"DESIGN & CONSTRUCTION OF A REVERBERATION METTER FOR THE MEASUREMENT OF REVERBERATION TIME BY DIGITAL METHOD."

A project report submitted to
The Department of Electrical Engineering,
Ahmadu Bello University,
Zaria.

as part of the requirements for the degree of

MASTER OF SCIENCE
in
ELECTRONICS & TELECOMMUNICATIONS,

BY

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February, 1970

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LIST OF SYMBOLS.

V = Volume of an enclosure

I_o = Average sound energy density at the instant the source is shut off.

I = Sound energy density at time t after starting the source.

c = Velocity of sound.

R.T. = Reverberation time in secs.

d_w = Thickness of wall

ρ_w = Density of wall

$\bar{\alpha}$ = Average absorption coefficient.

ρ_a = Density of air

Φ = Resultant sound wave function.

m = Attenuation constant of air.

X_c = Reactance of a capacitor C.

h_{21} = Transistor Input Impedance.

h_{11} = Transistor current gain.

K = Boltzmann's constant.

T = Absolute temperature.

I_s = Diode leakage current.

V_J = Voltage across an ideal diode

f = Frequency.

T.L = Transmission Loss .

D = Decay rate of sound.

e = Electronic charge.

N = Number of electrons.

q = Total charge of the electrons.

A = Total number of absorption units exposed within an enclosure.

S = Total surface area

d_w = Thickness of wall

t = Time in seconds.

SUMMARY

This thesis gives a report of the work carried out by the author on "The Measurement of the Reverberation Time by digital Method". The primary aims of the project were to build a reverberation time meter which would be suitable for measuring the reverberation time of a room and to use this to carry out reverberation time measurements in some of the rooms in the University, particularly those in the Electrical Engineering Department.

The first task was that of building the reverberation time meter. After considering the various alternative methods used previously, a new method was adopted in which reverberation time could be obtained on a digital counter. A section of the reverberation meter would therefore be responsible for starting the counter at the commencement of sound decay and the other part would be for stopping it when sound died down to a certain known level. A block Schematic diagram of the whole apparatus was constructed. Parts of the equipment required were already available but others had to be built. Among the equipment constructed were:- a microphone pre-amplifier, a variable gain amplifier, a logarithmic amplifier, a detector, a level comparator (Schmitt trigger) and also differentiating and clipping networks. The methods of design and construction of these equipment are given in Chapters 4, 5,6,7, and 8 respectively.

After completing the building of each section of the apparatus, various tests were carried out to check its successful operations. When all apparatus were ready, a final setting of the whole apparatus was made and tests were also performed to make sure that all of them connected together, worked well. All these are given in Chapter 9.

When it was clear that the reverberation time meter was working well, the measurement of the reverberation time of various rooms was undertaken. The accuracy of measurement was found to be within 6%. The first set of measurements was made in the Electrical Engineering Department in the Part III lecture room, Part II lecture room, and the Departmental Library. These rooms however, were classed as small rooms. The second set of measurements was made in larger rooms such as the University Assembly Hall, the Faculty of Engineering small Lecture Theatre and the Electrical Engineering High Voltage Laboratory. The results are given in Chapter 10.

The last chapter of the thesis contains the conclusions drawn from all the measurements and also suggestions for the improvement of the acoustic qualities of the rooms concerned.

INTRODUCTION

The need for good acoustics is of fundamental importance in order to obtain satisfactory hearing of speech and music. The acoustical behaviour of a room or an enclosure directly affects the audibility and intelligibility of sounds produced in the room. Especially in rooms intended for listening such as classrooms, theatres, auditoria, chapels and any musical studio, acoustical defects may have highly detrimental effects on the intelligibility of the sound heard by an audience, and as a result of this, the designer should be able to predict the acoustical performances of the room and arrange for the desired result before constructions.

Acoustics is mainly concerned with the transmission and reception of sound and in a more limited sense, it is concerned with those properties of auditoria and other rooms which provide good hearing conditions for audiences. The main problem involved in obtaining satisfactory acoustics is centred on the reverberation of sound. Reverberation is the prolongation of sound in an enclosure after the original source has been shut off and it is due to successive reflections of the sound energy at the boundaries of the enclosure.

Because of the need for proper control of the reverberation in a room, a standard of measure called the "reverberation time" has been established. This reverberation time is a measure of the time required for sound to die away to one-thousandth of its initial pressure or one-millionth of its initial intensity (i.e. 60 dB decay) after the source is shut off.

One of the chief requirements for optimum listening conditions in a room or auditorium is that the audience can hear clearly the sound source with sufficient intensity and in a pleasing manner, and this can generally be achieved if the reverberation time of the enclosure is of suitable value. Too long a reverberation time produces lack of clarity and too short a time results in reduction of the intensity or a "deadness" in the quality of sound.

The earliest scientific study of the acoustics of buildings started at about 1895 when Professor W.C. Sabine of Harvard University began his work on the acoustical properties of rooms and his studies were directed primarily on the influence of reverberation on the acoustics of chambers. This however, led to the correlation of the reverberation time of a room with the volume and the amount of sound absorbing materials in the room. In his method for the measurement of reverberation time of a room, he used an organ pipe for the source of sound and measured the time of duration of the sound after the source was shut off. On the assumption that the intensity of sound in a chamber dies down in a continuous manner, he derived a formula for the reverberation time as:-

$$T = \frac{0.049V}{S\bar{\alpha}}$$

Where T = Reverberation Time in seconds

V = The volume of the room.

$\bar{\alpha}$ = The average absorption coefficient of the absorption materials in the room.

S = The total surface area in square feet.

However, it was later shown by Schuster, Waetzmann and C.F. Eyring that this formula applies only to chambers with small absorption and large reverberation. The equation holds for reverberation times of the order of 0.5 second or more. On the other hand, it breaks down for times less than 0.5 second because it can no longer be assumed that the intensity of sound dies down in a continuous manner. Sabine's theory over-simplifies the actual behaviour of sound waves in an enclosure for it neglects important factors like the effects of interference, extraneous noises, echoes and possible air absorption of sound.

The first modification of Sabine's formula was made by W.S. Franklin in 1903. He assumed diffuse distribution of sound and continuous absorption in a room and derived mathematically the equations for the building up and decay of sound in a rectangular room with dimensions large compared with the wavelength for the particular frequency concerned. The equation for the building up of sound was given as

$$I = I_o\left[1 - \exp\left(-\frac{c\bar{A}t}{4V}\right)\right]$$

Where V = Volume of the enclosure

I = The sound energy density at time t after starting the source

I_o = The average sound energy density at the instant the source is stopped.

A = The total number of absorption units exposed within the room.

$= a_1S_1 + a_2S_2 + a_3S_3 + \text{----------}$

Where a_1, a_2, a_3 are the absorption coefficients of the various surfaces S_1, S_2, S_3.

$I_o = 4P_o/cA$

P_o = The rate at which sound is being generated by the constant source.

c = Velocity of sound.

After the sound source has been stopped, the energy density at the time t seconds after this period was also expressed as:-

$$I = I_o \exp\left(-\frac{cAt}{4V}\right)$$

Where all terms have the previous definitions.

Since the reverberation time of a room is the time which it takes sound to decrease to one-millionth of its original intensity after stopping the source, substituting $10^{-6}\ I_o$ for I in the above equation and solving for t gives

$$10^{-6}\ I_o = I_o \exp\left(-\frac{cAT}{4V}\right)$$

$$\text{i.e. } 10^{-6} = \exp\left(\frac{-cAT}{4V}\right)$$

$$T = \frac{0.049V}{A}$$

Where t = T = Reverberation Time of the enclosure.

Around 1930, C. F. Eyring assumed that the sound absorption process in an enclosure was discontinuous and as a result, he worked out a more general relation for the decay of sound in a closed room as:-

$$I_o = I_o \exp\left[cS\log_e(1-\bar{\alpha})t/4V\right]$$

where I, I_o, c, V and t have been previously defined.

$\bar{\alpha}$ = average value of the absorption coefficient of the walls.

i.e. $$\bar{\alpha} = \frac{a_1S_1 + a_2S_2 + a_3S_3 + \text{----------}}{S_1 + S_2 + S_3 + \text{----------}}$$

a_1, a_2, a_3 are the absorption coefficients of the various surfaces S_1, S_2, S_3.

From the previous relation, Sabine's formula for the reverberation time of a closed room was shown to be:-

$$T = \frac{0.049V}{-S[\log_e(1-\bar{\alpha})]}$$

This is similar to Sabine's equation for the reverberation time of an enclosure but the factor $\log_e(1-\bar{\alpha})$ replaced $\bar{\alpha}$ in Sabine's equation. V. O. Knudsen also showed that at higher audio-frequencies, absorption by air may form an appreciable proportion of the total absorption in an enclosure, and on the basis of this, he developed a formula for the reverberation time which included the effect of air absorption, i.e.

$$T = \frac{0.049V}{-S\log_e(1-\bar{\alpha}) + 4mV}$$

where m = Attenuation constant of the air.

Other early works on the measurement of reverberation time of an enclosure include,

(a) The Oscillograph method of measuring reverberation time by Meyer and Just.

(b) Reverberation Time meter by W. H. Hall which involved the use of two relays, a vacuum tube and a milliammeter which was calibrated to read directly in seconds.

(c) The Spark Chronograph method for measuring reverberation time by Wente and Bedell. In this method, certain times were measured during which a fixed sound source died down to various predetermined values.

In the work on which this thesis was written, the author used quite a different method from those given previously for the measurement of the reverberation time of a room. This involved measurement by digital means. It involved the use of a timer counter to obtain the time taken for sound to die down through a known fraction of the reverberation time.

The fraction chosen for the measurement was the time for 10d.B. decay of sound after the source was shut off. The reverberation time was then obtained by multiplying the time for a 10d.B. decay by 6 thus giving the time for a 60 dB sound decay. The use of a known fraction of the reverberation time to obtain the whole time was made possible by a previous conversion of the exponential decay of sound to voltage which decays linearly with time. This conversion was effected by means of a logarithmic amplifier in the reverberation meter constructed.

THEORY OF THE ACOUSTICS OF ROOMS

1.1 Properties of Sound Waves In a Room.

When sound waves strike the boundary of a room, some of its energy is reflected from the boundary surface, some is absorbed at the boundary while some is transmitted through it.

1.1.I Reflection of Sound Waves from the Walls of a Room.

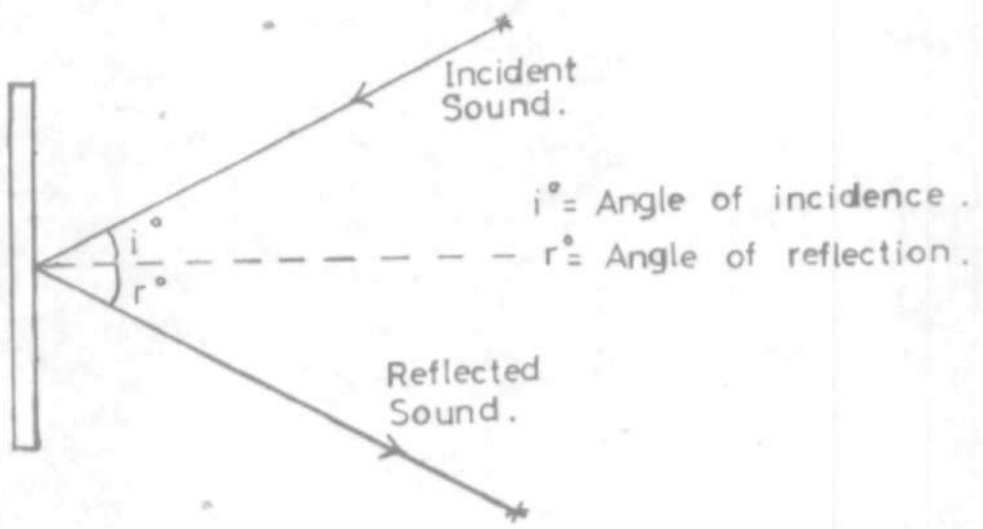
Sound waves undergo the process of reflection whenever there is a discountinuity of the medium. For regular reflection of sound waves at the boundary of a room, the same law is obeyed as in optics viz· that the angle of incidence of the sound waves is equal to the angle of reflection as shown in fig.(1;1). Therefore, when spherical sound waves reach the boundary surface of a room with normal incidence their direction will be reversed and the reflected waves will still be spherical with the same curvature but they will be travelling away from the surface as shown in fig. (1;2).

1.1.2 Absorption of Sound In Rooms:

The absorption of some of the energy of a sound source produced in a room also takes place at the boundary of the room. The nature of the surface on which the sound waves fall determined how much sound will be absorbed. When the sound energy is absorbed, it is converted into heat energy although the amount of heat is very small. The efficiency of the absorption process is rated by a number called the "absorption coefficient" of the material varying between zero and one. If no sound is absorbed the sound absorption coefficient is zero, and if all the sound is absorbed the coefficient is one. The absorption of any material in a room is not constant at all parts of the frequency range. An example showing the variation of the absorption coefficient of some absorbent materials with frequency is given in table (1;1). The absorption also depends to a slight extent on the position of the absorbent material in the room and its relation to other surfaces.

REFLECTION AT AN ANGLE

F I G . (1 ; 1)



SPHERICAL REFLECTION OF SOUND WAVES AT A SMOOTH BOUNDARY.

F I G . (1 ; 2)

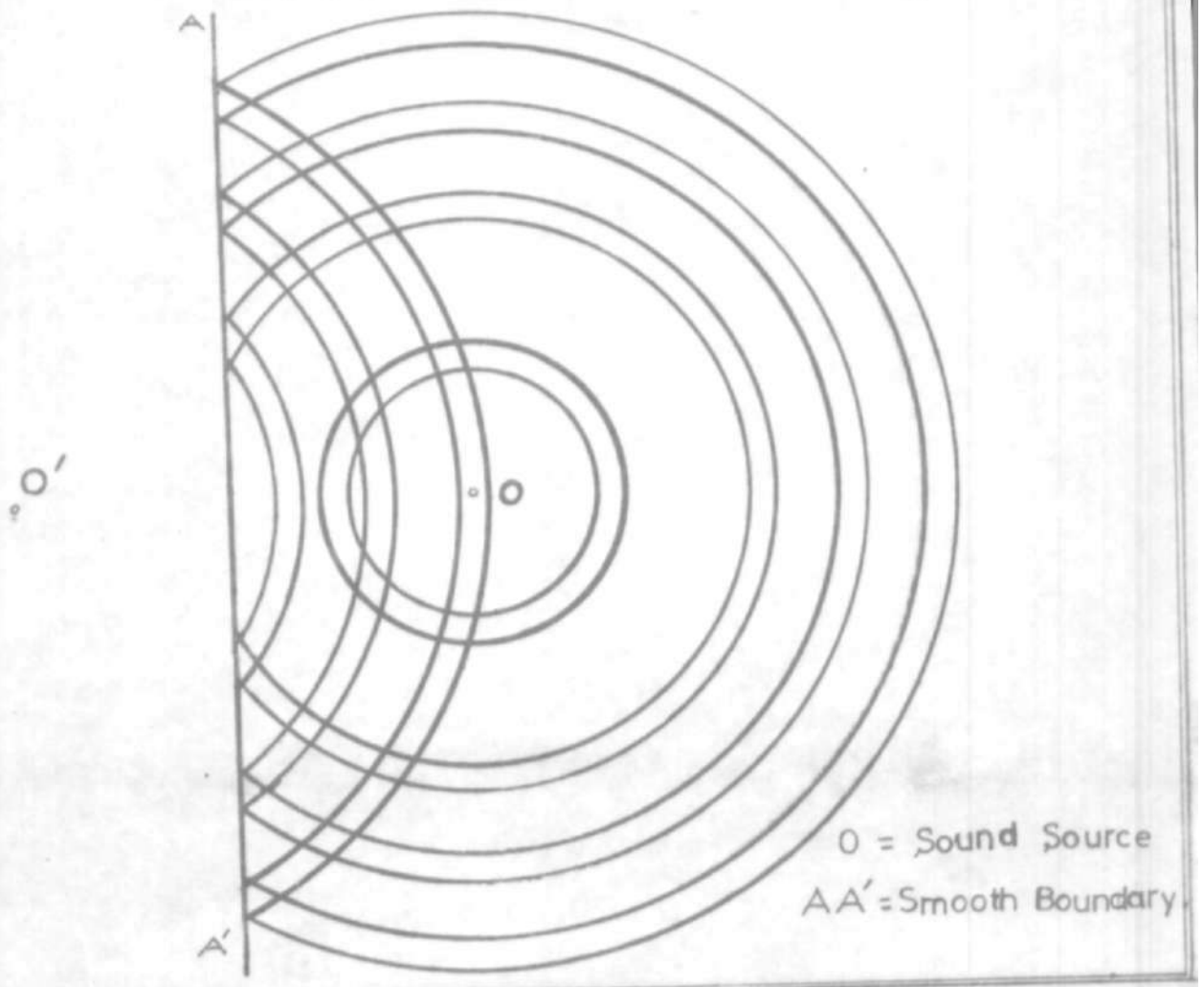


Table (1.1)

Absorbent Materials	Absorption Coefficient at frequency of 125Hz	500Hz	4.000Hz
Concret	0.01	0.02	0,05
Wooden floor	0.05	0.06	0.20
Carpet	0.09	0.21	0.37
Fibre board tiles Perforated, 1$\frac{1}{2}$" on Solid	0.05	0.54	0.60
Plywood: 3m.m. thick; 5cm air space	0.25	0.20	0.10
Acoustic Plaster	0.18	0.32	0.58
Air	-	-	0.01

1.1.3. Transmission of Sound Waves through the Boundaries of a Room

The transmission of sound waves in a room can take place at the boundary surface through the walls, doors and windows of the room.

(a) Transmission through walls

If a sound wave of frequency $f = w/2\pi$ is incident normally on a plane, homogeneous wall of mass $M_w = \rho_w d_w$ per unit area where ρ_w = density of the wall and d_w = the thickness of the wall.

the transmission loss can be written approximately as :-

$$\text{T. L. (through walls)} = 20 \log \frac{wM_w}{2\rho_a c}$$

where ρ_a = density of air

c = velocity of sound in air

M_w = mass of wall

(b) Transmission through Doors and Windows.

Doors and windows present both direct and indirect paths for the transmission of sound waves. The direct path is through the airgaps at the perimeter or edges of the doors and windows. The indirect path is through the body of the doors and windows which is caused to vibrate and radiate sound.

Transmission through these paths is reduced by sealing the edges of the doors and windows with rubber or felt strips and by using double doors.

1.2. Standing Or Stationary Waves In a Room

When sound waves strike the walls of a room and are reflected at normal incidence to the plane boundary, the incident and reflected waves interfere with one-another to form stationary or standing waves. From the acoustic viewpoint, a room may be treated as a complex resonator having numerous allowed modes of vibration each with its own characteristic frequency of damped free vibration. When a sound source is started in such a resonator, a steady state vibration having the frequency of the source is set up together with a transient free vibration composed of the various normal modes of vibration.

The transient vibration would be composed of the normal modes required to satisfy the initial acoustic conditions of the room. If the sound source is operated continuously, each component frequency of the transient dies out exponentially at its own particular rate eventually leaving only the steady state vibration.

When the sound source is shut off, the steady state waves having the frequency of the impressed sound almost immediately disappear. However, at any instant, the steady state vibration can be considered as being compounded of a large number of standing waves characteristic of the room. The amplitude of each of these standing waves depends primarily on the frequency of the source and the position of the source in the room. Consequently, as the source is shut off, the acoustical energy in the room is considered to reside in these standing waves, largely in those having natural frequencies near that of the source.

As these waves damp out exponentially, according to their individual free vibration properties, they often interfere with one-another and thus produce beat notes. The resulting damped pulsations of these free vibrations constitute the phenomena of reverberation.

1.3 Standing Wave Function

The equation of motion of a sinusoidal sound wave of amplitude A, frequency f, and velocity c is given by:-

$$\Phi(x,t) = A \sin 2\pi f(t - x/c)$$

where the wave moves in the direction of increasing x.

A similar wave moving in the opposite direction has the form:-

$$\Phi'(x,t) = A \sin 2\pi f (t + x/c)$$

If both waves exist simultaneously, then the resultant wave function Φ is given by:-

$$\Phi(x,t) = \Phi + \Phi'$$

$$\therefore \Phi(x,t) = A \sin 2\pi f (t-x/c) + A \sin 2\pi f(t + x/c)$$

$$= 2A \sin (2\pi f) \cos \left(\frac{2\pi fx}{c}\right)$$

The equation $\Phi(x,t) = 2 A \sin (2\pi f) \cos \left(\frac{2\pi fx}{c}\right)$ represents a stationary or standing wave.

If $\frac{2\pi fx}{c} = \pi/2$, i.e. $x = \frac{c}{4f} = \frac{1}{4}\lambda$

where $\lambda = c/f$ = wave length of the sound wave.

$\therefore \cos \left(\frac{2\pi fx}{c}\right) = \cos (\pi/2) = 0.$

Therefore the wave amplitude at this point is always zero.

Similarly, If $\frac{2\pi fx}{c} = 0$; i,e, $x = 0$

then $\cos \left(\frac{2\pi fx}{c}\right) = \cos (0) = 1$

and the wave amplitude is 2A and

$\Phi(t) = 2A \sin ft$ which varies sinusoidally with time with an amplitude twice that of the component travelling waves. Thus, the amplitude at a given point lies somewhere between the two limits 2A and 0.

1.4. Resonance in Rooms.

In a rectangular room of dimensions L_x, L_y, L_z feet, the natural

resonant frequencies are given by:-

$$f(l,m,n) = \frac{c}{2}\left[\left(\frac{l}{L_x}\right)^2 + \left(\frac{m}{L_y}\right)^2 + \left(\frac{n}{L_z}\right)^2\right] \text{ Hz.}$$

where c is the speed of sound in air = 1100ft/sec.

l,n,r are positive integers or zero.

The characteristic vibration that occurs for each different combination of values l, m, and n is called a mode and f(l,m,n) is referred to as the modal frequency.

In normal life, these resonant frequencies are not usually troublesome as the surroundings are sufficiently sound absorbent to eliminate them, but in certain situations, they can be troublesome. It is therefore one of the criteria of good acoustic design that a room should not have pronounced resonances.

If there is a pronounced resonant frequency in a room, this frequency will be heard as a "honk" whenever a speaker happens to sound at the frequency.

The sound energy will be caught by the resonant mode and it will persist for a longer time thus quickly standing out above the other rapidly decaying frequencies and produce a characteristic honking sound which may be annoying.

In order to reduce such a resonance, one must first try to estimate or measure its frequency. This will thus give a guide to the mode form or pattern of the resonance. Having found the standing waveform that is resonating, it is then possible to deal with it, and this can be done by placing a sound absorbing material on the walls.

1.5. "Live" and "Dead" Rooms

Rooms in which the absorption of sound is small and consequently the reverberation times are high are known as "live" rooms while those with very large absorption and very low reverberation times are called "Dead" rooms.

In the case of live rooms the sound suffers several reflections at the boundaries of the rooms during the growth or decay of the sound and the reflections are sufficient to provide a uniform energy density distribution in the rooms; but for dead rooms, the reflections are very few and the energy density in the rooms cannot therefore be assumed to be uniform.

1.6 Decay of Sound in Live Rooms

If the sound source in a live room is shut off at the time t = 0 and if E_o represents its uniformly distributed energy density at this instant, then as t increases,

$$E = E_o \exp\left(-\frac{Ac}{4V}t\right)$$

where c = velocity of sound

A = the total number of absorption units exposed within the room

V = the volume of the room.

Similarly, the intensity I at any time t is related to the initial intensity I_o by:-

$$\frac{I}{I_o} = \exp\left(-\frac{Ac}{4V}t\right)$$

$$\therefore 10 \log\left(\frac{I}{I_o}\right) = 10 \log\left[\exp.\left(-\frac{Ac}{4V}t\right)\right]$$

$$= \frac{10}{2.3} \ln\left[\exp\left(-\frac{Ac}{4V}t\right)\right]$$

$$= -\frac{10}{2.3} \cdot \frac{Ac}{4V}t$$

$$= -\frac{1.087\ Act}{V}$$

where $10 \log\left(\frac{I}{I_o}\right)$ represents the change in the intensity level in decibels. Therefore, the intensity level in a live room and correspondingly the sound pressure level decreases at a constant decay rate D given by:-

$$D\ (\text{Decay Rate}) = \frac{1.087Ac}{V}\ \text{dB per sec.}$$

1.7. Decay of Sound in a Dead Room

The equation of the decay of sound energy in a dead room is given by:-

$$E = E_o \exp.\left[\frac{cS \ln (1-\bar{\alpha})t}{4V}\right]$$

where $\bar{\alpha} = \frac{\sum \alpha_i S_i}{S}$

where $\sum \alpha_i S_i = \alpha_1 S_1 + \alpha_2 S_2 + \alpha_3 S_3 + \text{----------}$

and $\alpha_1, \alpha_2, \alpha_3, \text{----------}$ are the respective absorption coefficients of materials having surface areas $S_1, S_2 S_3, \text{----------}$ in the room.

The decay rate of sound in such rooms is therefore given in decibels per second by:-

$$D \text{ (Decay Rate)} = -\frac{1.087\, cS \ln (1-\bar{\alpha})}{V} \text{ dB./sec.}$$

and the reverberation Time is also given by:-

$$T = \frac{0.049V}{-S\ln(1-\bar{\alpha})} \text{ sec.}$$

1.8 Effect of the absorption of Sound in Air on Reverberation

Sound waves lose some energy during their propagation through a fluid medium. The intensity of a sound wave thus decreases in air in accordance with the equation:-

$$I = I_o \exp(-mx)$$

where m = the attenuation constant of air

and x = the distance travelled by the sound wave.

Therefore, during a time t, an acoustic wave travels a distance $x = ct$.

and consequently, the former equation can be written as:-

$$I = I_o \exp(-mct)$$

Since the intensity I at any time t is also related to the initial intensity I_o by:-

$$\frac{I}{I_o} = \exp\left(\frac{-Act}{4V}\right)$$

Therefore, introducting the effect of air absorption into this equation gives:- $I = I_o \exp. \left(-\frac{A}{4V} + m\right)ct.$

and the corresponding expression for the reverberation time becomes:-

$$T = \frac{0.049V}{A + 4mV}$$

where 4mV is introduced as a result of the absorption by air.

1.9 Optimum Reverberation Time of a Room

Apparently, the reverberant reinforcement of sounds in a room can have a benefitial effect on intelligibility. Therefore, there must exist an optimum compromise between sound reinforcement and excessive reverberation. The intelligibility of speech increases to a maximum as the reverberation time is decreased from large values; but as it is decreased below the optumum point, the intelligibility decreases because of decreased sound intensity in the room. Because it is usually impracticable to vary the shape or size of a room to obtain optimum reverberation time, this condition may be produced by the adjustment of the sound absorption coefficient of the wall, ceiling and floor surfaces of the room.

In small rooms such as living rooms, reception rooms, etc., it is usually possible to achieve the desired results by the use of carpets and draperies.

However, in large rooms, especially those designed for listen it is necessary to apply a specially designed absorbing material to the surfaces of the room.

1.10 Anechoic Chambers

With the increasing importance of accurate and absolute acoustic measurements, open air condition for testing are essential but uncontrollable as regard temperature, humidity, ground reflections etc This has led to the construction of the anechoic or dead rooms.

An anechoic chamber is one whose walls are so treated that practically all the incident sound energy is absorbed at the boundary surface, and consequently, the reverberation time is very low. Since no reflection occurs, the source then behaves as if it were located in an unbounded space. Under this condition, the intensity of the sound varies as the inverse square of the distance from the source.

Such rooms have walls, floors and ceilings covered with wedges of rock-wool, plastic foam, or fibre-glass each enshrouded in loosely woven cloth. The wedges are usually tapered to a narrow edge and inclined at different angles to each other so that the effective absorbing area is much increased and a slightly inclined wave striking an inclined face will suffer many reflections with diminishing amplitude before being returned to the body of the room with a much reduced intensity. Anechoic rooms are used for measuring radiation patterns and sensitivities of microphones and loud speakers and also for testing acoustical devices.

1.11 Factors Affecting the Acoustics of a Room.

For small rooms, a proper control of the reverberation time will usually insure good hearing; but for large rooms, a number of additional factors contribute to the lack of quality of the rooms. Chief among these considerations are interference, sound focussing, extraneous noise and the presence of echoes.

CHAPTER 2

CONSTRUCTION OF THE BLOCK SCHEMATIC DIAGRAM OF THE CIRCUIT USED

Since the project was concerned with the measurement of the reverberation time of a room by digital method, it was therefore decided that the time would be measured by means of a digital counter. To this end, the Advance Electronics Timer Counter T.C. 7 was chosen for the time measurement.

Before embarking on the measurement of the decay time of sound in a room, it was thought necessary that a sound source which would produce sound of required intensity should first be obtained. A loudspeaker was chosen for this purpose. To operate the loudspeaker, an audio frequency signal generator would be required to produce audio tones and these would pass through a power amplifier having an adjustable power gain. The output from the power amplifier would then be connected to the loudspeaker. This arrangement could be used to produce sound whose intensity could be varied. Therefore, the blocks representing the audio frequency signal generator, the power amplifier and the loudspeaker were first located in the block schematic diagram of the circuit as shown in figure (2.1).

After producing a steady sound source in the room, the next action would be to stop the sound source, and allow the sound to decay in the room. To stop the sound source, a switch could be connected between the power amplifier and the audio-frequency signal generator and this could be used in controlling the starting and stopping of the loudspeaker. The decay of the sound produced in the room would commence as soon as the source was put off. Therefore, to measure the time of decay of the sound, it would be necessary that the Timer Counter employed was started immediately the sound source was shut off. This counter was started by a closed contact to ground. It was therefore decided that a two position switch with one make and one break contacts could be used so that in the normal position of the key, nothing happened, but as the key was ~~push~~ pushed up, it would close the contact between the audio-frequency signal generator and the power

amplifier, thus starting the loudspeaker. This action would also break the contact between the start terminal of the counter and ground. However, on depressing the key of the switch, the converse of the previous action would occur. This time the contact between the start terminal and ground would be closed leaving the other contact open. Consequently, the counter would start counting in the unit of time employed while the loudspeaker was off. A capacitor of magnitude 0.05 μf was to be connected between the start terminal and ground to suppress contact bounce.

After starting the timer counter immediately the sound source was put off, the next action would be to arrange to stop it when the sound decayed to a certain level. It was then decided that a microphone would be needed to convert the sound energy received in the room to electrical energy and a certain voltage would be developed in it. The voltage developed by this microphone would have the same decay characteristic as that of the decaying sound in the room.

Since the voltage developed would be a very small one (2mV), a linear amplifier would therefore be needed to amplify it. As a result of this, the block representing this microphone pre-amplifier was positioned in the block circuit as shown in fig. (2,1)

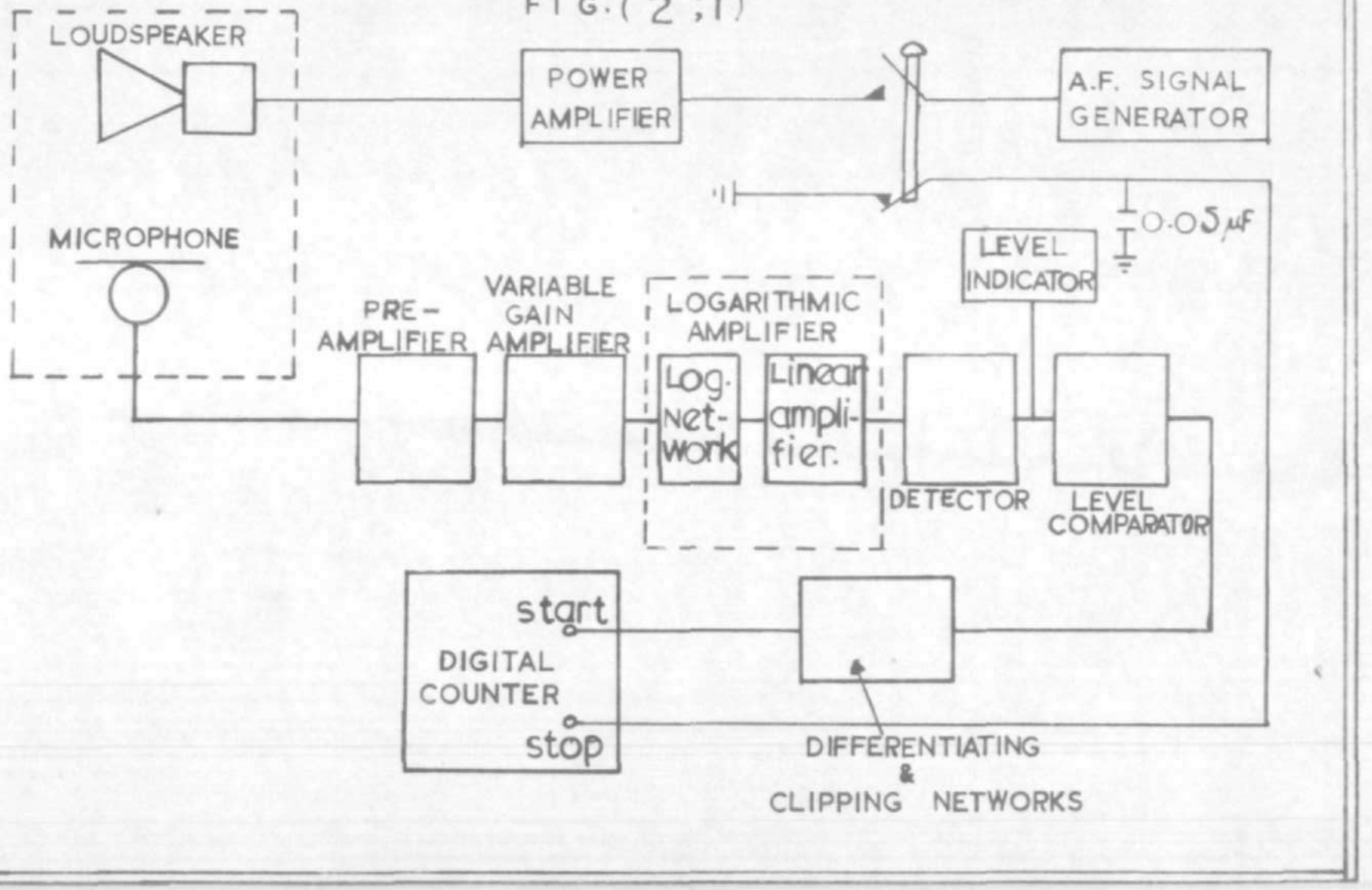
It might be necessary to vary the output voltage from the amplifier and to effect this, a variable gain amplifier would then be needed. This led to the addition of the block representing the variable gain amplifier to the circuit,. This block was then connected to the one representing the microphone pre-amplifier.

The output voltage from the variable gain amplifier would also have an exponential decay characteristic similar to that of the decaying sound.

Since sound intensity is measured in decibel which is a logarithmic measure, the aim should be to measure the sound intensity logarithmically. Because a logarithmic is the converse of an exponential, the exponential decay of sound is converted to a voltage which decays linearly with time. To obtain the logarithm of the output voltage from the variable gain amplifier

BLOCK SCHEMATIC DIAGRAM OF THE REVERBERATION TIME METER.

FIG.(2;1)



a certain logarithmic network could be connected to the output of the amplifier. As the output voltage from this logarithmic network would be small, a linear amplifier would be needed to amplify it. The combination of the logarithmic network and the linear amplifier was called a logarithmic amplifier. The output voltage from the logarithmic amplifier would then have a linear decay with time istead of the exponential decay of sound. This output a.c. voltage could then be passed through a detector which would convert it to a d.c. voltage having the same linear decay as the a.c. voltage.

At the detector output, a level indicator (Vacuum tube voltmeter) could be connected in parallel with the circuit. This level indicator would be used to give the level of the decaying voltage at any instant of time. If a timing device is used to note the time to change from one level of sound to another, the time for a specified change of level (dB) can be calculated because the decay time curve is linear.

The next step would be to find a certain equipment which would stop the counter when the voltage and correspondingly the sound died down to a particular level as given on the level indicator. This therefore, necessitated the use of a voltage level comparator (Schmitt trigger). This level comparator would be designed so that it would change state (Switching would occur) when its input voltage decreased to or passed a stipulated level. Consequently, a positive-going pulse would be given at the output of the Schmitt trigger. However, the amplitude of this voltage pulse should be greater than 4.5V p.p. but it should not exceed 8V.p.p, in accordance with the specifications for the use of the timer counter employed. Since a negative voltage stop would be required to stop the counter, a differentiating circuit could then be inserted to differentiate the pulse given at the output of the Schmitt trigger, thus converting,it into two narrow spikes one positive and the other negative. The positive spike could then be clipped off using a diode clipping network. The negative spike produced at the output of the clipping network would therefore be capable stopping the counter, and the time for the decay could then be read off on the timer counter.

Blocks representing the logarithmic amplifier, the detector, the level recorder, the level comparator and also the differentiating and clipping networks respectively were located in the block schematic diagram of the circuit as shown in figure (2;1). This figure gives a block schematic diagram of the whole arrangement.

CHAPTER 3

GENERAL DESCRIPTION OF EQUIPMENT USED

3.1 List of equipment used.

After constructing the block diagram of the circuit to be used, the next step taken was the compilation of a list of all the equipment which would be needed. The list included:

1. a timer counter
2. a Loudspeaker
3. a microphone
4. an audio frequency signal generator
5. a make and break switch
6. a microphone Pre-amplifier
7. a variable gain amplifier
8. a logarithmic amplifier
9. a detector
10. a Level recorder
11. a level comparator
12. Differentiati-ng Network
13. Power amplifier.

Out of these, those which were then available as ready units were:-

1. The timer counter
2. The Loudspeaker
3. Microphone
4. A.F. Signal generator
5. Make and break switch
6. Power amplifier
7. The level recorder
8. D.C. Power Supply Unit.

The remaining ones were to be designed and constructed.

3.2 Description of the ready pieces of Equipment

The timer counter

The timer counter employed was the Advance Electronics type T.C. 7; Serial No. 434. This counter is a four digit instrument which is capable of time, period, and frequency measurements and it can also be used for the counting of random or regular pulses.

In the project, the counter was started by a closed contact to ground and contact bounce was suppressed by a capacitor of magnitude 0.05μF connected between the start terminal and earth. Counting was stopped by the application of a pulse to its terminal. The time between the starting and stopping of the counter was then read directly from the instrument.

This timer counter could be used to make time measurements in three ways, viz:- Manual, 1 Line and 2 Line as given on the counter.

(i) Manual :- In this case, the manual start and stop push buttons control the count gate while the function switch may be on any position.

(ii) 1 Line :- When the function switch is set to this position, the time units are counted between the first and second pulses received on the start terminal after reset is made.

(iii) 2 Line :- With the function switch at this position, the time units selected will be counted between the application of a pulse to the start terminal and a pulse to the stop terminal.

The counter would respond only to pulses with negative-going steps of amplitude greater than 4.5 V. p.p. and a maximum of 8V. p.p.

Voltage steps of magnitude greater than 8V. p.p. would damage the instrument. The counter is capable of measuring time in any one of the three units:- 10μsec.unit, 1 Millisec. Unit, and 0.1 Sec. Unit.

The third method of time measurement (with the function switch at 2 LINE) was employed in the project and the 1 millisecond unit was also used because it gave a more accurate measurement than the 0.1 Second Unit; and it also gave a wider range of measurement than the 10μSecond Unit.

3.3. The Loud Speaker

This is an electro-acoustic transducer designed to radiate acoustic power into a room or open air. The two most widely used types of loudspeakers are the direct radiation or dynamic loudspeaker and the horn loudspeaker. Both of these utilize the electrodynamic coupling existing between the motion of a vibrating surface called the diaphram, and the current in a voice-coil.

An ideal loudspeaker should:-

(a) have a high electro-acoustic efficiency.

(b) faithfully reproduce transients as well as steady input signals.

(c) have an acoustic output response which is independent of frequency over the entire audible range.

(d) Introduce neither harmonic nor intermodulation distortion into its output.

(e) be capable of producing a non-directional radiation pattern.

(f) be of as small a size as possible considering the required acoustic output.

3.4. The Direct Radiation Loud Speaker

These are of two types:- the moving coil or electrodynamic type and the crystal loudspeakers. The moving coil loudspeaker is the commonest type. It often consists of a voice coil situated in a magnetic field produced by either a permanent magnet or an electro-magnet. An audio-frequency current flows through the voice-coil and generates a force $2\pi r$ NBI

where I = the a. f. current

N = Number of turns of the coil.

B = the flux density

r = the radius of the coil

The coil thus moves in and out in sympathy with the force on it.

3.5. The Horn Loud-Speakers

This type of loudspeakers consists of a transducer of moving coil type matched acoustically to the medium by means of a horn placed in front of, but not touching the diaphram. A constriction is placed behind the throat of the horn to prevent any destructive interference of the waves at the throat. The main advantage of the horn loudspeaker is its very high transfer efficiency i.e. a high ratio of acoustical output energy to electrical input energy.

The loudspeaker provided for use in the project was the moving coil type which was enclosed in a reflex cabinet i.e. a sealed box with a rectangular hole air vent.

3.6. The Microphone

This electro-acoustic transducer is used for converting acoustic energy into electrical energy. It serves two principal purposes. First, it can be used for converting music or speech into electrical signals which are then reproduced. Second, it can be used as a measuring instrument convertin; acoustic signals into electrical signals which actuate certain indicating meters. For most applications a microphone should be highly sensitive, suffer minimum background noise, and in addition, a linear response with frequency may be required. However, no microphone is ideal in all of these repects and a compromise has to be made in selecting one for a particular application.

It is also important to specify direction because there is generaly a variation in the sensitivity of the microphone as its diaphram is rotated relative to the direction of the sound waves.
In general, the directional properties of a microphone may be classified as follows:-

(a) Omnidirectional

(b) Figure of 8

(c) Cardioid . These are shown in the polar diagrams in fig. (3.1).

An omnidirectional response may be obtained by placing a moving coil microphone in such a position that its diaphram is horizontal The figure of eight response occurs for a ribbon microphone for which there is a phase difference of 180 degrees for symmetric pairs of points on opposite sides of the ribbon. The cardioid response is obtained when a moving coil and a ribbon microphones are placed in the same housing.

Microphones may be classed into two main groups:-

(a) the pressure microphones

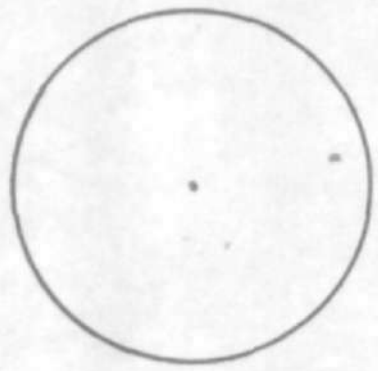
(b) the pressure gradient microphones

DIRECTIONAL RESPONSES OF MICROPHONES

F I G. (3 ; 1)

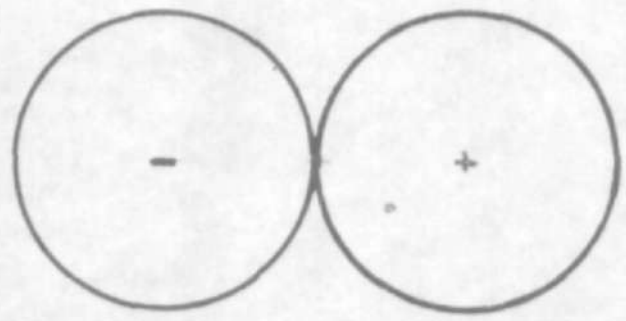
(a)

Omni-Directional



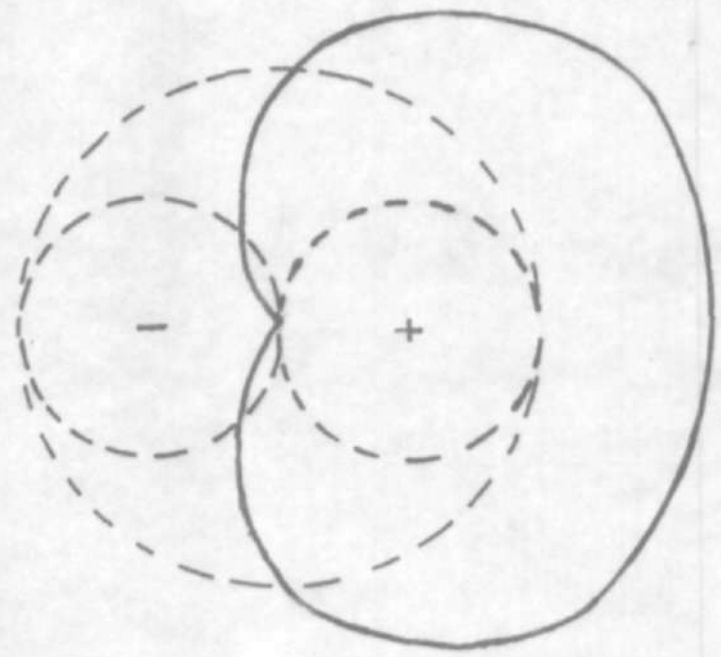
(b)

Figure of 8



(c)

Cardioid.



In pressure microphones, the electrical response corresponds to the variations in the acoustic pressure while in the Pressure gradient or Velocity operated microphones, the electrical response corresponds to the particle velocity resulting from the propagation of sound waves in air.

Table (3,1) gives a list of different types of microphones and their respective performances. In the project, a pressure gradient microphone was made available. The microphone has a low impedance, a good frequency response and a good quality. Its output is low (2mV.)

Table (3,1)

Microphones and Their Performances

CLASS	TYPE	OUTPUT	ELECTRICAL IMPEDANCE	FREQUENCY RESPONSE	QUALITY
PRESSURE MICROPHONES	Carbon	Very High	Low	Poor	Very Poor
	Condenser	Low	Very High	Good	Excellent
	Crystal	High	High	Good	Good
PRESSURE-GRADIENT OR VELOCITY-OPERATED MICROPHONES	Moving Coil	Low	Low	Fairly Good	Good
	Ribbon	Very Low	Very Low	Good	Excellent

3.7. The A. F. Signal Generator

The signal generator used in the project was the "Advance" Audio Frequency Signal Generator Type J 2B No. S 790. This instrument provides sinusoidal outputs in the frequency range of 15Hz to 50 kHz. A signal of any frequency in this range is set by using a range switch in conjunction with a calibrated dial; and continuous adjustment is by means of a slow-motion control situated centrally on the dial.

Two separate output arrangements with a continuous level control are provided on the instrument. One output is of 600Ω impedance. It is isolated from earth and has a maximum output level of 1 Watt. The alternative output has an impedance of 5Ω connected to earth and with an output level of about 500 m Watt. The output level is indicated on a front panel meter.

A 20 dB attenuator is also in-corporated in the instrument and when this attenuator is used in the circuit and the output is loaded with 600Ω, the output voltage is one-tenth of that indicated.

3.8. Power Amplifier

Sinclair Z 12 was used. It was taken as a ready piece of equipment. It is a high quality transistor power amplifier. It contains 8 transistors and it can be used on any supply voltage down to about 6 volts or up to about 20 volts. In the project, a supply voltage of 12 volts was employed. Also, to reduce the number of power supply units used in the circuit, 16 U2 batteries were used to supply the voltage.

3.9. The Level Indicator

The Hewlett Packard Vacuum tube voltmeter model 410 B was used as d.c. level indicator.

Both a.c. and d.c. ranges are provided in the instrument. The d.c. ranges have full-scale values of 1,3,10,30, 100, 300 and 1000 volts. The d.c. input resistance is approximately equal to 122 megohm.

Other ready pieces of equipment used are:- a two position switch with one make and one break contacts and also a Solartron Power Supply unit, A S 1414 No. P818.

CHAPTER 4

THE MICROPHONE PRE - AMPLIFIER

4.1 Design Considerations.

As the microphone employed for the acoustic measurement is placed in sound field, it receives sound energy which is consequently converted to an electrical signal. This signal is a weak one having a maximum of about 2 mV. A linear amplifier known as the microphone pre-amplifier is then required to amplify it.

This pre-amplifier should be such that:-

(a) the voltage gain is about 40 (or 32 dB.) so that a 2mV. input signal would yield about 80mV. at the output.

(b) the frequency response is flat over a frequency range of about 100Hz to 10kHz.

(c) the output has minimum distortion.

4.2 Choice of Configuration

There are three basic transistor configurations.

These are:-

(i) The common Base Configuration in which the input is applied to the emitter, the output is taken from the collector and the base is common to both the input and output circuits.

(ii) The Common Emitter Configuration having the input applied to the base, the output taken from the collector and the emitter is common to the input and output circuits.

(iii) The Common Collector Configuration in which the input is applied to the base, the output is taken across the load in the emitter and the collector is common to the input and output circuits.

The common base amplifier is characterized by a low input impedance and a high output impedance.

The current gain is slightly less than 1, but the voltage gain is usually large and thus it has a relatively high power gain.

The common emitter has an input impedance somewhat higher than that for the common base amplifier. The current gain is large and the voltage gain is also large and thus the power gain is considerably greater than that for the common base amplifier.

The common collector amplifier is characterized by a very high input impedance and a relativcly low output impedance. The current gain is large but the voltage gain is slightly less than 1.

The amplifier has a power gain generally lower than those for the common base and common emitter amplifiers.

The common emitter configuration was therefore used in the design of the microphone pre-amplifier because of its high gain and because it has adequate frequency response for audio amplifiers.

4.3. Biasing

The operating point of a transistor is capable of shifting from its desired value as a result of two factors. The first is the temperature dependence of the collector base leakage current I_{CBo} and the other is the wide variations in the large signal value of the common emitter current amplification factor α.

Therefore, to prevent this shifting of the quiescent operating point which may cause the distortion of the output voltage, a proper d.c. biasing of the transistor stage should be employed so as to establish and maintain the operating point inside a prescribed region in spite of temperature changes and replacement of the transistor.

There are five fundamental biasing techniques

viz:- (a) the fixed bias
(b) the emitter bias
(c) the cut-off bias
(d) the self bias
(e) the emitter-resistor and base potential devider bias (the single battery bias).

The fixed bias technique provides a simple and in-expensive means for establishing an operating point. However, the stability of the operating point is extremely poor when compared with that of the other circuits.

BIASING CIRCUITS FIG (4;1)

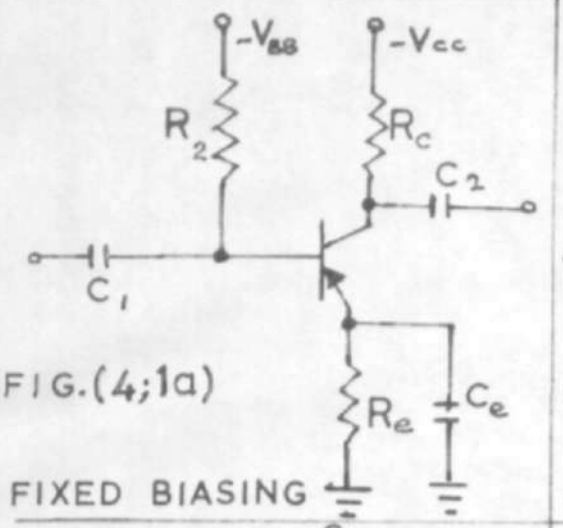


FIG.(4;1a)

FIXED BIASING

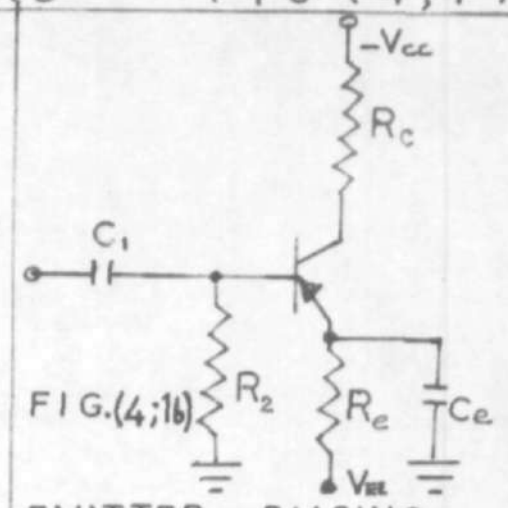


FIG.(4;1b)

EMITTER BIASING

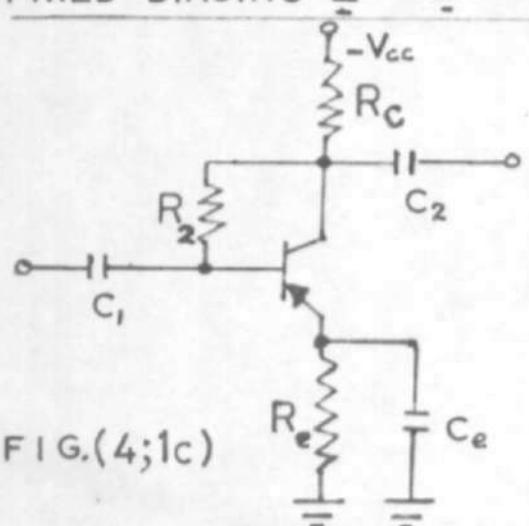


FIG.(4;1c)

SELF BIASING

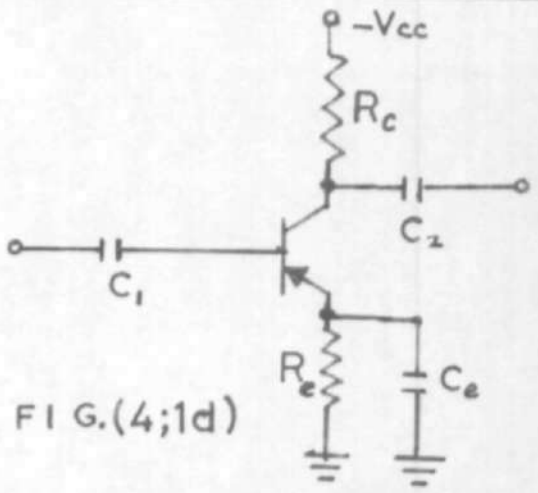


FIG.(4;1d)

CUT-OFF BIASING

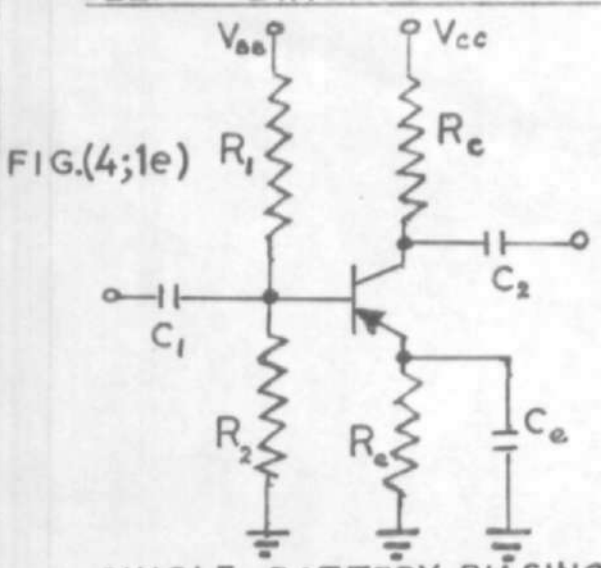


FIG.(4;1e)

SINGLE BATTERY BIASING

EQUIVALENT CIRCUIT FOR FIG.(4;1e)

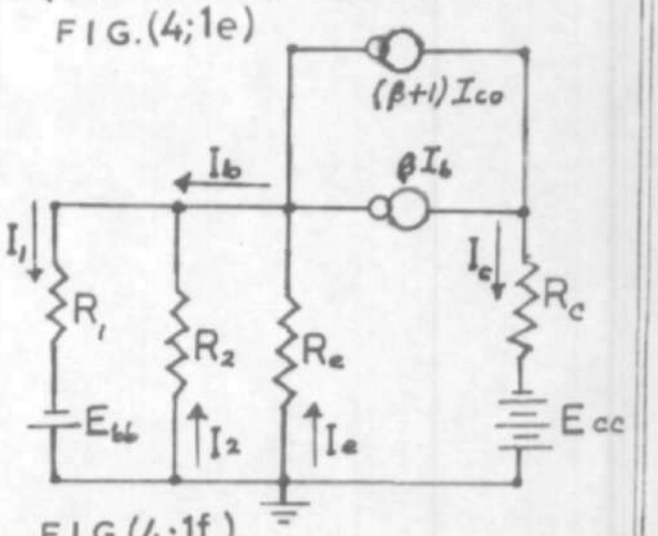


FIG.(4;1f)

The emitter bias shown in fig.(4;1b) provides a high degree of stability but has a disadvantage that two d.c. supplies of opposite polarity are often needed.

The cut-off bias is inexpensive since it uses only a single battery source, but its stability is also poor. This biasing method is shown in fig.(4·1d).

The single battery bias and the self bias shown in fig. (4·1e) and fig.(4·1c) respectively also make use of a single supply source and they provide a high degree of stability against temperature and α variations.

The single battery bias method was therefore employed in the amplifier circuits constructed in the project because of the single supply source and stability considerations.

4.4. Stability Factor S

A common way of determining whether a particular circuit has a good or poor thermal stability is by the use of the stability factor S. The stability factor of a circuit is the ratio of the change in the collector current to the change in the collector-base leakage current IcBo which causes the change in the collector current.

i.e. S (stability factor) = $\frac{Ic}{IcBo}$ or $\frac{Ic}{Ico}$

This factor gives the effect a change in Ico has upon the collector current.

4.5. The Stability Factor S for Single Battery Bias Circuit.

Considering the equivalent diagram of the single battery Bias circuit given in fig.(4;1f) and writing down the loop and nodal equations,

we have:- $I_c = I_c - I_b$ ----------eqn. (1)

$I_c = \beta I_b + (\beta+1) I_{co}$ ---------- ,, (2)

$I_b = I_1 - I_2$ ---------- ,, (3)

$E_{bb} = I_1 R_1 + I_2 R_2$ ---------- ,, (4)

$0 = I_2 R_2 - I_c R_c$ ---------- ,, (5)

Solving the above equations for I_c and replacing β by $\frac{\alpha}{1-\alpha}$

We have

$$I_c = \frac{\alpha R_c E_{bb} + I_{co} R_c (R_1+R_2) + R_1 R_2}{R_c (R_1 + R_2) + (1-\alpha) R_1 R_2}$$

But $$S = \frac{\Delta I_c}{\Delta I_{co}}$$

$$= \frac{R_c (R_1 + R_2) + R_1 R_2}{R_c (R_1 + R_2) + (1-\alpha) R_1 R_2}$$

Putting $\frac{R_1 R_2}{R_1+R_2} = R_b$, we have:-

$$S = \frac{R_c + R_b}{R_c + (1-\alpha) R_b}$$

$$= \frac{1 + R_c/R_b}{1 - \alpha + R_c/R_b}$$

If R_c is small compared to R_b, $S = \frac{1}{1-\alpha} = \frac{1}{0.02} = 50$ (for α=0.98)

If R_c is large ,, ,, R_b, $\therefore$ S = 1

To ensure good stability, S should have magnitude between 1 and 10.

4.6. Design Procedure

4.6.1. Calculation of the Load Resistor R_3

For this amplifier stage, a low power transistor suitable for low level stages, OC 71 was employed. The maximum output voltage developed by the microphone used was 2mV.

The desired voltage gain of the pre-amplifier = 40 or 32 dB

The current gain for the transistor OC 71 employed = 47, input impedance of the transistor 0.8kΩ .

But voltage gain of amplifier = $\frac{h'_{21}R_1}{h'_{11}} = 40$

where h'_{11} = Input impedance of the transistor.

and h'_{21} = Current gain of the transistor

$$R_1 = \frac{R_o R_3}{R_o + R_3}$$

where R_o = Output impedance of the transistor.

R_3 = Collector load resistance of the amplifier.

$$R_1 = \frac{40h'_{11}}{h'_{21}} = \frac{40 \times 0.8}{47}$$

$$= 0.68 \text{ k}\Omega$$

The makers quote $R_o = 1$ kΩ

$$R_1 = 0.68 \text{ k}\Omega = \frac{R_o R_3}{R_o + R_3} = \frac{R_3}{1+R_3}$$

$$\therefore 0.68 = R_3 (1 - 0.68) = 0.32 R_3$$

$$\therefore R_3 = \frac{0.68}{0.32} \text{ k}\Omega = 2.15 \text{ k}\Omega$$

The nearest preferred value of R_3 = 2.2kΩ

4.6.2. Choice of Operating Point

Since on applying an input voltage of 2mV. to the pre-amplifier, an output voltage of about 80mV would be required. Therefore, the operating point was chosen so as to permit more than 2mV. base drive, and give more than 80 mV at the output. i.e. $V_{ce} > V_{out}$.

Choosing V_{ce} = 3V gives I_{cq} = 3.2mA.

where V_{ce} = Collector emitter voltage at the operating point

I_{cq} = quiescent collector current

The current gain for the transistor, $h'_{21} \simeq 50$.

$$\therefore h'_{21} = \frac{I_c}{I_b} \text{ : i.e. } I_b = \frac{I_c}{h'_{21}}$$

$$\therefore I_b \text{ max} = \frac{I_c}{h'_{21}} = \frac{3.2\text{mA}}{50} = \frac{6.4 \times 10^3 \mu\text{A}}{10^2}$$

Therefore, the maximum base current swing was 64μA. A d.c. loadline was drawn. Its slope gave $\frac{1}{R_3 + R_e}$

where $R_e + R_3 = 2.8\ k\Omega$ as shown in fig. (4:2).

4.6.3 Calculation of Emitter Resistor R_4 (R_e is designated R_4)

Considering the transistor d.c. equation, i.e

$$V_{cc} = R_e I_e + V_{ce} + I_c R_c$$

for small I_b, $I_c = I_e$.

$$V_{cc} = (R_e + R_c) I_c + V_{ce}.$$

But $V_{ce} = 3V$ and $R_c = R_3 = 2.2k\Omega$.

$$V_{cc} = -12V$$

$$\therefore \quad 12 = 3 + (R_e + R_c) I_c$$

$$= 3 + (R_e + R_c) \times 3.2$$

$$9 = 3.2 (R_e + R_3); \text{ But } R_e = R_4$$

$$\therefore (R_4 + R_3) = \frac{9}{3.2}\ k\Omega = 2.84\ k\Omega$$

Similarly, from the d.c. loadline in fig. (4;2),

$$R_e + R_c = 2.8k\Omega = R_4 + R_3.$$

But $R_3 = 2.2k\Omega$

$$R_4 = (2.84 - 2.2)\ k\Omega$$

$$= 0.64\ k\Omega.$$

$\therefore$ The nearest preferred valued of $R_4 = 0.68k\Omega$.

4.6.4 Calculation of the Bias Components R_1, R_2

If R_b represents the effective resistance of the biasing resistors R_1 and R_2; and V_b represents the base voltage,

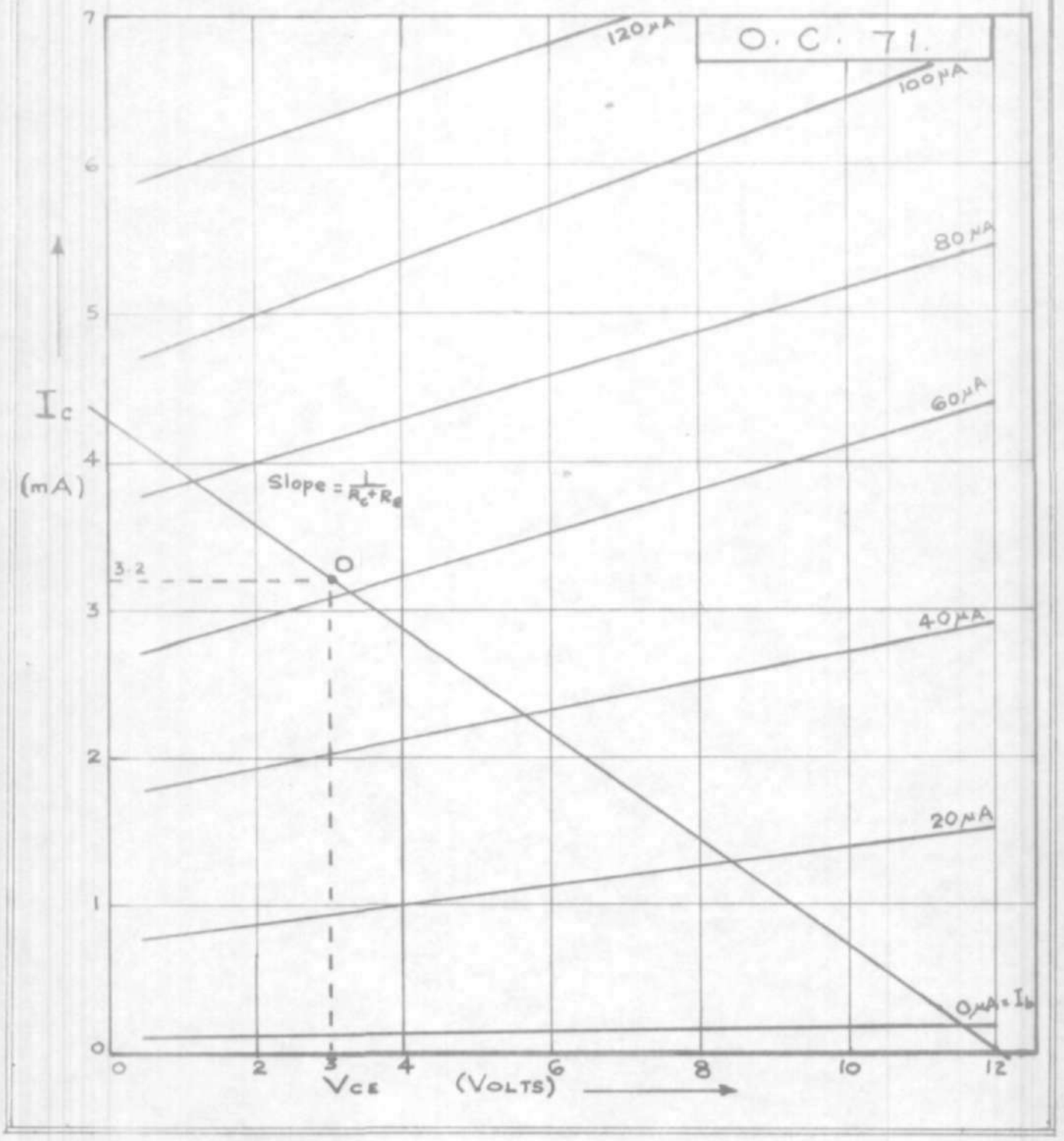
$$\therefore R_b = \frac{R_1 R_2}{R_1 + R_2}$$

where R_1, R_2 are the bias resistors.

$$V_b = \frac{V_{cc} R_2}{R_1 + R_2} - \left(R_c I_e + I_b \frac{R_1 R_2}{R_1 + R_2}\right)$$

FIG. (4;2)

OPERATING POINT & D.C. LOAD-LINE FOR MICROPHONE PRE-AMPLIFIER.



$$V_b = \frac{V_{cc} R_2}{R_1 + R_2} - \left(R_4 I_e + I_b \frac{R_1 R_2}{R_1 + R_2}\right)$$

But $I_e = I_c$ when I_b is small

$$\therefore \frac{V_{cc} R_2}{R_1 + R_2} > I_e R_e = I_c R_e = I_c R_4$$

i.e. $$\frac{V_{cc} R_2}{R_1 + R_2} > I_c R_4.$$

$R_4 = 0.68 k\Omega$ and $I_c = 3.2$ mA.

$$\therefore \frac{V_{cc} R_2}{R_1 + R_2} > 3.2 \times 0.68 = 2.18 \text{ volts.}$$

i.e. $$\frac{V_{cc} R_2}{R_1 + R_2} > 2.18 \text{ volts.}$$

To ensure good stability, a stability factor of magnitude between 1 and 10 is necessary.

Choosing $S = 9$ and using the single battery bias method,

$$S = \frac{1 + R_e/R_b}{1 - \alpha + R_e/R_b} = 9$$

where $R_e = R_4$ and $R_b = \frac{R_1 R_2}{R_1 + R_2}$

and $\alpha = 0.98$

$$\therefore S = \frac{1 + R_4/R_b}{1 - 0.98 + R_4/R_b} = 9$$

$$= \frac{1 + R_4/R_b}{0.02 + R_4/R_b}$$

$$\therefore 1 + R_4/R_b = 0.18 + 9\ R_4/R_b$$

$$\therefore 8\ R_4/R_b = 1 - 0.18 = 0.82$$

$$\therefore R_4/R_b = \frac{0.82}{8} \ldots k\Omega = 0.1\ k\Omega$$

But R_4 = 0.68 kΩ

$\therefore R_b = \frac{R_4}{0.1}$ kΩ = 6.8kΩ

= 7kΩ

But $\frac{V_{cc}R_2}{R_1+R_2}$ = 2.18 volts

If $\frac{V_{cc}R_2}{R_1+R_2}$ = 2.2 volts,

V_{cc} = -12 V.

$\therefore 12\ R_2 = 2.2\ (R_1+R_2)$.

$\therefore 9.8\ R_2 = 2.2\ R_1$

i.e. $R_1 = \frac{9.8}{2.2} R_2$

$\therefore R_1 = 5\ R_2$

But R_b = 7kΩ = $\frac{R_1R_2}{R_1+R_2} = \frac{5R_2^2}{5R_2+R_2}$

$= \frac{5}{6} R_2$

$\therefore R_2 = \frac{7x6}{5}$ kΩ = 8.4 kΩ

and R_1 = 5x8.4 kΩ = 42kΩ

Preferred values of R_1 and R_2 are:-

R_1 = 39kΩ

R_2 = 8.2kΩ

4.6.5 The Coupling Capacitor C_1

The capacitor C_1 coupling the microphone with the pre-amplifier was used to present a short circuit to the a.c. signal going from the microphone to the base of the pre-amplifier. Therefore, if the lowest frequency of the signal considered is f Hz, and the input impedance of the transistor amplifier is r_{in};
and if the effective bias resistance of the amplifier is R_b,

then, the reactance of the capacitor, $X_{C1} \ll \frac{r_{in} R_b}{r_{in}+R_b}$

But $X_{C1} = \frac{1}{wC_1} = \frac{1}{2\Pi fC_1}$

where $w = 2\Pi f$

$\therefore \frac{1}{2\Pi fC_1} \ll \frac{r_{in} R_b}{r_{in} + R_b}$

Assuming that $\frac{1}{2\Pi fC_1} = \frac{1}{20} \frac{r_{in} R_b}{r_{in}+R_b}$

$\therefore C1 = \frac{20 (r_{in}+R_b)}{2\Pi f\ r_{in} R_b}$

f = 100Hz: $r_{in} = 0.8K\Omega$: $R_b = 7K\Omega$

$\therefore C_1 = \frac{20 \times 7.8 \times 10^3}{2 \times 100 \times 7 \times 0.8 \times 10^6}$ F.

$= \frac{7.8 \times 10^6}{7.0.8 \times 10^4}$ μF.

$= 44.5\mu$F.

The nearest preferred value of $C_1 = 50\mu$F.

4.6.6. The Decoupling Capacitor C_3

The capacitor C_3 was used to decouple the emitter resistor R_4 to a.c. signal in order not to introduce unwanted feedback. Therefore, the reactance of the capacitor X_{C3} at the lowest signal frequency considered must be small compared with the value of the emitter resistor R_4

i.e. $X_{C3} \ll R_4$ (Emitter resistor)

But $X_{C3} = \frac{1}{wC_3} = \frac{1}{2\Pi fC_3}$

Assuming that $X_{C3} = \frac{1}{20} R_4$

$R_4 = 0.68k\Omega$

i.e: $\frac{1}{2\pi f C_3} = \frac{R_4}{20} = \frac{0.68 \times 10^3}{20}$

$\therefore C_3 = \frac{20}{2\pi f \times 0.68 \times 10^3}$

f = 100Hz

$\therefore C_3 = \frac{20}{2\pi \times 0.68 \times 10^2 \times 10^3}$ F

$= \frac{10^6}{0.68 \times 10^4}$ μF

$= 47$ μF.

$\therefore$ The nearest preferred value of $C_3 = 50\mu$F.

4.7. Tests for the Microphone Pre-amplifier.

Two main tests were performed for the pre-amplifier after construction. The first test was the one for the frequency response of the amplifier while the other was for the linearity characteristic.

(a) Frequency Response of Amplifier:

This was investigated by introducing an input a.c. signal voltage of 2 mV. into the input of the pre-amplifier. The signal voltage was obtained from the Hewlett Parckard 651B Test Oscillator No. G. 1439. The input was kept constant for all frequencies considered. The input and output voltages were measured by means of the Hewlett Packard vacuum tube voltmeter 400H and the output was also displayed on the Cathode Ray Oscilloscope Telequipment type D34 No. 0.896 so as to know when the distortion of the output voltage would set in.

The output voltage was then measured for varying values of frequencies between 20 Hz and 50K.Hz. keeping the input voltage at 2mV. for each measurement.

The readings of the input voltage, the frequencies and the corresponding output voltages were tabulated and the gain of the pre-amplifier in decibel was calculated at each frequency. This result is given in Table (4;1). A graph of the gain of the amplifier (in dB) against frequency was then plotted using the readings in Table (4·1) This graph is shown in fig. (4;3).

From the graph, it was observed that the pre-amplifier had a maximum gain of 36.2dB and it gave a flat frequency response within the range of about 70 Hz. to 10K.Hz.

However, this result did not give a considerable difference from those specified for the design. The amplifier was designed for a gain of about 32 dB and to have a flat frequency response between 100Hz. and 10K. Hz.

(b) Linearity:- The linearity characteristic of the pre-amplifier was also investigated. This was carried out by measuring the corresponding output voltages for different values of input voltages at certain frequencies in the audio frequency range. The measurements of the input and output voltages were obtained from the vacuum tube voltmeter 400F (Moulett Packard) as before.
The frequencies at which the input and output voltage variations were measured were 200 Hz, 500 Hz., 1000Hz, 5000Hz, 10000Hz and 15000Hz. The results are shown in Table (4;2) and the graph of the output voltage against the input voltage drawn with the results displayed in this table is given in fig.(4;4). It was observed that the input and output voltage variations have an identical linear relation at the frequencies considered.

Table (4;1)

For Microphone Pre-Amplifier Frequency Response.

Frequency in Hz.	Input Voltage (r.c.) in nV	Output Voltage (r.c.) in nV	Voltage Gain in dB
20	2	50	28
50	,,	90	33
70	,,	110	34.8
100	,,	120	35.6
200	,,	130	36.2
300	,,	130	36.2
400	,,	130	36.2
500	,,	130	36.2
700	,,	130	36.2
1000	,,	130	36.2
2000	,,	130	36.2
5000	,,	130	36.2
7000	,,	120	35.6
10000	,,	110	34.8
15000	,,	90	33
20000	,,	80	32
30000	,,	60	29.6
40000	,,	40	26

Test Equipment:

Hewlett Packard Vacuum Tube Voltmeter 400Hz, No.M.1765

Hewlett Packard 65 1B Test Oscillator No. G.1439

Cathode Ray Oscilloscope, Telequipment Type D43 No.0.896

Solartron Power Supply Unit AS 1414 No. P818.

FREQUENCY RESPONSE OF MICROPHONE PRE-AMPLIFIER

FIG.(4 ; 3)

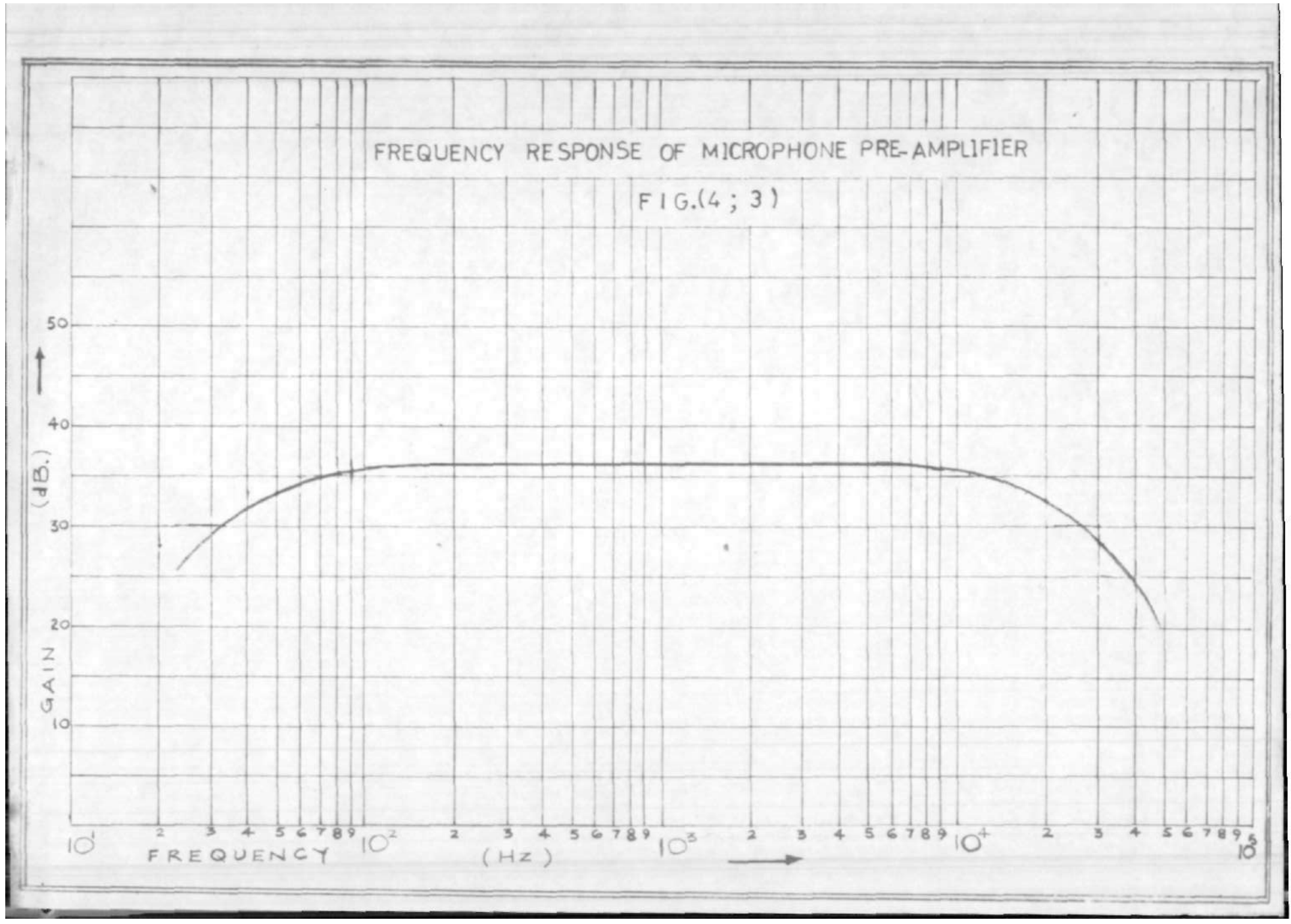


Table (4.2)

Input - Output voltage variations of the Pre-amplifier

(To show the linearity of the Pre-amplifier)

Input voltage	Output voltage at 200Hz	Output voltage at 500Hz	Output voltage at 1000Hz	Output voltage at5000Hz	Output voltage at 10000Hz	Output voltage at15,000Hz
(a.c.)in mV	(a.c.)in mV	(a.c.)in mV	(a.c.)in mV	(a.c.)in mV	(a.c.)in mV	(a.c.)in mV
0.2	8	8	8	8	8	7
0.4	25	25	26	26	25	25
0.6	40	42	43	43	42	42
0.8	54	56	56	57	55	54
1.0	68	72	72	73	72	69
1.2	86	91	90	91	89	85
1.4	105	105	110	110	108	105
1.6	132	134	136	136	134	132

Test Equipment:

Solartron Power Supply Unit. AS 1414 No. P818

C. R. O. Telequipment Type D43 No. 0.896

Howlett Packard 651B Test Oscillator No. G 1439

Howlett Packard Vacuum Tube voltmeter 400H, No. M.1765

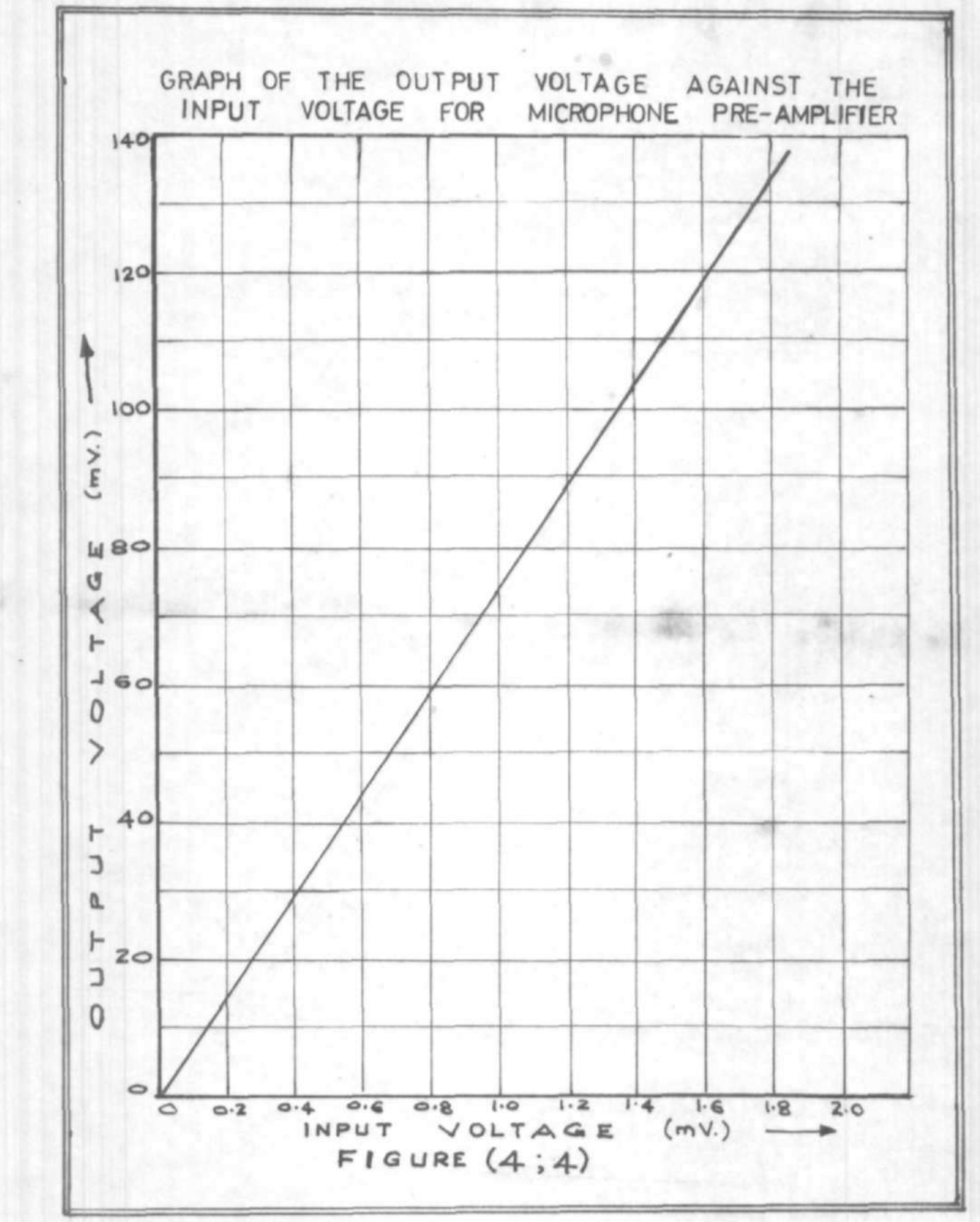

GRAPH OF THE OUTPUT VOLTAGE AGAINST THE INPUT VOLTAGE FOR MICROPHONE PRE-AMPLIFIER
OUTPUT VOLTAGE (mV.)
140
120
100
80
60
40
20
0
0
0·2
0·4
0·6
0·8
1·0
1·2
1·4
1·6
1·8
2·0
INPUT VOLTAGE (mV.)

FIGURE (4;4)

CHAPTER 5

THE VARIABLE GAIN AMPLIFIER

5.1. Specifications

Since it might be necessary to increase or reduce the amplitude of the a.c. signal going from the pre-amplifier to the next stage of the circuit, an amplifier with an adjustable gain control would therefore be needed between the microphone pre-amplifier and the following stage. The variable gain amplifier would be such that:-

(a) its maximum voltage gain is about 32 or 30dB and its minimum voltage gain is 0.

(b) its gain could be controlled to have any value between the maximum and the minimum voltage gains specified.

(c) its frequency response would be flat over a range of about 100Hz to 10KHz.

(d) its output would be free from distortion.

5.2. Choice of Components

A low power transistor would also be needed for this amplifier stage being a low level stage. OC 71 was therefore chosen again. The common emitter configuration was also employed.

5.3. Determination of the Load Resistor R_8

The desired voltage gain of the amplifier = 32

The transistor's current gain h'_{21} = 47

The input impedance h'_{11} of the transistor = 0.8 kΩ

The voltage gain can be expressed as $\frac{h_{21}R_1}{h_{11}}$

$$\therefore \frac{h'_{21}R_1}{h'_{11}} = 32$$

where $R_1 = \frac{R_o \cdot R_8}{R_o + R_8}$; $h'_{21} = 47$; $h'_{11} = 0.8k\Omega$

and R_8 = collector load resistor of the amplifier

R_o = the output impedance of transistor $\approx 1k\Omega$

$\therefore R_1 = \frac{32 \times 0.8}{47} = 0.55\ k\Omega$

$= \frac{R_o R_8}{R_o + R_8} = \frac{R_8}{1 + R_8}$ $(R_o \approx 1k\Omega)$

$\therefore 0.55\ (R_8 + 1) = R_8$

i.e. $0.55 = R_8\ (1 - 0.55)$

$= 0.45\ R_8$

$\therefore R_8 = \frac{0.55}{0.45}\ k\Omega$

$= 1.22\ k\Omega$

$\therefore$ The nearest preferred value of $R_8 = 1.2\ k\Omega$.

5.4. Choice of Operating Point.

If a signal voltage of 80mV was fed to the input of the amplifier the output voltage would be about 2.5V. since the desired maximum voltage gain was 32. Therefore, with a supply voltage V_{cc} of -12V, the operating point was chosen so as to allow an input voltage drive of about 80mV or more; and at the operating point, $V_{cE} > V_{out}$. Where V_{out} was the output voltage from the amplifier. The chosen values were: $V_{cE} = 5V$, $I_{cq} = 3.5mA$

I_{cq} = the collector current at the quiescent operating point.

Also, since the current gain of the transistor is 47 (for grounded emitter configuration),

therefore, for $I_c = 3.5mA$, max. $I_b = \frac{3.5}{47}\ mA$

i.e. $I_b\ (max.) = \frac{3.5}{47} \times 10^3\ \mu A$

$= 73\ \mu A$

A d.c. loadline was drawn and the slope of this gave $\frac{1}{R_8 + R_e}$, where R_e = the emitter resistor.

This is shown in fig.(5;1).

5.5. Determination of the emitter Resistor R_9 (R_e)

Considering the d.c. equation of the amplifier

$$V_{cc} = V_{ce} + R_e I_e + R_c I_c$$
$$= V_{ce} + R_9 I_e + R_8 I_c$$

For small I_b, $I_e = I_c$

$$\therefore V_{cc} = V_{ce} + (R_8+R_9) I_c$$

$V_{cc} = -12$, $I_c = 3.5$ mA, $V_{ce} = 5$V.

$$\therefore 12 = 5 + (R_8 + R_9) \times 3.5$$

$$\text{i.e. } R_8+R_9 = \frac{12-5}{3.5} = \frac{7}{3.5} \text{ k}\Omega$$

$$= 2\text{k}\Omega$$

Similarly, from the d.c. loadline in fig. (5;1), the slope of the graph $= \frac{6}{12} = \frac{1}{2} = \frac{1}{R_c+R_e} = \frac{1}{R_8+R_9}$

$$\therefore R_8+R_9 = 2\text{k}\Omega$$

But $R_8 = 1.2\text{k}\Omega$

$$\therefore R_e = R_9 = (2-1.2)\ \text{k}\Omega = 0.8\text{k}\Omega$$

$\therefore$ The preferred value of $R_9 = 0.68\ \text{k}\Omega$.

5.6. The Biasing Resistors R_6, R_7

The base resistor $R_6 = \frac{R_6 R_7}{R_6 R_7}$

The base voltage of the transistor. V_b is given by

$$V_b = \frac{V_{cc} R_7}{R_6+R_7} - \left(I_e R_9 + I_b \frac{R_6 R_7}{R_6+R_7}\right)$$

Notations for the resistors are as given

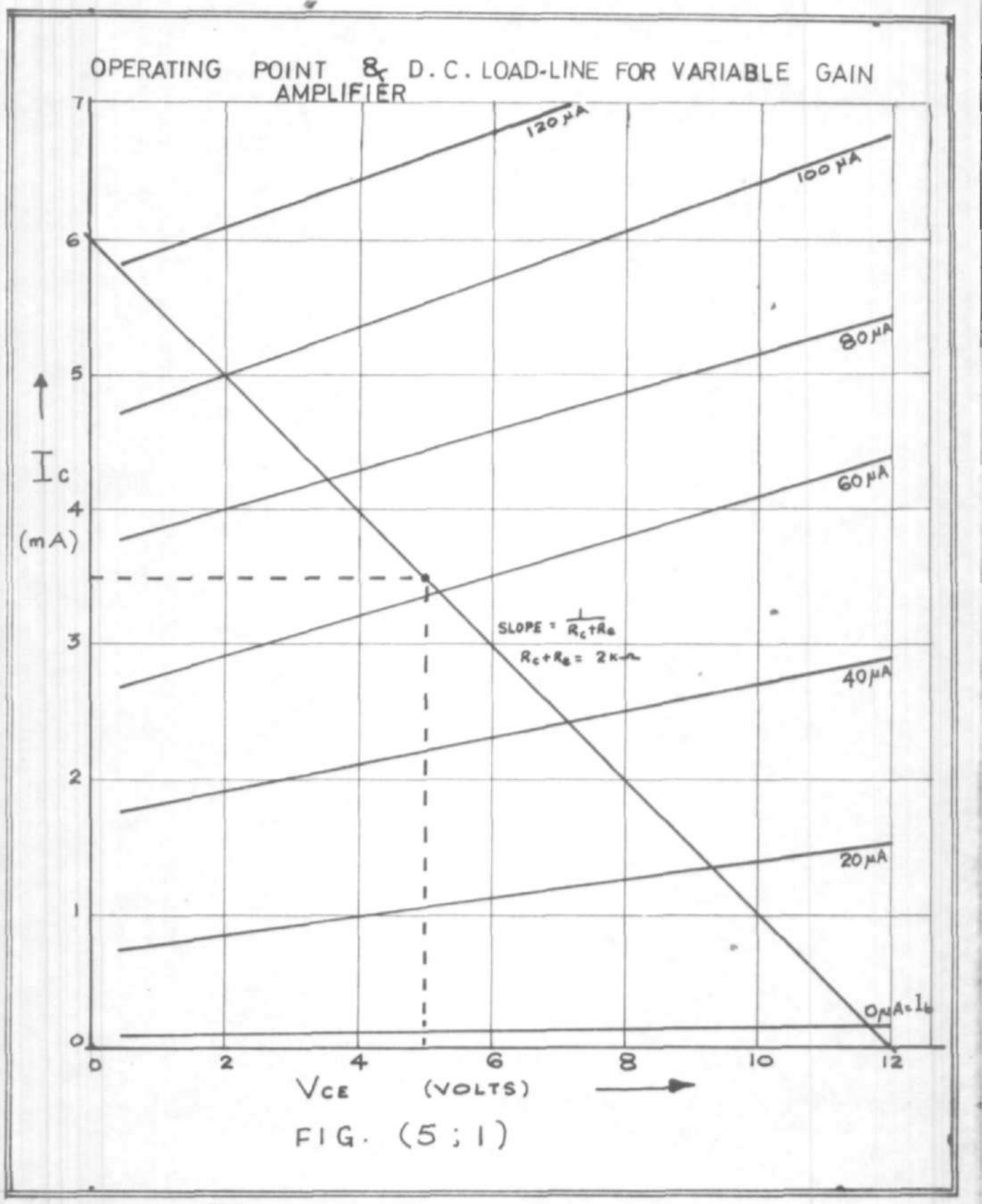

OPERATING POINT & D.C. LOAD-LINE FOR VARIABLE GAIN AMPLIFIER
120 μA
100 μA
80 μA
60 μA
40 μA
20 μA
0 μA = I_b
SLOPE = $\frac{1}{R_c+R_e}$
R_c+R_e = 2K
I_C (mA)
V_{CE} (VOLTS)
0
1
2
3
4
5
6
7
0
2
4
6
8
10
12

FIG. (5 ; 1)

in the circuit diagram in fig. (5;2).

for small I_b, $I_e = I_c$

$$\therefore \frac{V_{cc}R_7}{R_6+R_7} > I_c R_9 = 3.5 \times 0.68 \text{ volts.}$$

$$\therefore \frac{V_{cc}R_7}{R_6+R_7} > 2.38 \text{ volts.}$$

Choosing $\frac{V_{cc}R_7}{R_6+R_7} = 2.4$ V

and also S(stability factor) = 10.

$$\therefore S = \frac{1 + R_e/R_b}{1-\alpha + R_e/R_b} = \frac{1 + R_9/R_b}{1-\alpha+R_9/R_b}$$

Since $\alpha = 0.98$, R_e is designated R_9 and has a value 0.68 kΩ.

$$\therefore S = \frac{1 + R_9/R_b}{0.02 + R_9/R_b}$$

$$\therefore 1 + R_9/R_b = 0.2 + 10\ R_9/R_b$$

$$\therefore R_9/R_b = \frac{0.8}{9} = 0.09$$

But $R_9 = 0.68$ kΩ = 0.7kΩ

$$\therefore R_b = R_9 x 9 = \frac{0.68x9 \text{ k}\Omega}{0.8}$$

$$= \frac{0.7x9}{0.8} = \frac{6.3}{0.8} = 8.\text{k}\Omega$$

Since $\frac{V_{cc}R_7}{R_6+R_7} = 2.4 = \frac{12R_7}{R_6+R_7}$

$$\therefore 9.6\ R_7 = 2.4\ R_6$$

$$\therefore R_6 = 4R_7$$

Substituting this relation in $R_b = \frac{R_6R_7}{R_6+R_7}$

$$\therefore \frac{4R_7^2}{R_7 + 4R_7} = \frac{4}{5} \quad R_7 = 8k\Omega$$

$$\therefore R_7 = \frac{8x5}{4} \quad k\Omega = 10k\Omega$$

$$\therefore \text{Since } R_6 = 4R_7, \quad \therefore R_6 = 40k\Omega$$

The preferred values of R_6 and R_7 were

$$R_6 = 39\ k\Omega$$
$$R_7 = 10\ k\Omega$$

5.7. The Coupling Capacitor C_4

The input impedance of the amplifier = 1kΩ
The base resistance R_6 = 8kΩ
The lowest signal frequency considered = 100Hz
For C_4 to present a short circuit to the a.c. signal going to the base of the amplifier, its reactance,

$$X_{C4} << \frac{1 \times 8}{1 + 8} \quad k\Omega$$

Assuming $X_{C4} = \frac{1}{20} \frac{1 \times 8}{1 + 8} = \frac{1}{wC_4} = \frac{1}{2\Pi x 100 C_4}$

$$\therefore C_4 = \frac{20 \times 9}{2x8\ 10^3 x 10^2} \quad F.$$

$$= \frac{20 \times 9 \times 10^6}{2 \times 8 \times 10^3} \ \mu F$$

$$= \frac{18 \times 10^2}{16} \ \mu F$$

$$= 36 \ \mu F$$

A capacitor of 8 μF was used instead of 36 μF when a 5KΩ potentiometer was connected to the base of the amplifier (as a gain control).

5.8. The Decoupling Capacitor, C_6

The emitter resistor R_9 was decoupled to a.c. signal by the capacitor C_6. Therefore, the reactance of C_6 should be small

compared with the resistance R_9.

i.e. $X_{C6} << R_9$

If $X_{C6} = \frac{R_9}{20}$

Since $X_{C6} = \frac{1}{\omega C_6} = \frac{1}{2\pi f C_6} = \frac{1}{2\pi \times 100 C_6}$

$\therefore \frac{1}{2\pi \times 100 C_6} = \frac{R_9}{20} = \frac{0.68 \times 10^3}{20}$

$\therefore C_6 = \frac{20}{2\pi \times 0.68 \times 10^5}$ F.

i.e. $C_6 = \frac{20 \times 10^6}{2\pi \times 0.68 \times 10^5}$ μF.

$= \frac{10^2}{0.68\pi}$ μF

$= 46 \mu$F.

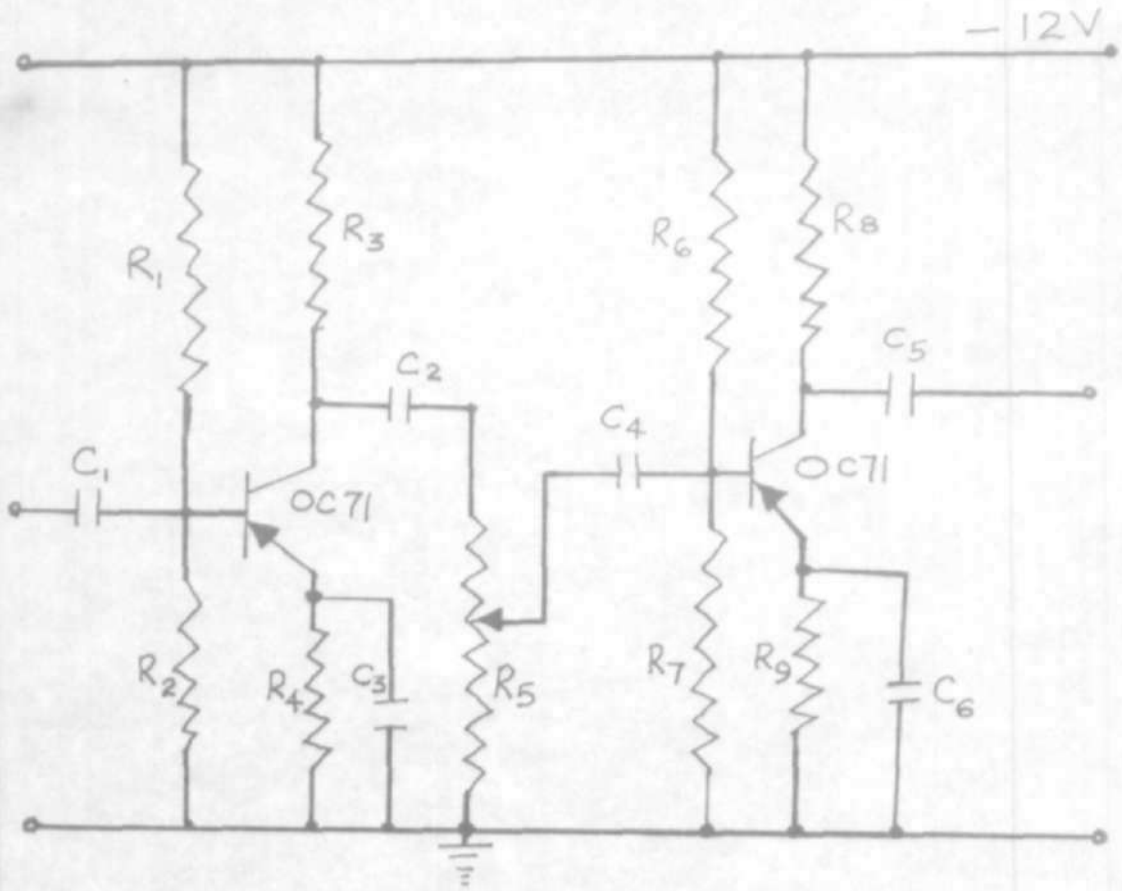
$\therefore$ The nearest preferred value of $C_6 = 50\mu$F. When making adjustments for frequency, this was changed to 100μF.

5.9. Gain Control

A 5kΩ variable potentiometer was connected between the output of the microphone pre-amplifier and the base of the next amplifier stage (variable gain amplifier). The potentiometer could be varied to obtain any gain of the next stage between zero and its maximum value (i.e. about 30 dB). The over-all gain of the two amplifier stages would consequently be controlled by varying the potentiometer. This potentiometer was coupled with the output of the microphone pre-amplifier by a capacitor C_2 of magnitude 8μF as shown in the circuit diagram in fig. (9;1)

FIG. (5;2)

PRE-AMPLIFIER & VARIABLE GAIN AMPLIFIER TOGETHER



5.10 Tests For The Variable - Gain Amplifier

Two tests were performed for the amplifier before connecting it to the gain control network. The first was to show the frequency response of the amplifier while the other was to show its linearity After these tests, another one was then conducted for the frequency response of the amplifiers coupled together (i.e. the pre-amplifier and variable gain amplifier together.) as shown in fig. (5.4).

(a) Test for the frequency response of the amplifier.

This was performed by introducing an input a.c. signal voltage of 10mV. to the base of the amplifier. The signal voltage was obtained from the Hewlett Packard 65 IB Test Oscillator No.G. 1439. This voltage was kept constant at all the frequencies considered. The input voltage and the corresponding output voltage were measured by means of the Hewlett Packard Vacuum tube voltmeter 400H and the output was also displayed on the C.R.O. Teloquipment Type D34, No. 0.896 to know the onset of distortion.

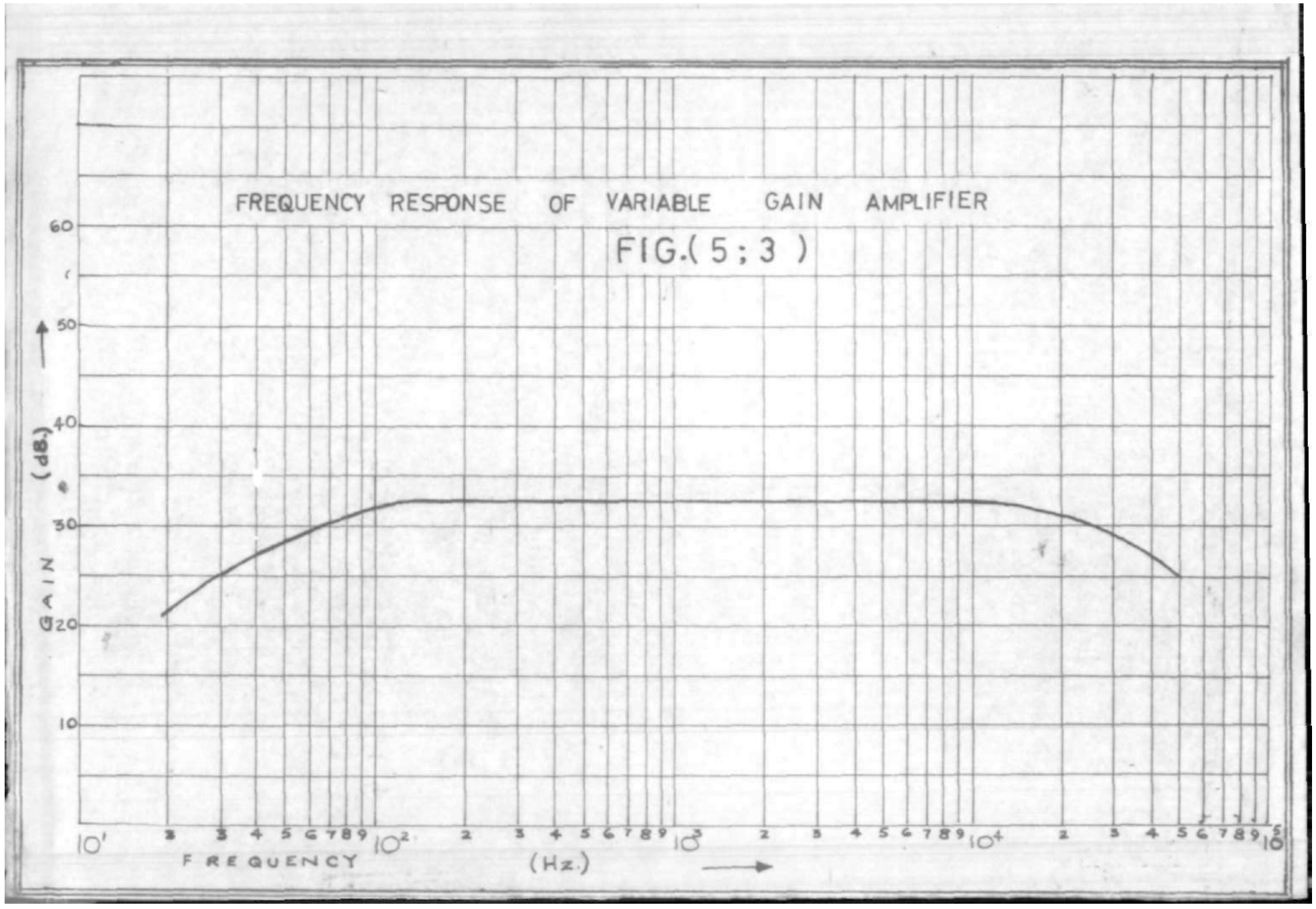
The readings for the variation of the output voltage with frequency for a constant input voltage of 10mV. were then tabulated as shown in Table (5:1). The gain in dB of the amplifier for each frequency considered was calculated and recorded.

A graph of the gain (in dB) against frequency (in Hz) was plotted and this is shown in fig. (5:3).

From this graph, it was observed that the amplifier had a maximum gain of 33.6 dB and it had a flat frequency response within a range of about 80Hz to 20KHz.

(b) Linearity

The linearity of the amplifier was investigated by measuring the corresponding output voltages for different values of input voltages. The measurement was carried out at certain chosen


FREQUENCY RESPONSE OF VARIABLE GAIN AMPLIFIER
FIG.(5 ; 3)
GAIN (dB.)
60
50
40
30
20
10
10'
2
3
4
5
6
7
8
9
10²
10³
10⁴
10⁵
FREQUENCY (Hz.)

frequencies in the audiofrequency range. These frequencies were: 200 Hz; 500Hz; 1KHz; 5K.Hz. 10K.Hz; and 15K.Hz. The readings obtained were tabulated as shown in Table (5;2). Using these readings, graphs of output voltage against the input voltage were drawn for the frequencies considered. These graphs are shown in fig. (5;5).

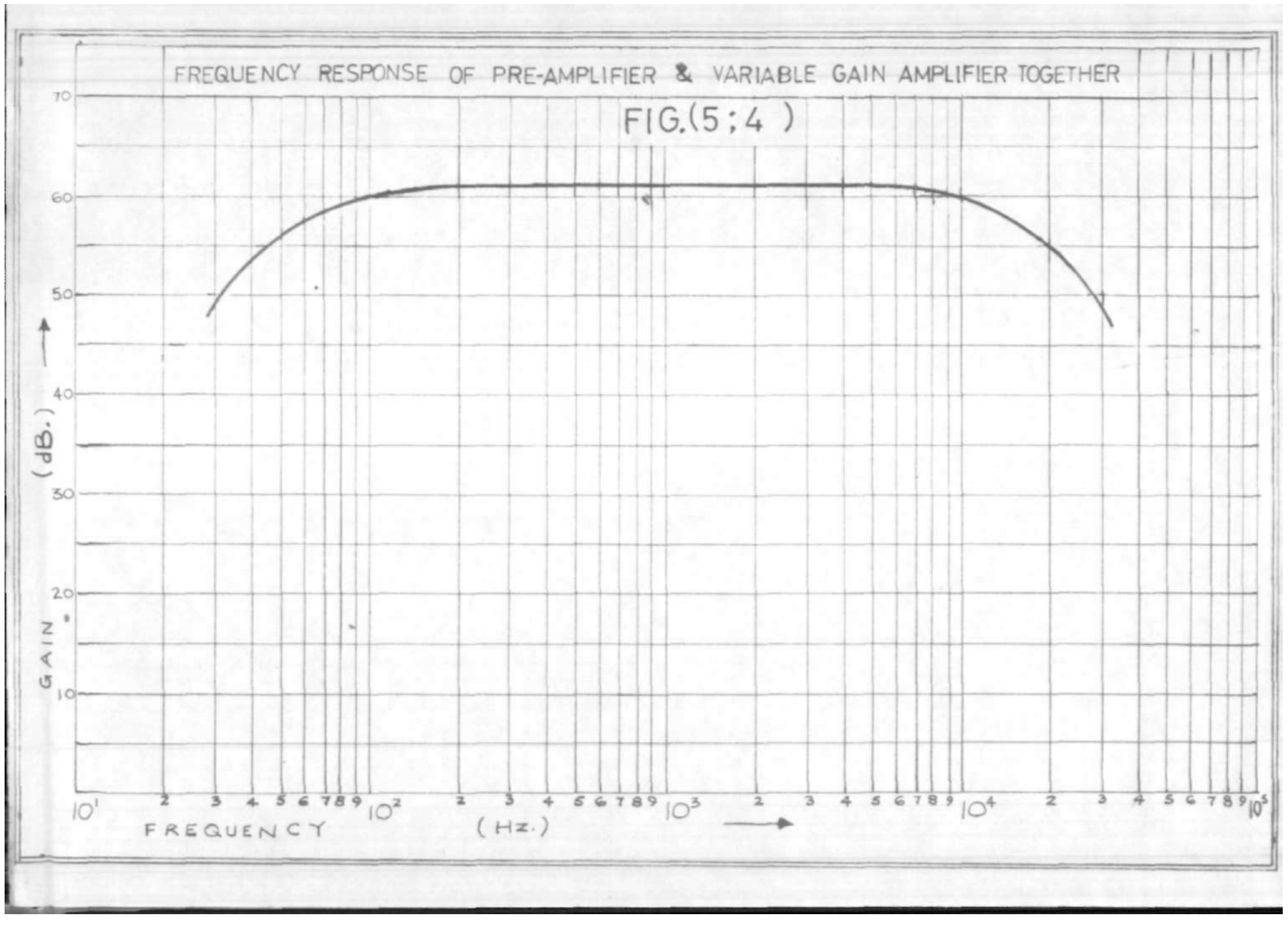
From the graphs, it was observed that there was a linear relationship between the input and output voltages of the amplifier. Two linear graphs were obtained; one representing the input - output voltage variation at 200 Hz and 15,000 Hz and the other representing the input - output voltage variation at a frequency range between 500 Hz and 10 KHz.

The input and output voltages were measured by using the Hewlett Packard Vacuum Tube Voltmeter 400 H No. M 1765.

(c) Frequency Response of the Pre-amplifier and the Variable Gain amplifier combined.

After coupling the pre-amplifier to the base of the variable gain amplifier through a 5 kΩ potentiometer (used as a volume or gain control) the frequency response of the combination was investigated. This was carried out by feeding a constant input voltage of 2mV. to the input of the pre-amplifier. With the gain control at the position of maximum gain the output voltage was measured from the variable gain amplifier. Care was taken to make sure that the input voltage was constant at all the frequencies considered. The output voltages were measured for various values of frequency between 20 Hz and 50,000 Hz. The Hewlett Packard Vacuum Tube Voltmeter 400 H No. M 1765 was also employed in measuring the voltages. The output was also connected to the C.R.O., D 43 (Telequipment) to be able to detect any on-set of distortion. The signal generator used was the Hewlett Packard 65 1B Test Oscillator No. G 1439. The readings for the frequencies and the output voltages were tabulated as shown in Table (5;3). The gains were calculated in dB. and a graph of the voltage gain in dB. was plotted against frequencies (in logarithmic Scale) and this

is displayed in fig.(5·4). From this graph it was observed that the maximum gain of the combination of the two amplifiers was 62 dB and that a flat frequency response was obtained between 100Hz and 10KHz.


FREQUENCY RESPONSE OF PRE-AMPLIFIER & VARIABLE GAIN AMPLIFIER TOGETHER
FIG.(5;4)
GAIN (dB.)
70
60
50
40
30
20
10
FREQUENCY (Hz.)
10¹ 2 3 4 5 6 7 8 9 10² 2 3 4 5 6 7 8 9 10³ 2 3 4 5 6 7 8 9 10⁴ 2 3 4 5 6 7 8 9 10⁵

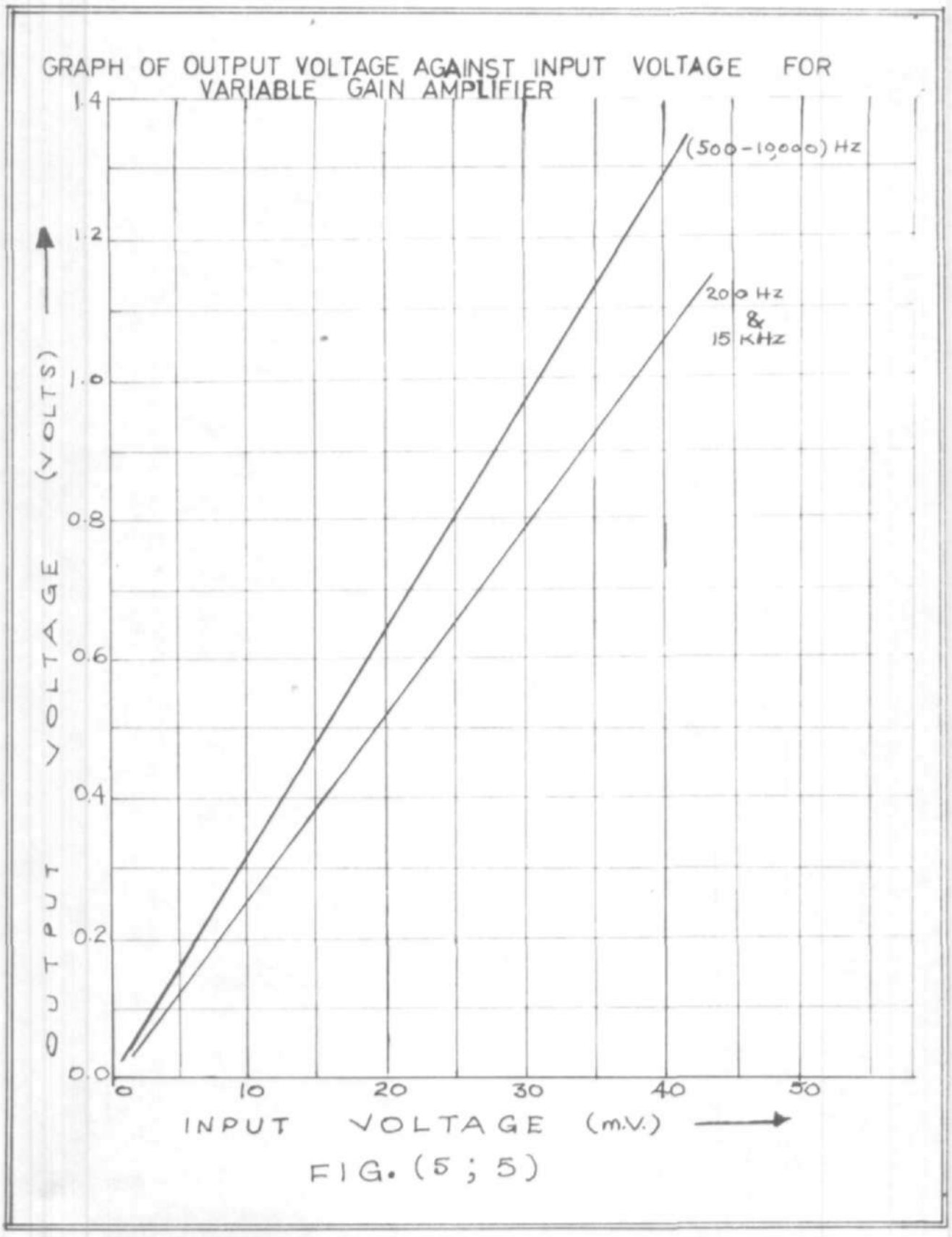

GRAPH OF OUTPUT VOLTAGE AGAINST INPUT VOLTAGE FOR VARIABLE GAIN AMPLIFIER
(500-10,000) Hz
200 Hz & 15 KHz
OUTPUT VOLTAGE (VOLTS)
1.4
1.2
1.0
0.8
0.6
0.4
0.2
0.0
0
10
20
30
40
50
INPUT VOLTAGE (m.V.)

FIG. (5 ; 5)

Table (5:1)

Variable Gain Amplifier Frequency Response.

Frequency in Hz	Input Voltage (a.c.) in m.V.	Output Voltage (a.c.) in m.V.	Voltage Gain in dB
20	10	120	21.6
50	,,	270	28.6
70	,,	330	30.4
100	,,	390	32.0
200	,,	450	33.0
300	,,	470	33.4
400	,,	480	33.6
500	,,	480	33.6
700	,,	480	33.6
1,000	,,	480	33.6
2,000	,,	480	33.6
5,000	,,	480	33.6
7,000	,,	470	33.4
10,000	,,	440	32.8
15,000	,,	400	32.0
20,000	,,	350	31.0
30,000	,,	280	29.0
40,000	,,	220	26.8
50,000	,,	180	25.0

Test Equipment.

Solartron Power Supply Unit AS 1411 No P818

C. R. O. Telequipment Type D43 No. 0896

Hewlett Packard 651B Test Oscillator No. G1439

,, ,, Vacuum Tube Voltmeter 400H No M1765.

Table (5.2)

Input - Output voltage variation of the variable gain amplifier
(to show the linearity of the amplifier.)

INPUT VOLTAGE in m.V.(a.c.)	OUTPUT VOLTAGE at 200Hz (a.c.)in Volts	at 500Hz (a.c.)in Volts	at 1KHz (a.c.)in Volts	at 5KHz (a.c.in Volts	at 10KHz (a.c.)in Volts	at 15KHz (a.c.) in Volts
5	0.16	0.19	0.20	0.20	0.20	0.20
10	0.26	0.32	0.33	0.34	0.34	0.42
15	0.40	0.48	0.50	0.51	0.51	0.57
20	0.54	0.64	0.67	0.68	0.67	0.57
25	0.67	0.80	0.82	0.82	0.81	0.69
30	0.79	1.00	1.00	1.00	1.00	0.79
35	0.94	1.15	1.15	1.15	1.15	1.00
40	1.06	1.25	1.30	1.30	1.30	1.10

Test Equipment:-

Solartron Power Supply Unit As 1414 No. P818

C. R. O. Telequipment Type D43 No. 0896

Hewlett Packard 651E Test Oscillator No. G1139

,, ,, Vacuum Tube Voltmeter 400H No. M1765

Table (5·3)

Frequency Response of the Pre-amplifier and Variable Gain amplifier together

FREQUENCY IN Hz	INPUT VOLTAGE (a.c.) in nV.	OUTPUT VOLTAGE (a.c.) in nV.	VOLTAGE GAIN in dB
20	2	300	43.6
50	,,	1300	56.2
70	,,	1700	58.6
100	,,	2000	60.0
200	,,	2400	61.6
300	,,	2400	61.6
400	,,	2500	62.0
500	,,	2500	62.0
700	,,	2500	62.0
1,000	,,	2500	62.0
2,000	,,	2500	62.0
5,000	,,	2500	62.0
7,000	,,	2400	61.6
10,000	,,	2000	60.0
15,000	,,	1500	57.6
20,000	,,	1100	54.8
30,000	,,	600	49.6
40,000	,,	400	46.0
50,000	,,	200	40.0

Test Equipment:-

Solartron Power Supply Unit AS 1414 No. P818

C. R. O. Telequipment Type D43 No. 0896

Hewlett Packard 651B Test Oscillator No. G. 1439

,, ,, Vacuum Tube Voltmeter 400H, No. K.1765

CHAPTER 6

THE LOGARITHMIC AMPLIFIER

6.1. Specifications

The combination of a logarithmic network and a linear amplifier formed the logarithmic amplifier. It was necessary to couple the linear amplifier to the output of the logarithmic network because the signal at the output of the logarithmic network would be very small compared with the input signal, and it would therefore need amplification.

The logarithmic network should be such that its output voltage should be a logarithm of its input voltage.

Also, the linear amplifier to be coupled to the output of this logarithmic network should be such that:-

(i) it would permit an input voltage drive of about 200mV or more.

(ii) its voltage gain should be about 32 dB.

(iii) its output should be a faithful reproduction of the input voltage (i.e. there should be no distortion.)

6.2. The Logarithmic Network (Theory)

A simple logarithmic circuit using a semiconductor diode is shown in fig. (6.1). Considering the diode to be ideal, therefore, the value of the current I flowing through the diode is given by the equation:-

$$I = I_s \left[\exp\left(\frac{q V_j}{KT}\right) - 1 \right]$$

where I_s = Diode leakage current

if e = the electron charge, q = Total charge of electrons

V_j = the voltage across the ideal diode

K = Boltzmann's constant

T = absolute temperature

I = Current flow through the diode.

From the previous equation, we have:-

$$\frac{I}{I_s} = \exp\left(\frac{qV_J}{KT}\right) - 1$$

$$\text{i.e. } \frac{I + I_s}{I_s} = \exp. \frac{(qV_J)}{KT}$$

$$\therefore \frac{qV_J}{KT} = \text{In.} \left(\frac{I + I_s}{I_s} \right)$$

$$\therefore V_J = \frac{(KT)}{(q)} \text{ In } \left(\frac{I + I_s}{I_s} \right)$$

Therefore, the value of the voltage V_J across the ideal diode is given by

$$V_J = \frac{(KT)}{(q)} \text{ In } \left(\frac{I + I_s}{I_s} \right)$$

But considering a real diode and taking into account its ohmic resistance R_D, the actual voltage across the diode would be larger by the amount of the ohmic voltage drop IR_D. Therefore, the former equation can be rewritten for the voltage V_D across a real diode as:-

$$V_D = \left(\frac{KT}{q}\right) \text{ In } \left(\frac{I + I_s}{I_s} \right) + IR_D$$

From this equation, it can be seen that a logarithmic range may be obtained when I is much larger than I_s and IR_D is small compared with V_D.

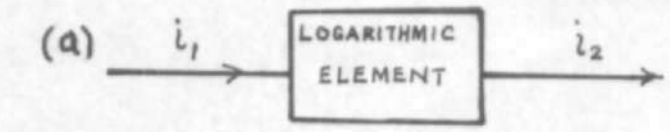
Thus V_D can be expressed as:

$$V_D = a \log I/I_s$$

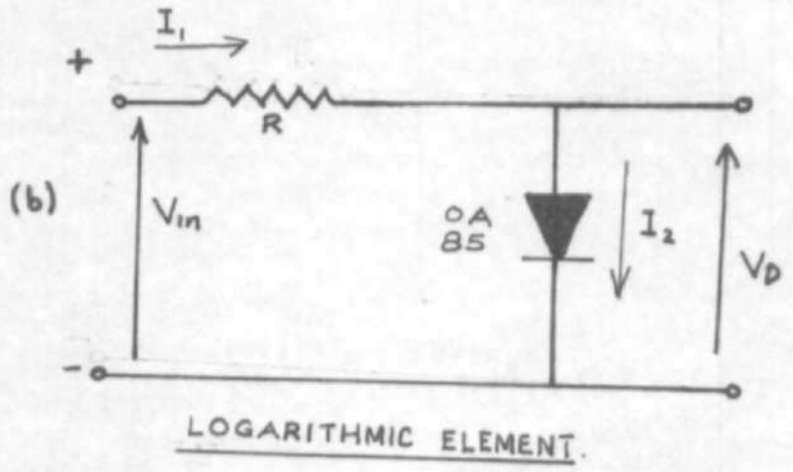
$$I = \frac{V_{in} - V_D}{R}$$

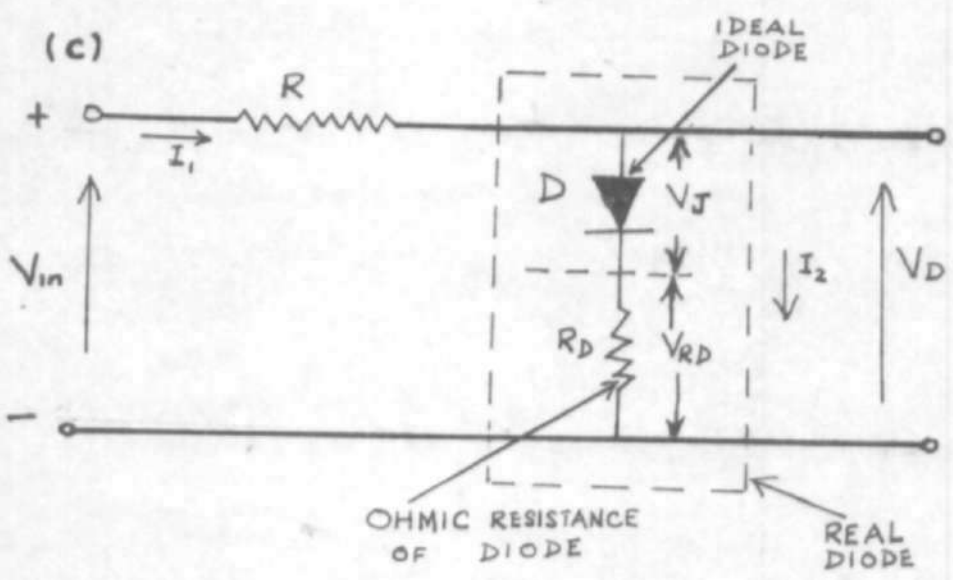
$$\therefore a \log \left(\frac{V_{in} - V_D}{I_s R} \right) = V_D$$

LOGARITHMIC NETWORK.



$$i_2 = k \log i_1 + C$$





LOGARITHMIC ELEMENT EQUIVALENT DIAGRAM

FIG. (6;1)

6.3. Choice of Components for the Logarithmic Element.

To obtain the logarithmic element, a resistor was connected in parallel with a diode biased in in the forward direction as shown in fig. (6;1b). A germanium diode OA85 was used. The value of the resistor employed was determined as a result of a certain test performed to know which value was suitable.

6.4. Test for a Suitable Value of Resistor

For this test, resistors R = IKΛ and R = 10KΛ were selected. The 1KΛresistor was first connected in the logarithmic circuit as shown in fig. (6,1b). A solartron Power Supply unit, AS1414, No. P818 was set to 10volts, D.C. and the voltage was fed to the input of the logarithmic element. A 5KΛ potentiometer was used to control the voltage applied to the diode. Both the applied voltage and the voltage across the diode were measured using the Hewlett Packard Vacuum Tube Voltmeter Model 410B, No. M 1773. The readings obtained for both the applied voltage V_{in} and the voltage across the diode, V_D were tabulated as shown in Table (6,1). The same process was repeated when the 10KΛresistor was connected in the circuit.

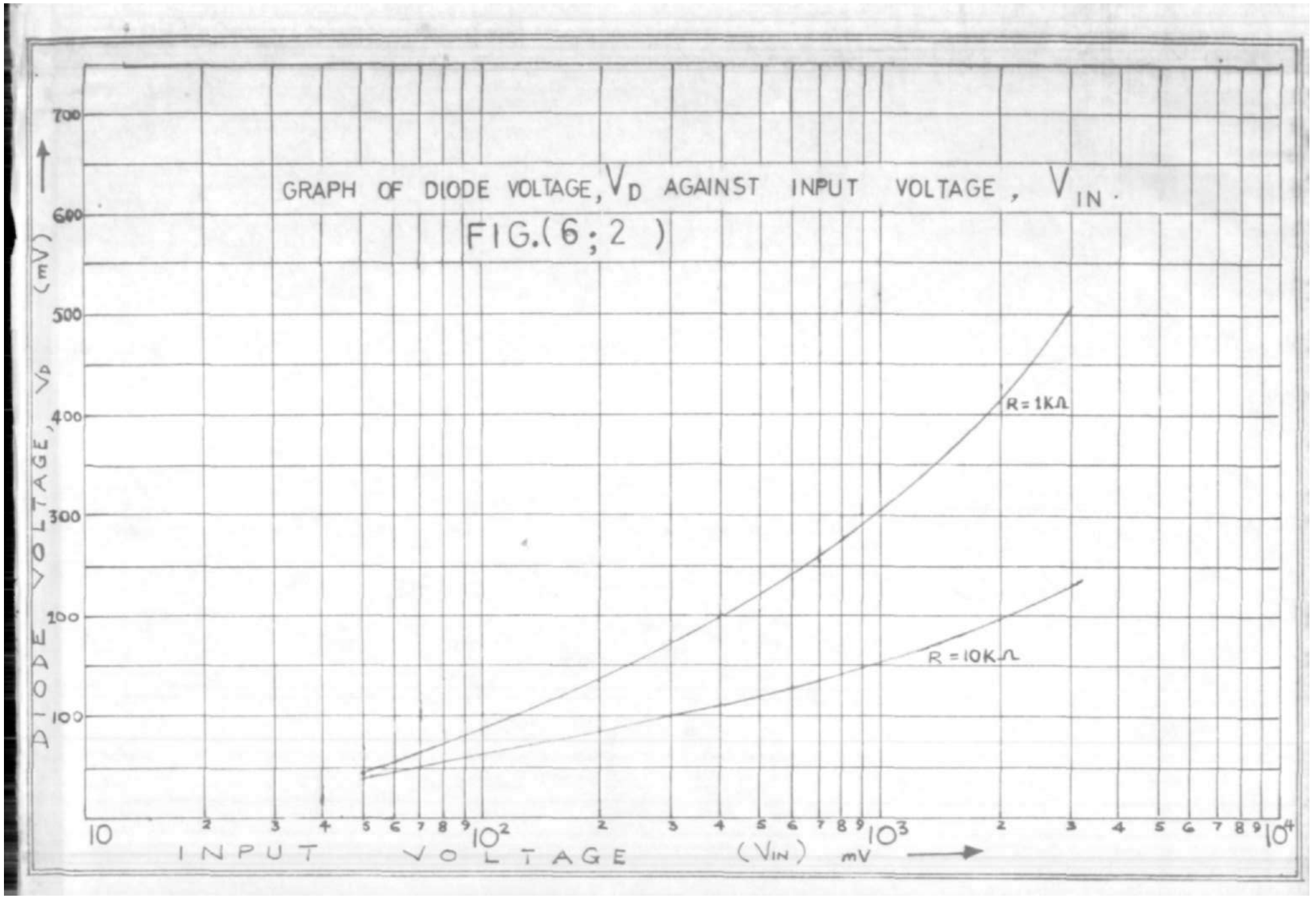
A graph of V_D against V_{in} (in log. scale) was plotted for each value of resistor used and the graphs are shown in fig.(6,2). It was observed that the graphs showed a linear relationship between the voltage across the diode, V_D and the applied voltage, V_{in} (in log. scale) over different ranges for the two cases. But for the linear section of each of the graphs, the equation $V_D = K \log (V_{in}) + C$ is satisfied.
where K and C are constants.

For small values of C.
$V_D = K \log. V_{in}$.

When the 1KΛ resistor was used in the circuit, the range of $V_D$over which the logarithmic relation was satisfied = (80 - 200)mV. While for the 10KΛ resistor, the range was = (50 - 130) mV. The 1KΛ resistor was therefore, selected because it could be used to obtain a wider range of the

GRAPH OF DIODE VOLTAGE, V_D AGAINST INPUT VOLTAGE, V_{IN}.

FIG.(6;2)



voltage V_D across the diode than the 10 kΩ resistor.

Diode Measurements

Table (6.1)

1 KΩ RESISTOR

APPLIED VOLTAGE V_{IN} (mV)	VOLTAGE ACROSS DIODE V_D (mV)
50	45
100	88
200	138
300	172
400	200
600	241
800	275
1000	305
1500	365
2000	415
3000	506

10 KΩ RESISTOR

APPLIED VOLTAGE V_{IN} (mV)	VOLTAGE ACROSS DIODE V_D (mV)
50	38
100	55
200	85
300	100
400	112
600	130
800	145
1000	155
1500	180
2000	198
3000	230

6.5. The Linear Amplifier

The linear amplifier following the logarithmic network should be an amplifier which could handle greater power than the previous amplifiers. Therefore, a power amplifier was necessary for this amplifier stage to be able to cope with the high value of collector current.

6.6. Choice of Class of Operation for the Power Amplifier Stage

Power amplifiers can be grouped into four main classes depending upon their operation, viz:-

(i) Class A operation
(ii) Class B ,,
(iii) Class AB ,,
(iv) Class C ,,

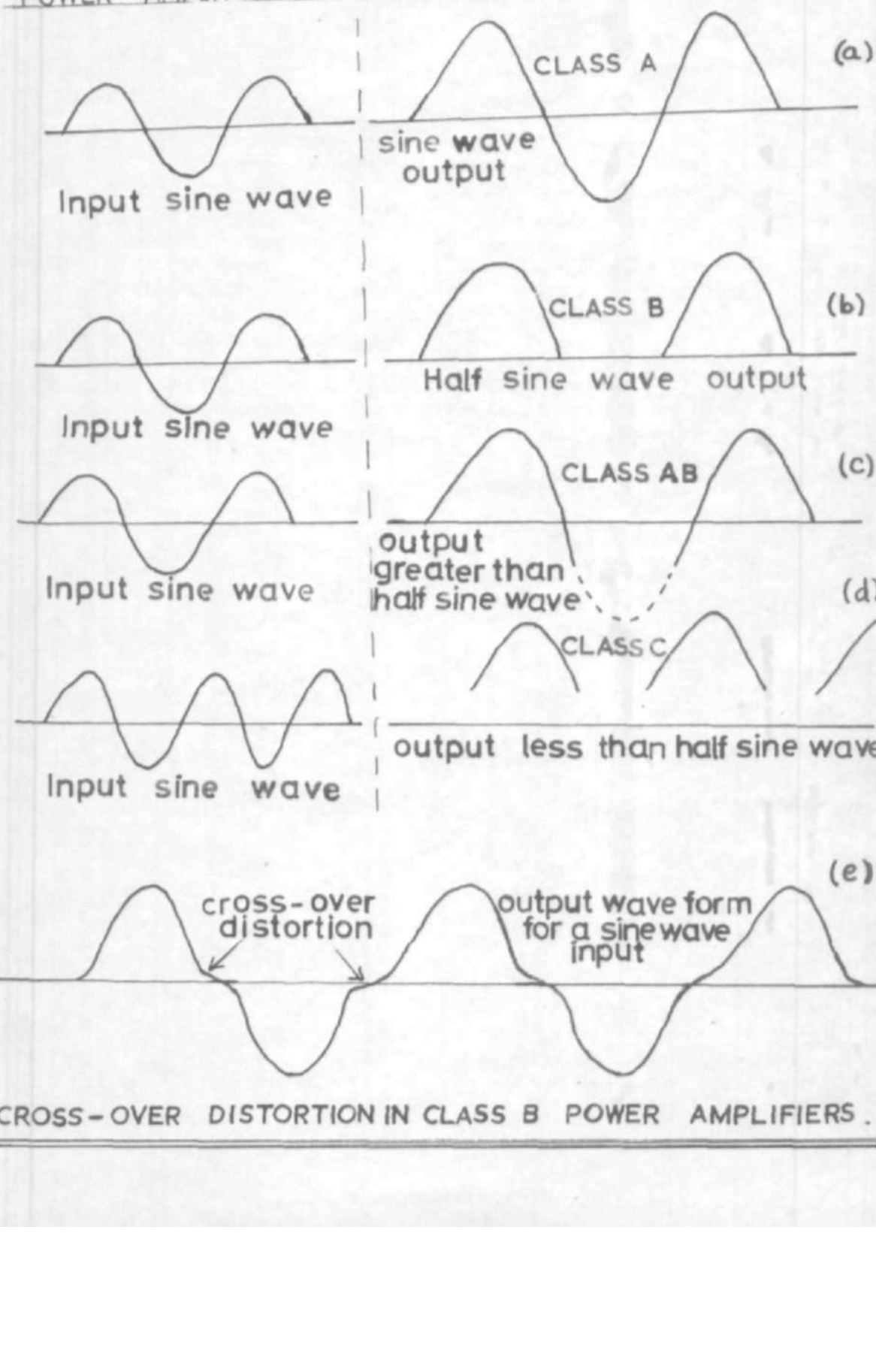
In class A operation, the amplifier has its operating point in the active region. The entire input signal is thus reproduced faithfully at the output. This type of operation offers the lowest efficiency. The power delivered by the supply is constant and it is not affected by the input signal; i.e. power may be dissipated by the circuit even if no signal is present. Distortion is not very significant in this case. The distortion in the output waveshape due to non-linearity of characteristics is tolerable

In class B operation, the operating point is in the cut-off region. The device thus conducts over one half of the entire input signal cycle so that in order to amplify successfully a sinusoidal signal, two transistors are necessary each working on successive alternations. The class B circuits are very efficient power amplifiers. The efficiency is about 78.5%. On the other hand, class B operation introduces considerable distortion which may result from (a) the non-linearity of the I_c / I_b transfer characteristic, (b) Cross-over distortion (as shown in fig.(6·3c)) and (c) the mismatch of the two transistors.

In class AB operation, the amplifiers have output waveforms which are not a full replica of the inputs.

FIG. (6;3)

POWER AMPLIFIERS INPUT AND OUTPUT WAVE-FORMS.



CROSS-OVER DISTORTION IN CLASS B POWER AMPLIFIERS.

In this case, the operation is between class A and class B. The output voltage is therefore bigger than $\frac{1}{2}$ sine wave for an input which is a sine wave (as shown in fig. (6.3 c)). This type of operation eliminates the cross-over distortion which is present in class B amplifiers. The distortion is not therefore as significant as in the case of class B amplifiers.

In class C operation, the transistor is biased well below cut-off. The output voltage is less than half sine wave for a sine wave input. (as shown in fig. (6.3d). Cross-over distortion is even more noticable in this case than in the Class B case.

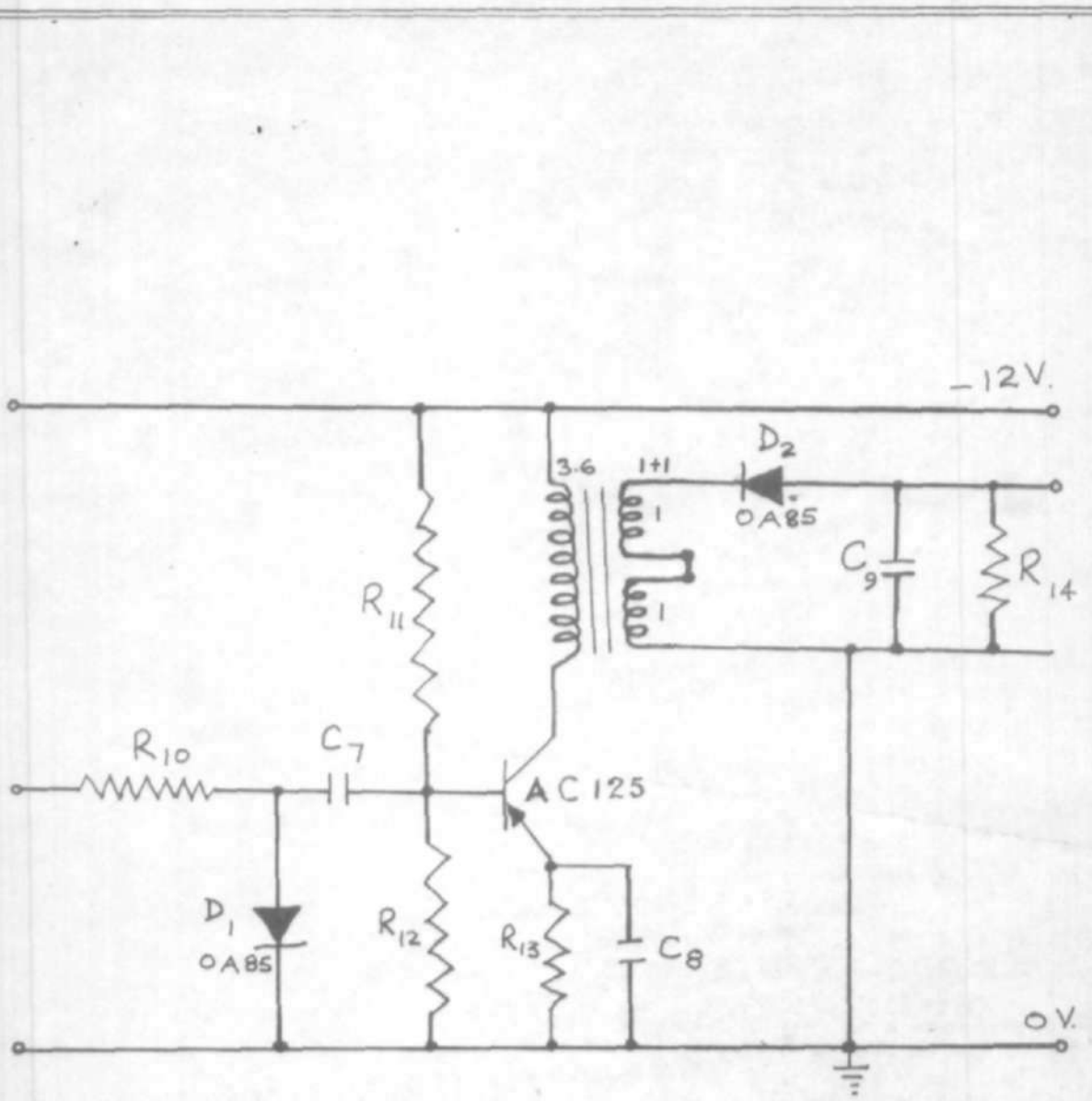
Class A operation was employed because it would give a faithful reproduction of the input signal at the output. Its distortion is low and it is also simple to design.

6.7. Design Procedure

In the design of the class A power amplifier, a transistor AC 125 was used. Its maximum power dissipation at 25°C is 500 mW.

Transformer coupling was used because the small d.c. losses in the transformer would enable the realization of a relatively high conversion efficiency. A transformer type T/T3 with turns ratio 3.6·1+1 (1:8·1) was employed and this was connected to the collector of the transistor AC.125 as shown in fig. (6.4). The maximum a.c. voltage presented to the base of the amplifier at the output of the logarithmic network was 200 mV. and the desired maximum output voltage from this amplifier stage = 10 volts. The supply voltage obtained from the solartron Power Supply unit AT 1414 No. P818 was - 12 volts. Therefore, using an emitter resistor $R_e = R_{13} = 62\Omega$, the operating point was chosen such that V_{ce} = 10volts and the quiescent collector current I_{cq} = 15mA. Using h'_{21} (current gain) for the transistor = 75

$$\therefore \quad I_b \text{ max} = \frac{I_c}{h_{21}} = \frac{15}{75} \text{ mA} = 200\mu\text{A}.$$



THE LOGARITHMIC AMPLIFIER

FIG. (6;4)

From the dynamic loadline, R_1, the a.c. resistance at the primary of the transformer was obtained to be $R_1 = 2.1K\Omega$

$\therefore$ The effective secondary resistance $= \frac{2.1}{n^2}$ where n = 1.8.

$\therefore$ The effective secondary resistance $= \frac{2.1}{(1.8)^2} K\Omega$

$= 0.68K\Omega = 680\Omega$

The quiescent collector power dissipation is given by

$$P = I_{cq} \times V_{ce}$$

$$= 15 \times 10mW$$

$$= 150 mW$$

Since the maximum theoretical efficiency of a class A power amplifier stage is 50%, therefore, the maximum power output of this stage = 75 mW.

6.8 The Biasing Components R_{11} and R_{12}

The base resistance R_b of the amplifier could be written as:-

$$R_b = \frac{R_{11}R_{12}}{R_{11}+R_{12}}$$

where R_{11}, R_{12} are the biasing resistors.

If the stability factor S = 5,

$$\therefore S = \frac{1 + R_c/R_b}{1 - \alpha + R_c/R_b} = \frac{1 + R_{13}/R_b}{1 - \alpha + R_{13}/R_b}$$

For $\alpha = 0.98$,

$$\therefore S = 5 = \frac{1 + R_{13}/R_b}{1 - 0.98 + R_{13}/R_b} = \frac{1 + R_{13}/R_b}{0.02 + R_{13}/R_b}$$

i.e. $0.1 + 5R_{13}/R_b = 1 + R_{13}/R_b$. $\therefore \frac{R_{13}}{R_b} = 0.022$

But $R_c = R_{13} = 0.062K\Omega$

$$\therefore R_b = \frac{R_{11}R_{12}}{R_{11}+R_{12}} = \frac{0.062}{0.022} = 3K\Omega$$

But $V_b = \frac{V_{cc}R_{12}}{R_{11}+R_{12}} - \left\{I_c R_{13} + I_b \frac{R_{11}R_{12}}{R_{11}+R_{12}}\right\}$

where V_b = the base voltage of the amplifier and R_{11}, R_{12} are the biasing components of the amplifier as shown in fig.(6.5).

For small I_b, $I_e = I_c$

$\therefore \frac{V_{cc}R_{12}}{R_{11}+R_{12}} > I_c R_{13} = 0.062 \times 15 = 1.0$ volt.

$\therefore$ If $\frac{V_{cc}R_{12}}{R_{11}+R_{12}} = 2$ volts.

$$12R_{12} = 2R_{11} + 2R_{12}$$

$$\text{i.e. } 2R_{11} = 10R_{12}$$

$$\text{or } R_{11} = 5R_{12}$$

But $\frac{R_{11}R_{12}}{R_{11}+R_{12}} = 3.$

$\therefore \frac{5R_{12}^2}{5R_{12}+R_{12}} = \frac{5}{6} R_{12} = 3K\Omega$

$\therefore 5R_{12} = 18K\Omega = R_{11}$

and $R_{12} = \frac{18}{5}K\Omega = 3.6K\Omega$

The nearest preferred value of R_{12} is given by $3.3K\Omega$

Also, the preferred value of R_{11}

$= 15K\Omega$

$\therefore R_{11} = 15K\Omega$

$R_{12} = 3.3K\Omega$

6.9. The Coupling Capacitor C_7

C_7 was used to couple the logarithmic network to the base of the following amplifier stage. The input impedance of the amplifier = $1K\Omega$. The base resistance $R_b = \frac{R_{11}R_{12}}{R_{11}+R_{12}} = 3K\Omega$

$\therefore$ For C_7 to present a short circuit to the a.c. signal going from the logarithmic network to the base of the amplifier, its reactance

$$X_{C7} << \frac{1 \times R_b}{1 + R_b}$$

i.e. $X_{C7} << \frac{1 \times 3}{1 + 3}$ KΩ

$\therefore$ If $X_{C7} = \frac{1}{20} \cdot \frac{1 \times 3}{1 + 3}$ KΩ

$\therefore$ $X_{C7} = \frac{1}{wC_7} = \frac{1}{2\pi fC_7} = \frac{3}{80}$ KΩ

where f = the lowest frequency considered = 100Hz

$\therefore$ $\frac{1}{2\pi \times 10^2 C_7} = \frac{3}{80}$

$\therefore$ $C_7 = \frac{80}{2\pi \times 3 \times 10^2 \times 10^3}$ F

$= \frac{80}{6} \times \frac{10^6}{10^5} \mu$F

$= 43 \mu$F.

$\therefore$ The nearest preferred value of $C_7 = 50 \mu$F.

6.10 The Decoupling Capacitor C_8

For C_8 to decouple the emitter resistor R_9 to a.c. signal, the reactance of C_8 should be small compared with R_9.

i.e. $X_{C_8} << R_9$

But $R_9 = 62\ \Omega$

$\therefore$ If $X_{C_8} = 6.2\ \Omega = \frac{1}{wC_8} = \frac{1}{2\pi fC_8}$

But f = 100Hz.

$\therefore$ $C_8 = \frac{1}{2\pi f \times 6.2}$ F $= \frac{1}{2\pi \times 6.2 \times 10^2}$ F

$$\therefore C_8 = \frac{10^6}{1.24 \times 10^3}\ \mu F = 250\,\mu F$$

This was later changed to 100 μF. when making adjustment for frequencies.

6.11 Test for Logarithmic Amplifier

Two main tests were performed for the logarithmic amplifier after construction. One was to investigate the frequency response of the amplifier and the other was to show the logarithmic characteristic of the amplifier.

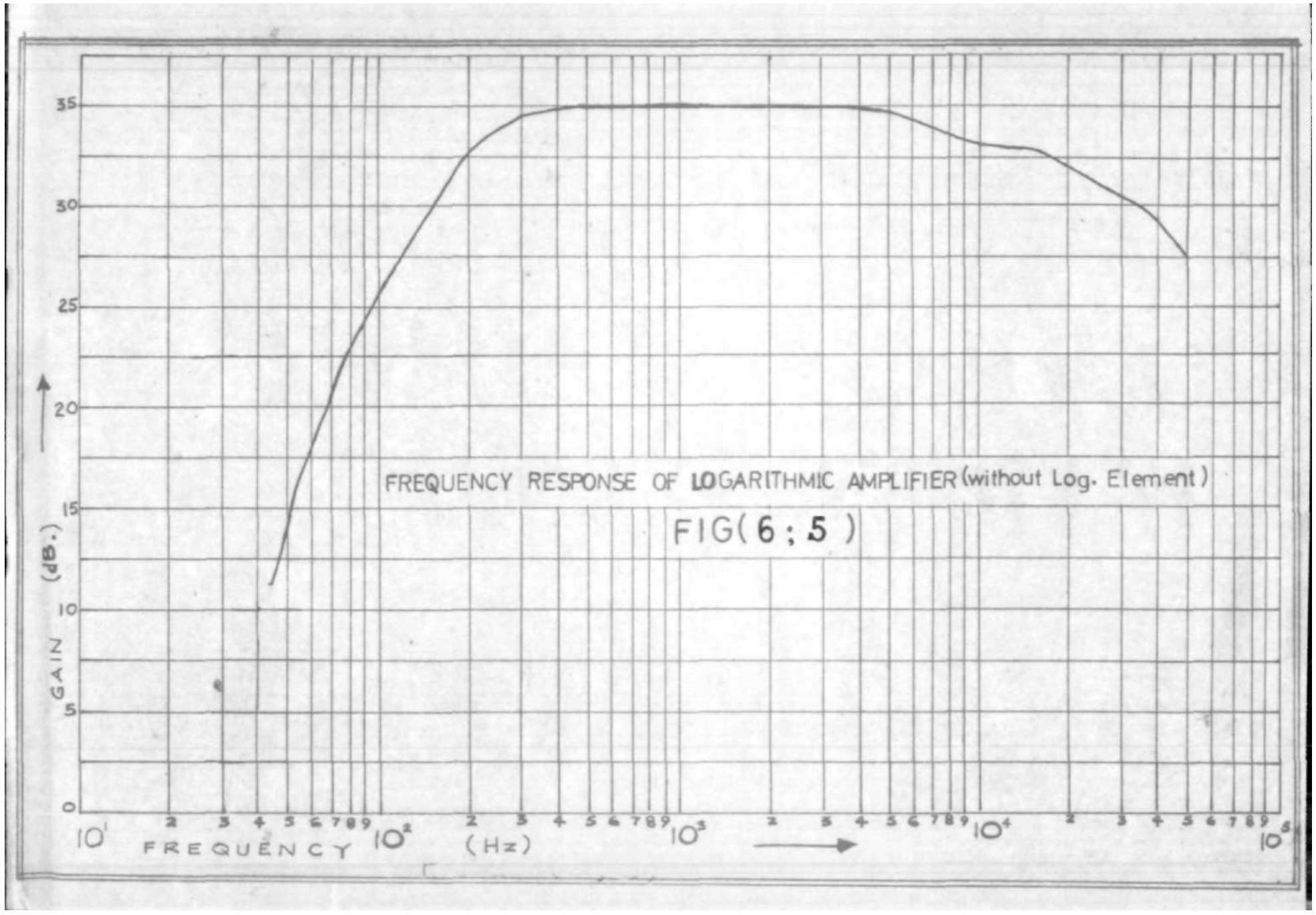
(a) Frequency Response of the Amplifier (without the log. element)

The frequency response of the amplifier without the logarithmic element was first investigated. An input signal voltage of 0.1 volt from the Hewlett Packard 65 1B test Oscillator No.G 1439 was introduced to the base of the amplifier and it was kept constant at all frequencies considered.

The corresponding output voltages for various values of frequency between 30Hz and 50Hz were measured using the Hewlett Packard Vacuum Tube voltmeter 400H, No. M.1765. The C.R.O., D43 No.0896 (Telequipment) was also used to detect any distortion in the output voltage. The gain of the amplifier for each frequency considered was calculated and the readings for the frequency, input voltage, output voltage and voltage gain were tabulated as shown in Table (6;2). Using these readings, a graph of the voltage gain in dB was plotted against frequencies (in log.scale) as shown in fig. (6,5).

From this graph, it was observed that there was a flat frequency response between a frequency range of 200Hz and 15KHz.

The poor response of the amplifier at low frequencies was due to the transformer T/T3 employed in the circuit. In any case, the result obtained was in accordance with the manufacturer's specification for the transformer (i.e. the transformer would give a flat frequency response over a frequency range of about 200Hz and 20KHz.)


35
30
25
20
15
10
5
0
GAIN (dB.)
FREQUENCY RESPONSE OF LOGARITHMIC AMPLIFIER (without Log. Element)
FIG(6 ; 5)
10¹
2 3 4 5 6 7 8 9
10²
2 3 4 5 6 7 8 9
10³
2 3 4 5 6 7 8 9
10⁴
2 3 4 5 6 7 8 9
10⁵
FREQUENCY (Hz)

(b) Logarithmic Characteristics of the Amplifier

To investigate the logarithmic characteristics of the logarithmic amplifier, two tests were conducted. One was to show the logarithmic characteristic when the input voltage was applied to the logarithmic element and the other when the input voltage was applied from the input of the microphone pre-amplifier.

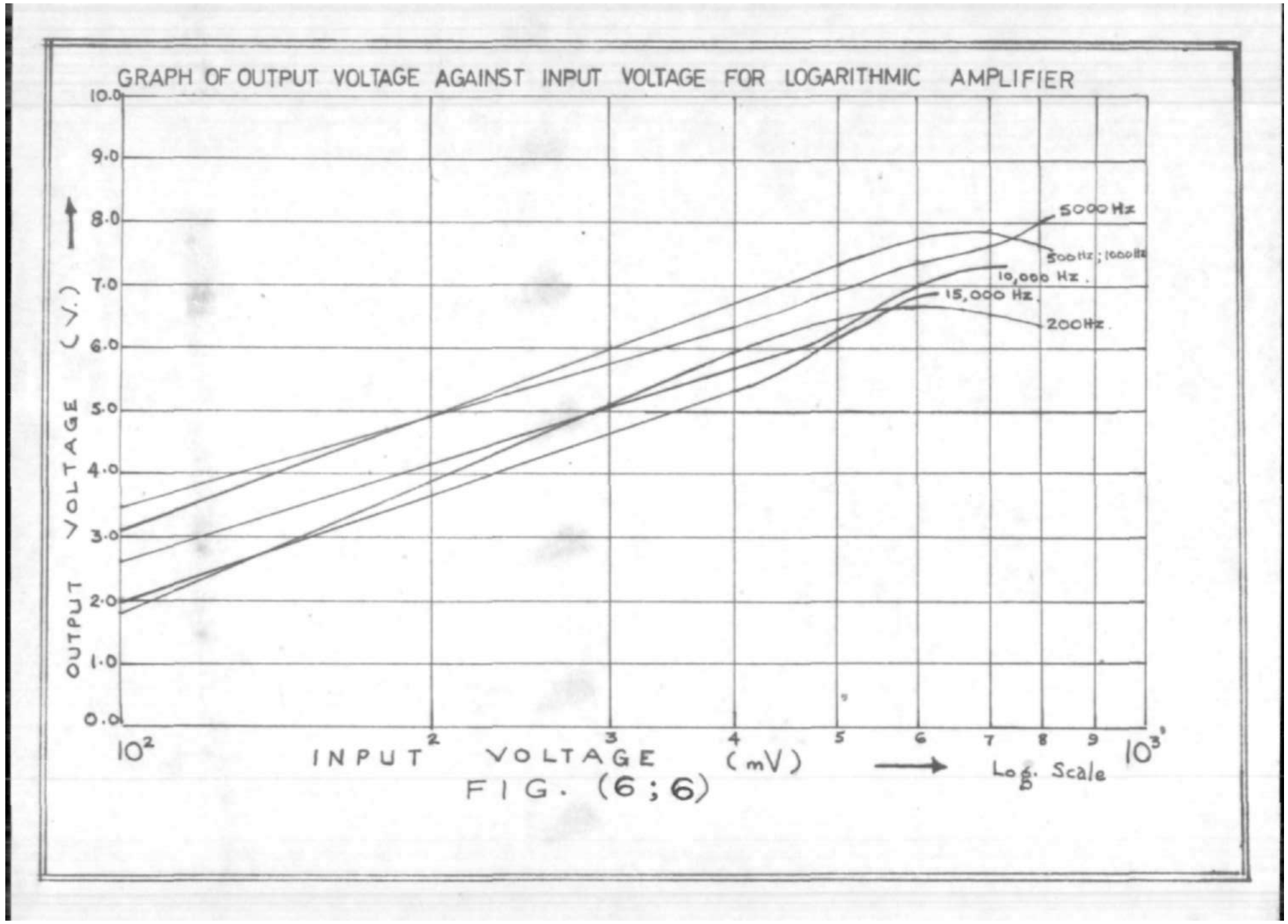
(i) Log. Characteristic of Amplifier with the Input Voltage Applied to the Log. Element

For this case, the output voltage of the amplifier were measured for different values of the input voltage applied to the input of the log. element. The input and output voltages were measured by the Hewlett Packard Vacuum Tube voltmeter 400 H No. M1765 and the frequencies at which these input - output voltage variations were measured were 200KHz; 5,000kHz; 10,000KHz & 15,000kHz

The output of the amplifier was also connected to the C.R.O. to know the onset of distortion. The results obtained are shown in Table (6;3) and from these, graphs of output voltage against the input voltage (in log. scale) were drawn for each frequency considered as shown in fig. .(6,6) From these graphs, it was observed that for all the frequencies considered, the logarithmic characteristic was generally satisfied between an output voltage range of 3.5 volts and 6 volts.

(ii) The same test was carried out when the input voltage was applied to the input of the pre-amplifier. The results obtained are shown in Table (6:4) and the graphs of the output voltage against the input voltage (in log. scale) were plotted for the respective frequencies considered as in the former case. These graphs are shown in fig. (6;7). From the graphs, the logarithmic relation $V_{(output)} = K \log. V_{in.}$ was generally satisfied within an output voltage range of 3 volts and 6 volts for the frequencies considered.

In all the previous cases where the C.R.O. was used to display the output voltage, waveforms observed were sine waves for undistorted output voltages.


GRAPH OF OUTPUT VOLTAGE AGAINST INPUT VOLTAGE FOR LOGARITHMIC AMPLIFIER
OUTPUT VOLTAGE (V.)
10.0
9.0
8.0
7.0
6.0
5.0
4.0
3.0
2.0
1.0
0.0
5000 Hz
500 Hz ; 1000 Hz
10,000 Hz.
15,000 Hz.
200 Hz.
10^2
2
3
4
5
6
7
8
9
10^3
INPUT VOLTAGE (mV)
Log. Scale
FIG. (6;6)

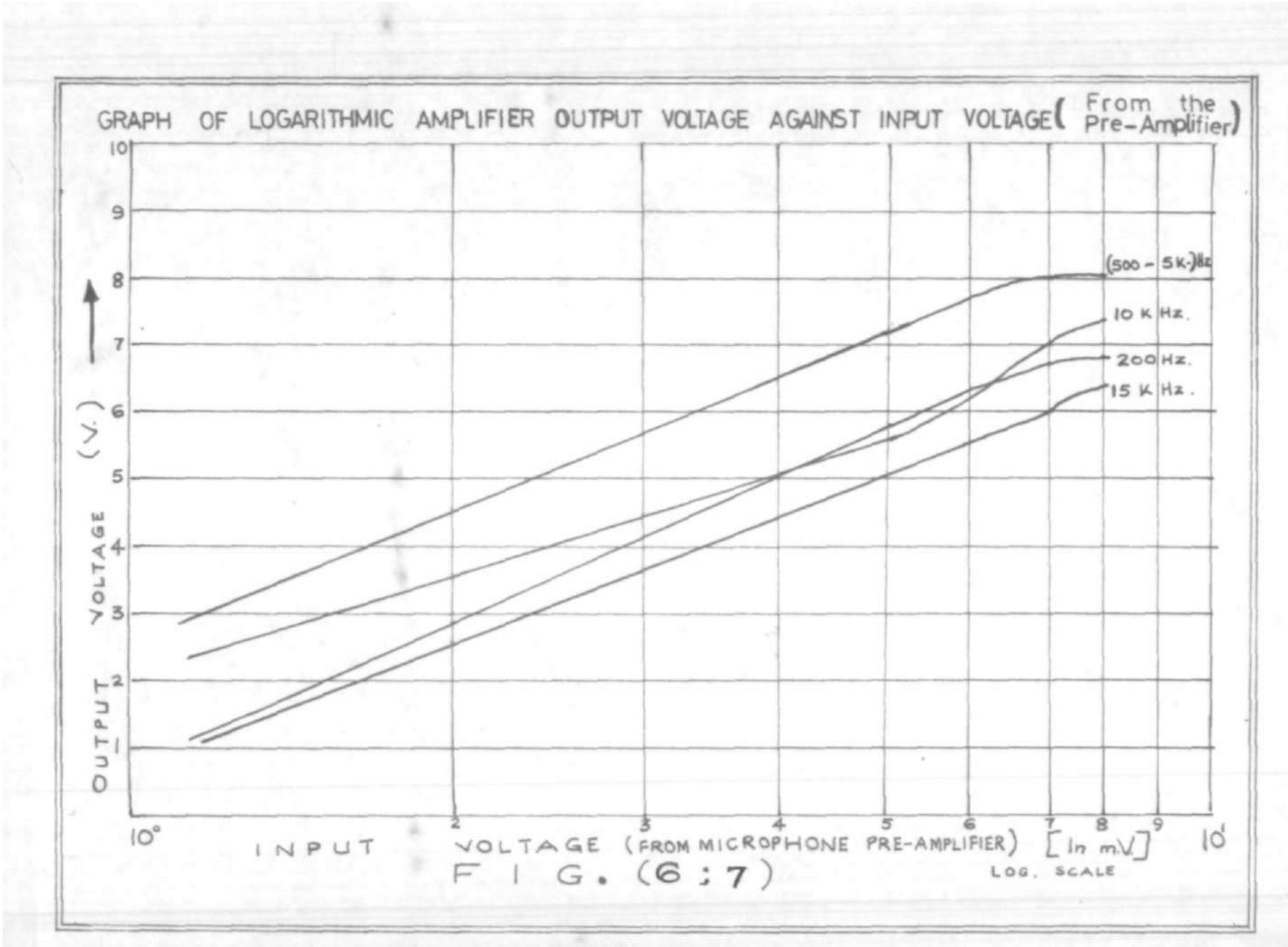

GRAPH OF LOGARITHMIC AMPLIFIER OUTPUT VOLTAGE AGAINST INPUT VOLTAGE(From the Pre-Amplifier)
OUTPUT VOLTAGE (V.)
10
9
8
7
6
5
4
3
2
1
(500 - 5K.)Hz
10 K Hz.
200 Hz.
15 K Hz.
10°
2
3
4
5
6
7
8
9
10¹
INPUT VOLTAGE (FROM MICROPHONE PRE-AMPLIFIER) [In mV]
LOG. SCALE

FIG. (6;7)

Table (6:2)

Frequency Response of Amplifier (Without log. Element)

Frequency in Hz.	Input Voltage (in volt (a.c.)	Output Voltage in volts (a.c.)	Voltage Gain in dB.
30	0.1	0.1	0
50	,,	0.5	1.4
70	,,	1.1	20.8
100	,,	2.0	26.0
200	,,	4.4	33.0
300	,,	5.1	34.2
400	,,	5.4	34.6
500	,,	5.5	34.8
700	,,	5.7	35.1
1,000	,,	5.7	35.1
2,000	,,	5.7	35.1
5,000	,,	5.4	34.6
7,000	,,	5.2	34.4
10,000	,,	4.8	33.6
12,000	,,	4.6	33.2
15,000	,,	4.4	33.0
20,000	,,	4.0	32.0
30,000	,,	3.5	30.8
40,000	,,	2.8	29.0
50,000	,,	2.4	27.6

Test Equipment:

Hewlett Pachard 651B Test Oscillator No.G 1439

C. R. O. Telequipment Type D43 No.0896

Hewlett Packard Vacuum Tube Voltmeter 400H No. M1765

Solartron Power Supply Unit AS1414 No. P818

Table (6:3)

Logarithmic Amplifier Input - Output Voltage Variations.

INPUT VOLTAGE in mV.(a.c.)	OUTPUT VOLTAGE (IN VOLTS) a.c. at 200Hz	at 500Hz	at 1000Hz	at 5000Hz	at 10000Hz	at 15,000Hz
100	1.9	3.1	3.6	3.5	2.6	2.1
150	3.0	4.1	4.1	4.2	3.5	3.0
200	3.9	4.9	5.1	4.9	4.2	3.7
300	5.2	6.0	6.1	5.7	5.1	4.7
400	5.9	6.6	6.7	6.3	5.8	5.4
500	6.5	7.3	7.4	6.9	6.4	6.2
600	6.6	7.7	7.8	7.4	7.0	6.7
700	6.5	7.9	8.1	7.8	7.4	7.2
800	6.4	7.7	8.1	8.0	7.7	7.4
900	6.2	7.2	7.7	7.8	7.6	7.6
1000	6.0	6.8	7.2	7.3	7.3	7.4

Table (6:4)

Variation of Logarithmic Amplifier Output Voltage with input from the Microphone Pre-Amplifier.

INPUT VOLTAGE in mV.	OUTPUT VOLTAGE IN VOLTS (a.c.) at 200Hz	at 500Hz	at 1000Hz	at 5000Hz	at 10,000Hz	at 15,000Hz
1.0	1.5	3.2	3.7	3.4	2.3	1.5
1.5	1.9	3.7	4.1	3.8	2.9	1.9
2.0	2.9	4.6	4.9	4.5	3.6	2.6
3.0	4.3	5.7	5.9	5.5	4.6	3.7
4.0	5.2	6.5	6.6	6.1	5.3	4.4
5.0	6.0	7.2	7.3	6.8	5.9	5.1
6.0	6.5	7.6	7.7	7.2	6.4	5.6
7.0	6.8	8.0	8.2	7.7	7.0	6.0
8.0	6.8	8.1	8.4	7.9	7.3	6.5
9.0	6.7	7.9	8.3	8.2	7.6	6.9
10.0	6.5	7.5	7.9	8.0	7.8	7.2

Test Equipment:

Hewlett Packard 651B Test Oscillator No.C.1439

,, ,, Vacuum Tube voltmeter; 400H No. M1765

C.R.O. Telequipment Type D43 No. 0896

Solartron Power Supply Unit AS 1414 No. P818.

CHAPTER 7

DETECTOR

.7.1. Specifications

Since a d.c. voltage was required to actuate the next stage of the circuit (i.e. the Schmitt trigger) it was therefore, necessary to construct a device capable of converting the a.c. voltage from the output of the logarithmic amplifier to a d.c. voltage. A diode detector (linear detector) was therefore chosen to effect this because of its simplicity of design. The requirements of the diode detector were:-

(i) to convert the a.c. input voltage to a d.c. voltage at the output.

(ii) the output d.c. voltage should contain minimum ripple voltages.

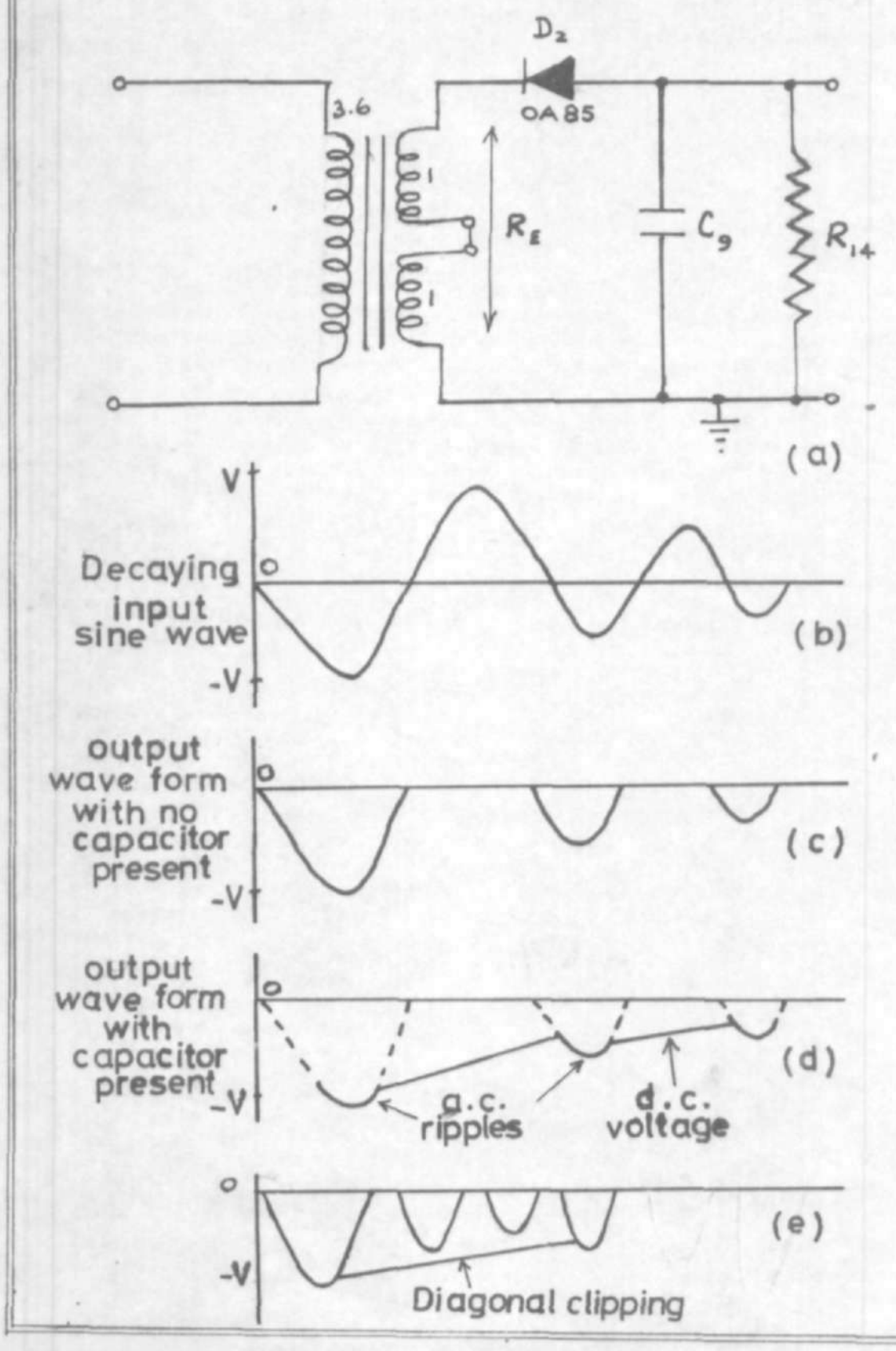
(iii) the output voltage should also be free from diagonal clipping.

7.2. Diode Detector

The process of rectification of an alternating electrical signal consists ideally in either (i) removing or (ii) reversing the polarity of those parts of the waveform which have polarity opposite to the one required. The first process is called Half-wave rectification while the second is known as full-wave rectification. Half-wave rectification was employed because few components were required.

The semi-conductor diode used was connected as shown in fig. (7;1a) so that if a sine wave voltage was applied to the circuit when the capacitor C_9 was not present, the positive half cycles of the waveform would be removed. On connecting the capacitor C_9 across R_{14}, it would charge to the level of the input voltage on the negative half cycle. The presence of the load resistor R_{14} would permit the capacitor to discharge.

DIODE DETECTOR. FIG.(7:1)



The conduction of the diode would occur when the voltage across the capacitor was less than the peak value of the input voltage. The time of diode conduction however would depend on the RC time constant.

It is nearly impossible to obtain absolutely pure, unvarying d.c. voltage from a rectifier. There is always some amount of a.c. components left and these are known as ripple voltages. The small a.c. ripples are superimposed on the d.c. voltage as shown in fig. (7;1d). The ripples are due to the small discharge of the capacitor between the succeeding negative half-cycles.

Also, if the RC time constant is long compared with the modulation period, the voltage across the capacitor will not decrease sufficiently rapidly to follow the decaying voltage waveform. This will thus result in diagonal clipping as shown in fig. (7;1e).

7.3. Design Procedure

A direct voltage nearly equal to the peak value of the carrier amplitude would exist across C and R, and this output voltage from the detector would be able to follow the decaying voltage if the period RC constant of the detector was short compared with the period of the decaying voltage.

Then considering this decay period of the voltage amplitude as being approximately equivalent to the reverberation time (i.e. about 2secs. for large halls and about 1 sec. for small rooms). Therefore, in order that the output voltage from the detector may follow the decay of the input voltage,

$$CR \ll 2\text{sec.}$$

$\therefore$ Assuming that this relation would hold

$$\text{when } CR = \frac{2}{20} \text{ Sec.}$$

Therefore $CR = 0.1\text{Sec.} = 100\text{mSec.}$

But C is designated C_9 and R is designated R_{14} in fig.(7.1a).

$\therefore$ $C_9 R_{14} = 100\text{m}$Sec.

Choosing $R_{14} = 15\text{K}\Omega$ gives $C_9 = \frac{100 \times 10^3}{15 \times 10^3}\ \mu\text{F}$.

$= \frac{100}{15}\ \mu\text{F} = 6.7\mu\text{F}$

$\therefore$ Preferred value of $C_9 = 8\ \mu\text{F}$.

$\therefore$ $C_9 = 8\mu\text{F}$

$R_{14} = 15\text{K}\Omega$

7.4. Test for the Linearity of the Diode Detector

After building the diode detector, a test for its linearity was performed. This test involved measuring the input and output voltage variations of the detector at certain chosen frequencies of the input signal (i.e. 200Hz, 500Hz, 1000Hz, 5000Hz, 10,000Hz and 15KHz.) The detector a.c. input voltages (from the logarithmic amplifier) were measured by the C.R.O. Telequipment Type D43 No.0896 and the corresponding output voltages (d.c.) were measured by the Hewlett Packard Vacuum tube voltmeter Model 410B No. M1773.

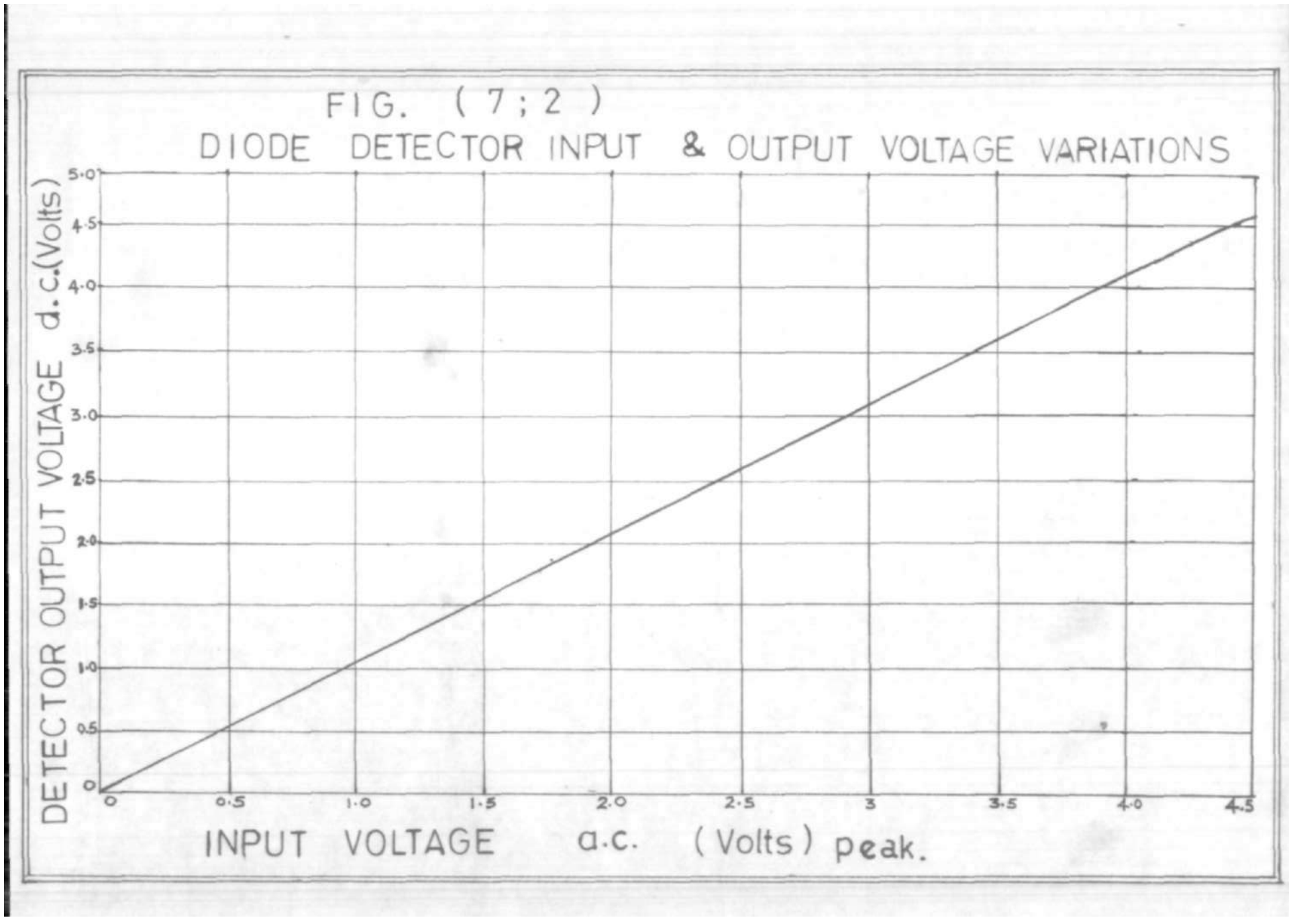
The readings were compiled and tabulated as shown in table (7;1). Since the readings of the input - output voltages obtained at different input signal frequencies did not show any marked difference, it was therefore decided to plot the graph of the output voltage against the input voltage for the input signal frequency of 1000Hz. alone since other cases would be almost identical to it. The graph is shown in fig. (7;2)

From this graph, it was observed that a linear relation was obtained for an output voltage range of (0 - 4.5)V.

Since the logarithmic range of the output voltage from the logarithmic amplifier is about (3 - 6) volts, it was then decided to use a logarithmic range of (3.5 - 4) volts which would be equivalent to 10dB decay, as the working range.

FIG. (7 ; 2)

DIODE DETECTOR INPUT & OUTPUT VOLTAGE VARIATIONS



This thus meant that the switching level of the level comparator (the Schmitt trigger) would be 3.5 volts.

Therefore, the voltage was expected to decay from 4 volts to 3.5 volts before the occurence of the switching of the Schmitt trigger and the decay would correspond to a 10 dB decay. Therefore, if the decay time for a 10 dB decay could be obtained, the time for a 60 dB decay corresponding to the reverberation time could also be estimated.

One problem which came up with the detector after connecting it to the Schmitt trigger was that the level indicator which was connected at the output of the detector to give the d.c. voltage going into the input of the Schmitt trigger was still indicating a certain residual voltage of about 1.5 volts when the power supply was on but no signal was applied to the detector input. However, this was later found to be due to the fact that under no signal condition, with the power supply on, the residual voltage indicated on the level indicator was the d.c. base voltage of the following transistor in the level comparator.

Table (7:1)

Detector Input - Output Voltage Variations

INPUT VOLTAGE (a.c.) In volts p	DETECTOR OUTPUT VOLTAGE (d.c.) IN VOLTS With Input Signal freq 200Hz	With Input Signal freq 500Hz	With Input Signal freq 1KHz	With Input Signal freq 5KHz	With Input Signal freq 10KHz	With Input Signal freq 15KHz
0	0.0	0.0	0.0	0.0	0.0	0.0
0.5	0.55	0.6	0.6	0.6	0.55	0.55
1.0	1.0	1.15	1.2	1.15	1.10	1.00
1.5	1.5	1.6	1.6	1.55	1.5	1.55
2.0	2.10	2.2	2.2	2.15	2.1	2.1
2.5	2.55	2.6	2.6	2.6	2.5	2.5
3.0	3.00	3.1	3.1	3.05	3.0	3.05
3.5	3.50	3.6	3.65	3.6	3.5	3.55
4.0	3.95	4.0	4.0	4.05	3.95	4.0
4.5	4.55	4.6	4.6	4.55	4.50	4.50
5.0	4.90	4.95	4.95	4.95	4.90	4.90

Test Equipment:

Solartron Power Supply Unit AS1414 No. P.818

Vacuum Tube Voltmeter (Newlett Packard) Model 410B No. M.1773.

C. R. O. (Telequipment) Type D43 No.0896.

CHAPTER 8

THE LEVEL COMPARATOR (SCHMITT TRIGGER) DEFFERENTIATING & CLIPPING NETWORKS.

8.1. The Level Comparator

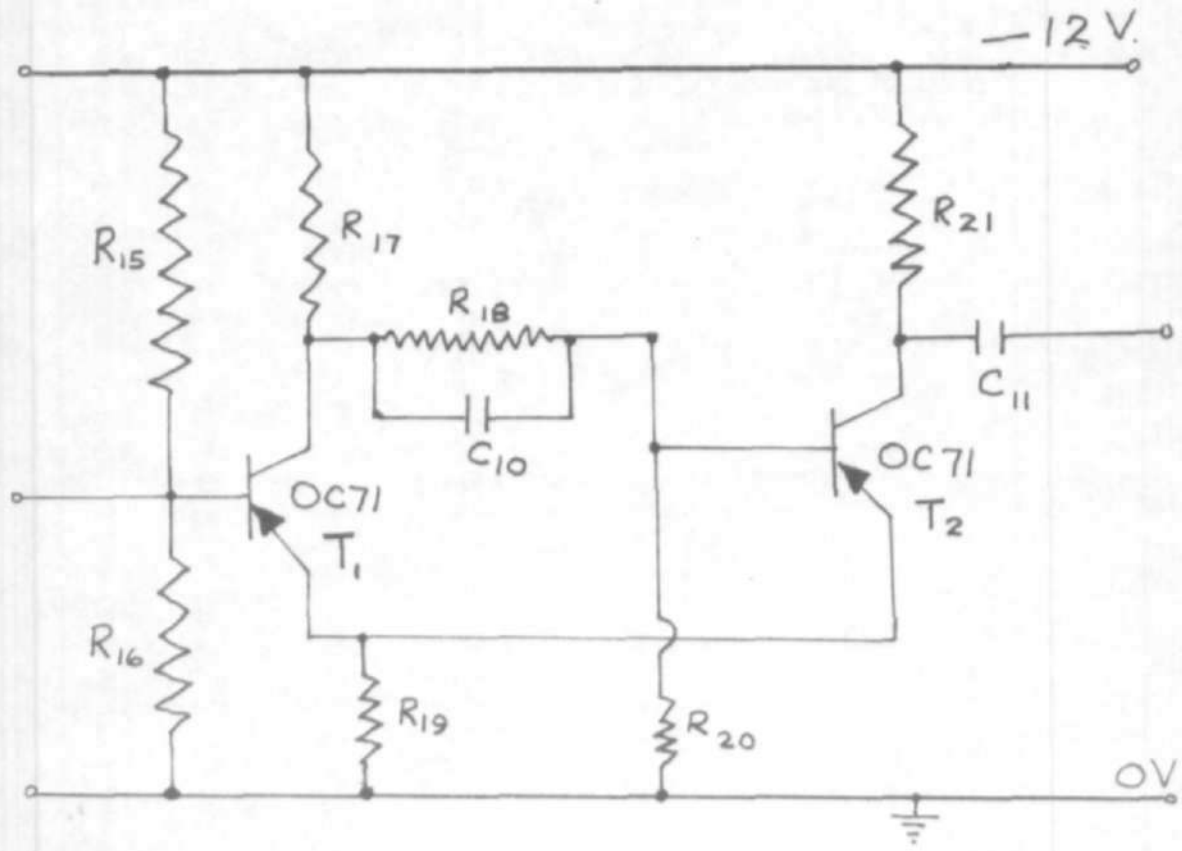
The d.c. voltage from the detector circuit was fed to the input of the level comparator. This level comparator was supposed to be a device which could change state when its input voltage crossed a specified level and a pulse would be produced at its output. For the level comparator, a Schmitt trigger circuit was chosen because it has all the properties of the level comparator required.

8.2. The Schmitt Trigger (Theory)

The first account of a basic cathode-coupled Bistable multivibrator whose state depended only on the d.c. voltage level at the input terminal was given by Otto Schmitt in 1938. The original circuit used valves and the transistor version is often called the emitter coupled asymmetrical trigger circuit or the Schmitt trigger.

The transistor Schmitt trigger used is shown in fig.(8:1). In the circuit two p.n.p. transistors OC71 were employed. The resistors were so chosen that normally, when no voltage is applied to the input, the transistor T_1 is cut off and the collector voltage $V_{C1} = V_{cc}$ volts where $V_{cc} = 12V$. This negative voltage is coupled to the base of T_2, and it forward biases the base to emitter junction thus causing T_2 to operate in the saturation region. The circuit will remain in this condition until a negative going signal of sufficient amplitude to overcome the reverse bias at the base of T_1 is applied. This turns on that stage. The collector voltage of T_1 then decreases in magnitude and the change is coupled to the base of T_2. This causes the emitter current of T_2 to decrease.

LEVEL COMPARATOR



[SCHMITT TRIGGER]

FIG. (8:1)

It also lowers the potential across R_{19} which in turns causes the collector current of T_1 to increase even more. As a result of the regenerative circuit action, in a short time, T_1 is fully on (saturated) and T_2 is fully off. The output voltage taken from the collector of T_2 is then a maximum negative voltage $V_{C2} = V_{cc} = 12V$.

8.3. Design Procedure

The desired output voltage from the Schmitt circuit, $V_o = -7$ volts p.p.

The desired Switching voltage, $V_e = 3.5V$(d.c.)

The supply voltage $V_{cc} = -12$volts.

When T_2 was on, it would pass a current

$$I_{C2} = \frac{V_o}{R_{21}}$$

$$\therefore R_{21} = \frac{V_o}{I_{C2}}$$

For $I_{C2} = 4.5mA$; $R_{21} = \frac{7}{4.5}$ KΩ

$= 1.55K\Omega$

$\therefore$ The Preferred value of $R_{21} = 1.6K\Omega$

The current $I_e = I_{C2}$ would pass through R_{19} with a voltage V_e developed across R_{19}

$$\therefore V_e = R_{19} I_e = R_{19} I_{C2}$$

$$\therefore R_{19} = \frac{V_e}{I_{C2}}; \quad \text{But } V_e = -3.5 \text{ volts.}$$

$$\therefore R_{19} = \frac{3.5}{4.5} \text{ K}\Omega = 0.77K\Omega$$

The preferred value of $R_{19} = 0.68K\Omega$

$= 680\ \Omega$

Also $V_e = \frac{10R_{20}V_o}{R_{21}h_{fe\,(min.)}}$

$\therefore\ R_{20} = \frac{V_e R_{21} h_{fe(min.)}}{10V_o}$

where $h_{fe(min.)} = 41$

$\therefore\ R_{20} = \frac{3.5 \times 1.6 \times 41}{70}$ K

$= 3.3$ K

Choosing $R_{17} = R_{21} = 1.6$ K

$R_{18} = \left(\frac{V_{cc}}{V_e} - 1 - \frac{R_{17}}{R_{20}} - \frac{R_{17}}{R_{19}}\right) R_{20}$

$= \left(\frac{12}{2.7} - 1 - \frac{1.6}{3.3} - \frac{1.6}{0.68}\right) \times 3.3$ K

$= (4.46 - 3.86) \times 3.3$ K

$= 2.0$ K

8.4 Biasing Resistors R_{15}, R_{16}

Choosing a stability factor S = 4.5

$\therefore\ S = \frac{1 + R_e/R_b}{1 - \alpha + R_e/R_b} = 4.5$

i.e. $\frac{1 + R_e/R_b}{1 - 0.98 + R_e/R_b} = \frac{1 + R_e/R_b}{0.02 + R_e/R_b} = 4.5$

$\therefore\ R_e/R_b = 0.25$

But $R_e = R_{19} = 0.68$K

$\therefore\ R_b = \frac{0.68}{0.25}$ K $= 2.7$ K

But the base resistance $R_b = \frac{R_{15}R_{16}}{R_{15}+R_{16}}$

$\therefore$ If $R_{15} = 5R_{16}$,

$$R_b = \frac{5R_{16}^2}{6R_{16}} = \frac{5}{6} R_{16}$$

$$\therefore \frac{5}{6} R_{16} = 2.7 \text{ K}\Omega$$

$$\therefore 5R_{16} = 2.7 \times 6 \text{ K}\Omega$$

$$\text{i.e. } R_{16} = \frac{2.7 \times 6}{5} \text{ K}\Omega$$

$$= 3.24 \text{ K}\Omega$$

$$\therefore R_{15} = 5R_{16} = 3.24 \times 5 \text{ K}\Omega$$

$$= 16.2 \text{ K}\Omega$$

The preferred values of R_{15}, R_{16} are:-

$R_{15} = 15\text{K}\Omega$

$R_{16} = 3.3 \text{ K}\Omega$

The capacitor C_{10} was applied as shown in fig. (8;1) to speed up the switch over of the circuit.

A value of C_{10} less than 500 p.F. would be applicable.

The value of C_{10} used was given by:-

$C_{10} = 200\text{p.F.}$

8.5. Testing of the Level Comparator

Test Equipment:-

Oscilloscope Teloquipment Type D43 No.0896

Harrison 622 1B D.C. Power Supply, No. P1442.
Solartron Power Supply Unit AS1414 No. P.818.

The supply voltage for the Schmitt trigger circuit = -12volts supplied from the Solartron Power Supply, Unit A.S. 1414 No. P818.

A negative-going d.c. voltage was applied to the base of the triggerr circuit from the Harrison 622 6D d.c. power supply and the output voltage was displayed on the Oscilloscope.

The voltage level at the base was then varied slowly and carefully in amplitude from about -6volts to 0V. taking note of the switching level of the Schmitt trigger. It was observed that the switching occurred when the voltage level was -3.5V. This process was repeated about three.times and the same result was obtained in each case. The measured values of V_o and V_e were:- V_o = 5.5volts and V_e = 3.5 volts.

8.6. The differentiating Network

The capacitor C and resistor R forming the differentiating network were connected to the output of the Schmitt trigger circuit to differentiate the output voltage pulse from the Schmitt circuit. The values of R and C were respectively determined by feeding the output from the differentiating network to an Oscilloscope (C. R O.) Telequipment Type D43 No.0896 and noting the values of R and C which gave a good differentiated output as shown in fig.(8,2).

These values are:-

R = 10KΩ

C = 0.25μF

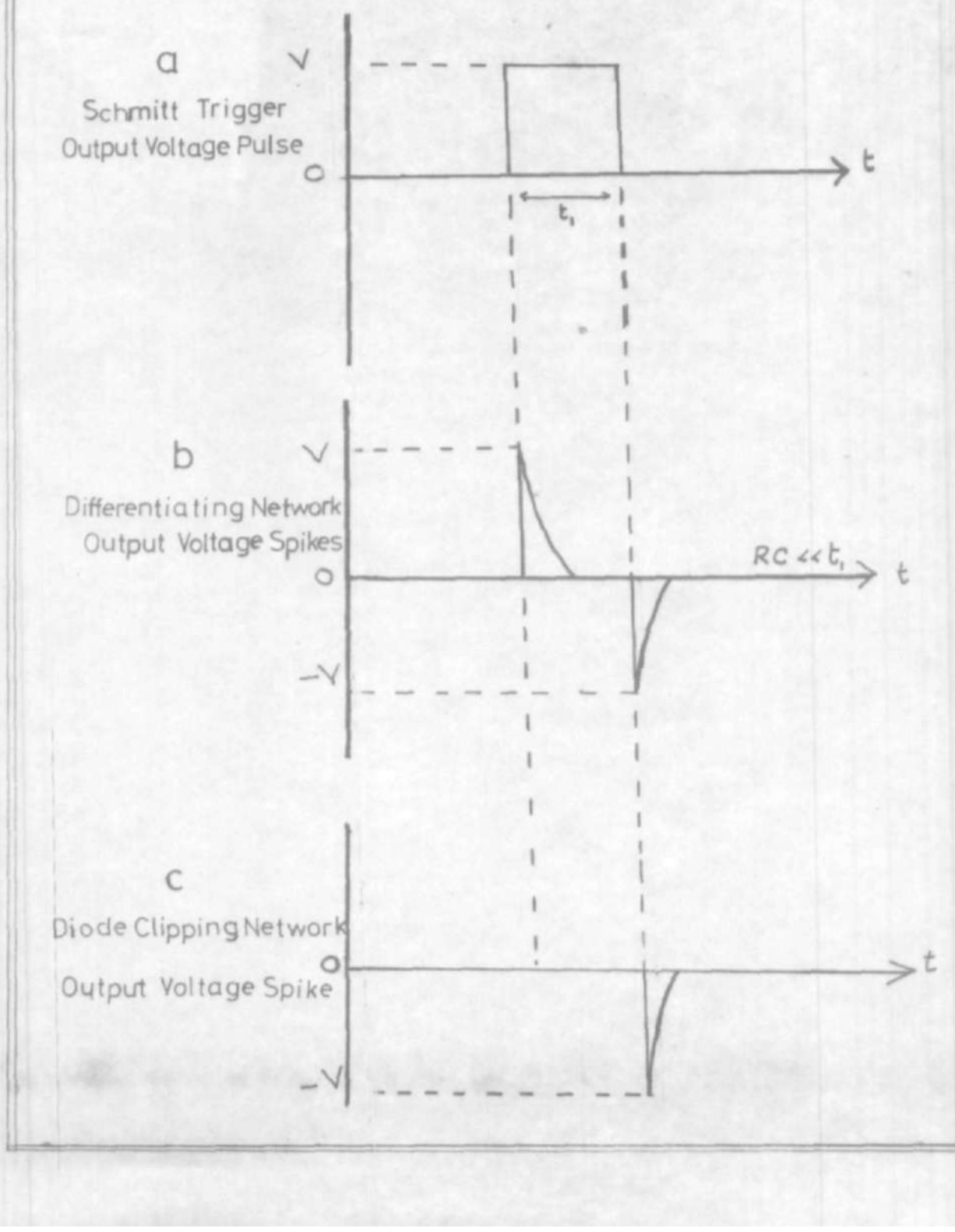
i.e. RC= 2.5 mSec.

8.7 Clipping Circuit

The differentiating network produced two spikes at its output (one negative and one positive). Since only the negative spike would be required to stop the timer counter used in the project, the positive spike was then clipped off by connecting a diode OA 85 as shown in fig. (9,1). The amplitude of the final negative spike going to the Timer Counter = 7.5 volts p.p . This voltage spike would be suitable for stopping the timer counter since it fell within the range specified for the counter by the makers.

FIG. (8;2)

OUTPUT VOLTAGE PULSE & SPIKES



CHAPTER 9

TESTING OF THE FINAL SET UP OF APPARATUS

The circuit diagram showing the set up of the complete reverberation time meter is shown in fig. (9.1).

9.1. Testing of the Switch, Loudspeaker and Counter

The first test performed was to make sure that the two position switch (with one break and one make contacts) employed worked satisfactorily. This switch was connected between the signal generator and the loudspeaker power amplifier in one position and in the other position, it was connected between the start terminal of the counter and ground. When the key was in its normal position, nothing happened, but when it was pushed up, it closed the contact between the signal generator and the loudspeaker power amplifier.

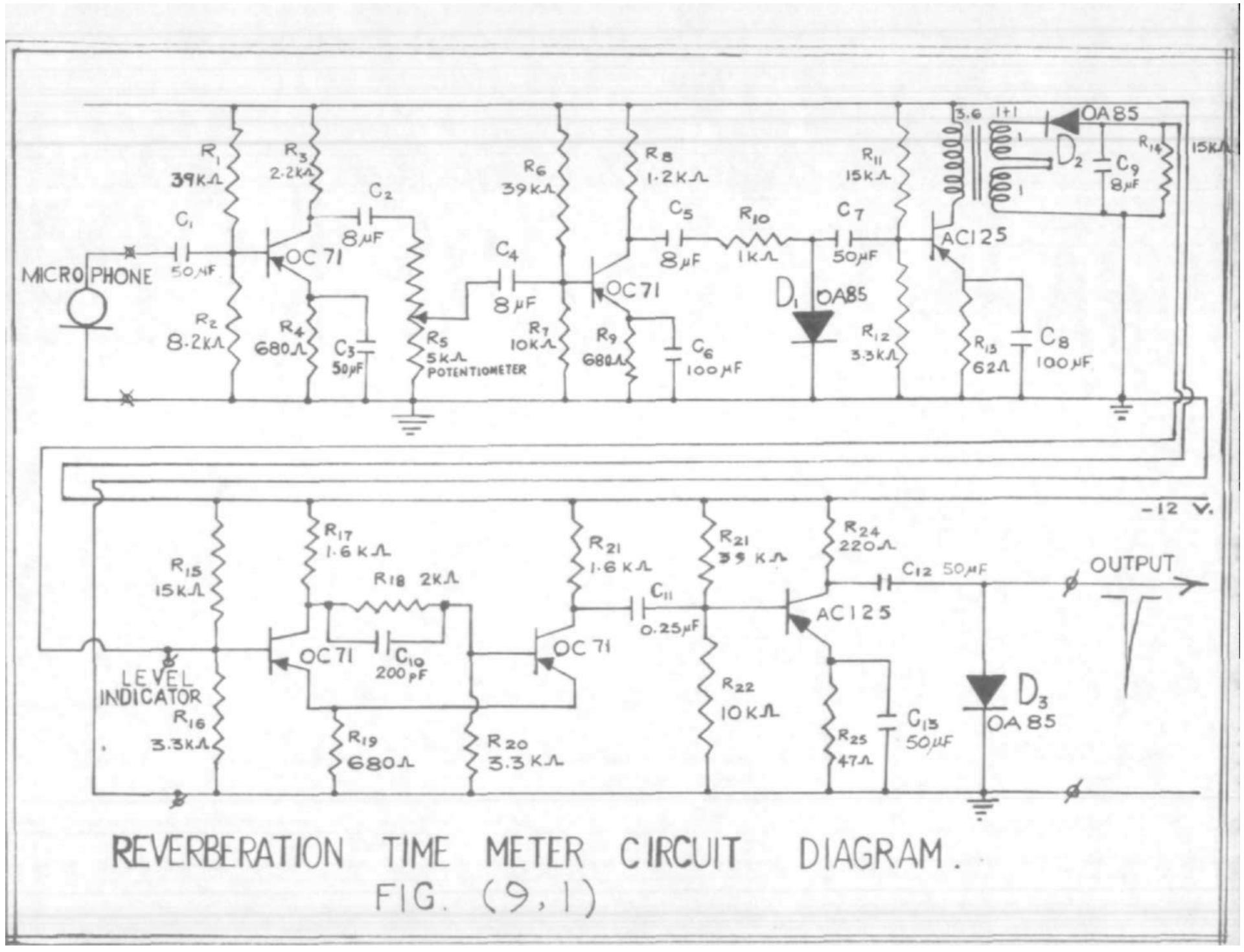
With the signal generator and power amplifier on, the loudspeaker started to produce sound whose intensity could be varied by varying the gain control of the power amplifier.

On depressing the key, the signal generator contact was broken and the one between the start terminal of the counter and ground was closed. This action simultaneously stopped the loudspeaker output and started the counting. This test thus gave the assurance that the switch could be effectively used to control both the loudspeaker and the timer counter as desired.

9.2 Testing the Microphone and the Level Indicator

The second test performed was to check whether the microphone was alright and if so, to investigate the action of the level indicator (Hewlett Packard Vacuum Tube Voltmeter Model 410B, No. M1773) when the microphone was placed in sound field.

The microphone was connected to the input of the pre-amplifier. The power supply unit was also connected to the circuit and all other necessary connections were made. Sound was first produced by clapping of hands and whistling.



REVERBERATION TIME METER CIRCUIT DIAGRAM.

FIG. (9,1)

and in each case, there was a deflection of the pointer on the level indicator. The noise made by people walking along the corridor and the occasional traffic noise in the street were also picked up by the microphone and this also caused the deflection of the pointer on the level indicator. Deflection was proportional to the intensity of sound received by the microphone.

The loudspeaker was then used to produce sound and the intensity was varied from a minimum level to a maximum level. This consequently caused the deflection of the pointer from a minimum to a maximum level. This test showed that the microphone was alright and that the level indicator too was working well.

9.3. Testing the Whole Set-Up

The third test was performed for the whole set-up (i.e. when all connections necessary were made to the timer counter and all other equipment used.) This test was carried out in the night because external noise was then a minimum. The loudspeaker was used to produce sound and the intensity was increased until the level indicator read 4.0 volts. The loudspeaker was then stopped and the sound started to decay in the room. As this happened, the level of the pointer on the indicator scale was decreasing until it crossed the switching level of the Schmitt trigger used as the level comparator (Switching level = -3.5V) and the counter stopped counting. The time for the decay was then given on the counter. This process was repeated about five times and in each case, the counter was stopped at the 3.5volts level of the level indicator.

After this, the intensity of the sound produced by the loudspeaker was lowered such that the level indicator recorded a lower voltage than the trigger switching voltage (-3.5volts) and the loudspeaker was then shut off. The timer counter started counting immediately the loudspeaker was shut off, but the counting continued indefinitely until the stop terminal of the counter was operated manually.

This was performed about four times and the same thing happened in each case. This test showed that the stopping of the counter could only be effected if the level indicator gave a voltage equal to -3.5volts.

9.4. Test for the Consistency of the Counter Readings

A test was carried out to check the consistency of any reading of the timer counter. This check was made by adjusting the intensity of the sound produced by the loudspeaker so that the level indicator initially recorded 4.0volts. The loudspeaker was then shut off and the timer counter started to count. The counter stopped when the level indicator read 3.5 volts and the time recorded on the counter gave the time for a 10 dB sound decay. This time was 167 mSec. at a signal frequency 400Hz. This process was repeated three times again at the same signal frequency and these other attempts gave 170mSec., 168mSec., and 179mSec. respectively. Each of these values obtained for the time of decay of the sound through 10d.B. did not show a considerable difference from the average of the four readings, 169mSec. thus showing that the readings are consistent.

9.5. Effect of Increasing the Decay beyond 10 dB on the time Measurement

Another test which was carried out was to investigate the effect of increasing the level indicator reading beyond the 4 volts level on the time recorded by the counter. It was observed that the time recorded by the counter was getting higher and higher as the pointer level on the level indicator increased beyond the 4volts level. This was the expected result since the increase in the pointer deflection beyond 4volts would cause an increase in the decay beyond 10 dB and so, higher values of time should therefore be obtained.

9.6. Final Test for the accuracy of the Reverberation time meter

The arrangement shown in the fig. (9;2) was used in finding the accuracy of the reverberation time meter. Since sound exhibits an exponential decay in a room and the reverberation time meter constructed was to be used for measuring this sound decay, therefore, to test the accuracy of the meter,

an arrangement consisting of an R.C. network was made (as shown in fig. (9;2)) to provide an exponentially decaying voltage similar in characteristic to the sound decay. The timer counter was used to measure the decay time corresponding to the reverberation time using this arrangement as the input voltage source of the reverberation time meter. Since the time constant of the R.C. network was known, and this time constant would give the time for the decay of the voltage across the capacitor C_1 by 63.2% of its initial value when discharging through the resistor R; the time corresponding to the reverberation time could therefore be estimated theoretically using the R.C. time constant and hence the accuracy of the instrument could be estimated.

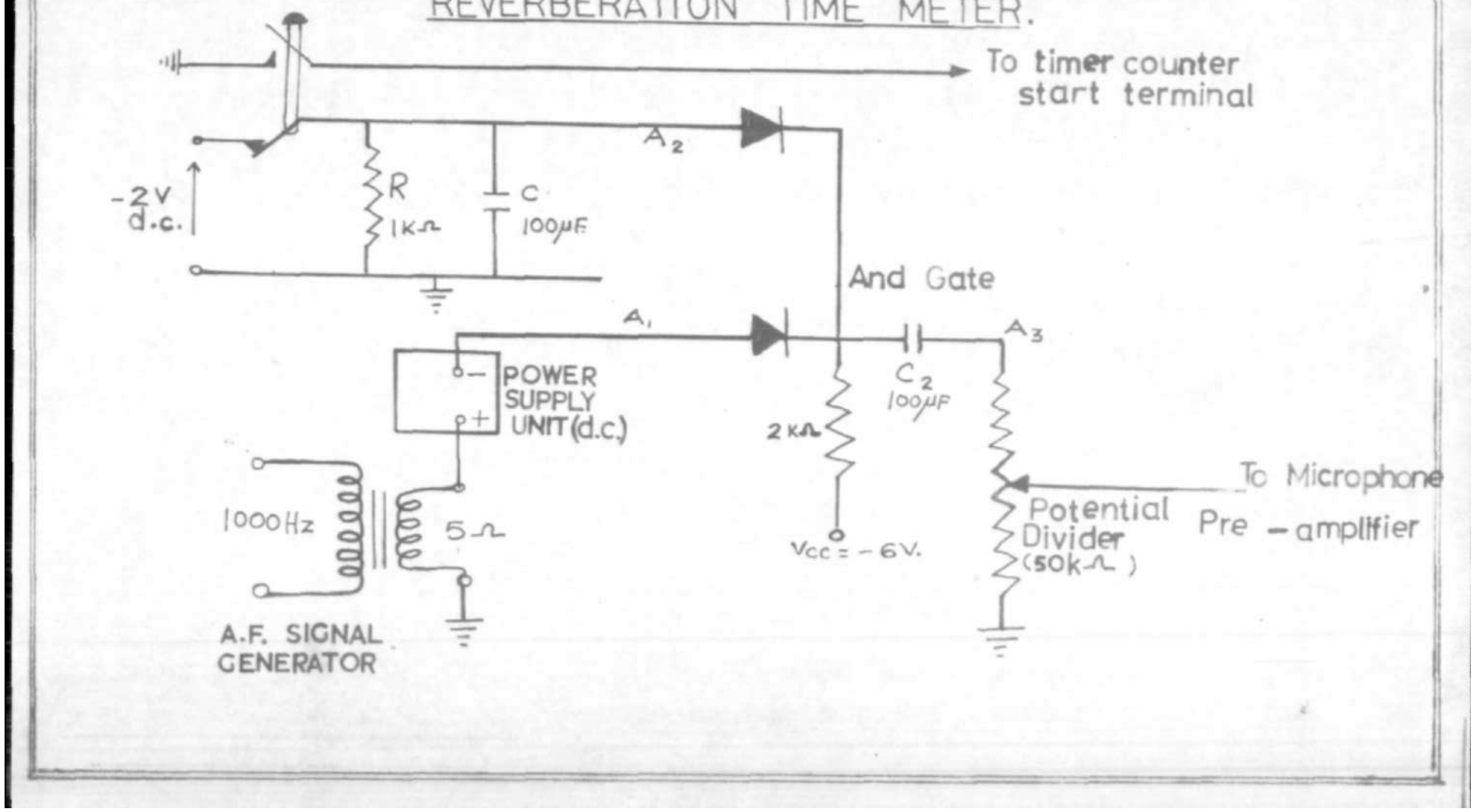
The arrangement involved the use of the AND gate shown in the fig.(9;2). The gate has two inputs A_1 and A_2 and one output A_3. The output A_3 would be on if only A_1 and A_2 were on.

An a.c. voltage of 2volts from a signal generator was introduced to the input A_1 of the gate through a d.c. power supply unit which was used as a biasing unit. The other input of the gate A_2 was connected to the RC network. Through a two position switch with one make and one break contacts, a 2v. d.c. signal was introduced to the RC network. This switch was also used to start the timer counter as soon as it switched off the input signal to the RC network When the two inputs A_1 and A_2 were on, an output voltage was obtained at A_3 and the amplitude of this was controlled by a potential divider arrangement. This output voltage was then fed to the input of the reverberation meter (base of the pre-amplifier). This made the pointer on the level indicator to rise and it was adjusted to a level of 4volts d.c.

When the key of the switch was depressed, the d.c. voltage at the input of the RC network was switched off and the counter started to count. The capacitor C in the RC network then started to discharge exponentially through the resistor R and the output at A_3 of the AND gate also gave a similar exponential voltage decay.

FIG. (9;2)

CIRCUIT USED FOR TESTING THE ACCURACY OF THE REVERBERATION TIME METER.



The pointer of the level indicator then started to fall from its initial level of 4volts and as the voltage passed the switching level of the Schmitt trigger (3.5volts), the counter was stopped and the time corresponding to a 10d.B. decay was read off from the counter. This process was repeated three times and from the average result, the time corresponding to the reverberation time was obtained as given below:

Attempts	Counter Reading of time for a 10 dB. decay (T_{10})
1st attempt	120 mSec.
2nd ,,	117 mSec.
3rd ,,	121 mSec.
	Average T_{10} = 119mSec.

$\therefore$ T_{60} (Reverberation Time) = 119 x 6 mSec. = 714mSec. = 0.71sec.

For the RC network, R = 1KΩ; C = 100μF.

But the RC time constant T corresponds to a voltage decay through 63.2% of its initial value.

$\therefore$ The R.C. time constant T corresponds to the time for 20 log. $\frac{100}{36.8}$ dB.decay

i.e. T corresponds to the time for 20 x 0.45 dB = 9dB decay.

$\therefore$ T_9 = 0.1 sec.

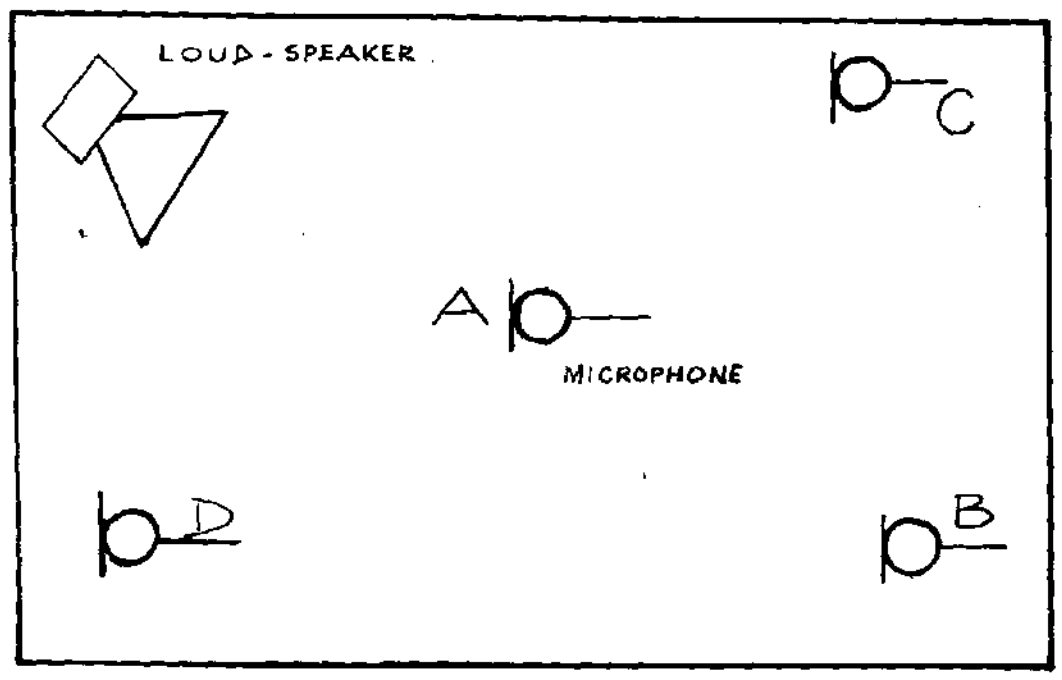
and so, T_{60} (R.T.) = $\frac{60}{9}$ x 0.1Sec. = 0.67 Sec.

But the Reverberation time meter gave T_{60} = 0.71sec.

$\therefore$ The accuracy of the meter = $\pm \frac{0.71 - 0.67}{0.67}$ x 100%

$\pm \frac{0.04}{0.67}$ x 100% = $\pm$ 6.0%.

$\therefore$ The accuracy of the Reverberation time meter = $\pm$ 6.0%.



MICROPHONE & LOUD-SPEAKER POSITIONING IN A ROOM

FIG. (9;3)

CHAPTER 10

MEASUREMENT

Before going to make measurements for the reverberation time of different rooms, certain procedure of measurement and precautions were agreed upon. These would make the readings taken to be orderly and reasonable. All the modes of measurement and precautions taken are given in the following paragraphs.

10.1 Sound decay time employed in the Measurement.

Since the diode detector used gave a linear characteristic over a range of about (0 - 4.50) volts and the switching level of the comparator is 3.5V. measurement of sound decay could therefore be made for a maximum decay of:-

$$(4.50 - 3.5) \times 20 \text{ dB} = 20 \times 1.0 \text{ dB}$$
$$= 20 \text{ dB}$$

In the measurements made, a decay of 10 dB was then chosen. This meant that the level indicator would first be set to a level of 4 volts by varying the intensity of sound produced. The sound source would then be shut off at this level and when the sound decayed to a level at which the level indicator reading = 3.5V., switching would occur in the level comparator and the counter would be stopped. The reading on the counter would then give the time for a decay of sound through 10 dB (i.e. T_{10}). The reverberation time could then be obtained by multiplying this time by six to give the time for a 60 dB decay corresponding to the reverberation time.

10.2 Frequency Range

The range of frequencies used in the measurements was 100Hz - 8KHz. Frequencies lower than 100Hz were not employed because the readings obtained with them would not be reliable since the amplifiers used (particularly the logarithmic amplifier) have poor response at such low frequencies.

The upper limit of the frequency range could not be taken up to 10KHz or 15KHz because after a frequency of 8KHz, measurements for higher values of frequencies were not possible because the intensity of sound produced by the loudspeaker was not high enough to produce a voltage of 3.5 volts on the level indicator even with the gain control of the power amplifier turned to the maximum point.

10.3. Periods of Measurement

Reverberation time measurements should be made at a time when extraneous noise is a minimum. Therefore, the nights are the most suitable periods because there will be very little or no external noise produced. The nights were therefore chosen for the measurements. Afternoon measurements were also made, but they were to show the effect of the extraneous noise on the reverberation time. Rampant among the extraneous noises produced in the afternoons were traffic noise in the roads, the footsteps and voices of people walking along the corridors, the rumbling of the air-conditioning systems, the hum of some electrical machines and also the noise from the departmental workshops below.

10.4. Loudspeaker Positioning

The loudspeaker was located in a corner of the room in which measurements were made. This was because the greatest number of normal modes of vibration would be excited at the corner. This also would ensure maximum number of room resonances which would be an advantage in obtaining a smooth over-all reverberation curve(for better acoustic coupling.) Mounting in a corner would also inprove the low frequency response of the loudspeaker.

10.5. Microphone Positioning

Reverberant rooms have a standing wave patter and so, if the microphone was held at a fixed point in the room, the result obtained might not give a real indication of the over-all sound field in the room. Therefore, the microphone employed was used to make reverberation time measurements at four different locations in the room.

The aim being to average out the point to point variations of the reverberation time in the room. The positions where the microphone was placed during measurements were denoted A, B, C and D as shown in fig. (9:3). These notations were also used in the graphs for the reverberation times.

10.6. Room Conditions for Measurements

Measurements were made when all doors and windows in the rooms were closed. This step was taken because if the doors and windows were left open, they would present direct paths for the transmission of sound to the outside and the result obtained for the reverberation time would not be reliable. In each room, except where it was impossible to remove the materials inside the rooms, two sets of measurements were taken; one for the empty rooms and the other when the materials were in the room.

10.7. Measurements

The first measurements were performed in the Part III lecture room. Four sets of measurements were made in all for the room. Two in the afternoon and two in the night. The afternoon and night measurements each consisted of one set of measurement for the empty room and the other set for the room with tables and chairs. The readings obtained for variation of reverberation time with frequency were compiled and tabulated as shown in tables [10, (1 - 4)].

The same procedure was repeated for the Part II lecture room. After these measurements, it was observed that the afternoon measurements did not show very considerable difference from the night ones except in certain cases. It was therefore, decided that for other rooms only the night measurements would be made. This decision was first carried out in the departmental library where only the night measurements were made. It was not possible to remove the books off the shelves and also the curtains when making measurements for empty room.

The readings were thus obtained with the books and curtains inside. The other set of readings were obtained when the tables and chairs were also brought inside.

After this, it was decided that reverberation time measurements should also be made in the Assembly Hall for the sake of comparison. However, afternoon measurements could only be made. This was because the Hall was already booked for the nights by some other users and no other time could be convenient except in the afternoon. Measurements were therefore made in the afternoon on a Saturday, when noise level was not excessive because no sporting activities were in progress. Two sets of measurements were made; one for the empty hall and one for hall with chairs and stage curtains. After this, the next place where measurement was also taken was the small Lecture Theatre of the faculty of Engineering. Here, only one set of reading was taken. Measurement was taken for only one position of the microphone (on the lecturer's table near the black-board).

The measurement was however taken in the night (between 8.oop.m. and 10.oop.m.).

After a visit to the High Voltage Laboratory it was discovered that the acoustics of that room was quite unsatisfactory. The sound produced by the clapping of hands or whistling took about three seconds to die down. It was therefore decided to make a reverberation time measurement for the laboratory. In any case, it was not possible to use more than one microphone position because of the equipment installed in the room. Measurment was also made in the night (8.oop.m. - 10.oop.m.)

After taking the reverberation time - frequency measurement for each room, the dimensions of the room were measured so as to be able to estimate the volume of the room and also its likely value of reverberation time.

For all the measurments made in the rooms concerned, the readings were compiled and tabulated and the respective reverberation time frequency graphs were plotted. The results of rooms are given in the table and figures named below:-

Measurement for Part III Lecture Room	(Tables - Tables [10,(1-4)] (Graphs - Figs. [10,1(a-d)]
Measurement for Part II Lecture Room	(Tables-Tables [10,(5-8)] (Graphs -Figs.(10,(10,2(a-d)
,, ,, The departmental Library	(Tables -Tables[10,(9 -10)] (Graphs - Figs.[10, 3(a - b)]
,, ,, ,, Assembly Hall	(Tables -Tables [10,(1 - 12)] (Graphs - Figs. [10 4(a - d)]
,, ,, ,, Lecture Room	(Tables -Tables (10,13) (Graph - Fig. (10,5))
,, ,, ,, High Voltage Lab.	(Table - Table (10,14) (Graph - Fig. (10,6))

CHAPTER 11

ANALYSIS OF RESULTS

The readings obtained from the measurements made in different rooms were used to plot the graphs of reverberation time against frequency and all the graphs are displayed in figs.[10: (1 - 6)]. In addition to these graphs, a graph showing the recommended optimum reverberation times (for speech, sound broadcasting, music and television) for different volumes of rooms (After Knudsen & Harris) is shown in figure (11:1). The dimensions of each room where reverberation time measurements were made were afterwards measured and their measured reverberation times compared with the optimum reverberation times obtained from fig. (11.1).

11.1. R. T. Measurement in Part III Lecture Room

11.1.1 Night Measurement

(a) Empty Room

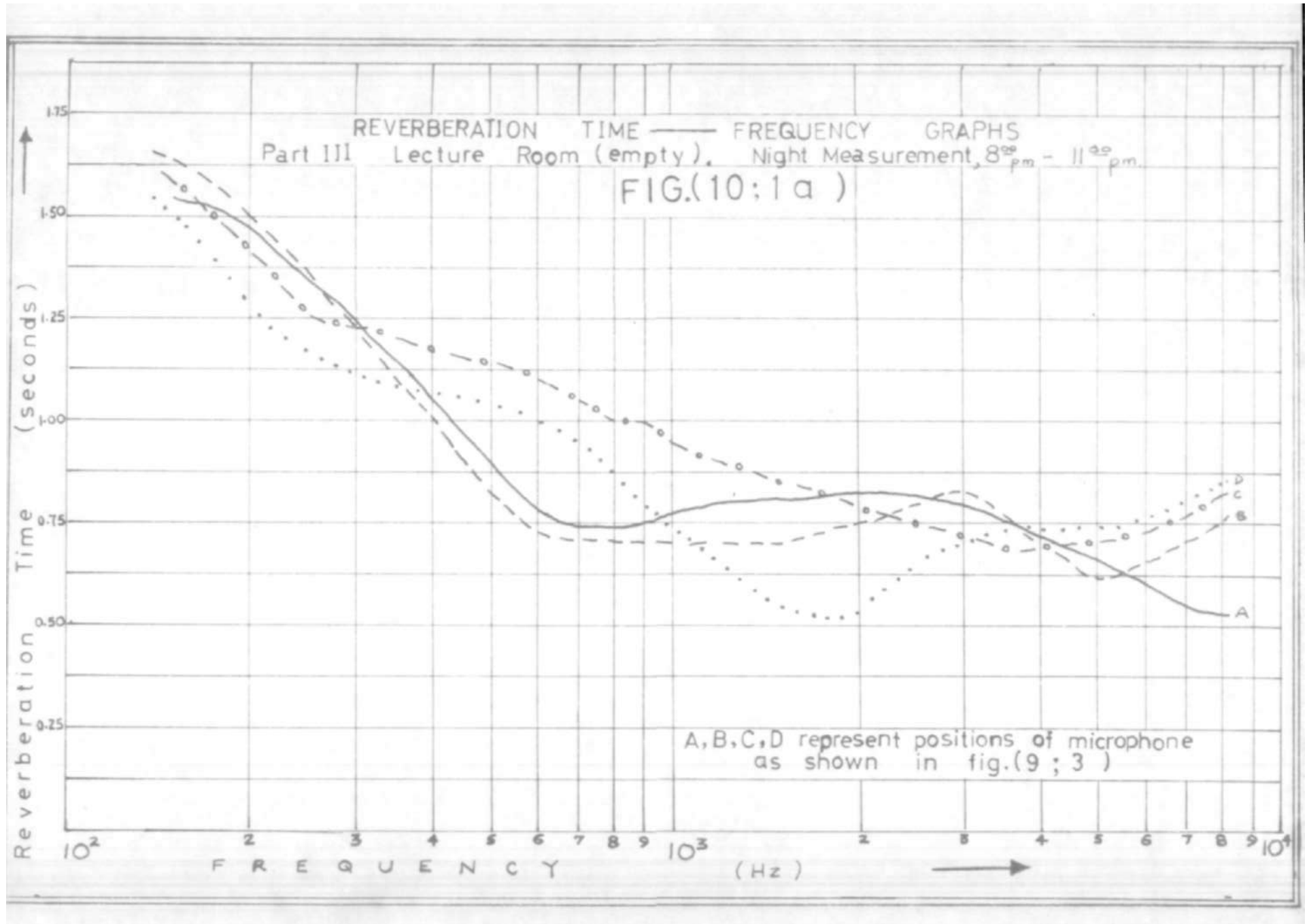
The graphs shown in fig. (10;1a) for the empty room measurement gives the variation of the reverberation time with frequency at each of the four positions A, B, C, & D in the room. From these graphs, it is observed that the reverberation is not uniform over the audio-frequency range considered. The reverberation times are high at the low frequency section of the graphs and they decrease in value as the frequency increases. However, at higher audio-frequencies, they assume almost a constant value as shown in fig.(10,1a).

The average reverberation time of the room is obtained from the graphs at an intermediate frequency of 1000Hz. This can thus be compared with the average value at a frequency of 200Hz. (denoting a typical low frequency value) as shown in the next page:

REVERBERATION TIME —— FREQUENCY GRAPHS

Part III Lecture Room (empty). Night Measurement, 8^{00}_{pm} - 11^{30}_{pm}.

FIG.(10:1a)



Position of Microphone	R.T. (in secs.) at 1000Hz.
A	0.80
B	0.70
C	0.95
D	0.75
Total =	3.2secs.
Average R.T. =	0.8secs.

Position of Microphone	R.T. (in secs.) at 200 Hz.
A	1.45
B	1.50
C	1.40
D	1.30
Total =	5.65secs.
Average R. T. =	1.41secs.

i.e. The average reverberation time of the room (at the intermediate frequency of 1000Hz) = 0.80sec. and the average R.T. at 200Hz. (low frequency) = 1.41sec.

The dimensions of the room are given by:-

Length of room = 40ft.

Breadth ,, ,, = 31.2ft.

Height ,, ,, = 12ft.

∴ Volume ,, ,, = 15,000cub.ft.

Since this room is a lecture room, it is supposed to be designed for speech and from figure (11;1), the optimum reverberation time corresponding to a room volume of 15000 cub.ft. = 0.7sec.

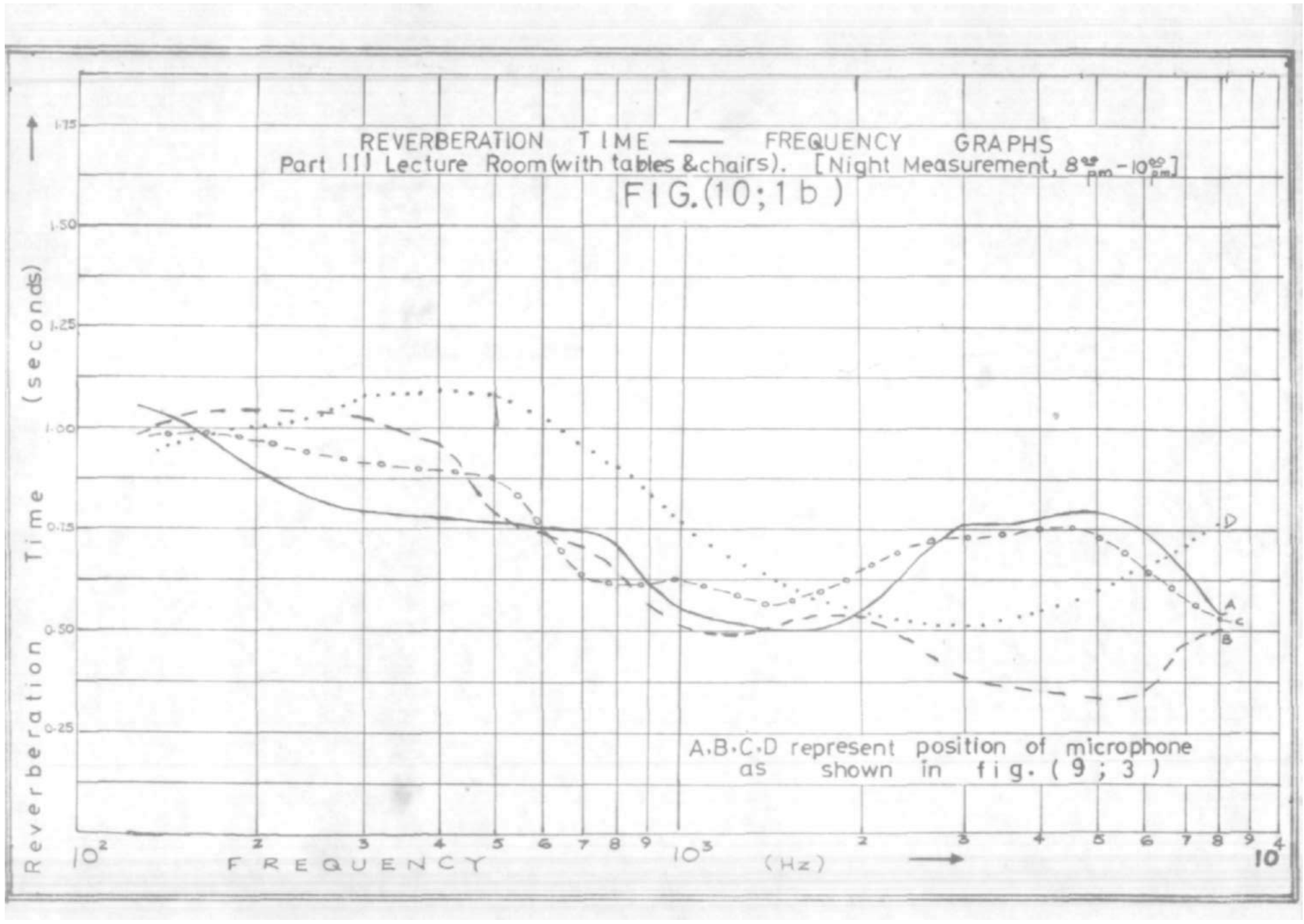
This differs little from the measured reverberation time of 0.8sec. at 1KHz., and it may be concluded that at this frequency, and higher audio-frequencies considered, intelligibility of speech is not badly affected. On the other hand, at 200Hz, the measured reverberation time is much greater than the recommended optimum value of the reverberation time and lack of clarity of speech results at these low frequencies.

(b) Room with Tables and Chairs.

The graphs shown in fig.(10·1b) represent the results obtained for the variation of reverberation time with frequency when some absorbent materials like wooden tables and chairs were present in the room. Some of the sound energy falling on the surfaces of the chairs and tables was absorbed while some was reflected.

REVERBERATION TIME —— FREQUENCY GRAPHS
Part III Lecture Room (with tables & chairs). [Night Measurement, 8^{30}_{pm} - 10^{30}_{pm}]

FIG. (10;1b)



From the graphs shown in fig.(10·1b) it is observed that the high values of the reverberation time at the low audio frequency section have decreased considerably. There is a small decrease at higher frequencies too. For comparison, the new average values of the reverberation time at the intermediate and low audio frequencies are given below:-

Position of Microphone	R.T. (in seconds) at 1000Hz.
A	0.58
B	0.52
C	0.63
D	0.75
Total	= 2.48secs.
Average R.T.	= 0.62secs.

Position of Microphone	R.T.(in secs.) at 200 Hz.
A	0.90
B	1.05
C	0.95
D	1.00
Total	= 3.9secs.
Average R.T.	= 0.98secs.

The previous result shows that the low frequency value of the reverberation time decreased from its empty room value by (1.41 - 0.98)sec. = 0.43sec. while the intermediate frequency value decreased by (0.8 - 0.62)sec. = 0.18sec.

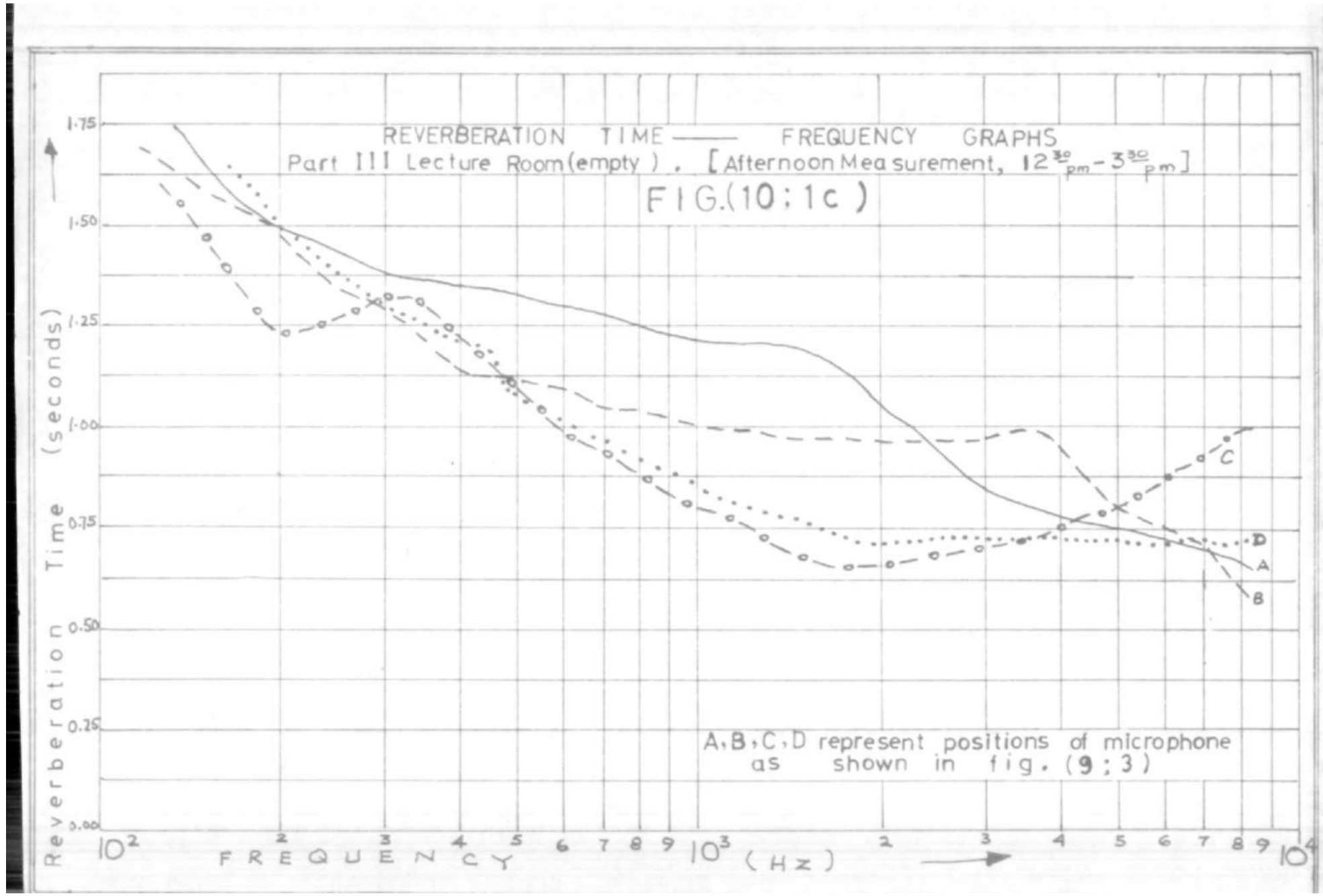
Also comparing the new values of the reverberation time with the recommended optimum value (for speech), i.e. 0.7sec, it can be realized that intelligibility of speech is greatly improved at low frequencies by the inclusion of chairs and tables.

11.1.2 Afternoon Measurements: (Part III Lecture Room)

These measurements were taken in the afternoon between 12.30 p.m. and 3.30 p.m. The results of the measurements are shown in the graphs displayed in figure (10·1c) and figure (10·1d) respectively; Fig. (10·1c) represents the empty room condition of measurement while fig.(10·1d) represents the measurement taken with tables and chairs present.

(a) Afternoon Measurement for Empty Room

The graphs in fig.(10·1c) for the empty room measurement show higher values than the previous ones for the reverberation times at different frequencies.


REVERBERATION TIME —— FREQUENCY GRAPHS
Part III Lecture Room (empty) . [Afternoon Measurement, 12 30 pm - 3 30 pm]
FIG.(10:1c)
Reverberation Time (seconds)
1·75
1·50
1·25
1·00
0·75
0·50
0·25
0·00
C
D
A
B
A, B, C, D represent positions of microphone as shown in fig. (9:3)
10²
FREQUENCY
10³
(Hz)
10⁴

This might be due to the effect of external noise which will tend to prolong the time taken for the sound to die down inside the room.

Resonance also occurs in the room and where the resonant frequency is excited, the sound would stand out distinctly from its original level and it would take a longer time to die down and consequently causing longer values of reverberation time.(This also applies in the evening too). Therefore, in all graphs, the resonant frequencies in the rooms at the positions of measurement are often shown where the reverberation time curves bulge out from the original levels.

The average value of the reverberation time at 1000Hz (intermediate frequency) and at 200Hz (representing a typical low frequency) are obtained as follows:-

Position of Microphone	R.T. (in secs.) at 1000Hz.
A	1.20
B	1.00
C	0.95
D	0.80
Total	= 3.95secs.
Average R.T.	= 0.99secs.

Position of Microphone	R.T.(in secs.) at 200Hz.
A	1.5
B	1.5
C	1.24
D	1.5
Total	= 5.74secs.
Average R.T.	= 1.44secs.

From these results, it is observed that the average intermediate frequency value of the reverberation time obtained in this case is greater than any of the previous cases. The same thing is true of the low frequency value of the reverberation time (i.e. at 200Hz.).

It may not be necessary to compare the results obtained in this case with the recommended optimum value because the reverberation time at any position of the room and at any frequency considered could vary depending on the level of the external noise coming into the room.

(b) Afternoon Measurement for the room.(with tables and chairs)

The graphs in fig.(10·1d) represent the measurment made when the chairs and tables were present in the room. From the graphs, it is observed that the absorbing materials in the room have helped to chop

off or reduce the value of the reverberation time considerably at low frequencies. At higher audio-frequencies, there is also a small reduction in the value of the reverberation time. These results may be better realized from the estimation of the average values of the reverberation time at the intermediate frequency of 1000Hz. and also at a typical low frequency of 200Hz. as given below:-

Position of Microphone	R.T. (in secs.) at 1000Hz.
A	0.95
B	0.85
C	0.75
D	0.70
Total	= 3.25secs.
Average R.T.	= 0.81secs.

Position of Microphone	R.T. (in secs.) at 200Hz.
A	1.15
B	0.90
C	1.00
D	1.30
Total	= 4.35secs.
Average R.T.	= 1.09secs.

One can then compare these values of the reverberation time with those obtained for the empty room (afternoon measurement). For the case of the empty room:-

The average value of R.T. (at 1000Hz.) = 0.99sec.
and ,, ,, ,, ,, (at 200Hz) = 1.44sec.
therefore, the reduction in the reverberation time
(at 1000Hz) = (0.99 - 0.81)secs.
= 0.18secs.
Also the reduction in R.T., (at 200Hz) = (1.44 - 1.09)secs.
= 0.35secs.

The above result thus shows that the reduction in the reverberation time at low frequencies is greater than the reduction at high frequencies.

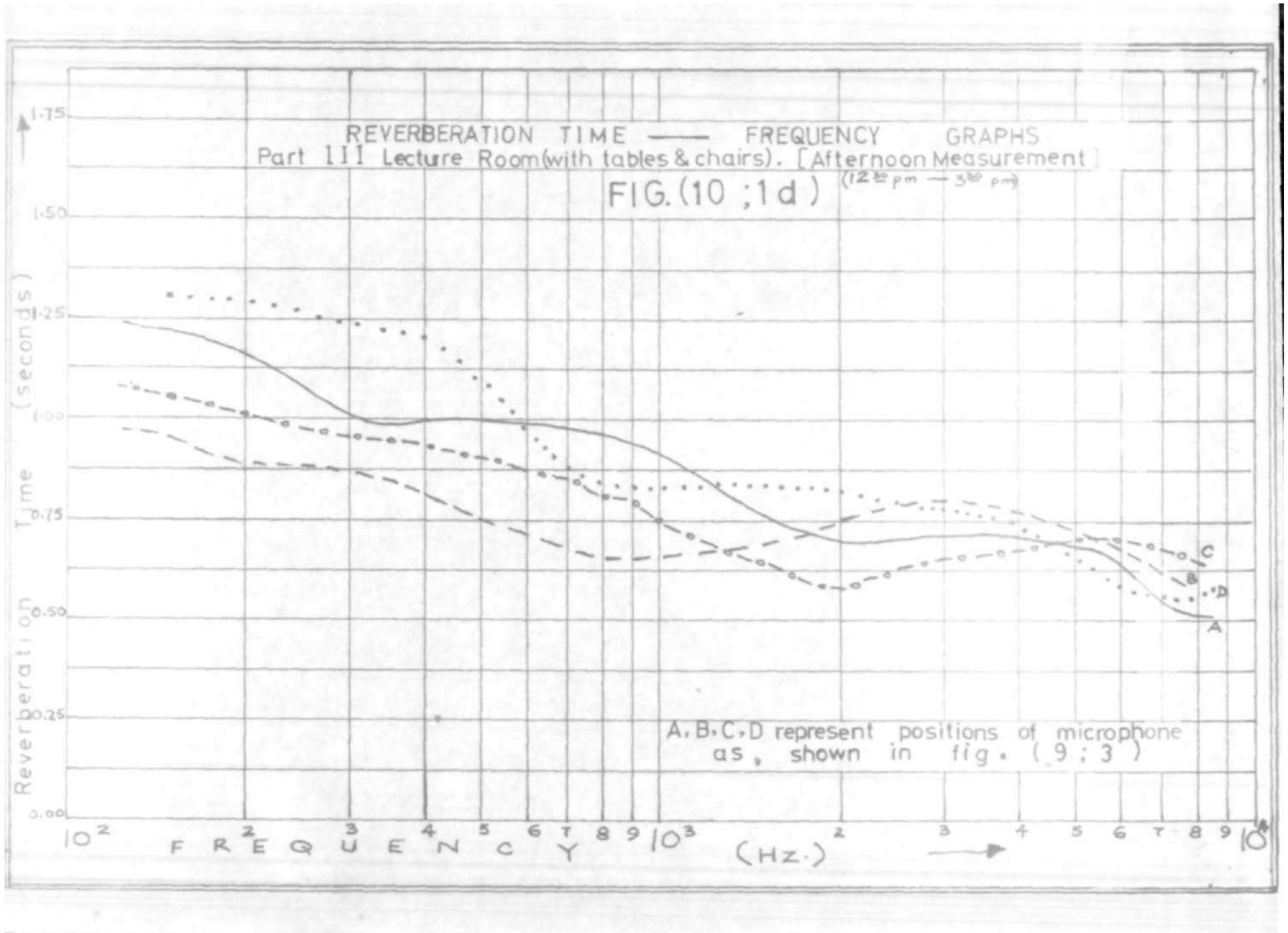
11.2 R. T. Measurement in Part II Lecture Room:

In this case, both night and afternoon measurements were also taken. One set of readings was taken for the empty room and one set for room with tables and chairs. These results are displayed in the graphs of fig. (10:2(a - d)).

REVERBERATION TIME —— FREQUENCY GRAPHS

Part III Lecture Room (with tables & chairs). [Afternoon Measurement (12.30 pm — 3.30 pm)]

FIG. (10 ; 1d)



11.2.1. Night Measurement (Part II Lecture Room)

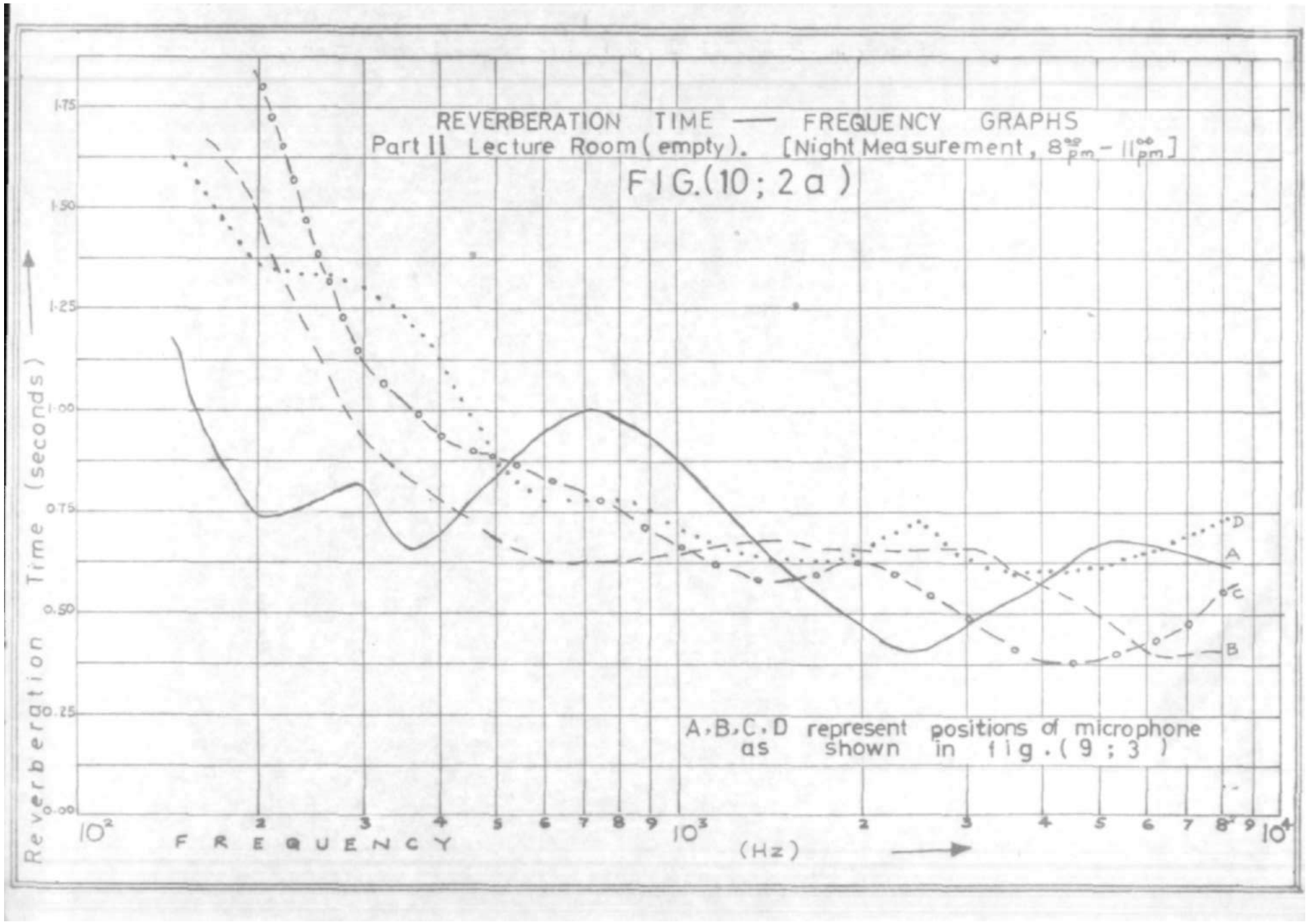
(a) Empty Room

The results of these measurements are represented in the graphs shown in fig.(10;2a). From these graphs, it can be observed that like the case of the Part III lecture room, there are higher values of the reverberation time at low audiO-frequencies and these reduce to an almost constant value at the higher audio-frequencies considered. The average values of the reverberation times are given below for the intermediate and low audio-frequencies.

Position of Microphone	R.T. (in secs.) at 1000Hz.
A	0.88
B	0.65
C	0.65
D	0.72
Total	= 2.90secs.
Average R.T.	= 0.78secs.

Position of Microphone	R.T.(in secs.) at 200Hz.
A	0.75
B	1.50
C	1.80
D	1.35
Total	= 5.40secs.
Average R.T.	= 1.4secs.

These results show that the average R.T. of the room at the intermediate frequency of 1000Hz = 0.78sec. and at low frequencies (e.g. 200Hz) the R.T. = 1.4sec. Since this room has the same dimensions as the Part III lecture room, therefore, their volumes are equal i.e. 15,000 cub. ft. and the corresponding value of the recommended optimum reverberation time for rooms of this volume is equal to 0.74sec. Comparing the result obtained with this optimum value of R.T. it is realized that the average reverberation time obtained from the graphs at the frequency of 1000Hz has just a slight difference from the recommended value which means that speech will be quite intelligible at this frequency and at higher audio-frequencies considered. In the case of lower frequencies intelligibility will be badly affected since there is a significant difference between the recommended optimum value of reverberation time and the value obtained at low frequencies.


REVERBERATION TIME — FREQUENCY GRAPHS
Part II Lecture Room (empty). [Night Measurement, 8pm - 11pm]
FIG.(10;2a)
Reverbergation Time (seconds)
1·75
1·50
1·25
1·00
0·75
0·50
0·25
0·00
D
A
C
B
A,B,C,D represent positions of microphone as shown in fig.(9;3)
10²
2 3 4 5 6 7 8 9
10³
2 3 4 5 6 7 8 9
10⁴
FREQUENCY (Hz)

(b) Room with Tables and Chairs

The Part II lecture room has many things in common with the Part III lecture room. The dimensions of the rooms are the same the materials for the ceilings and floors of the rooms are the same and also other room materials were almost the same during the period of measurements. This fact is also shown in the graphs obtained for the two rooms. The curves for one room resemble the same type for the other. When the tables and chairs were present in the Part II lecture room, there was a noticeable reduction in the value of the reverberation time at low frequencies while there was only a slight difference in the higher audio-frequency values of the reverberation time as shown in fig. (10·2b). This can be shown better numerically as given below:-

Position of Microphone	R.T.(in secs.) at 1000Hz.
A	0.73
B	0.57
C	0.85
D	0.70
Total	= 2.55secs.
Average R.T.	= 0.71secs.

Position of Microphone	R.T. (in secs.) at 200Hz.
A	1.22
B	0.95
C	1.20
D	1.17
Total	= 4.54secs.
Average R.T.	= 1.4.secs.

The results show that the average values of the reverberation time of the room (with tables and chairs) at 1000Hz and 200Hz. respectively are 0.71secs., and 1.14secs. It can therefore be seen that the reverberation time of the room at the mid-frequency and higher audio-frequency range can be favourably compared with the recommended optimum reverberation time for the room, but at low audio-frequencies, this breaks down because the reverberation time of the room is well above the recommended optimum value of the reverberation time for speech. This room thus has poor acoustic qualities for speech at low frequencies.

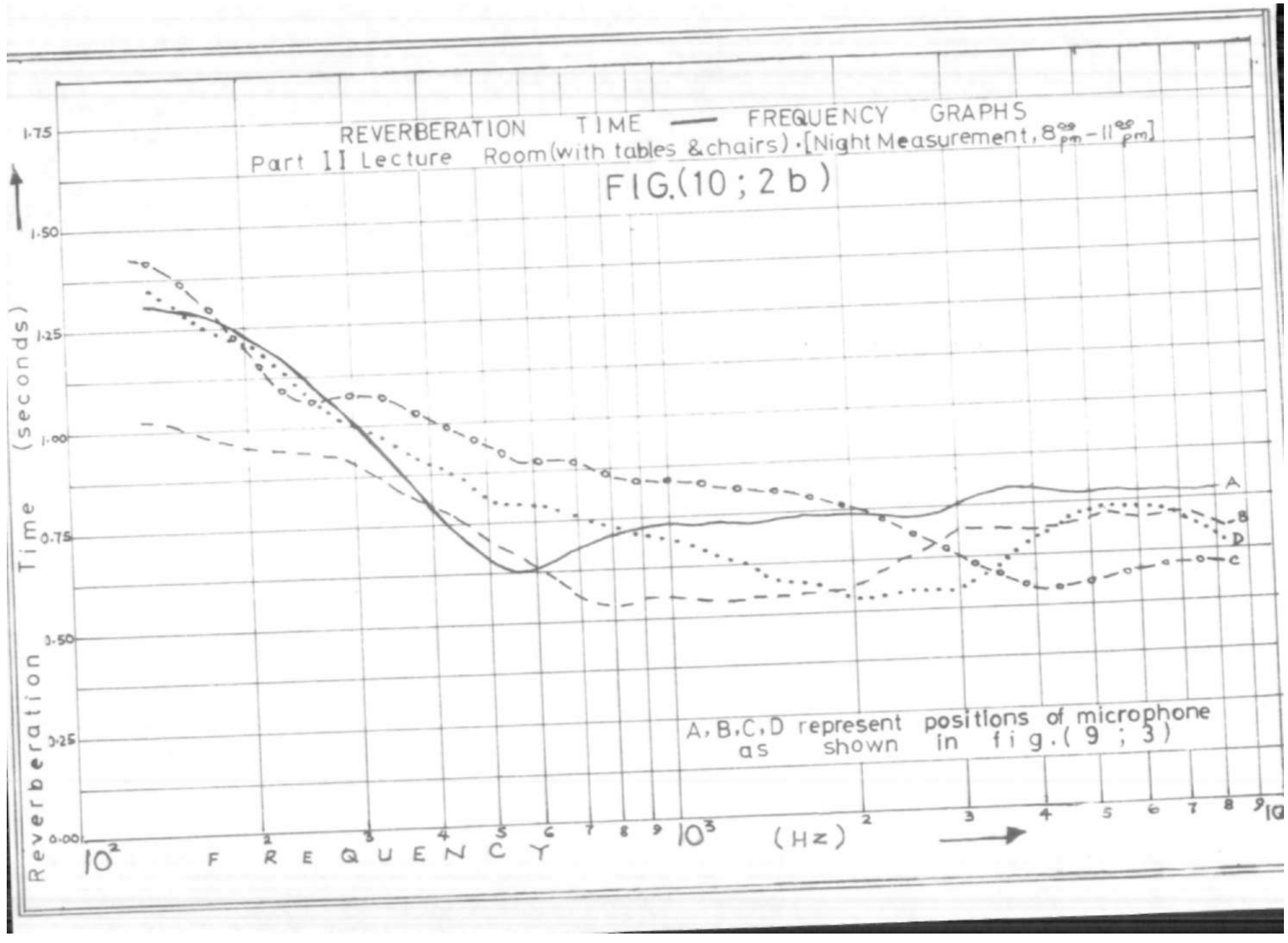
11.2.2. Afternoon Measurements (Part II Lecture Room)

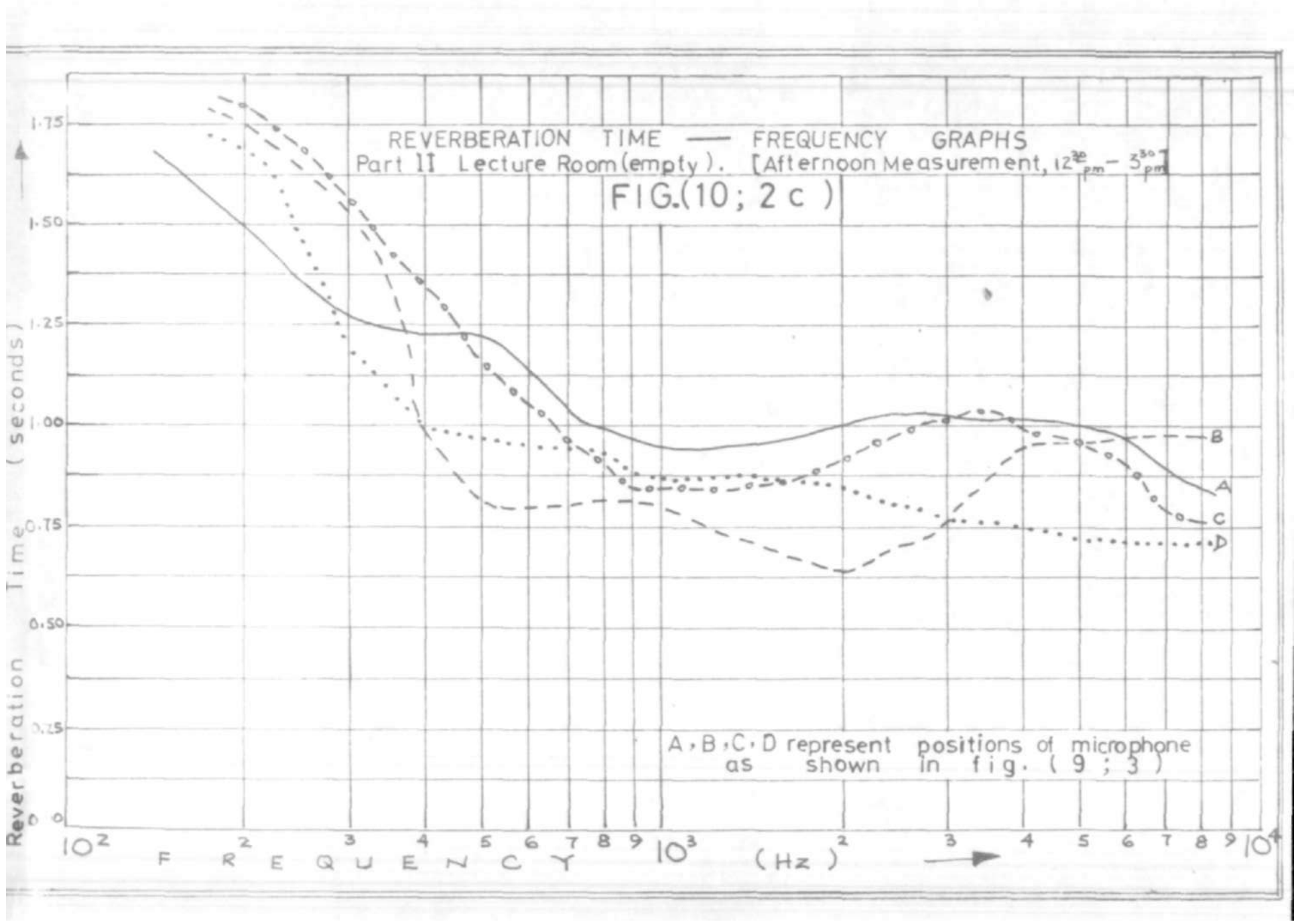
The afternoon measurement for this room were also taken between 12.30 p.m. and 3.30 p.m. and the results of the measurements are shown

REVERBERATION TIME — FREQUENCY GRAPHS

Part II Lecture Room (with tables & chairs). [Night Measurement, $8^{00}_{pm} - 11^{00}_{pm}$]

FIG. (10 ; 2 b)

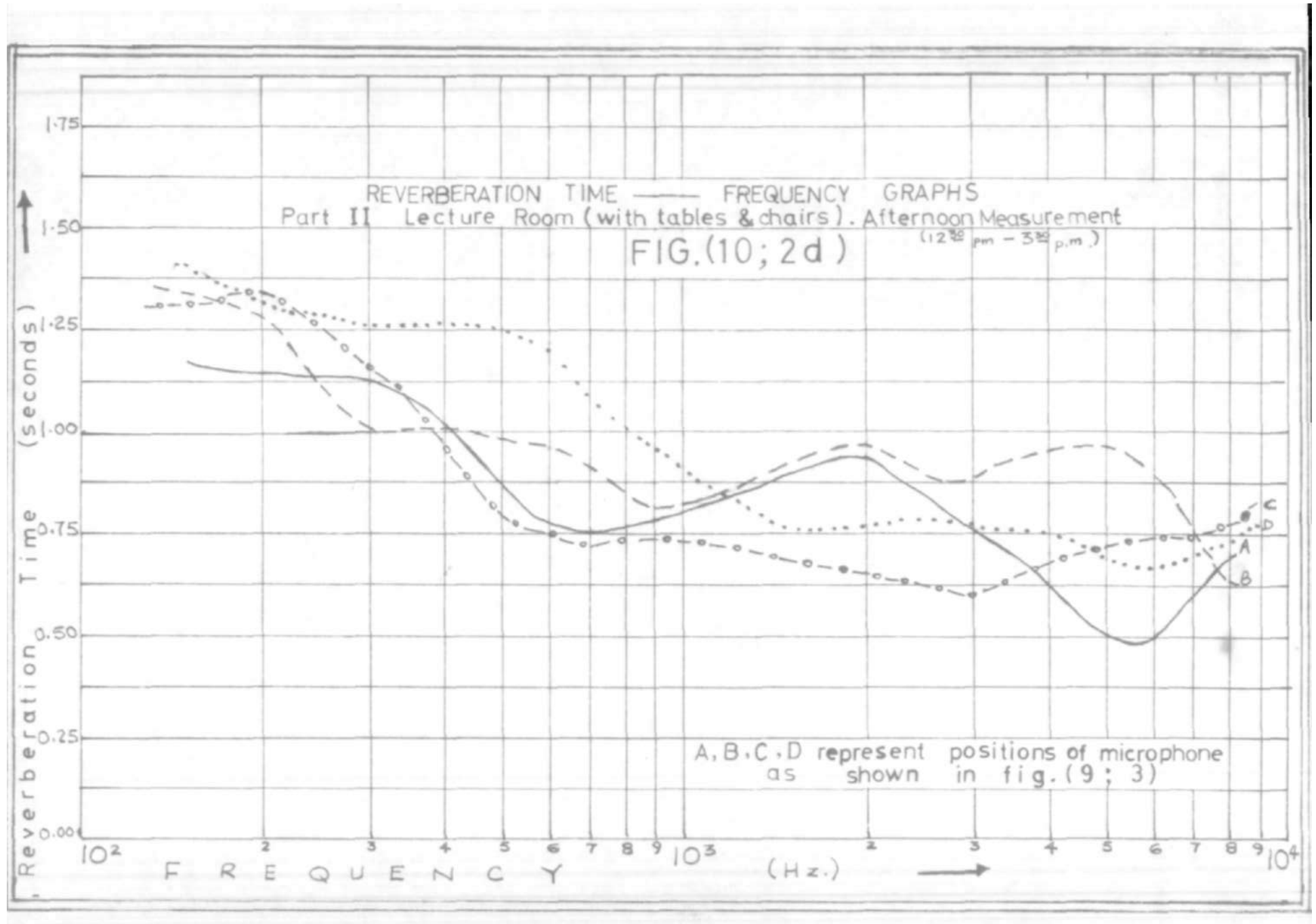



REVERBERATION TIME — FREQUENCY GRAPHS
Part II Lecture Room (empty). [Afternoon Measurement, 12^30 pm – 5^30 pm]
FIG.(10; 2c)
Reverberation Time (seconds)
1·75
1·50
1·25
1·00
0·75
0·50
0·25
0
A
B
C
D
A, B, C, D represent positions of microphone as shown in fig. (9; 3)
10^2
FREQUENCY
10^3
(Hz)
10^4

REVERBERATION TIME —— FREQUENCY GRAPHS

Part II Lecture Room (with tables & chairs). Afternoon Measurement (12³⁰ pm – 3³⁰ p.m.)

FIG.(10; 2d)



in fig. (10:2c) and fig. (10;2d) respectively. The results obtained for this case are similar to those obtained for the Part III lecture room in the afternoon (treated previously.) During the measurements made, external noise was also encountered as in the case of Part III lecture room. The two graphs representing the results of these afternoon measurements also show that at low frequencies, the reverberation time of the room is high and the room lacks quality for speech, but at higher audio-frequencies, the acoustic condition improves considerably. The average reverberation times of the room at mid-frequency and at a typical low frequency (200Hz) are given below (as obtained from the graphs in the two figures:-

Room condition for measurement.	Average R.T. (in secs.) at 100Hz	Average R.T.(in secs.) at 200Hz (low frequency)
Empty Room	0.87	1.7
Room with tables & Chairs	0.80	1.28

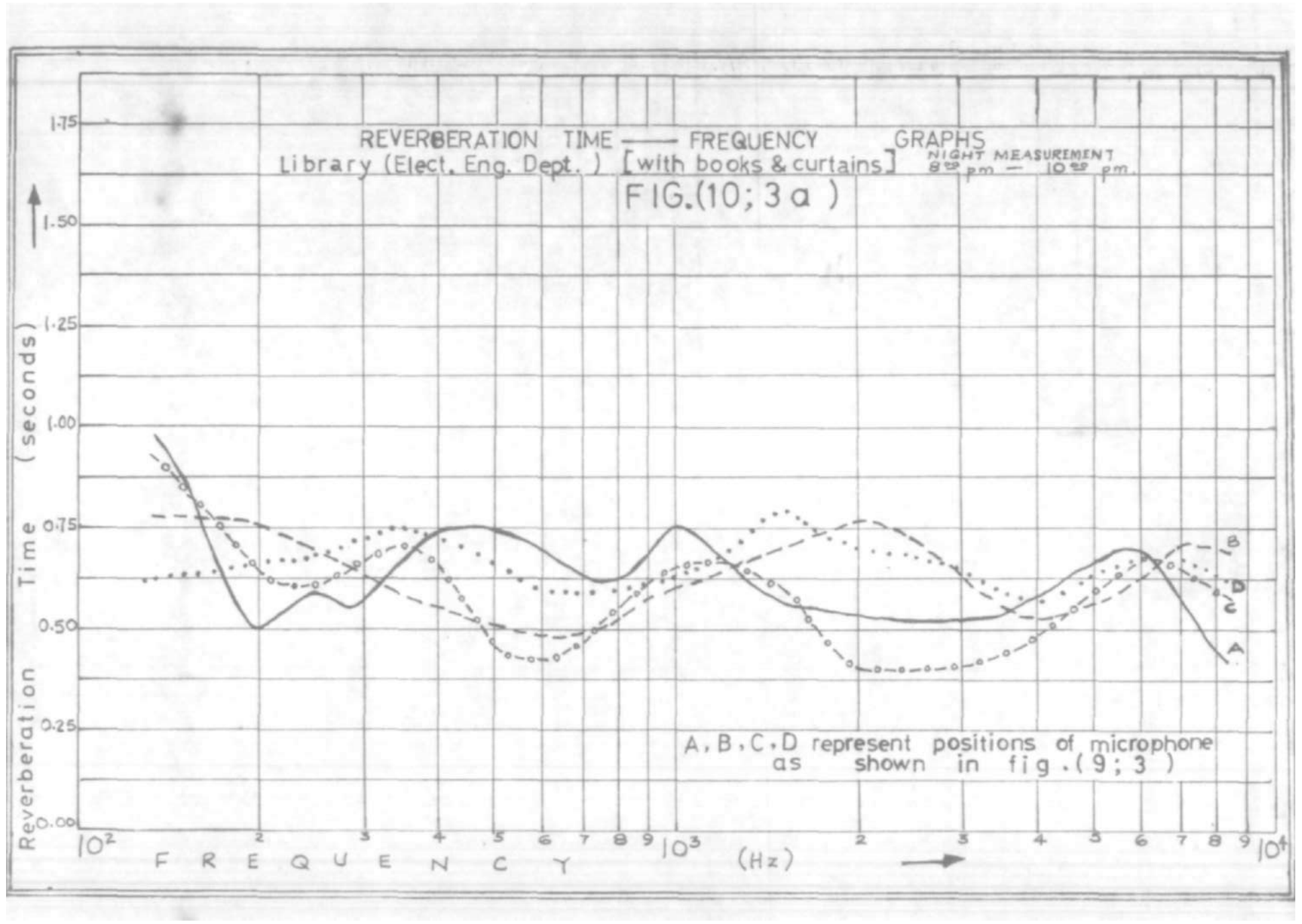
11.3 R.T. Measurement in Electrical Engineering Department Library

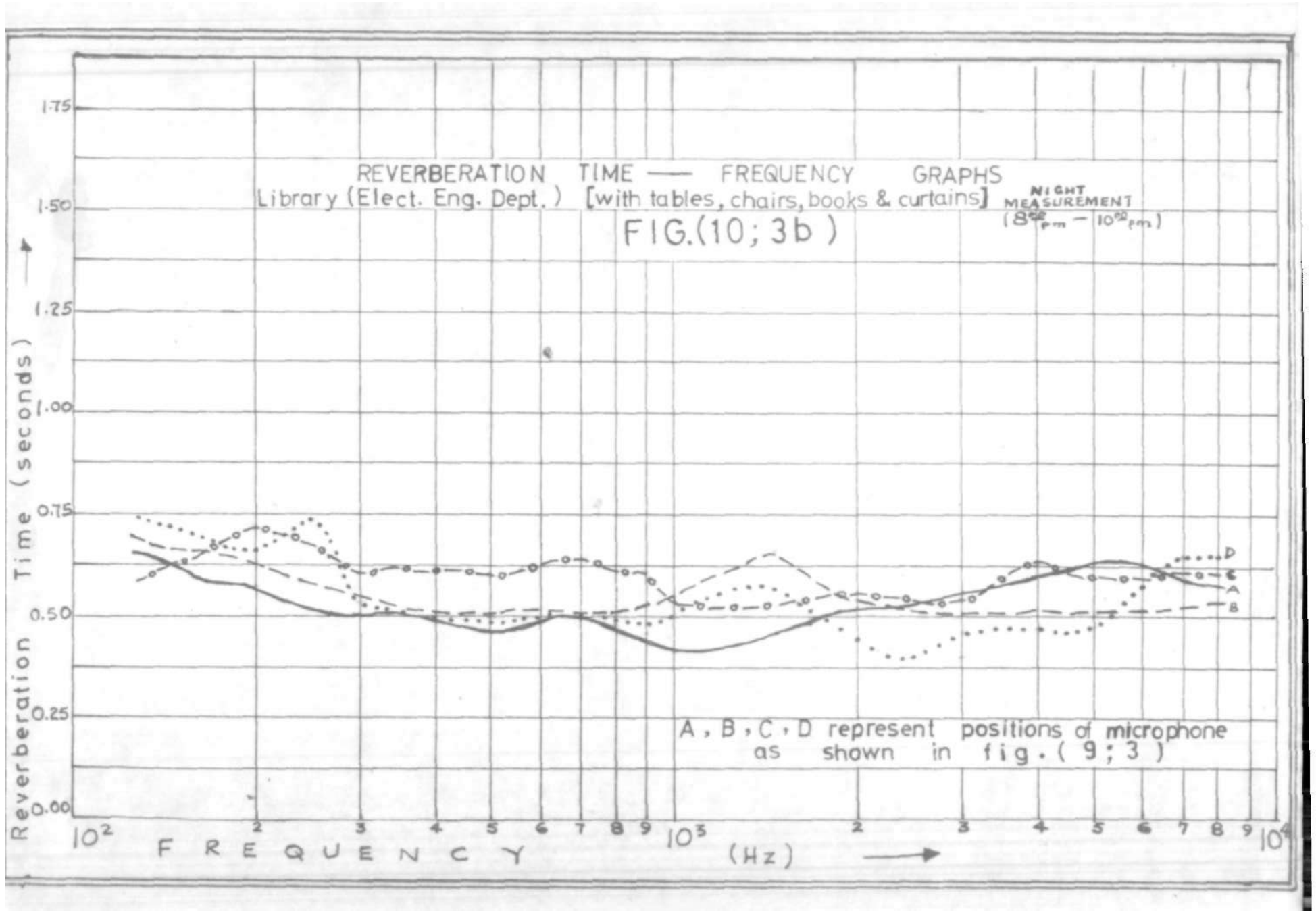
(a) With Books and Curtains

The analysis of the result of this can be made from the graphs in figure (10;3a). From these graphs, it is observed that the average reverberation time over the audio-frequency range considered assumes almost a constant value. The measured value of reverberation time can then be estimated at the intermediate frequency (i.e. 1000Hz) as shown below. This time, it may not be necessary to estimate the value of the reverberation time at low frequencies since it will be almost equal to the value at the intermediate frequency· but it is however given for comparison.

Position of Microphone	R.T. (in secs.) at 1000Hz.
A	0.75
B	0.60
C	0.65
D	0.63
Total	= 2.63secs.
Average R.T.	= 0.66secs.

Position of Microphone	R.T. (in secs.) at 200Hz.
A	0.50
B	0.75
C	0.65
D	0.67
Total	= 2.57secs.
Average R.T.	= 0.64secs.


REVERBERATION TIME —— FREQUENCY GRAPHS
Library (Elect. Eng. Dept.) [with books & curtains]
NIGHT MEASUREMENT
8 pm — 10 pm
FIG.(10;3a)
Reverberation Time (seconds)
1.75
1.50
1.25
1.00
0.75
0.50
0.25
0.00
10²
2 3 4 5 6 7 8 9
10³
2 3 4 5 6 7 8 9
10⁴
FREQUENCY (Hz)
A, B, C, D represent positions of microphone as shown in fig.(9;3)
B
D
C
A


REVERBERATION TIME — FREQUENCY GRAPHS
Library (Elect. Eng. Dept.) [with tables, chairs, books & curtains]
NIGHT MEASUREMENT
(8^50 pm — 10^50 pm)
FIG.(10; 3b)
Reverberation Time (seconds)
1.75
1.50
1.25
1.00
0.75
0.50
0.25
0.00
D
C
A
B
A, B, C, D represent positions of microphone as shown in fig. (9; 3)
10^2
2 3 4 5 6 7 8 9
10^3
2 3 4 5 6 7 8 9
10^4
FREQUENCY (Hz)

Therefore, the average reverberation time of the room at 1000Hz
= 0.66sec.

Since the length of the library = 34ft
Breadth " " " = 29.5ft
Height " " " = 10ft.
∴ Volume " " " = 34 x29.10)cub.ft.
= 10,000 cub.ft.

considering the graph of the recommended optimum reverberation time (for speech) in fig.(11·1)· for this room, the recommended optimum reverberation time = 0.7sec. Therefore, by comparing this optimum value with the results obtained from the graphs, it can be realised that speech will be quite intelligible over the audio-frequency range considered since the results from graphs compare favourably with the optimum value. This result was achieved because of the presence of books and curtains inside the room. They help to chop off the high values of the reverberation time at low frequencies.

(b) Measurement with Tables and Chairs Inside

The result of this measurement is also shown by the graphs in fig.(10·3b). From the graphs, it is observed that like the previous case, the reverberation time is almost constant over the audio-frequency range considered. In this case, the value of the reverberation time is lower than in the former case. Room resonance effect can also be observed in these curves and they are shown by the occasional rising up in the curves.

11.4. R.T. Measurement in the A.B.U. Assembly Hall.

Owing to the fact that the Assembly Hall was already booked for the nights, measurements could therefore not be performed at the period. The measurements were therefore made in the evening and during the time,

external noise from the surrounding was negligible. Readings were obtained for the cases of

(a) Empty Hall

(b) Hall with stage curtains and chairs.

The results of measurement are shown in the graphs given in fig (10;4a) and fig. (10;4b) respectively.

(a) R.T. Measurement for Empty Hall

This measurement was taken between 4.00 p.m. and 6.30 p.m. and graphs of the results are shown in fig.(10;4a). These graphs also show that at low audio frequencies, the reverberation times are high and they decrease in value as the frequency increases. The average reverberation times at the intermediate frequency of 1000Hz and also at a typical low frequency of 200Hz are estimated below.

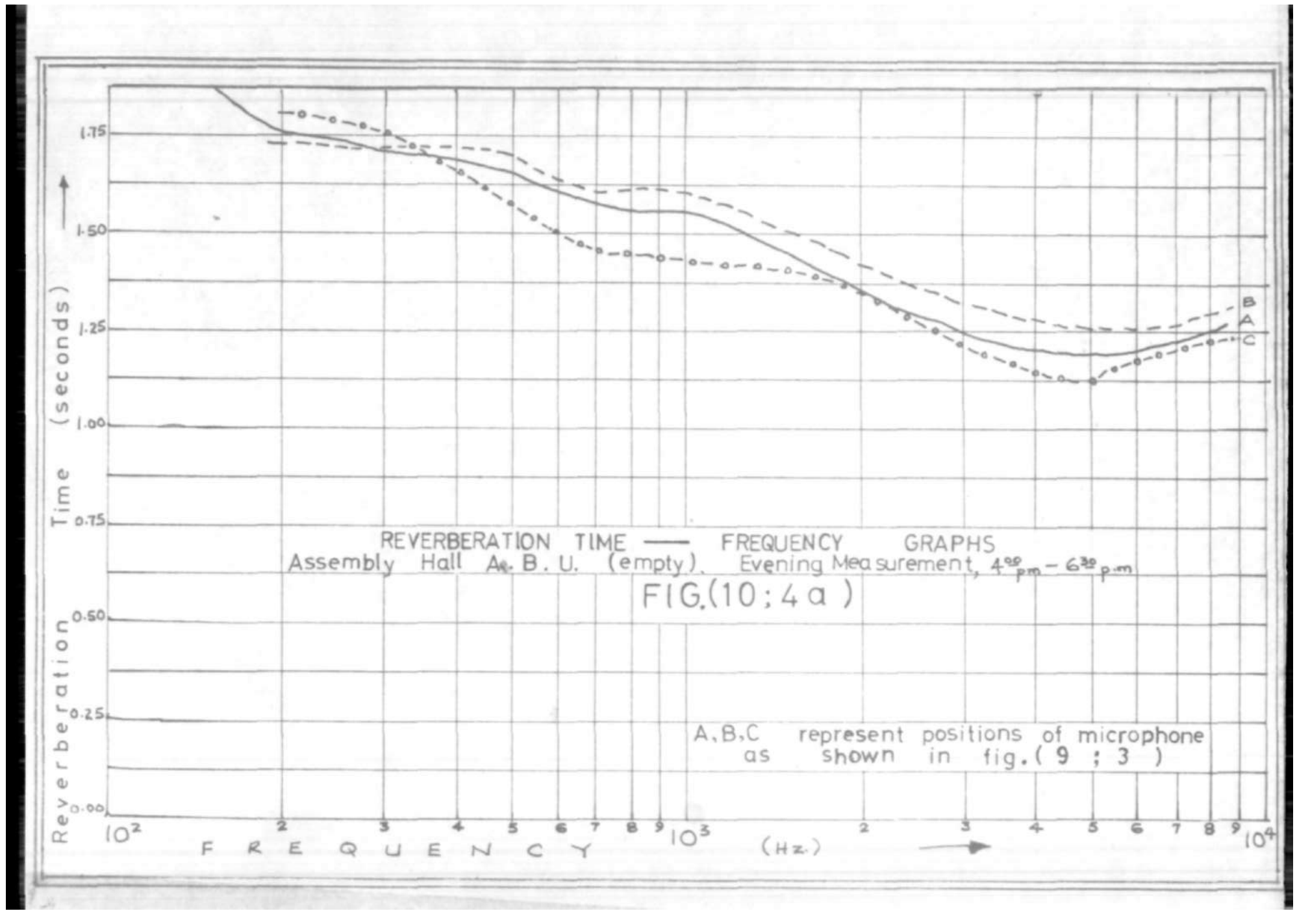
Position of Microphone	R.T. (in sec.) at 1000Hz.
A	1.55
B	1.60
C	1.45
Total	= 4.6secs.
Average R.T.	= 1.53secs.

Position of Microphone	R.T.(in sec.) at 200Hz.
A	1.75
B	1.72
C	1.81
Total	= 5.28secs.
Average R.T.	= 1.76secs.

The dimensions of the hall are given below:-

Length of Hall = 130ft.
Breadth,, ,, = 41ft.
Height ,, ,, = 50ft.
∴ Volume ,, ,, = 130 x 41 x 50 cub.ft.
= 260 x 10^3 cub. ft.
= 300 x 10^3 cub. ft.

From the graph of the recommended optimum reverberation time against the room volumes in fig. (11;1), the value of the optimum reverberation time of a room of volume 300 x 10^3 cub.ft is 0.97sec. (i.e. for speech). Comparing this with the results obtained above for empty room, it can be seen that when the hall is empty, it is not



REVERBERATION TIME —— FREQUENCY GRAPHS
Assembly Hall A. B. U. (empty). Evening Measurement, $4^{00}_{pm} - 6^{30}_{p.m}$

FIG.(10;4a)

A, B, C represent positions of microphone as shown in fig.(9;3)

suitable for speech purposes. However, for rooms having such volumes, the optimum reverberation time as obtained from fig. (11;1) for music and sound broadcasting is 1.6sec. for each.

This thus compares favourably with the results obtained from graphs showing that the Assembly Hall when empty is quite suitable for sound broadcasting and music purposes over the audio-frequency range considered. The reverberation times are however too long for speech.

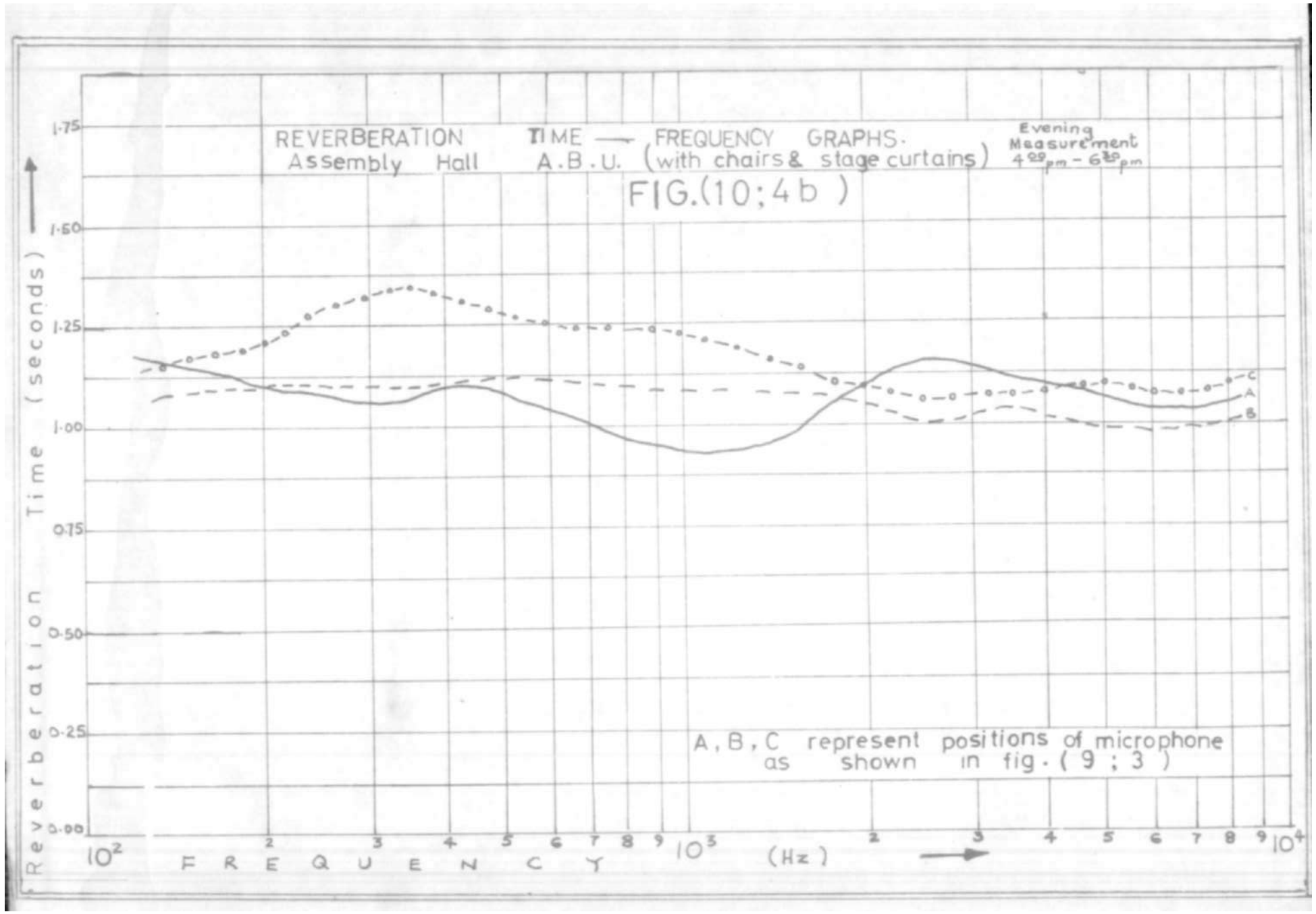
(b) R. T. Measurement For the Assembly Hall with Chairs and Stage Curtains.

The results obtained here are shown in the graphs given in fig. (10;1b). From these graphs, it is observed that the presence of those materials in the Hall helped to reduce the high values of the reverberation time at low frequencies. There is also some reduction in the value of the reverberation times at higher audio-frequencies too. The average values of the reverberation time at low and intermediate frequencies considered (i.e. 200Hz. and 1000Hz respectively) are obtained as shown below:

Position of Microphone	R.T. (in secs.) at 1000Hz.
A	0.93
B	1.10
C	1.22
Total	= 3.25secs
Average R.T.	= 1.08secs.

Position of Microphone	R.T. (in Secs.) at 200Hz.
A	1.13
B	1.10
C	1.20
Total	= 3.4secs.
Average R.T.	= 1.13secs.

Since the value of the recommended optimum reverberation time (for speech) for the hall = 0.97sec, It is realized that the results obtained from graphs do not differ much from the optimum value. This shows that with certain absorbent materials in the room (like chairs and the curtains at the stage) the reverberation time in the room is reduced; and the acoustic conditions in the room are then made favourable for speech purposes.


REVERBERATION TIME — FREQUENCY GRAPHS.
Assembly Hall A.B.U. (with chairs & stage curtains)
Evening Measurement 4.29 pm - 6.39 pm
FIG.(10;4b)
Reverberation Time (seconds)
1·75
1·50
1·25
1·00
0·75
0·50
0·25
0·00
A
B
C
A, B, C represent positions of microphone as shown in fig. (9;3)
10²
2 3 4 5 6 7 8 9
10³
2 3 4 5 6 7 8 9
10⁴
FREQUENCY (Hz)

11.5 The Small Lecture Theatre (Faculty of Engineering)

In this case, night measurement was made and only one set of reading was taken because the chairs and tables in the room are fixed to the ground and it was therefore difficult to make measurments at different parts of the room. The only set of readings taken was obtained with the microphone placed on the lecturer's table in front of the room (i.e. near the blackboard). The measurement was taken in the night between 8.00 p.m. and 10.00p.m. The readings obtained were used to plot a graph of reverberation time against frequency as shown in fig.(10;5). From this graph, it is observed that the reverberation time at the intermediate frequency of 1000Hz = 0.69sec. and the value of the reverberation time at a typical low frequency (e.g. 200Hz.) is 0.72sec.

But the room dimensions are:-

Length of room = 48.6ft
Breadth ,, ,, = 47 ,,
Average Height ,, ,, = 12 ,,

∴ Volume ,, ,, = 48.6 x47 x12 cub.ft.
= 27×10^3 cub. ft.

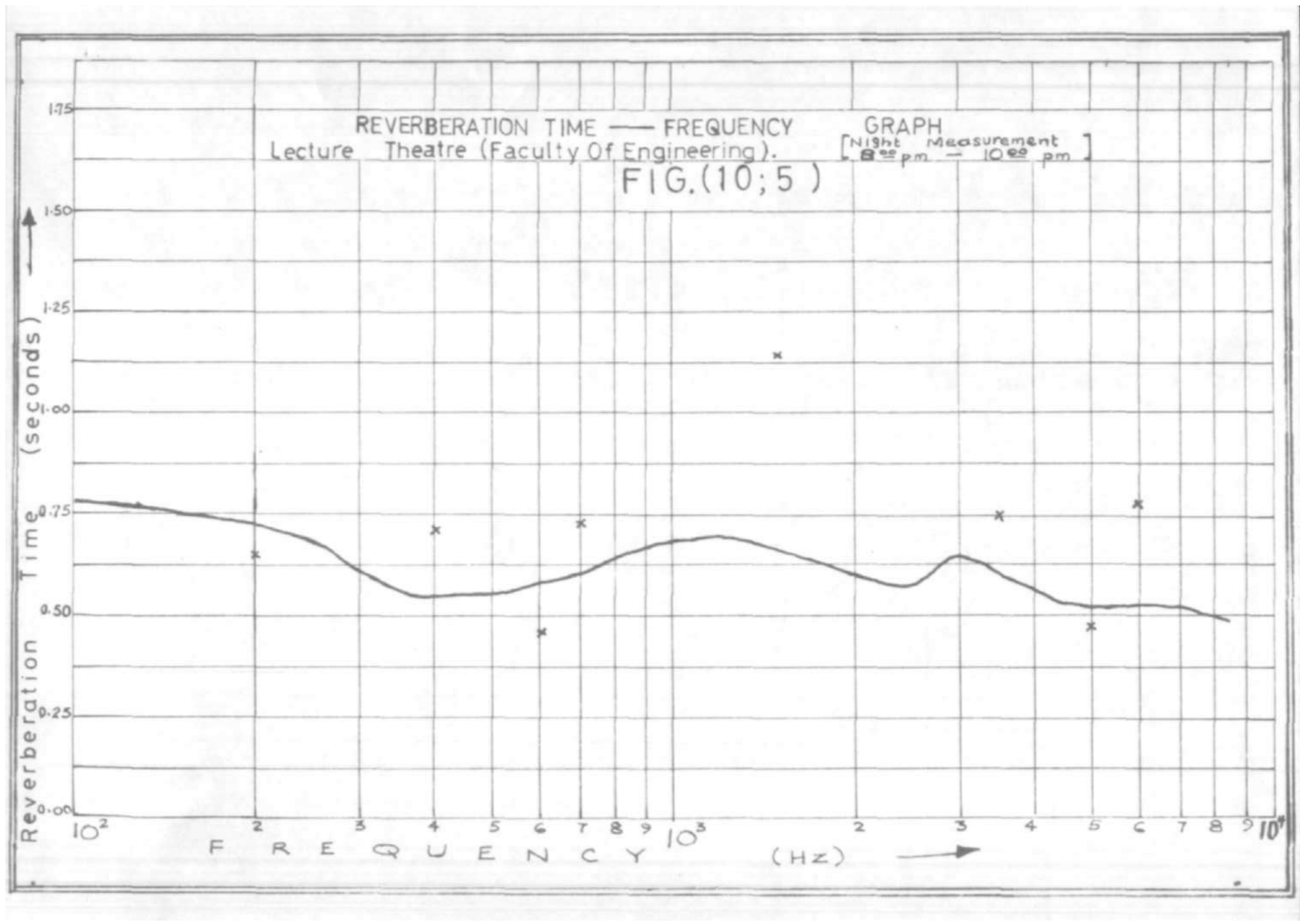
Using the graph of the recommended optimum reverberation times for different room volumes given in fig. (11·1), the optimum reverberation time for speech for a volume of 27×10^3 cub.ft. is 0.76sec. Therefore, the reverberation times obtained by the measurement taken in the Lecture Threatre both at low and higher audio-frequencies compare favourably with the recommended optimum value of reverberation time for the room. This means that the room has excellent acoustic qualities for speech purposes over the audio-frequency range.

11.6 R. T. Measurement for the Higher Voltage Laboratory

This was the room in which the last set of measurment was made. The measurement was performed in the night: between 8.00 p.m. and 10.00p.m. Only one set of readings could also be taken here as a result of the

REVERBERATION TIME —— FREQUENCY GRAPH
Lecture Theatre (Faculty Of Engineering). [Night Measurement 8^00 p.m — 10^00 p.m]

FIG.(10;5)



locations of certain High Voltage equipment which made the shifting of the apparatus about in the room difficult. However, the readings were taken with the microphone placed in the middle of the laboratory. A graph of the reverberation time of the room against frequency was plotted and this is shown in fig. (10:6). From this graph, it is observed that the values of the reverberation time of the room at different frequencies are very high. At the intermediate frequency in the graph (i.e. 1000Hz.), the reverberation time is 2.6secs. while at a typical low frequency of 200Hz, the reverberation time is 3.0secs.

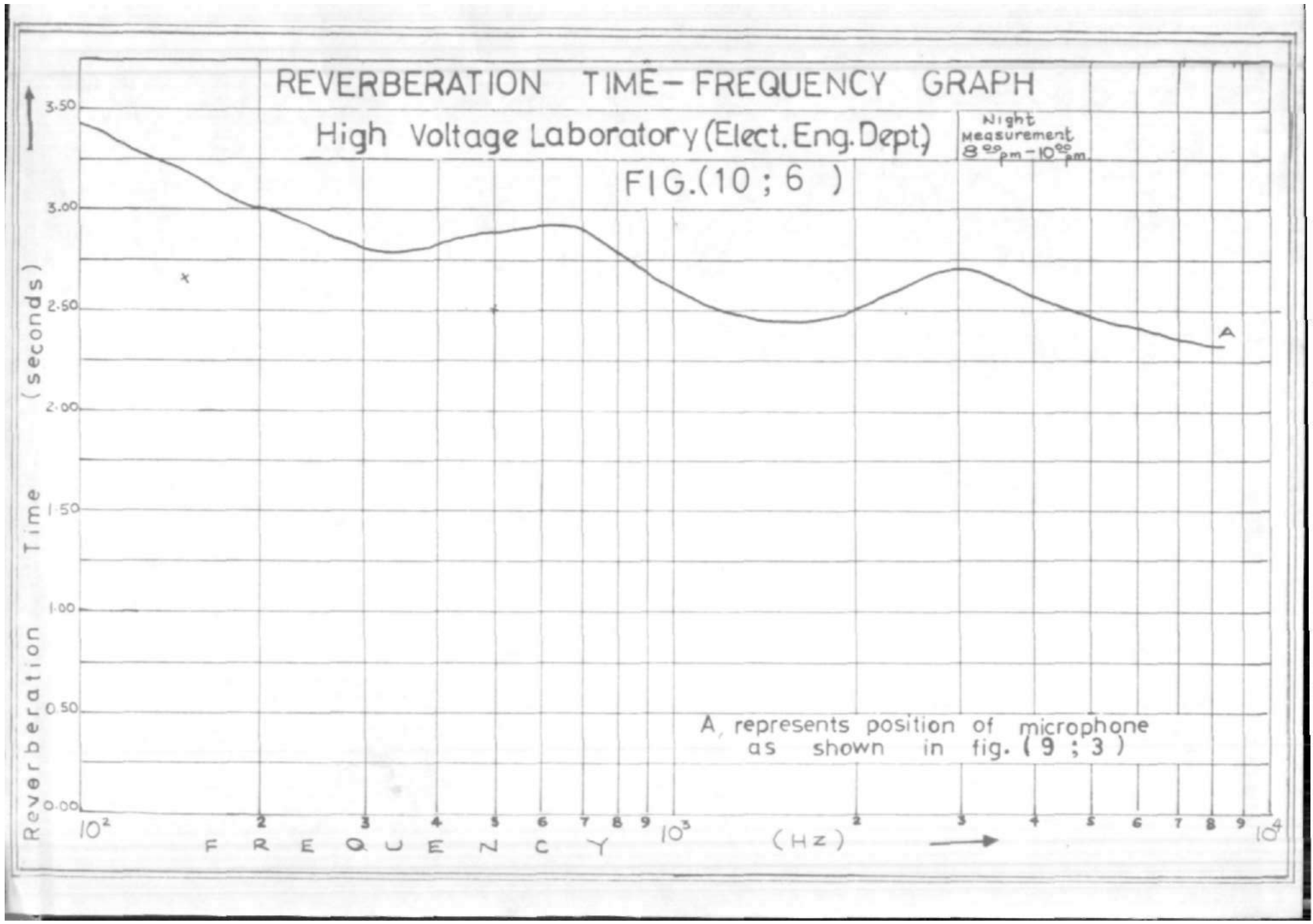
The room dimensions are given by:-

Length of the room = 39.6ft.
Breadth ,, ,, ,, = 39.6,,
Height ,, ,, ,, = 40ft
$\therefore$ Volume ,, ,, ,, = 62×10^3 cub.ft.

Using the graph of the recommended optimum reverberation time in fig. (11,1), for rooms having this volume, the value of the reverberation times are given by: Optimum R.T. for room volume of 62×10^3 cub.ft.

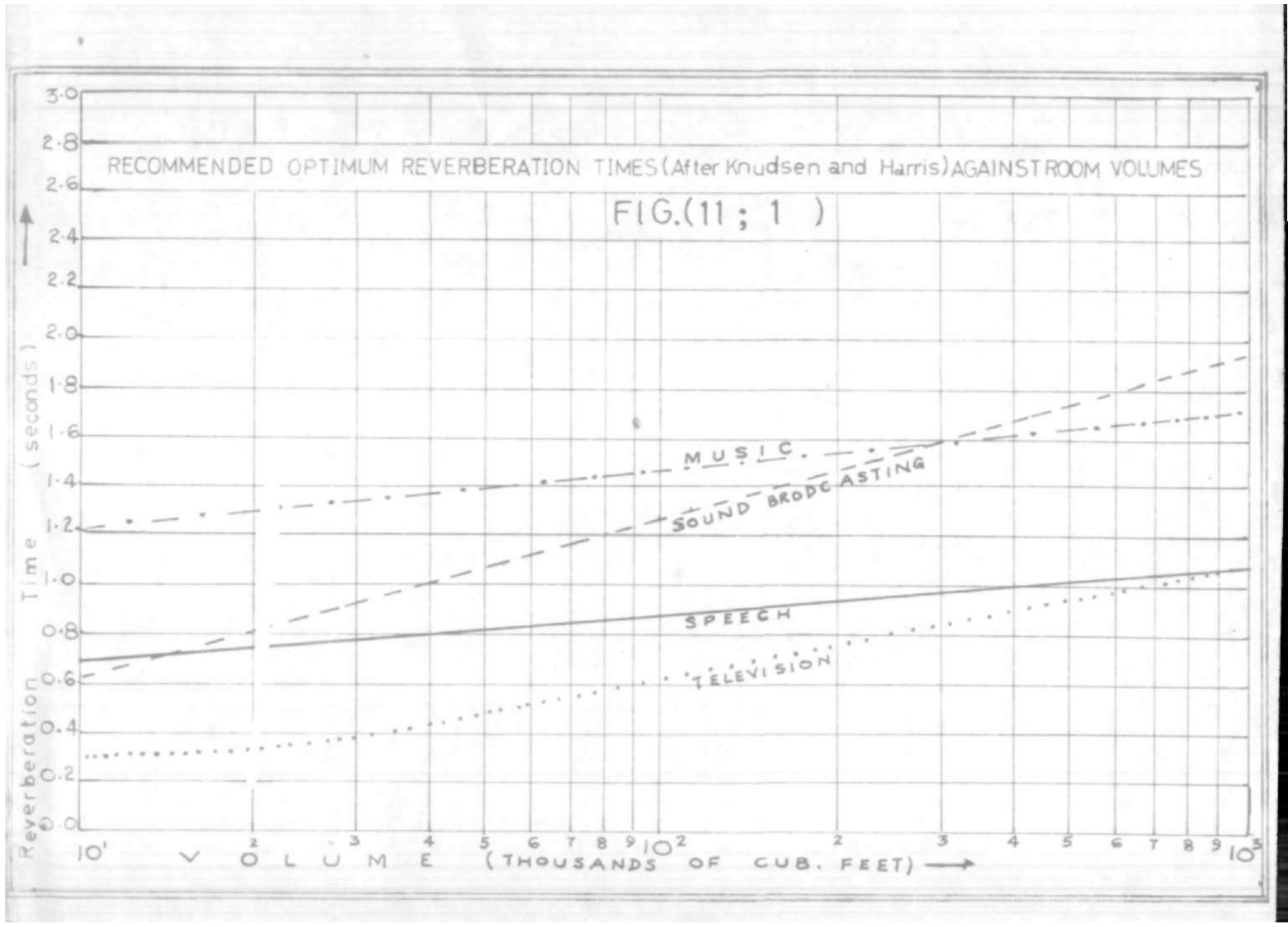
(a) For speech = 0.84 sec.
(b) For sound broadcasting = 1.13 ,,
(c) For music = 1.40 ,,

Comparing each of the above values of the optimum reverberation time with the value of the reverberation time obtained at the mid-frequency (i.e. 1000Hz) of the audio frequency range considered in the graph, it can be deduced that the High Voltage Laboratory, with a reverberation time of 2.6secs. was not designed for any of the purposes mentioned previously since its reverberation time is well above each of the optimum values given. The high value of the reverberation time of this laboratory may be due to the fact that no furnishing is made in the room and further more, the boundaries of the laboratory consist of solid walls only.


REVERBERATION TIME- FREQUENCY GRAPH
High Voltage Laboratory (Elect. Eng. Dept.)
FIG.(10 ; 6)
Night Measurement 8 pm - 10 pm
Reverberation Time (seconds)
3.50
3.00
2.50
2.00
1.50
1.00
0.50
0.00
A
A represents position of microphone as shown in fig. (9 ; 3)
10²
2 3 4 5 6 7 8 9
10³
2 3 4 5 6 7 8 9
10⁴
FREQUENCY (HZ)

RECOMMENDED OPTIMUM REVERBERATION TIMES (After Knudsen and Harris) AGAINST ROOM VOLUMES

FIG. (11 ; 1)



There are no windows through which some of the sound energy produced inside could be transmitted outside. This thus means that most of the sound energy falling on the boundaries of the laboratory is reflected; hence high values of the reverberation time are obtained. Since the laboratory is large, echoes of long delay of the reflected sound are often created and these tend to increase the time taken for sound to decay and consequently the reverberation time is high.

11.7 Comparison of Results

Part III and Part II lecture rooms (when empty) are examples of rooms in which the acoustic condition is quite unfavourable for speech purposes over the low audio frequency range but improves considerably within the mid-frequency and higher audiO frequency range.

The small Lecture Theatre (Faculty of Engineering) is an example of a room whose acoustic properties allow good qualities for speech. The third case is that of a room with very poor acoustics over the whole audio frequency range. An example of this is the High Voltage Laboratory in the Electrical Engineering Department. In the case of the Part III and Part II lecture rooms, to improve the poor acoustic condition of the room (at low frequencies) it is necessary that selective absorbents which are effective over a small frequency range are provided to reduce the reverberation time over the low frequency range where the reverberation times are high. These absorbents should consist of fibrous materials placed in form of pads or blankets behind perforated facing materials or resonant absorbents consisting of layers of materials such as felt placed over certain enclosed air space. These would bring the reverberation time at the low frequency range close to the mid-frequency value and make it suitable for speech purposes over a wide audio frequency range including the low frequency part.

The case of the small lecture theatre where the reverberation time agrees approximately with the recommended value (for speech) over the whole audio-frequency range considered can be seen in fig. (10·5). The room thus has excellent acoustic properties for speech purposes. This conditon has been achieved as a result of the presence of certain absorbing materials in the room.

Concerning the case of the High Voltage Laboratory where the reverberation times are very high over the audio-frequency range as can be observed in the graph of fig. (10;6), speech is not intelligible in this room as a result of the long echo delay which tends to prolong the time of decay of sound in the room. To make the room suitable for speech purposes it would therefore be necessary to reduce its height or to equip it with certain absorbent materials which are effective over a wide range of frequency. Thick fibrous materials could be arranged by the walls so that the excessive reflection of sound at the walls constituting long echo delay and destructive interference may be checked. Also, if necessary, the floor of the room should be covered with heavy carpets, and perforated fibre board tiles can also be arranged at the ceiling of the room. All these will help to reduce the the high values of the reverberation time in the room and will make the room perfectly suitable for speech purposes:

11.8 Calculation of R.T. Using Sabine's Formula.

Sabine's formula for Reverberation Time is given by:-

$$R.T. = \frac{0.049V}{S\ \bar{\alpha}}$$

where V = Volume of room in cub. ft.

S = Total Surface area

$= S_1 + S_2 + S_3 - - -$

(S_1, S_2, S_3 --- are areas of the various surfaces)

$\bar{\alpha}$ = Average absorption coefficient

$$= \frac{\alpha_1 S_1 + \alpha_2 S_2 + \alpha_3 S_3 \text{ ---}}{S_1 + S_2 + S_3 + \text{------}}$$

(α_1, α_2, α_3 --- are the absorption coefficients of the surfaces S_1, S_2, S_3 --- resp.)

If A = total absorption $= \alpha_1 S_1 + \alpha_2 S_2 + \alpha_3 S_3 + \text{---}$

$\therefore \bar{\alpha} = \frac{A}{S}$ and R.T. $= \frac{0.049V}{S\bar{\alpha}} = \frac{0.049V}{A}$ secs.

In this calculation, two cases will be considered.

(i) The High Voltage Laboratory

(ii) The Small Lecture Theatre (Faculty of Engineering)

(i) <u>High Voltage Laboratory</u>

Volume of the laboratory $V = 62 \times 10^3$ cub. ft.

Area of the ceiling S_1 (consisting of fibre boards) = 1600sq.ft.

Absorption coefficient of fibre board ($\frac{1}{2}$" thick) $\alpha_1 = 0.3$ (at 1000Hz.)

$\therefore \alpha_1 S_1 = 1600 \times 0.3 = 480.$

Total surface area of concrete walls, $S_2 = 6400$ sq.ft.

Absorption coefficient of concrete, $\alpha_2 = 0.05$ (at 1000Hz.)

$\therefore \alpha_2 S_2 = 6400 \times 0.05 = 320$

Surface area of the cement floor, $S_3 = 1600$ sq.ft.

Absorption coefficient of cement floor, $\alpha_3 = 0.05$ (at 1000Hz).

$\therefore \alpha_3 S_3 = 1600 \times 0.05 = 80$

Total surface area of 9 windows (5' by 3' each) covered with black canvas, $S_4 = 135$ sq.ft.

Absorption coefficient of canvas, $\alpha_4 = 0.25$ (at 1000Hz)

$\therefore \alpha_4 S_4 = 135 \times 0.25 = 34.$

$$\therefore A = \alpha_1 S_1 + \alpha_2 S_2 + \alpha_3 S_3 + \alpha_4 S_4 + \text{----------}$$

$$= 480 + 320 + 34$$
$$= 914.$$

$$\therefore \text{R. T.} = \frac{0.049 \times 62 \times 10^3}{914} \text{ secs.} = 3.3 \text{ secs.}$$

But the value of the R.T. obtained from the graph at 1000Hz is 2.6 secs. The difference between the calculated value and the measured value of the R.T. could be due to the effect of the absorption which took place as a result of the presence of some High Voltage equipment during measurement .

Also at a low frequency of about 100Hz.

$\alpha_1 = 0.3, \alpha_2 = 0.05, \alpha_3 = 0.05, \alpha_4 = 0.05$

$\therefore A = 480 + 320 + 80 + 6.7 = 886.7$

$\therefore \text{R.T.} = \frac{0.049 \times 62 \times 10^3}{886.7} = \text{secs.} = 3.42\text{secs.}$

Results from the graph show that at 100Hz. R.T. is 3.4secs. In this case the calculated value of R.T. is approximately in agreement with the value from graph.

(ii) The Small Lecture Theatre (Faculty of Engineering)

Length of room = 48.6ft.

Breadth,, ,, = 46 ft.

Average height of room = 11.5ft.

$\therefore$ Volume of room = 26×10^3 cub.ft.

Plywood materials in the room are:-

(a) the blackboard (16' by 8)
(b) 10 fixed tables (38' by 1')
(c) 189 ,, chairs (3.2' by 1.4')
(d) Lecturer's table.
(e) 4 doors (6.8' by 4.9').

Total surface area of plywood materials, S_1 = 1930 sq.ft.

Absorption coefficient of perforated board, $\alpha_1 = 0.15$(at 1000Hz.).

$\therefore \alpha_1 S_1 = 1930 \times 0.15 = 289$

Surface area of perforated boards, $S_2 = 120$sq.ft.

Absorption coefficient of perforated board, $\alpha_2 = 0.3$(at 1000Hz).

$\therefore \alpha_2 S_2 = 120 \times 0.3 = 36$

Surface area of glass windows, $S_3 = 96$ sq.ft.

Absorption coefficient of glass, $\alpha_3 = 0.03$ (at 1000Hz).

$\therefore \alpha_3 S_3 = 96 \times 0.03 = 3.$

Surface area of concrete walls, $S_4 = 1460$sq.ft.

Absorption coefficient of concrete, $\alpha_4 = 0.04$ (at 1000Hz).

$\therefore \alpha_4 S_4 = 1460 \times 0.04 = 58.4.$

Surface area of ceiling (made of corktiles), $S_5 = 2240$sq.ft.

Absorption coefficient of cork tiles, $\alpha_5 = 0.55$ (at 1000Hz).

Surface area of the floor (made of plastic tiles), $S_6 = 2240$sq.ft.

Absorption coefficient of plastic tiles, $\alpha_6 = 0.05$ (at 1000Hz)

$\therefore A = \alpha_1 S_1 + \alpha_2 S_2 + \alpha_3 S_3 + \alpha_4 S_4 + \alpha_5 S_5 + \alpha_6 S_6.$

$= 289 + 36 + 3 + 58.4 + 1232 + 112. = 1730$

$\therefore$ R. T. $\dfrac{0.049 \times 26.0 \times 10^3}{1730} = \underline{0.73 secs.}$

Since at 1000Hz, the reverberation time obtained from the graph is 0.69· this thus means that the value obtained by using Sabine's equation does not differ much from the measured R.T. at 1000Hz.

11.9 General Performance of Equipment

(a) Effect of Variation of Supply Voltage.

All equipment used in the project worked satisfactorily for a supply voltage of -12 volts. A test was performed to determine the range of the supply voltage which could be used without seriously affecting reverberation measurements. The calibrating R-C circuit of fig. (9.2) was constructed and used to give an exponentially decaying voltage which gave an R.T. of 0.67sec. The circuit was connected to the input of the pre-amplifier. This has the advantage of giving a known decay time with no masking effect from noise which could have been troublesome if the microphone had been used. The variation of reverberation time with frequency was measured for different values of supply voltage as given in table (11;1), and the analysis of the readings is given below.

Supply Voltage	Max.R.T.	Min.R.T.	Average R.T.	Error %
-10 volts	0.57sec	0.29sec	0.38sec.	43
-11 "	0.78 "	0.59 "	0.69 "	3
-12 "	0.79 "	0.63 "	0.70 "	4
-13 "	0.80 "	0.61 "	0.71 "	6
-14 "	0.61 "	0.33 "	0.47 "	30

From these results, it was observed that the reverberation meter could operate well with any of the supply voltages of -11 volts, -12 volts and -13 volts. Greater or lower values of supply voltage resulted in unstable readings.

(b) Effect of Short and Longtime Operation

In order to know the time to be allowed for the equipment to warm up before starting measurement, the variation of R.T. with time was observed for about 30 minutes. The readings are given in table (11;2). From the readings obtained, it was observed that measurement could commence about one or two minutes after switching on the reverberation meter since the reading obtained did not show any marked difference in

the variation of R.T. with time. It is however, not known for how long the equipment could be used without any appreciable change in calibration, but during certain measurements, it was used continuously for about 3 hours.

The variations of R.T. with time was also used to check the stability of measurements. This result showed that the equipment gives stable readings and could therefore, be relied upon for any measurement of reverberation time.

CHAPTER 12.
Conclusion

This method of reverberation time measurement gives direct reading on the timer counter T.C. 7 for the time of sound decay through 10 dB. and from this, the reverberation time is obtained. It has been shown that the method gives satisfactory and stable readings within an error of $\pm 6\%$.

The whole reverberation meter consists of separate units like the signal generator, the power supply unit, the loudspeaker, the timer counter etc., and it is not possible to combine them together to form a single unit because it will be very heavy to carry from one place to another. The combination is not also possible considering the correct positioning of some of the equipment in the room during measurement. For example, the Loudspeaker and the microphone should be placed at different positions in the room.

When sound is produced in a room where measurement is to be made and the sound source is shut off, the exponential decay of the voltage developed in the microphone is converted to a linear decay with time by means of a logarithmic amplifier. This amplifier is preceded by two other linear amplifiers in the circuit i.e. the microphone pre-amplifier and the variable gain amplifier. The logarithmic amplifier could not be made to precede the linear amplifiers in the circuit because it requires a large input voltage but the output voltage from the microphone is very small (i.e. maximum voltage 2mV) Also a set of linear amplifiers which would be required to amplify the output voltage may introduce some distortion to it so that there may be a departure from the expected linear decay with time. In the circuit used, the logarithmic amplifier precedes the diode detector. This position cannot however be changed so that the detector will come before the logarithmic amplifier because if the change is made the logarithmic amplifier will be amplifying a d.c. voltage and this is not desirable.

In the reverberation meter, the key of the two position switch with one break and one make contacts employed for starting the timer counter is operated vertically. This mode of operation is more convenient than the horizontal form.

The same switch is also used to shut off the sound source (the loudspeaker) when the pointer on the level indicator reaches a certain voltage level (4 volts). The stopping of the sound source is thus based on a visual judgement of the pointer position on the level indicator. The readings obtained from measurements could therefore be improved if an automatic control is employed for the voltage level at which the sound source is to be shut off.

When making measurements the knob on the signal generator for controlling the frequency of sound produced in the loudspeaker should be tightly fixed because it may introduce some error if it is slack.

This method provides an easy means of measuring reverberation time. Both the connection and also the operation of the equipment are simple; but for the equipment to be used by any unskilled operator, a means should be devised for the whole reverberation time to be obtained on the timer counter instead of a fraction of it.

As a by-product of the investigation of the performance of the equipment, it was noted that a room should be designed to suit a particular purpose for which it is to be used. For example, for successful and clear production of speech and music, different acoustic conditions are required (Speech requires R.T. of about 0.7 sec. while music requires about 1.5 secs.). In the project, attention was concentrated on lecture rooms which are designed for speech, but the Assembly Hall with a reverberation time of 1.53 secs. when some of the absorbent materials in the hall are removed was also found to be quite suitable for music.

Table (10.1)

Reverberation Time-Frequency Measurements
Part III Lecture Room (EMPTY)
Night Measurement (8.oo p.m. - 11.oo p.m.)

FREQUENCY IN Hz.	REVERBERATION TIME (IN SECONDS)			
	Position A	Position B	Position C	Position D
100	2.06	2.02	1.56	1.87
150	2.08	1.63	0.82	1.78
200	1.03	1.44	1.25	1.24
250	1.74	1.36	0.87	1.00
300	1.24	1.48	1.15	1.00
350	0.87	0.83	1.31	1.07
400	1.25	1.01	1.31	1.05
500	0.58	1.25	1.17	1.25
600	0.78	0.73	0.98	1.25
700	1.12	1.14	1.30	1.26
800	0.49	0.84	1.02	0.86
900	0.74	0.50	1.00	0.64
1000	0.77	0.67	1.12	0.67
1500	0.65	0.71	1.08	0.54
2000	0.81	0.75	0.97	0.53
2500	0.98	0.77	0.64	0.64
3000	0.43	0.84	0.83	0.92
3500	0.92	0.76	0.60	0.73
4000	0.50	0.67	0.70	0.90
5000	0.44	0.63	0.83	1.06
6000	0.71	0.37	0.64	0.76
7000	0.55	0.72	0.77	0.62
8000	0.53	0.88	0.84	0.82

Table (10.2)

Reverberation Time-Frequency Measurements
Part III Lecture Room (EMPTY)
Night Measurement (8.00 p.m. - 11.00 p.m.)

FREQUENCY IN Hz.	REVERBERATION TIME (IN SECONDS). Position A	Position B	Position C	Position D
100	1.64	1.56	0.97	1.11
150	1.37	1.10	0.76	0.93
200	0.76	0.95	0.94	0.98
250	1.39	1.07	0.94	1.32
300	0.83	0.86	0.92	0.60
350	0.79	1.23	1.03	1.10
400	0.79	0.53	1.14	1.11
500	1.43	1.56	1.01	1.20
600	1.52	1.45	0.70	1.17
700	1.24	0.86	0.64	0.70
800	0.67	0.98	0.58	0.70
900	0.89	0.55	0.37	1.25
1000	0.46	0.51	0.80	0.79
1500	0.48	1.00	0.56	0.64
2000	0.58	0.54	0.67	0.87
2500	0.82	0.31	0.54	0.36
3000	0.76	0.36	0.93	0.49
3500	0.36	0.78	0.96	0.84
4000	0.80	0.34	0.75	0.73
5000	1.00	0.31	0.74	0.79
6000	0.76	0.32	0.41	0.44
7000	0.56	0.47	0.44	0.55
8000	0.59	0.49	0.56	0.78

Table (10·3)

Reverberation Time - Frequency Measurements
Part III Lecture Room (EMPTY)
Afternoon Measurement (12.30 p.m. - 3.30 p.m.)

FREQUENCY in Hz	REVERBERATION TIME (IN SECONDS) Position A	Position B	Position C	Position D
100	2.14	1.83	2.10	2.10
150	1.82	1.87	1.50	1.14
200	1.51	1.53	1.21	1.53
250	1.38	1.40	0.58	1.11
300	1.50	1.43	1.32	1.31
350	1.35	1.64	1.34	1.32
400	1.16	1.08	1.18	0.91
500	1.28	1.45	1.38	1.10
600	0.54	0.85	0.41	0.65
700	1.42	0.65	0.54	0.66
800	1.09	1.03	0.75	1.16
900	1.24	0.46	0.85	0.54
1000	1.20	0.62	1.03	0.85
1500	1.21	1.26	0.52	1.40
2000	1.06	0.74	0.32	1.09
2500	0.65	0.93	0.63	0.68
3000	0.84	0.65	1.35	0.33
3500	0.82	0.93	0.39	0.98
4000	0.96	0.63	1.00	0.88
5000	0.76	1.24	0.80	0.37
6000	0.79	1.00	0.87	0.97
7000	0.42	0.56	0.85	0.41
800	0.55	0.68	1.02	0.40

Table (10:4)

Reverberation Time - Frequency Measurments
Part III Lecture Room (With tables and chairs)
Afternoon Measurement (12.30 p.m. - 3.30 p.m)

FREQUENCY IN Hz	REVERBERATION TIME (IN SECONDS) Position A	Position B	Position C	Position D
100	1.62	1.67	1.30	1.36
150	1.35	1.78	1.06	1.44
200	1.19	0.87	1.02	1.31
250	0.65	1.28	0.62	1.11
300	0.99	0.87	0.43	1.23
350	1.40	0.93	1.30	1.31
400	0.98	1.22	0.92	1.22
500	0.45	0.49	1.10	1.13
600	0.54	0.99	0.88	0.46
700	1.21	0.46	0.93	0.87
800	1.11	0.66	1.13	1.14
900	1.05	0.50	0.49	0.66
1000	0.89	0.54	0.38	0.87
1500	0.74	0.66	0.42	0.85
2000	0.44	0.74	0.86	0.95
2500	0.66	1.04	0.58	0.69
3000	0.97	0.82	0.40	0.78
3500	0.72	0.39	0.80	0.58
4000	0.88	0.58	0.84	0.72
5000	0.66	0.71	0.80	0.43
6000	0.46	0.85	0.66	0.33
7000	0.46	0.56	0.46	0.57
8000	0.67	0.96	0.61	0.72

Table (10,5)

Reverberation Time - Frequency Measurements

Part II Lecture Room (EMPTY)

Night Measurement (8.00 p.m. - 11.00 p.m.)

FREQUENCY IN Hz	REVERBERATION TIME (IN SECONDS) Position A	Position B	Position C	Position D
100	2.10	1.20	2.26	1.78
150	1.09	1.37	2.04	1.85
200	0.72	1.46	1.76	1.36
250	0.96	1.66	0.88	0.75
300	0.79	0.92	0.85	1.31
350	0.65	0.65	0.96	1.24
400	0.70	0.97	0.92	1.12
500	0.84	0.67	0.89	0.67
600	1.65	0.61	0.68	1.32
700	1.01	0.75	1.13	0.67
800	0.73	0.62	0.75	0.75
900	1.38	0.47	0.99	1.04
1000	0 44	0.79	0.50	0.54
1500	1.38	0.69	0.70	0.75
2000	0 44	0.64	0.55	0.63
2500	0.38	0.48	0.83	0.46
3000	0.72	0.82	0.48	0.62
3500	0.52	0.47	0.39	0.74
4000	0.56	0.70	0.37	0.63
5000	0.67	0.48	0.35	0.62
6000	0.68	0.38	0.50	0.65
7000	0.65	0.39	0.47	0.49
8000	0.63	0.40	0.55	0.72

Table (10,6)

Reverberation Time - Frequency Measurments

Part II Lecture Room (With tables and chairs)

Night Measurment (8.00 p.m. - 11.00 p.m.)

FREQUENCY IN Hz	REVERBERATION TIME (IN SECONDS)			
	Position A	Position B	Position C	Position D
100	1.90	2.30	2.06	1.55
150	1.48	1.40	1.47	1.35
200	1.25	0.92	1.20	1.18
250	1.14	0.72	1.07	1.14
300	1.00	1.25	1.10	1.00
350	0.75	0.68	1.13	0.64
400	0.79	0.92	1.02	1.02
500	0.97	0.70	0.94	1.06
600	0.61	0.63	0.83	0.69
700	0.44	1.11	0.88	0.75
800	0.94	0.62	0.92	0.71
900	0.70	0.42	0.84	0.89
1000	0.79	0.56	0.85	0.52
1500	0.85	0.44	0.69	0.59
2000	0.71	0.52	0.93	0.76
2500	0.76	0.78	0.71	0.51
3000	0.63	0.39	0.73	0.43
3500	0.79	0.83	0.47	0.85
4000	1.05	0.53	0.56	0.65
5000	0.60	0.70	0.72	0.73
6000	0.96	0.68	0.83	0.47
7000	0.76	0.57	0.51	0.48
8000	0.98	0.69	0.60	0.68

Table (10.7)

Reverberation Time - Frequency Measurements

Part II Lecture Room (EMPTY)

Afternoon Measurement (12 30 p.m h. - 3.30 p.m.)

FREQUENCY IN Hz	REVERBERATION TIME (IN SECONDS) Position A	Position B	Position C	Position D
100	2.11	2.15	2.15	2.40
150	2.04	1.48	1.48	1.99
200	1.54	1.85	1.80	1.73
250	1.38	1.80	1.70	1.43
300	0.77	1.73	1.58	0.89
350	1.19	0.67	1.27	1.00
400	0.94	0.95	0.88	1.02
500	1.22	0.80	1.15	0.61
600	1.42	0.80	0.85	0.79
700	0.98	0.77	1.15	0.91
800	1.29	0.93	1.20	0.76
900	0.95	1.26	0.42	0.68
1000	0.92	1.07	1.05	1.13
1500	1.25	0.70	1.25	1.03
2000	1.33	0.61	0.87	0.97
2500	1.04	0.58	1.30	0.45
3000	0.79	1.00	0.76	0.96
3500	1.27	0.92	1.05	1.04
4000	0.77	0.94	0.54	0.58
5000	1.07	0.99	0.96	0.70
6000	0.94	1.13	0.93	0.45
7000	0.66	0.73	1.10	1.00
8000	0.85	0.98	0.79	0.68

Table(10,8)

Reverberation Time - Frequency Measurements

Part II Lecture Room (WITH TABLES AND CHAIRS)

Afternoon Measurement. (12.30 p.m. - 3.30 p.m.)

FREQUENCY IN Hz	REVERBERATION TIME (IN SECONDS)			
	Position A	Position B	Position C	Position D
100	1.60	1.34	1.22	1.47
150	1.34	1.35	1.05	1.58
200	1.15	1.30	1.35	1.33
250	1.18	1.13	1.32	1.25
300	0.95	0.98	1.15	1.34
350	1.09	1.19	0.97	1.12
400	0.55	1.14	0.96	0.80
500	0.98	1.20	0.70	1.27
600	0.76	0.96	1.37	0.70
700	1.10	0.66	0.71	1.30
800	0.51	1.25	0.70	1.12
900	1.40	1.03	1.06	0.47
1000	0.63	0.37	0.94	0.90
1500	1.20	0.50	0.55	0.77
2000	0.93	0.96	0.45	0.77
2500	0.80	0.89	0.48	1.21
3000	0.36	0.52	0.61	0.79
3500	0.72	0.66	0.95	0.65
4000	0.55	1.07	1.02	0.60
5000	0.52	0.96	0.68	0.83
6000	0.49	0.93	0.54	0.69
7000	0.60	0.73	0.77	0.85
8000	0.67	0.52	0.75	0.70

Table 10,9

Reverberation Time - Frequency Measurements

Departmental Library (Elect. Eng.) (EMPTY) WITH BOOKS & CURTAINS

Night Measurement.

FREQUENCY IN Hz	REVERBERATION TIME (IN SECONDS)			
	Position A	Position B	Position C	Position D
100	0.70	0.74	0.53	0.61
150	0.88	0.54	0.84	0.64
200	0.49	0.75	0.62	0.87
250	0.60	0.92	0.60	0.53
300	0.59	0.48	0.50	0.73
350	0.55	0.44	0.71	0.76
400	0.75	0.55	0.65	0.98
500	0.99	0.50	0.44	0.51
600	0.70	0.80	0.94	0.95
700	0.42	0.48	0.46	0.61
800	0.63	0.65	0.43	0.62
900	0.72	0.57	0.86	0.89
1000	0.78	0.61	0.65	0.52
1500	0.58	0.89	0.60	0.80
2000	0.33	0.77	0.39	0.71
2500	0.53	0.70	0.34	0.98
3000	0.70	0.87	0.56	0.65
3500	0.54	0.55	0.42	0.61
4000	0.39	0.45	0.53	0.33
5000	0.84	0.57	0.61	0.62
6000	0.71	0.63	0.69	0.71
7000	0.59	0.61	0.65	0.67
8000	0.45	0.69	0.66	0.65

Table 10,10)

Reverberation Time - Frequency Measurements)

Departmental Library (Elect.Engg.) (WITH TABLES & CHAIRS BOOKS & CURTAINS)

Night Measurement (8.00.p.m. - 10.00. p.m.)

FREQUENCY IN Hz	REVERBERATION TIME (IN SECONDS) Position A	Position B	Position C	Position D
100	0.56	0.60	0.55	0.62
150	0.40	0.65	0.56	0.70
200	0.57	0.64	0.72	0.63
250	0.56	0.82	0.64	0.74
300	0.49	0.57	0.54	0.57
350	0.53	0.41	0.67	0.51
400	0.62	0.40	0.60	0.64
500	0.46	0.51	0.42	0.48
600	0.47	0.70	0.62	0.40
700	0.74	0.90	0.74	0.59
800	0.47	0.70	0.50	0.34
900	0.44	0.41	0.62	0.56
1000	0.38	0.41	0.69	0.52
1500	0.42	0.67	0.58	0.58
2000	0.52	0.75	0.57	0.34
2500	0.38	0.46	0.57	0.35
3000	0.64	0.38	0.50	0.56
3500	0.67	0.52	0.63	0.65
4000	0.44	0.77	0.71	0.36
5000	0.51	0.51	0.59	0.50
6000	0.64	0.72	0.60	0.58
7000	0.61	0.65	0.55	0.64
8000	0.60	0.58	0.57	0.61

Table (10,11)

Reverberation Time - Frequency Measurements
A.B.U. Assembly Hall (EMPTY)
Afternoon Measurement. (4.30 p.m. - 6.30 p.m.

FREQUENCY IN Hz	REVERBERATION TIME (IN SECONDS) Position A	Position B	Position C
100	1.98	1.94	1.90
150	1.95	2.44	1.41
200	1.76	1.73	1.47
250	1.40	1.36	2.34
300	1.50	1.70	1.21
350	1.70	1.20	1.20
400	2.04	1.39	1.75
500	1.65	1.70	2.06
600	1.42	1.90	1.85
700	1.06	1.60	1.45
800	1.54	1.95	1.25
900	1.90	1.15	1.41
1000	1.55	1.65	1.60
1500	1.80	0.95	1.35
2000	1.33	1.69	1.38
2500	1.00	1.09	1.45
3000	1.41	1.08	1.03
3500	1.08	1.31	1.06
4000	1.20	1.56	1.93
5000	1.00	1.11	1.12
6000	1.20	1.27	1.17
7000	1.50	1.43	1.45
8000	1.24	1.38	1.23

Table (10,12)

Reverberation Time - Frequency Measurments
A.B.U Assembly Hall (WITH CHAIRS & CURTAINS)
Afternoon measurement. 2.30 p.m. - 4.30 p.m.

FREQUENCY IN Hz	REVERBERATION TIME (IN SECONDS) Position A	Position B	Position C
100	1.23	1.32	1.95
150	1.16	1.23	1.18
200	1.34	1.10	1.22
250	0.71	0.96	1.32
300	1.34	1.04	1.60
350	1.07	1.10	1.35
400	1.11	1.33	1.11
500	0.69	1.12	2.16
600	1.21	0.82	1.27
700	2.42	1.23	1.16
800	1.22	0.52	1.70
900	0.95	0.82	1.26
1000	0.82	1.25	1.08
1500	0.94	1.65	1.32
2000	1.10	1.05	0.95
2500	1.16	1.00	1.06
3000	1.50	1.15	1.28
3500	1.26	1.04	1.08
4000	0.99	1.79	0.78
5000	0.75	0.85	1.10
6000	1.03	0.97	1.08
7000	0.90	1.10	1.00
8000	1.05	1.00	1.12

Table (10,13)

Reverberation Time - Frequency Measurements

Faculty of Engineering Lecture Theatre

Night Measurement (8.00 p.m. - 10.00 p.m.)

Microphone Positioning:- Front, (Centre) on the Lecturer's table

FREQUENCY IN Hz	REVERBERATION TIME (IN SECONDS)
100	0.77
150	0.74
200	0.64
250	0.68
300	0.50
350	0.56
400	0.70
500	0.54
600	0.45
700	0.72
800	0.60
900	0.62
1000	0.67
1500	1.14
2000	0.66
2500	0.56
3000	0.65
3500	0.74
4000	0.56
5000	0.47
6000	0.77
7000	0.53
8000	0.50

Table(10,14)

Reverberation Time - Frequency Measurements

High Voltage Laboratory (Elect. Eng. Dept.)

Night Measurement (8.00 p.m. - 10.00 p.m.)

Microphone Positioning:- Centre of the Laboratory.

FREQUENCY IN Hz	REVERBERATION TIME (IN SECONDS)
100	3.40
150	2.40
200	3.00
250	2.90
300	2.80
350	2.80
400	3.72
500	2.50
600	1.30
700	2.80
800	2.80
900	3.75
1000	2.60
1500	3.70
2000	2.50
2500	3.10
3000	2.70
3500	2.90
4000	1.80
5000	0.93
6000	2.40
7000	1.40
8000	

Table 11.1.

Variation of R.T. with Supply Voltage

Frequency	R.T.for -10VSupply	R.T.for -11VSupply	R.T. for -12VSupply	R.T. for -13VSupply	R.T. for -14VSupply
150Hz	0.53sec	0.72sec	0.69sec.	0.61sec	0.60sec.
200,,	0.42 ,,	0.70 ,,	0.63 ,,	0.73 ,,	0.41 ,,
300,,	0.35 ,,	0.66 ,,	0.72 ,,	0.65 ,,	0.36 ,,
400,,	0.57 ,,	0.65 ,,	0.64 ,,	0.69 ,,	0.34 ,,
500,,	0.36 ,,	0.69 ,,	0.68 ,,	0.76 ,,	0.61 ,,
600,,	0.48 ,,	0.59 ,,	0.71 ,,	0.72 ,,	0.53 ,,
700,,	0.32 ,,	0.62 ,,	0.65 ,,	0.67 ,,	0.36 ,,
800,,	0.35 ,,	0.71 ,,	0.74 ,,	0.78 ,,	0.32 ,,
900,,	0.51 ,,	0.68 ,,	0.66 ,,	0.75 ,,	0.37 ,,
1000,,	0.40 ,,	0.76 ,,	0.70 ,,	0.68 ,,	0.40 ,,
2000,,	0.33 ,,	0.67 ,,	0.76 ,,	0.80 ,,	0.38 ,,
3000,,	0.29 ,,	0.70 ,,	0.69 ,,	0.74 ,,	0.39 ,,
4000,,	0.37 ,,	0.78 ,,	0.77 ,,	0.67 ,,	0.45 ,,
5000,,	0.34 ,,	0.72 ,,	0.73 ,,	0.68 ,,	0.33 ,,
6000,,	0.42 ,,	0.65 ,,	0.65 ,,	0.70 ,,	0.39 ,,
7000,,	0.32 ,,	0.69 ,,	0.72 ,,	0.73 ,,	0.36 ,,
8000,,	0.41 ,,	0.72 ,,	0.75 ,,	0.69 ,,	0.40 ,,
9000,,	0.38 ,,	0.67 ,,	0.68 ,,	0.74 ,,	0.36 ,,
10000,,	0.32 ,,	0.64 ,,	0.69 ,,	0.76 ,,	0.35 ,,

Table 11.2.

Variation of R.T. with Time

Time in (Minutes)	R.T.(Secs.) at 1000Hz	R.T. (Secs.) at 200Hz.
0	0.59	0.62
2	0.70	0.73
4	0.76	0.78
6	0.68	0.78
8	0.74	0.72
10	0.66	0.73
12	0.78	0.76
14	0.68	0.80
16	0.73	0.78
18	0.71	0.73
20	0.69	0.77
22	0.75	0.74
24	0.70	0.74
26	0.67	0.69
28	0.73	0.70
30	0.72	0.75

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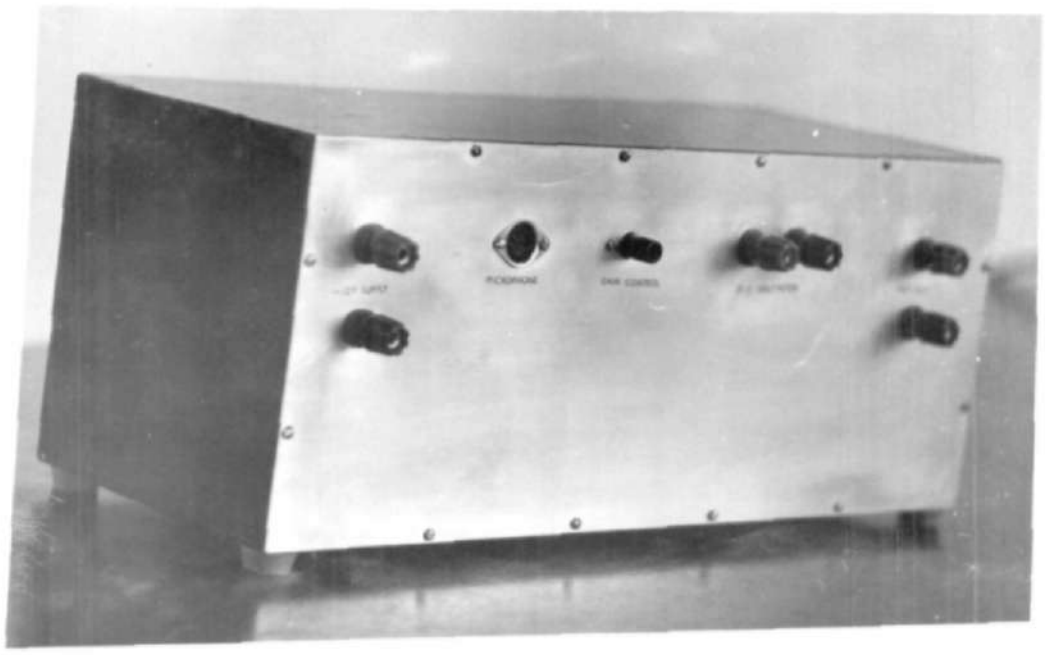
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ACKNOWLEDGEMENTS.

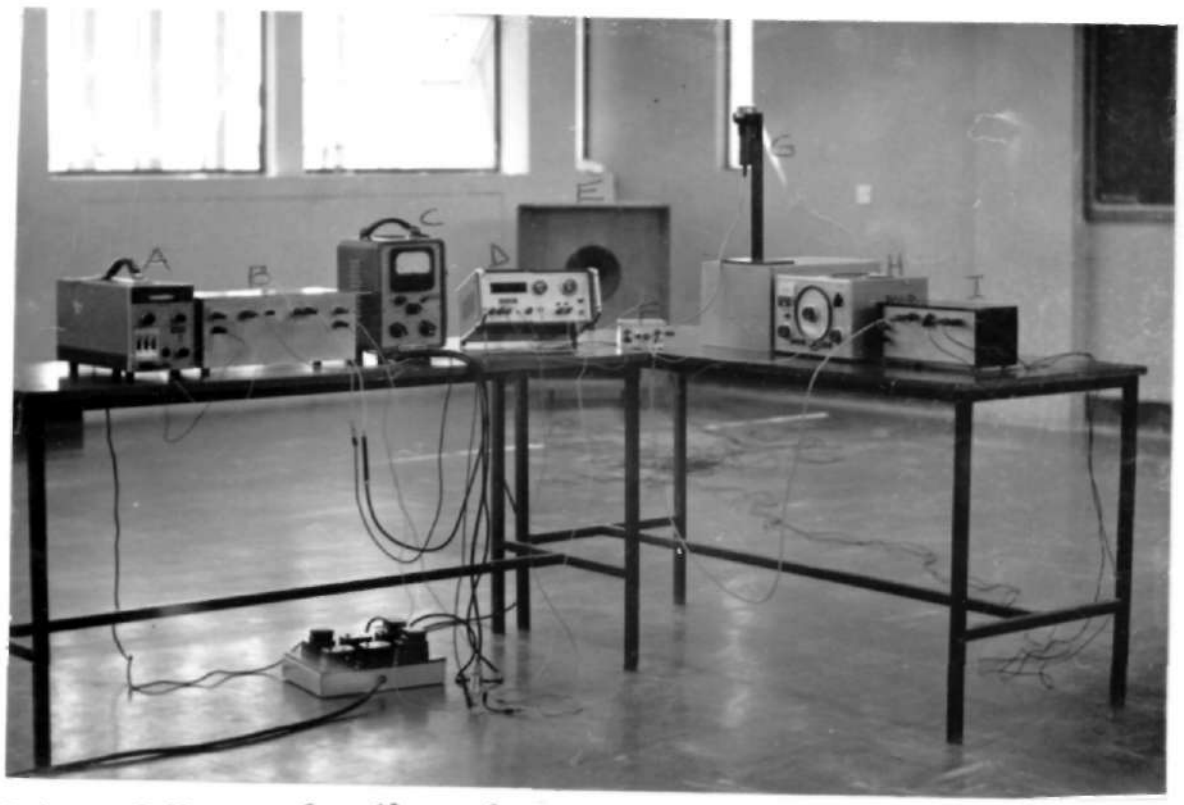
The author is highly indebted to Mr. J. Buczkowski, M.Sc., Senior Lecturer, Department of Electrical Engineering, Ahmadu Bello University, Zaria for supervising the project and for making useful suggestions when necessary.

He also wishes to express his profound gratitude to Professor K.R. Sturley, B.Sc., Ph.D., C.Eng., F.I.E.E.E., F.I.E.E., Professor and Head of the Department of Electrical Engineering Ahmadu Bello University, Zaria for his kindness and the keen interest shown in the work, and also for providing adequate facilities for the project.

Finally, he would like to thank the members of the technical staff of the department for their cooperation.



Part of the reverberation meter consisting of a microphone pre-amplifier, a variable gain amplifier, a logarithmic amplifier a detector and differentiating & clipping circuits.



Set-up of the reverberation meter.

A = Power supply unit.

B = Box containing linear amplifier, a logarithmic amplifier, detector and differentiating & clipping circuits.

C = Vacumm tube voltmeter.

D = Advance timer counter T.C. 7

E = Loudspeaker

F = Two way switch with one make & one break contacts

G = Microphone

H = Signal generator

I = Loudspeaker power amplifier.