4422

Instructions and Applications

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Reverberation Processor Type 4422

An instrument designed specifically for the measurement of the reverberation characteristics of enclosures and structures using the SCHROEDER-KUTTRUFF method of impulsive excitation.

The 4422 acts as a source of impulsive signals, a processor of the enclosure or structural response signals, an automatic indicator of reverberation time and a controller of associated Brüel & Kjær instruments.

BRÜEL & KJÆR

030-0566

REVERBERATION PROCESSOR TYPE 4422

August 1972

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1. INTRODUCTION

It has long been recognized that one significant parameter for describing the subjective quality of auditoria and concert halls etc. is the reverberation time. Reverberation time is defined as the time for the sound level in a room or enclosure to decay through 60 dB. The usual method of determining the reverberation time, to date, has been the Interrupted Noise Method. This method involves excitation of the room or enclosure by either filtered white noise or a warble tone. The sound is abruptly switched off, and the resulting sound decay in the room recorded on a logarithmic level recorder. The reverberation time is then obtained from the best straight line drawn through the decay curve.

Although widely accepted, the Interrupted Noise Method does suffer from several deficiencies. As the room excitation is random in nature, the greatest deficiency is the statistical uncertainty of the noise at the instant of interruption, and hence poor repeatability of the resulting decay curve. This statistical uncertainty necessitates the averaging of many decay curves to obtain an accurate result. It is impossible to make detailed investigations of the decay curve as any irregularities tend to be swamped by superimposed random fluctuations. The uneven quality of the decay curves also necessitates averaging over a wide dynamic range (35 dB), to obtain a representative result.

The Interrupted Noise Method has also been applied to the investigation of the absorption properties of building materials. Such investigations involve measurement of the reverberation time of a reverberant room before and after a panel of the material under investigation has been introduced into the room. It can readily be seen that such investigations suffer from the basic difficulties of the Interrupted Noise Method.

Atal, with others, (1) performed an investigation in which the "subjective" reverberation time was compared with the measured reverberation time of an auditorium. He found that large differences between subjective and measured reverberation times can occur when the reverberation time is measured over 35 dB. Better agreement was found when the reverberation time was evaluated over the initial part of the decay curve. Kuttruff (2) investigated the reverberation characteristics of enclosed spaces, such as reverberant rooms, theoretically explained the occurence of bending of reverberation curves and concluded it was the initial part of the decay curve which was the most significant for evaluating the damping or absorption characteristics of absorbent materials. Jordan (3) in an investigation of the reverberant characteristics of model auditoria again considered that the reverberation time obtained from the initial part of the decay curve was an important parameter.

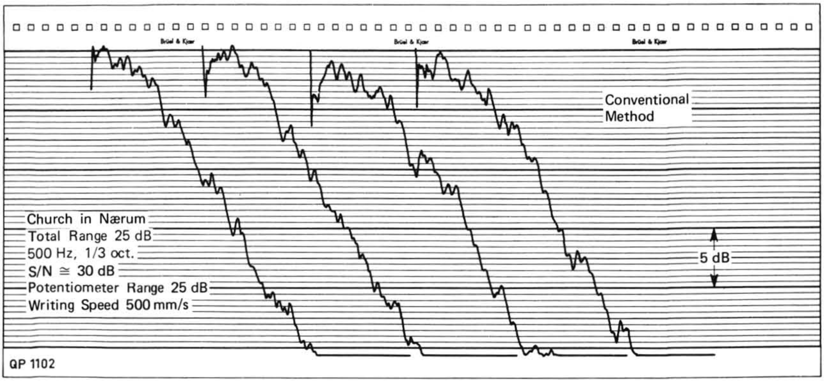
It is clear from these investigations that an instrument is required which can perform reverberation measurements free from statistical uncertainty. This would eliminate the necessity of averaging over many decay curves, allow determination of the reverberation time from only the initial portion of the decay curve and permit detailed investigation of non-exponential decays.

Schroeder (4) developed a theory, the result of which was that the ensemble average of many reverberation curves could be obtained by squaring and integrating the enclosure response to excitation of impulsive nature. Kuttruff (5) later modified the final result of this theory and suggested a practicable measurement set-up. The Schroeder – Kuttruff method was used as the basis for the design of the Reverberation Processor Type 4422. The Reverberation Processor acts as a source of rectangular pulses which can be third octave or octave filtered over the entire audio range. The filtered pulses are used for enclosure excitation, and involve no statistical uncertainty. The instrument receives the enclosure response to the excitation, squares it, and integrates it. The resulting squared decay curve can be logarithmically recorded. The Processor also automatically evaluates the Early Decay Time which is defined here as the Reverberation Time evaluated from the time for the sound level to decay from the −1 dB point to the −10 dB (or −15 dB) point.

Fig.1.1 shows reverberation decay curves obtained by the conventional Interrupted Noise Method and by the recent Schroeder – Kuttruff method, using the Reverberation Processor.

The characteristic irregularity of the curves obtained by the conventional method is clearly visible, and in contrast the high repeatability of the Schroeder – Kuttruff curves can also be seen. In cases where results obtained by the two methods differ, greater belief can be given to those obtained by the Schroeder – Kuttruff method because of the high repeatability and accuracy of the curves.

The Reverberation Processor as well as being specifically designed for



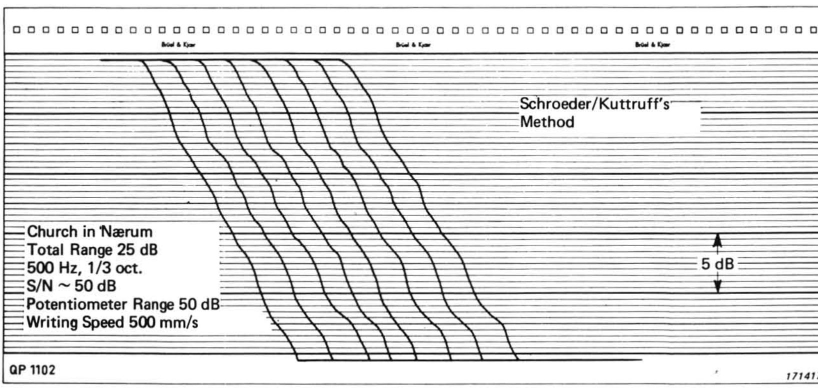


Fig.1.1. Typical Reverberation Curves Obtained by the Interrupted Noise Method and the Schroeder–Kuttruff Method

auditorium investigations and the laboratory testing of sound absorbent materials is well suited to structural vibration response testing.

References:

1. B.S. Atal, M.R. Schroeder, G.M. Sessler; "Subjective Reverberation Time and its Relation to Sound Decay" 5th International Congress on Acoustics Liege (1965) Paper G.32.

2. H. Kuttruff; "Eigenschaften und Auswertung von Nachhallkurven" Acustica Vol 8 (1958) p. 271

3. V.L. Jordan; "Acoustical Criteria for Auditoriums and their Relation to Model Techniques" J.A.S.A. vol. 47 (1970) p. 408

4. M.R. Schroeder; "New Method of Measuring Reverberation Time" J.A.S.A. Vol. 37 (1965) p. 409

5. H Kuttruff, M.J. Jusofie; "Nachhallmessungen nach dem Verfahren der integrierten Impulsantwort" Acustica Vol. 19 (1967/68) p. 56.

2. CONTROLS

2.1. FRONT PANEL

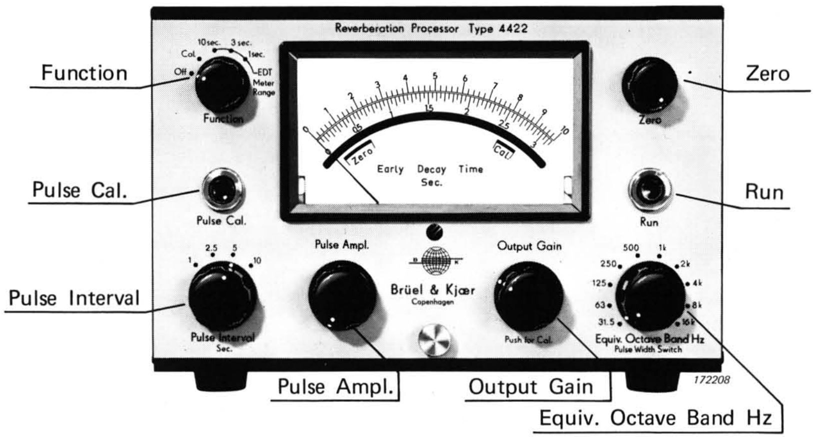


Fig.2.1. Front Panel

METER SCALE:

Fulfils the dual purposes of providing "Zero" and "Cal" ranges for calibration of the instrument and also scale markings for the direct reading of Early Decay Time.

FUNCTION:

Switching this knob to "Cal" turns the instrument on, and selects the calibration mode. Further switching allows the Early Delay Time to be read directly from the meter scale with settings of 10 sec., 3 sec. and 1 sec. full scale deflection.

PULSE CAL.:	Pressing this button results in the emission of a single pulse for calibration purposes. Holding the button pressed also activates the peak hold circuit.
PULSE INTERVAL:	Determines the time interval between the twin pulses on a measurement run. Pulse intervals provided are 1 sec., 2.5 sec., 5 sec. and 10 sec..
PULSE AMPLITUDE:	Provides adjustment of the pulse amplitude for calibration purposes.
OUTPUT GAIN:	Pushing this knob results in the emission of a pulse and activation of the peak hold circuit for calibration purposes, and turning the knob results in adjustment of the output gain of the integrator.
EQUIV. OCTAVE BAND Hz:	Determines the pulse width, and hence the equivalent octave band of the pulse.
RUN:	Pressing this button results in the emission of twin pulses for a measurement run.
ZERO:	Provides adjustment of the meter scale deflection to the "Zero" level.

2.2. REAR PANEL

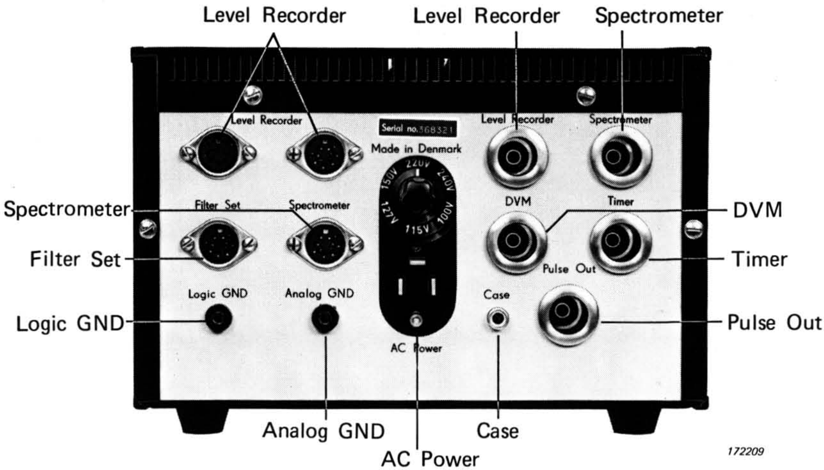


Fig.2.2. Rear Panel

AC POWER SOCKET: For the connection of the AC mains supply.

VOLTAGE SELECTOR: To select the correct AC mains supply remove the central fuse holder and turn the selector with a wide bladed screwdriver or a small coin.

REMOTE CONTROL SOCKETS:

LEVEL RECORDER: 8 pole DIN socket for connection to Level Recorder Remote Control socket (see also Fig.4.4).

7 pole DIN socket for connection to Level Recorder Two-Way Channel Selector socket (see also Fig.4.4).

SPECTROMETER: 7 pole DIN socket for connection to Frequency Spectrometer REMOTE CONTROL socket (see also Fig.4.4).

ANALOG GND:
LOGIC GND:
CASE:

As both signal and remote control connections to the Level Recorder are made, the possibility of ground loops may arise. In the event of noise problems, the signal cables should be grounded to ANALOG GND or CASE and control cables grounded to LOGIC GND.

FILTER SET: 7 pole DIN socket for connection to Band Pass Filter Set Remote Control Socket (see also Fig.4.4).

SIGNAL SOCKETS:

LEVEL RECORDER: For output of the squared and integrated reverberation decay curve to the Level Recorder.

SPECTROMETER: For input of the received enclosure response from the Frequency Spectrometer.

PULSE OUT: For output of rectangular pulse to the Band Pass Filter Set.

DVM: A socket for direct measurement of the Early Delay Time on a digital voltmeter. A DC voltage between 0 and + 10 V is emitted whose level is proportional to the Early Decay Time, 1 V being equivalent to 1 s.

TIMER: Emits a pulse of + 5 V amplitude, the width of which is proportional to the Early Decay Time, which can be calculated from the following table.

EDT range		EDT
Range 1	−1 to −10 dB	6.67 x pulse width
Range 2	−1 to −15 dB	4.29 x pulse width

3. OPERATION

As the Reverberation Processor was primarily designed for the investigation of the reverberation characteristics of rooms and enclosures, this application will be described in full. Further applications may be found in Chapter 5. The measuring arrangement for the recording of reverberation decay curves with automatic Level Recorder control using the Reverberation Processor and associated Brüel & Kjær instrumentation is shown in Fig.3.1.

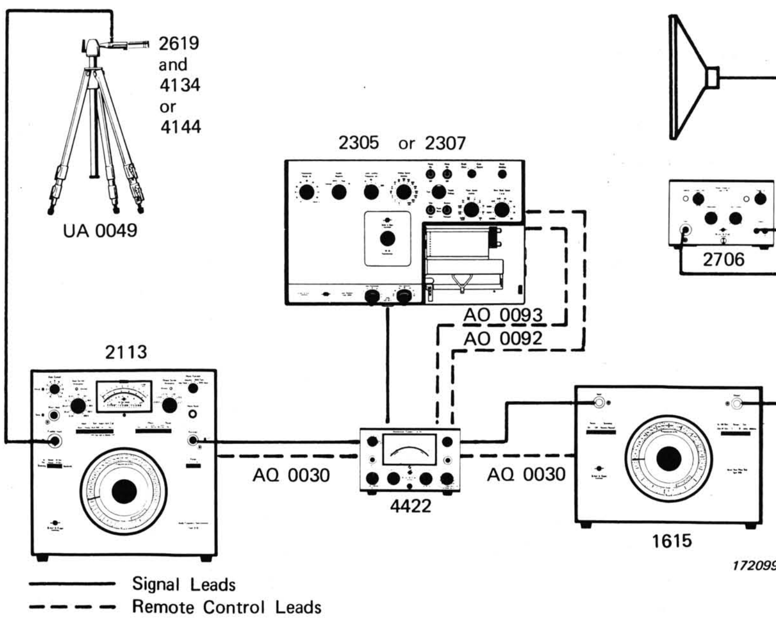


Fig.3.1. Measuring Arrangement for Reverberation Investigations

Initial Settings:

Reverberation Processor Type 4422

FUNCTION	"Cal"
PULSE INTERVAL	"2.5 sec."
PULSE AMPLITUDE	Max. clockwise
OUTPUT GAIN	Max. clockwise

Frequency Spectrometer Type 2113

POWER	"On"
INPUT	"Preamp."
FILTERS	"Int."
SCANNING	"Manual"
FILTER	"1/3 octave"
RECORDER	"AC"
GAIN CONTROL	"9"
INPUT SECTION ATTENUATOR	"1 V"
OUTPUT SECTION ATTENUATOR	"x 1"

Level Recorder Type 2305

POWER	"On"
MOTOR	"On"
POTENTIOMETER RANGE	"50 dB"
RECTIFIER RESPONSE	"DC"
LOWER LIMITING FREQUENCY	"50 Hz"
WRITING SPEED	"500 mm/sec." (for 100 mm paper) or "250 mm/sec." (for 50 mm paper)
SINGLE CHART/CONT. RECORD	"Cont. Record"
STOP/START	"Start"
REVERSE/FORWARD	"Forward"
PAPER SPEED	"100 mm/sec." (gear button in)
DRIVE SHAFT SPEED	"12 rpm"
INPUT POTENTIOMETER	"0"
INPUT ATTENUATOR	"0 dB"

Power Amplifier Type 2706

POWER	"On"

CURRENT LIMIT	"1.8 A RMS"
ATTENUATOR	"40 dB"
GAIN CONTROL	Max. clockwise

Band Pass Filter Set Type 1614

POWER	"On"
SCANNING	"Manual"
1/3 OCTAVE – 1/1 OCTAVE	"1/3 Octave"
LIN 1.8 Hz – 200 kHz	Out

The following procedure assumes the use of a 50 dB potentiometer in the Level Recorder.

The OUT–IN CONTROL for the Two Channel Selector on the Level Recorder should be rotated with a screwdriver until the notched portion is uppermost, and the pen lowered. It may be necessary to adjust the cam on the Two Channel Selector to achieve correct synchronization of paper feed and pen lowering. Dummy runs for this purpose can be initiated by pressing the RUN button on the Reverberation Processor.

The EQUIV. OCTAVE BAND knob on the Reverberation Processor should now be switched to the required frequency. The Frequency Spectrometer and Band Pass Filter Set should be set to one of the three third octave bands in the octave band setting of the Reverberation Processor. The measurement set-up is now ready for the calibration procedure.

3.1. CALIBRATION PROCEDURE

This procedure ensures that the Reverberation Processor is operating in its optimum state, and must be performed prior to every run at a new frequency.

1. Adjust ZERO knob to obtain a meter scale deflection within the "zero" limits.

2. The Power Amplifier must now be adjusted to give the maximum power output to the loudspeaker.

   It is advisable at this stage to ascertain the loudspeaker resistance and maximum power rating. Fig.3.2 shows the variation of the maximum output power available from the Power Amplifier with loudspeaker resistance.

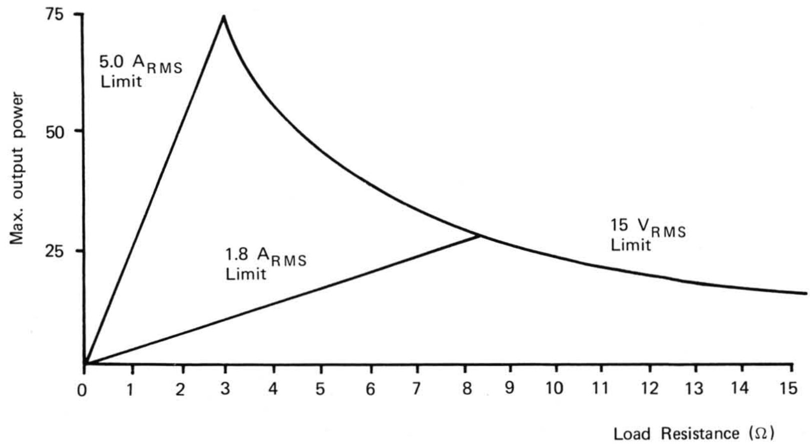


Fig.3.2. Variation of the Maximum Output Power of the Power Amplifier Type 2706 with Load Resistance

The loudspeaker should be capable of withstanding 100% overload without damage for impulsive excitation. If the power for such an overload is less than that output by the Power Amplifier for a "1.8 A_{RMS}" current limit, then the power to the loudspeaker should be suitably monitored and restricted. If the loudspeaker is rated at a higher power than that available with a "1.8 A_{RMS}" current limit, then the CURRENT LIMIT SWITCH should be changed to "5.0 A_{RMS}". The following instructions should only be followed for loudspeakers which can withstand the power available.

Press PULSE CAL button, and observe clipping light on the Power Amplifier. If this does not come on, decrease the ATTENUATOR by 10 dB, and repress PULSE CAL button and repeat the process until the clipping light does come on. Again press PULSE CAL button and reduce the GAIN CONTROL setting on the Power Amplifier until the clipping light just fails to come on.

3. Press the PULSE CAL button and hold depressed to operate the peak hold circuit. Note the meter scale deflection. Release PULSE CAL button. If there is no deflection, or the deflection is small, turn the INPUT SECTION ATTENUATOR control on the Frequency Spectrometer anticlockwise one position. Allow the needle

to reach a steady level, and re-press PULSE CAL button. If the INPUT SECTION ATTENUATOR overload light comes on, increase the INPUT SECTION ATTENUATOR control by one position and decrease the OUTPUT SECTION ATTENUATOR by one position. Repeat this procedure until a full scale deflection on the Reverberation Processor is obtained. Turn the GAIN CONTROL on the Frequency Spectrometer anti-clockwise to obtain a meter scale deflection on the Reverberation Processor within the "Cal" limits on the scale. This final adjustment could be performed using the PULSE AMPL. control on the Reverberation Processor, but this would result in a smaller pulse being emitted into the room, and possibly a poor signal to noise ratio.

4. Press the OUTPUT GAIN knob, and hold depressed to operate the peak hold circuit. Note the meter scale deflection, release knob, and adjust OUTPUT GAIN setting, if necessary. Allow the needle to reach a steady deflection, re-press the OUTPUT GAIN knob, and repeat until a deflection within the "Cal" limits is obtained. If such a deflection cannot be obtained, the instrument will still operate, but the results may be affected by poor signal to noise ratio. In this case, a better initial Level Recorder deflection can be obtained by turning the INPUT POTENTIOMETER clockwise.

3.2. MEASUREMENT PROCEDURE

1. If automatic indexing of the Frequency Spectrometer and the Band Pass Filter Set to the next third octave band after each measurement run is required, switch the SCANNING push buttons on both instruments to "Remote". The EQUIV. OCTAVE BAND on the Reverberation Processor should be reset manually to the next frequency setting whenever the Frequency Spectrometer and Band Pass Filter Set move to the next octave band.

2. Check the "Zero" deflection on the Reverberation Processor and adjust if necessary. If the needle moves erratically, then a poor signal to noise ratio may affect the results. Wait until the needle reaches a steady deflection, and press the RUN button. The Level Recorder will be automatically started, the pen lowered to record the decay curve squared for the second measurement pulse, and the paper feed stopped after two chart lengths.

3. A run can be started remotely by connecting pin 2 to pin 5 in the FILTER SET 7 pole DIN socket, see Fig.4.4.

4. If long reverberation times are being measured, the PAPER SPEED and DRIVE SHAFT SPEED on the Level Recorder could be decreased by one setting. Paper can be saved by one of three methods:– having the DRIVE SHAFT SPEED one setting higher than that specified, using a double notch cam (It should be checked that part of the decay curve will not be lost if this procedure is adopted) or using Paper loops. The Level Recorder Instruction Manual should be consulted for further information.

3.3. ANALYSIS OF DECAY CURVES

1. The Reverberation Time can be obtained from the Decay Curve by two methods, first by calculating the time for a certain dB drop of the curve, and second by using the Reverberation Protractor Type SC 2361. Both methods are facilitated by drawing the best straight line through the curves, ignoring the lower part of the curve if noise is present (see Section 4.5).

Fig.3.3 shows a typical curve with the necessary construction for the determination of the Reverberation Time by the first method.

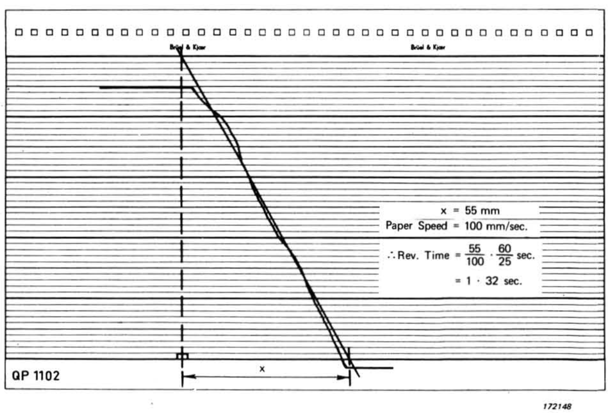


Fig.3.3. Analysis of Reverberation Decay Curves

The distance "x" should be measured in millimetres. The Reverberation Time is then given by the formula

$$\text{Rev. Time} = \frac{x}{\text{Paper Speed}} \cdot \frac{60}{25} \text{ sec.}$$

Use of the Reverberation Protractor is illustrated in Fig.3.4.

Select the required quadrant (either 50 dB, 10 mm/sec or 50 dB, 30 mm/sec) and place the heavy line to the left of the suitable quadrant on the best straight line. It will be found convenient to place the centre of the Protractor on a heavy horizontal line on the paper. The correct number should be interpolated from the scale markings on the Protractor. For 50 mm paper, this number should be doubled (because of squaring of the decay curve) to give the correct Reverberation Time and for 100 mm paper it should be multiplied by four. This is necessary because the Reverberation Protractor was designed for RMS decay curves recorded on 50 mm paper. Factors of 10 may also have to be introduced to adjust for paper speeds other than those given on the Protractor.

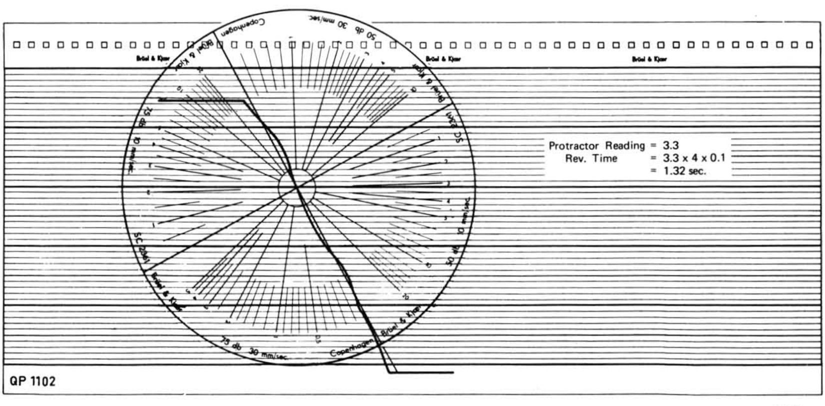


Fig.3.4. Use of the Reverberation Protractor

3.4. READING OF EARLY DECAY TIME

The required Early Decay Time range should first be selected. Range I is from −1 dB to −10 dB (which is the setting on instruments leaving the

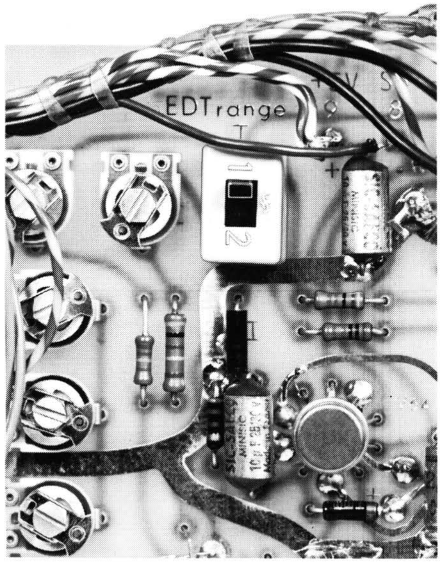


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Fig.3.5. The EDT Range Switch

factory) and Range 2 is from −1 dB to −15 dB. The EDT Range switch is found by removing the right hand panel as viewed from the front of the instrument, unscrewing the unpainted screws holding the EDT printed circuit board, and swinging the board out. The switch is shown in Fig.3.5.

The switch "up" position gives EDT range 1 (−1 to −10 dB) and "down" gives EDT range 2 (−1 to −15 dB). The Early Decay Time can be read directly from the Reverberation Processor by turning the FUNCTION knob to EDT METER RANGE and selecting the range which gives a suitable deflection.

The Early Decay Time can also be read externally by two methods:

1. Digital Voltmeter — connect the Digital Voltmeter to the socket marked DVM. A DC voltage between 0 and 10 V will be produced, 1 V being equivalent to an EDT of 1 s.

2. Timer Output — a pulse is available at this socket of 5 V amplitude, and width proportional to the Early Decay Time.

EDT Range 1	−1 to −10 dB	EDT = 6.67 x pulse width
EDT Range 2	−1 to −15 dB	EDT = 4.29 x pulse width

At this stage, the Early Decay Time and the setting of the PULSE INTERVAL knob should be compared. The pulse interval should be greater than, or equal to, the Early Decay Time, but as low as possible. If necessary, the PULSE INTERVAL knob should be reset, and the measurement repeated.

3.5. ADJUSTMENT OF INPUT GAIN

There is a facility in the Reverberation Processor for increasing the input gain. Such an adjustment will be necessary when using the instrument in conjunction with Sound Level Meters with a low output level (e.g. the 2209), but is unnecessary when using the Frequency Spectrometer Type 2113. The input gain control can be found by removing the left hand

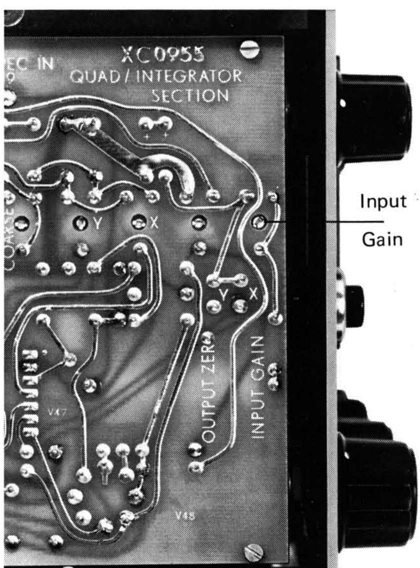

Fig.3.6. The Input Gain Control

panel of the Processor as viewed from the front of the instrument. The control is situated at the front of the printed circuit board as shown in Fig.3.6.

The input gain is set to minimum on leaving the factory, and can be increased by turning the screw clockwise.

4. DESCRIPTION

4.1. APPLICATION OF THE INTEGRATED IMPULSE METHOD

The theory of the integrated impulse method can be found in Chapter 6. The result of this theory, which forms the basis for the design of the Reverberation Processor is as follows:

$$\langle s^2 (t) \rangle = N^2 \int_o^\infty r^2 (x)\, dx - N^2 \int_o^\infty r^2 (x)\, dx$$

where $\langle s^2 (t) \rangle$ is the ensemble average of infinitely many squared reverberation curves, N is the RMS value of the impulse and r (x) the time variation of the enclosure response to unit impulse.

The measurement method as suggested by Kuttruff is shown in Fig.4.1. A suitable impulse generator emits impulses of RMS value N, which are transmitted to the enclosure. Enclosure response to the impulse N will be Nr (x) and this is received by a microphone. This signal is squared, giving $N^2 r^2 (x)$, and integrated giving $N^2 \int r^2 (x)\, dx$. Now the impulse generator emits two identical impulses. The enclosure response to the first impulse is integrated completely, this giving the first integral $N^2 \int_o^\infty r^2 (x)\, dx$ in the above equation.

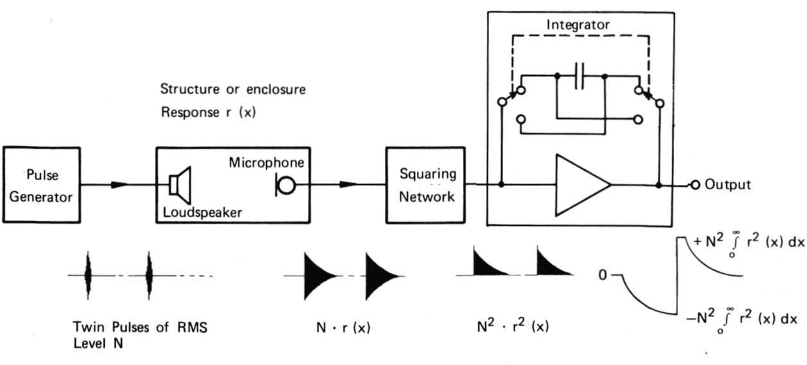

Fig.4.1. Application of the Integrated Impulse Method

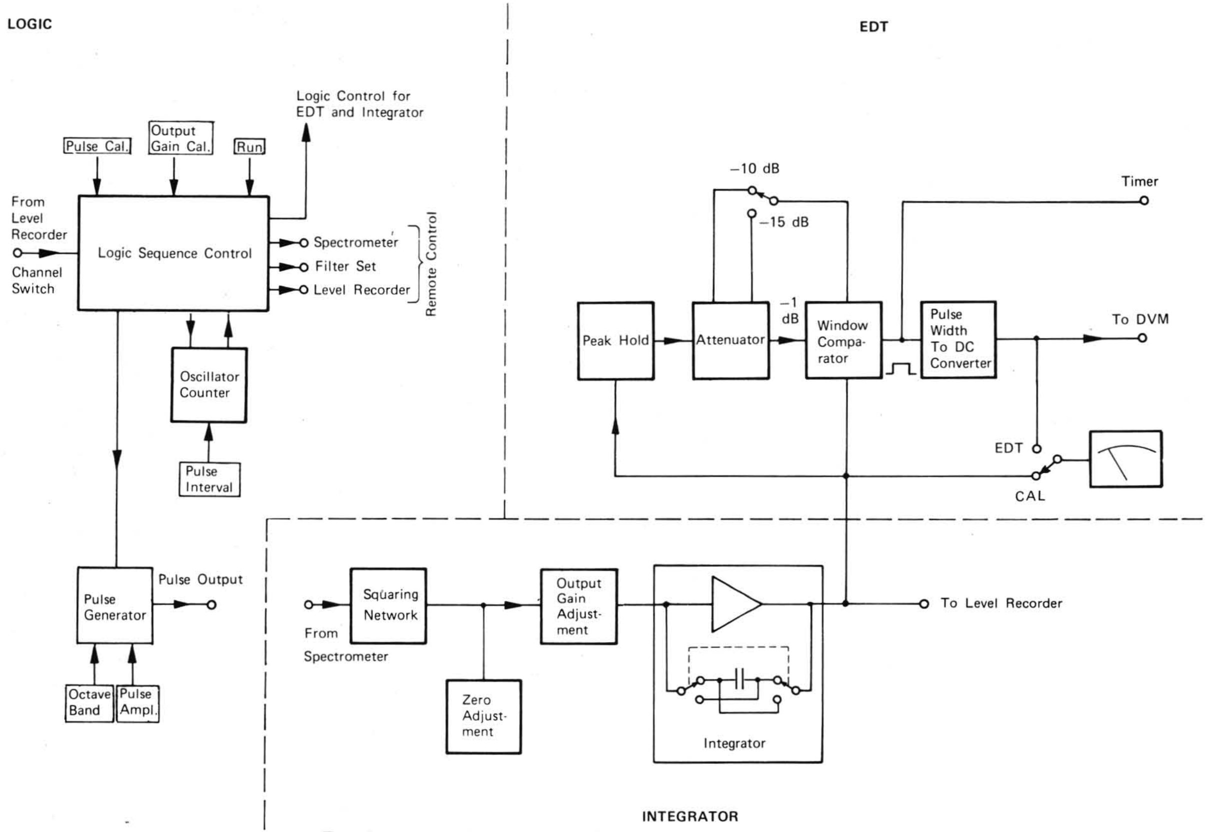

Fig.4.2. Block diagram of the Reverberation Processor

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This integral is stored as a negative voltage on the integrator feedback capacitor. The capacitor is reversed, and the integral is set as a positive initial condition for the integration of the second impulse. The integrator then subtracts the progressing integral $N^2 \int_0^t r^2(x)\,dx$ from this initial value, giving the complete right hand side of the above equation, and hence the required quantity $\langle s^2(t)\rangle$. It should be stressed at this stage that Schroeder's theory is strictly true only for a broad band white noise pulse.

In actual fact the rectangular pulse emitted by the Reverberation Processor is third octave or whole octave filtered, as is standard practice in reverberation measurements. The effect of this filtering is discussed in section 4.5.

A block diagram of the Reverberation Processor Type 4422 is shown in Fig.4.2. The diagram is divided into the following parts:

Integrator Section
Early Decay Time Section
Logic Section

4.2. INTEGRATOR SECTION

The twin pulses returned to the Reverberation Processor are first fed into a squaring network. Squaring is achieved by feeding the signal to both input terminals of a solid-state multiplier. After squaring, the signal is fed to the zero adjustment circuitry which allows the needle deflection to be brought within the "zero" limits on the meter scale, so that the background noise level can be seen by observing the needle deflection. If the needle moves erratically after zeroing, then the signal to noise ratio will be poor.

After zero adjustment the signal is fed into the output gain attenuator. This provides variation of the output gain to give a suitable initial deflection on the Level Recorder.

The signal is now integrated by means of a high gain operational amplifier with capacitor feedback. As such integrators tend to suffer from drift, due to integration of extraneous noise over long periods of time, some method of zero compensation is required. When the RUN button is pressed the signal level at that time is taken as the average zero level, and compensated for during the remainder of the measurement. It is clear that erratic needle movement could easily result in a false zero level being set as the initial condition on pressing the RUN button. It is for this reason that it is suggested in section 3.2 that the needle is allowed to reach a steady deflection before initiating a run.

The integrator capacitor is provided with switching to reverse the voltage across the integrator between the twin pulses.

The switching is achieved electronically by logic circuitry. The output from the integrator is fed to the output socket for the Level Recorder, and also to the Early Decay Time section.

4.3. EDT SECTION

The squared integrated room response to the second impulse is fed to a window comparator and also to a peak hold circuit which stores the initial level of the decay curve after reversal of the integrator capacitor. This peak value is fed into an attenuator which transmits levels −1 dB, −10 dB and −15 dB down on the peak value (referenced to input signal levels, not squared levels). The −10 dB and −15 dB levels are fed to the EDT range switch, which allows selection of either the −10 dB or −15 dB level for the determination of the Early Decay Time. The selected level is passed to the Window Comparator along with the −1 dB level. The Comparator compares the decaying signal level first with the −1 dB level, and switches the + 5 V amplitude timer pulse on. When the decaying signal reduces to the −10 dB (or −15 dB) level, the comparator switches the timer pulse off. The width of the timer pulse is equal to the time to decay through either 9 dB or 14 dB depending upon which position of the EDT range switch is selected. The width of the pulse is thus directly proportional to the EDT.

The pulse from the Window Comparator is passed to a pulse width to DC converter which feeds a DC voltage to the DVM output socket. This DC level is directly proportional to the Early Decay Time, 1 V being equivalent to 1 s. The DC level is also applied to the meter when the FUNCTION knob is switched to one of the EDT meter range positions, and is automatically stored until the next measurement run.

4.4. LOGIC SECTION

The logic section acts as a central control unit for the instrument, and also for the Level Recorder, Spectrometer and Band-Pass Filter Set. Fig.4.3 shows a time record of the functions associated with the instrument control, for a PULSE INTERVAL setting of "10 sec".

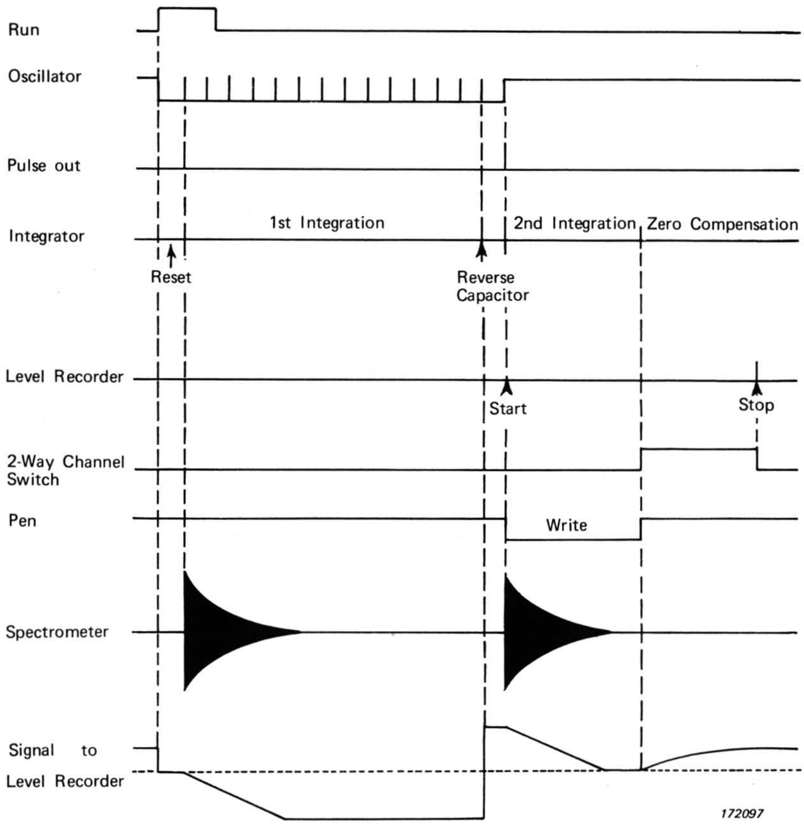

Fig.4.3. Instrument Control Time Record

On pressing the "RUN" button, the oscillator counter commences to output timing pulses of approximately 0.8 s period. The first timing pulse initiates the resetting of the integrator, and the second results in the emission from the pulse generator of the first measurement pulse. The width of this pulse is determined by the setting of the EQUIV. OCTAVE BAND knob setting. The period of the pulse is approximately 1/3 of the period of a pure tone at the EQUIV. OCTAVE BAND knob setting.

The amplitude of the rectangular pulse is determined by the position of the PULSE AMPLITUDE knob, being continuously variable from 0 to 5 V.

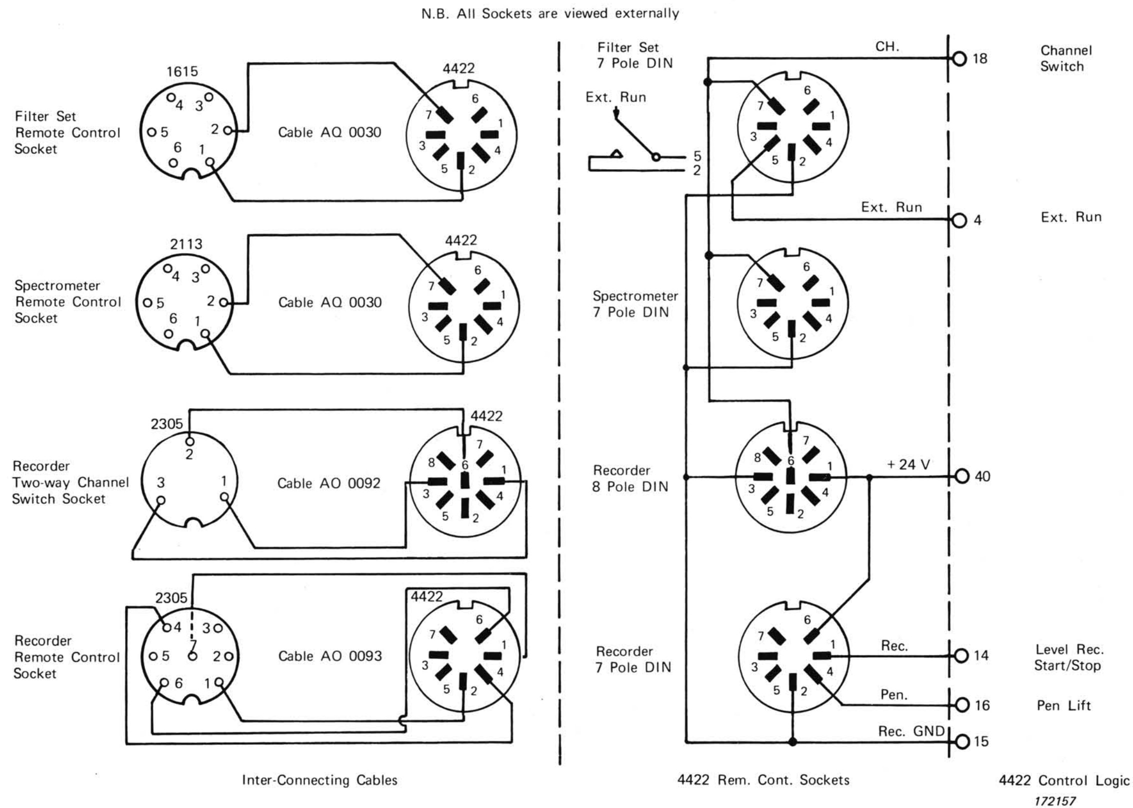

Fig.4.4. Remote Control Connections for Level Recorder, Frequency Spectrometer and Band Pass Filter Set

The remaining timing pulses after the measurement pulse are counted until the required number to approximately give the PULSE INTERVAL knob setting have been emitted. On the last but one pulse the integrator capacitor is reversed, and on the last timing pulse the second measurement pulse is emitted. The Level Recorder is also started and the pen lowered, allowing recording of the decay curve. The two-channel selector on the Level Recorder is then used first to raise the pen, and then to stop the Recorder after two chart lengths (500 mm) of paper have been run out. At this stage a switching pulse is output to the Spectrometer and Band Pass Filter Set for automatic switching to the next third octave band. The connections for operation of the Level Recorder, Spectrometer and Filter Set are shown in Fig.4.4.

The integrator is then returned to its zero compensation mode. The Processor will not allow another run to be started until the Level Recorder has stopped.

4.5. INSTRUMENT ACCURACY

It has been stated earlier in this chapter that Schroeder's theory is strictly applicable only to excitation by a broad band white noise pulse. It is, however, the usual practice in reverberation measurements, including those using Schroeder's method that the excitation is third octave filtered. Schroeder atempted to overcome this difficulty by including the filter response in his lumped response term r(x). This manipulation only postpones the problem, as the enclosure or structural response and the filter response must be separated at a later stage. However, as long as the pulse decay time is much less than the enclosure or structural decay time, then the response obtained will be correct, and independent of the filter response.

This last statement implies a restriction on the minimum reverberation time which the Reverberation Processor can measure accurately. As a first step in determining this minimum reverberation time at a given frequency, the pulse decay time was investigated for third and whole octave filtering. Fig.4.5 shows pulses filtered by third and whole octave filters of centre frequency 31.5 Hz. Also shown is the pulse after squaring and integrating. The pulse decay times (for 0 to −60 dB decay) were 0.55 sec and 0.40 sec for single third octave and octave filtering respectively. This preliminary investigation enabled the variation of pulse decay time with frequency to be drawn on Figs.4.6 and 4.7.

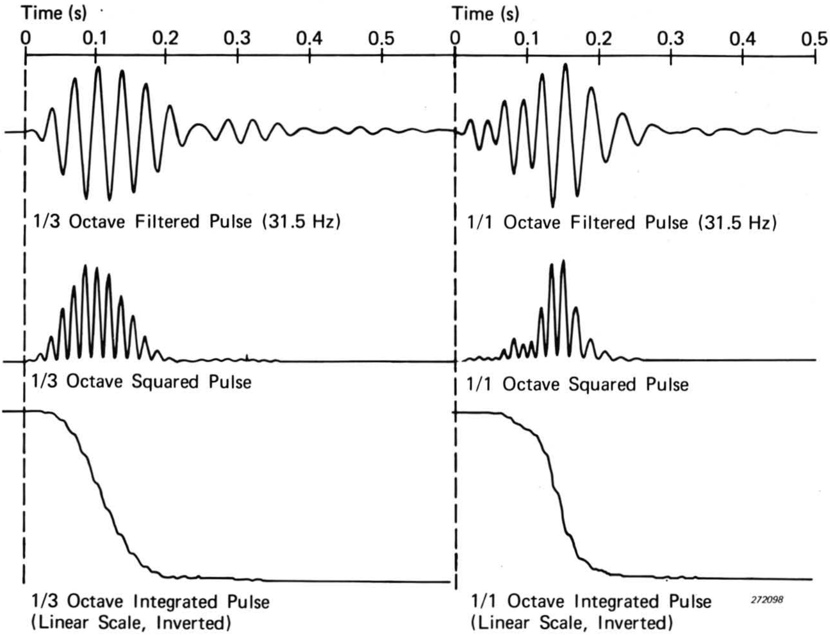

Fig.4.5. Third Octave and Whole Octave Filtered Pulses

It was then determined by how much the enclosure or structural decay time must exceed the pulse decay time at a given frequency to attain a certain accuracy. The accuracy requirement was stated as ± 3%. It was also decided to take "worst case" conditions for this determination. For this reason the Early Decay Time measurement was considered, as this would give less accurate results than those averaged by eye over an entire decay curve and also as the EDT circuitry can measure reverberation times less than 0.25 seconds which is the lowest the Level Recorder can follow with settings in the stable range. As practical reverberation decay curves were very difficult to obtain in this region, a decay curve was built up to contain an initial direct sound energy decay followed by several succesive reflected energy decays. Such a curve will be obtained from reverberation time measurements at low frequencies with repeated sound reflection between parallel walls. Again this is a worst case situation. More diffuse sound fields would produce decay curves which are closer to true exponential decay.

From the decay curve it was found that a ± 3% error would result when the EDT circuitry found the −10 dB point to be at the end of the third

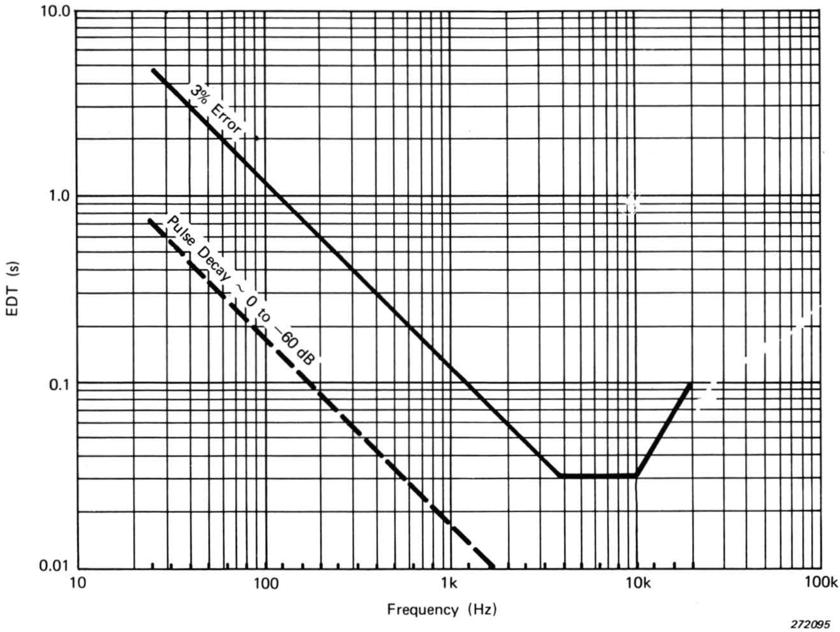

Fig.4.6. Lower Limit of Early Decay Time for ± 3% Error – Third Octave Filtering

reflected wave. By a consideration of the number of cycles occuring up to this −10 dB point, a frequency and time scale was assigned to the decay curve, allowing the 3% error lines to be drawn on Fig.4.6 for third octave filtering and Fig.4.7 for octave filtering.

The remainder of the error curves at the high frequency end were obtained by testing the Reverberation Processor with an electronic decay circuit. The decrease in accuracy at the high frequency end is due to failure of the integrating capacitor to follow the initial part of the decay curve.

A further source of error, common to all methods of reverberation measurement is the presence of noise in the enclosure. A poor signal to noise ratio tends to have a worse effect on results obtained by the Schroeder – Kuttruff Method because of the necessity to integrate between the twin pulses. It is for this reason that the pulse interval must be reduced as much as possible without cutting off the last portion of the enclosure

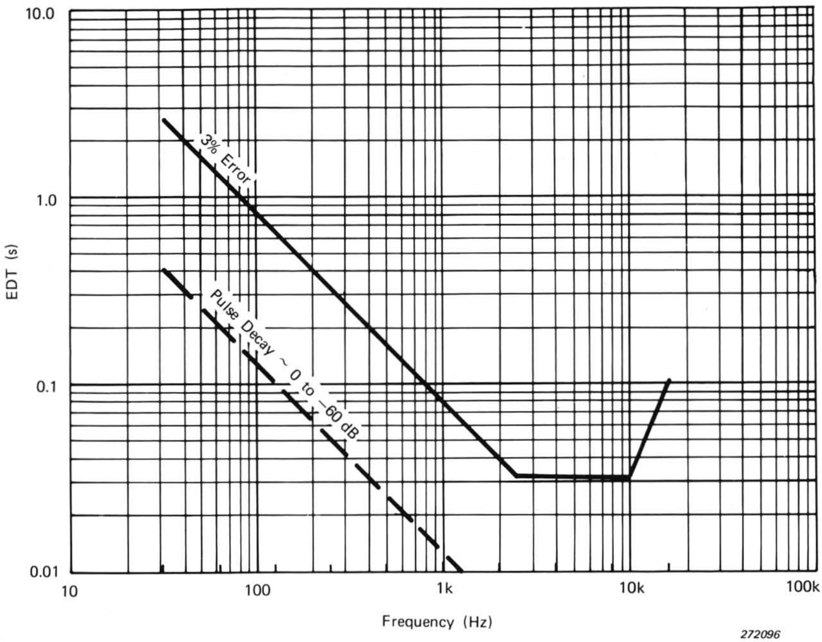

Fig.4.7. Lower Limit of Early Decay Time for ±3% Error — Octave Filtering

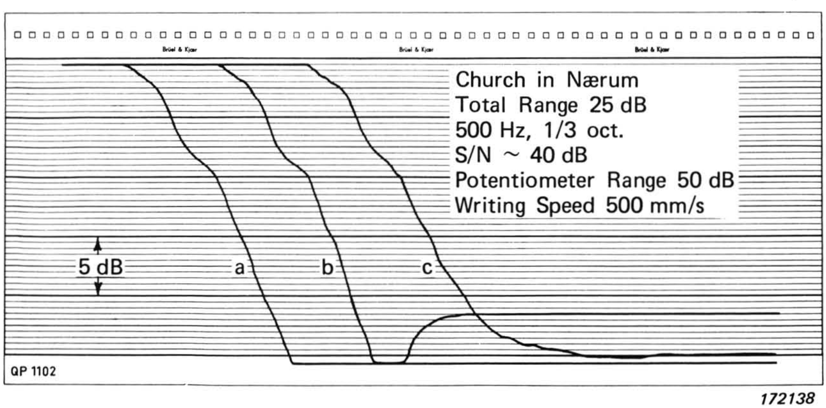

Fig.4.8. Typical Decay Curves with Poor Signal to Noise Ratio

response to the first pulse. Typical curves recorded in the presence of noise are shown in Fig.4.8.

Decay curve (a) is unaffected by noise. Curve (b) shows the effect of noise between the two pulses (a similar effect can result from pulses of different magnitude being transmitted to the room) and curve (c) indicates the effect of excess noise during the response to the second pulse. Curves of this nature can be predicted by erratic movement of the needle after zeroing. It can be seen that only about 20 dB of the decay curve remains unaffected, and thus as a general rule the signal to noise ratio should be approximately twice the dynamic range required. When analysing curves affected by noise, the lower portions should be neglected.

5. FURTHER APPLICATIONS

5.1. SEMI-PORTABLE MEASUREMENT OF REVERBERATION TIME

Fig.5.1 shows the measurement set up for an alternative method of obtaining the Early Decay Time using semi-portable instrumentation.

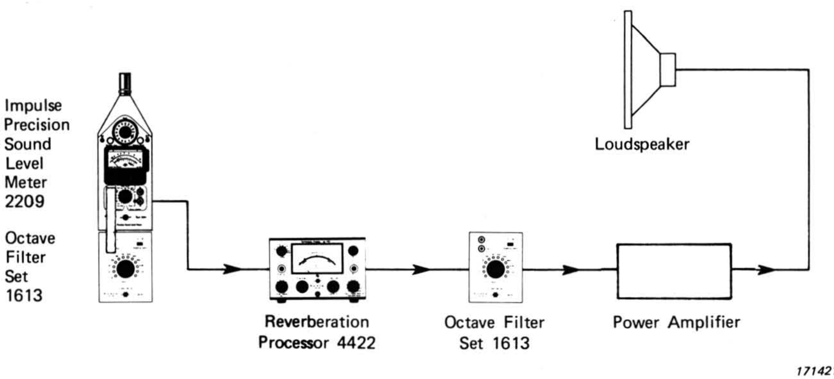

Fig.5.1. Set-up for the Semi-Portable Measurement of Reverberation Time

Pulses from the Reverberation Processor are filtered by a portable Octave Filter Set Type 1613, amplified by a Power Amplifier (which must have an input impedance greater than 150 kΩ) and fed to a loudspeaker. The room or enclosure response is registered by an Impulse Precision Sound Level Meter Type 2209 with associated Band Pass Filter Set Type 1613. The output from the Sound Level Meter is returned to the Reverberation Processor.

As the Sound Level Meter is a low output device the Input Gain of the Reverberation Processor must be increased. The procedure for this adjustment can be found in section 3.5. The calibration and measurement procedures for this set up closely resemble those already described in section 3.

5.2. MEASUREMENT OF REFLECTION COEFFICIENT (ρ)

Fig.5.2 shows the set up for the measurement of the Reflection Coefficient (ρ) using the Reverberation Processor.

The instrumentation is very similar to that described in section 3, except that the Level Recorder has been replaced by a Storage Oscilloscope.

The loudspeaker, microphone and reflector under investigation are first placed as in position 1. The distance 2r is accurately measured.

The Reverberation Processor is calibrated as in section 3. The RUN button is then pressed, and the resulting sound decay recorded on the

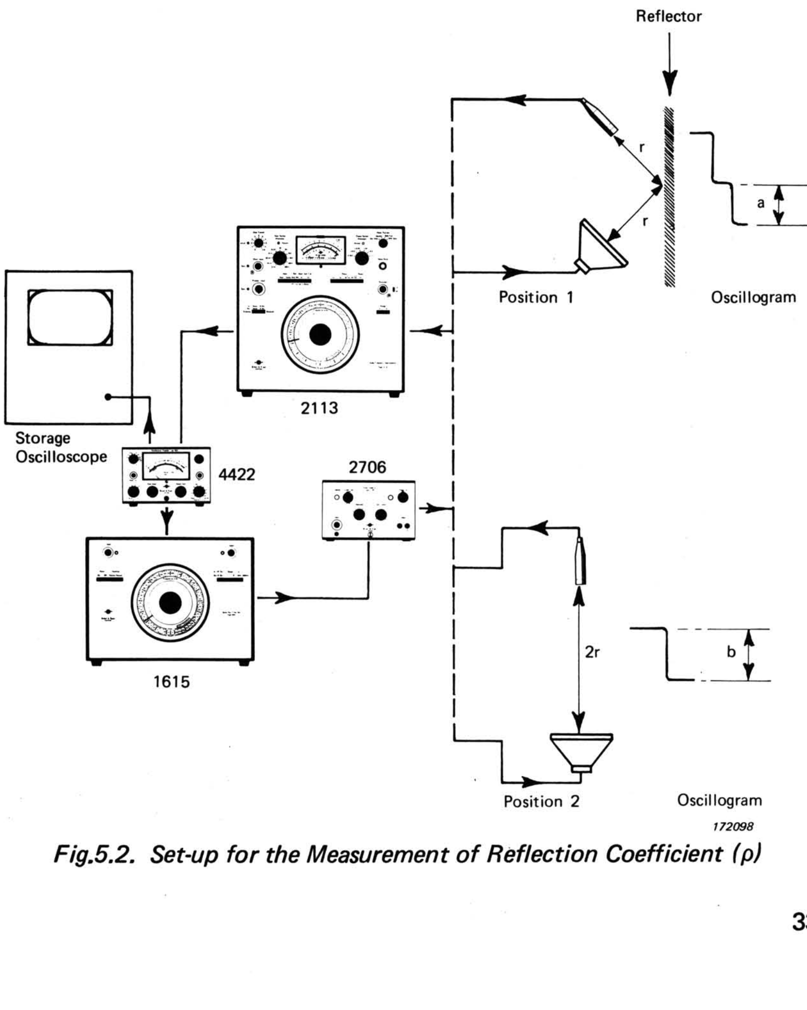

Fig.5.2. Set-up for the Measurement of Reflection Coefficient (ρ)

Storage Oscilloscope. The initial part of the curve obtained represents the energy received by the microphone directly from the loudspeaker, and the second part represents the energy reflected by the reflector. The distance "a" on the oscillograph corresponds to the reflected energy which should be measured and recorded.

The reflector is then removed, and the microphone and speaker placed as in position 2, with the same distance 2r between them. The previous settings should not be altered. The RUN button is again pressed, and the sound decay again recorded on the Storage Oscilloscope. The distance "b" which represents the energy received by the microphone should be measured and recorded. The Reflection Coefficient is then given by:

$$\rho = \frac{a}{b}$$

The Absorption Coefficient (α) may also be calculated for the same angle of incidence using the expression

$$\alpha = 1 - \rho$$

5.3. MEASUREMENT OF THE QUALITY FACTOR (Q) OF STRUCTURES

It can be shown that the Quality Factor (Q) equal to the reciprocal of the loss Factor (d) is given by

$$Q = \frac{Tf_r}{2.2}$$

where T = the reverberation time of the structure in seconds and f_r = the resonant frequency of the structure in Hz. Thus if the resonant frequency of a structure is known, it can be impulsively excited by the Reverberation Processor at this frequency to determine its reverberation time, allowing Q to be calculated directly. The measurement set up for this determination is shown in Fig.5.3.

The correct sized Power Amplifier and Vibration Exciter System should be chosen to give suitable excitation of the structure under investigation. The Accelerometer should also be chosen to avoid undue loading of the structure. For further information as regards the correct choice of these

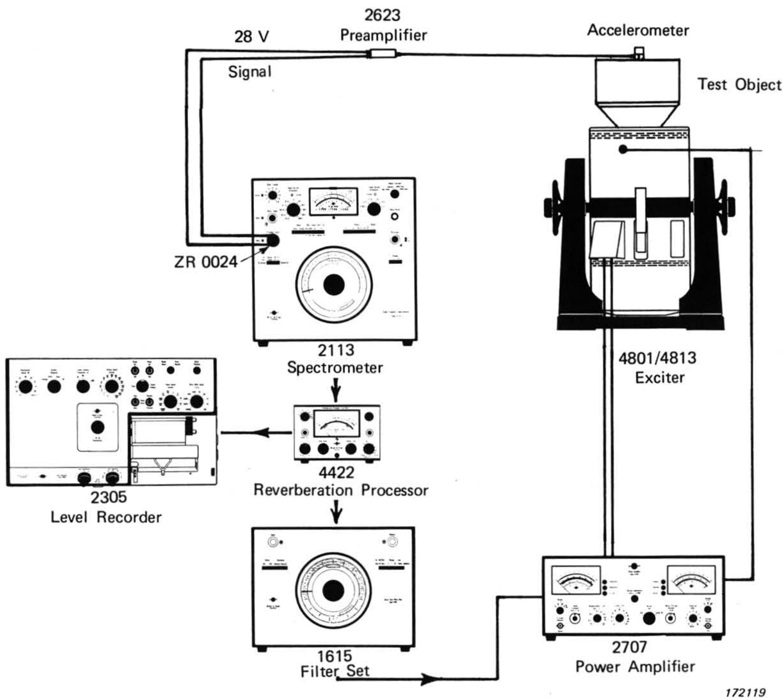

Fig.5.3. Set-up for the Measurement of Quality Factor (Q) of Structures

components, the relevant Instruction Manuals should be consulted. The measurement procedure is very similar to that outlined in section 3. Further information on this method can be found in Brüel & Kjær Technical Review 1957 No. 4 page 5.

6. THE THEORY OF THE INTEGRATED IMPULSE METHOD

The theory outlined below is that originally presented by M.R. Schroeder in J.A.S.A. Vol. 37, 1965 Numbers 1–6 Page 409.

First, the following quantities are defined with reference to Fig.6.1.

$n(\tau)$ The noise transmitted to the enclosure. The noise is assumed to have been switched on at time $-\infty$, allowed to reach a steady level, and switched off at time 0.

$r(t-\tau)$ The response of the enclosure (and associated filters, amplifiers and transducers in the measuring system) at time t to a unit impulse emitted at time τ.

$s(t)$ The total signal received at time t by a transducer in the enclosure.

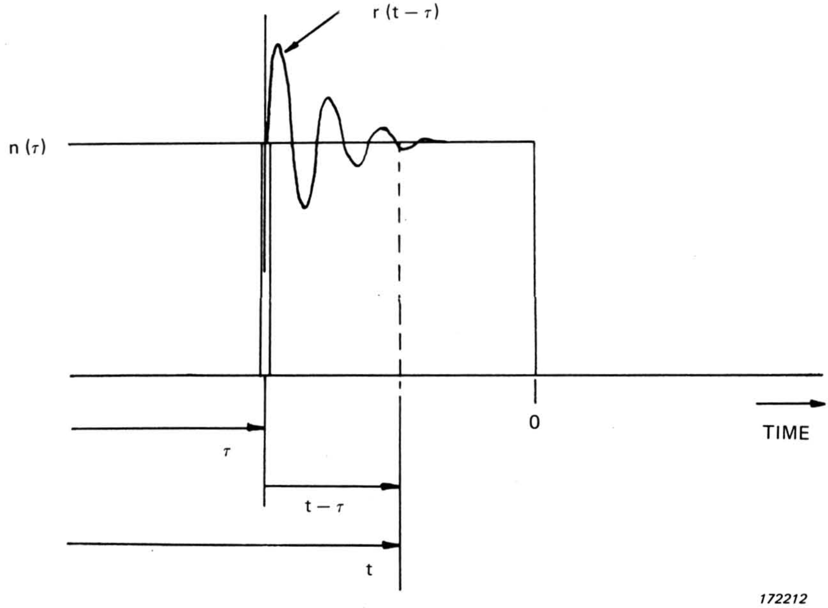

Fig.6.1. Definition of Quantities and Time Scales in the Integrated Impulse Theory

The signal received at time t due to a single impulse n at time τ will be n.r $(t-\tau)$. Now the noise transmitted to the enclosure, $n(\tau)$, can be thought of as being a series of impulses n, and hence the total signal received is given by:

$$s(t) = \int_{\infty}^{t} n(\tau)\, r(t-\tau)\, d\tau$$

It can be seen that the upper limit t may be taken as 0 as no further signal is received after that time

$$s(t) = \int_{\infty}^{0} n(\tau)\, r\,(t-\tau)\, d\tau \qquad (1)$$

By considering the enclosure as a linear network, and squaring equation (1)

$$s^2\,(t) = \int_{\infty}^{0}\int_{\infty}^{0} n(\tau)\; n\;(\theta)\; r\,(t-\tau)\; r\,(t-\theta)\; d\tau\, d\theta \qquad (2)$$

where θ is a further time scale.

By averaging equation (2) over many noise signals, we obtain

$$\langle s^2\,(t) \rangle = \int_{\infty}^{0}\int_{\infty}^{0} \langle n(\tau)\; n(\theta) \rangle\; r(t-\tau)\; r(t-\theta)\; d\tau\, d\theta \qquad (3)$$

Now, the auto-covariance function of the noise signal $n(\tau)$ may be written as

$$\langle n(\tau)\; n(\theta) \rangle$$

using the two time scales τ and θ as above. If we now specify the noise signal n(t) to be stationary white noise, two simplifications can be made, First, the specification of stationarity implies that the auto-covariance function depends only on the time delay $\theta - \tau$. Secondly, the specification of whiteness implies that the auto-covariance function is zero everywhere, except when $\theta = \tau$.

Thus the auto-covariance function may be written as

$$\langle n(\tau)\; n(\theta) \rangle = N^2\, \delta\; (\theta-\tau) \qquad (4)$$

where N is the RMS value of the signal $n(\tau)$ and $\delta(\theta-\tau)$ is the Dirac δ function. Substituting for $\langle n(\tau)\, n(\theta)\rangle$ from equation 4 into equation 3 gives

$$\langle s^2(t)\rangle = \int_{\infty}^{o}\int_{\infty}^{o} N^2\, \delta(\theta-\tau)\, r(t-\tau)\, r(t-\theta)\, d\tau\, d\theta$$

But as $\delta(\theta-\tau)$ is zero except when $\theta = \tau$

$$\langle s^2(t)\rangle = N^2 \int_{-\infty}^{o} r^2(t-\tau)\, d\tau$$

or substituting a new integration variable $x = t-\tau$

$$\langle s^2(t)\rangle = -N^2 \int_{\infty}^{t} r^2(x)\, dx \qquad (5)$$

and finally, reversing the limits of the integration to dispose of the minus sign

$$\langle s^2(t)\rangle = N^2 \int_{t}^{\infty} r^2(x)\, dx \qquad (6)$$

This is the result obtained by Schroeder. It may be stated in words as follows:

The average of **infinitely many** squared reverberation decay curves may be obtained from the squared and integrated response of an enclosure to a **single** impulse.

The only problem with the result is the lower limit t of the integration. Schroeder overcame this problem by tape recording the decay curve, and playing it back in reverse, thus effectively using Equation (5) with t as the upper limit of integration. This process makes analogue integration far simpler, but makes for a time-consuming process.

A practical solution was proposed by H. Kuttruff and M.J. Jusofie in Acustica, Vol. 19, 1967/68 Page 56

$$\langle s^2(t) \rangle = N^2 \int_0^\infty r^2(x)\,dx - N^2 \int_0^t r^2(x)\,dx \qquad (7)$$

Equation 7 forms the complete basis of the design of the Reverberation Processor. The practical application of the theory is discussed in section 4.1.

7. SPECIFICATIONS

Measurement Range:	Approx. 0.25 to 10 s Reverberation Time using Level Recorder Type 2305. Approx. 0,2 to 10 s Early Decay Time (EDT) indicated on built-in meter, measured over −1 to −10 dB or −1 to −15 dB selected by internal switch. Section 4.5 should be referred to for further information.
Frequency Range:	20 Hz to 20 kHz
Pulse Output:	Amplitude continuously variable 0 to 5 V. Pulse width variable in 10 steps corresponding to the octave bands 31.5 Hz to 16 kHz. Pulse interval nominally 1, 2.5, 5 or 10 s.
Input From Spectrometer Output:	Input impedance ⩾ 10 kΩ. Max. Level adjustable 10 to 50 V peak-to-peak.
Squaring:	30 dB input dynamic range at ± 1 dB linearity.
Level Recorder Output:	± 8.5 V max. output level nominally. Load impedance ⩾ 18 kΩ.
Digital Voltmeter (DVM) Output:	0 to + 10 V DC proportional to EDT, + 1 V DC per second EDT. Load impedance ⩾ 18 kΩ.
Timer Output:	Rectangular pulse, + 5 V amplitude, width proportional to EDT.

EDT range		EDT
Range 1	−1 to −10 dB	6.67 x pulse width
Range 2	−1 to − 15 dB	4.29 x pulse width

Load Impedance ⩾ 1 kΩ.

Remote Control:

Reverberation Processor: run initiation

Level Recorder: start/stop and pen lift.

Filter Set and Spectrometer: stepping after each measurement.

Operating Ambient Temperature: 5^{o} to 40^{o}C (41^{o} to 104^{o}F).

Power Supply: 100 V, 115 V, 127 V, 150 V, 220 V and 240 V AC
50 or 60 Hz
Approximately 15 W.

Dimensions:

Height: 13.26 cm (5.2 in)
Width: 20.95 cm (8.3 in)
Depth: 20.00 cm (7.9 in)

Cabinet: KK 0025

Weight: Approximately 5 kg. (11 lb)

Accessories Included:

2 control cables for Filter Set and Spectrometer AQ 0030
1 control cable, 8-pin DIN to 3-pin 2305 AO 0092
1 control cable, 7-pin DIN to 7-pin 2305 AO 0093
Power Cable European AN 0005
or Power Cable American AN 0006
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