centre frequencies are set by using an analog multiplexer to select the appropriate frequency determining resistor. The particular topology chosen had the property that changes in the centre frequency have negligible effect on the Q factor of the filter and thus the resultant bandwidth of the two filter sections remains at one octave. The final output of the filters section is taken via a front panel breakjack to the detector card.

The detector card uses a precision full wave rectifier and an averager followed by a wide dynamic range logging amplifier. The output of the logger is digitised by an 8 bit analog to digital converter each time a sample is requested by the controlling microcomputer. The effective averaging time of the rectifier is equivalent to a reverberation time of 80ms and linearity of the combined detector and logger enables an 80dB range to be encoded with the error of around 1.5dB concentrated in the last 10dB of range.

However the arrangement of signal levels within ARK are such that the last 10dB of logging range covers the noise output of the switched capacitor filters, producing a total useable dynamic range for the system of approximately 135dB. The analog to digital converter scaling is such that each bit represents a level of 0.32dB.

Each of the main functions above are implemented on a standard eurocard size board. An extender card which also carries out certain test facilities is provided. All of the cards reside in a simple frame which fits within a 2U rack mounting profile. This frame also carries the digital interface, control circuitry and the power supply.

The power supply produces all of the necessary voltages ranging from the 200 volts for the microphones to 5 volts for the interface. An advantage of using a remote CRT display is that only simple grounding and screening techniques are required.

As with many remotely controlled devices, an indicator panel is provided that confirms to the operator that the equipment is functioning correctly. The display comprises 8 LEDs which indicate the following:

(a) which input socket is routed to the preamplifier

(b) preamplifier gain has been changed

(c) filter centre frequency has been changed

(d) that the receive filter has been changed from the switched capacitor filter to the analog one

(e) the analog to digital converter is in use

(f) the noise output level has been changed

(g) the noise output is enabled

(h) the motorised microphone boom arm is moving.

4. INTERFACE

The simple parallel interface protocol required that the simultaneous presentation of a 4 bit nybble and an 8 bit data bus is accompanied by a simple strobe pulse. The bus is bidirectional and may use address 0 to 14 for communication. Data being read back into the controlling micro is buffered at the interface card and a read to address 15 is used exclusively for this. The system has the advantage over alternatives such as GPIB, of cheapness and ease of use, whilst not denying the possibility of change to other systems.

5. TECHNIQUES AND SOFTWARE

5.1 Introduction

The existing method of measuring reverberation times using the pen recorder has several disadvantages, a major one being the difficulty in interpreting the traces obtained. Typical decay curves obtained in radio broadcast studios often deviate markedly from a straight line, especially at the lower frequencies, as in Fig 5. Variable fluctuations are often evident and are often due purely to the statistical nature of the summation of individual reflections from the studio walls or beats between two closely spaced room modes. These variations severely reduce the accuracy and repeatability of results obtained by this method. Other limitations of the present system are the writing speed and paper speed of the pen recorder. The writing speed of the recorder sets a limit to the fastest reverberation time at approximately 100ms and this limiting reverberation time increases at lower frequencies due to the time domain response of the receiving bandpass filters. However this limitation is sufficient in most studio environments but is a variable that must be carefully chosen to avoid erroneous results. When measuring fast reverberation times typically found in broadcast studios, even the use of the maximum paper speed causes the recorder decay slope to be very steep, thus increasing the scope for error when using the measurement protractor.

A further disadvantage of the existing measurement system was inconvenience of the data storage medium for archival and further analysis purposes. The paper rolls obtained were bulky and retrieval of a particular set of traces was a lengthy and tedious procedure.

5.2 Sampling and Averaging

The development of ARK was instigated to overcome or, at least, minimise the problems detailed above. A fundamental design aim was to enable the averaging of successive traces at a particular frequency so as to reduce the random fluctuations encountered on decay curves, especially at the lower frequencies, and so highlight the true decay pattern. This is possible since the samples taken are referenced to a fixed point in time, the cut-off of the noise source, and therefore each data point of the new trace can be averaged with the same point of the previous trace. ARK implements a serial average process (see Appendix 1) whereby the 'averaged' curve is updated by the new data on each successive test. This method allows between 2 and 256 traces to be averaged in real time and also reduces the memory space required. Each decay measured has an equal weighting on the final averaged decay curve. This serial averaging process is illustrated schematically in Fig 6 and a typical averaged decay is shown in Fig 7. The number of samples taken during each decay is fixed to 249 although the period over which they are taken may be varied via the controller keyboard.

An advantage of digitally sampling the decay curves is the ease of processing the data samples during or after averaging. In addition to the serial averaging process a simple weighting process (see Appendix 2) is available which emphasises the new data and reduces the significance of the previous averaged data. This process can operate continuously and so is useful during investigative and remedial work on the studio acoustics.

5.3 Reverberation Time Analysis

The ability to control the excitation noise level and mic amp gain independently allows the controlling microprocessor to perform totally automatic reverberation time data collections. Using the serial averaging algorithm used for the decay curve processing but averaging successive single data samples, the background noise in any specific octave band is measured. The third octave filtered pink noise, at the same centre frequency, is adjusted to produce sufficient dynamic range, as recommended in British Standard Specification BS:3638[4], to make a valid reverberation time measurement. This avoids excessive excitation levels which may cause non-linear effects in some sound absorbers or resonance to occur in studio fittings. Once set, an accurate measurement of the excitation noise level can be made. After allowing a sufficient delay for the noise level to reach equilibrium in the studio, the level is ramped down at a rate of 3dB per sample to minimise any transitory effects that may occur if the signal was not at a zero-crossing point. The data sampling commences at the onset of the ramp down and continues until the completion of 249 samples. The data is serially averaged between each burst.The noise source is then reinstated and the sampling process continues until completion of the pre-selected number of traces.

The averaged decay curve is then plotted onto a scaled axis on a monitor and a cursor appears on the screen which can be aligned, via the keyboard, with the final averaged trace. The cursor start point is set at 5dB below the steady state noise level before cut-off such that alignment of the cursor parallel to the curve will not obscure the trace itself. The reverberation time represented by the slope of the cursor is continuously displayed on the monitor, see Fig 7. The British Standard BS:3638 specifies that the reverberation time may be measured over a range of 30dB, starting at a level 5dB lower than the initial steady state. This has been the procedure adopted by the IBA, although it should be stressed that it is the slope between these points, and not the time interval between them, that yields the true reverberation time. The fitting of the cursor to the decay conforms to this approach.

It was initially envisaged that an automated curve fitting routine would be used to calculate the slope but this is complicated by the presence of secondary decays encountered in some studio environments. Secondary and possible tertiary decays indicate the existence of two or more modes of decay with different time constants causing curvature in the logarithmic display of the decay. Gilford and Jones[5] showed that noticeable colouration occurred when the ratio of the primary to secondary excitation levels was less than approximately 30dB and this is reflected in the IBA's Code of Practice limits for secondary decays[1]. It was therefore considered essential that any automated computation of reverberation times should yield not only primary decay times, independent of any subsequent decay modes, but also the decay rate and relative level of the secondary decay, if present. Further work is being carried out using polynomial curve fitting and smoothing algorithms in order to assess the different decay modes. Until this technique has been proved reliable, it is considered that manual interpretation of the decays will produce the acceptable level of accuracy required.

This data may then be filed onto floppy disc in the form of a 256 byte binary file, consisting of 249 bytes of data and 7 bytes of information related to the particular test ie time window, number of averaged traces, frequency etc. See Appendix 3 for details.

The current edition of software is based on a selection of options from menus. For example, in the immediate mode, a data gathering and plot sequence is performed as defined by user controllable settings. In the unattended mode a complete sequence of measurements over a range of frequencies may be initiated. The results are automatically filed and may be reviewed later. For the IBA's Code of Practice, data is collected in the band between 63Hz and 5kHz and either analysed on site and filed or stored for later analysis. A microphone calibration routine is also provided and the results for each of the two microphones are also filed on disc.

The IBA Code of Practice requires measurements to be made in the vicinity of each usual microphone and monitoring location in a studio, and that a number of measurements should be made at each of these locations by varying the position of the measurement microphone slightly. The results at these positions are then averaged to obtain the reverberation time for that location and frequency. This positional averaging reduces the effect of standing waves which can be numerous and very localised in the studio environment. In order to minimise the number of traces to be filed and save the necessity for further numerical averaging, the measuring microphone is automatically moved to the three or more different positions for each frequency. This technique results in a spatial average result for several positions in that measuring location. The position, time and speed of movement of the microphone rotator are controlled by the software.

6. DEVELOPMENTS

Although ARK was primarily developed to improve the measurement procedure for reverberation times, it is equally suited to measure background noise levels and the isolation between areas. As described in the measurement of background noise, incorporated into the reverberation time procedure, serial averaging of successive single data samples is possible. The number of samples to be averaged can be varied, as can the period between samples. Additional software routines enable the mean, peak and standard deviation of the background noise results to be calculated and displayed in the form of a histograph in various time windows. The background noise data is filed, as for the reverberation times, in 256 byte binary files.

This method of detecting background noise levels can also be used to measure the impulsive noise isolation characteristics of a studio. A tapping machine complying to the requirements of ISO 140 can be placed in the source room and the peak capture algorithm of ARK implemented to record the peak transmitted sound levels in each octave band.

An extension of the background measurement procedure is used for evaluation of the inter-area isolation performance. The two calibrated microphone input sockets available on ARK, facilitate alternate measurement of noise levels in two areas by the controller switching to the required input. The level of interfering source can be controlled automatically to produce an adequate sound pressure level in the source area in a particular octave band. The IBA Code of Practice sensibly defines the necessary interfering source level depending on the use of the area. The total sound pressure levels in the receiving room and in the interfering area are then measured. A correction for the difference between the actual source level and the Code definition is then calculated. This procedure is repeated for all necessary octave bands. The data points can be plotted against the octave centre frequencies and stored as binary files. This method of operating ARK can thus yield the sound reduction index of a building element.

7. ACKNOWLEDGEMENTS

The authors are grateful to many colleagues within the IBA and in other broadcasting organisations for all their thoughts, inspiration and guidance and to the Director of Engineering of the IBA for permission to publish this article.

8. REFERENCES

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[2] Schroeder M. R. 1965 New Methods of Measuring Reverberation Time. J Acoustic So. Am. 1965, vol 37, 3 pp 409-412

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[4] British Standards Institution - Method for the Measurement of Sound Absorption Coefficients (ISO) in a Reverberation Room - BS 3638;1963.

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9. LIST OF APPENDICES

1. Serial average process.

2. Weighting average process.

3. Memory map of ARK reverberation time data file.

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APPENDIX 1 - Serial Average Process

The 'new' data samples collected are averaged sequentially with the present 'running' average samples using the algorithm below. Each trace measured has an equal weighting on the final serial average.

$$avg_n = \frac{n-1}{n} (avg_{n-1}) + \frac{1}{n} (data_n)$$

$$= \frac{(n-1 (avg_{n-1}) + data_n)}{n}$$

where: n = number of trace

avg_n = serial average of first n traces

$data_n$ = new data of nth trace

This averaging process is completed in real time using machine language programs divided into the following three steps

STEP 1 $(n-1) \times avg_{n-1} = x$

STEP 2 $x + data_n = y$

STEP 3 $y \div n \; (= avg_n)$

APPENDIX 2 - Weighted Average Process

The weighted average routine differs from the serial average in that each trace does not contribute equally to the displayed result. The new data has greater significance than the previous results. This is achieved by introducing a weighting factor into the averaging 'algorithm', as shown below.

$$avg_n = (1-w).(avg_{n-1}) + w.(data_n)$$

where: avg_n = weighted average of first n traces

$data_n$ = new data of nth trace

w = weighting factor

This process is completed in a real time machine language routine, divided into the following steps.

STEP 1	$(1-w) \times avg_{n-1}$	$= x$
STEP 2	$w \times data_n$	$= y$
STEP 3	$x + y$	$(= avg_n)$

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APPENDIX 3 - Memory Map of ARK Reverberation Time Data File

The data stored for each reverberation time measurement is filed as a block of 256 bytes, comprising 249 data samples and 7 bytes of information. The memory map of the binary file is shown below:-

BYTE	
0	249TH DATA SAMPLE
|	|
F8	1ST DATA SAMPLE
F9	MEASURED 'ON' LEVEL OF PINK NOISE
FA	PREAMPLIFIER GAIN SETTING
FB	PINK NOISE LEVEL
FC	NUMBER OF TRACES AVERAGED
FD	POINTER OF CURSOR SLOPE
FE	TIME WINDOW OF TRACES (0.5, 1, 2 OR 4 SECONDS)
FF	FREQUENCY

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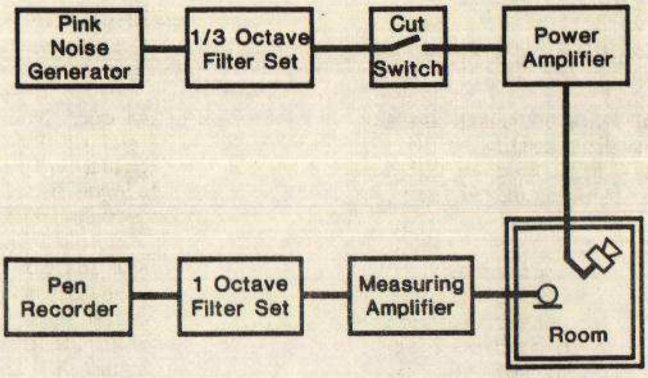

Fig 1 Conventional arrangement used to measure reverberation times

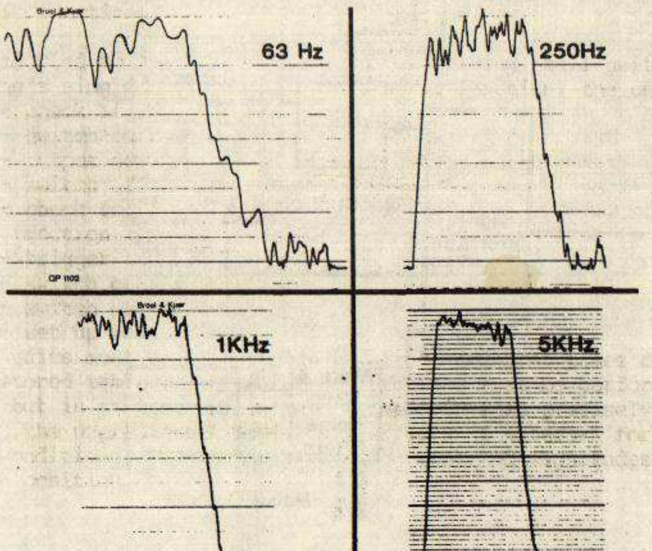

Fig 2 Typical pen traces using conventional equipment

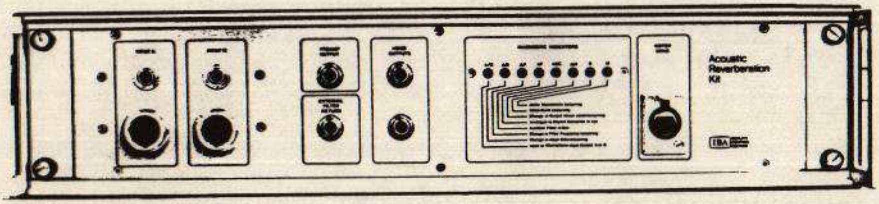

Fig 3 A view of the production model of ARK

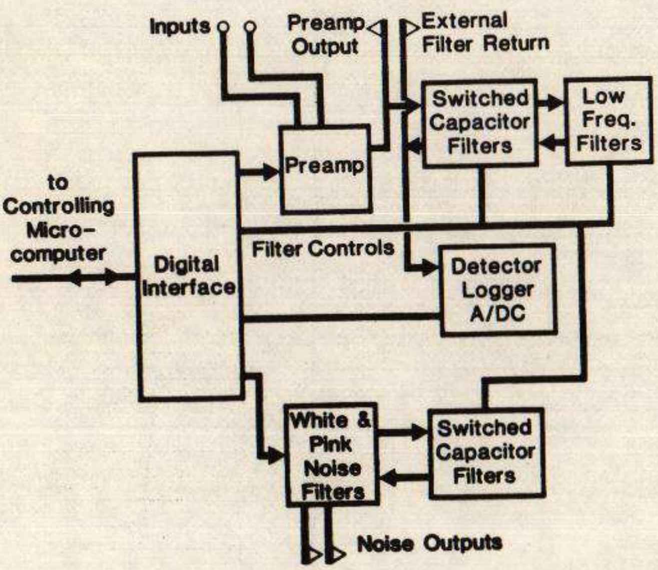

Fig 4 System diagram of ARK

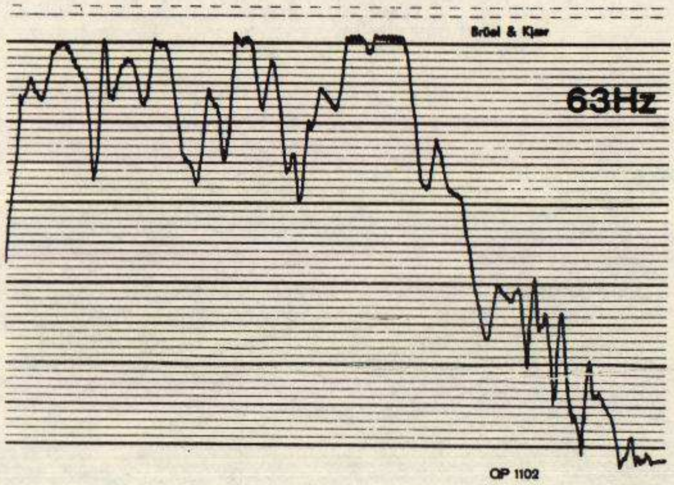

Fig 5 Typical reverberation delay obtained with the arrangement of Fig 1

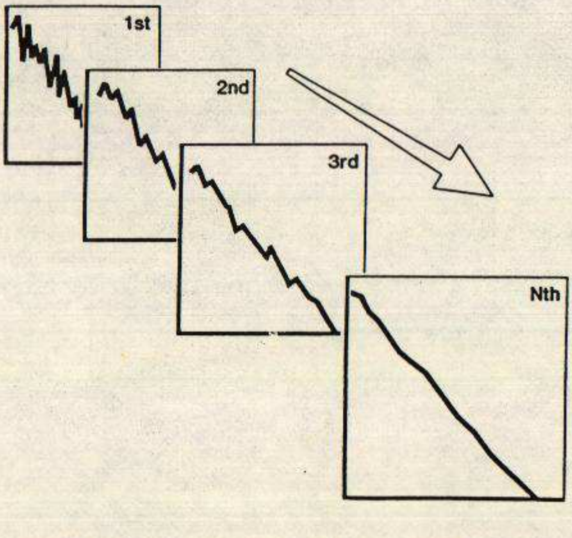

Fig 6 Schematic representation of the serial average process

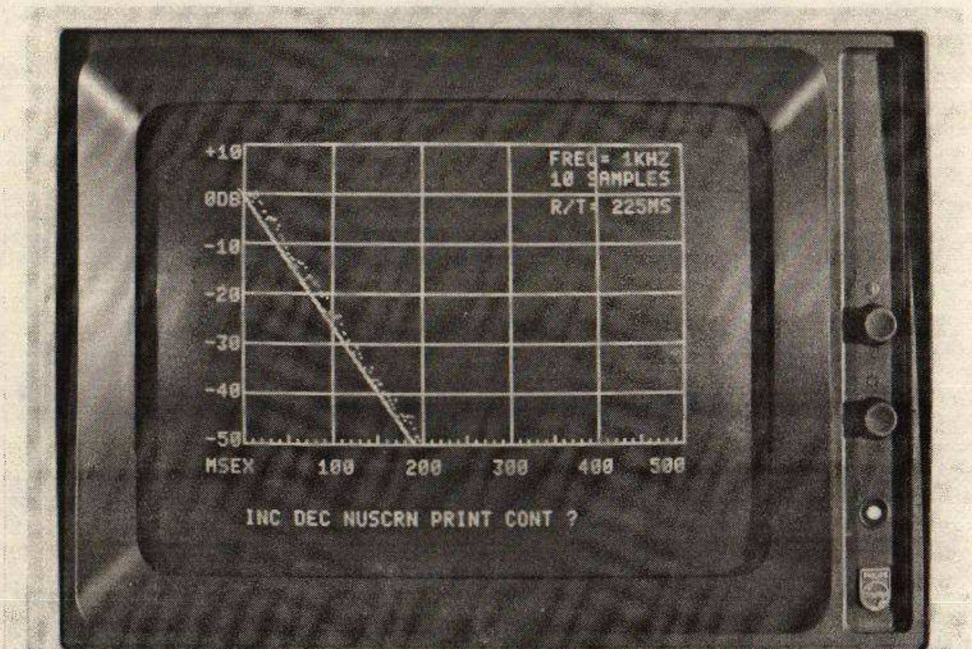

Fig 7 Typical averaged decay obtained using ARK. Note that the straight line is ARK's user moveable cursor.