DEVELOPMENT OF AUDIO SYSTEM TO STUDY THE COEFFICIENT OF ATTENUATION OF SOUND WAVES IN AIR

BY

OSINOWO M. OLATUNDE
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Abstract

Audible sound waves travel farther in dry air than in moist air; as attenuation occurs due to the presence of atmospheric water vapour. To study the rate of attenuation, an audio equipment was developed which is made up of: a variable audio oscillator with less than 0.2% distortion in the frequency range 20Hz to 50kHz, a 50Watt audio power amplifier with total harmonic distortion (THD) less than 0.1% to drive the transmitting device and a linear sound pressure meter as the receiver. The sound pressure meter was found to be accurate to within ±3dB in the range 50dB to 120dB.The audio equipment was set up for some months and from the data collected, the attenuation coefficient of sonic waves traveling in the atmosphere was found to be between –0.0063dB and –0.0311dB per meter.

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CERTIFICATION

I certify that this research work and the results obtained were carried out by **Osinowo M. O.** of the department of physics, Federal University of Technology Akure and has been approved as meeting the requirements for the award of M.Tech. degree in Physics of the Federal University of Technology Akure, Ondo State, Nigeria.

Dr M.T. Babalola
B.Sc, PhD (Ibadan) – Supervisor

Dedication

To

Maume, Mauci, and Maunu.

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CHAPTER ONE

1.1 Introduction

Sound is a physical phenomenon, which may be defined as a time varying disturbance of the density of a fluid (or solid) medium. It is usually associated with very small vibrational movements of particles of the medium. The frequency range of interest is the audio frequency range, which extends from about 20Hz to 20kHz.

Audio- frequency vibrations can also occur in solid materials such as steel or wood. These are always accompanied by sound in any fluid with which the solids are in contact. Sound is a longitudinal wave motion in material media – solids, liquids and gases. Sound in a fluid [or air] depends for its existence or transmission upon two properties of the medium (Subrahmanyam and Lal, 1974):

1 the generation of pressure in response to a change in the volume available to a fixed mass of fluid, i.e. change of density;

2 the possession of inertia, that is, that property of matter which resists attempts to change its momentum.

Both the forces generated by volumetric strain of fluid elements, and the acceleration of those elements, are related to their displacements from positions of equilibrium. The resulting interplay produces the phenomenon of wave motion, whereby disturbances are propagated throughout the fluid, often to very large distances.

1.2 SOUND AND SOUND FIELDS.

The physical phenomenon known as "sound" in a fluid essentially involves time varying disturbances of the density of the medium from its equilibrium value. These changes of density are in most cases extremely small compared with the equilibrium density, typically of the order of 10^{-7} to 10^{-5} (Fahy, 1989). They may be attributed to changes in the volume of space occupied by a given mass of fluid, changes of shape not being of consequence in the case of sound waves. It is the volumetric strain, or dilatation, undergone by an elemental mass of fluid which matters. The static elastic nature of air in its response to volumetric strain is easily demonstrated by closing the outlet hole in a bicycle pump with a finger and depressing, and then releasing. The plunger returns almost to its original position on release. It is not so easy to find everyday phenomenon to demonstrate that liquids, such as water, are also elastic; this is because, in response to a given volumetric strain, they generate much larger internal stresses, and hence, reaction forces.

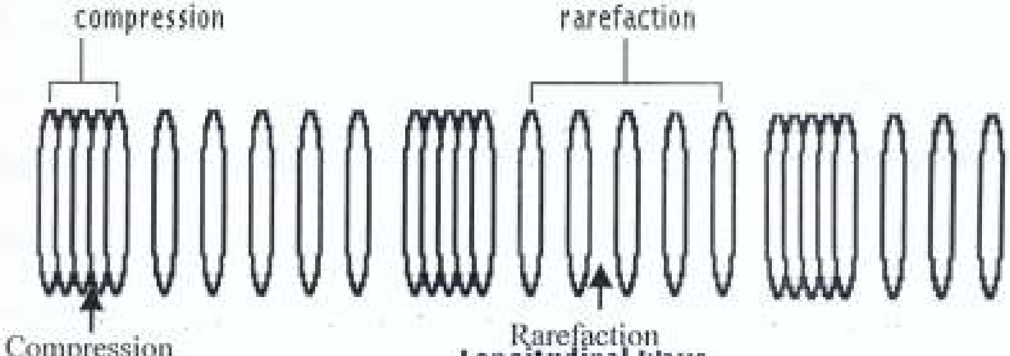
Sound waves in air are produced by contact with a vibrating surface. The resulting displacement of air particles causes the pressure to increase and decrease alternately above and below the steady atmospheric value in sympathy with the vibration. Due to the elasticity and inertia of the medium, these pressure variation form waves which travel outward from the point of origin in ever increasing circles. The wave motion is longitudinal, the air particles moving backwards and forwards away from and towards the source of sound i.e. along the direction of propagation of the sound wave. The waves of alternate compression and rarefaction of the air can be illustrated by a graph shown in figure1.1 (Microsoft® Encarta® Reference Library, 2002) which is a simple sine curve representing a pure and sustained note. In this diagram, the direction of propagation is from left to right, forward displacements of air particles being represented by upward ordinates and backward displacements by downward ordinates. During the time from O to A the air particles are displaced forward and, from A to B backwards. Point A represents a compression, maximum pressure coinciding with maximum change in displacement. Point C where displacement is in the opposite direction represents a state of rarefaction. The dashed curve shows that the air pressure varies above and below the steady state atmospheric value (represented as O) in proportion to the displacement caused by the vibrating surface that originates the sound. The change in air pressure is very small relative to the normal atmospheric pressure but the ear or the microphone acts as a barometer and converts these minute changes into the sensation of hearing or electrical signals.

1.3 PHYSICAL CHARACTERISTICS

1.3.1 Characteristics of Sound Waves

Each musical instrument we play produces a characteristic vibration. The vibrations travel through the air in sound waves that reach our ears, allowing us to identify the instrument being played even when we cannot see it. The four sound waves shown in figure 1.2 here demonstrate signature waveforms of some common instruments. A tuning fork makes a pure sound, its vibration is nearly sinusoidal. A violin generates a bright sound and a jagged waveform. The flute produces a mellow, true sound and a relatively curved waveform. The tuning fork, violin, and flute were all playing the same note, so the distance between the peaks (the high points of the wave) is the same for each waveform. A gong does not vibrate in a regular pattern, as do the first three instruments. Its waveform is jagged and random, and its pitch is generally unrecognizable(Microsoft®Encarta® Reference Library, 2002).

Any simple sound, such as a musical note, may be completely described by specifying three perceptual characteristics: pitch, loudness (or intensity), and quality (or timbre). These characteristics correspond exactly to three physical quantities: frequency, amplitude, and harmonic constitution, or waveform, respectively. Noise is a complex sound, a mixture of many different frequencies or notes not harmonically related (Microsoft®Encarta® Reference Library, 2002).



Longitudinal Wave

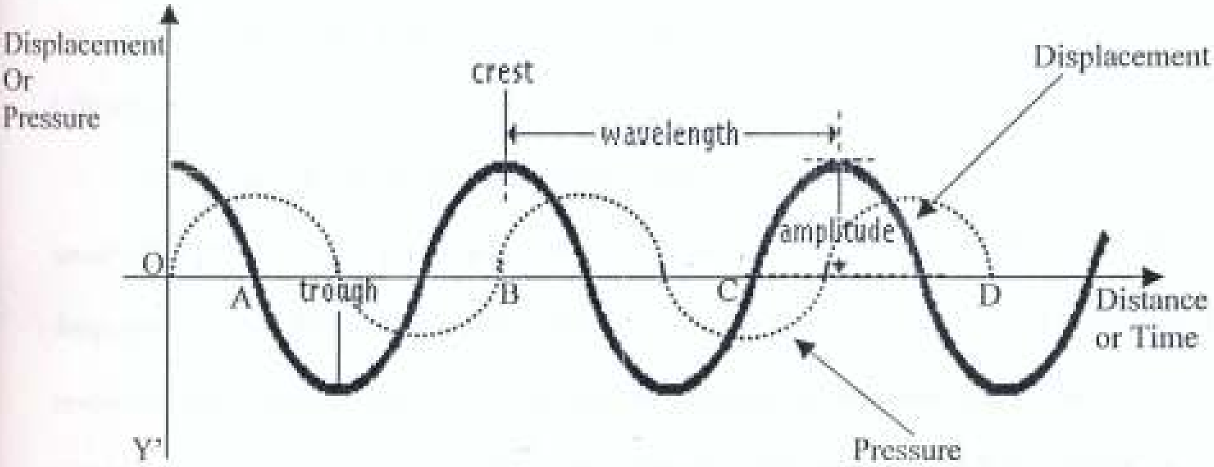


Figure1.1 Representation of Sound Wave

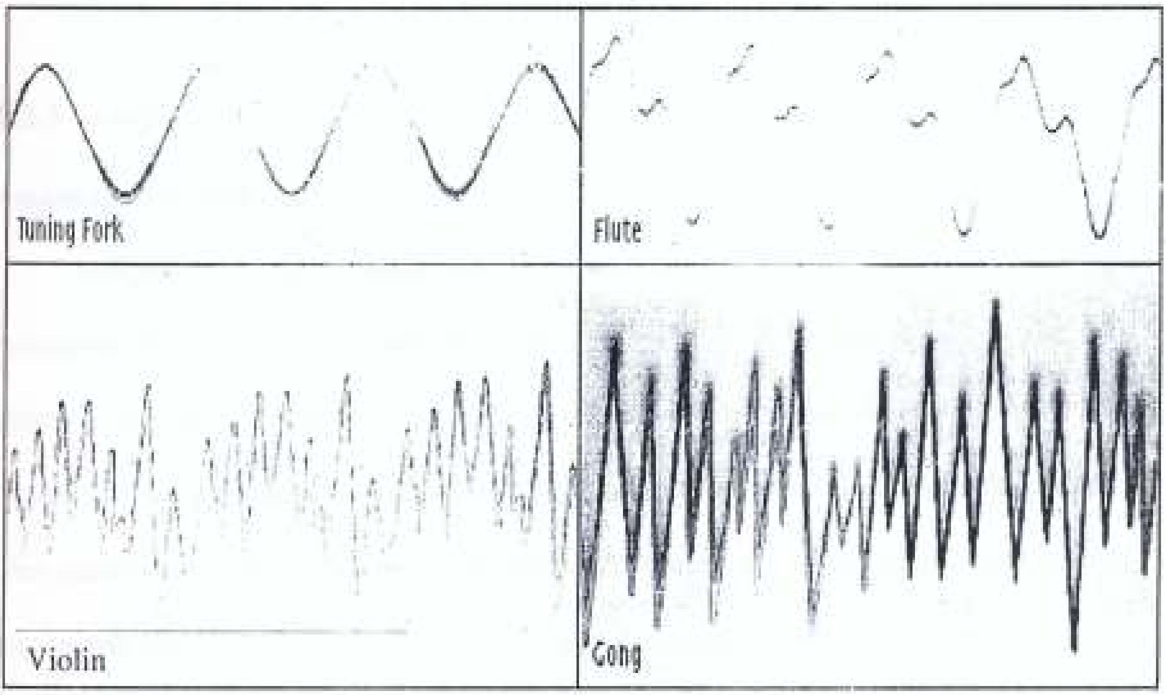


FIGURE 1.2: Characteristic Sound Waves

1.3.2 Frequency

Frequency is perceived as "higher" or "lower" sounds. The frequency of a sound is the number of complete oscillations a sound wave completes per second. Frequency is measured in hertz (Hz), or cycles per second. Because the amplitude (maximum height) of the waves above remains constant, we are able to hear the same note at different frequencies, from 110.00 Hz to 880.00 Hz. Waves propagate at both higher and lower frequencies, but humans are unable to hear them outside a relatively narrow range (Microsoft®Encarta® Reference Library, 2002).

Sounds can be produced at a desired frequency by different methods. For example, a sound of 440 Hz can be created by actuating a loudspeaker with an oscillator tuned to this frequency. An air blast can be interrupted by a toothed wheel with 44 teeth, rotating at 10 revolutions per second; this method is used in operating an ordinary siren. The sound of the speaker and that of the siren at the same frequency are very different in quality, but will correspond closely in pitch, equivalent to the A above middle C on a piano.

1.3.3 Amplitude

Amplitude and Volume

Amplitude is the characteristic of sound waves that we perceive as volume. The maximum distance a wave travels from the zero position is the amplitude (figure 1.3). This distance corresponds to the degree of motion in the air molecules of a wave. As the degree of motion in the molecules is increased, they strike the eardrum with progressively greater force. This causes the ear to perceive a louder sound. A comparison of samples at low, medium, and high amplitudes demonstrates the change in sound caused by altering amplitude.

These three waves have the same frequency, and so should sound the same except for a perceptible volume difference.

The amplitude of a sound wave is the degree of motion of air molecules within the wave, which corresponds to the extent of rarefaction and compression that accompanies the wave. The greater the amplitude of the wave, the harder the molecules strike the eardrum and the louder the sound that is perceived. The amplitude of a sound wave can be expressed in terms of absolute units by measuring the actual distance of displacement of the air molecules, or the pressure differential in the compression and rarefaction, or the energy involved. Ordinary speech, for example, produces sound energy at the rate of about one hundred-thousandth of a watt. (Microsoft®Encarta® Reference Library, 2002). All of these measurements are extremely difficult to make, however, and the intensity of sounds is generally expressed by comparing them to a standard sound, measured in decibels dB.

1.4 Sound Intensities

Sound intensities are measured in decibels (dB). For example, the intensity at the threshold of hearing is 0 dB, the intensity of whispering is typically about 10 dB, and the intensity of rustling leaves reaches almost 20 dB. Sound intensities are arranged on a logarithmic scale, which means that an increase of 10 dB corresponds to an increase in intensity by a factor of 10. Thus, rustling leaves are about 10 times louder than whispering. Graphs of sound pressures of some common sources of sound are shown in figure 1.4 (Microsoft®Encarta® Reference Library, 2002).

The distance at which a sound can be heard depends on its intensity, which is the average rate of flow of energy per unit area perpendicular to the direction of propagation.

In the case of spherical waves spreading from a point source, the intensity varies inversely as the square of the distance, provided that no loss of energy is due to viscosity, heat conduction, or other absorption effects. Thus, in a perfectly homogeneous medium, a sound will be nine times as intense at a distance of 1 unit from its origin as at a distance of 3 units; that is, intensity varies inversely as the square of the distance (Condon and Odishaw, 1967). In the actual propagation of sound through the atmosphere, changes in the physical properties of the air, such as temperature, pressure, and humidity, produce attenuation and scattering of the directed sound waves, so that the inverse-square law generally is not applicable in direct measurements of the intensity of sound.

1.5 Velocity of Sound

Sound waves travel more swiftly and efficiently in water than in air, allowing animals such as whales to communicate with one another over great distances. (Microsoft®Encarta® Reference Library, 2002). Whales and tortoises also use sound waves to help them navigate in dark water, directing and receiving sound waves in much the same way as the sonar on a ship or submarine. The distance between two successive crests of the wave is called the wavelength. The product of the wavelength and the frequency must equal the speed of propagation of the wave, and is the same for sounds of all frequencies (if the sound is propagated through the same medium at the same temperature). The speed of propagation of sound in dry air is directly proportional to the absolute temperature. At a temperature of 0° C the speed is 331.6 m/s and at 20° C, it is 344 m/s. Changes in pressure at constant density have virtually no effect on the speed of sound. In general, the velocity of sound in gases depends only on their density. If the molecules are heavy, the speed decreases. Thus, sound travels slightly faster in moist air than in dry air, because moist air contains a greater number of lighter molecules. The velocity of sound in most gases depends also on one other factor, such as the specific heat, which affects the propagation of sound waves (Khanna and Bedi, 1989).

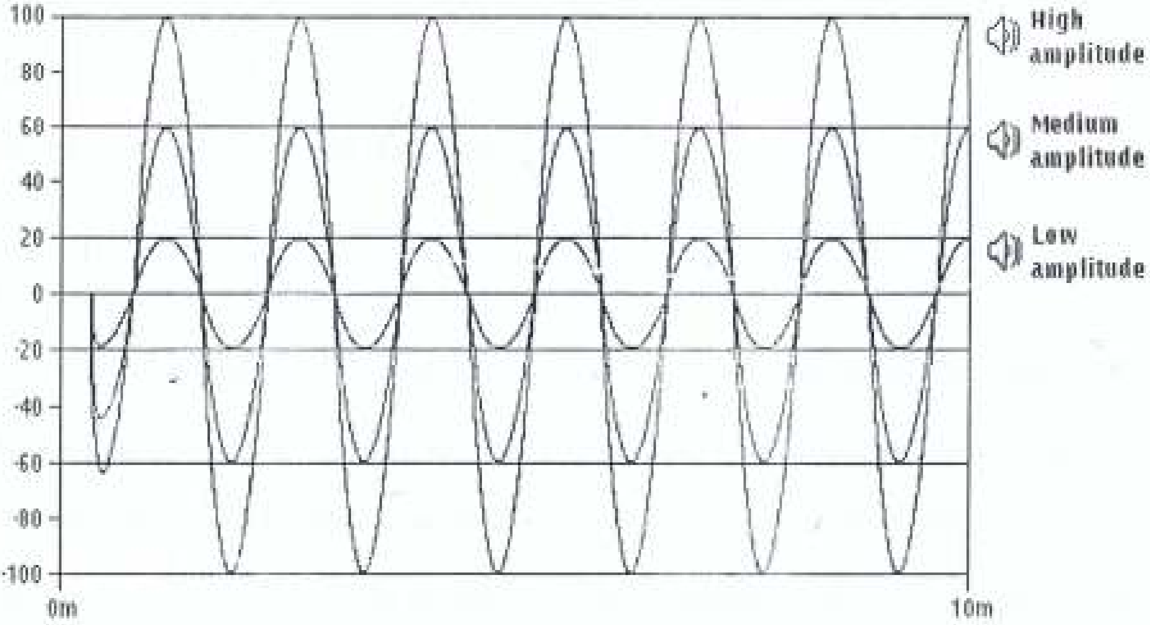


FIGURE 1.3: Amplitude of Waves.

Sound generally moves much faster in liquids and solids than in gases. In both liquids and solids, density has the same effect as in gases. That is, velocity varies inversely as the square root of the density. The velocity also varies directly as the square root of the elasticity. The speed of sound in water, for example, is about 1,525 m/s at ordinary temperatures but increases greatly with an increase in temperature. The speed of sound in copper is about 3,353 m/s at ordinary temperatures and decreases as the temperature is increased (owing to decreasing elasticity). In steel, which is more elastic than copper, sound moves at a speed of about 4,877 m/s. (Khanna and Bedi, 1989).

1.6 Refraction, Reflection, and Interference

Echo. An echo is a reflected sound wave. The perceptible gap between the emission and repeat of the sound represents the time it takes waves to travel to an obstacle and back. The echoed sound is often fainter because not all of the original wave is reflected. Generally, echoes such as those heard in the mountain ranges are caused by sound waves striking large surface 30 m or more away from their source (Microsoft®Encarta® Reference Library, 2002). An echo in a different medium, such as a steel pipe, may be created and observed by rapping the metal when the ear is against it

Sound moves forwards in a straight line when travelling through a medium having uniform density. Like light, however, sound is subject to refraction. In Polar regions, for example, where air close to the ground is colder than air that is above, a rising sound wave entering the warmer region, in which sound moves with greater speed, is bent downwards by refraction. (Microsoft®Encarta® Reference Library, 2002). The excellent reception of sound downwind and the poor reception upwind are also due to refraction. The velocity of the wind is generally greater at an altitude of many meters than near the ground; a rising sound wave

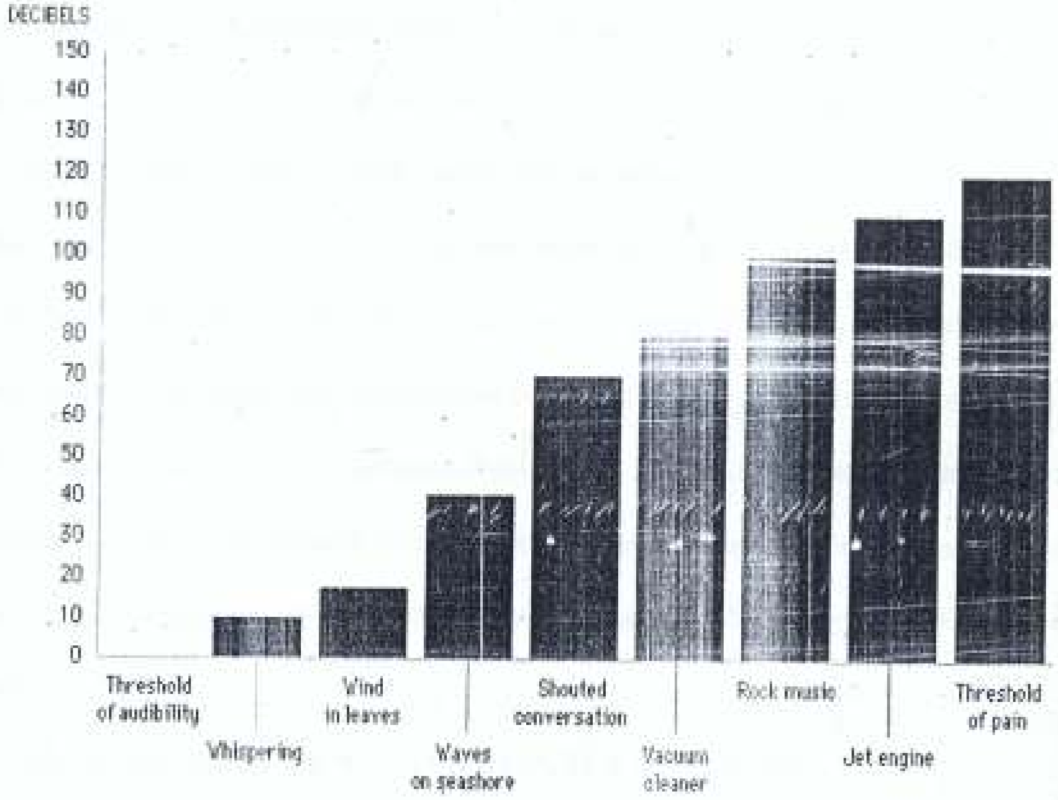


FIGURE 1.4: Intensities of Different Sources of Sound

moving downwind is bent back towards the ground, whereas a similar sound wave moving upwind is bent upwards over the head of the listener.

Sound is also governed by reflection, obeying the fundamental law that the angle of incidence equals the angle of reflection. An echo is the result of reflection of sound. Sonar depends on the reflection of sounds propagated in water. A megaphone is a funnel-like tube that forms a beam of sound waves by reflecting some of the diverging rays from the sides of the tube. A similar tube can gather sound waves if the large end is pointed at the source of the sound; an ear trumpet is such a device. (Microsoft®Encarta® Reference Library, 2002).

Sound is also subject to **diffraction** and **interference**. If sound from a single source reaches a listener by two different paths—one direct and the other reflected—the two sounds may reinforce one another; but if they are out of phase they may interfere, so that the resultant sound is actually less intense than the direct sound without reflection. Interference paths are different for sounds of different frequencies, so that interference produces distortion in complex sounds. Two sounds of different frequencies may combine to produce a third sound, the frequency of which is equal to the sum or difference of the original two frequencies.

1.7 FACTORS AFFECTING THE VELOCITY OF SOUND

Sound waves in fluids involve local changes [generally small] in the pressure, density and temperature of the media, together with motion of the fluid elements. Fluid elements in motion have speed, and therefore possess kinetic energy. In regions where the density increases above its equilibrium value, the pressure also increases; consequently, energy is stored in these regions, just as it is in a compressed spring. This form of energy is termed potential energy.

In simple harmonic motion, such motion is natural to fluid particles disturbed from equilibrium. Fluid particles will oscillate continuously only if waves are continuously generated by a source, or if, once generated, they repeatedly retraverse a fluid region via reflections from surrounding boundaries. It is not intuitively obvious that in either case energy will be transported from one location to another; it seems much more likely that it will just be transferred to-and-fro between adjacent fluid elements.

Therefore, considering a transient sound created in the open air, for example by a handclap, a thin shell of disturbance will spread out all around the source, travelling at the speed of the sound. Within this disturbed region the fluid particles will be temporarily displaced from their equilibrium positions, and the pressure, density and temperature will temporarily vary from their equilibrium values. Once the disturbance has passed, everything is just as it was before- the fluid particles are once more at rest in their original positions and do not continue to oscillate. It is quite clear from this qualitative description of wave propagation that the potential and kinetic energies created by the action of the source on the air immediately surrounding it are transported with the disturbance. They cannot disappear, except through the action of fluid friction (viscosity), and other dissipative processes, which are known to have rather small effect at audio frequencies. The actual wave motion depends therefore on the kind of initial impulse given from outside, the elasticity and the mass contents of the medium.

Newton first showed that in a homogeneous medium the velocity of a longitudinal wave is given by

$$v = \sqrt{\frac{E}{\rho}} \qquad 1$$

where E is the modulus of elasticity for the particular type of strain set up and ρ, the density of the medium.

1.8 VELOCITY OF PLANE WAVES IN AIR BY ADVERSE – WIND METHOD

Considering a wave traveling towards the right in the air contained in a long tube of unit cross section as shown in figure 1.5. If the medium is caused to move towards the left with a

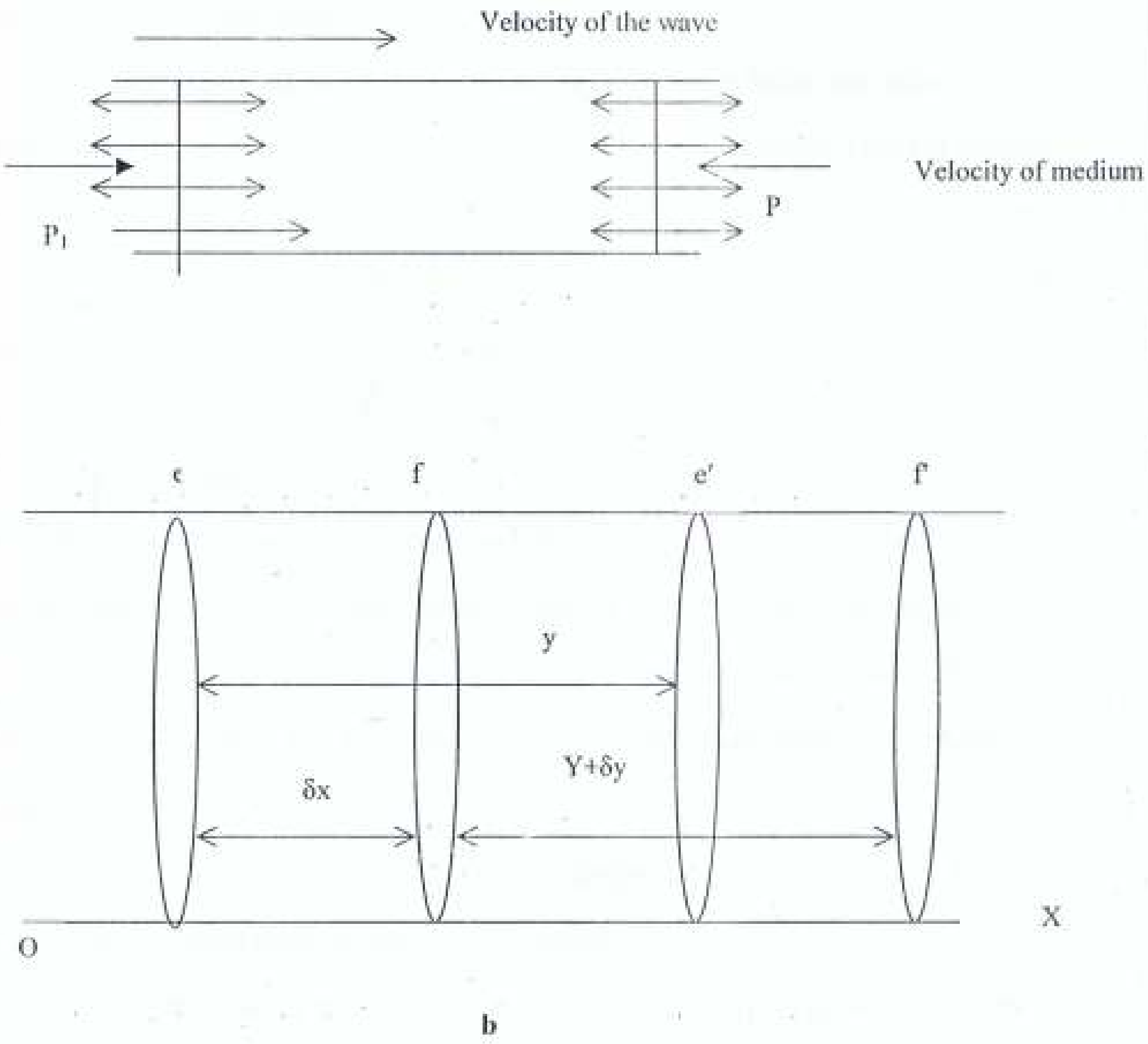


Figure 1.5 a) Deducing Velocity of a wave in air b) Deducing the velocity of a longitudinal wave

velocity equal to that of the wave, then the wave, though moving with respect to the air, will be at rest relatively to the ground.

In the small region surrounding any point, the pressure, velocity and density of the medium will be constant. These quantities will, of course, have different values at different points along the tube. Assuming two partitions A and B are placed across the tube, since there is no change in the average density of the medium and no accumulation of the medium between A and B, the mass of the air entering the space between the two partitions in a time t must be equal to the mass leaving it in the same time. Since the particle velocities at A and B are, in general, not identical, there may be a gain or loss of momentum accompanying the transfer of mass. For simplicity, suppose that A is a region of normal density and B a region of condensation, then since the same mass of medium must flow out of the partition B per second as it enters the partition A in the same time, the velocity of outflow at B will be less than the velocity of inflow at A as the same volume of a given mass of fluid depends upon its pressure.

Let v, P and ρ denote the velocity, pressure and density respectively at B and v_1, P_1 and ρ_1 the corresponding quantities at A relative to the ground.

(i) since the mass of air flowing in at A is equal to that flowing out at B, we have,

$$m = v\rho = v_1\rho_1 \qquad 2$$

(ii) in one second a mass of gas m equal to that flowing through either partition has its velocity decreased from v to v_1 and therefore it suffers a loss of momentum/sec= m (v-v_1) and this is also equal to the gain of momentum/sec of the second medium contained in the cylinder AB.

Applying Newton's second law of motion it is due to some external force. Since the only force available is the slight difference of pressure P_1-P, acting on the medium between the partitions A and B, we have therefore

$$P_1 - P = m\,(v - v_1)$$

$$P_1 - P = \frac{m}{v}\left(1 - \frac{v_1}{v}\right) \text{...} 3$$

From equation (1)

$$m = v\rho\,, and\ \ \frac{v_1}{v} = \frac{\rho}{\rho_1}$$

Substituting these values in equation (3) above, we have

$$P_1 - P = v^2\rho(1 - \frac{\rho}{\rho_1}) = v^2\rho\frac{\rho_1 - \rho}{\rho_1},$$

or

$$v^2\rho = \frac{P_1 - P}{\frac{\rho_1 - \rho}{\rho_1}}$$

Now the volume elasticity of the medium is given by

$$E = \frac{Elasticity\ \ pressure}{Volumetric\ \ strain} = \frac{P_1 - P}{(V - V_1)/V}$$

where V and V_1 are the specific volumes at A and B, then since

$$V\alpha\frac{1}{\rho}$$

$$\frac{\rho_1 - \rho}{\rho_1} = \frac{V - V_1}{V}$$

Therefore,

$$E = \frac{P_1 - P}{\frac{\rho_1 - \rho}{\rho_1}}$$

Setting this value in (iv) above

Also, Schultz (1956) made a major contribution to the development of practical sound intensity measurement system. He implemented the principle by which a particle velocity signal can be obtained by integrating the difference between the signals produced by two small pressure –sensitive transducers spaced a small distance apart in terms of the wavelength of sound at the highest frequency of interest. Unfortunately for the development of practical measurement systems, Schultz's disc-like transducers were placed back to back, with their surfaces a very small distance apart. This configuration placed extreme demands on the electronic circuitry of the day, and, although he demonstrated satisfactory performance under laboratory conditions in relatively simple sound fields, attempts to survey the sound fields generated by a sound source in a rigid-walled enclosure were disappointing. As Schultz et al. (1975) explained later, it was not only the inadequate performance of the measurement system under highly reactive conditions which produced the problems. The lack of a comprehensive theoretical analysis, and therefore of physical understanding, of the enclosed sound field, also contributed to the resulting lack of confidence in the measured results.

In the late1960's, Mechel (1968), Odin (1967) and Kurze (1968) demonstrated the relationships between the active and reactive components of sound intensity, and at the same time, the spatial gradients of phase and squared pressure. These relationships are today effectively implemented in the indirect spectral technique of intensity measurement.

More pioneering contributions in the area of sound intensity measurement to the determination of sound power radiated by complex sources, were made in the early 1970's by research workers in South Africa such as Van Zyl and Anderson (1975) and Burger, et al (1973) among others. Although they used a combination of pressure-and-velocity microphones in their early work, they later realized the superiority of the combination of two nominally identical pressure microphones, and developed the first analogue intensity meter to have a wide frequency range and large dynamic range. This instrument was developed from a prototype constructed by Van Zyl in 1979. Subsequently this group developed a range of intensity meters of increasingly high performance as a small commercial venture.

Current research and development in the field concerned, among other things, improved survey procedures for the determination of the sound power of one source operating in the presence of others (this is of particular importance for the development of good measurement standards); applications in duct acoustics and building acoustics; power flux line mapping; and imaging of acoustic sources by near field microphone array techniques, which generate intensity distributions as a by-product.

1.10 SCOPE OF THIS PRESENT WORK

Sound waves travel farther in dry air than in moist air. Thus much attenuation is experienced in dry air than in moist air. The scope of this recent research is to study this coefficient of attenuation by constructing audio equipment (both the transmitter and the linear sound pressure meter as receiver) using single probe method at the receiver end.

CHAPTER TWO

2.1 ATTENUATION OF SOUND WAVES

Sound propagating away from a source diminishes in strength at a rate depending on a variety of circumstances. It also encounters situations that can cause changes in amplitude and direction. At increasing distances from a source of sound, the intensity is expected to decrease. The rate at which it decreases is dictated by the directional properties of the source and the environment into which it radiates. A source of sound that is small compared with the wavelength of the sound being radiated, (a condition that includes many common situations), the sound spreads outward as a sphere of ever increasing radius. The sound energy from the source is distributed uniformly over the surface of the sphere, meaning that the intensity is the sound power output divided by the surface area at any radial distance from the source.

Since the surface area of a sphere is $4\pi r^2$, the relationship between the sound intensities at two different distances is:

$$\frac{I_1}{I_2} = \frac{r_2^2}{r_1^2}$$

where I_1 is the intensity at radius r_1, I_2 is the intensity at radius r_2 .

The level difference $= 10\log \frac{r_2^2}{r_1^2} = 20 \log \frac{r_2}{r_1}$ dB

Sound propagating outdoors through the atmosphere generally decreases in level with increasing distance between the source and the receiver. That is, attenuation occurs. This attenuation (units in dB) is the result of several mechanisms (Cyril, 1979) which include:

1. Attenuation (A_{div}), which is caused by geometrical divergence from the source and receiver.

2. Attenuation (A_{barr}), which results if a barrier, is placed between the source and receiver.
3. Attenuation (A_{atm}) due to the absorption of acoustic energy by the air in which the sound waves are propagated. As sound propagates through the atmosphere, its energy is gradually converted into heat (i.e. sound is absorbed) by a number of molecular processes in the air called atmospheric absorption.

The attenuation due to atmospheric absorption during propagation, (A_{atm}), through a distance d meters, is given by

$$(A_{atm}) = \frac{\alpha d}{100} \text{ dB}$$

Where α is the atmospheric attenuation coefficient in dB per 100m.

This attenuation coefficient strongly depends on frequency and relative humidity and less strongly on temperature.

4. Attenuation (A_{excess}) in excess of that from the above mechanisms, caused mainly by propagation over ground (usually called "ground effect").

2.2 EFFECT OF ATMOSPHERIC PARAMETERS ON SOUND ENERGY

2.2.1 Effect of Amplitude and Wavelength.

The expression for velocity is independent of the amplitude and the wavelength of the note and therefore we can safely conclude that velocity does not depend upon the amplitude and the wavelength. This holds so long as the amplitude is small as is generally the case. For waves of large amplitude the relationship is not true and it has been found that loud sounds such as those of a cannon travel faster than ordinary sounds in the immediate *neighborhood* of the source. In such a wave, pressure is not proportional to density and the volume elasticity increases as the density is increased by compression and diminishes as the density is reduced by rarefaction. Consequently, a compression wave travels faster and a rarefaction wave slower, resulting in a change of waveform. Moreover, due to the rapid compression in the early stages, there will probably be a rise in temperature, which will cause an increase of velocity. At distance from the source the sound velocity settles down to normal value corresponding to that for small amplitudes.

2.2.2 Effect of Pressure

The change in the pressure of a given mass of gas would produce a corresponding change in volume. From Boyle's law,

PV=a constant.

$$\text{Since} \quad V = \frac{m}{\rho} \qquad 2.1$$

Therefore,

Pm/ρ= a constant.

or P/ρ remains unaltered .

The velocity is thus independent of any change in the atmospheric pressure so long as that change is sufficiently slow and Boyle's law holds good.

This has been verified experimentally and it has been found that the velocity of sound at high altitudes is the same as at sea level, although the atmospheric pressure at the two places is different (Subrahmanyam and Lal. 1974).

2.2.3 Effect of Density

The pressure remaining the same, the velocity of sound in two gases are inversely proportional to the square root of their densities, provided γ (the ratio between the specific heat at constant pressure and the specific heat at constant volume) has the same value for both of them e.g. oxygen and hydrogen. Oxygen is sixteen times denser than hydrogen; therefore the velocity of sound in hydrogen is four times that in oxygen,

$$\frac{velocity\ in\ hydrogen}{velocity\ in\ oxygen} = \sqrt{\frac{\gamma P}{\rho_h}} \div \sqrt{\frac{\gamma P}{\rho_o}}$$

Where P is the total pressure, ρ_h and ρ_o are hydrogen and oxygen densities respectively and γ is the ratio between the specific heat at constant pressure and the specific heat at constant volume. (Subrahmanyam and Lal, 1974).

2.2.4 Effect of Temperature

When the temperature increases, the density of the gas decreases without affecting the pressure. As the numerator in the expression $\sqrt{\gamma P / \rho}$ would remain unchanged, a rise

$$v_o = \sqrt{\frac{\gamma P}{\rho_o}} \quad and \quad v_t = \sqrt{\frac{\gamma P}{\rho_t}}$$

of temperature would increase the velocity, and vise versa. If v_o is the velocity of sound in air at 0°C and v_t, the velocity at t°C, and ρ_0 and ρ_t are the corresponding densities at these temperatures, then if the pressure remains unchanged,

Therefore, (by Charle's law)

$$\frac{v_t}{v_o} = \sqrt{\frac{\rho_t}{\rho_0}} = \sqrt{(1+\alpha t)},$$

where α is the coefficient of volume expansion.

or

$$\frac{v_t}{v_o} = \sqrt{1 + \frac{t}{273}} \qquad where\ \alpha = \frac{1}{273}$$

whence

$$\frac{v_t}{v_o} = \sqrt{\frac{273+t}{273}} = \sqrt{\frac{T}{T_o}}$$

where T and T_0 are Kelvin temperatures that is, velocities are directly proportional to the square root of Kelvin temperatures.

If, however, the variation of t is not too large, we have

$$\frac{v_t}{v_0} = (1+\alpha t)^{1/2} \approx \left(1+\frac{1}{2}\alpha t\right)$$

expanding by binomial theorem and neglecting the higher powers.

Or $$\frac{v_t}{v_0} = \left(1+\frac{1}{2}.\frac{t}{273}\right) \approx \left(1+\frac{t}{546}\right)$$

Taking the value of $v_0 = 332\ ms^{-1}$,we have

$$v_t = 332\left(1+\frac{t}{546}\right)$$

or $$v_t = (332+0.61t)ms^{-1}$$

Thus the velocity of sound at 0^0C increases by 0.61 ms^{-1}. This is called temperature coefficient of velocity of sound and agrees very well with the experimental value. (Subrahmanyam and Lal. 1974). This approximation holds good for air, and at ordinary temperatures. Sound waves produced in the air by sources that have a speed greater than that of sound itself in no way resemble ordinary sound waves. They are called supersonics. Besides them there are ultrasonics-sound waves of frequencies much higher than the audible limit.

2.3 Physical Aspects of Sound

Sound is the mechanical vibration of an elastic medium. Influenced by an outer force, the particles of the elastic material are displaced and because of the elastic force and the inertness they begin to vibrate. Vibration propagates in solids, liquids and gaseous materials, as well. As the human ear perceives generally airborne sounds, the generation, propagation and perception of airborne sound is of essential importance. Airborne sound appears as the fluctuation of air pressure. Sound pressure can therefore be regarded as an alternating component superposed to constant (or very slowly varying) atmospheric

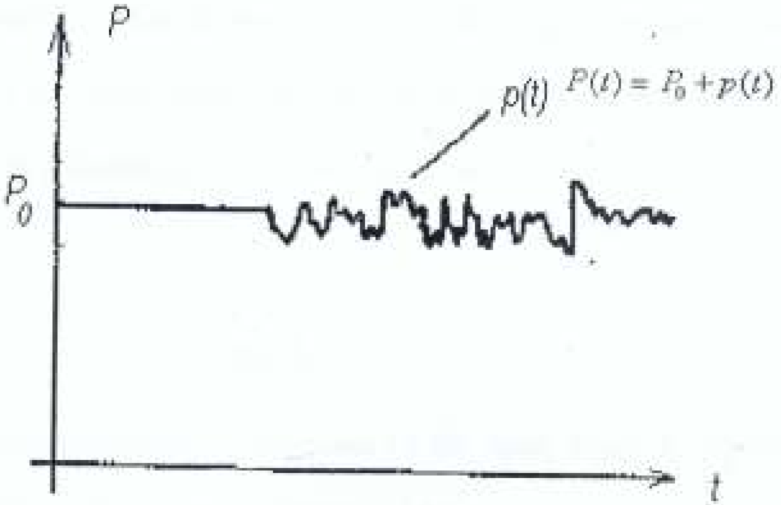


Figure 2.1 The Air Pressure

pressure (Figure2.1.). Air pressure P *(t)* in a given point of space can be expressed as the sum of the atmospheric pressure P_0 and of the sound pressure *p(t)*.

To characterize the magnitude of sound, the effective value of sound pressure is used. The standardized unit of sound pressure is the Pascal (1 Pa = 1 N/m^2). (Atmospheric pressure is about 100.000 Pa). Sound pressure is measured by microphones. Instead of its actual value, sound pressure is usually given in dB being compared to certain reference pressure. 20 mPa is used as reference, since this is the level of the 1 kHz sinusoidal sound just yet audible by the average human ear. The sound pressure level (SPL) can therefore be given as:

$$SPL = 20 \log p/p_0.$$

Pressure difference generated in one point of the space tends to equalize towards the adjacent parts. During this process, particles of the air are displaced generating thus a new pressure difference in the neighboring space. So the sound pressure variation propagates in the form of sound waves. The distance between two points having the same phase of the sound wave is called the *wavelength*. The product of the frequency and the wavelength is equal to the *propagation velocity* of the sound wave:

$$C = f\lambda$$

The propagation velocity of the sound is 340 m/s. If the sound source is concentrated in a single point and if there is no obstacle in the surrounding field then the sound waves are spherical. Far enough from the source, the curvature of the sphere can be neglected and the wave is supposed to be plain (Fig. 2.2.). For plain waves the ratio of sound pressure and particle velocity is constant.

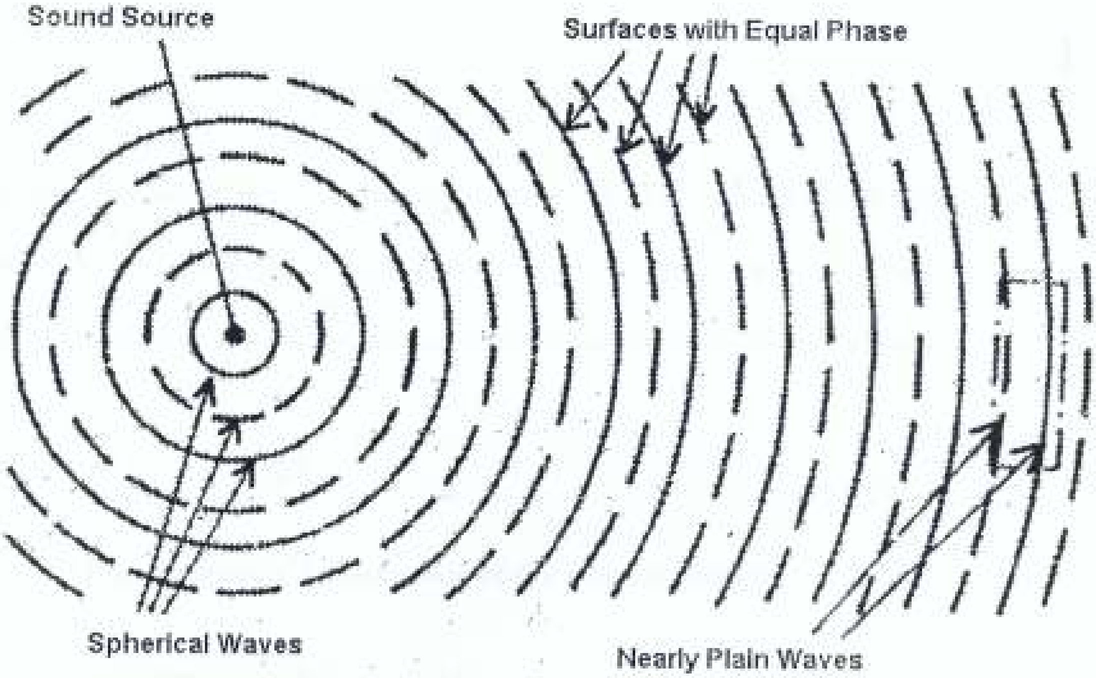


Figure 2.2 Spherical and Plane Waves

$$\frac{P}{V} = \rho_0 c = 410 \frac{Kg}{m^2 . s}$$ where ρ_0 is the air density.

Sound can also be characterized by the acoustic power, which comes through a unit of area. This quantity is called *sound intensity* and its value can be expressed as the product of sound pressure and particle velocity:

$$I = pv = \frac{p^2}{\rho_0 c}$$

Intensity is usually given in related form in dB, too. The reference is the same as for sound pressure. It can be seen easily that the reference value is 10^{-} 1 pW/m^2 which is the intensity of a just yet audible 1 kHz tone. Intensity level can, therefore, be expressed as

$$L_I = 10 \log \frac{I}{I_0}$$

2.4 Physiological Characteristic of the Human Hearing

Human hearing is limited both in frequency and amplitude. According to the test measurements carried out on a very large amount of people, levels of just audible sound pressures were determined. The average of the measured values is called the *threshold of audibility*. The threshold of audibility is strongly frequency dependent. The ear is most sensitive in the range of some few kHz. Below and beyond this range the sensitivity is smaller (Figure. 2.3.) It can be seen that the range of the audible signals is between 20 Hz and 20 kHz. (Cyrl,1979). Too loud sounds cause pain, the lowest of such a sound pressure is called the *threshold of pain*. Musical sounds and the speech are within these limits. It can also be seen that the frequency range and the amplitude range of music are remarkably greater than that of speech.

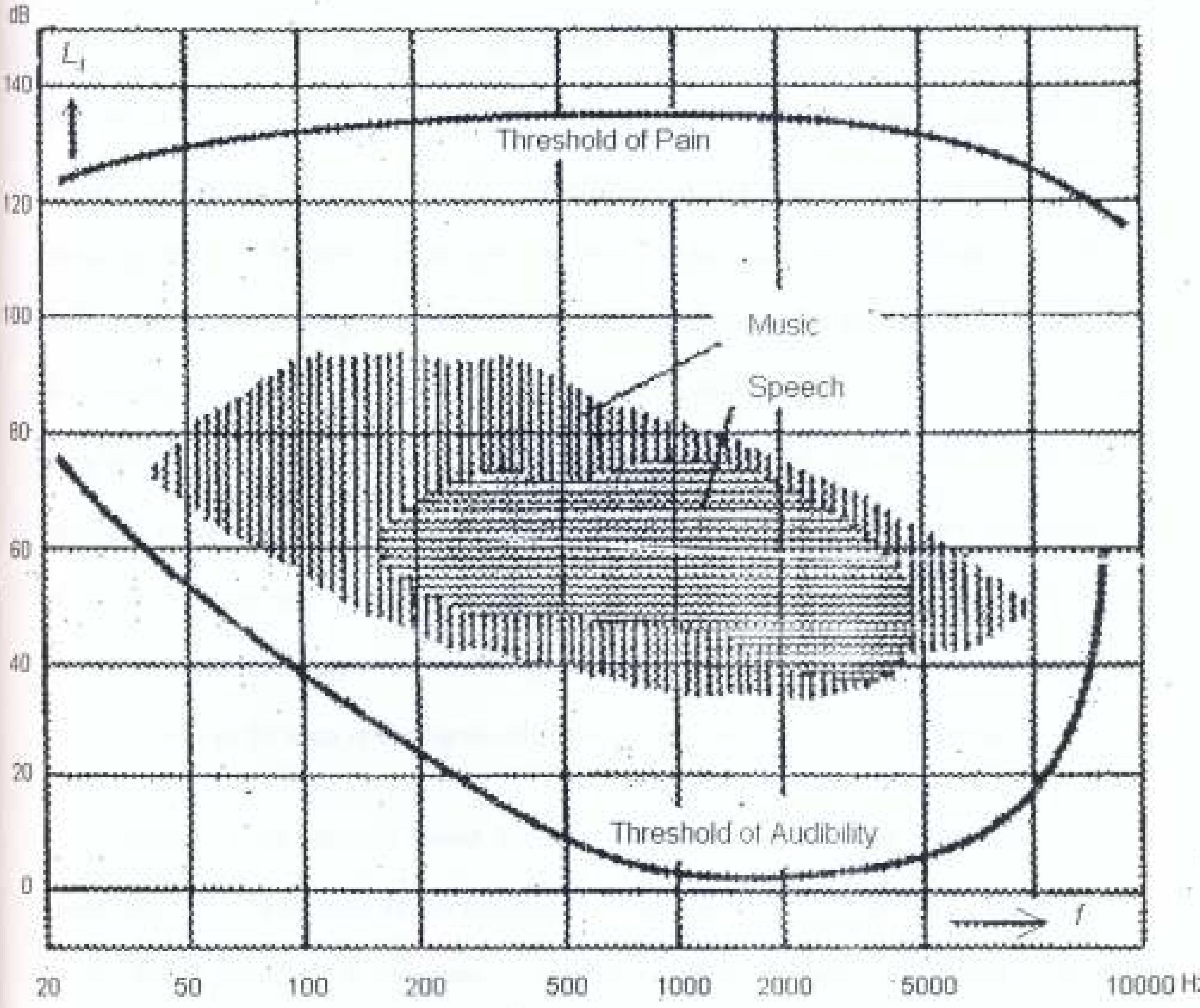


Figure 2.3 Limits of Human Hearing

For the subjective judgement of the sound level, the term of *loudness level* was introduced. Loudness level of an arbitrary sound is as many *phon*-s as many dB-s is the sound pressure level of the 1 kHz tone of the same loudness. (In the loudness evaluation test, the listener alternately listens to and compares the measured sound with the Connecting the points with the same loudness along the frequency axis, we obtain the so-called Fletcher-Munson curves (Figure.2.3). Loudness level of a sound at a given frequency and pressure level can be read as the value of L_N belonging to a certain curve on the diagram. Loudness is thus suitable for the comparison of sounds with different frequencies. To evaluate the resulting level of simultaneous sounds, the term *loudness* has been introduced. Loudness is denoted as N and its unit is called *phon*. If the loudness level is greater than 40 phon then the loudness can be computed as *masking which* is another term related to the simultaneous presence of two different sounds. Masking means covering the weaker sound with a stronger sound when each has a different frequency. Masking has been examined for sinusoidal sounds and for narrow- and broadband noises, respectively. Figure 2.4, (Cyril, 1979) shows the increase of the audibility threshold caused by 1 kHz narrow-band masking noises. As it can be seen in the figure, high frequency sounds are easier to mask.

Spatial parameters of sound are also very important. First of all, *direction of the sound* has to be mentioned. In the horizontal plane, the sound is localized by the difference of the sound pressures at our ears. At low frequencies, the phase difference is detected while on higher frequencies a difference in intensity arises due to the shadowing caused by the head. To perceive direction in the vertical plane, the head has to be moved up and down.

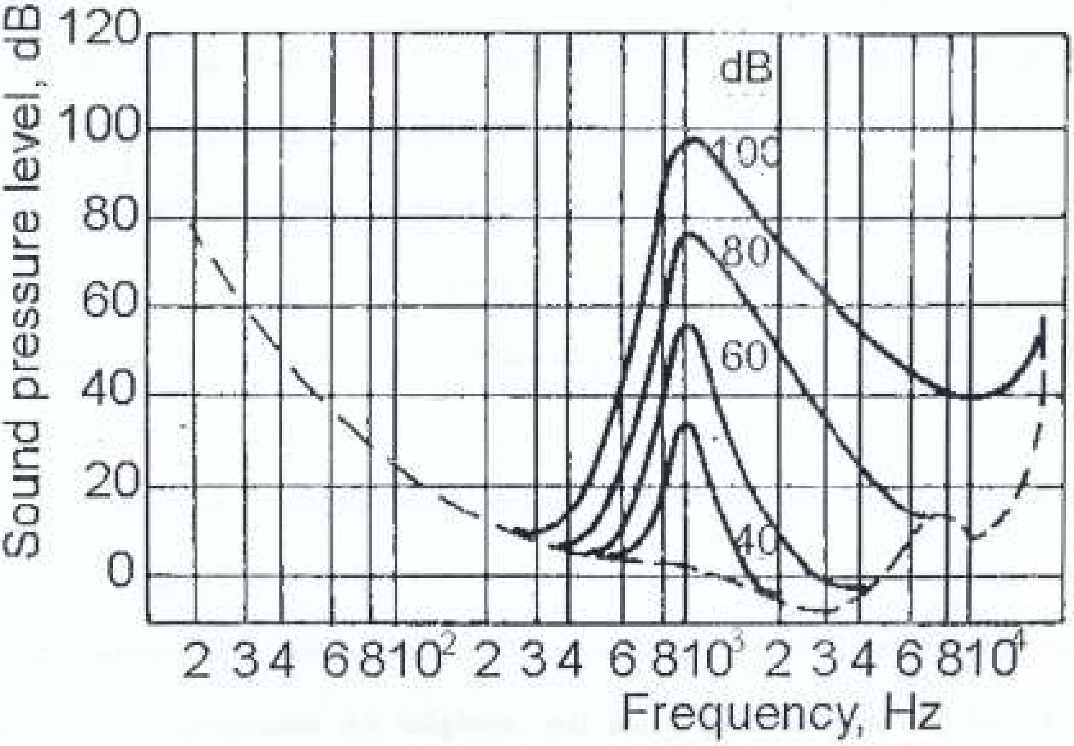


Figure 2.4 Masking

2.5 Electroacoustic Transducers

Electroacoustic transducers are devices transforming electrical energy into acoustic energy or vice versa. Transformation is carried out in two steps. First, electric energy is converted into mechanical energy by means of an electromechanical transducer and second, from mechanical energy to acoustic energy. The important part of such a transducer is a mechanical vibrating system, which is rigidly attached to a diaphragm. The mechanical vibration forces the air particles adjacent to the diaphragm to move so that mechanical energy is converted into the acoustic energy of the propagating sound waves. In the *inverse* effect, incoming sound waves bring the diaphragm and the mechanical system into motion so that a signal proportional to the motion is generated at the electrical output. In certain transducers the diaphragm and the mechanical part cannot be clearly distinguished from each other (Amos, 1977).

The so-called *controlled* transducers use the input signal to control output energy of an external source. This principle has the advantage of being able to control much greater energies than the input energy. That is why these transducers are usually called *active* as well. Carbon microphones for telephone sets are a good example for such kind of transducers.

The *electromagnetic transducer* (Figure 2.5) consists of a permanent magnet, soft magnetic pole-shoe, an anchor and a spring. At the standstill position the gap size is *s*/2, attracting force of the magnet is in equilibrium with the force of the spring. In the presence of a current flowing through the coil, attracting force gets stronger and the air gap becomes smaller. On the contrary, a current flowing in the opposite direction weakens the attracting force thus the anchor moves off. In the inverse operation, force generated by the sound pressure moves the anchor. In accordance with the direction of the movement, a flux change will induce voltage in the coil.

The *electrodynamic transducer* (Figure. 2.6) has a fixed air gap. Inside the air gap, there is a magnetic field in which a conductor is moving. At the ends of the conductor, a voltage is induced which is proportional to the flux density in the gap, to the length of the conductor and to the velocity of the moving conductor. Thus the motional energy is transformed into electrical energy. When a current drives the conductor, a force is generated which is also proportional to the flux density, to the conductor length and to the magnitude of the current. To increase the efficiency of the transducer, a moving coil is used instead of a single piece of a straight conductor, since the entire wire length is of importance.

The moving electrode is made of a thin metal foil acting also as the microphone diaphragm. The fixed electrode is called back plate and is made of a thick piece of metal. To ensure linear operation of the transducer, a DC voltage source is connected to the microphone through a high value resistor. The electrostatic force attracts the thin diaphragm towards the back plate. As the edges of the diaphragm are fixed, the foil becomes deformed. This deformation is strengthened or weakened by an additional AC voltage, depending on its sign with respect to the DC voltage so that the diaphragm starts to move. If sound pressure is acting on diaphragm, it is again more or less deformed and due to this, capacity of the condenser formed by the two electrodes will change. As the charge of the capacitor is unable to change during rapid changes of the capacitance, the voltage changes and the change can be monitored on the resistance R. The *piezoelectric transducer* makes use of the ability of certain materials to produce unbalanced charge distribution on their surface when they are deformed. In the inverse effect, deformation appears when electrical field is applied.

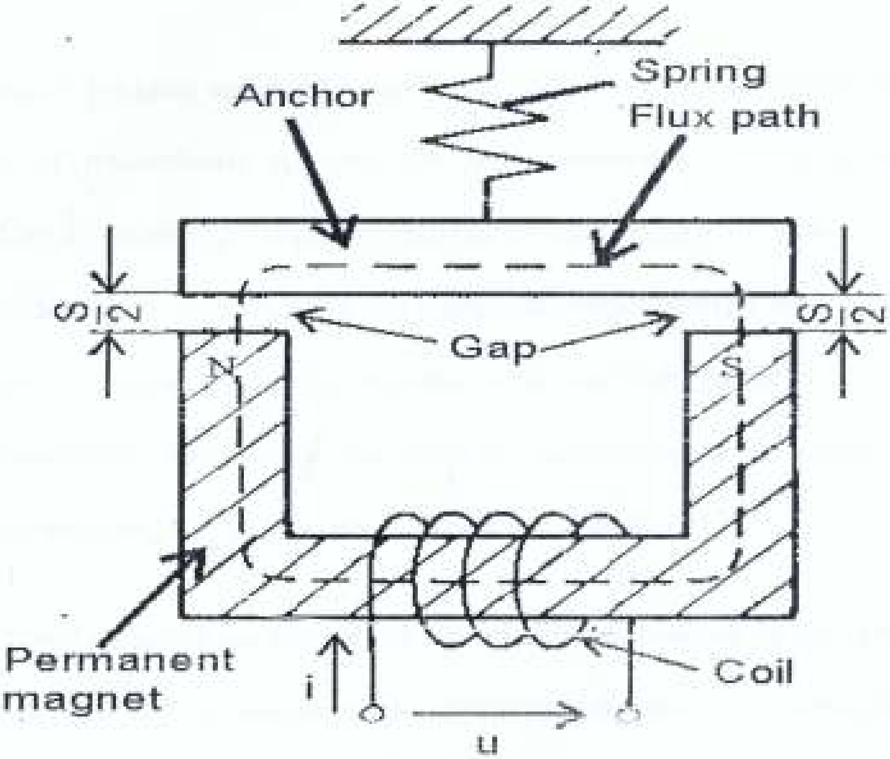


Figure 2.5 Electromagnetic Transducer

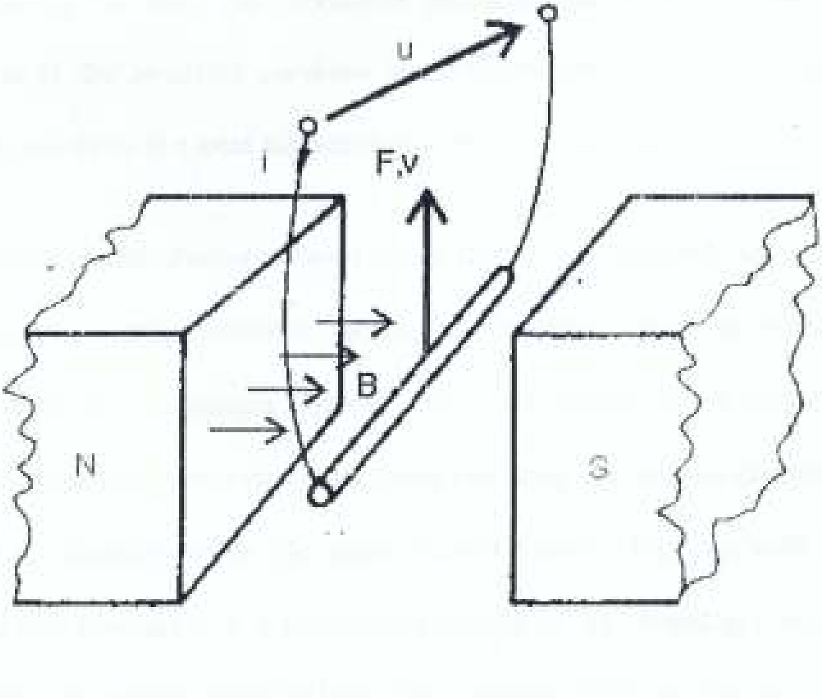


Figure 2.6 Electrodynamic Transducer

2.5.1 Microphones

Because of the great variety of requirements for the transmitted and recorded sound, several types of microphones are used for sound reception. A microphone can be characterized by its sensitivity, frequency response of the sensitivity and by the directional characteristic. *Sensitivity* is defined as the change in output voltage with corresponding change of pressure applied. *Frequency response* is the sensitivity given as the function of frequency. *Directional characteristic* expresses the dependence of the sensitivity on the direction of the incoming waves as shown in figure 2.8. (Amos, 1977)

Directional characteristic depends on the microphone housing. In the case of closed housing, the microphone is *omnidirectional* (sensitivity is the same in all directions). If the housing is opened (both sides of the diaphragm can be accessed by the sound pressure) the microphone is *bilateral*, i.e. it is entirely insensitive to side sound-waves and most sensitive to sounds coming in from the directions perpendicular to the diaphragm. The third important type is the so-called *cardioid* microphone which has great sensitivity to one direction only and there is a total suppression in the backward direction.

In telephone sets, mass-produced cheap *carbon microphones* are used (figure 2.9). Such a microphone acts as a resistor, varying its resistance according to the change of the sound pressure on the diaphragm. This is due to the change of the contact resistance of carbon powder placed between two gold plated contacts. The bottom electrode is fixed and isolated from the housing while the upper electrode moves together with the diaphragm. Since the contact resistance is a nonlinear function of the diaphragm displacement, the distortion of the microphone is rather high. The widespread use of this type of microphones

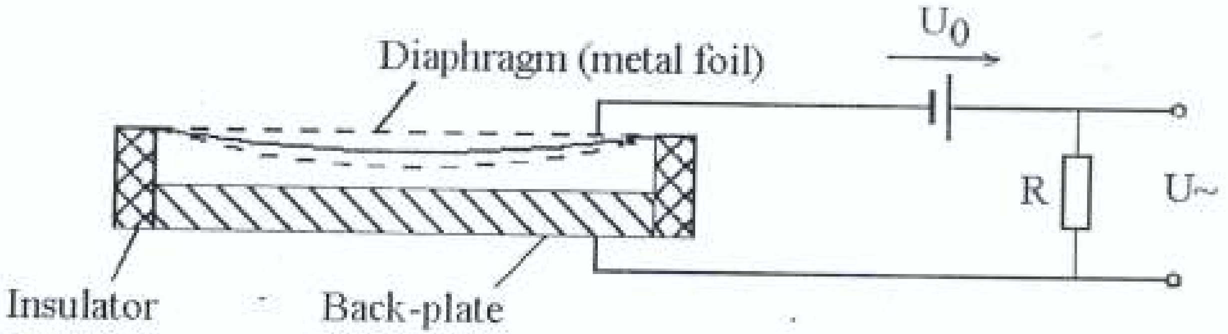


Figure 2.7 Electrostatic Transducer

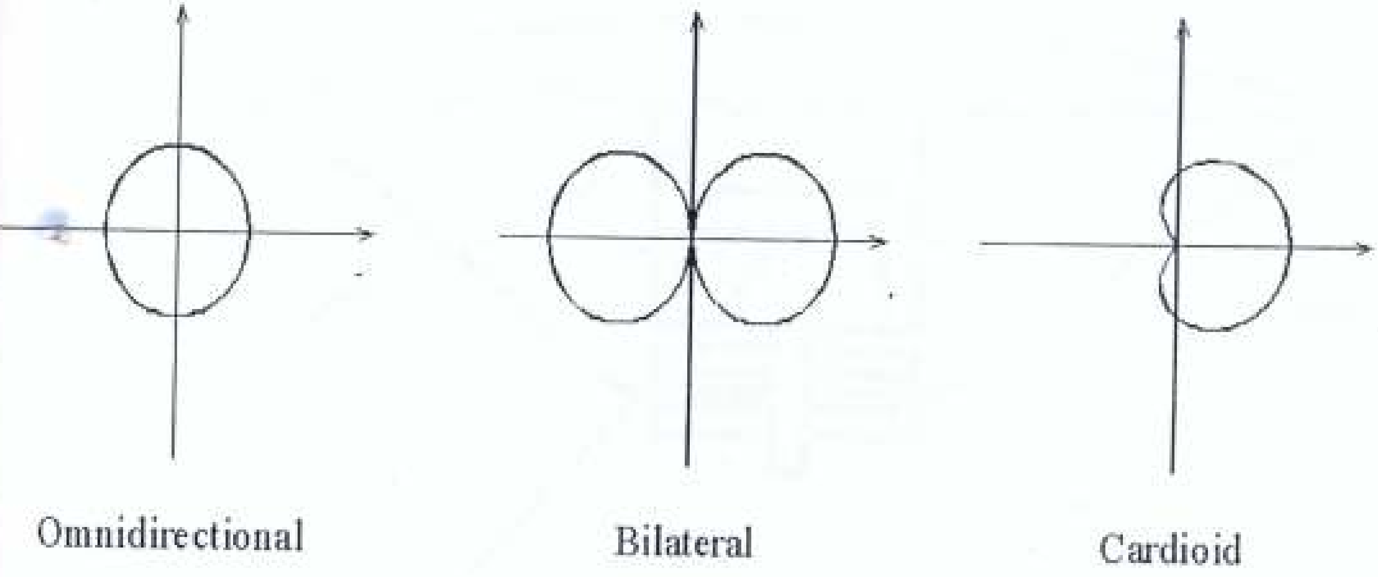


Fig. 2.8 Microphone Directivity Patterns

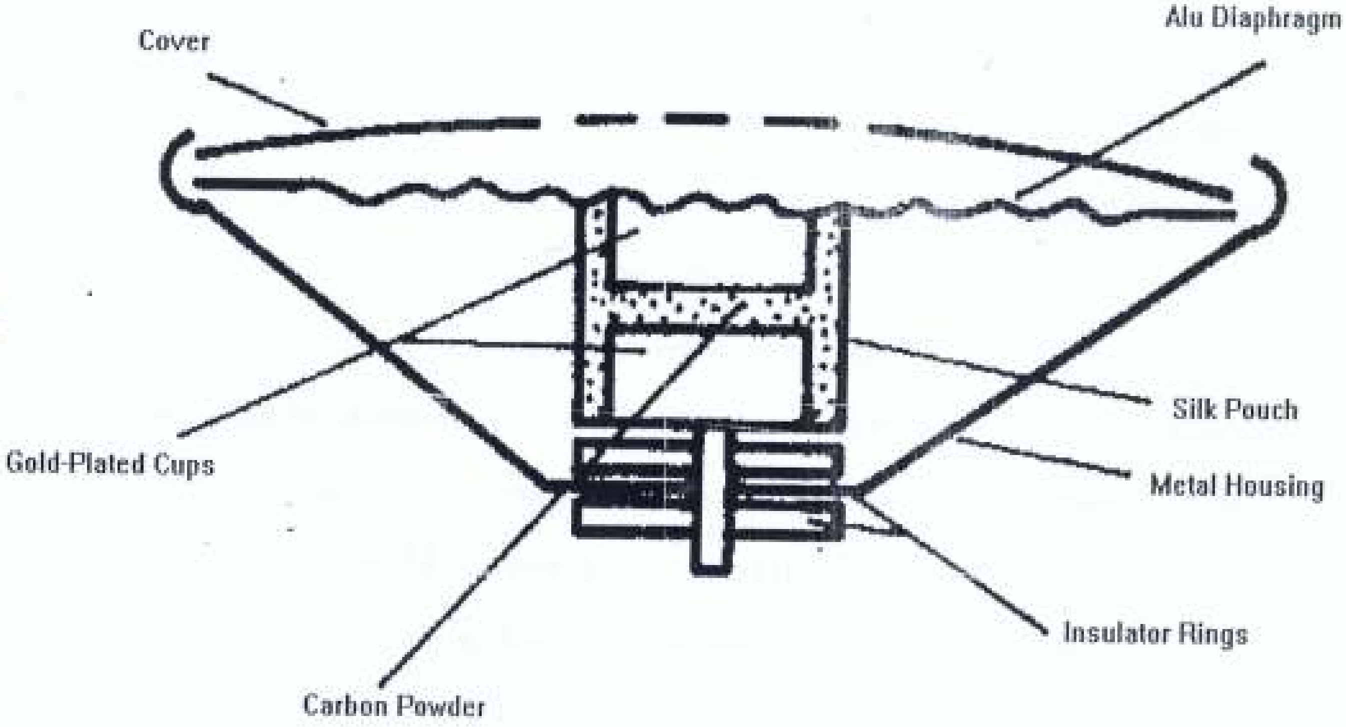


Figure 2.9 Carbon Microphone

is due to its great output signal. Currently, they are loosing importance as fully electronic telephone sets enter the market.

Both in studio and consumer applications, *dynamic microphones* (figure 2.10) are frequently used. Induced output voltage appears at the ends of the moving coil, inserted into the air gap of a permanent magnet. The coil moves together with the diaphragm and the motion is proportional to the incoming sound pressure. The magnet and the diaphragm are placed into a housing closed from the front side by a protecting grid. Opening the housing, choosing proper grid parameters and placing additional acoustic elements,

broadband frequency response and arbitrary directional characteristic can be realized. The *condenser microphone*, figure 2.11, is a device suitable for both studio applications and for measurements. There is a disk-shaped back-plate, isolated from the cylindrical metal housing at the end of which a metal diaphragm is stretched. The distance between the two electrodes is about 0.01 mm. Alternating voltage, which appears on the resistance is amplified by a low-noise high input impedance preamplifier. The sensitivity of such a microphone can be calibrated and remains stable during a long period of time because of the precise construction. For consumer applications, *piezoelectric microphones* (figure 2.12) are used . A piece of a dual crystal (bimorph) is fixed at one end to the housing while its other end is attached to the diaphragm. The incoming pressure bends the bimorph to produce two voltages of opposite sign, which are then simply added. As the output impedance of the device is high, high input impedance amplifier has to be used for further signal amplification.

2.5.2 Loudspeakers

The last step of artificial sound field production is the transformation of electric energy into acoustic energy. This transformation is performed by loudspeakers. Like with microphones, different types of loudspeakers have been developed.

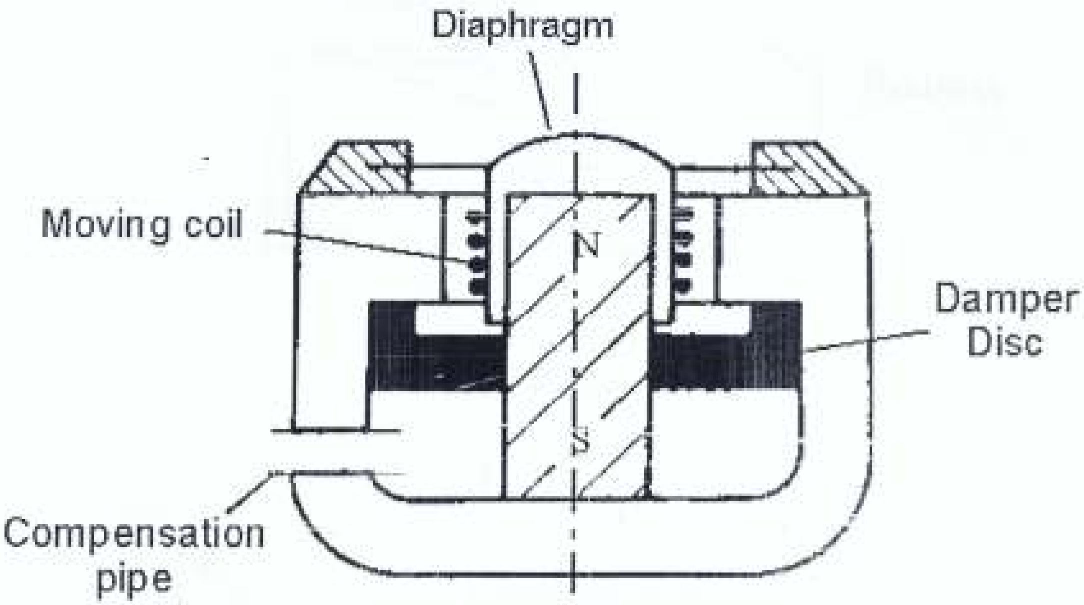


Figure 2.10 Dynamic Microphone

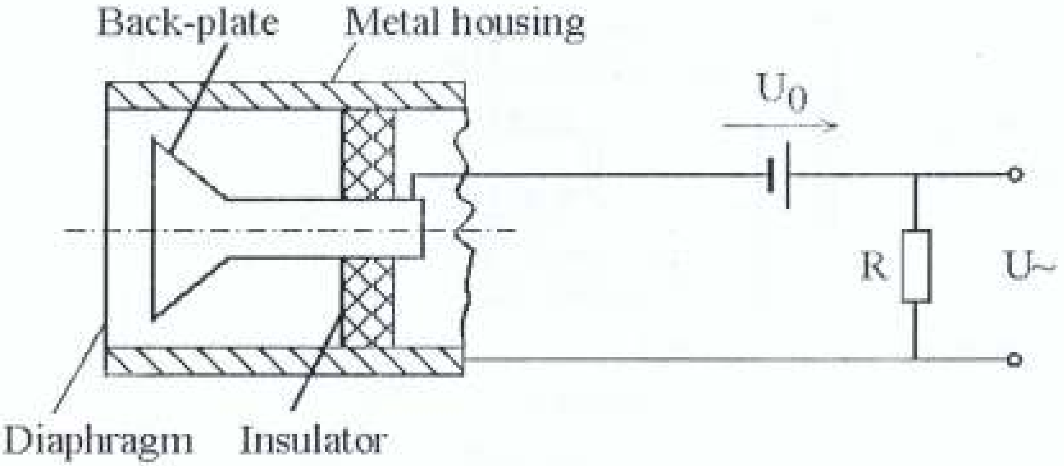


Figure 2.11 Condenser Microphone

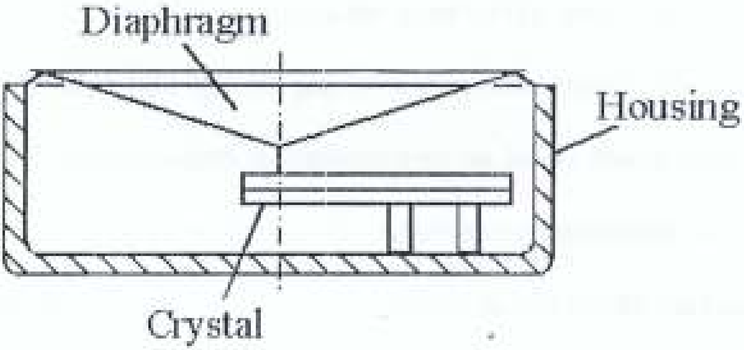


Figure 2.12 Piezoelectric Microphone

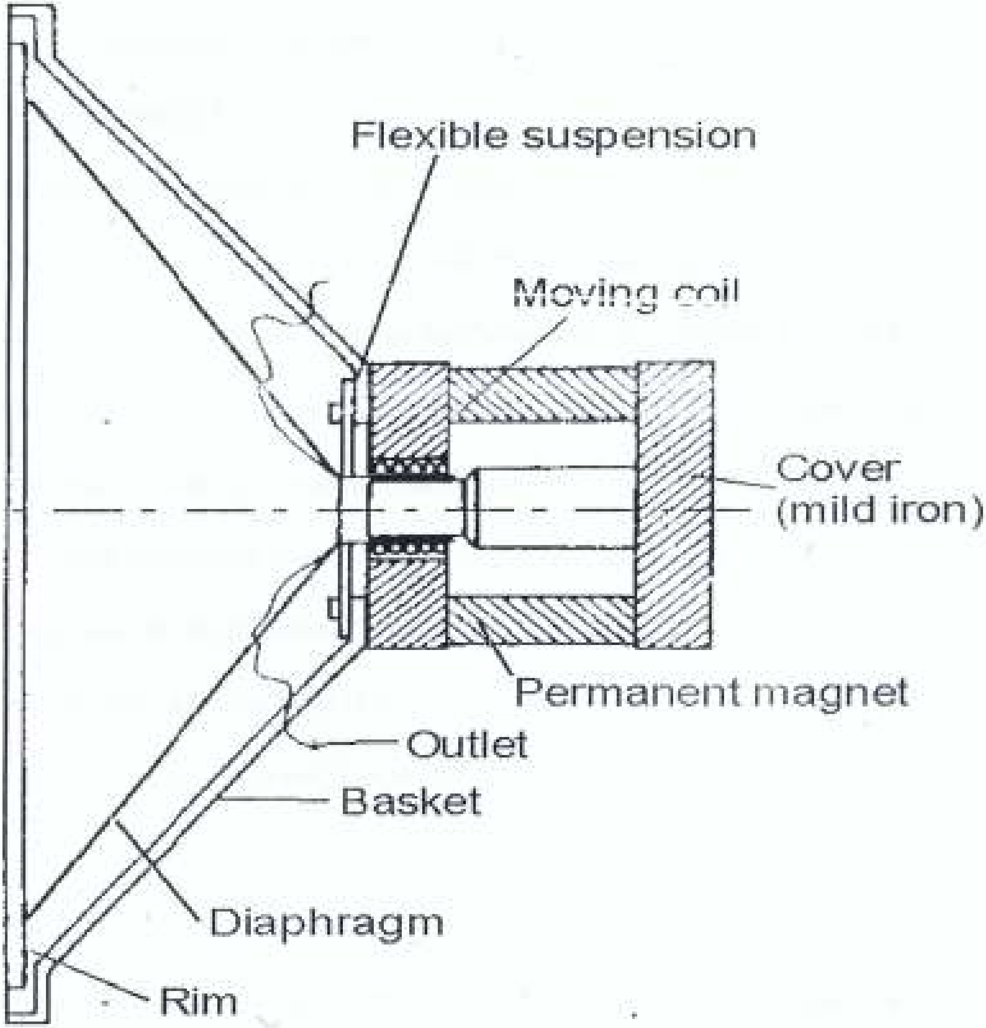


Figure 2.13 Dynamic Loudspeaker

The *dynamic loudspeaker* (Fig. 2.13) is the most widely used type. The moving coil of the loudspeaker is placed into the air gap of a magnetic circuit. The moving coil is attached to a conical diaphragm which is supported by an inside spider and by an outside rim to keep the motion axial. Frame attached to the magnet supports the rim and the leads of the moving coil end also on the frame. By the mutual action of the moving coil current and the magnetic field, an axial force is generated. This force brings the diaphragm into motion thus producing sound waves. For modest quality requirements (e.g. for AM radio receivers) one loudspeaker is satisfactory. Two or three loudspeakers can cover the full audio range, each designed for a different frequency range.

Efficiency of sound emission can be improved by better acoustic matching. Such an improvement can be achieved by a *horn* with exponentially growing cross sectional area. The transducer is placed into the throat of the horn. The only drawback of the horn is that a very great horn is needed for a good low frequency response, so that horn is used only for sound reinforcement requiring modest sound quality (where the lack of the basses can be tolerated). In a *condenser loudspeaker*, to achieve good low frequency response, the diaphragm area has to be extremely great and a special high voltage power supply and matching transformer are also needed to provide proper dc bias and impedance matching. Distortion of such a device is small, frequency response is flat but the price is rather high.

2.6 Offset Error Compensation.

Real operational amplifiers usually have small dc output voltage even when the input voltage is zero. This small dc output voltage, in excess of ±15mV, depends on the types of transistors used in the input circuitry of the op-amp. This voltage is called an input offset voltage, which is an error and undesirable in many applications. It is also an input voltage, which must be applied across the inputs of the op-amp to force the output voltage down to zero. Manufacturers often quote it as V_{IO}. Any circuit that enables us to achieve this is known as input-offset voltage nulling network. Offset voltage nulling is usually termed as internal and external nulling. (Franco, 1988).

Internal Offset Nulling: The nulling procedure is straightforward as shown in figure 2.14 for a ua741 op-amp used in this project. It must be noted that for nulling to be carried out, the op-amp must have provision for it.

External Offset Nulling: Also external offset nulling is based on the injection of an adjustable voltage into the input circuit to compensate for the existing error. This external nulling does not introduce additional imbalance in the input stage, hence there is no degradation in temperature drift and common mode rejection ratio. This external nulling is shown in figure 2.15a and 2.5b for both noninverting and inverting amplifier respectively. The potentiometers in the circuits are adjusted to drive V_0 to zero in each case.

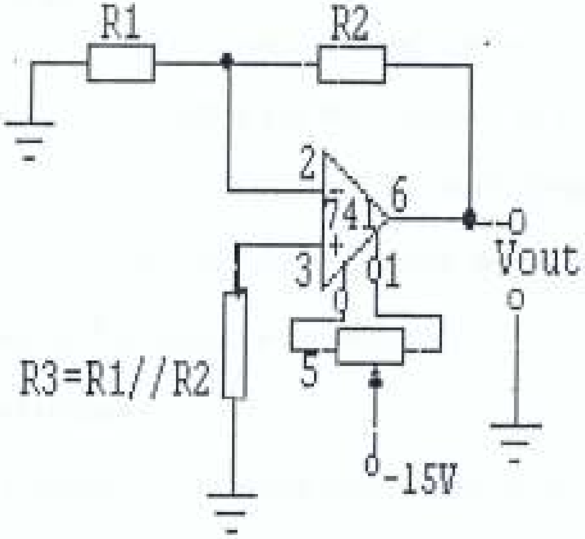
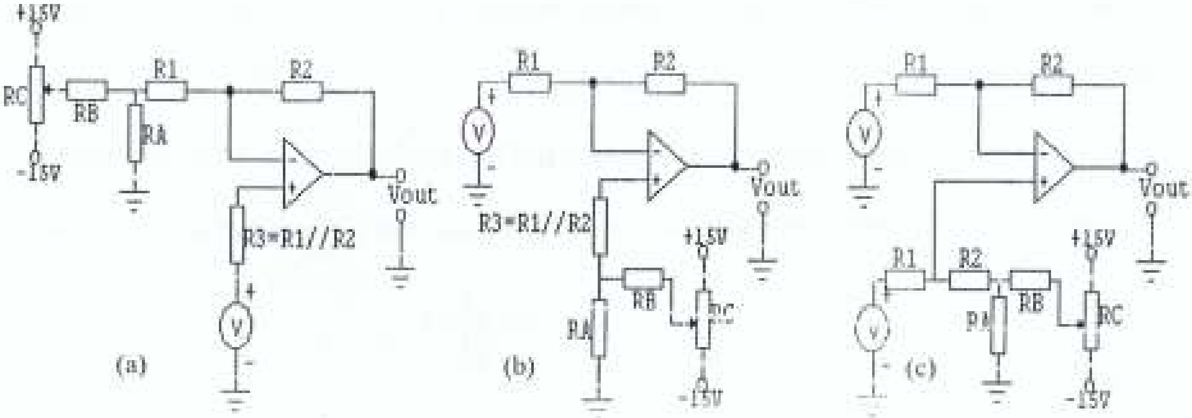


Fig 2.14: Resistor feedback configuration with internal offset nulling



(a) (b) (c)

Fig 2.15 External offset nulling for (a) non-inverting amplifier, (b) inverting amplifier and (c) difference amplifier

CHAPTER THREE

3.0 THE DESIGN PROCEDURE

To determine the coefficient of attenuation of sound waves in the atmosphere two circuits were developed to form the transmitter and the receiver .The arrangement is illustrated in figure 3.1.The transmitter comprises of the oscillator, audio power amplifier and the electro acoustic transducer. The receiver is simply a sound pressure meter that measures the intensity at fixed distances away from the transmitter.

3.1 The Audio Oscillator

The audio oscillator is simply a Wien bridge arrangement that generates a sine wave output. The basic Wien bridge oscillator configuration is as shown in figure 3.2. The active device is an operational amplifier to which both negative and positive feedback is applied. Negative feedback is provided by resistive network R_1 and R_2 while positive feedback is by the reactive network consisting of a parallel and a series RC pair. The reactive components need not be of the same value but making them so simplifies the circuit's analysis.

The circuit can behave as a non-inverting amplifier, which amplifies V_p by the amount

$$A = \frac{V_0}{V_p} = 1 + \frac{R_2}{R_1}$$

V_0 is the output voltage and V_p is the voltage across parallel components.

V_P is in turn supplied by the operational amplifier itself according to

$$\frac{V_P}{V_0} = \frac{Z_P}{(Z_P + Z_S)}$$

where $Z_p = R // 1/\left[1/(j\omega C)\right]$ and $Z_S = R + \frac{1}{(j\omega C)}$

Z_p and Z_s are parallel and series impedances respectively, j is a complex term and ω is the angular frequency.

(a) Sound Wave Transmitter

(b) The Receiver

Figure 3.1 Block diagrams of the transmitter and receiver

Substituting and expanding yields

$$B=\frac{V_P}{V_0}=\frac{1}{3+j\left(\frac{f}{f_0}-\frac{f_0}{f}\right)}$$

where f_0 is the output frequency given as $f_0=\frac{1}{2\pi RC}$

The overall gain experienced by a signal in going around the loop is

T=AB,

that is $$T(jf)=\frac{1+R_2/R_1}{3+j(f/f_0-f_0/f)}$$

This is a band pass function since at both high and low frequencies it approaches zero. Moreover, it peaks for $f=f_0$ where

$$T(jf_0)=\frac{1+R_2/R_1}{3}$$

Depending on the magnitude of T (jf_0), we have three possibilities : (Franco,1988)

1. T $(jf_0)<1$.This correspond to a stable situation in which all signals in the circuit eventually converge to zero
2. T $(jf_0)>1$. This corresponds to an unstable situation, since any disturbance with spectral content in the vicinity of f_0 will be amplified regenerative and cause the circuit to break out into oscillation of growing magnitude.
3. T $(jf_0)=1$. This is a sustained oscillation. It means keeping the amplitude of oscillation below the saturation levels, yet providing enough loop gain to prevent oscillation from decaying to zero. This is achieved by making T (jf_0) exactly unity. This requires that

$$\frac{R_2}{R_1}=2$$

The condition leading to T $(jf_0)=1$ is also referred to as the neutral stability, which is achieved when the resistive and the reactive feedback networks form a balanced bridge. To keep the bridge balanced exactly at all times, a form of automatic control was used to continuously adjust the ratio R_2/R_1. This is achieved by making the ratio level dependent so that at low signal levels R_2/R_1 is slightly greater than 2 to ensure oscillation start up, and at high signal levels it is slightly less than 2 to provide amplitude limiting. Once the oscillation has started, it will grow and automatically stabilize at the level that makes $R_2/R_1=2$ exactly.

The circuit of figure 3.3 uses a diode-resistor network to make the effective value of R_2 level sensitive. At low signal levels, where the voltage drop across the diode network is insufficient to turn the diodes on, the 200kΩ resistor is effectively open- circuited and $R_2/R_1=21/10=2.1$. This causes oscillation to build up. As the oscillation grows, the diodes are gradually brought into conduction, thus making the effective value of R_2 change from 21kΩ to 21//200=19kΩ. At high signal levels $R_2/R_1=19/10=1.9$. Signal amplitude will automatically stabilize at some immediate level where $R_2/R_1=2$. (Franco, 1988). With the component values shown in figure 3.3, the output amplitude yields peak to peak of about 1.5 volts which is good enough to be fed into the input of the audio power amplifier.

3.2 THE AUDIO POWER AMPLIFIER

The amplifier is powered from a dual power supply of ±30V, which determines the maximum possible power that can be delivered to an 8Ω loud speaker as:

$$P=\frac{V^2}{2R_L}=\frac{(30)^2}{16}=56.25W$$

In practice, the available power will be rather less, say approximately 50W, because of the inclusion of the 0.5Ω series resistor for overload safety purposes, bottoming and cut off limitations in the output transistors and the impossibility of perfect biasing. To conserve standby power and for maximum efficiency, the output stage of figure 3.4 is arranged as class AB, complementary symmetry, push- pull configurations.

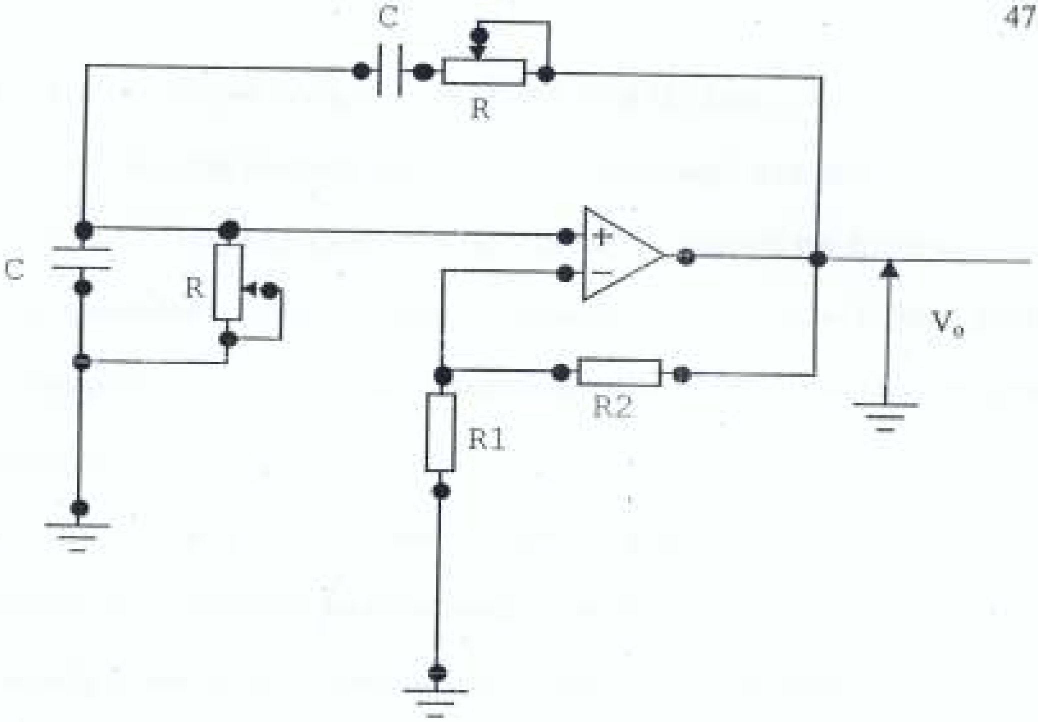


Figure 3.2: Basic Wien Bridge Oscillator Configuration

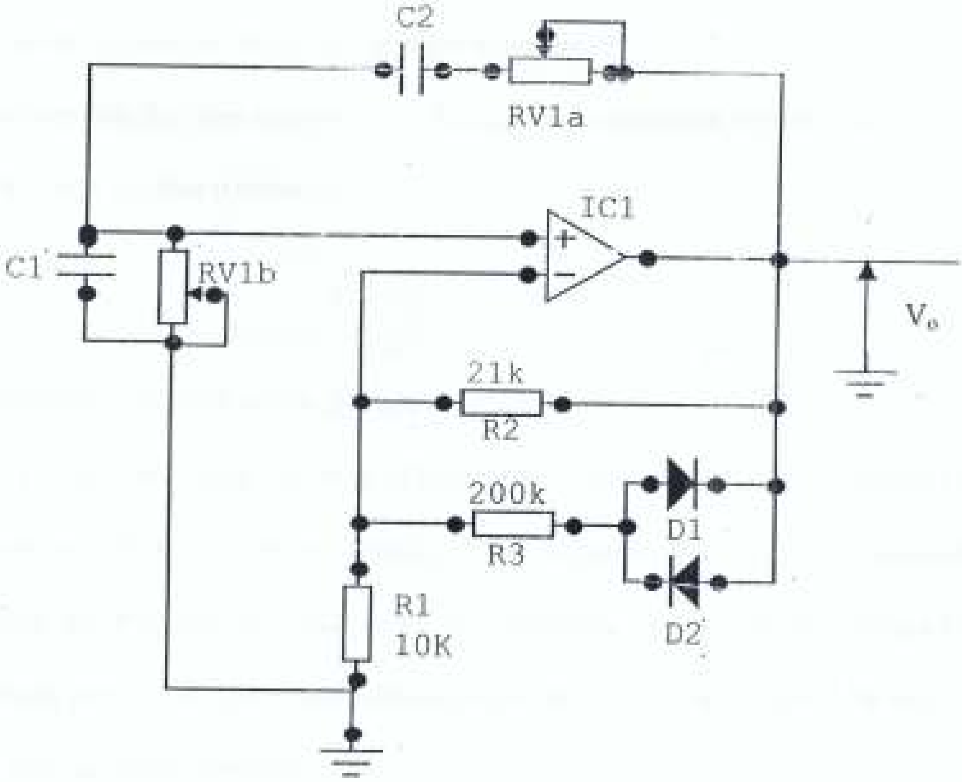


Figure 3.3: Wien Bridge Oscillator With Automatic Amplitude Stabilization

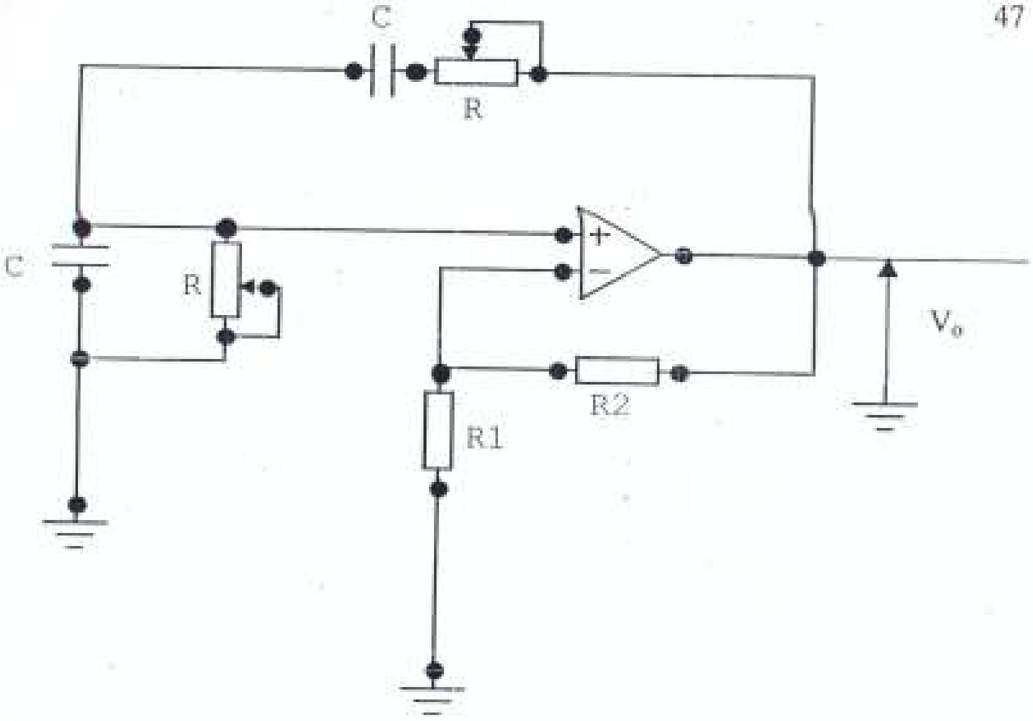


Figure 3.2: Basic Wien Bridge Oscillator Configuration

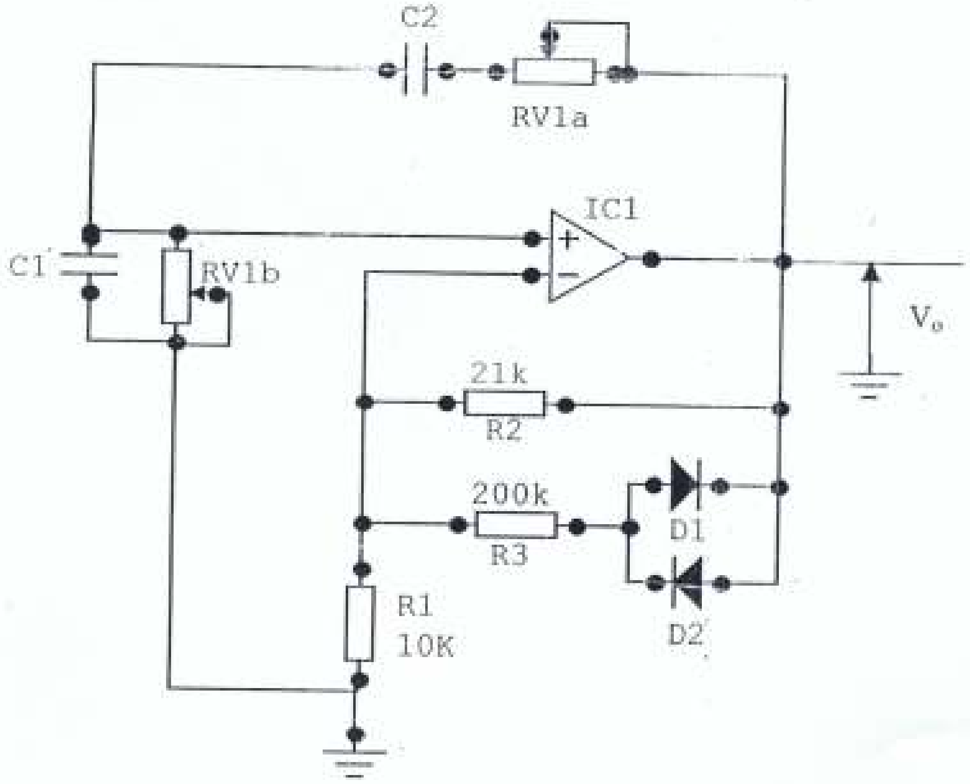


Figure 3.3: Wien Bridge Oscillator With Automatic Amplitude Stabilization

The common collector configurations formed from the Darlington pair T_5 and T_6 yield almost 100% negative feedback and endow the output stage with unit voltage gain, very low distortion and very high input resistance to avoid overloading the driver stage and very low output resistance for driving the low impedance speaker. Due to high power dissipation, T_5 and T_6 were mounted on a common heat sink taking appropriate care over electrical insulation.

Transistors T_7 and T_8 protect T_5 and T_6 respectively against overload which might arise, for example, from accidental short-circuiting of the loudspeaker. The bases of T_7 and T_8 sample the output current and, if this becomes too great, these two transistors switch on and limit the input to T_5 and T_6.

Diodes D_1 and D_2 protect T7 and T_8 against reverse current flow while D_3 and D_4 protectT_1 and T_2 against reverse inductive voltage surges. Transistor T_4 in conjunction with R_4 , R_5 and R_6 act as a base emitter voltage (V_{BE}) multiplier and provides bias for T_5 and T_6 that can be almost perfectly thermally compensated.

Assuming that the base current of T_4 is negligible compared with the current through R_4, R_5 and R_6, then for this transistor,

$$V_{CE} = \left[\frac{(R_4 + R_5 + R_6)}{R_6}\right] V_{BE}$$

Adjustment of R_5 allows the potential difference between the bases of the Darlington's T_5 and T_6 to be set within the range three to four times base emitter voltage (V_{be}) which is enough to turn them just on. Ideally, R_5 is adjusted for minimum crossover distortion. Because the bias between the bases is a multiple of V_{be}, thermal compensation of the operating point of the push-pull Darlington is automatically achieved by mounting T_4 close to T_5 and T_6 on the same heat sink.

Transistor T_4 presents almost a short circuit to signals between its collector and emitter. Thus, R_3 is the collector load of the driver transistor T_3, which is biased to operate in class A. The ambient dissipation of T_3 is given as 30V x 10mA =300mW and a small heat sink was attached to it also.

The long -tailed pair, formed from matched transistors T_1 and T_2 feeds T_3 and provides valuable differential action with common mode rejection. Substantial negative feedback through R_1 and R_2 keeps the overall amplifier gain at 28 with low distortion and in conjunction with the differential input stage, maintains the ambient output voltage very close to zero without the need for a preset adjustment The full rated output is delivered when the input is $1V_{rms}$. Larger gain is of course easily obtained by increasing R_2.

The input resistance of the amplifier is simply the 22kΩ resistor because the parallel input resistance of the feedback long-tailed pair is very high. The amplifier is completely dc coupled at the input, its low frequency response is very good without the need for very large coupling capacitor. High frequency stability is assisted by including a 1nF capacitor in the collector circuitry of T_3. This capacitor together with R_3 constitutes a low pass filter that attenuates signals at frequencies well above the audible range.

The frequency response of the amplifier was quite good between 30Hz to about 50kHz, and it can drive 8Ω or 4 loudspeakers.

The amplifier was powered by a rectified ac voltage using a center tapped 50V step-down transformer which was well filtered to remove most of the ripple voltages which can cause noise and humming at the amplifier's output. The rectified output gives ±30V and can provide more than 2A of current.

3.3 THE LINEAR SOUND PRESSURE METER

Usually sound intensity or pressure meters are calibrated on a logarithmic scale but the one designed in this project is calibrated on a linear scale. This linear scale does not affect the output reading but makes it easier. This sound pressure meter uses a special integrated circuit, AD636, that is a true root mean square (RMS) converter.

3.4 AD636 PRINCIPLE OF OPERATION

This AD636 embodies an implicit solution of the RMS equation that overcomes the dynamic range as well as other limitations inherent in a straightforward computation of rms value of ac or dc signals. (Application sheet of AD636). The actual computation performed by the AD636 is given by the equation:

$$V_{rms} = Avg\left[\frac{V_{in}^{2}}{V_{rms}}\right]$$

where A_{vg} is the average value of the ac signal.

V_{rms} is the root mean square value, which is the effective value in any ac signal.

V_{in} is the input voltage.

Figures 3.5 and 3.6 shows the schematic and the internal circuitry of the AD636 respectively. The schematic is subdivided into four major sections namely:

(i) absolute value circuit (that is, active rectifier) (ii) squarer/divider circuit

(iii) current mirror circuit and (iv) the output buffer amplifier.

The internal circuitry of AD636 is shown in figure 3.6. The active rectifiers A_1 and A_2 convert the input voltage Vin, which can be an alternating current or direct current, to a unipolar current I_1. I_1 drives one input of the squarer/divider, which has the transfer function:

$$I_4 = \frac{I_1^{2}}{I_2^{2}}$$

The output current, I_4, of the squarer/divider drives the current mirror through a low pass filter formed by R_1 and the externally connected capacitor, C_{AV}. If the time constant,

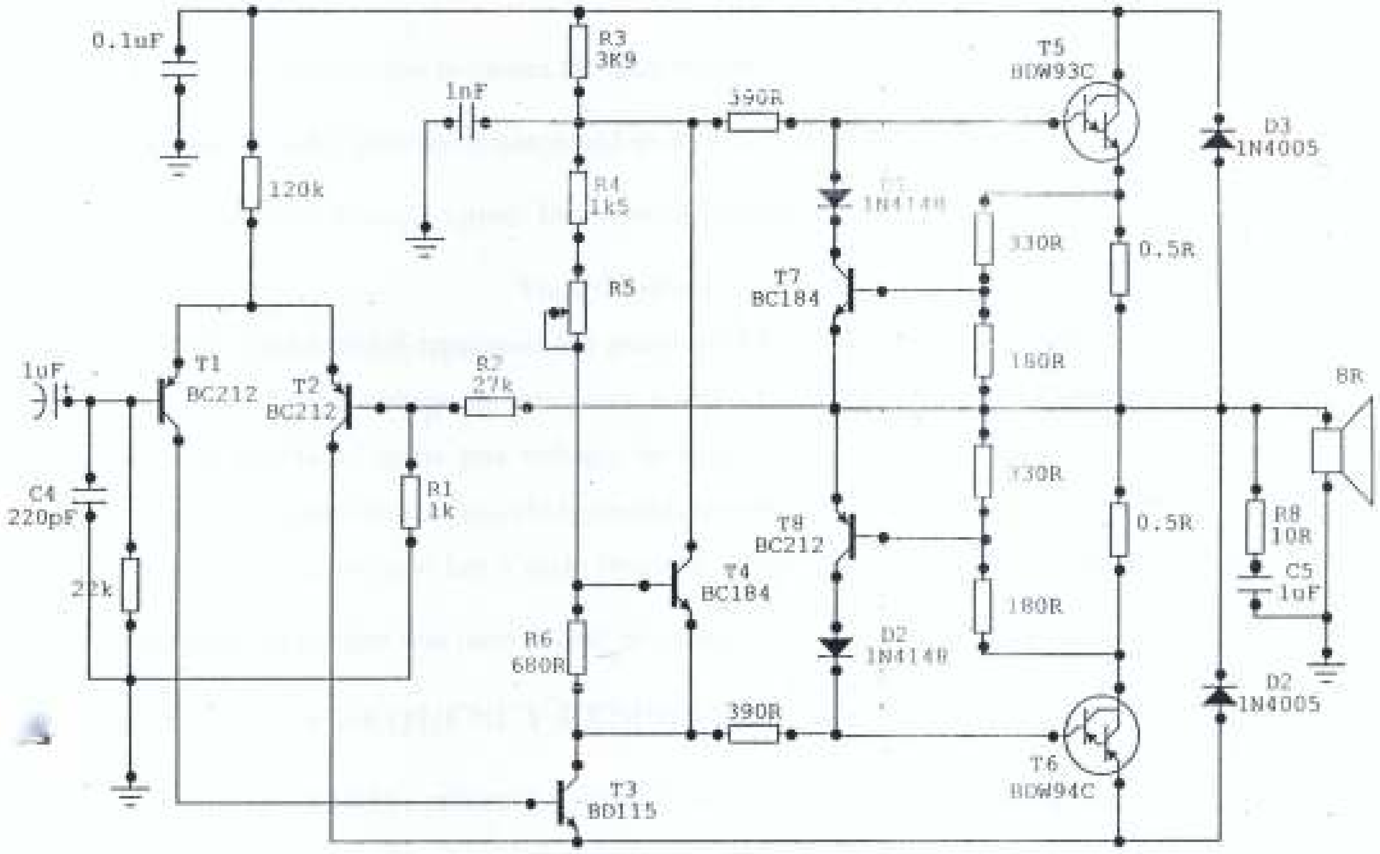


Figure 3.4 50 Watts Audio Amplifier

R_1C_{AV} is much greater than the longest period of the input signal, then I_4 is effectively averaged. The current mirror returns a current I_3 , which equals $A_{vg}.(I_4)$, back to the squarer/divider to complete the implicit RMS computation. Thus: $I_4 = Avg\left[\frac{I_1^2}{I_4}\right] = I_1 rms$

The current mirror also produces the output current, I_{OUT}, which equals $2I_4$.

I_{OUT} can be used directly or converted to a voltage with R_2 and buffered by A_1 to provide a low impedance voltage output. The transfer function of the AD636 thus results:

$$V_{OUT}=2R_2Irms=V_{IN}rms$$

The dB output which represents the pressure of the input signal is derived from the emitter of Q_3, since the voltage at this point is proportional to $-\log V_{IN}$. Emitter follower, Q_5, buffers and level shifts this voltage, so that the dB output is zero when the externally supplied emitter current (I_{REF}) to Q_5 approximates I_3.

This dB output voltage has a scale factor of -3mV per every decibel of input voltage. It is this scale factor that was used to give the sound pressure meter a linear scale.

3.5 FREQUENCY RESPONSE

The AD636 utilizes a logarithmic circuit in performing the implicit rms computation. As with any log circuit, the bandwidth is proportional to the signal level. The solid lines in figure 3.7 represent the frequency response of the AD636 at input levels from 1 millivolt to 1-volt rms. The dashed lines indicate the upper frequency limits for 1%, 10%, and ±3 dB of reading additional error. For example, note that a 1 volt rms signal will produce less than 1% of reading additional error up to 220 kHz. A 10 millivolt signal can be measured with 1% of reading additional error (100 mV) up to 14 kHz.

3.6 CIRCUIT DESCRIPTION OF THE SOUND PRESSURE METER

The circuit of figure3.8 is straightforward where all possible measures have been taken in the design to obtain good linearity and accuracy. The crystal microphone receives radiated signals from the transmitter. The microphone has an inbuilt FET, which is used as

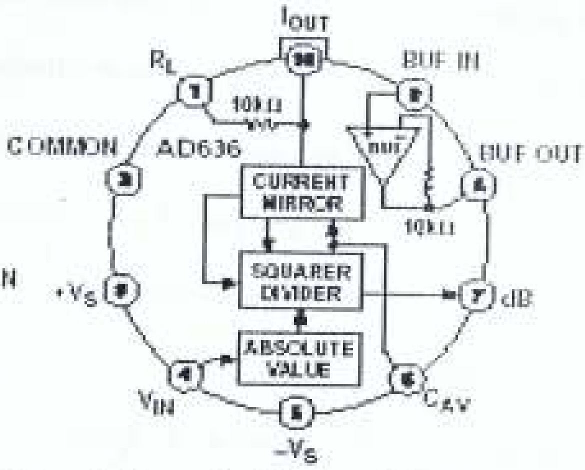


Figure 3.5 Schematic Circuit of AD636

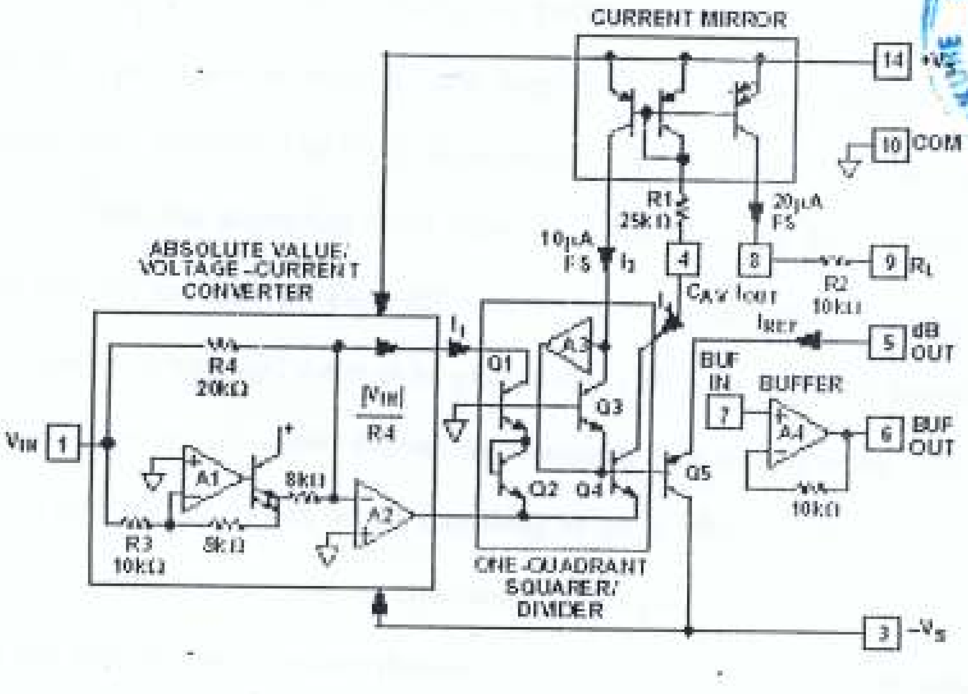


Figure 3.6 The Internal Circuitry of AD636

an amplifier: R_2 and R_3 serve as drain resistors. The network R_2-C_2 provides frequency compensation at low frequencies while R_1-C_1 combination decouples the power supply line to the microphone. The amplification provided by IC_1 and IC_2 is determined by the position of switch sw1. In position 120dB, R_6 is short-circuited. IC_1 then functions as a voltage follower and the gain A_2 produced by IC_2 is given by;

$$A_2=R_9/(R_7+R_8)=1$$

because the value of R_7 plus R_8 is equal to that of R_9. This means that the combined amplification of IC_1 and IC_2 is equal to 1. In position 100dB, the gain A_1 produced by IC_1 is given by,

$$A_1=(R_5+R_6)/R_5=10$$

Since the amplification of IC_2 remains at 1, the combined amplification is 10.

For the switch to be in position of 80dB, the amplification of IC_1 remains at 10. Since R_7 is now short circuited, IC_2 also amplifies by 10. The total amplification becomes, therefore, 100. The output of IC_2 is taken directly into the input of the true RMS converter, which converts the alternating input signal to a direct current. The IC also provides an output that delivers 3mV per every dB voltage variation at the input. This makes it possible for a standard moving coil meter to be given a linear dB scale. A buffer amplifier, IC4 that drives the moving coil meter, follows the converter. The amplification provided by IC_4 makes it possible for meters with a sensitivity of 30 to 100μA to be used. The full scale current is matched to the moving coil meter by the potentiometer P_2 .The value of this preset has been chosen to ensure that the meter has a range of 30dB full-scale deflection and that the ranges overlap one another by 10dB.(Tech/Elector Electronics,1995).

The sound pressure meter is powered with two rechargeable batteries of 6volts, 4Ah each were used.

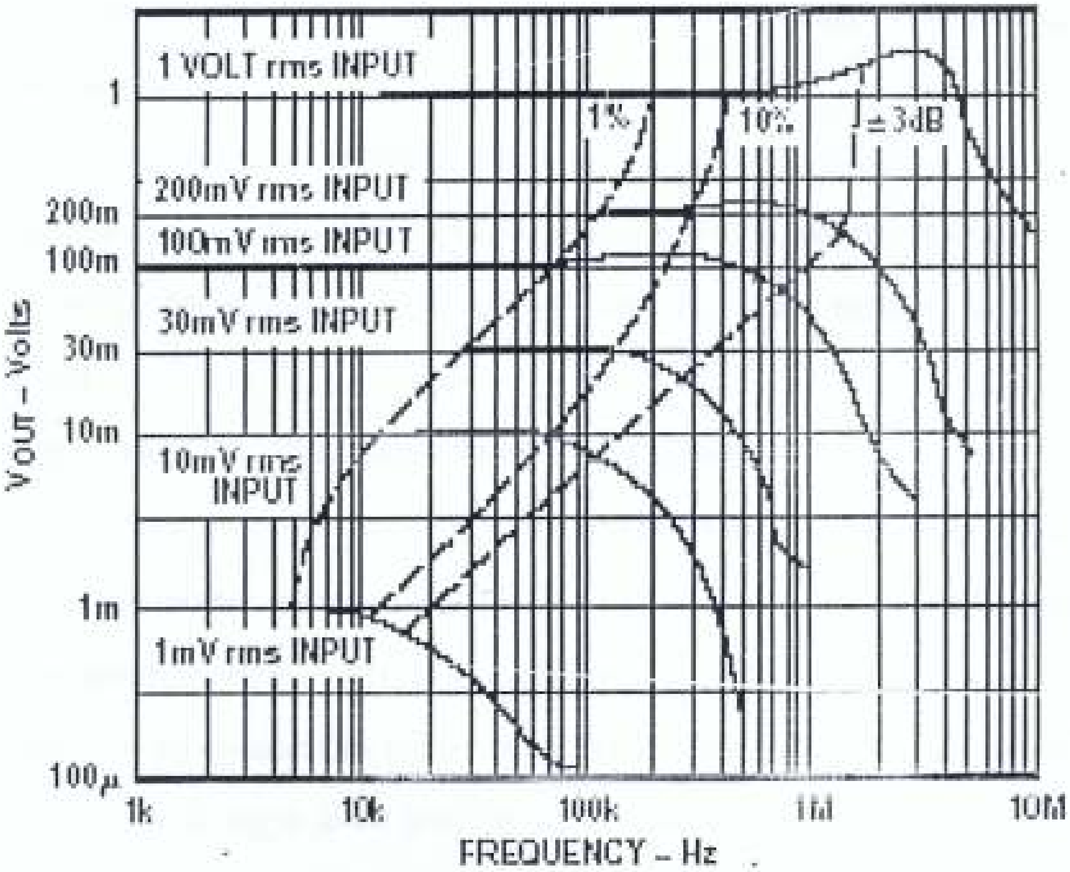


Figure 3.7 **AD636 Frequency Response**

3.7 CALIBRATION

The equipment was calibrated by removing the microphone and then connecting an alternating signal at a frequency of about 1kHz to the input. At this stage, the range switch was set to 120dB.The input signal was the set to 8mV$_{rms}$ measured across R$_4$ with a digital voltmeter. Potentiometer P$_1$ was then adjusted so that the pointer of the moving coil meter is exactly at zero.

For full-scale deflection, the input signal was now adjusted to 253mV$_{rms}$ and potentiometer P$_2$ adjusted so that the meter reads 120dB at full-scale deflection.

When the sound pressure meter has been calibrated it was found to be accurate within ±2dB.

3.8 TESTING THE EQUIPMENT FOR CONSISTENCY

After construction it was desirable to test the equipment to evaluate the performance of the constructed sound pressure meter because it forms a vital part of this research. This was done by feeding a 1kHz sine wave to the auxiliary line input of a digital sound power system (Sony compact Disc Player, Model MXVR456 built in 1997) having 5-band graphic equalizer with digital sound level display in decibel. The graphic equalizer of the sound system was set to activate only the 1kHz tone with the volume set to 65dB while other frequencies were disabled to ensure that the system worked within the midrange of frequencies. The microphone was directly in front of the loud speaker to avoid the effect of atmospheric wind. The readings of the meter were then recorded every minute for consecutive 10 minutes. The variations noted were due to little effects of the wind movements. The same test was performed at four dB levels and the results are given in table 3.1 to 3.4.The reading of the sound system was taken as the standard at a room temperature of 28^{0}C.

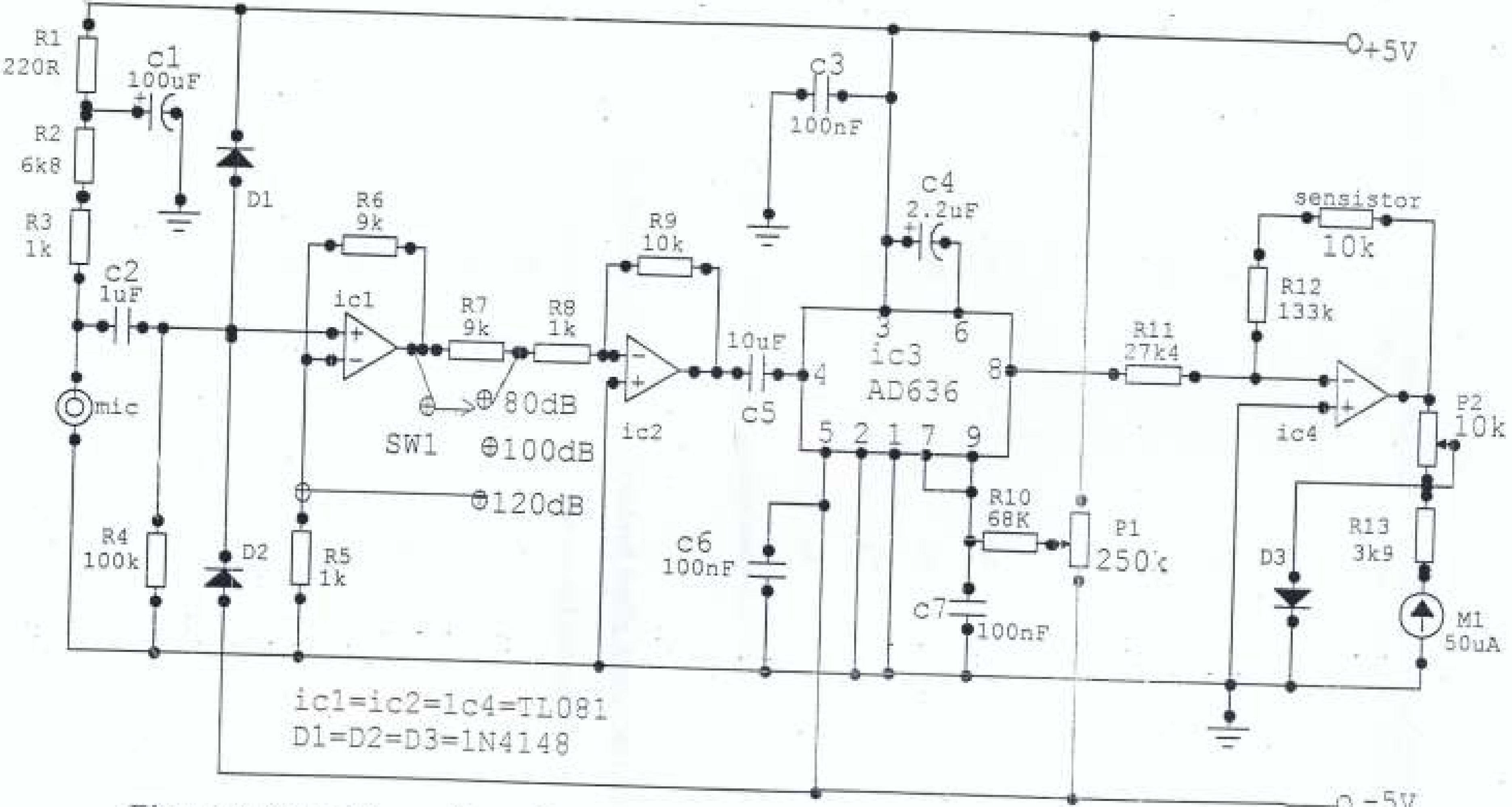


Figure 3.8 The Linear Sound Presssure Meter

Table 3.1

At 1kHz sine wave at the input of a standard audio amplifier (Sony model) at 65dB

	Standard (dB)	Constructed meter (dB)
1	65	66
2	65	66
3	65	65
4	65	65.5
5	65	64
6	65	67
7	65	68
8	65	66
9	65	67
10	65	66

The mean for the constructed sound pressure meter is 65±1

Table 3.2

At 1kHz Sine Wave at the Input of a Standard Audio Amp. (Sony model) at 82dB

	Standard (dB)	Constructed meter (dB)
1	82	80
2	82	81
3	82	82
4	82	82
5	82	81
6	82	81.5
7	82	80
8	82	83
9	82	82.2
10	82	81

The mean reading for the constructed sound pressure meter is 82±1.2

Table 3.3

At 1kHz Sine Wave at the Input of a Standard Audio Amp. (Sony model) at 100dB

	Standard dB	Constructed Meter dB
1	100	99
2	100	99.5
3	100	100
4	100	100
5	100	99
6	100	99

The mean for the sound pressure meter at this level is 100±0.5.

Table3.4

At 1kHz Sine Wave at the Input of a Standard Audio Amp. (Sony model) at 115dB

	Standard dB	Constructed Meter dB
1	115	113.5
2	115	113
3	115	114
4	115	115
5	115	115
6	115	114.5

The mean for the sound pressure meter at this level is 115±0.5.These results show that the sound pressure meter is consistent for the same reading ignoring the wind effect.

3.9 SETTING THE EQUIPMENT

After the parts have been constructed and then coupled together, the experiment was carried out in a big pipe opened at one end to minimize the effect of atmospheric wind on the sound waves being transmitted. Long metal sheets were covered with sound absorbing materials and then folded and joined together to form a big long pipe of about 7 metres. The sound absorbing materials prevent reflections from sides of the pipe that might shoot-up the microphone readings.

The transmitting transducer was used to block one end of the pipe. Holes of 1 meter apart were made on the sides of the pipe where microphones will be inserted to take readings at intervals of 1 metre away from the transmitter. This arrangement was set up for some months to evaluate the equipment's performance and the data collected. To take readings the variable oscillator was adjusted to a particular frequency and the audio amplifier's output adjusted for a very loud sound. Then, pressures at interval of 1 metre away were recorded. Temperatures at any particular reading were also recorded. Graphs of pressures (in dB) were plotted against the distance for different frequencies. Also the graphs of computed attenuations (in dB) were plotted against the distances at different frequencies. The results obtained from the sound pressure meter are given in tables 3.5 to 3.8 at constant frequencies. The corresponding graphs are shown in figures 3.10 to 3.14.

Table 3.5 Readings at 30^{0}C.

	200Hz	500Hz	1000Hz	3000Hz	5000Hz	8000Hz	10000Hz	15000Hz
1	78	91.5	92	91	96	73	92.5	77
2	73	90	87	76	86.5	73	81	70
3	68	83.5	80	67	76	68.5	73	65
4	65	78.5	76	56	68.5	62	65	58
5	60	73.5	71	52	59	55	57.5	55
6	54	71	72	51.5	52	54	53	52

Table 3.6 Readings at 22^{0}C.

	100Hz	200Hz	500Hz	1000Hz	5000Hz	10000Hz	15000Hz
1	75	95.7	91	91	73	77.8	94
2	65.5	92	87.7	84	65.5	72	82
3	56	93	82.5	78	60	67.5	76
4	53	90	76	67	57	63	69
5	52	85	72.5	57.5	52.5	61	63.5
6	52	83	71	52	51	60	53

Table 3.7 Readings at 27^{0}C.

	200Hz	500Hz	1000Hz	3000Hz	5000Hz	10000Hz	15000Hz
1	95	100.6	98	80	90	89	73
2	87	90.5	81.6	70.9	73.5	73.5	69
3	80	81	73	65	65	66.5	62
4	71	72	62	59.8	55.5	56	55
5	70	71	55	58.3	50.5	52	51
6	68	71	51	53	51	52	51

Table 3.8 Readings at 24^{0}C.

	200Hz	500Hz	1000Hz	3000Hz	5000Hz	8000Hz	10000Hz	15000Hz
1	91	95	91	86	93	76	78	79
2	91	95	81.5	74	92	76	78	74
3	86	88.5	76	66	83	72	73	65
4	82	78	68.5	59	75	60	66	60
5	78.5	70	59	53	72	55	56	52.5
6	71	68	52	53	65	54	55	48

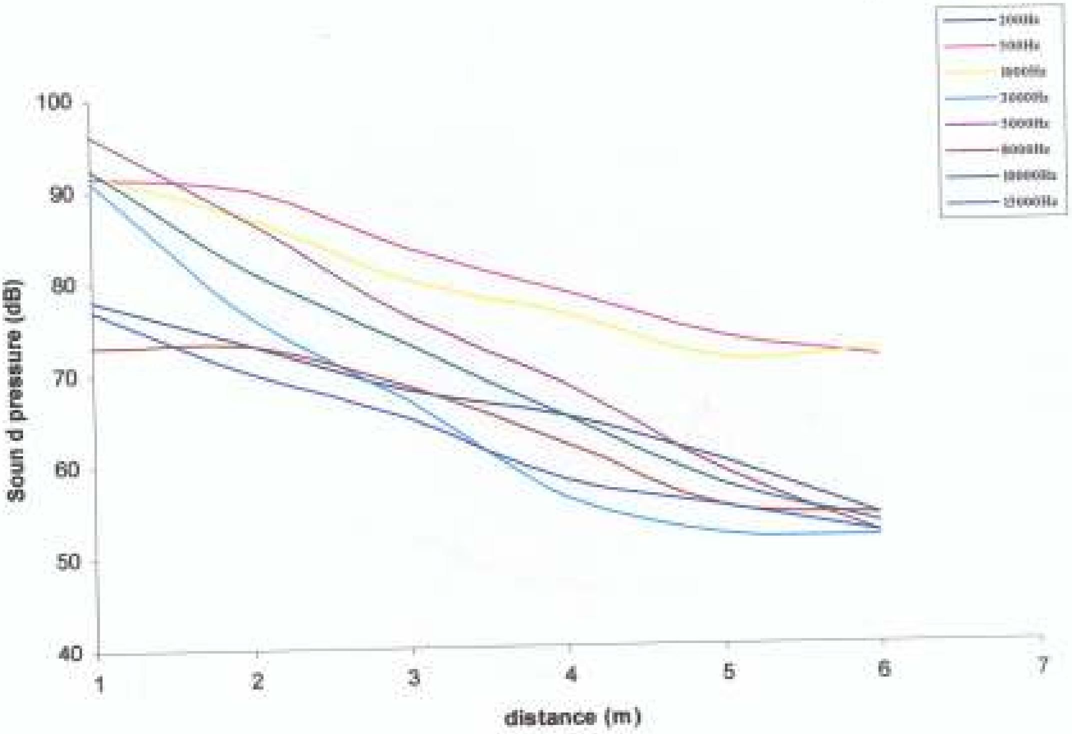


Figure 3.10 Graph of Sound pressure versus distance from Table 3.5

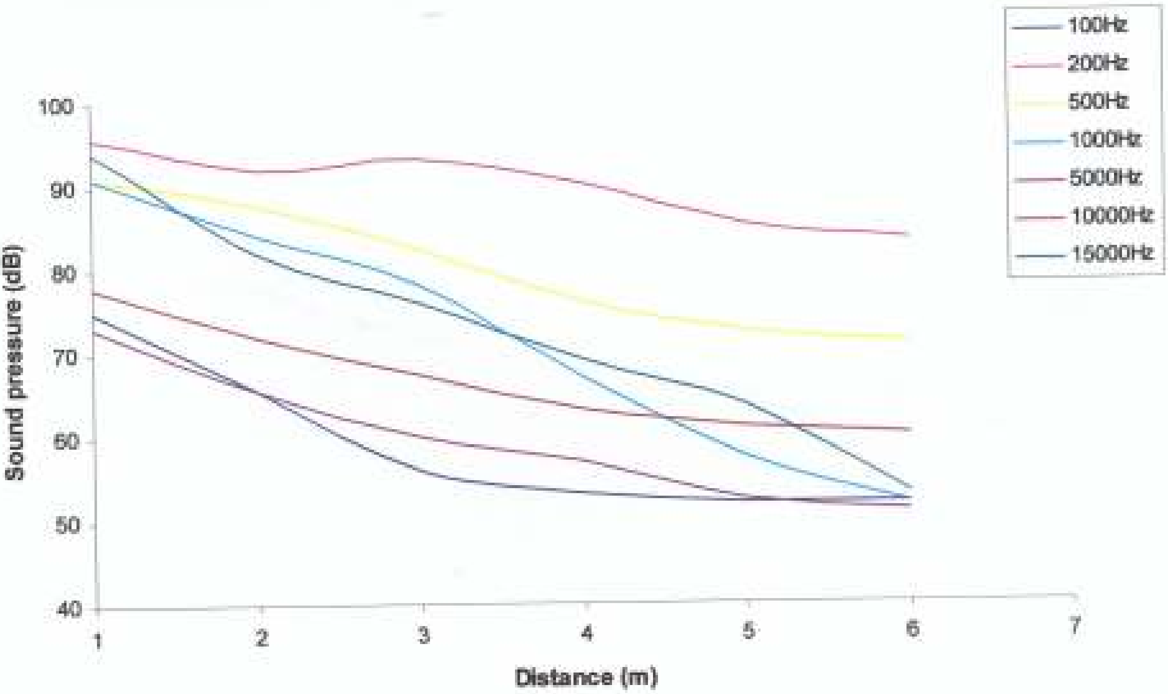


Figure 3.11 Soud Pressure(dB) versus Distance(m) from table 3.6

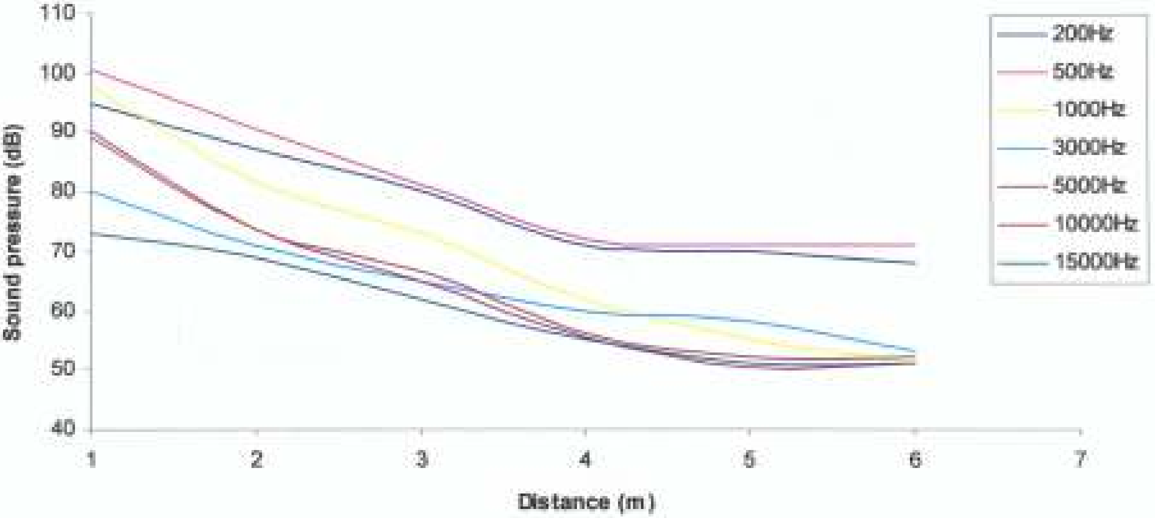


Figure 3.12 Graph of Sound pressure(dB) Versus Distance (m) Table 3.7

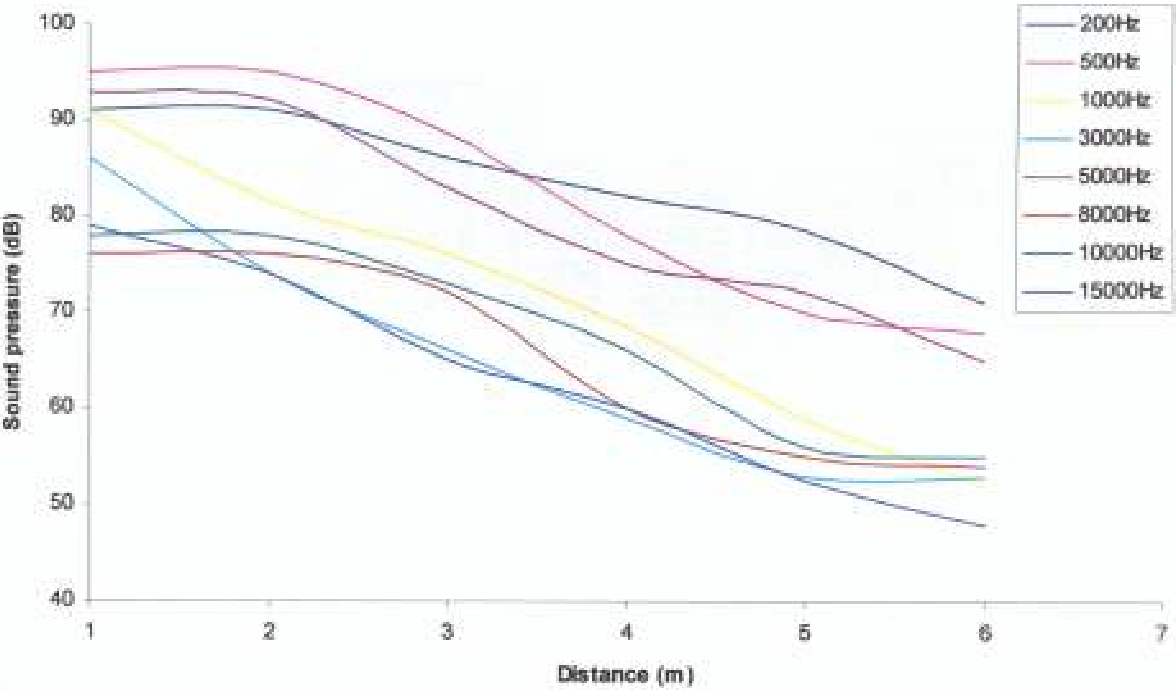


Figure 3.13 Graph of Sound pressure(dB) versus Distance(m) from Table 3.8

Table 3.9 Log of sound pressures computed from table 3.5

	200Hz	500Hz	1000Hz	3000Hz	5000Hz	8000Hz	10000Hz	15000Hz
1	1.892095	1.961421	1.963788	1.959041	1.982271	1.863323	1.966142	1.886491
2	1.863323	1.954243	1.939519	1.880814	1.937016	1.863323	1.908485	1.845098
3	1.832509	1.921686	1.90309	1.826075	1.880814	1.835691	1.863323	1.812913
4	1.812913	1.89487	1.880814	1.748188	1.835691	1.792392	1.812913	1.763428
5	1.778151	1.866287	1.851258	1.716003	1.770852	1.740363	1.759668	1.740363
6	1.732394	1.851258	1.857332	1.711807	1.716003	1.732394	1.724276	1.716003

Table 3.10 Log of sound pressures computed from table 3.6

	100Hz	200Hz	500Hz	1000Hz	5000Hz	10000Hz	15000Hz
1	1.875061	1.980912	1.959041	1.959041	1.863323	1.89098	1.973128
2	1.816241	1.963788	1.943	1.924279	1.816241	1.857332	1.913814
3	1.748188	1.968483	1.916454	1.892095	1.778151	1.829304	1.880814
4	1.724276	1.954243	1.880814	1.826075	1.755875	1.799341	1.838849
5	1.716003	1.929419	1.860338	1.759668	1.720159	1.78533	1.802774
6	1.716003	1.919078	1.851258	1.716003	1.70757	1.778151	1.724276

Table 3.11 Log of sound pressures computed from table 3.7

	200Hz	500Hz	1000Hz	3000Hz	5000Hz	10000Hz	15000Hz
1	1.977724	2.002598	1.991226	1.90309	1.954243	1.94939	1.863323
2	1.939519	1.956649	1.91169	1.850646	1.866287	1.866287	1.838849
3	1.90309	1.908485	1.863323	1.812913	1.812913	1.822822	1.792392
4	1.851258	1.857332	1.792392	1.776701	1.744293	1.748188	1.740363
5	1.845098	1.851258	1.740363	1.765669	1.703291	1.716003	1.70757
6	1.832509	1.851258	1.70757	1.724276	1.70757	1.716003	1.70757

Table 3.12 Log of sound pressures computed from table 3.8

	200Hz	500Hz	1000Hz	3000Hz	5000Hz	8000Hz	10000Hz	15000Hz
1	1.959041	1.977724	1.959041	1.934498	1.968483	1.880814	1.892095	1.897627
2	1.959041	1.977724	1.911158	1.869232	1.963788	1.880814	1.892095	1.869232
3	1.934498	1.946943	1.880814	1.819544	1.919078	1.857332	1.863323	1.812913
4	1.913814	1.892095	1.835691	1.770852	1.875061	1.778151	1.819544	1.778151
5	1.89487	1.845098	1.770852	1.724276	1.857332	1.740363	1.748188	1.720159
6	1.851258	1.832509	1.716003	1.724276	1.812913	1.732394	1.740363	1.681241

The results obtained for the log of pressures are given in tables 3.9 to 3.12 at different distances.

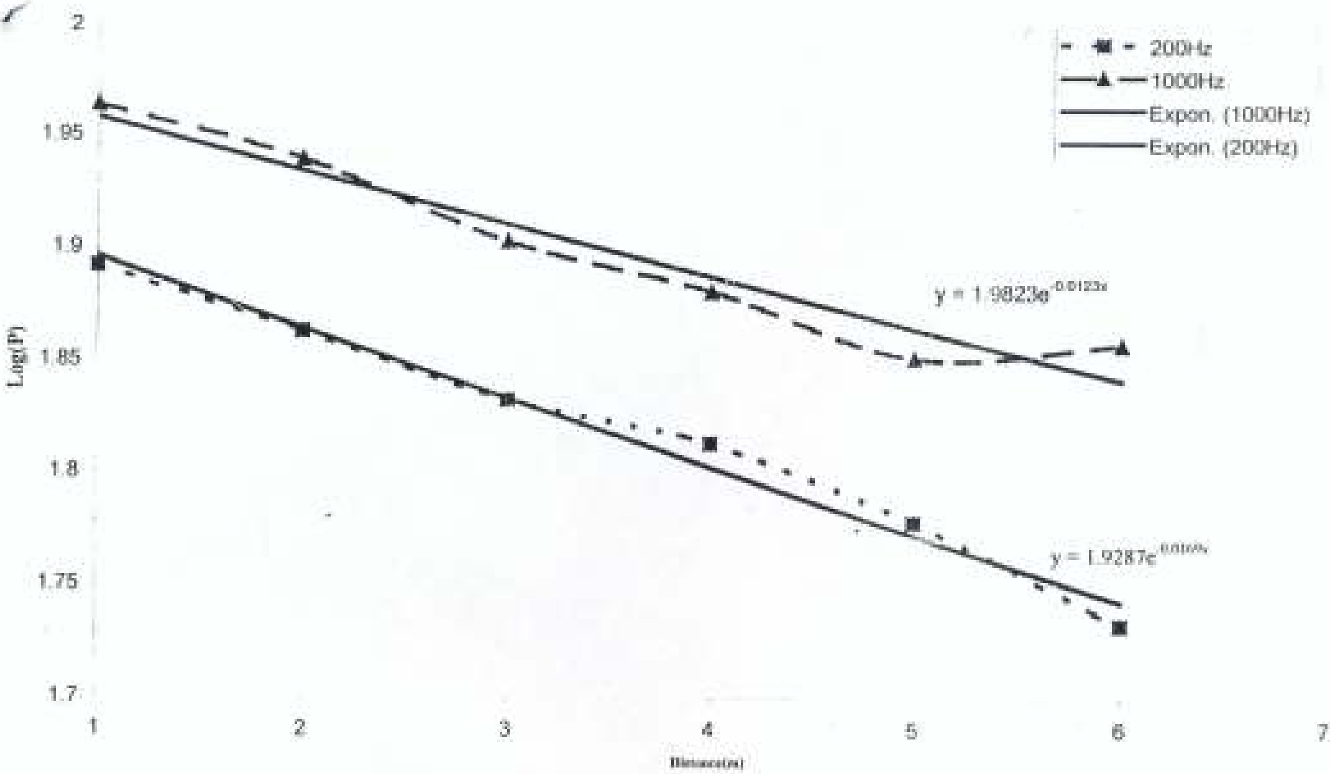


Figure 3.13 Log(P) versus Distance(m) from Table 3.9

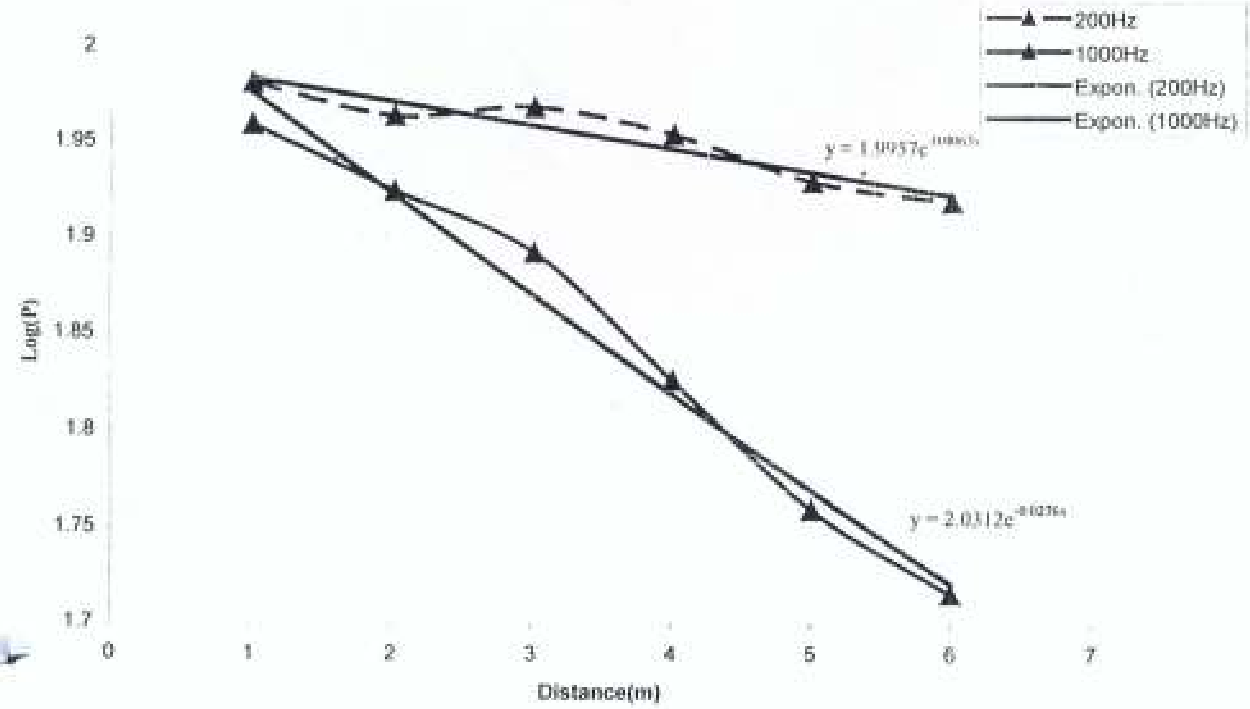


Figure 3.14 Log(P) versus Distance(m) from Table 3.10

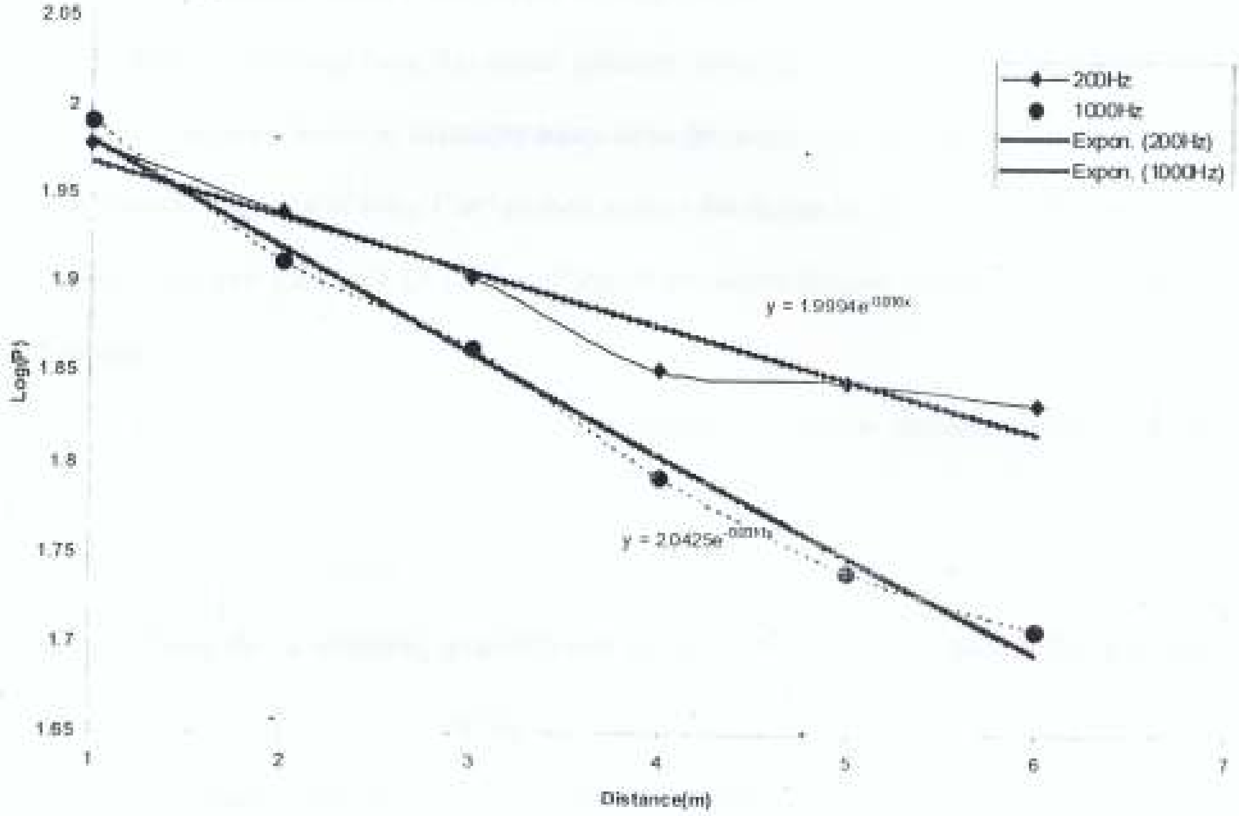


Figure 3.16 Log(P) versus Distance(m) from Table 3.11

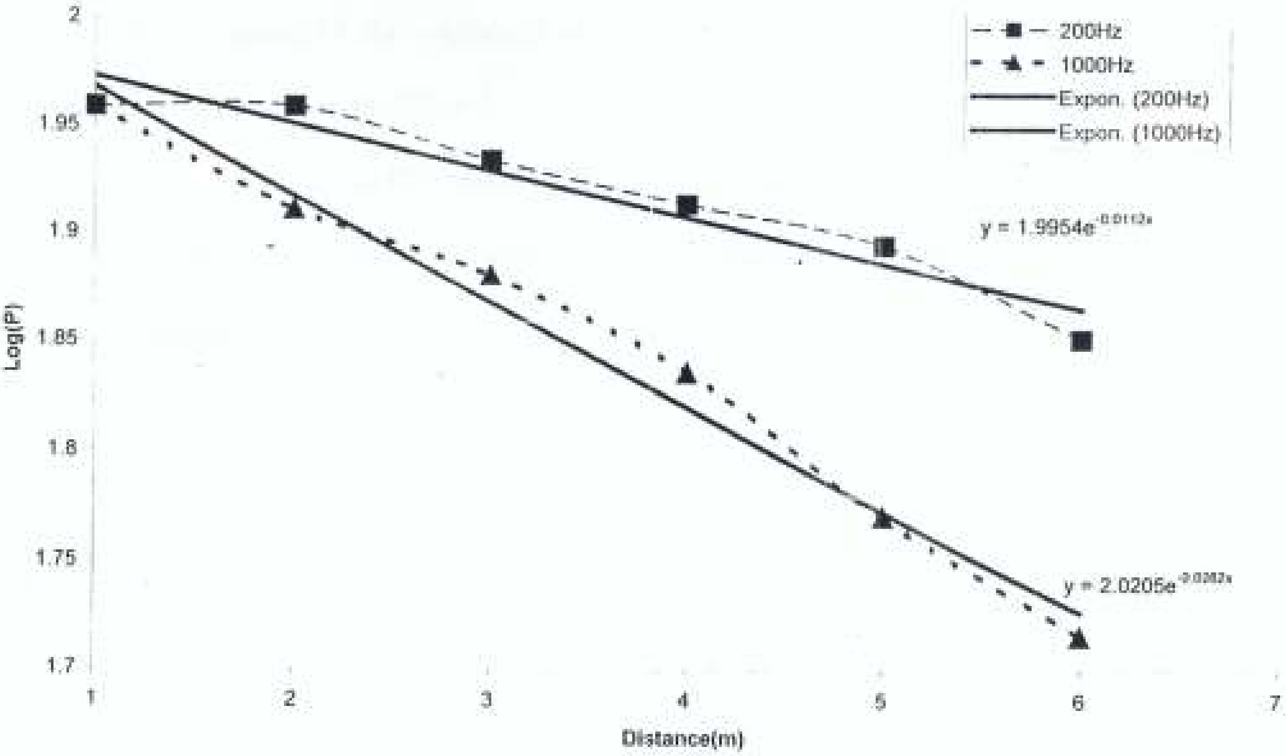


Figure 3.16 Log(P) versus Distance(m) from Table 3.12

3.10 Estimation of the Coefficients of Attenuation

Results obtained from the sound pressure meter given in tables 3.5 to 3.8 are only the sound pressure levels at distances away from the sound source. To deduce coefficients of attenuation, graphs of $\log_{10} P$ are plotted against the distances for different frequencies as shown in figures 3.14 to 3.17.Tables of $\log_{10}P$ are given in tables 3.9 to 3.12 at different distances.

From the graphs of $\log_{10}P$ versus the distance in metres, the relationship is of the form:

$$Y=Ae^{-\alpha x}$$

Where A= a constant, α=coefficient of attenuation in dBm^{-1} and x the distances away from the sound source in metres.

The negative sign implies that as one moves away from the source there is reduction in intensity levels. For clarity purpose, some frequencies are only selected. The selected frequencies are 200Hz and 1000Hz from each table, graphs of which are plotted in figures 3.14 to 3.17. From the graphs, the coefficients of attenuation are given directly.

From figure3.17, the coefficient of attenuation for 200Hz is given as $-0.0112dBm^{-1}$ and for 1000Hz it is $-0.0276dBm^{-1}$.

Also, values of coefficients at different temperatures versus frequencies are plotted in figure 3.18, where effects of frequency and temperature on coefficients of attenuation can be compared.

Table3.13 of coefficients of attenuations of some selected frequencies at different temperatures.

f(Hz)	30^{0} C	22^{0} C	27^{0} C	24^{0} C
200	-0.0169	-0.0063	-0.016	-0.0112
500	-0.0126	-0.0123	-0.0167	-0.0177
1000	-0.0123	-0.0276	-0.0311	-0.0262
5000	-0.029	-0.0175	-0.0282	-0.0172
10000	-0.0265	-0.0126	-0.0266	-0.0194

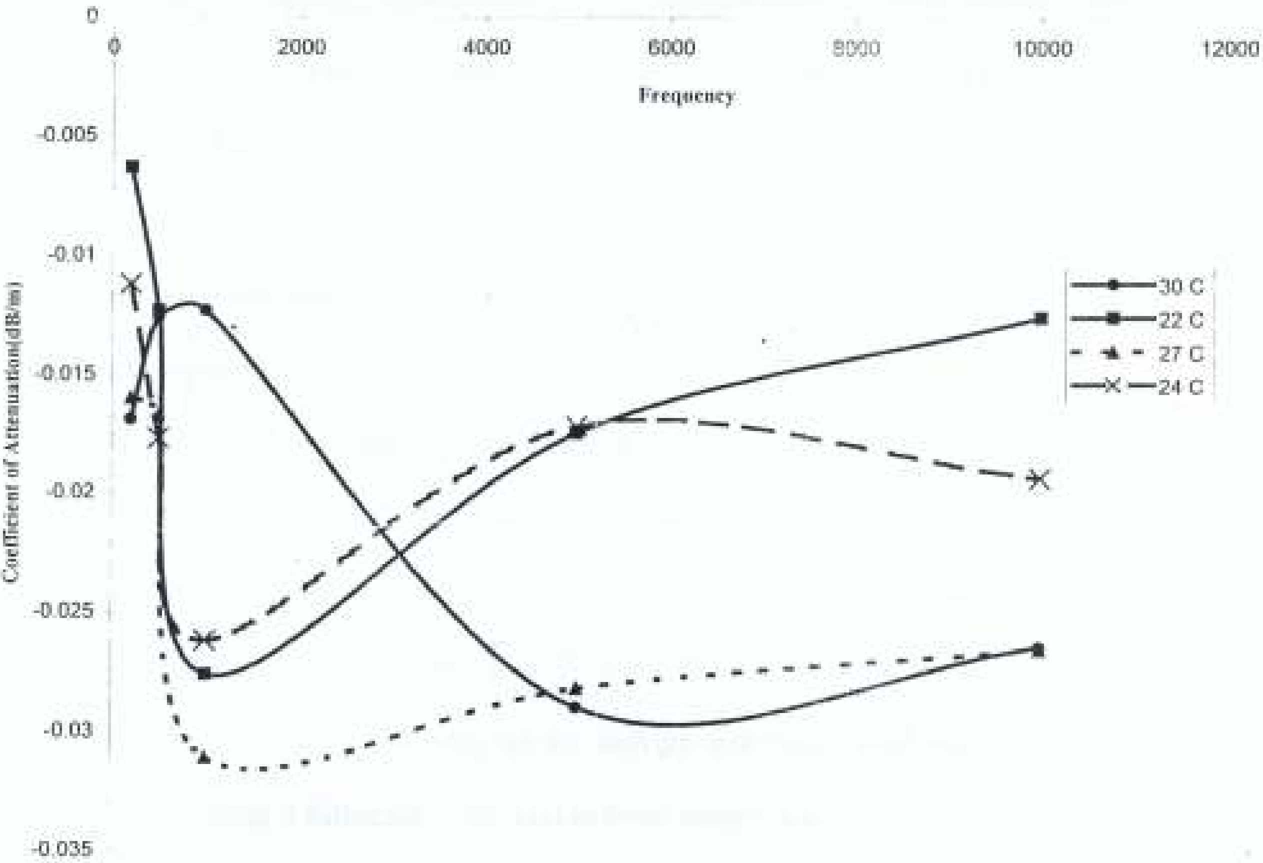


Figure 3.17 Coefficient of Attenuation at different temperatures Versus Frequency

CHAPTER 4.

4.0 ERROR ESTIMATION, DISCUSSION AND CONCLUSION.

4.1 Dealing With Errors In The System.

No system is absolutely free from errors. To make measurements precise, there are some errors that can be accounted for while some cannot be accounted for but they can be ignored if their error contributions are negligible or insignificant.

Errors that occur from input offset voltage of operational amplifiers used as preamp in the sound pressure meter and the op-amp that drives the moving coil meter can be accounted for. Due to offset voltage that occurs at differential input of the op-amp TL081, both internal and external nulling were applied that reduced the offset voltage from13mV to 120μV (that is 0.12mV). This value, that is 120μV is very small since it has little or no effect on the amplified signal output. The system performed very well at input signal from 0.1mV to well over 200mV.

Also, errors from calibration of the moving coil meter used was drastically reduced by choosing a full-scale deflection in three ranges that overlap one another by 10dB.

Errors due to the effect of wind could not be totally eliminated. The lagged pipe that was used to carry out the experiment was still opened at one end and therefore fluctuations on the moving coil meter were still noticeable due to little air currents coming into the pipe. Another error that could not be accounted for is the effect of humidity and temperature variations on the components used, which internally affects signal processing. Moreover, this issue of temperature variations were adequately taken care of in the transmitting amplifier by the use of a very large heat sink because its own variation was mainly from the power transistors used.

Part of the errors in the determination of sound pressure levels can be accounted for by straightforward calculations. For the determination of errors that may arise from roughly assuming the measured values of sound pressures, the standard deviation, σ of the two tests carried out in chapter three was estimated to be 1.3 and 1.2 respectively which gave us an idea that the uncertainty of the estimate was fairly low. These gave us a rough estimate of how many readings we ought to take at each test to give us a reasonable precision in the mean value of sound pressure level.

From statistical table, we would expect 99% of each mean to be within three times these values of the true mean. Therefore, for better results, two to three readings at each test level were taken, which ensured that the mean values of sound pressure level were nearly all within about 1% of the true values.

4.2 DISCUSSION.

All the data and analysis so far given are meant to test and to evaluate the performance of the sound equipment. To be able to reach a reasonable conclusion on the applications of the equipment it has to be used extensively for a very long time; taken into consideration the atmospheric effects.

Atmospheric effects refer to: (1) atmospheric absorption, that is, the sound attenuation by air and water vapor; (2) atmospheric refraction, that is, the sound refraction caused by temperature and wind gradients. Atmospheric absorption is a function of the frequency of the sound, the temperature, the humidity and the atmospheric pressure between the source and the receiver. Over distances greater than 30 meter, the attenuation due to atmospheric absorption can substantially reduce sound levels, especially at high frequencies (above 3000Hz). Atmospheric refraction is the bending of sound waves due to wind and temperature gradients. Near-ground wind effects are, typically, the most substantial contributor to sound refraction. Upwind conditions tend to refract sound waves away the ground resulting in a decrease in sound levels at receiver. Conversely, downwind conditions tend to refract sound waves towards the ground resulting in an increase in sound levels at a receiver. To minimize the effect of wind, measurements should not be performed in conditions where strong winds exist in the direction of propagation.

Temperature effects can also contribute to sound refraction. During daytime weather conditions, when the air is warmer closer to the ground (temperature decreases with height), sound waves tend to refract upward away from the ground (temperature lapse). This may result in a decrease in sound levels at a receiver. Conversely when the air close to the ground cools during nighttime weather conditions (temperature increases with height), sound waves tend to refract downward towards the ground (temperature inversion). This may result in an increase in sound levels at a receiver. Generally, refraction effects due to temperature do not exert a substantial influence on sound levels at few meters away from the source.

Nevertheless, the constructed equipment has so many applications, which can also be deduced from the plotted graphs.

Knowing the frequency of a source of noise, the dominant regions of the noise radiation can be located or predicted yielding substantial economic and environmental benefits through effective noise control.

In work places or industrial noise control, the minimum distance between a worker and the machinery can be determined straight away knowing the pressure and frequency the machine radiates including application to building acoustics, vehicular and engine technology.

The sound pressure meter can be used to study the transmission loss through partitions or materials. Also, the pressure meter is a very useful equipment to community health workers in assessing the environmental noise index.

4.3 CONCLUSION

Graphs of attenuations versus the distances of some selected frequencies, which have been linearized, are shown in figures 3.14 to 3.17 where coefficients of attenuations are deducted. From these graphs, it is clearly seen that;

(i) the sound pressure is attenuated rapidly as one moves away from the source.

(ii) there are different coefficients of attenuations (for same power of transmission) at different frequencies. That is, attenuation strongly depends on frequency.

(iii) since the variations of atmospheric temperature affects the velocity of sound in the atmosphere , it is also noted in the graphs that attenuation depends on temperature.

4.4 Recommendations

All attempts have been made at this level to make this research work a perfect one, however within the focus of this project, I wish to make the following recommendations for future research in the area of acoustics.

(i) since the audio frequency range extends well above 20kHz (over 300kHz), future construction should be made to go beyond the audible range.

(ii) acoustic transducers (both the microphone and the loudspeaker) that have higher bandwidths should be used.

(iii) taking readings over a longer time.

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