DESIGN OF AN AUDITORIUM AUDIO ADDRESS SYSTEM

CASE STUDY OF ENGINEERING LECTURE THEATRE GIDAN KWANU

BY

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DEDICATION

I dedicate this project to Almighty Allah. who has been my source of inspiration and guidance. Also to my parents Sheikh Alhaji Magaji Hamza and Hajiya Hajarat Gogo Hamza.

Declaration

I, ABDULLAHI UMAR, hereby declare that this work was done by me and never been presented elsewhere for the award of a degree, also hereby relinquish the copy right to the Federal University of Technology, Minna.

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ABSTRACT

This design illustrates how to determine appropriate amplifier and loudspeaker units in the volume of an auditorium, ensuring that everybody receives the same comfortable sound pressure level (80dB) no matter where one is seated. Six loudspeakers were chosen which are in regular array arrangement to ensure uniform distribution of sound. Ten listening points were selected in which distances between the speakers to the points were determined. The power from each speaker to every point depends on the distance between them. The cumulative power from the six speakers gives the average total power in the room. That power is an acoustic power output and the input power which is an electrical power to the loudspeakers depends on the acoustic power and the efficiency of the speaker. The efficiency of the speakers takes care of the system losses. Each loudspeaker was found to be 50Watts and therefore,300Watts of amplifier gives the appropriate wattage to drive the six loudspeakers.

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CHAPTER ONE

1.0 INTRODUCTION

When building is to be erected especially an auditorium, the quality of the listening environment should be given high priority. It's scientific investigation should begin during the preliminary design stage before plans are committed to blue prints. At the same time seating capacity is being decided, acoustic design of the auditorium should begin taking a lead position in determining the layout and shape of the building.

Auditorium is part of a theatre where audience sits and transfer message from one person to another. If that message cannot be heard clearly by every person in the room, the design could be considered a design failure. Acoustics is an engineering science with result that can be predicted through vigorous mathematical computation and investigation.

As a skill, it requires experience and the development of reliable intuition of one of the most complex natural sciences known to modern physics. Very few architects are equipped with the skills to undertake even basic acoustic design, so it is usually overlooked until late in the project, or not addressed at all.

In some cases, the acoustic consultant is asked to look at the completed plans and suggest some remedial modifications. Ideally, an acoustic consultant should be engaged at the same time as the architect, saving redesign and therefore money [1]

One of the most important things an auditorium must do is to provide a place where speech can be clearly understood. This means a good auditorium should have a good intelligibility rating. The set of minimum acoustic requirements that are met by a working auditorium starts with the direct sound from the loudspeaker being loud enough, that means it replicates conversational sound levels. The background noise in the has to be fairly quiet. The hall acoustics should be clearly free from echoes and other types of late reflections. And finally the hall acoustic is not reverberant at all. [4]

The relative sound level reaching the audience may be increased by the use of reflective surfaces, ensuring an optimum RT and reducing background noise as much as possible. However, in large auditoria, there are physical limits to the effectiveness of these techniques.

Under normal circumstances, the human voice is quite audible in room up to $300m^3$ well designed rooms with good reflective reinforcement may increase this limit up to as high as $1500m^3$ for high standard of intelligibility. As a rule of thumb, however, it is generally accepted that some form of active reinforcement will be required for rooms greater than $1700m^3$ or where the direct sound must travel more than 18metres to the listener. This reinforcement must often takes the form electrical amplification, using a microphone and loudspeakers.[4]

However, in this design, engineering lecture theatre at main campus was given as a case study which has the internal dimensions as follows: length—25.75metres, width—14.1meres and the height--8.4metres, which gives a total volume of 3049.83metres. Hence, an amplification is necessary inside the auditorium.

1. To increase the sound level when a source is too weak to be heard
2. To provide additional sound to audiences beyond the intended range of the source
3. To alter the reverberation time and other impression of an auditoria
4. To reduce the relative effects of background noise

Some of these functions come under the umbrella of public address systems as they produce artificial sound in some remote location. Whilst many of the concepts and principle are the same, we are going to concern ourselves with sound amplification systems which reinforce the sound coming from a visible source.

1.1 HISTORICAL BACKGROUND

Since the prehistoric time man in common with his variegated cousins throughout the animal kingdom has relied on his five senses to keep informed about his surrounding, to move around safely, to seek out food and shelter, and warn him of dangers. Of five senses, hearing was a relatively late in arrival in the evolutionary process but has perhaps become the busiest. Our sense of hearing seems always to be at work and it has certainly evolved as a uniquely complex and finally turned mechanisms which are only now beginning to understand properly. It is therefore hardly surprising that senses of hearing and the nature of the sound that surrounds us has interested scientist since the earliest times, as sound come to be used for communication speeches and promoting pleasures (music). The nature of sound and hearing become a necessary study.

The word microphone first appeared in 1827 in Whetstone's description of acoustic device and was later used by Berliner in 1877 and Hughes in Alexandra. Graham is generally credited with the inventing of the first workable microphone and loudspeaker/earpiece

Amplification is the process where the power of a signal is increased without altering its basic information carrying characteristics. Amplification in communication has found itself in virtually all human endeavors from vacuum tubes to transistors and magnet, amplifier, amplify tiny electrical voltage to a level at which they can operate loudspeakers. There is no doubt that without amplification, communication engineering would have been impossible. The earlier pioneers could never envisaged the ways in which their technology could expand and diversify to cover first every field of human interest. Undoubtedly, communications will continue to expand not minding the present cable satellite television, cellular phone, internet affecting more and more areas of our lives.

Communicating could be defined as a process of sending and receiving signals or messages. The earlier means of communication were by smoke, drums, fire e.t.c. The dumb also used certain signs to communicate with their colleaques.

Furthermore, engineers and scientists have developed many means of communication.These include the use of radio links, telephone lines, microwave links, television and public address system etc. The telephone system is a two way communication channel that is fast and reliable.

However, it is not readily available in the remote places in Nigeria of today. The radio link is a free space propagation operating at a narrow band FM-VHF and approximately 150MHz. Radio link has the following advantages; power authority can own it. It has high data rates than the telephone link. However, it has a specrum limitation and more expensive than the telephone.

In the public address system, a very important part is the audio amplifier. The microphone serves as the input transducer while the loudspeaker servers as the output transducers. The input transducer converts the sound energy into electrical signal. The output transducer converts the electrical signal to sound [6]

1.1 AIM AND OBJECTIVES

1. To determine appropriate amplifier and loudspeaker units in the volume of the auditorium
2. To obtain a uniform sound pressure level at every point in the room.
3. To increase the sound level when a source is too weak to be heard
4. To provide additional sound to audiences beyond the intended range of the source
5. To alter the reverberation time and other impression of an auditoria
6. To reduce the relative effects of background noise

1.3 CHALLENGE

This design was born out of need to solve the problems of sound quality in our auditoriums and to come up with an effective means of public address system to complement the existing means of communication in our environment.

1.4 PROJECT LAYOUT

There are five chapters which made up the sections of this project.

CHAPTER ONE : Contains the general introduction and background information

CHAPTER TWO : Carries the description of the design system

CHAPTER THREE : Explains how design was carried out

CHAPTER FOUR : carries the result

CHAPTER FIVE : Carries the conclusion and recommendations for further improvement

CHAPTER TWO

2.0 LITERATURE REVIEW

2.1 Fundamental Concept of Acoustics

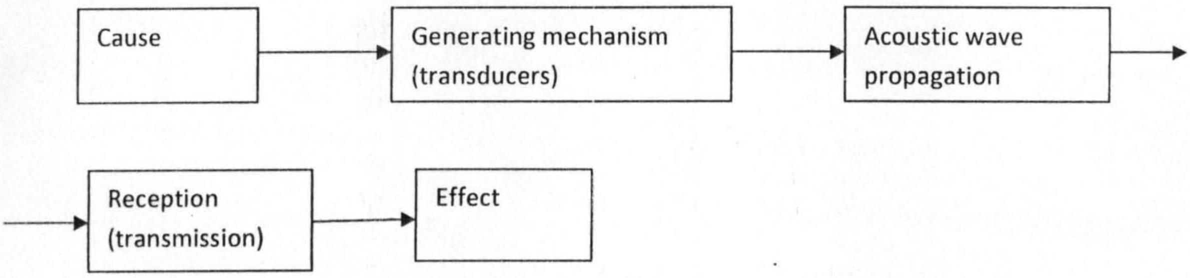


Figure 2.0

The steps shown in the above diagram can be found in any acoustical event or process. There are many kinds of cause, both natural and volitional. There are many kinds of transduction process that convert energy from some other form into acoustical energy producing the acoustic wave.

The wave carries energy throughout the propagation medium. Eventually this energy is transduced again to other forms, in ways that again may be natural or volitional. The final effect may be purely natural or volitional also.[1]

The central stage in the acoustical process is the wave propagation. This falls within the domain of physical acoustics, in fluids, sound propagates primarily as a pressure wave. In solids, mechanical waves can take many forms including longitudinal waves, transverse and surface waves.

Acoustics looks first at the pressure levels and frequencies in the sound wave.Transduction processes are also of importance.

2.1.1 **Wave Propagation**

2.1.2 **Pressure levels**

In fluids such as air and water, sound wave propagates as disturbances in the ambient pressure level. While this disturbance is usually small, it is still noticeable to the human ear. The smallest sound that a person can hear known as threshold of hearing, is nine orders of magnitude smaller than the ambient pressure. The loudness of this disturbances is called the sound pressure level, and is measured on a logarithmic scale in decibels.

Mathematically, sound pressure level is defined as $spl = 20\times\log(P/P_{ref})$ [3]

Where P_{ref} is the threshold of hearing and P is the change in ambient pressure.

2.1.3 **Frequency**

Frequency is what physicists and acoustic engineers used to describe sound pressure Levels. The entire levels can be divided in to three sections: audio, ultrasonic, and Infrasonic. The audio range falls between 20Hz and 20,000Hz. This range is important because its frequencies can be detected by human hear. This range a number of applications including speech communication and music. The ultrasonic range refers the very high frequencies: 20,000Hz and higher. This range has shorter wavelengths which allows better resolution in imaging technologies. Medical applications such as ultrasonography and elastrography rely on the ultrasonic frequency range.

The lowest frequencies are known as the infrasonic range. The frequencies can be used to study geological phenomenon such as earthquakes. [4]

2.1.4 Transducers in acoustics

A transducer is just a device for converting one form of energy into another. This means in acoustics context, converting sound energy in to electrical energy (or vice versa). Acoustics transducers include loudspeakers, microphones, hydrophones, sonar projectors, and ultrasonic imaging equipment. [2]

2.2.0 Block diagram of an auditorium audio address system

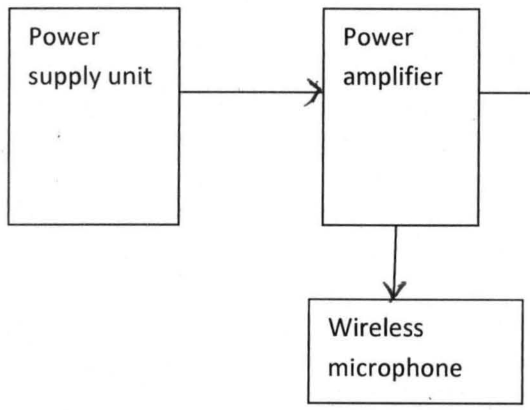


Figure 2.1

Block diagram of auditorium audio address system

2.2.1 power supply unit

The power supply provides the power amplifier with an electrical energy input of 240V AC or 24V DC (2×12V car Battery) operation. The amplifier should ensure uninterrupted operation in case of power failure. That is, provision for automatic changeover from AC to Battery operation to ensures continuity of program.

2.2.0 **Microphones**

Microphones are transducers which detect sound signals and produce an electrical image of the sound, i.e., they produce a voltage or a current which is proportional to the sound signal. The most common microphones for musical use are dynamic, ribbon, or condenser microphones. Besides the variety of basic mechanisms, microphones can be designed with different directional patterns and different impedances. [6]

2.2.3 **Dynamic Microphones**

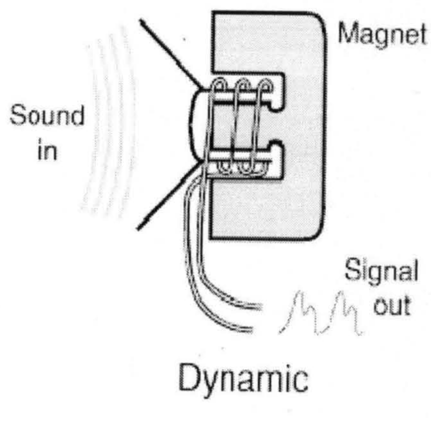


Figure 2.2

Pripriniciple: sound moves the cone and the attached coil of wire moves in the field of a magnet. The generator effect produces a voltage which "images" the sound pressure variation- characterized as a pressure microphone.

Advantages

- Relatively cheap and rugged.

- Can be easily miniaturized.

Disadvantages:

- The uniformity of response to different frequencies does not match that of the ribbon or condenser[6]

2.2.4 **ribbon microphones**

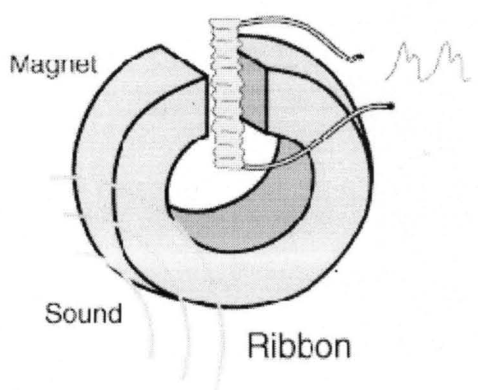


Figure 2.3

Principle: the air movement associated with the sound moves the metallic ribbon in the magnetic field, generating an imaging voltage between the ends of the ribbon which is proportional to the velocity of the ribbon - characterized as a "velocity" microphone.

Advantages:

- Adds "warmth" to the tone by accenting lows when close-miked.

- Can be used to discriminate against distant low frequency noise in its most common gradient form.

Disadvantages:

- Accenting lows sometimes produces "boomy" bass.
- Very susceptible to wind noise. Not suitable for outside use unless very well shielded.[2]

2.2.5 **Condenser Microphones**

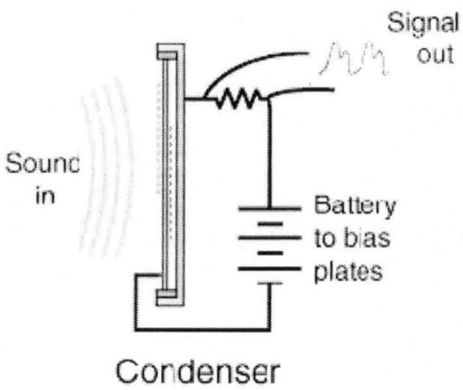


Figure 2.4

Principle: sound pressure changes the spacing between a thin metallic membrane and the stationary back plate. The plates are charged to a total charge

$$Q = CV = \frac{\alpha(\text{Area of plate})(\text{voltage})}{(\text{plate spacing})}$$

where C is the capacitance and V the voltage of the biasing battery.

Advantages:

- Best overall frequency response makes this the microphone of choice for many recording applications.

Disadvantages:

- Expensive
- May pop and crack when close miked
- Requires a battery or external power supply to bias the plates.
- A change in plate spcing will cause a change in charge Q and force a current through resistance R. This current "images" the sound pressure, making this a "pressure" microphone.

2.3.0` **Directional Patterns of Microphones**

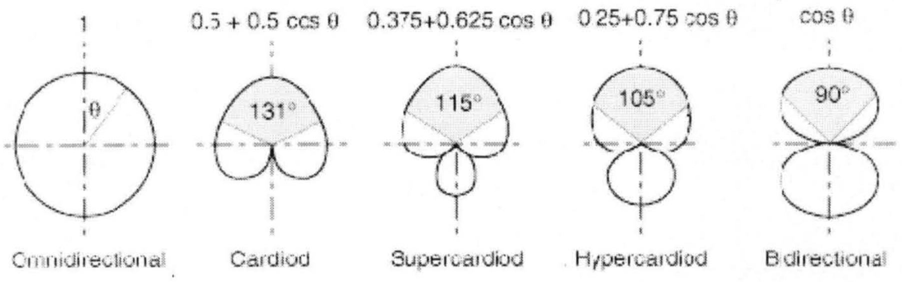


Figure 2.5

With any of the microphone mechanisms, different directional patterns of response can be produced. Common directional patterns are called cardiod (for heart shaped) and omnidirectional. There are also "figure-eight" patterns which accept sound from front and rear. The parabolic microphone is unidirectional in the extreme. This project will employ the use of cardiod microphone.

2.3.1 **Wireless Microphone**

Wireless microphones offer extreme convenience, freedom of movement, and simplicity. But they are not without some pitfalls. Here is how to use them properly to maximize these benefits for your application.[10]

About Frequencies

1. There must be one transmitter and one receiver to make a complete wireless system, and they both must be on the same frequency.

2. Wireless frequencies are shared with TV stations, communications equipment, and other wireless mic systems.

3. Because of frequency sharing, chances are someone else in the area might be using the same frequency as your wireless system.

4. Government regulations also set other technical requirements, including limits on maximum transmitter power. - If any two transmitters are operating on the same frequency, interference will result and the wireless system will be unusable. Two transmitters cannot be used with one receiver at the same time.

5. A higher squelch setting on the receiver provides better protection against interference, but can cause a reduction in operating range. Set squelch to the lowest position that reliably mutes the interference.

6. Turn off unnecessary electronic equipment, especially computers, CD players, and other digital devices. These are a relatively common cause of wireless interference, especially if they are near the receiver.

7. If the use of computers or digital devices is necessary, keep them at least 3 feet (1 meter) away from the wireless receiver and its antennas.

8. The practical maximum operating range of a wireless system will vary from as little as 100 feet in heavily crowded indoor situations to approximately 1,000 feet under open outdoor conditions.

9. Diversity systems will almost always have better operating range than non-diversity systems.

10. Receivers must have either one or two external antennas, and there should be a clear open-air path between these antennas and the transmitter. [12]

2.3.2 Loudspeakers

Loudspeaker, or loudspeaker system is an electromechanical transducer that converts electrical signal to sound. [3]

To adequately reproduce a wide range of frequencies, most loudspeaker systems require more than one driver, particularly for high sound pressure level or high accuracy applications. Individual drivers are used to cover different frequency ranges. The drivers are named subwoofers (very low frequencies), woofers (low frequencies), mid-range speakers (middle frequencies), tweeters (high frequencies) and sometimes super tweeters which are drivers optimized for higher frequencies than a normal tweeter. [2]

2.3.3 Types of loudspeaker

2.3.4 Horn speakers

Horn speakers are the oldest form of loudspeaker system, having been used from very early on for cylinder recording players. They use a shaped waveguide in front of or behind the driver to increase the directivity of the loudspeaker and to transform a small diameter, high pressure condition at the driver cone surface to a large diameter, low pressure condition at the mouth of the horn. This increases the sensitivity of the loudspeaker and focuses the sound over a narrower area. A horn loaded speaker can have a sensitivity as high as 110 dB @ 2.83 volts (1 watt @ 8 ohms) @ 1 meter [11]

2.3.5 Piezoelectric speakers

Piezoelectric speakers are frequently used as beepers in watches and other electronic devices, and are sometimes used as tweeters in less-expensive speaker systems, such as computer speakers and portable radios. Piezoelectric speakers have several advantages over conventional loudspeakers: they are resistant to overloads and they can be used without a crossover due to their electrical properties. There are also disadvantages: some amplifiers can oscillate when driving capacitive loads like most piezoelectrics, which results in distortion or damage to the amplifier. Additionally, their frequency response, in most cases, is inferior to that of other technologies. This is why they are generally used in single frequency (beeper) or non-critical applications.

Piezoelectric speakers can have extended high frequency output, and this is useful in some specialized circumstances; for instance, sonar applications in which piezoelectric variants are used as both output devices (generating underwater sound) and as input devices (acting as the sensing components of underwater microphones). They have advantages in these applications.

2.3.6 Electrostatic loudspeakers

Electrostatic loudspeakers use a high voltage electric field (rather than a magnetic field) to drive a thin membrane between two perforated conductive plates called stators. Because they are driven over the entire membrane surface rather than from a small voice coil, they can provide a more linear and lower distortion response than dynamic drivers. They have the disadvantage that the diaphragm excursion is severely limited because of practical construction limitations. The further apart the stators are positioned, the higher the voltage must be to achieve acceptable efficiency, which increases the tendency for attracting dust and producing electrical arcs. For many years electrostatic loudspeakers had a reputation as a generally unreliable and occasionally dangerous product. Arcing remains a potential problem with current technologies, especially when the panels are allowed to collect dust or dirt, or when driven with high signal levels.

Electrostatics are inherently dipole radiators and due to the thin flexible membrane cannot be used in enclosures to reduce low frequency cancellation as with common cone drivers. Due to this and the low excursion capability, full range electrostatic loudspeakers are large by nature, and even so are not outstanding performers at the lowest frequencies. To reduce the size of commercial products, they are often used as a high frequency driver in combination with a conventional dynamic driver which handles the bass frequencies

2.4 **Cables**

Speaker cable is used to connect the speaker outputs on audio devices such as a receiver or amplifier to the inputs on the speaker(s).

Speaker cable can be used bare or with pin, banana, spade or ring terminals

The primary variations in speaker wire are in gauge, insulation color, cable quality and cable style.

Gauge indicates the size of the conductor, with a smaller gauge indicating a larger wire. Larger gauge wire is a more effective conductor.

The insulation colors vary for individual preference, to allow the wires to best blend in against a wall

Types of cables

Standard speaker cable

Figure 2.6

Megacable

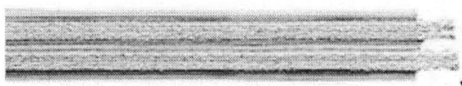


figure 2.7

Table 2.1 : cable quality and advantage

Cable quality	Advantage
Standard	Costs less
Megacable	High copper content and large conductors lower resistance, allowing for higher power transfer
High-amperage	Thicker insulation and rated for higher temperatures which can occur with high-amperage operation

Table 2.2 : cable style and advantage

Cable style	Advantage
Round	Standard cable
Flat	For routing along walls or under carpet or baseboard
Dual-conductor	Standard for stereo speaker connections
Single-conductor	Standard for mono speaker connections

2.5 Power amplifier

An electronic amplifier is a device for increasing the power and/or amplitude of a signal. It does this by taking power from a power supply and controlling the output to match the input signal shape but with a larger amplitude. In this sense, an amplifier may be considered as modulating the output of the power supply.[3]

When installing power amplifiers, it is essential that adequate provision is made to ventilate them.

2.5.1 Classes of amplifier

Power amplifier circuits (output stages) are classified as A, B, AB and C for analog designs, and class D and E for switching designs based upon the conduction angle.

The amplifiers mostly used in Public address systems are Class AB and Class D amplifiers. Class AB amplifiers need bulky transformers made of copper wiring and large metal heat sink for cooling. However, Class D amplifiers, which are more much efficient, weigh much less than Class AB amplifiers producing an equivalent power output.

2.5.2 **Amplifier circuit**

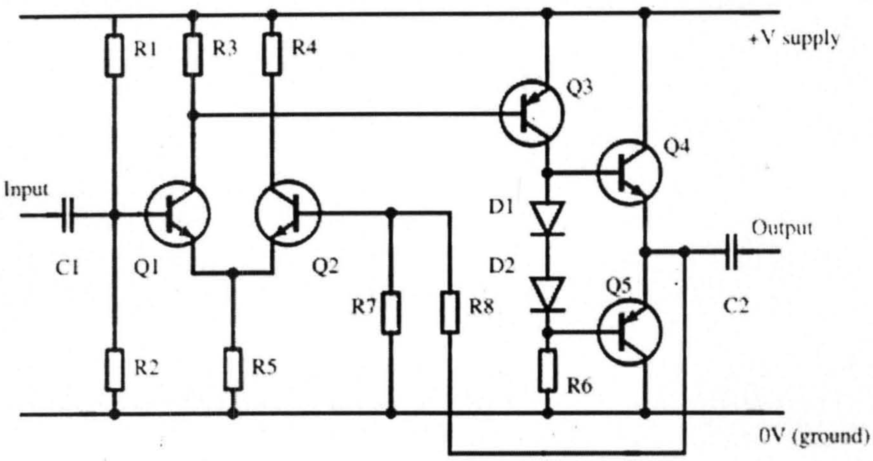


Figure 2.8

The input signal is coupled through capacitor C1 to the base of transistor Q1. The capacitor allows the AC signal to pass, but blocks the DC bias voltage established by resistors R1 and R2 so that any preceding circuit is not affected by it. Q1 and Q2 form a differential amplifier (an amplifier that multiplies the difference between two inputs by some constant), in an arrangement known as a long-tailed pair. This arrangement is used to conveniently allow the use of negative feedback, which is fed from the output to Q2 via R7 and R8.

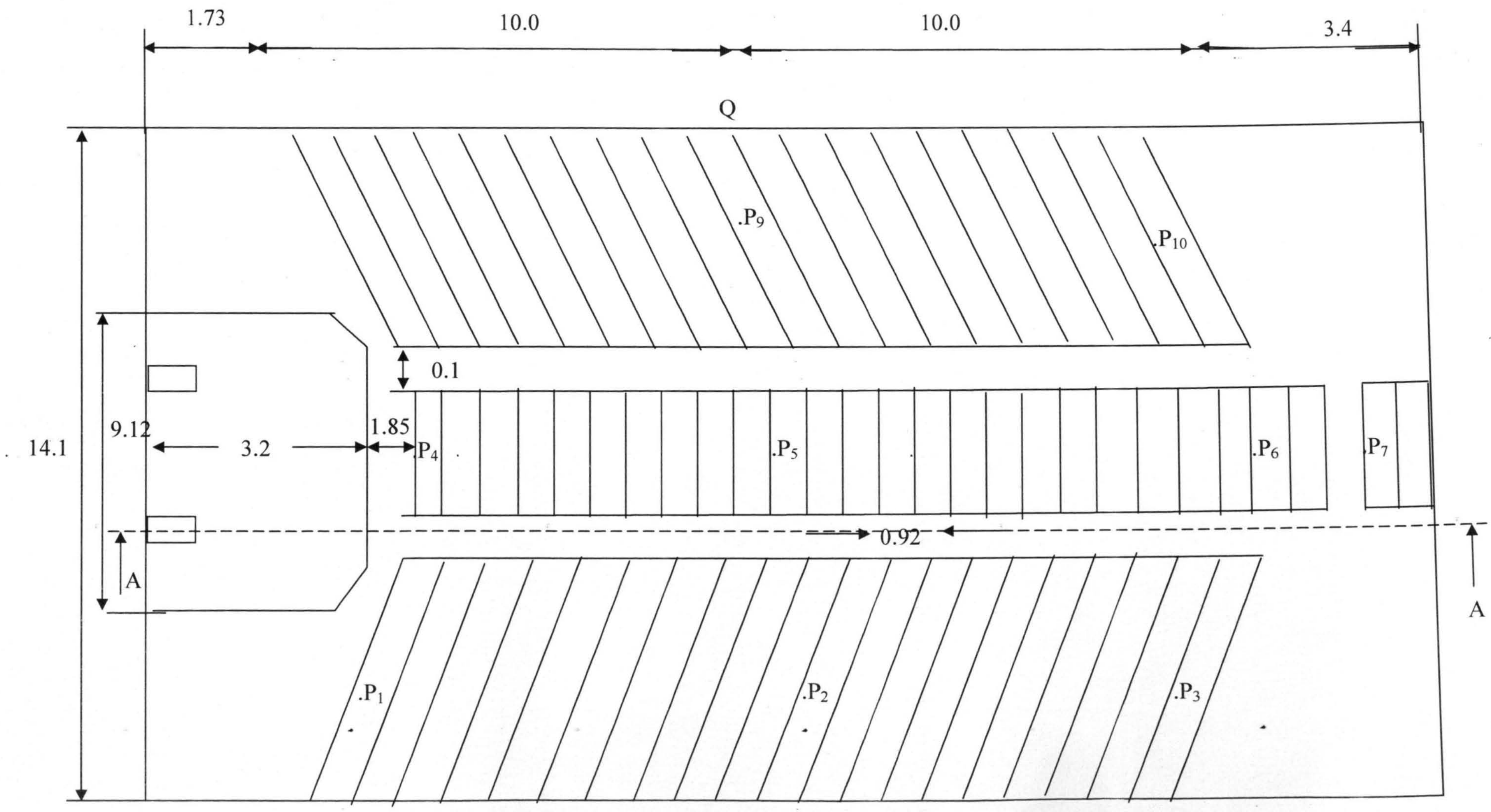
The negative feedback into the difference amplifier allows the amplifier to compare the input to the actual output. The amplified signal from Q1 is directly fed to the second stage, Q3, which is a common emitter stage that provides further amplification of the signal and the DC bias for the output stages, Q4 and Q5. R6 provides the load for Q3 (A better design would probably use some form of active load here, such as a constant-current sink). So far, all of the amplifier is operating in Class A. The output pair are arranged in Class AB push–pull, also called a complementary pair. They provide the majority of the current amplification and directly drive the load, connected via DC-blocking capacitor C2. The diodes D1 and D2 provide a small amount of constant voltage bias for the output pair, just biasing them into the conducting state so that crossover distortion is minimized. That is, the diodes push the output stage firmly into class-AB mode (assuming that the base-emitter drop of the output transistors is reduced by heat dissipation. [11]

CHAPTER THREEE

3.0 DESIGN ANALYSIS

Auditorium design begins with the loudspeaker and how it plays sound into the hall. It ends with how the hall returns reflections of the sound back to the audience. There are six loudspeakers to be used; each was designed to be 50 watts, making up a total of 300 watts. The designed amplifier should also be 300Watts. The speakers should produce a sound pressure level at about 80 dB everywhere in the seating area. That is, the loudspeaker system should sound similar no matter where a person is seated. This is achieved when the speaker system is tested and confirmed to provide a fairly flat frequency response curve for every seat in the house. The Figures 3.0 and 3.1 show the plan and the sectional view of the auditorium respectively. And figure 3.3 shows the loudspeakers distribution on the ceiling.

3.1 The plan view of the auditorium



3.2 The sectional view of the auditorium

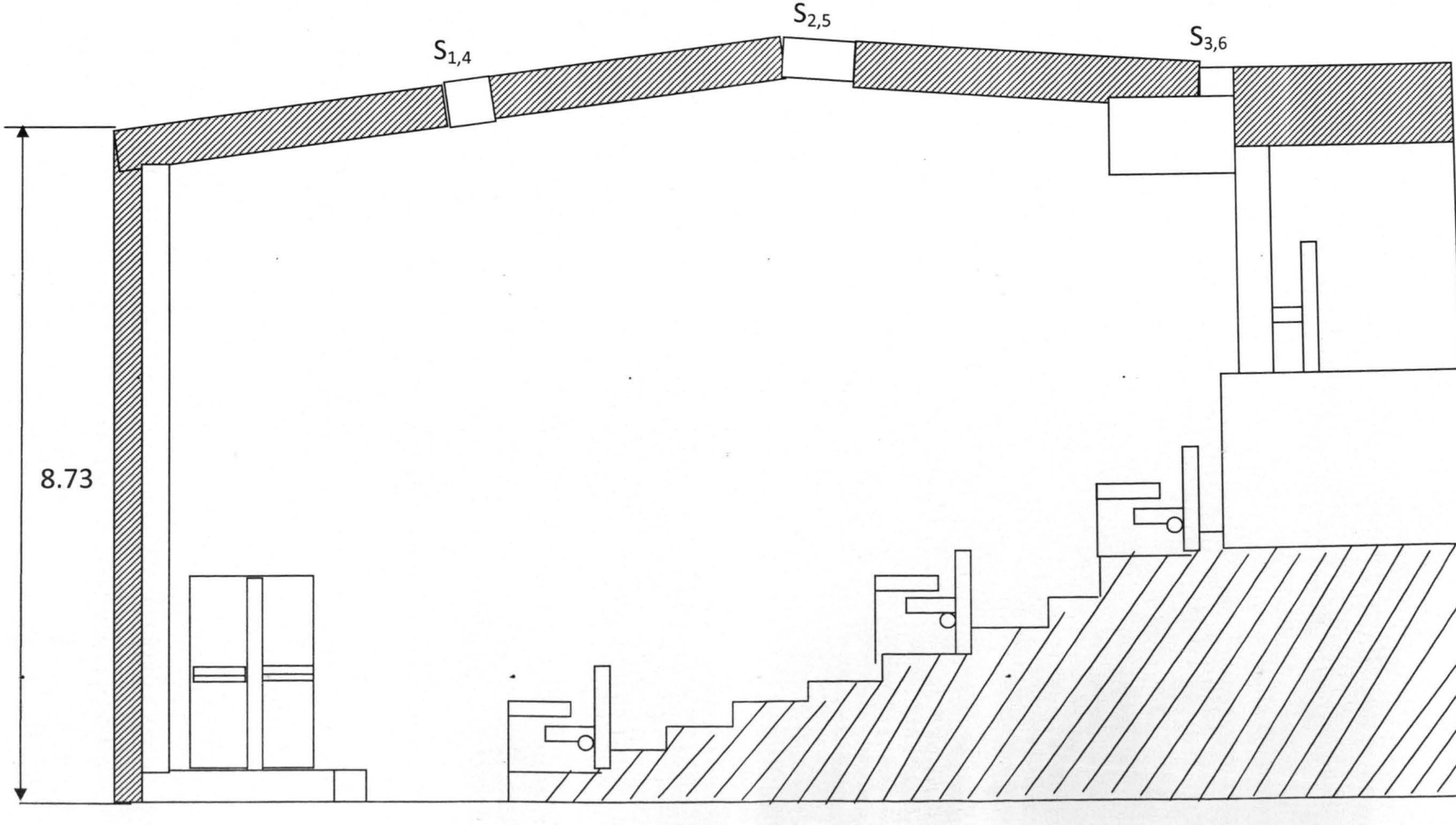
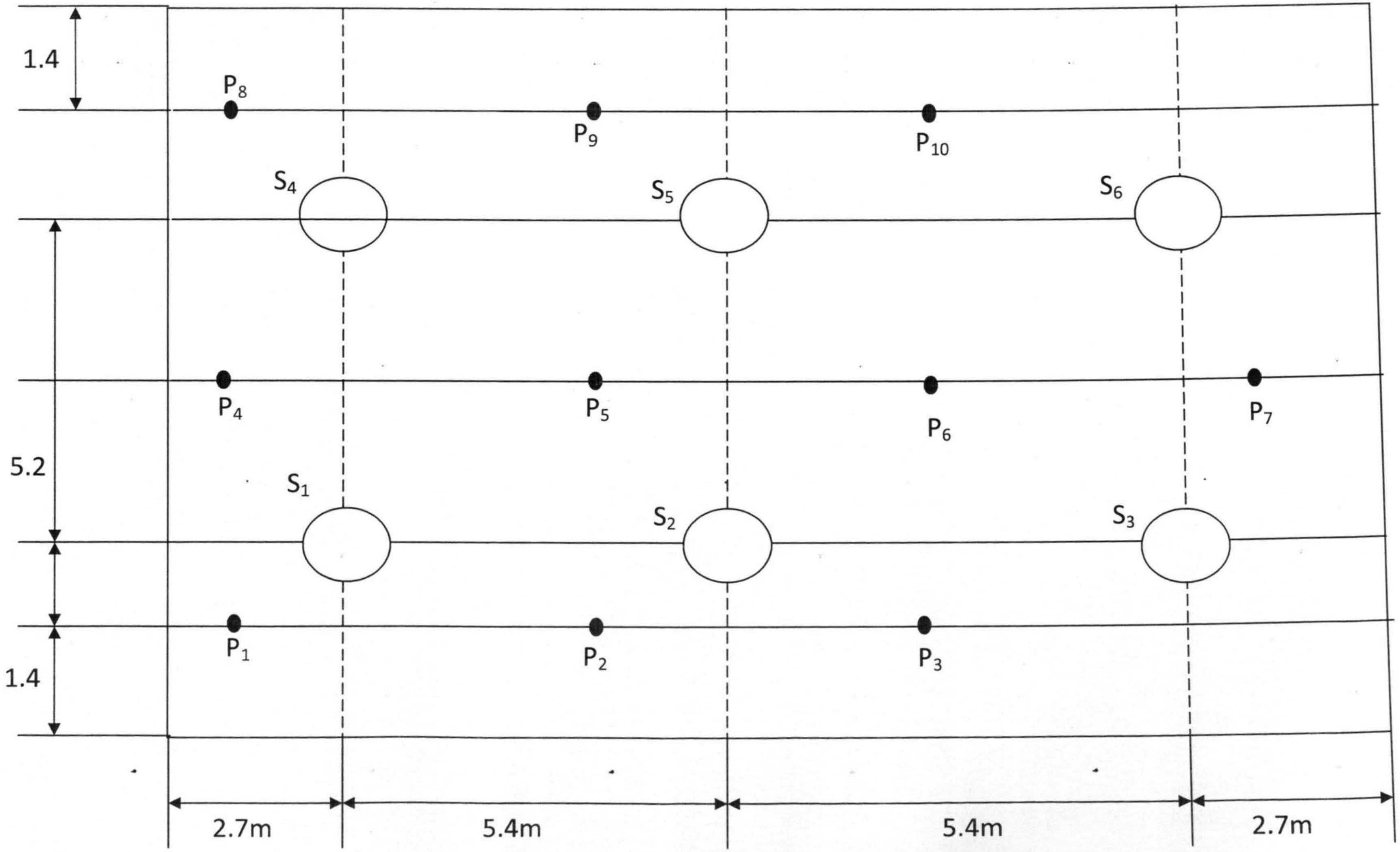


Fig 3.2 Sectional view of the auditorium

3.3 **The loudspeakers distribution on the ceiling**



3.4 **Distance Of The Loudspeakers To The Point Of The Listeners**

Before the distances can be determined, the following measurements are first determined :

1. Total height of the room = 8.73 metres
2. Internal width of the hall = 14.1 metres
3. Internal length of the hall = 25.13 metres
4. Floor to ear level when on seat= 1.13 metres
5. Distance between seats = 0.92 metres
6. Each steps of the floor = 0.17 metres
7. Length of walk ways = 1.0 metres

3.4.1 Distance of the loudspeaker S1 to point P1

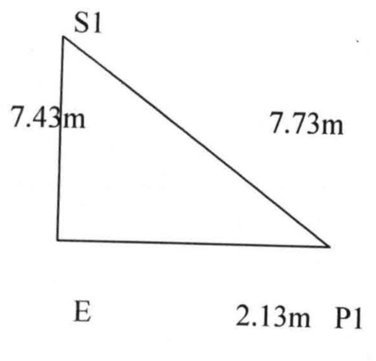


Figure 3.4

From figure 3.4 above, the distance of loudspeaker s1 to point p1 is as follows;

Where : S1 = loudspeaker 1,

P1 = listener point 1,

E = ear level

S1E = mounting height

S1P1 = distance of S1 to P1

Hence, by using Pythagoras theorem,

$(S1p1)^2 = (EP1)^2 + (S1E)^2$ ....(i)

$S1P1 = \sqrt{((EP1)^2 + (S1E)^2)}$

$S1P1 = \sqrt{((2.13)^2 + (7.43)^2)}$

$= \sqrt{59.7418}$

S1P1 = 7.73 metres

Similarly, at point P2;

The distance from loudspeaker s1 to point P2 is

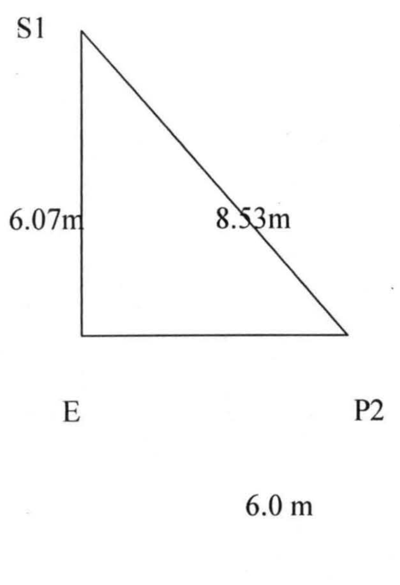


Figure 3.5

From figure 3.5 above,

Mounting height S1E = 6.07m

Therefore,

$(S1P2)^2 = (S1E)^2 + (EP2)^2$ .... (ii)

S1P2 = $\sqrt{((6.07)^2 + (6.07)^2)}$

S1P2 = √(72.8449

S1P2 = 8.53 metres

The remaining distances were obtained and all presented in to a tabular form as shown in the table below;

Table 3.0 distance between the loudspeakers and the listening points

Speaker number Si	Listening positions									
	P1	P2	P3	P4	P5	P6	P7	P8	P9	P10
S1	7.73	8.53	15.36	8.38	9.12	15.61	17.65	11.72	12.16	17.47
S2	12.14	6.63	8.07	12.35	7.34	8.62	9.93	14.97	10.91	11.78
S3	19.17	11.98	5.08	19.22	11.70	5.92	4.30	20.84	14.65	9.96
S4	11.42	12.10	17.47	8.38	9.12	15.61	17.65	11.42	8.53	15.36
S5	11.52	10.91	11.78	12.35	7.34	8.62	9.93	12.14	6.63	8.07
S6	20.84	14.65	9.96	19.23	11.70	5.92	4.30	19.17	11.98	5.08

3.5 **Design of loudspeaker power (w)**

The total power by a loudspeaker is given by

$W = P^2/\ell C \times 4\pi r^2$..(iii)

Where P = sound pressure (N/m^2)

ℓ = density of air (kg/m^3)

C = velocity of sound in air (m/s)

r = distance of the loudspeaker to the listener (m)

But $\ell = 1.18$ kg/

C = 344 m/s

3.5.1 **sound pressure level of a loudspeaker**

The sound pressure level of a loudspeaker is given by;

$Spl = 10\times\log P^2/P_{ref}$..(iv) [9]

Where: spl = sound pressure level (dB)

P = sound pressure (N/m^2)

P_{ref} = reference pressure (N/m^2)

3.5.2 **Sound pressure level for an auditorium**

The comfortable or optimum sound pressure level for a healthy ear in an auditorium is between 79 to 80dB [5]

However, in this design, 80dB will be chosen

That is spl = 80dB

$P_{ref} = 2\times10^{\wedge-5}$ N/m^2

Now, converting pressure level (dB) in to pressure(N/m^2)

From equation (iv),

The pressure P is obtained as follows;

$Spl = 10\times\log P^2/P_{ref}$

But spl = 80dB (standard value in auditorium)

Therefore,

$80 = 10\times\log P^2/P_{ref}$

$80/10 = \log P^2/P_{ref}$

$P^2/P^2_{ref} = 10^{\wedge[80/10]}$

$P^2/P^2_{ref} = 10^{\wedge 8}$

$P^2 = P^2_{ref}\times 10^{\wedge 8}$

$P^2 = (2\times 10^{\wedge -5})^2\times 10^{\wedge 8}$

$P^{2=}(4\times 10^{\wedge -10})\times 10^{\wedge 8}$

$P^2 = 0.04\ N/m^2$

$P = 0.2\ N/m^2$

Now, to design the power of loudspeaker by using equation (iv) as below;

$W = P^2/\ell C \times 4\pi r^2$

Where P = pressure = $0.2N/m^2$

ℓ = density of air = 1.18kg/m^3

C = velocity of sound in air = 344m/s

r = loudspeaker distance to listener Position

Then equation (v) becomes

$W = 0.2^2/(1.18 \times 344) \times 4\pi\, r^2$

$W = 1.2383 \times 10^{\wedge -3}\, r^2$...(vi)

Now, the power by loudspeaker S1 to point P1 is

$W_{11} = 1.2383 \times 10^{\wedge -3}\, r^2$

From table 3.0 r = distance between S1 to P1 = 7.73m

Therefore, $W_{11} = 1.2383 \times 10^{\wedge -3}(7.73)^2$

$W_{11} = 1.2383 \times 10^{\wedge -3}(59.7529)$

$W_{11} = 0.074$Watts

Similarly, At position P2, the power by loudspeaker S1 is

$W_{12} = 1.\ 2383 \times 10^{\wedge -3}\, r^2$

but r = distance between S1 to P2 = 8.53m

$W_{12} = 1.2383 \times 10^{\wedge -3}(8.53)^2$

$W_{12} = 1.2383 \times 10^{\wedge -3}(72.7609)$

$W_{12} = 0.090$Watts

The power by each speaker to the various positions inside the hall is tabulated in the table below;

Table 3.1 power by the loudspeakers to point of listeners

Speaker number Si	Power at a point due to speaker (watts) W = 1.2383×10^-3 r^2									
	P1	P2	P3	P4	P5	P6	P7	P8	P9	P10
S1	0.074	0.090	0.292	0.087	0.103	0.302	0.386	0.170	0.183	0.378
S2	0.183	0.054	0.081	0.189	0.067	0.092	0.122	0.278	0.147	0.172
S3	0.455	0.178	0.032	0.457	0.171	0.043	0.023	0.538	0.266	0.123
S4	0.161	0.181	0.378	0.087	0.103	0.302	0.386	0.161	0.090	0.292
S5	0.164	0.147	0.172	0.189	0.067	0.092	0.122	0.183	0.054	0.081
S6	0.538	0.266	0.123	0.457	0.171	0.043	0.023	0.455	0.178	0.032

Similarly, the average power by each speaker is

$$W_{av1} = (W11 + W12 + W13 + W14 + W15 + W16 + W17 + W18 + W19 + W110)/10$$

From table 3.1

$$W_{av1} = (0.074 + 0.090 + 0.292 + 0.87 + 0.103 + 0.302 + 0.386 + 0.170 + 0.183 + 0.378)/10$$

$$W_{av1} = 2.848/10$$

$$W_{av1} = 0.2848 \text{ Watts}$$

$$W_{av2} = (0.183 + 0.054 + 0.081 + 0.189 + 0.067 + 0.092 + 0.122 + 0.278 + 0.147 + 0.172)/10$$

$$W_{av2} = 1.385/10$$

$$W_{av2} = 0.1385 \text{ Watts}$$

$$W_{av3} = (0.455 + 0.178 + 0.320 + 0.457 + 0.171 + 0.043 + 0.023 + 0.538 + 0.266 + 0.123)/10$$

$$W_{av3} = 2.574/10$$

$$W_{av3} = 0.2574 \text{ Watts}$$

$$W_{av4} = (0.161 + 0.181 + 0.378 + 0.087 + 0.103 + 0.302 + 0.386 + 0.161 + 0.090 + 0.292)/10$$

$$W_{av4} = 2.141/10$$

$$W_{av4} = 0.2141 \text{ Watts}$$

$$W_{av5} = (0.164 + 0.147 + 0.172 + 0.189 + 0.067 + 0.092 + 0.122 + 0.183 + 0.054 + 0.081)/10$$

$W_{av5} = 1.271/10$

$W_{av5} = 0.1271$ Watts

$W_{av6} = (0.538 + 0.266 + 0.123 + 0.457 + 0.171 + 0.043 + 0.023 + 0.455 + 0.178 + 0.032)/10$

$W_{av6} = 2.286/10$

$W_{av6} = 0.2286$ Watts

3.5.3 **Total Acoustic Power In The Room**

$W_{tot} = w_{av1} + w_{av2} + W_{av3} + W_{av4} + W_{av5} + W_{av6})/6$

$W_{tot} = 1.2505$ Watts

Hence, the acoustic power by each speaker

$W(_{tot)av} = W_{tot/6}$

$W(_{tot)av} = 1.2505/6$

$W(_{tot)av} = 0.2084$ Watts

3.5.4 **Attenuation Factor in the room**

There are many attenuation factors in the room including variation in the temperature of air, Audience and furniture in the room. Therefore, 10% of the acoustic power is added to the value obtained. That is, 10% of 0.2084 = 0.02084Watts now, the final value will be

0.2084 + 0.02084 =0.22924Watts

3.6 **Efficiency of loudspeakers**

Loudspeaker efficiency is the defined as the sound power output divided by the electrical power input. Most loudspeakers are inefficient transducers; about 1% of the electrical energy sent by amplifier is converted . To the acoustic energy we can hear. [10] The remaining is converted to heat Typical loudspeaker have efficiency of 0.5—4% [3]

3.6.1 **Input Electrical Power Of loudspeaker**

The input electrical power can be obtained using efficiency formula as below;

Efficiency = (output power/input power)$\times$ 100..vi

In this design however, efficiency of speaker is assumed to be 0.5%

But the output sound power is = 0.22944 Watts for one speaker

Input power is = ?

Then, from equation (vii)

input power = (output power/efficiency)$\times$ 100

input power = (0.2294/0.5)$\times$100

input power = 0.45848$\times$ 100

input power = 45.848 Watts (50 Watts)

That implies the input electrical power of each loudspeaker = 50 Watts

Then, for the six loudspeaker = 6 $\times$ 50 = 300 Watts

3.6.2 **Design of the amplifier**

Since the total electrical power of the loudspeakers is 300Watts, the required amplifier should be 350Watts to enable future load extension.

3.7 **Power level SWL at a point in the hall**

The sound power level in decibel is given by

SWL = 10log(sound power)/(reference power)……………………………………….vii

From table 3.3 above, the power level SWL at every point is thus,

SWL = 10logW/Wref ………………………………………………………………………viii

where : SWL = Sound power level

W =Power at a point

Wref = reference power = 10^{-12} Watts

Now, from equation viii, the power at each level was calculated and also tabulated in table 3.3 below,

Table 3.3 Sound power and power level at the point of listeners

points of listeners	average power (Watts) at a point	sound power level(dB)
P1	0.263	114
P2	0.153	112
P3	0.180	113
P4	0.244	114
P5	0.114	111
P6	0.146	112
P7	0.177	112
P8	0.298	115
P9	0.153	112
P10	0.180	113

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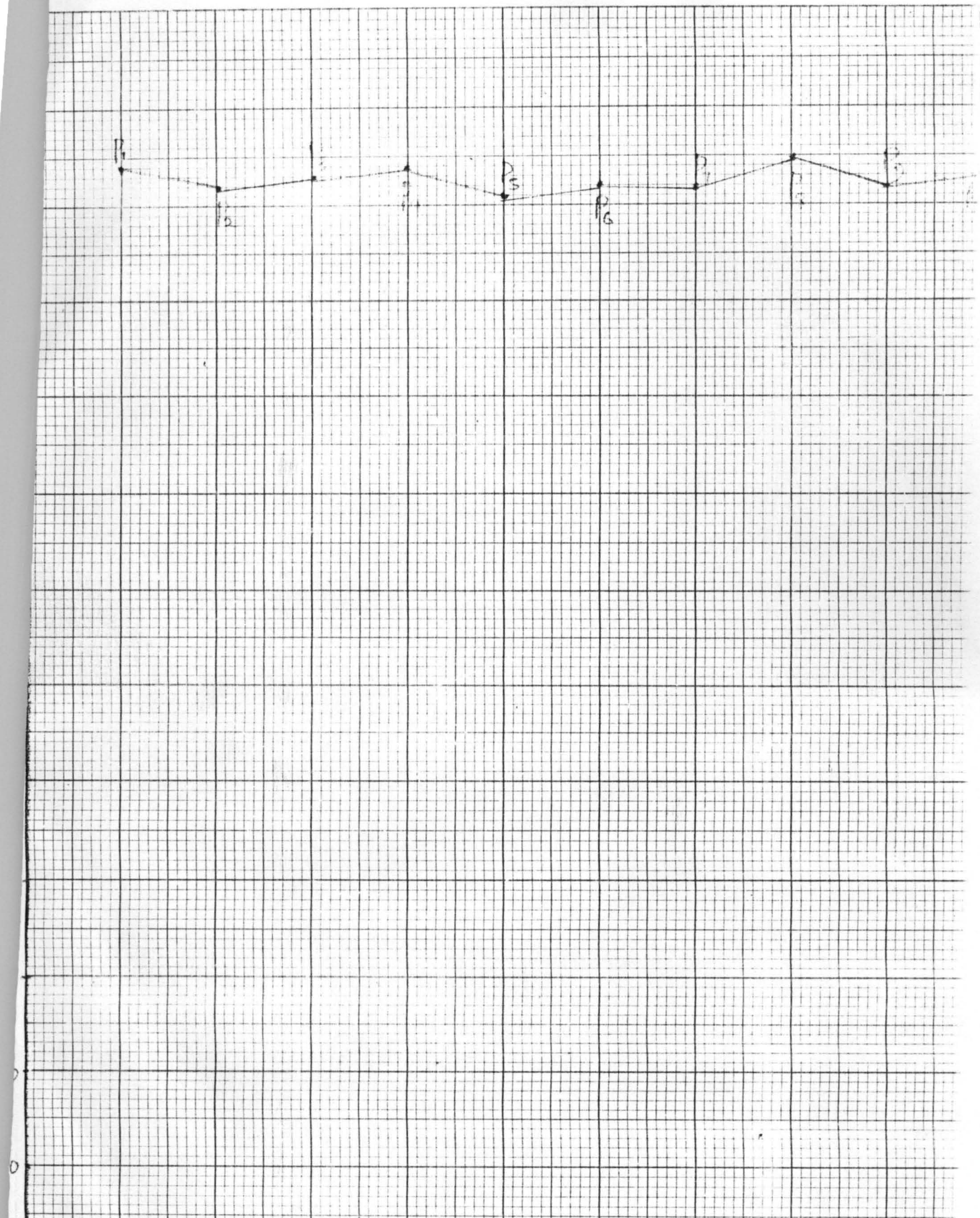

CHAPTER FOUR

4.0 Discussion Of Results

In an anechoic room, loudness is the strongest cue to perceived distance. When a dry source is produced through a loudspeaker, the perceived distance depends on its playback level and not on the loudspeaker distance [7].

In a reverberant room, the ratio between direct and late sound is the most significant cue to perceived distance and the level has less influence. With one distant loudspeaker, it is difficult to produce a sound image just in front of the listener unless reverberation in the room is removed.

The loudspeaker array which form a focus increases the ratio of direct part at listening position, and produces a very far from the array, the effect of the loudspeaker array is the same as that of a super directional loudspeaker system. [8].

The number of loudspeaker array determines the level of direct sound, whereas the interval between loudspeakers affects the total radiation or power of reflections in the room. The total power in the room is the cumulative power from the six loudspeakers. The acoustic power from each loudspeaker was obtainedto be 0.2084Watts but due to be attenuation factors, the value was increased by 10% to 0.22924Watts. The input power depends on the efficiency and the output power. With 0.5% of the efficiency, the electrical input of each loudspeaker was found to be 45.848Watts which is not available hence, 50Watts was chosen for the speaker thereby, making a total of 300Watts for the six loudspeakers.

The specification of the amplifier depends on the electrical power of each speaker and also the number of the loudspeakers to be used. Thus, the amplifier electrical output power also is 300Watts. From the table 3.3 and the graph in chapter three, it was seen that the difference between the sound power levels at the points of listeners is not more than 5dB, hence, uniform distribution of sound has been achieved.

CHAPTHER FIVE

5.0 CONCLUSION AND RECOMENDATIONS

5.1 Conclusion

A model for determining appropriate amplifier and loudspeaker units in the volume of the auditorium has been investigated which gives a 300Watts amplifier on six 50Watts loudspeaker each for the auditorium. Uniform distribution of sound was also achieved in the hall. The amplifier designed was done with consideration of some factors such as height of the room, losses in the system, attenuation due to the temperature of air, audience and furniture in the room.

5.2 Problems Encountered

The following problems were encountered :

1. Removal of reverberation in the room completely.
2. Architectural design of the room.

5.3 Recommendations :

1. I would recommend that further work be done on the area of electro-acoustics in rooms
2. More listening points should be considered in determining the average value of the power
3. Some other halls should be studied as well in terms of sound reinforcement systems design
4. Audio engineering books should be more included in the departmental liberaries.
5. Further work should include use of a projector in the hall.

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