DESIGN AND CONSTRUCTION OF A SUB FREQUENCY AMPLIFICATION LOUDSPEAKER SYSTEM CAPABLE OF AMPLIFYING FREQUENCIES BETWEEN 45HZ THROUGH 125HZ AND A COUPLED CLASS AB 2KW POWER AMPLIFICATION SYSTEM

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BENIN CITY, EDO STATE,
NGERIA.

SEPTEMBER, 2023.

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A PROJECT SUBMITTED TO THE DEPARTMENT OF ELECTRICAL AND ELECTRONIC ENGINEERING, FACULTY OF ENGINEERING, UNIVERSITY OF BENIN, BENIN CITY. IN PARTIAL FULFILLMENT OF THE REQUIREMENT FOR THE AWARD OF A BACHELOR OF ENGINEERING (B. ENG) IN ELECTRICAL AND ELECTRONIC ENGINEERING

SEPTEMBER, 2023

CERTIFICATION

This is to certify that this project was carried out by **ANINE ELOHO ENDURANCE** with matriculation number **ENG1704163**; **ASOWA ADRIAN OSEMUDIAMEN** with matriculation number **ENG1704169**; **ETANA NAIMAGBO AYOMIDE**with matriculation number **ENG1708900**; **IDOGHO IKPEMOSIMEH PERFECT** with matriculation number **ENG1707902**; **OSUNDE HELEN AISOSA** with matriculation number **ENG1707859**; **ODIZE ENAKENO GODSTIME** with matriculation number **ENG1607651**; **UWAJE OLUMWENSE HARRISON** with matriculation number **ENG1704252**; **UZOWURU JOSIAH GODWINS** with matriculation number **ENG1708910.**

ENGR. PROF. S. E. IKE **PROJECT SUPERVISOR**	DATE
ENGR. PROF K.O. OGBEIDE **HEAD OF DEPARTMENT**	DATE

DEDICATION

This project is dedicated to all practitioners and technocrats, whose primary pursuits is to achieve a response that is identical to the that of the source signal, and whose final aim is to obtain a linear response in the any listening space.

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Firstly, I want to appreciate God for His help and support throughout this study.

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UZOWURU JOSIAH GODWINS

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ABSTRACT

The primary pursuit of this project was to determine the response of a manufactured sub frequency amplification loudspeaker system capable of amplifying frequencies between 45 Hz through 125 Hz. Secondly, a 2000 W class AB power amplifier was also designed and manufactured to specifically power this loudspeaker design. Finally, using transfer function measurements, the critical identification and analysis of the character of the output wave forms in diverse listening spaces was done.

This project was carried out using a low frequency transducer/driver of size 18 inches, made of Neodymium permanent magnet, a copper voice coil of 4-inches, and a diaphragm made of sturdy paper film. This driver/transducer is tightly suspended within a casket which comprises of a robust acoustic (wooden) compartment made of birch plywood of thickness 4 millimeters(4mm), with an internal rigid structure, critically designed in a bandpass configuration for maximum acoustic power output within which the transducer is immersed/suspended and made to resonate.

The CLASS AB power amplification system was specifically manufactured using bipolar junction transistors, give a maximum output power of 2000 W. This was chosen since an efficiency of about 75% can be gotten.

In order for the frequency/amplitude response to be got, we used several third-party measurement softwares to determine and analyze various response traces using sweeps and test signals in various listening spaces. The third party software include: RATIONAL ACOUSTICS SMAART V8, ROOM EQ WIZARD(REW), DECIBEL-X.

And finally, for the determination of the dispersion(horizontal and vertical) characteristics, the acoustic behavior of our wave fronts and the simulation of our response respectively was predicted using: EASE FOCUS 3 and MEYER SOUND SIM.

CHAPTER ONE

1.0 INTRODUCTION TO THE STUDY

1.1 BACKGROUND OF THE STUDY

The landscape of loudspeaker manufacturing within Africa presents a narrative characterized by a lag in design and production processes compared to global standards. Research has unveiled a concerning trend where loudspeaker equipment produced in the region often falls short of meeting real-life application demands. The result is a prevalence of underpowered systems that introduce non-linearity and distortion into the audio signals within the listening environments where this equipment is deployed. Furthermore, the absence of detailed specification sheets outlining the operational parameters and maximum limits of loudspeaker systems perpetuates the perception of African manufacturers as being deficient in the field of loudspeaker design and manufacturing. This study seeks to challenge and overturn these notions by elevating and adhering to higher standards in the design, manufacturing, and measurement of loudspeaker systems, ultimately delivering efficient and reliable products for both users and technicians.

The ubiquity of loudspeakers in today's world cannot be overstated. They find applications in a vast array of devices, from the diminutive MP3 players to the omnipresent mobile phones and the commanding presence of professional sound systems. The range of frequencies these loudspeakers are capable of amplifying is a determining factor in their power requirements, thereby dictating the specifications of their accompanying systems. In this context, our study delves into the intricate components of the audio reproduction chain, encompassing the power supply, transmission cables, power amplifier, filters, and, crucially, the loudspeakers themselves. Our primary focus is on the design of a sub-low frequency amplification system, a specialized category of loudspeaker systems tailored to amplify frequencies within the first decade of human aural response, typically ranging from 20Hz to 200Hz. This frequency band encapsulates the fundamental frequencies produced by diverse sound sources, whether naturally occurring or artificially generated.

The heart of our design project is centered on the amplification of frequencies between 45Hz and 125Hz, a strategically chosen range given its significance as the locus of most fundamental frequencies produced by earthly sound sources. Our loudspeaker system is engineered to meet this specific requirement. It consists of an 18-inch low-frequency transducer housed within a carefully crafted bass reflex enclosure. This enclosure design plays a pivotal role in housing top-tier transducer components and a passive crossover network, ultimately influencing the system's loudness level or sound pressure level (dB SPL) in any acoustic environment.

Our design endeavor is not solely confined to the choice of frequency range and transducer size. Rather, it extends to address critical parameters such as coil size and electrical sensitivity of the loudspeaker transducer. These factors play a pivotal role in determining the transducer's performance. Our design strategy incorporates high-end components to maximize the coil radius for efficient heat dissipation during extended high-volume usage and increase electrical sensitivity. These design choices aim to ensure that the transducer converts electrical power into sound output with the utmost efficiency, adhering to the laws of physics.

In our quest to determine the frequency response of our designed system, we employ transfer function measurements. This process involves using an excitation signal that encompasses all frequencies of interest to drive the device under test (DUT). The DUT's response is then captured and compared with the original signal. However, it is essential to acknowledge the presence of noise, which inherently reduces measurement precision. Therefore, employing excitation signals with high energy becomes imperative to achieve a satisfactory signal-to-noise ratio across the entire frequency range of interest. Further improvements in signal quality are achieved through gating techniques, which suppress noise and undesired reflections. It is crucial to note that acoustical measurements inherently introduce non-linearity and time variance.

The role of the audio power amplifier in this ecosystem is paramount. It serves as the bridge between low-power audio signals and the loudspeakers, amplifying the former to a level suitable for driving the latter. Hence, the audio power amplifier assumes a critical position in the electronics of the amplification process. However, it must operate transparently, without introducing interference or distortion to the signal passing through it. To achieve this, the power amplifier must exhibit a linear response with known attributes such as Total Harmonic Distortion (THD), sensitivity, and frequency response, which ideally spans the human auditory range of 20Hz to 20KHz.

In contemporary times, electronic devices that emit sound, such as mobile phones, MP4 players, laptops, televisions, and audio equipment, all incorporate audio power amplifiers. The significance of the audio power amplifier in sound reinforcement cannot be overstated. Loudspeakers, no matter how capable, cannot fulfill their amplification potential without the assistance of audio power amplifiers. The transistor, which was invented in the 1940s, revolutionized amplification technology. By the 1970s, transistor amplification had matured, giving rise to various amplifier circuits, including combinations of Class A and Class B amplifiers, high-output current amplification circuits with minimal distortion, and more. These advancements made transistor amplifiers the cornerstone of audio technology.

A pivotal development in the field of audio technology occurred in the early 1960s when Jack Kilby introduced integrated circuits. By the early 1970s, integrated circuits had gained widespread acceptance within the audio industry due to their cost-effectiveness, compact size, and multifunctional capabilities. Thick-film audio integrated circuits and operational amplifier integrated circuits became staples in audio circuits, further advancing the field.

In the realm of transfer function measurement methods, dual-channel FFT analysis is a noteworthy mention. This methodology entails dividing the output spectrum of the Device Under Test (DUT) by that of the input signal, a process inherent to all FFT-based transfer function measurements. However, it is crucial to recognize that the noise source traditionally employed in dual-channel analyzers is non-deterministic, leading to unknown spectrum characteristics. This necessitates the simultaneous capture and processing of both the input and output signals. In acoustical measurements, the precise delay of the acoustical transmission path becomes a crucial factor, as the direct signal undergoes an identical delay, ensuring that corresponding segments of the excitation signal align for analysis. Additionally, to mitigate leakage effects, signal blocks submitted for FFT analysis must be windowed, a process that introduces a margin of error as delayed components undergo greater attenuation compared to the direct sound.

An advancement in dual-channel analysis can be realized by generating the impulse response for each measurement through inverse FFT. Windowing the impulse response introduces a critical degree of control over the level of reflections incorporated into the result and allows for the exclusion of noise occurring outside the windowed interval. This approach enhances convergence speed, leading to more accurate results. The measurement of transfer functions and their associated impulse responses stands as one of the daily tasks integral to acoustical applications. Fourier-transforming the room impulse response obtained from this process provides valuable insights, particularly for detecting low-frequency modes.

Sweep-based measurements hold distinct advantages, particularly in mitigating the adverse effects of time variance. In certain scenarios, they stand as the sole viable option, such as in long-distance outdoor measurements during windy conditions or when assessing analog recording equipment.

1.2 STATEMENT OF THE PROBLEM

The inception of this project stems from a recognition of several pressing issues:

1. The challenge of underpowered sub-low frequency sound emission from sound systems across small, medium, and large-scale applications.
2. The need for an efficient power amplification system that can deliver power to loudspeakers without adulterating the audio signals they carry.
3. The opportunity to introduce the concept of transfer function-based FFT (Fast Fourier Transform) and IDFT (Inverse Fast Fourier Transform) measurements to technocrats and professionals within the telecommunications field.
4. The absence of comprehensive specification sheets outlining system parameters and

applications limits, a void this project seeks to fill.

1.3 AIMS AND OBJECTIVES OF THE WORK

The overarching aim of this study is multi-faceted:

1. Design a sub-low frequency loudspeaker system configured in a bandpass configuration, tailored to amplify frequencies ranging from 45Hz to 125Hz, with a nominal impedance of 4 ohms.
2. Develop a 2000Watt CLASS AB power amplifier uniquely designed to power the loudspeaker cabinet, ensuring it remains transparent to the electrical test signal passing through.
3. Determine the frequency response of the loudspeaker system within controlled environments and unpredictable acoustic settings through the use of transfer function measurements.
4. Generate a comprehensive data sheet or specification sheet encompassing the system's amplitude response in various listening spaces, as well as its frequency response in a controlled environment, thereby providing valuable insights to industry practitioners.
5. Explore and determine critical parameters such as the maximum excursion of the system at rated power levels, the sound pressure level (dB SPL) produced by the system, the phase response, corner loading response, power response of the sub-frequency system, and the system's acoustic behavior and simulation, leveraging tools like EASE FOCUS 3 and MEYER SOUND SIM.

The subsidiary objectives are as follows:

I. Determine the system's frequency response (@ -6dB), revealing the range of frequencies the design can comfortably amplify without distortion.

II. Identify the crossover frequency, which signifies the cut-off frequency from which the design is proficient.

III. Determine the system's maximum sound pressure level (SPL max.) in dBSPL, starting from 1 meter away from the source design through to 50 meters.

IV. Assess the dispersion in both horizontal and vertical axes, shedding light on the sub-low frequency system's omnidirectional properties.

V. Recommend an amplifier (W RMS) that harmonizes efficiently with the loudspeaker's characteristics.

VI. Specify the nominal impedance of the system, ensuring clarity regarding the minimum load impedance compatible with the power amplifier.

VII. Define the input connector type necessary for successful current transmission to the loudspeaker system.

1.4 METHODOLOGY

To navigate the complex web of design, manufacture, and measurement with precision, this study employs a rigorous methodology involving the following steps:

1. In-depth research into the operational principles governing Class AB power amplifiers.
2. Generation of a comprehensive circuit diagram for the Class AB power amplifier, optimized for a minimum output impedance of 4 Ohms.
3. Thorough research into loudspeaker cabinet design and its impact on acoustic characteristics.

4. Calculations to determine precise component values critical for the project.
5. Extensive research into transfer function measurements using Fast Fourier Transform (FFT) techniques, underscoring their significance in analyzing and determining frequency response in various acoustic environments.
6. Implementation of acoustic wavefront prediction software analysis, particularly through the application of tools like EASE FOCUS 3.

1.5 SCOPE

The scope of this project encompasses a wide range of activities, spanning from sourcing high-quality materials to cost estimation, design, and rigorous testing of a sub-low frequency loudspeaker system, which is synergistically powered by a purpose-designed power amplifier. The study extends further to encompass the detailed analysis of the frequency response and acoustic characteristics of the designed system.

1.6 JUSTIFICATION OF THE STUDY

The rationale behind this study is rooted in the palpable need to address the critical issues facing developing countries, especially concerning the scarcity of high-quality power amplification equipment and maximum excursion sub-low frequency loudspeaker systems. The project seeks to offer a robust foundation for companies engaged in the design, manufacturing, testing, and integration of sub-low frequency systems and, by extension, live sound reinforcement practitioners. This, in turn, fosters economic growth by augmenting knowledge bases, driving innovation, and enhancing the application of system design software and measurement techniques. Furthermore, the study is poised to stimulate improvements in existing design patents and, ultimately, provide a more efficient system that safeguards the connected load (the sub-low frequency loudspeaker) from damage due to excessive excursion.

1.7 LIMITATIONS OF THE STUDY

While this study encompasses a wide array of objectives and ambitions, it is important to acknowledge certain limitations:

1. The intricate process of understanding transfer function measurements, including system connection configurations and the interpretation of measurement results, poses a significant learning curve.
2. The weight constraints associated with larger, more sophisticated designs present logistical challenges.
3. The bulk and size of transformers required for sub-low frequency power supplies contribute to the weight and size of the finished power amplifier.
4. Simulation software, such as EASE FOCUS 3, may not fully account for the complexities of real-world acoustic reflections, potentially impacting measurement data.
5. The presence of multiple reflections and ambient sounds in measurement spaces can introduce inconveniences and complexities in determining and analyzing measurement results.
6. The measurement system utilized in transfer function measurements has limitations, particularly in the maximum impulse response trace it can accommodate, which is restricted to 50 milliseconds, equivalent to 55 feet or 16.94 meters. This limitation is

attributed to the use of a demo version of the measurement software.

As this study progresses, it endeavors to address these limitations effectively, harnessing rigorous methodologies and innovative solutions to attain its ambitious objectives.

CHAPTER TWO

2.0 INTRODUCTION TO LITERATURE REVIEW

The history of the subwoofer is closely intertwined with the evolution of full-range loudspeakers. In the mid-1950s, several manufacturers of hi-fi loudspeakers began introducing compact, closed-box enclosures featuring long-throw woofers. It was during this period that a talented engineer named James Novak, working at Jensen, developed the equations describing a bass reflex (or vented) loudspeaker design.

The capability to reproduce extended low frequencies garnered significant interest from the movie industry. In the mid-1970s, Cinema (then Cinema-scope) and Todd-AO Studios introduced multichannel sound to large cinema screens, incorporating a center mono channel dedicated to providing 'bass extension.' The first movie to feature this 'bass extension' channel was Star Wars in 1977.

As subwoofers became an integral addition to standard stereophonic reproduction systems in the late-1970s, their significance has grown exponentially. Today, subwoofers play a crucial role in various sound reproduction systems, from live music PA systems to DJ setups and beyond, thanks to the widespread acceptance of multichannel audio and the emergence of music genres that encompass extreme low-frequency content.

Sub-low frequencies pose a unique challenge when it comes to directional perception in various acoustic environments. This challenge arises from their incredibly long wavelengths, which effectively fill the entire space. Consequently, sub-low frequencies experience both constructive interference (where output responses are reinforced) and destructive interference (where output responses cancel out), leading to interference and corruption of the final output response. This interference and distortion in the test signal input result in a non-linear and distorted response through the audio amplification system.

Figure 2.1: a sub low frequency loudspeaker system.

The construction of a sub low-frequency loudspeaker system incorporates an audio power amplifier and a loudspeaker, connected together via a transmission cable made of pure copper. The measurements conducted through transfer functions aim to determine the actual response of the amplified signal within any given room or space, providing specific insights into the response at particular positions.

An amplifier is an electronic device designed to increase the voltage, current, or power of a signal. Amplifiers find applications in various domains, including wireless communications, broadcasting, and all sorts of audio equipment. Amplifiers can be broadly categorized into weak-signal amplifiers and power amplifiers. In power amplification, two key considerations are power output and efficiency, where power output is typically measured in watts or kilowatts.

Figure 2.2: The Behringer EP4000 audio power amplifiers in a rack cabinet.

Efficiency is a crucial parameter in amplifier design, representing the ratio of signal power output to the total power input (the wattage demanded from the power supply or battery). This value is always less than 1 and is typically expressed as a percentage. In audio applications, power amplifiers typically exhibit an efficiency range of 30 to 50 percent.

Weak-signal amplifiers serve a different purpose and are primarily used in wireless receivers. They also find applications in acoustic pickups, audio tape players, and compact disc players. Weak-signal amplifiers are specifically designed to handle exceedingly small input signals, which can sometimes be as minuscule as a few nano-volts (units of 10^-9 volts). These amplifiers must generate minimal internal noise while significantly increasing the signal voltage. Field-effect transistors are often the most effective devices for this application. The effectiveness of a weak-signal amplifier is typically denoted by its sensitivity, which is defined as the number of microvolts (units of 10^-6 volt) of signal input required to produce a certain ratio of signal output to noise output, often set at 10 to 1.

Fourier transforms, named after the 19th-century French mathematician and physicist Jean-Baptiste Joseph Fourier, are based on the concept that complex signals, such as speech or music, can be constructed from or broken down into sine waves with varying amplitudes and phase relationships. Fourier transforms are extensively utilized in audio analysis to reveal the spectral content of time-domain signals. Inverse Fourier transforms (IFTs), on the other hand, are employed to reconstruct time-domain signals from spectral data.

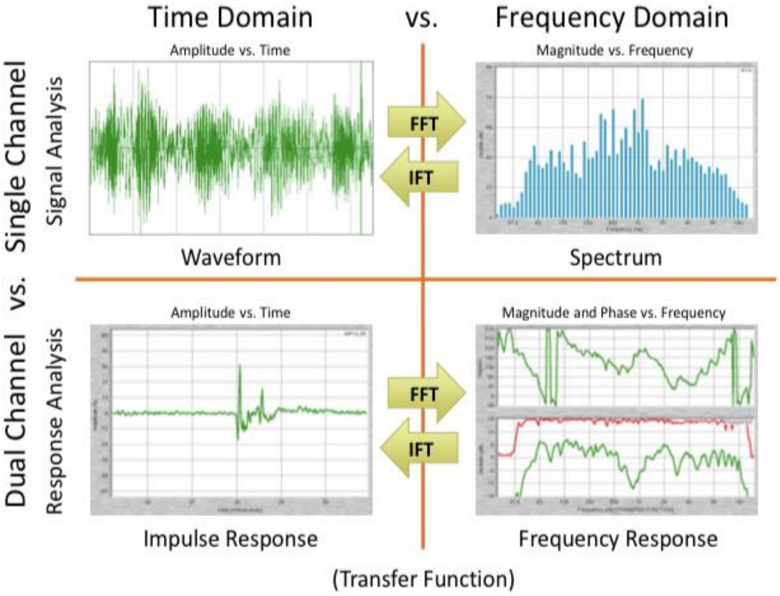


Figure 2.3: Dual-channel vs. single-channel measurements in the time domain and frequency domain.

Measuring transfer functions and their associated impulse responses is a fundamental practice in various domains of acoustics, and it plays a pivotal role in many applications. Here are some examples:

- **Loudspeaker Development:** Loudspeaker developers frequently assess the frequency response of new prototypes before releasing them for production. In addition to on-axis responses, a complete set of polar data, requiring multiple measurements, is essential for comprehensive characterization.
- **Room Acoustics:** In room acoustics, the impulse response takes center stage. It provides

critical insights into various acoustical parameters related to perceived sound transmission quality. For instance, Fourier-transforming the room impulse response helps identify modes at low frequencies.

- **Building Acoustics:** In building acoustics, assessing the frequency-dependent insulation against noise from outside or other rooms is a common concern.
- **Vibro-Acoustics**: Vibro-acoustics involves studying the propagation of sound waves in materials and their radiation from surfaces. This field relies on measurements with shakers and simulations.
- **Profiling by Detection of Reflections (Sonar, Radar):** The measurement of impulse responses is also crucial in fields related to reflection-based profiling, such as sonar and radar.

For acquiring room impulse responses (RIR) for convolution with dry audio material, an essential consideration is the dynamic range. Due to the wide dynamic range of human hearing and the logarithmic relationship between sound pressure level (SPL) and perceived loudness, any anomalies in the reverberant tail of an RIR are easily discernible. This is especially relevant when convolution is used with speech and monitored through headphones, as required for virtual reality experiences based on binaural responses. To ensure high-quality RIRs, it's essential to achieve a signal-to-noise ratio (S/N ratio) at least equivalent to the 16-bit CD standard, which is more than 90 dB.

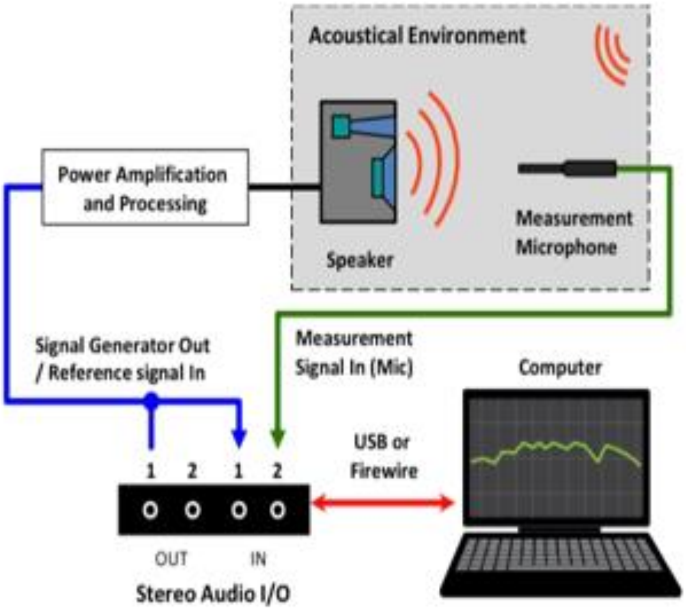
$$G(s) = \frac{C(s)}{R(s)}$$

$$\text{Where, } R(s) = \mathcal{L}r(t),\ C(s) = \mathcal{L}c(t)\ \&\ G(s) = \frac{\mathcal{L}c(t)}{\mathcal{L}r(t)}$$

Sweeps as excitation signals have proven to alleviate many of these limitations. By using a sweep somewhat longer than the RIR to be recovered, it becomes possible to exclude harmonic distortion products, leaving background noise as the primary limitation for the achievable S/N ratio. Sweep-based measurements are also less vulnerable to the effects of time variance, making them suitable for long-distance outdoor measurements under challenging weather conditions or for assessing analog recording equipment.

The dual-channel analysis can be significantly enhanced by generating the impulse response for each measurement using inverse Fast Fourier Transform (FFT). Windowing the impulse response provides the crucial flexibility to control the reflections entering into the result and to suppress noise outside the windowed interval, which expedites the convergence process.

The transfer function is a dual-channel measurement technique that allows the determination of a system's frequency response by comparing its input (the reference signal) to its output (the measurement signal). This measurement yields a complex signal that represents the difference between the two signals in both magnitude and phase. The results offer insights into the overall processing behavior of the system under test (SUT) across different frequencies or time. The transfer function is a valuable tool for assessing the frequency response of various components in a sound system, whether they are electrical (such as EQs and mixers) or electro-acoustical (including loudspeakers, their drive electronics, and their acoustic environment). It finds applications in loudspeaker design, equipment evaluation, equalization, and sound system optimization, among others.



Basic connections and routing should be pretty much the same for any audio I-O device you may encounter. We trust that you understand the audio cabling and connections necessary connect your equipment together. Note that the audio I-O device in this case could conceivably be the computer's built-in stereo line input and headphone output, in conjunction with a self-powered measurement microphone or external mic preamp and phantom power supply and a little bit of creative cabling.

Figure 2.4: transfer function configuration for acoustic output determination.

Dual-channel measurements compare two signals to find the similarities and differences between them. Transfer function and impulse response measurements are dual-channel measurements that compare the output of a device or system to the input signal that produced it. We can therefore say that we are measuring the response of the system to a given stimulus, and because both signals are known, the spectrum of the input signal becomes almost immaterial. We are also able to precisely measure time relationships between the two signals, enabling us to examine phase relationships and find delay times.

2.1 LITERATURE REVIEW

2.1.1 Review On Power Amplifiers

Development of an Audio Power Amplifier for the Management of Speaking Activities by Chibuike Chukwuma Onwubiko, Abiodun Alani Ogunseye, Ayodeji Olalekan Salau and Thomas Kokumo Yesufu(2017): The resource-activity or structure-property relationship of a conventional class AB audio power amplifier was modified to successfully create a flexible duration low power mode and, hence, allow for its dynamic and profitable selection.

Design of Class AB Power Amplifier for RADAR Applications K Karson Joshua, Shanthi. P, Swaraj Varshney (2017): A Class AB power amplifier is designed for RADAR application with higher efficiency and high output power using a GaN HEMT transistor device technology which works for C band and has a new technique of stabilizing the amplifier by using series and parallel resonance at output and input side of the power amplifier.

2.1.2 Review on Sub-low frequency loudspeakers

The main purpose of a loudspeaker enclosure is to prevent the front and rear-firing waves from canceling one another. The best way to examine this phenomenon is to remember that sound waves have peaks, consisting of high air pressure, and troughs, consisting of low air pressure. The air changes from high to low pressure at a speed determined by the pitch of the sound. The position of this waveform at any given time is referred to as its ‘phase’.

The working principle of the sub-low frequency loudspeaker system is based on Faraday’s law of electromagnetic induction which states that “when there is a relative motion between a magnetic field and a solenoid, Electromagnetic Force is induced which is directly proportional to the rate of change of magnetic flux”.

A driver/cone pushing air from both its front and back will create sound waves that are 180 degrees out of phase (peaks and troughs are in opposite alignment). This effect becomes problematic if the two different sound waves meet, as the interference between them will result in sound cancellation. Conversely, if the peaks and troughs are in alignment (in-phase), the sound summation will result in an increase in sound pressure (and hence loudness).

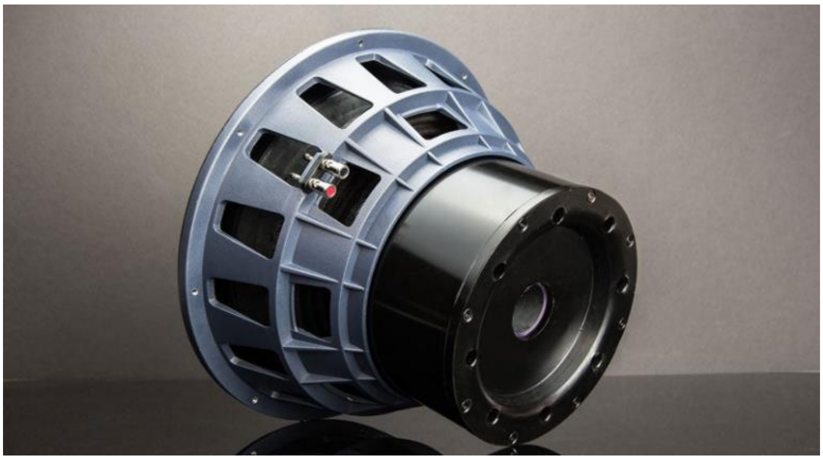


Figure 2.5: A low frequency transducer with ferrite permanent magnet

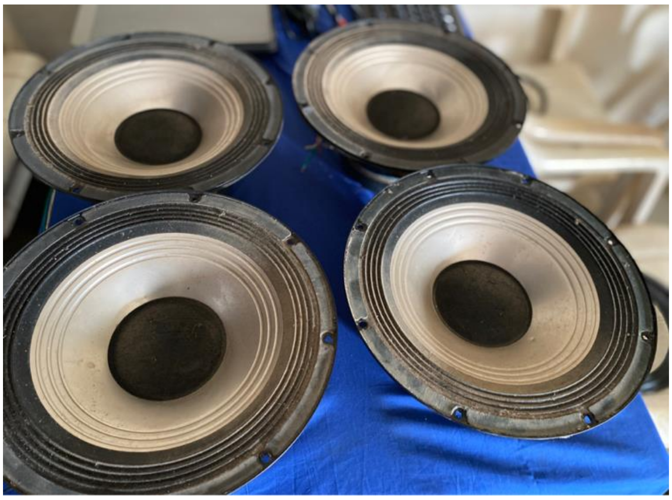


Figure 2.6: Quad 10” diaphragms by BUGERA.

Subwoofer designs configurations using gradient pairing drivers: Robert Evbas (2021): A design contractor in Nigeria, after the design of a dual 18” subwoofer system in bandpass ported configuration, demonstrated and confirmed that the maximum excursion of a subwoofer design is inherently dependent on the short term power rating of the diaphragm, which in turn, affects the maximum output rating of the loudspeaker design.

Christophe Anet of QSC Professional Systems (2023): The most essential property for a subwoofer is its ability to move (or resonate) air, and this is determined by its maximum driver excursion and effective piston area. To achieve a constant sound pressure level as frequency decreases, the air volume that must be displaced is proportional. In other words, the air volume displacement increases by 12 dB per octave with decreasing frequency.

2.1.3 Review on System response determination using Transfer function measurements.

Loudspeaker frequency response determination using impulse response evaluation by SMAART (2019): The word “system,” in this case means everything that affects the spectral energy content and timing of the reference signal, from the point where it was introduced into the signal chain, all the way through to the point where we picked up the resulting output as our measurement signal. In the case of acoustical measurements (captured by means of a microphone), the definition of SUT therefore includes the acoustical path from the loudspeaker as well as the electronic path and the loudspeaker system. If we were measuring just a single processor channel from its input to its output, then that one channel is the SUT – for electronic measurements of a single device we might say “device under test” (DUT).

Stability of the Frequency Response Estimate in Listening Rooms Aki Mäkivirta and Thomas Lund (September 2016): In-room estimates of loudspeaker responses for professional use are typically taken either at one microphone location, replacing the listener with a microphone, or averaging in space, at multiple locations at and relatively close to the listening location. However, the spatial averaging points used in taking a measurement should be chosen based on the room acoustics and on the application. Spatial averaging across a wide area may come with the risk of compromising the result at the main listening position.

Modal Equalization by Temporal Shaping of Room Response Matti Karjalainen, Poju Antsalo, and Aki Mäkivirta: May 2003, Copenhagen, Denmark: The low-frequency behavior of sound reproduction in listening rooms is often problematic due to long-ringing modes that are difficult and expensive to control by acoustic means. Modal equalization has been proposed recently to correct the low-frequency problems by shortening the decay times of problematic modes through modification of transfer function poles.

Stability of the Frequency Response Estimate in Listening Rooms A. Mäkivirta1, T. Lund (2016): The use of single point equalization is confirmed [3] as a safe choice for measurement of studio monitoring rooms having relatively low reverberation times and well controlled room modes. The use of spatial average is not likely to significantly change or improve the outcome of equalization in that case. Spatial averaging across a wide area may come with the risk of compromising the result at the main listening position. The spatial averaging positions should therefore be chosen based on the intention of the room equalization and the acoustic quality of the room.

2.2 THEORETICAL BACKGROUND

2.2.1 SUB-LOW FREQUENCY AMPLIFICATION SYSTEMS

Theory of Operation of an Audio Low Frequency Transducer

A transducer is a device that converts energy from one form to another. In loudspeaker transducers electrical energy is converted into mechanical wave energy or sound energy.

The principle behind the workings of a transducer is primarily based on Faradays law of Electro-Magnetic Induction, which states that: “when there is a relative motion between a solenoid and magnetic field, VOLTAGE (Electro Magnetic Force) is induced in the solenoid which is directly proportional to the rate of change of magnetic flux”.

$$\mathcal{E}=-N\frac{\Delta\Phi}{\Delta t}$$

Where,

E= Induced electromotive force

N= number of coil turns

$\frac{\Delta\Phi}{\Delta t}$ =B, magnetic field strength

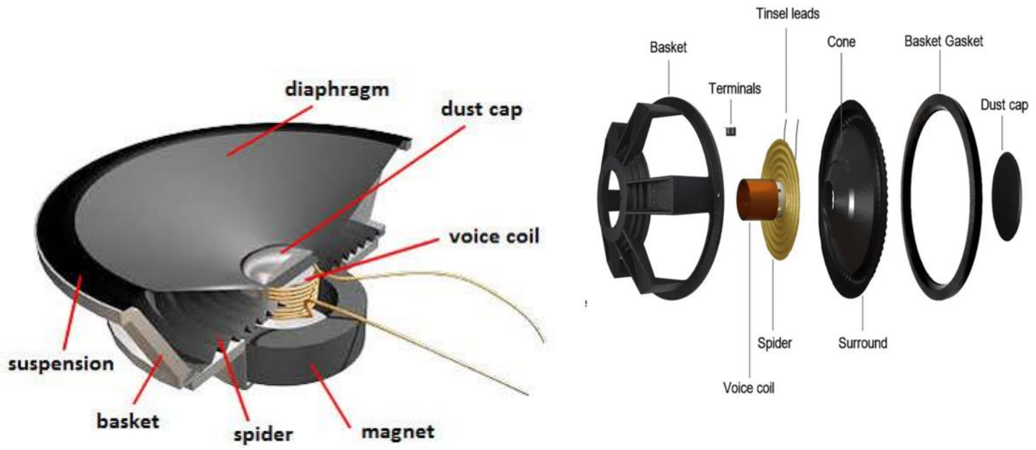


Figure 2.7: overview of the internal structure of a low frequency transducer

Magnet Structure: two pieces of oppositely oriented magnets that produce a radial field from the inner to outer magnet.

Voice Coil: carries the current so that it is always moving in a plane perpendicular to the magnetic field; thus the force always acts on the same axis.

Spider: vibrates rigidly with the voice coil and translates the mechanical energy to the cone.

Cone: produces pressure waves from its surface due to the oscillation of the spider. Basket: holds the components together firmly, preventing motion in parts like the magnet structure.

Dust Cap: protects the cone and circuitry from dust.

How the mechanism works:

- The electric signal passes through the wire in the form of an analog,
- sinusoidal (or other wave.
- The signal enters the voice coil, wrapping around the inner magnet (in the form of
- a solenoid)
- A force is exerted from the stable magnet structure to the free-moving voice coil.

- As the signal's amplitude and frequency change, the force on the voice coil undulates back and forth.
- The voice coil rapidly vibrates along the axis of the magnet structure, thereby vibrating the cone.
- As the voice cone vibrates, the air immediately around it is pressurized and rarified.
- The pressurized air molecules propagate as a wave-this is sound.

The mechanical waves energy is found in the sound waves. Sound waves cause localized differences in pressure within a medium (solid, liquid or gas). As the waves propagate through the medium, they cause instances of maximum compression (max pressure) and maximum rarefaction (min pressure) in the molecules they pass through.

A sub low frequency amplification system is built up to primarily amplify the frequency range of the first decade in the human aural response between 20Hz -200Hz. This encompasses a critically built enclosure housing top of the line transducers and a passive crossover network. This frequency edge which contains the fundamental frequencies of any sound source, naturally or artificially created requires a lot of power for their amplification, top of the line passive electronic crossover components, and a sturdy cabinet design which also determines the loudness level or the sound pressure level (dB SPL) of our system in any space when deployed.

2.3 SIGNALS

Signals play a crucial role in representing sound, and their characteristics are primarily determined by two key factors: frequency and amplitude. Here's how these factors correspond to the properties of sound:

- **Frequency:** Frequency refers to how rapidly the cone (or diaphragm) within a speaker oscillates. In simpler terms, it's about how quickly the speaker moves back and forth. The relationship between frequency and pitch is direct: a high frequency, or fast movement of the cone, produces a high-pitched sound, while a low frequency, or slow movement of the cone, results in a deep-pitched sound.
- **Amplitude:** Amplitude, on the other hand, tells the speaker how forcefully or gently to push the air. This "pushing force" is associated with the pressure variations in the sound wave, which, in turn, determine the volume or loudness of the sound. A larger amplitude corresponds to a louder sound, while a smaller amplitude results in a quieter sound.

Within the context of sound systems, there's a specific frequency range that holds particular significance, spanning from 45 Hz to 125 Hz. This frequency band is essential in musical applications, and it's crucial to control the directionality of the sound system within this range. Doing so helps avoid unnecessary interactions with the shape of the venue and minimizes unwanted reflections that can alter the overall sound quality at various points within the listening space, whether indoors or outdoors.

However, working with frequencies in this range presents unique challenges. Low-frequency sounds are harder to control in terms of their propagation, and they tend to travel further into the environment compared to higher frequencies. Additionally, air absorption is not as significant in the 45 Hz to 125 Hz range. Therefore, careful consideration is required when designing and deploying a subwoofer system to minimize undesirable reflections and ensure optimal sound quality.

Traditionally, subwoofer systems have been placed on the ground in various configurations. However, advancements in technology and logistics have allowed sound system engineers to "fly" subwoofer arrays, suspending them in mid-air. This aerial positioning offers certain advantages but also presents some acoustic challenges.

One of the challenges is related to the omnidirectional nature of woofers, which means that everyone in the room hears sound from all the woofers in the array. However, due to variations in the distances between each woofer and the listeners, there can be points where the sounds from multiple woofers cancel each other out, resulting in reduced or no bass output directly from the woofers. These cancellations create tonal imbalances and uneven bass levels in the venue, which can be especially noticeable in outdoor settings where there is no reverberation to mask these issues. When evaluating the performance of subwoofer systems, two key factors are typically considered:

1. **Low-Frequency Extension:** This refers to how deep the subwoofer can reproduce bass frequencies. A subwoofer with good low-frequency extension can produce deep, powerful bass.
2. **Sound Pressure Level (SPL):** SPL measures the loudness or intensity of the sound produced by the subwoofer. It quantifies how effectively the subwoofer can create high-intensity bass output.

While larger subwoofer drivers have the potential to deliver greater SPL and deeper bass due to the laws of physics, other factors such as accuracy, transient response speed, and the ability to seamlessly integrate with full-range speakers are equally important. Ultimately, the performance of a subwoofer driver also depends on the quality of its motor and amplifier components, so having a larger driver doesn't always guarantee better overall performance .Pitfalls of Big Subwoofer Drivers

Figure 2.8: a configuration of large low frequency amplification systems

Acoustically, subwoofers with 15-, 18- and 20-inch drivers are more susceptible to boomy responses and distortion compared to subwoofers with 10-, 12- or 13-inch drivers because the piston movement of such a large surface area is harder to control. Poorly designed big subwoofer drivers with inadequate motors don't produce accurate bass and struggle to start and stop on a dime. This results in low frequency output that can sound smeared, boomy or bloated and detract from the convincingness of an audio experience.

It's even possible for a subwoofer with a 10- or 12-inch driver to outperform a subwoofer with a larger driver if the motor magnets in the smaller subwoofer can generate greater force and exert better control over the driver. It’s also the reason why a subwoofer’s driver size can be overrated when determining the overall performance. Put simply, a big driver is harder to control, and offers no guarantee of greater SPLs or deeper bass than a small subwoofer.

Engineering Challenges of Big Subwoofer Drivers

A subwoofer driver is only theoretical bass. The bigger the driver, the more amplifier current and magnetic energy required to power and control the piston motion, and this current only gets you so far.

Figure.2.9: internal structure of a low frequency diaphragm showing the voice coils

The current must be fed efficiently through the voice coil that lives in the magnetic gap of the motor magnets. The amount and direction of the current through that coil of wire makes an electro-magnetic field which reacts against the permanent magnetic field of the motor magnets. This will push and pull on the voice coil, which then moves the driver cone to extreme excursion levels so enough air is moved. This movement of air is the sound we hear and feel, including palpable SPLs and bass notes below the limits of human hearing. The more challenging part is to simultaneously maintain pinpoint control of the big driver so it only plays the musical notes and subsonic frequencies that are meant to be heard and felt, and not unwanted distortion. Driver size, SPL and low frequency extension are always critical factors.

2.4 SUB-LOW FREQUENCY LOUDSPEAKER DESIGN CONFIGURATION

Bandpass Subwoofer Box

These are specially designed subwoofer boxes. The woofer cones are placed inside an enclosed chamber. It ensures that the cone can generate louder bass effects with deeper and smoother sounds, with the boom effect.

One bandpass box comes with two chambers. The 1st chamber is an encased chamber that contains the bandpass subwoofer cone. On the frontal section, a ported container/box is housed in a different cavity. The enclosed and ported design makes it a formidable soundbox.

Some advantages of Bandpass subwoofer designs include:

1. **Better Efficiency**

   The bandpass subwoofers are highly efficient. These machines produce the best quality bass and sub-bass waves. The sub ports in a subwoofer usually serve the purpose of low-pass filler. The enclosures only release the desired frequency range, suitable for the environment.

   Bandpass subs offer deep bass extension and produce better sound effects than other subwoofer variants. Hence, the sound level is way better even inside the limited space.

2. **Less Woofer Excursion**

The Bandpass subwoofers are known for their low woofer excursion. These subwoofers allow you to control the cone motion. As a result, it offers a slow excursion. Due to the limited motion of the cone, the woofer delivers lower frequencies and higher excursion. Hence, the air circulation is lower than estimated. The bandpass subwoofer boxes channel the airflow and air pressure through the enclosures. The controlled and rapidly changing airflow results in higher acoustic outputs without cone movements. As a result, the subwoofers do not push the drivers to their limits, resulting in higher durability.

3. **Highly durable design**

Another benefit of a Bandpass Subwoofer is its durability. These subwoofers are made with polypropylene cones. Polypropylene is a rigid and durable material. The cones are sealed in an enclosure to produce the bass. Since the cone is fixed, it offers the best possible quality bass. Moreover, it remains enclosed in a protective case. Hence, the speaker does not get damaged easily and lasts for a long time.

4. **Size Variety**

While bigger sizes are always best, in case you do not want high decibels of revelry. But, you will find different sizes like 4th order bandpass, 6th order bandpass, or even 8th order bandpass sizes as per your personal choice. Additionally, larger bandpass subwoofers usually drive lower frequencies. As a result, you get richer sounds and a better experience.

5. **Features of Sealed-&-Ported-Enclosure**

Bandpass Subwoofers have woofer cones enclosed in a box. The sealed and ported structure makes a lot of difference in the sound quality. The ported area units sound waves which make extra booming waves. The booming even remains consistent in a confined space. Moreover, the sealed construction prevents the sound waves from overlapping each other. Hence, the sound remains accurate and precise.

2.5 AUDIO POWER AMPLIFIER

Amplifiers are needed to boost the current source because the current originally carrying the signal is too weak to render audible sound. Amplifiers use transistors to allow a weak signal to depict the form of a much stronger signal. In this way, the amplifier takes in a signal of a certain frequency and amplitude and puts out a signal with the same frequency, but much larger amplitude.

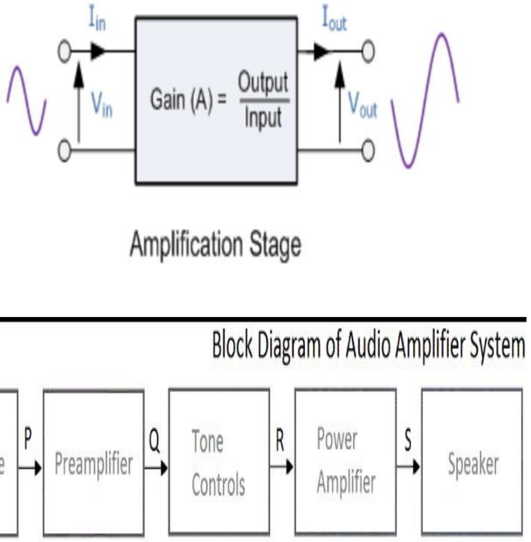


Figure 2.10: simple working of an audio power amplifier

2.5.1 Design Parameters

Key design parameters for audio amplifiers are frequency response, gain, noise, and distortion. These are interdependent; increasing gain often leads to undesirable increases in noise and distortion. While negative feedback actually reduces the gain, it also reduces distortion. Most audio amplifiers are linear amplifiers operating in class AB.

2.5.1.1 Class AB Operation

An alternate approach to overcome the cross-over distortion, is to use the AB amplifier. This approach means that the amplifier sacrifices a certain amount of potential efficiency for better linearity - there is a much smoother transition at the crossover point of the output signal. In this way, Class AB amplifiers sacrifice some of the efficiency for lower distortion. Accordingly, class AB is a much better option where a compromise between efficiency and linearity is needed.

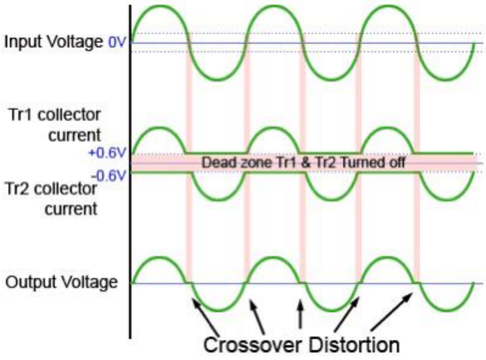


Figure 2.11 Class AB output waveform

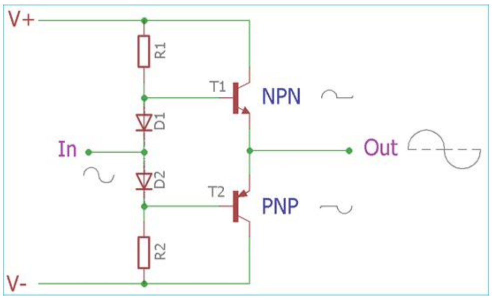


Figure 2.12: transfer curve of class AB and the push-pull configuration.

The Class AB Amplifier is a compromise between the Class A and the Class B configurations above. While Class AB operation still uses two complementary transistors in its output stage a very small biasing voltage is applied to the Base of the transistor to bias it close to the Cut-off region when no input signal is present. An input signal will cause the transistor to operate as normal in its Active region thereby eliminating any crossover distortion which is present in class B configurations. A small Collector current will flow when there is no input signal but it is much less than that for the Class A amplifier configuration. This means then that the transistor will be "ON" for more than half a cycle of the waveform. This type of amplifier configuration improves both the efficiency and linearity of the amplifier circuit compared to a pure Class A configuration.

2.5.2 Filters and Preamplifiers

Since modern digital devices, including CD and DVD players, radio receivers and tape decks already provide a "flat" signal at line level, the preamp is not needed other than as a volume control and source selector. One alternative to a separate preamp is to simply use passive volume and switching controls, sometimes integrated into a power amplifier to form an integrated amplifier.

2.5.3 Further Developments in Amplifier Design

Advancements in amplifier design have seen significant developments over the years, particularly in the transition from valve (tube) amplifiers to solid-state amplifiers. Initially, solid-state amplifiers were perceived to lack the exceptional audio quality associated with the best valve amplifiers. Audiophiles believed that the unique sound character of valve amplifiers was inherent to the vacuum tube technology itself. However, in 1972, a breakthrough occurred when Matti Otala uncovered a previously unnoticed form of distortion known as transitory intermodulation distortion (TIM), also referred to as slew rate distortion.

TIM distortion was observed during rapid increases in the output voltage of amplifiers. Notably, this distortion did not manifest itself during steady-state sine tone measurements, making it challenging to detect by design engineers prior to 1972. TIM distortion primarily resulted from limitations in the open-loop frequency response of solid-state amplifiers.

To address TIM distortion, Otala and other researchers proposed several solutions, including:

1. **Increasing Slew Rate:** Amplifiers were designed to have a higher slew rate, enabling them to respond more quickly to rapid changes in the input signal.
2. **Inserting Lag Compensation Circuits:** A lag compensation circuit was introduced in the input stage of the amplifier to counteract TIM distortion.

In modern high-quality amplifiers, the open-loop response typically extends to at least 20 kHz, effectively canceling out TIM distortion. However, it's worth noting that TIM distortion may still be present in many low-cost, consumer-grade amplifiers.

Another significant advancement in amplifier design was the introduction of the Baxandall Theorem, developed by Peter Baxandall in England. This theorem introduced the concept of comparing the ratio between input distortion and output distortion in an audio amplifier. This innovative approach provided audio design engineers with a valuable tool for assessing the distortion processes within an amplifier, leading to more refined and accurate amplifier designs.

2.5.4 Amplifier Gain

Measured at the output with the signal measured at the input. There are three different kinds of Then the gain of an amplifier can be said to be the relationship that exists between the signal Amplifier Gain, Voltage Gain, (Av), Current Gain (Ai) and Power Gain (Ap).

2.6 COMPONENTS REVIEW

2.6.1 Loudspeaker Design

The best loudspeakers recreate sound very accurately. In other words, they don't color the sound by changing it. The materials used in a loudspeaker design and fabrication are only part of the puzzle. The way those elements are assembled also affect the sound. If the design is not carefully integrated by convening the best-affordable components, it will negatively affect the desired acoustic output. If it is too wide, the sound can reverberate inside the cabinet, creating cross noise that interferes with the sound waves coming directly from the driver. If you don't brace the driver securely enough, it can rattle and create distortion.

The following components were put together and with transfer function, the resonant frequency range was measured. These include:

1. A low frequency transducer/driver of size 18 inches, made of Neodymium permanent magnet, a copper voice coil of 4-inches, and a diaphragm made of sturdy paper film. As the diaphragm size increases, the range of frequencies in the sub-low frequency spectrum that can be amplified increases.

2. A robust wooden housing made of birch plywood, critically designed in a bandpass configuration for maximum power output.
3. Trolleys for transportation.
4. A steel grill for protection.

2.6.2 Audio Power Amplifier Design

Every audio system uses audio power amplifier to amplify weak voice signals that are suitable to drive the speakers properly. Therefore, the audio power amplifier is considered an integral part of audio systems in the field of electronics (low power applications). In this research work, we have developed an audio power amplifier using two transistor types for low voltage amplification of our designed sub-low frequency loudspeaker (or subwoofer).

Components brought together include for the design if the audio power amplifier include:

Resistors

Resistors are passive electrical/electronic components used as current limiters within a circuit.

Figure 2.13: A series of resistors

Inductors

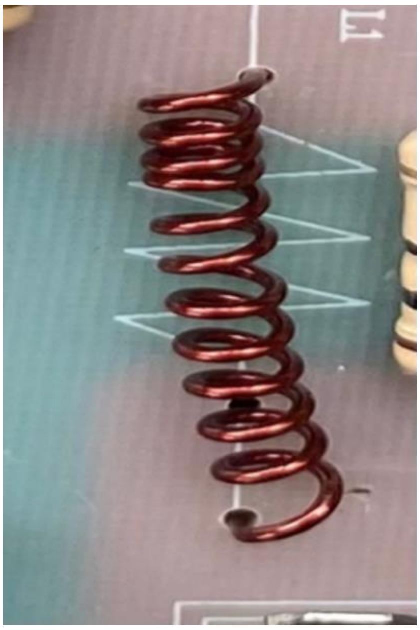


Figure 2.14: An inductor embedded in the main amplification panel

Capacitors

The function of a capacitor is to store an electrical charge. It will not allow a D.C. voltage potential to pass through it. It is used to pass A.C. through circuits without the D.C. reference causing problems.

Recall that the basic definition of a capacitor is two conductors separated by an insulator.

Figure 2.15: ceramic capacitors in the audio power amplifier module; large capacitors connected in series for voltage filtration

Transformer

A transformer is used to increase or decrease the voltage of an alternating current.

Basic Operation of a Transformer

In its most basic form, a transformer is simply a device that steps up or down voltage. In a step- up transformer, the output voltage is raised, while in a step-down transformer, it is lowered. The step-up transformer will decrease the output current and the step-down transformer will increase the output current such that the system maintains an equal input and output power.

Figure 2.16: A toroidal transformer

The transformer is essentially a voltage control device and is often used in the distribution and transmission of alternating current power. In 1831, Michael Faraday introduced the concept of a transformer, which was further developed by a number of other eminent scientists. The fundamental goal of utilizing transformers, however, was to maintain equilibrium between electricity produced at very high voltages and consumed at very low voltages.

Transformers come in several varieties and are categorized according to the following criteria: Depending on the voltage, Transformers are categorized based on voltage as:

- Step-up Between the power generator and the power grid, they are known as transformers. Input voltage is less than secondary output voltage.
- Step back Converting a high voltage main supply to a low voltage secondary output is the function of this type of transformer.

Depending on the type of core utilized: There are various core types that are used in transformers.

- **Transformer with an air core:** The air serves as the conduit for the flux between the primary and secondary windings. The windings or coils wound on the magnetically inert strip.
- **Iron core transformer:** Windings are wound on numerous iron plates that are layered on top of each other, creating the ideal linking path to produce flux. Based on the Winding Arrangement
- It will be an autotransformer with a laminated core and just one winding wound. The same coil is used for both the primary and secondary. In Greek, auto also means "self."

Working Principle of a Transformer

The transformer operates on the mutual induction and the faraday law of electromagnetic induction.

On the transformer core, there are typically two coils: a primary coil and a secondary coil. Strips are used to link the core laminations. On the transformer core, there are typically two coils: a primary coil and a secondary coil. Strips are used to link the core laminations. The mutual inductance of the two coils is very high. A changing magnetic flux results from the passage of an alternating current through the primary coil.

Parts of a Single-phase Transformer

A single-phase transformer's main components are;

1. Core

The winding is supported by the transformer's core. It also provides a magnetic flux flow path with low resistance. The winding is looped around the core, as indicated in the figure. A laminated soft iron core is present in a transformer to reduce losses. Variables including operational voltage, current, and power, among others, determine the composition of the core. The core diameter has a direct negative correlation to copper losses and a direct positive correlation to iron losses.

2. Windings

The copper wires that are wound over the transformer core are known as windings. Copper cables are employed because of

- Copper's high conductivity reduces transformer loss because resistance to current flow lowers as conductivity rises.
- Copper has a high degree of ductility, which allows for the production of very thin wires. The main and secondary coil windings are the two basic types of windings.
- The primary winding is the group of winding turns that receive supply current.
- Secondary winding: The set of turns of winding from which output is taken. The primary and secondary windings are insulated from each other using insulation coating agents.

3. Insulation Agents

Insulation is necessary in transformers to keep the windings separate and avoid short circuits. This facilitates mutual induction. Insulation agents have an effect on the stability and durability of transformers.

Transformer equations

EMF Induced In Primary & Secondary Windings:

$$E1 = 4.44\, fN_1\varphi_m = 4.44\, fN_1B_mA \quad \textit{Primary winding}$$

$$E2 = 4.44\, fN_2\varphi_m = 4.44\, fN_2B_mA \quad \textit{Secondary winding}$$

Where

- E_1 = EMF induced in primary winding
- E_2 = EMF induced in Secondary winding
- N_1 = Number of Turns in Primary winding
- N_2 = Number of Turns in Secondary winding
- f = Line frequency
- φ_m = Maximum Flux in Core
- B_m = Maximum flux density
- A = Area of Core

Bridge Diode Rectifier

This is the component of a power system that is responsible for the rectification of AC current to DC current since most components in electronic systems use low voltage.it is rated 50Amperes.

Figure 2.17: A 50Amp bridge diode rectifier

Transistors

The PNP Transistor is the exact opposite to the NPN Transistor device. Basically, in this type of transistor construction the two diodes are reversed with respect to the NPN type giving a Positive-Negative-Positive configuration, with the arrow which also defines the Emitter terminal this time pointing inwards in the transistor symbol.

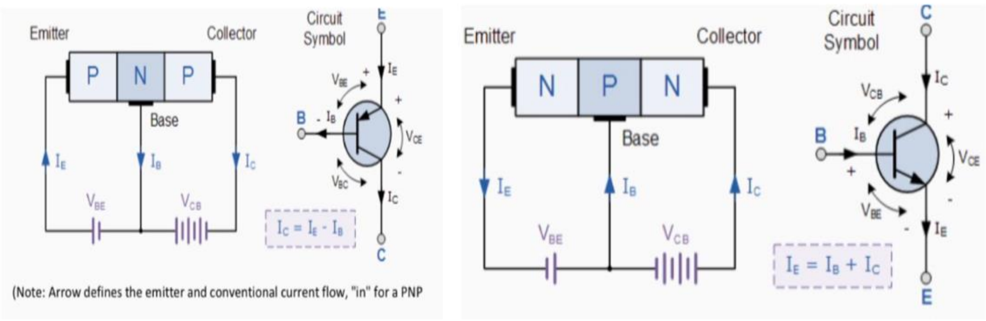
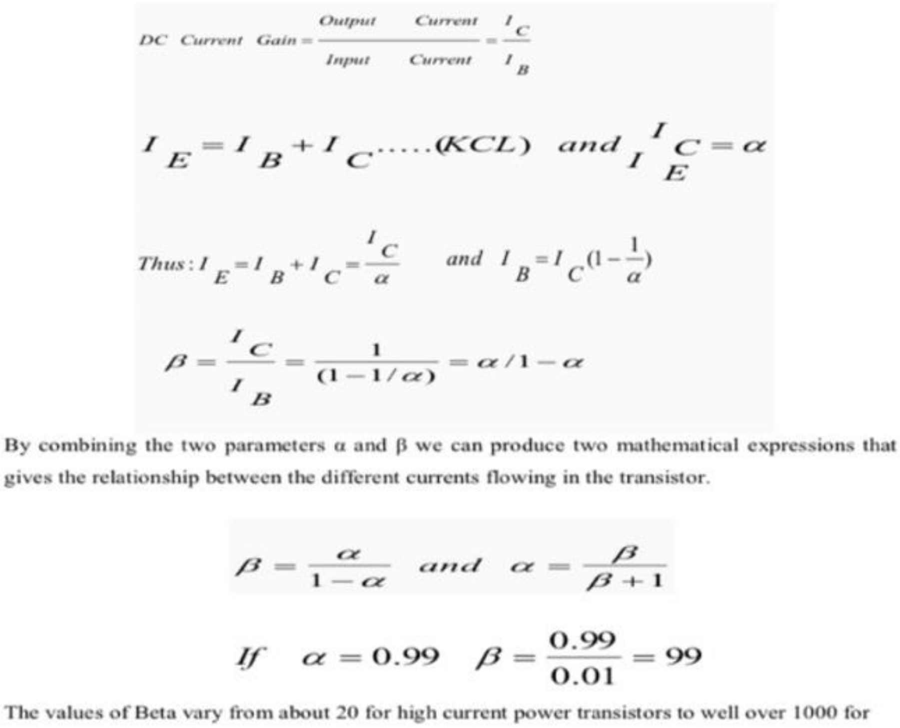


Figure 2.18: PNP transistor configuration circuit and NPN configuration circuit.



$$DC\ Current\ Gain = \frac{Output\ \ Current}{Input\ \ Current} = \frac{I_C}{I_B}$$

$$I_E = I_B + I_C .....(KCL) \quad and \quad \frac{I_C}{I_E} = \alpha$$

$$Thus: I_E = I_B + I_C = \frac{I_C}{\alpha} \quad and \quad I_B = I_C(1 - \frac{1}{\alpha})$$

$$\beta = \frac{I_C}{I_B} = \frac{1}{(1 - 1/\alpha)} = \alpha / 1 - \alpha$$

By combining the two parameters α and β we can produce two mathematical expressions that gives the relationship between the different currents flowing in the transistor.

$$\beta = \frac{\alpha}{1 - \alpha} \quad and \quad \alpha = \frac{\beta}{\beta + 1}$$

$$If \quad \alpha = 0.99 \quad \beta = \frac{0.99}{0.01} = 99$$

The values of Beta vary from about 20 for high current power transistors to well over 1000 for

Component list for the audio power amplifier design and fabrication include

Transistors

2 x C2073(NPN)

2 x A940 (PNP)

10 x 2SA1943(PNP)

10 x 2SC5200(NPN)

Transformer

A toroidal power transformer

2000VA, 220V

Capacitors

8 x 15000microfarads/80V

4 x 220pico-farads(ceramic)

4 x 680pico-farads(ceramic)

2 x 104, 0.1micro-farads/250V (polyester)

5 x 104, 0.1micro-farads/100V(polyester)

Resistors

12 x 10 Ohms, 1/4 watts

4 x 10 Ohms, 2Watts

10 x 100 Ohms,1/4Watts

4 x 330 Ohms,1/4Watts

5.6KOhm, 1Watt

Choke resistors

12 x 0.47Ohms, 5 Watts

4 x 330Ohms, 1/4Watts

4 x 100Ohms, 1Watts

4 x 1KOhms, 1/4Watts

4 x 18KOhms, 1/4watts

2 x 33KOhms, 1/4Watts

2 x 56KOhms, 1/4Watts

2 x 5KOhmss, 5Watts

Integrated Circuit(IC)

UA741(1)

Miscellaneous

10 mica insulators

IN4007(2)

2 aluminum heat sinks

Inductor

6 micro-henry

Diode

50A bridge diode rectifier

15V, 1 Watt zener diode

LEDs

1 Red,

1 yellow

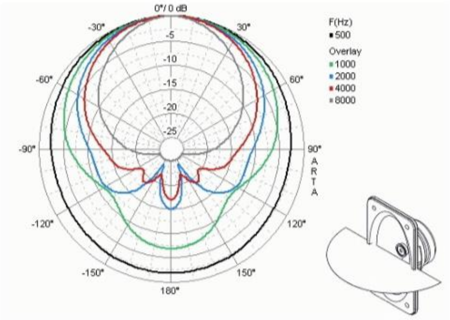
2.6.3 Determination of System Response (Amplitude & Frequency)

The aim of this work was to study the psychoacoustic determined performance requirements for the design of a loudspeaker system for high-quality reproduction of low-frequency sound in any space. Dual-channel methods provide a relative measurement (input vs. output), and can help to answer questions like “What is the crossover frequency in our system,” “How much boost or attenuation is there at 1 kHz,” or “When is energy from my main speaker system arriving at the measurement mic?” Sub-low frequencies are omnidirectional post amplification in diverse spaces, but the determination of their directivity and response of the wavefronts produced in combination with the boundaries of these spaces in which amplification is done is determined using a series of transfer function measurements.

The diligent reproduction of 45Hz-125Hz by our system confirms the accuracy of our design so a measurement of the system as a whole is done.

The third party softwares include:

- Rational Acoustics SMAART v8, and
- Open Sound Meter(OSM)
- EASE Focus 3 a three-dimensional, acoustic simulation software for the
- configuration and modeling of loudspeakers, will be used to determine the polar response or coverage pattern of our design.



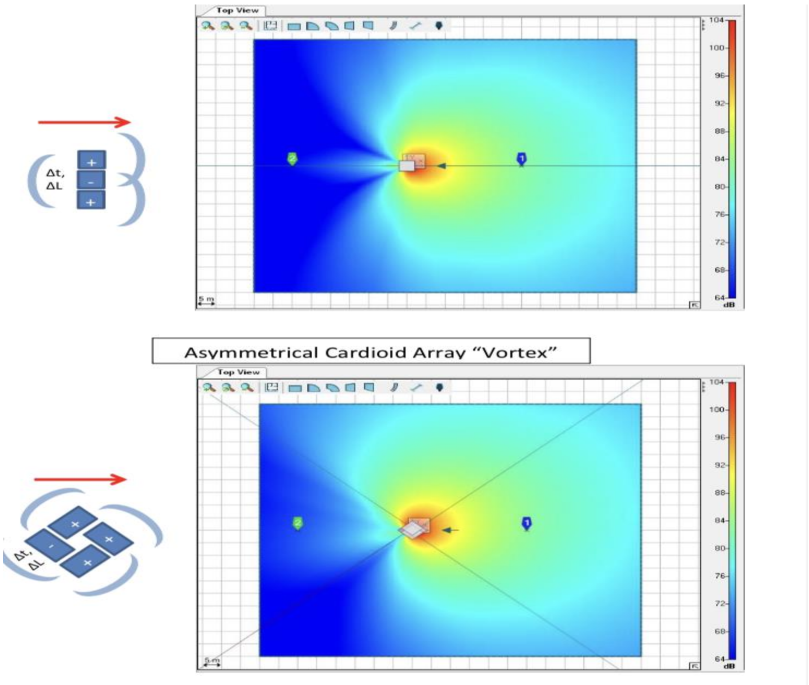


Figure 2.19: Ease focus 3 showing the polar response of a sub low frequency cardioid configuration within the audience area of a space.

2.6.4 System Measurement Equipments

The determination of the overall response of the system incorporates the integration of a measurement system on ground, from the beginning of the building process through the finished work.

The equipments include:

1. A semi anechoic chamber needed to isolate the source from the outside world during some isolated measurements
2. A series of Behringer ECM8000 electret flat frequency measurement microphones
3. An Analog-to-digital converter used in transfer function measurements and Real Time Analysis
4. A laptop computer running Third party softwares for transfer function measurements

5. A laptop computer running third party softwares for acoustic behavior prediction
6. A power amplifier
7. A mini electronic mixing console for small voltage amplification
8. Interconnection transmission cables

These measurements are done to unveil the following:

- The frequency response of the system in free space
- The amplitude response of the system in any space
- The maximum excursion of the system
- The sound pressure level (dab SPL) of the system
- The phase response of our system
- Corner loading response of the system

2.6.5 Target Curve

While there is a consensus that a neutral loudspeaker has a flat anechoic frequency response, several researchers have been commenting on the suitability of an all-pass target for the in-room frequency response. Some researchers suggest a down sloping character across the full or at least a part of the audio band. This seems to be largely motivated by listener preference while little attention has been paid to solving the problem of the ‘circle of confusion’. This refers a self-referenced system where existing recordings are used to evaluate the room sound.

2.6.6 One or More Measurement Locations

The sound pressure measurable in a room at a single microphone location is relatively local, and large variations in the pressure can be seen when the microphone location is moved. This is particularly evident at high frequencies where large and very local comb filtering effects can happen because of acoustic reflections. That effect is usually reduced by time domain windowing the impulse response estimate and by applying in-frequency smoothing with a sliding variable-width averaging window to the frequency response. These techniques are usually able to sufficiently reduce acoustic comb filtering and tend to reduce spatial locality of a measurement, thereby rendering the measurement usable for the practical purposes of evaluating room- induced sound colorations. Increasing the number of microphone positions can provide a more complete picture of room acoustics. Such measurements estimate the power output of the loudspeaker in the room and using that for system equalization particularly at low frequencies.

2.6.7 Room Selection Criteria

A variety of listening rooms were included in the study. Most rooms lack refined acoustic design or extensive acoustic treatment. Room selection criteria also included repeated availability, low background noise, acceptable reverberation time, and reasonably low level and acceptable direction of early reflections at the listening position.

CHAPTER THREE

METHODOLOGY

3.0 CONSTRUCTION OF THE 2000WATT AUDIO POWER AMPLIFIER

3.0.1 TRANSISTOR SPECIFICATIONS

From the specification sheet, the power rating of each transistor = 100Watts

Number of transistors

10x 2SA1943(PNP)

10x 100 = 1000watts

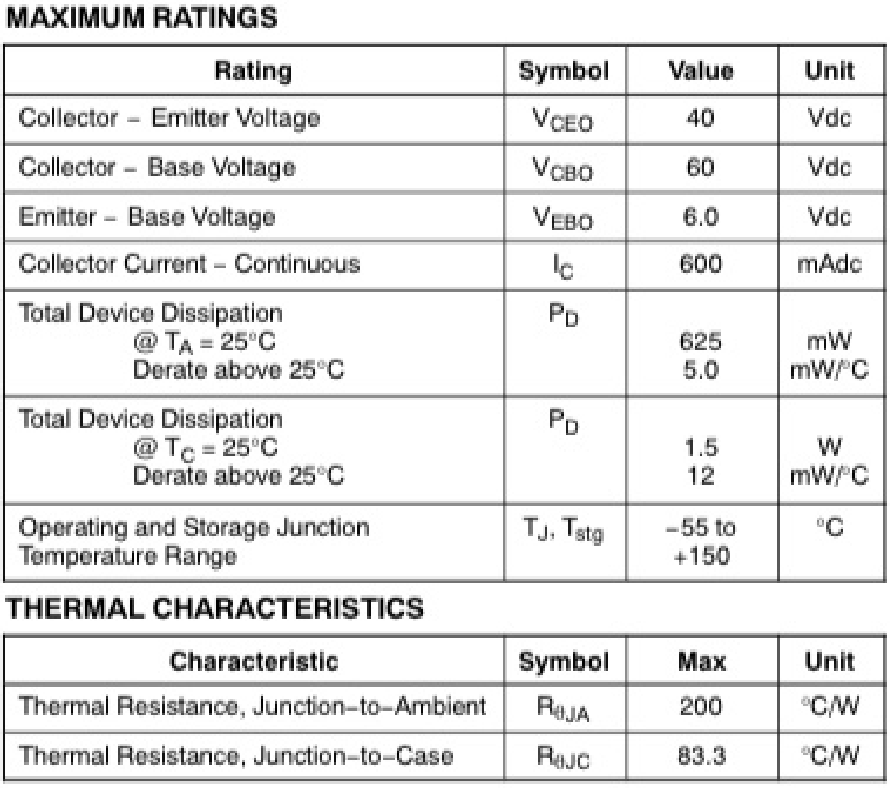
10x 2SC5200(NPN)

10x 100 = 1000watts

Total power rating of transistors =

1000 x 2(channels)= 2000watts

Specification sheet for the 2SA1943 transistor and the 2SC5200 transistors.

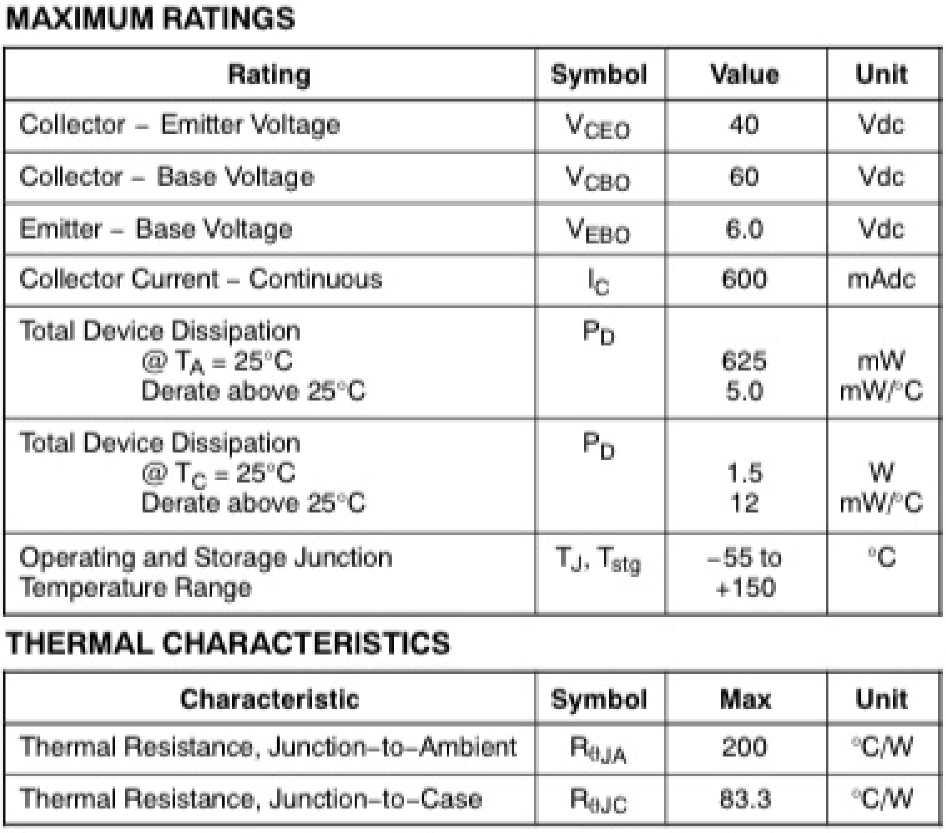


MAXIMUM RATINGS

Rating	Symbol	Value	Unit
Collector – Emitter Voltage	V_{CEO}	40	Vdc
Collector – Base Voltage	V_{CBO}	60	Vdc
Emitter – Base Voltage	V_{EBO}	6.0	Vdc
Collector Current – Continuous	I_C	600	mAdc
Total Device Dissipation @ $T_A = 25°C$ Derate above 25°C	P_D	625 5.0	mW mW/°C
Total Device Dissipation @ $T_C = 25°C$ Derate above 25°C	P_D	1.5 12	W mW/°C
Operating and Storage Junction Temperature Range	T_J, T_{stg}	−55 to +150	°C

THERMAL CHARACTERISTICS

Characteristic	Symbol	Max	Unit
Thermal Resistance, Junction–to–Ambient	$R_{\theta JA}$	200	°C/W
Thermal Resistance, Junction–to–Case	$R_{\theta JC}$	83.3	°C/W



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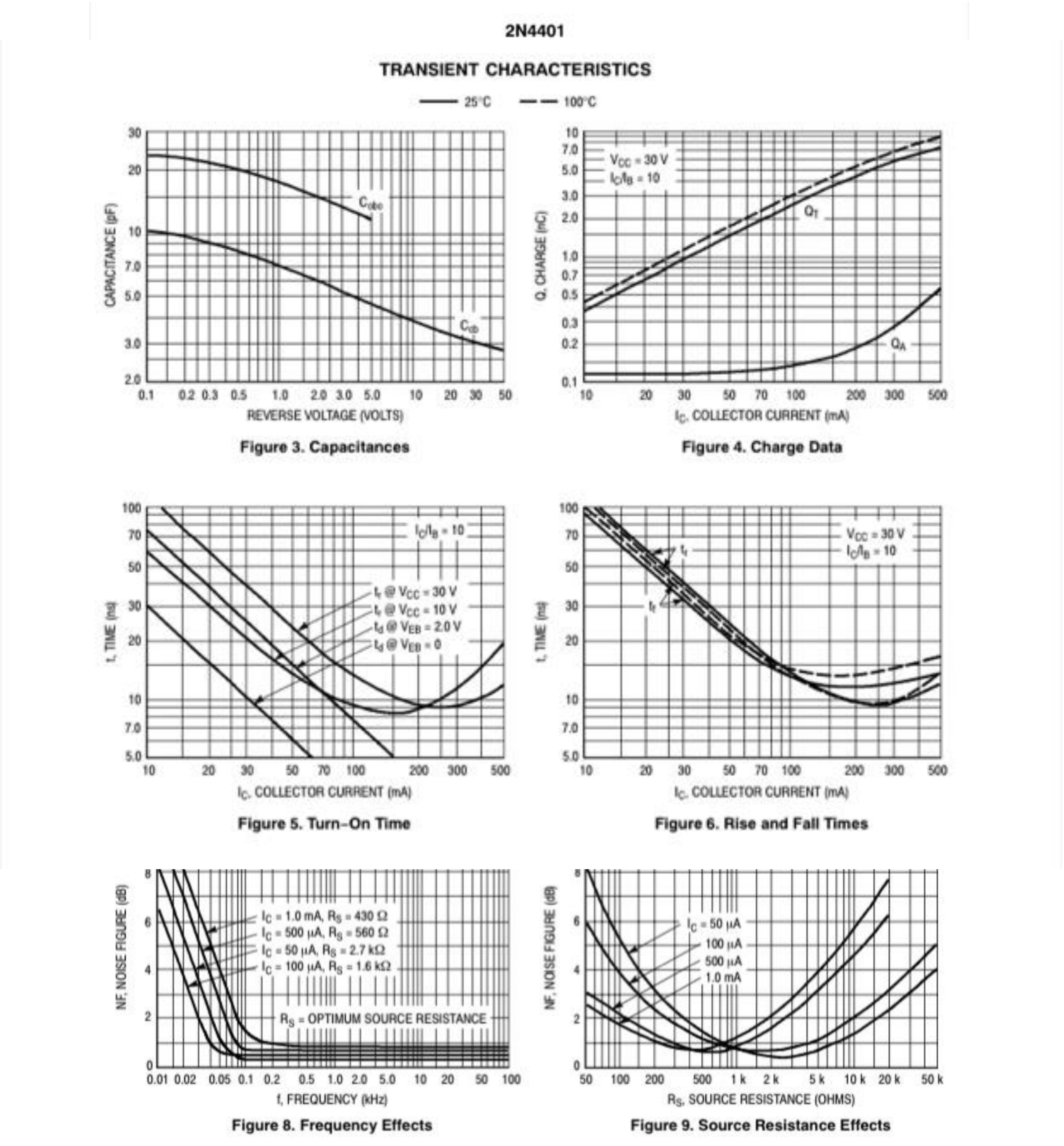

2N4401

TRANSIENT CHARACTERISTICS

Figure 3. Capacitances

Figure 4. Charge Data

Figure 5. Turn–On Time

Figure 6. Rise and Fall Times

Figure 8. Frequency Effects

Figure 9. Source Resistance Effects

Figure 3.1: specification sheets for the 2SA1943 and the 2SC5200 transistors respectively.

3.1 Design Calculations for Transformer:

3.1.1 DETERMINATION OF POWER SUPPLY AND FUSE RATING

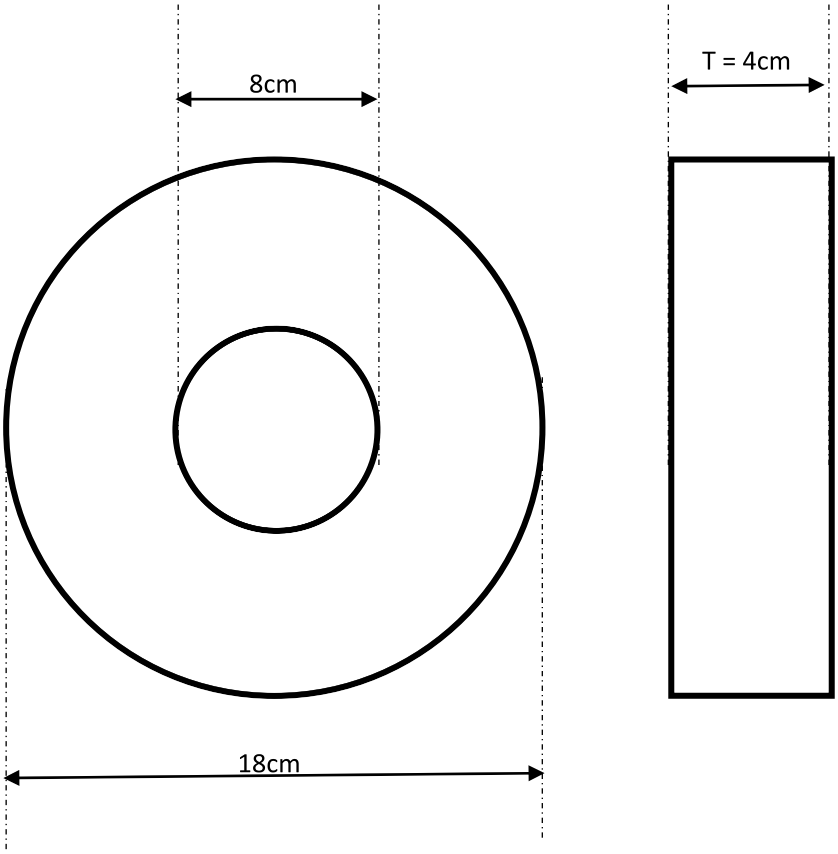

Figure 3.2: A toroidal Transformer

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Which works on the principle of Faraday's electromagnetic law

$$E = 4.44BANf$$

$$B = \frac{\emptyset}{A}$$

Area

L internal section = 8cm = 8 x 10^{-2}m

h exteral section = 18cm = 18 x x10^{-2}m

= (18 – 8) 10^{-2} = 10 x 10^{-2} = 101cm

Area = $40cm^2$

Error! Reference source not found.= $20cm^2$ = 20 x 10^{-2} m^2

$$f = 50Hz$$

$$\frac{N_1}{N_2} = \frac{V_1}{V_2} = \frac{I_2}{I_1}$$

For Iron core (toroidal) = 1.73 Tesla

$$T_{pv} = \frac{N}{E} = \frac{1}{4.44BAf}$$

$$T_{pv} = \frac{10^4}{4.44x1.73x20x50} = \frac{10000}{4.44x1.73x20x50} = 1.30\ t/v$$

For sound transistor circuit using 70v

$$N = 1.30\ x\ 70\ = 91t/v$$

For Number of turns for primary

$$\frac{N_1}{N_2} = \frac{E_1}{E_2} = \frac{I_2}{I_1}$$

$$N_2 = \frac{286}{220}\ x\ 70 = 91\ turns$$

For 15 VDC, LM358 Integrated circuit

$$N = 1.30\ x\ 15 = 19.5\ turns$$

For 12VDC cooling fan,

$$N = 1.30\ x\ 12 = 15.6\ \ turns$$

For 220 AC Fan,

$$N = 286\ turns$$

Fuse rating for sub circuits

$P = IV$

Power rating for transformer = 2000VA

$I = ?$

V = 220v

$$I_1 = \frac{p}{v} = \frac{2000}{220} = 9.01A$$

Fuse rating multiplication factor = 9.01

Using **Error! Reference source not found.**

$$I_{ph} = \frac{E}{4.44fN_{ph}} x\ 10^3 = \frac{70\ x\ 10^3}{4.44\ x\ 50\ x\ 91} = 3.5A$$

Fuse rating for

$3.5\ x\ 0.732 = 2.56A$

For IC LM358 over voltage

Using **Error! Reference source not found.**

$$I_{ph} = \frac{15\ x\ 10^3}{4.44\ x\ 50\ x\ 19.5} = 3.5A$$

Fuse rating = **Error! Reference source not found.**

For 12VDC cooling fan

$$I_{ph} = \frac{12\ x\ 10^3}{4.44\ x\ 50\ x\ 15.6} = 3.465A$$

Fuse rating **Error! Reference source not found.**

For AC fan

$N = 286\ turns$

$$I_{ph} = \frac{220\ x\ 10^3}{4.44\ x\ 50\ x\ 286} = 3.87A\ \cong 4A$$

Furse rating **Error! Reference source not found.**

Fuse rating for 15VDC pre amplifiers – JRC4558

$1.30\ x\ 15 = 19.5turns$

$$I_{ph} = \frac{15\ x\ 10^3}{4.44\ x\ 50\ x\ 19.5} = 3.465 \cong 4A$$

Fuse **Error! Reference source not found.**

<u>Primary side conductor calculations</u>

Number of primary turns Tp = t/volt x Vp

= 1.30 x 220

=286 turns

Area of primary conductor = 40square cm

= 40

conductor weight calculations = Diameter x gauge

Diameter = 1.26mm

Conductor gauge = 16

1.26 x 16 = 20.64 grams

Weight of copper conductor

1 meter of conductor = 20.64grams

89.60 meters = 1848.344grams

<u>Secondary side copper conductor calculations</u>

conductor weight calculations = Diameter x gauge

Diameter = 1.83mm

Conductor gauge = 13mm

1.83 x 13= 23.78 grams

Number of turns = 156

Total conductor length = area x number of turns in secondary

Total conductor length = 20 x 156

= 3120/100

=31.20meters

Weight of copper conductor

1 weight of conductor = 23.69 grams

31.2 meters length = 739.13 grams

Resistance of the secondary side

diameter of sec. conductor = 1.83×10^{-3} m

area = $3.142 \times (1.83 \times 10^{-3})^2 / 4$

Area of sec. conductor = 2.62×10^{-6} square meter

R= resistivity x length / area

Resistivity of copper = 1.7×10^{-7}

Area of sec. conductor = 2.62×10^{-6} square meter

Length of secondary copper conductor = 31.2meters

$R = 1.7 \times 10^{-8} \times 31.2 / 2.62 \times 10^{-6}$

R = 0.2024 ohm

magnetic flux = magnetic flux density x area

= 1.73 x 0.002

=0.00346 weber

EFFICIENCY OF THE TOROIDAL TRANSFORMER

Efficiency (%) = output / input = $V_sI_s\text{Cos}Ø / V_sI_s\text{Cos}Ø + I^2 R$

$V_s = 120V$

$V_p = 220V$

$I_s = 16.67A$

$I_p = 9.19A$

Power factor(cosØ) = 0.85

In toroidal transformers, the iron losses are very small compared to the losses due to copper.

Thus, copper loss>>>>iron loss

$= 120 \times 16.67 \times 0.85 / (120 \times 16.67 \times 0.85) + (16.67)^2 \times 2.65$

Efficiency = 0.9679 %

3.2 POWER CIRCUIT MODULE (PCM)

This audio power amplifier project is based on the TOSHIBA 2SC5200(NPN) transistors and TOSHIBA 2SA1943 (PNP) transistor components from TOSHIBA, Japan. It is able to deliver peak power up to 2000W using a 4 ohm load and dual 70 Volts DC power supplies. It is designed to operate with minimum external components with current limit and thermal shutdown protection features. Other features include high gain, fast slew rate, wide power supply range, large output voltage swing and high current capability.

The system components were integrated to get a power amplifier specifically made for a subsonic loudspeaker load. Below are the excerpts of the integration of the components/parts.

The schematic below shows how the +70V DC and -70V DC are obtained. In order to provide power supply for the mono amplifier, a power transformer rating of 2000VA with centre tapped secondary winding is used.

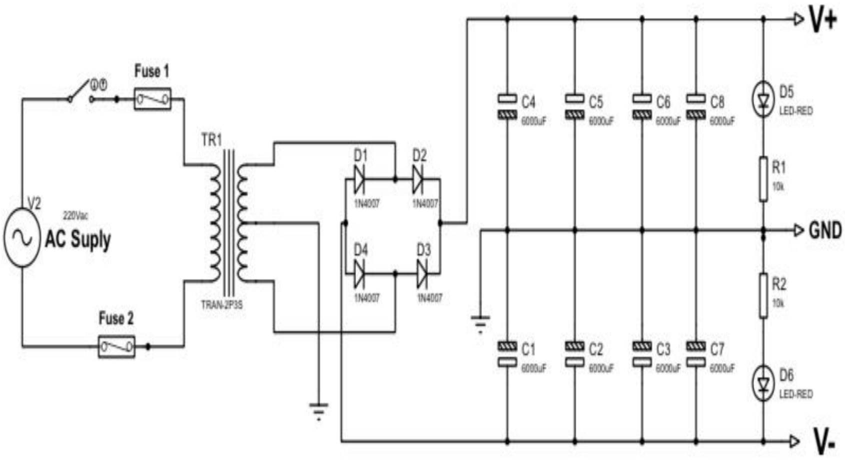


Figure 3.3: power supply circuit diagram

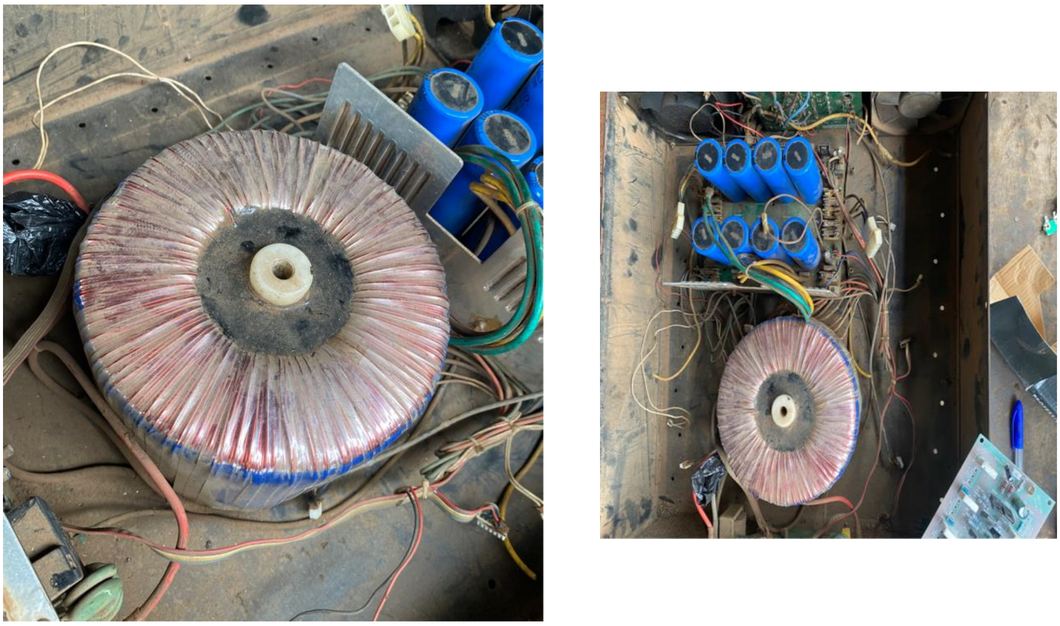


Figure 3.4: integration of the components of the audio power amplifier from the step down transformer to the filtration circuit.

The stepped down secondary output of the transformer is rectified by using a diode bridge rectifier rated 50Amps together with 8 electrolytic capacitors connected in series, each with 15000 micro-Farads/80WV, to smoothen the ripple voltage.

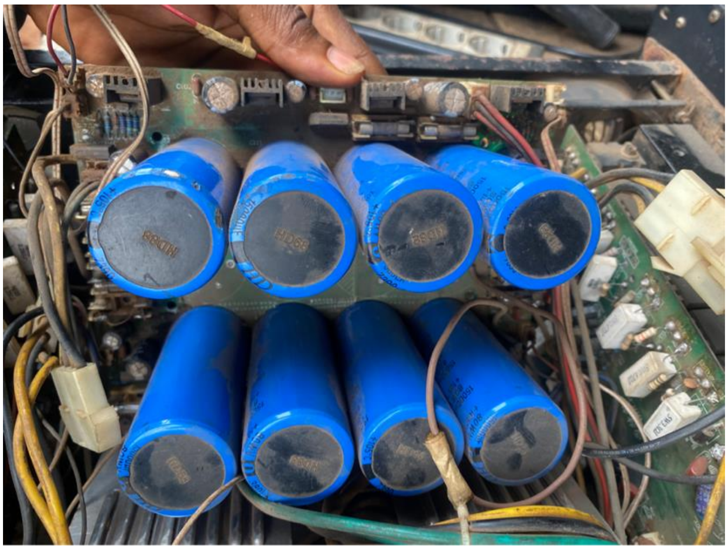


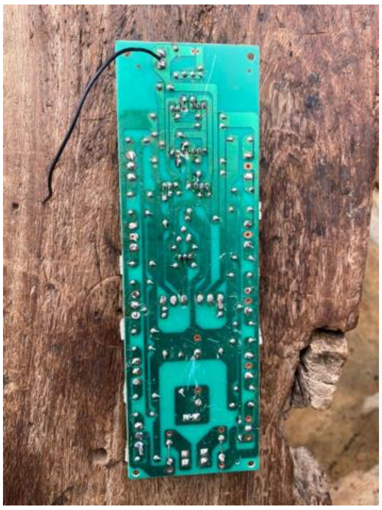
Figure 3.5: filtration capacitors in series

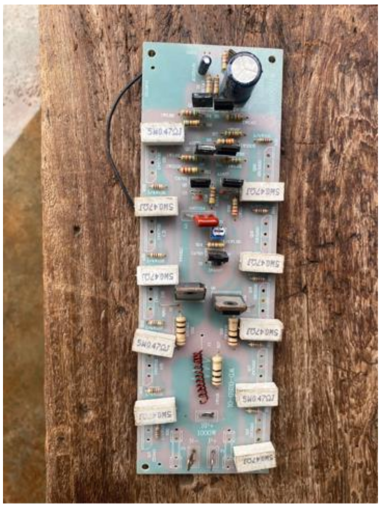
The light emitting diodes(LED) are indicator lights which tells when the system is either ON or OFF.

The output is connected to the audio power transistor-amplifier main circuit.

3.2.1 Audio Power Amplifier Module Connection

The +70V and -70V DC power supply were connected to the audio amplifier module through an integrated circuit with the peripheral devices shown in the schematic below.





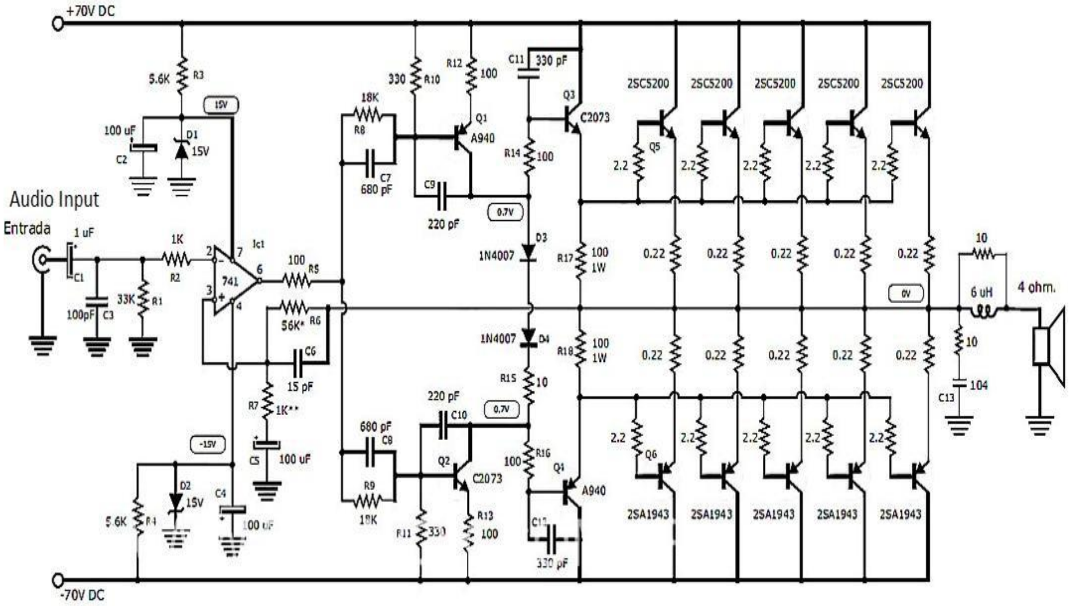


Figure 3.6: the PCB design power amplifier module pre-installation and the audio amplifier circuit diagram

The audio input signal to be amplified is coupled to pin 2 through the resistor R2 and an electrolytic capacitor C1. On board transistors C2073(4) and A940(4) boost the audio signal. Through the soldered connection points on the amplifier module, the positive DC current flows through the 2SC5200(NPN) transistors while the negative flows through the 2SA1943(PNP) transistors. After amplification, the collector terminal collects the signals and the emitter transfers the signals to the load(loudspeakers) input terminals through the small GP connectors. The output signal at pin 3 of UA741 can be used to directly drive a 4 ohm loudspeaker load.

An aluminum heat sink with a thermal resistance rating of 1.4 Celsius/Watt was used during the amplification of signals or else the amplifier module will be cut-off from operation due to the heat that will build up during the operation of the amplifier.

Figure 3.7: an aluminum heat sink used for the dissipation of heat from the transistors (22 x10×2.5 centimeters)

Transistors heat up quickly due to their high switching frequency and collector current, hence they are always used in conjunction with a heat sink. It's worth noting that the heat sink will also serve as the transistor's collector pin, thus it should be separated from the rest of the circuit. If this is not done, the negative rail will be shorted to ground. Finally, the heat is expelled from the system by an AC cooling fan connected directly to the supply power source of 220V. The second cooling fan is a DC fan which has its voltage rectified by a diode to 12Volts for it to be operational.

\

Figure 3.8: ac fan with ratings:220V 50/60Hz;80W

2000W Class AB Power Amplifier Specifications:

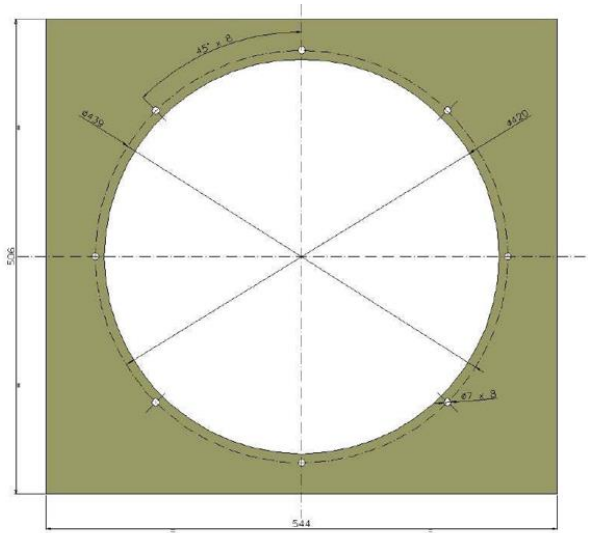
- Max. Output Peak Power: 2000 Watt Peak @ 4 Ohms Loudspeaker
- Max. RMS Output RMS Power: 1000 Watt RMS @ 4 Ohms Loudspeaker
- Max. Bridged RMS Output Power: 2000 Watt RMS @ 8Ohms Loudspeaker
- RMS Output Magnitude: 44.7v RMS
- Peak Output Magnitude: 65v Peak
- Frequency Response: 20Hz-20KHz
- Max. Output Gain: 22dB
- Input: 100mV Peak
- DC Power Supply: 70v DC (Dual Supply)

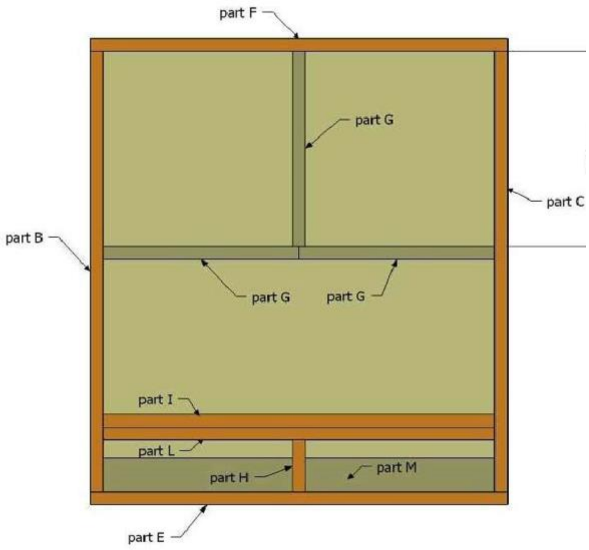
3. 3 LOUDSPEAKER DESIGN AND FABRICATION

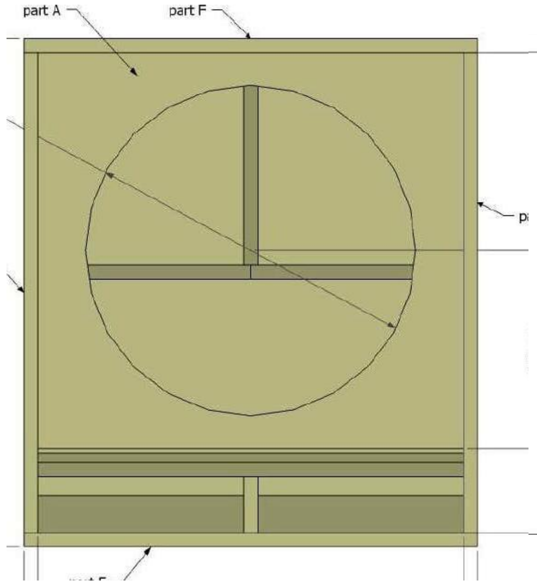
3.3.1 The design process

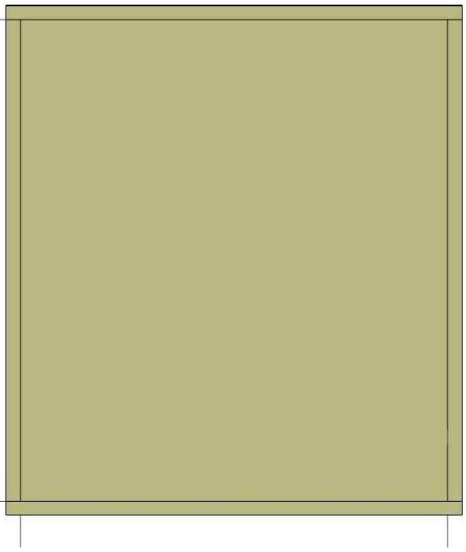
The bandpass design configuration was the preferred choice since the ported section aids in the increase of the sound pressure level (dB SPL) reaching the measurement system or in real cases, listeners.

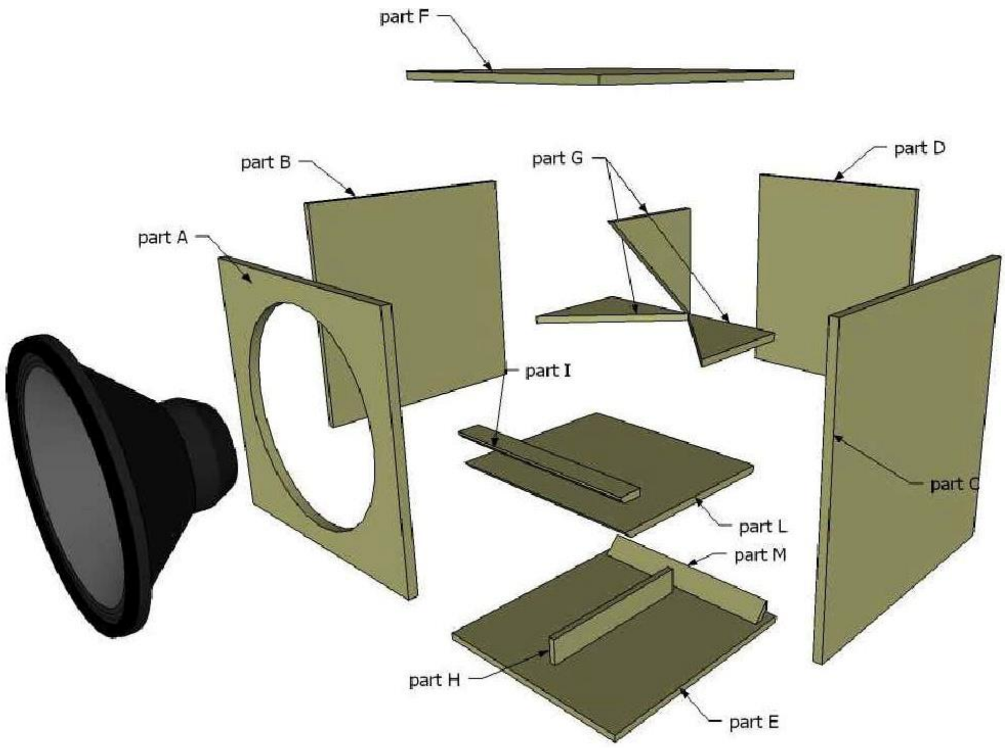
Below entails a 3-D drawing of the proposed design before the buildup process commenced. A design software DIALUX was used in the modeling process prior to the actual fabrication.

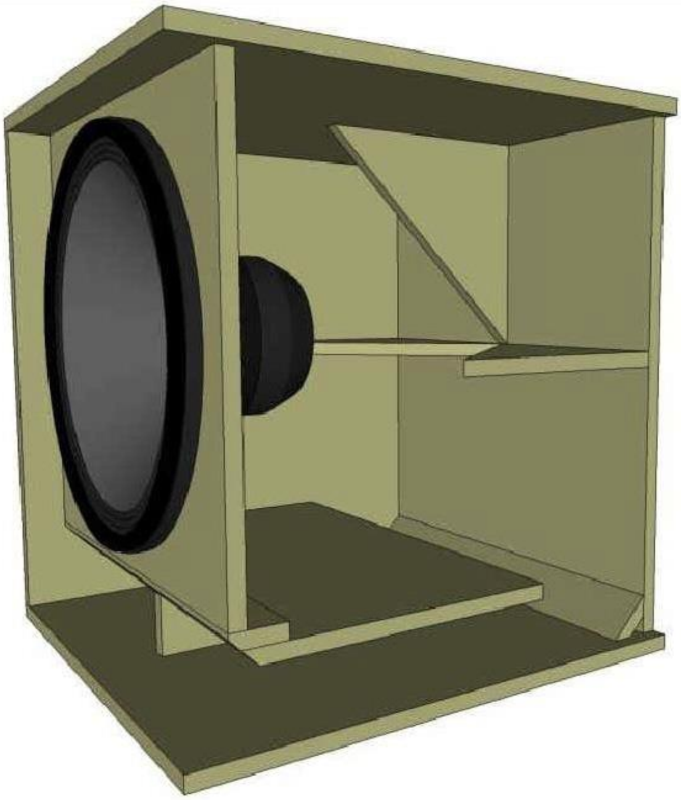

45° x 8
ø439
ø420
ø7 x 8
506
544


part F
part G
part C
part B
part G
part G
part I
part L
part H
part M
part E


part A
part F
pi




part F
part B
part G
part D
part A
part I
part C
part L
part M
part H
part E



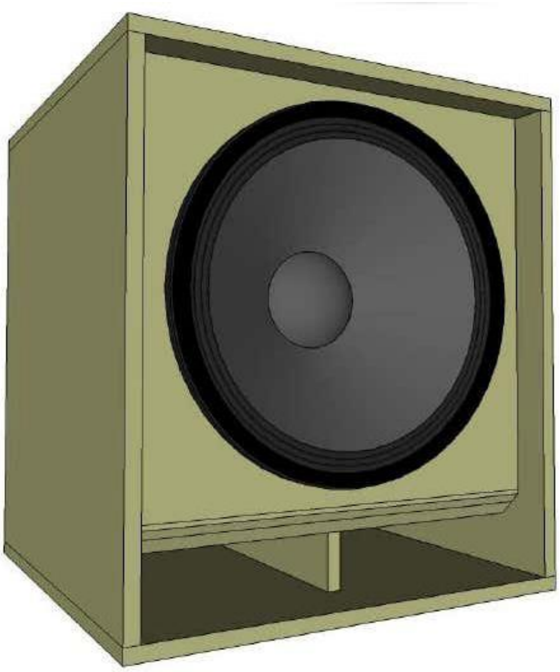


Figure 3.9: loudspeaker modeling prior to construction.

3.3.2 FABRICATION PROCESS

Birch plywood was the go to choice for the fabrication of the cabinet which houses the 18” transducer. The specification is a bandpass configuration which aims at increasing the sound pressure level of the amplified output signal.

Prior to the cutting and merging of the woodwork, the following specifications were used.

Table 3.1 specifications prior to buildup of cabinet

Width	563mm	(W)
Height	683mm	(H)
Depth	721mm	(D)
transducer port seat	439mm	(H)
Ports	266mm	bandpass configuration

By specification and detailed cutting, the diverse parts according to the modeled design were brought together using strong permanent adhesive materials. Top bond was chosen because of its fast and long lasting adhesive characteristics. Applications were on the edges of the already cut wooden parts. They were merged and allowed to dry.

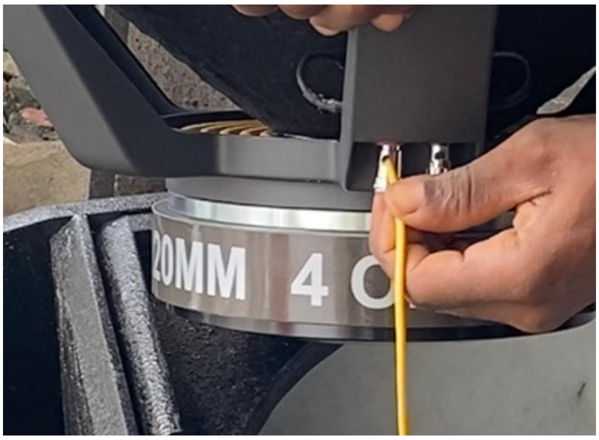
Figure 3.9: cabinet after cutting to size and applying adhesive.

Next was the spraying with urea paint to aid protection of the cabinet as well as to give it an appealing look.

Figure 3.10: cabinet right after painting and left to dry.

Finally, after the loudspeaker had dried up, the last phase was ready to be done which was embedding/housing the 18” transducer within the cabinet which entailed the soldering of the transducers electrical input contact points to the input terminals, which in turn connects the loudspeaker to the outputs of the designed audio power amplifier. A protective grill with an attached thin black foam was also put in place to secure the transducer’s diaphragm from unnecessary handling and sticky liquids including rainfall.

00:01



20MM 4 O


OHM
4000

Figure 3.11: buildup process prior to the mounting of the transducer

3.3.3 Mounting of the 18” Transducer

The conversion of the audio signal in the form of voltage (electrical energy) into sound energy is done by the transducer. The TITANIC 18” transducer was used here. This was held into place, within the wooden(acoustic) structure by screws and washers.

Figure 3.12: the transducer being held into place by screws.

This structure is able to hold and counteract the weight of the transducer due to a greater weight. This is needed because during the excursions(displacements) of the diaphragm when a test signal is played through the system, if the wooden cabinet's structure is unable to hold the transducer in place, a rattling noise is heard which in turn can compromise the response in the listening area. And in some cases it could even be audible to the point where it overrides the test signal.

Figure 3.12: the finished work post the embedded transducer.

3.4 SOUND SYSTEM MEASUREMENTS

3.4.1 Measurement Locations

A transfer function measurement is an efficiency test to verify the identical nature of two signals. The primary question a transfer function measurement asks is to confirm how close the amplified signal in the acoustic realm is to the generating signal which was electrical. As the variables within the system increases, the efficiency drops steadily.

The responses were measured with a BEHRINGER ECM8000 flat frequency measurement microphone and a portable computer running the acoustics measurement platform, Smaart V8.

Five (5) measurement points (microphone positions) were distributed to cover uniformly the seating area which allows for a good representation of the audience listening area. Measurement points were separated at least 5 meters away of each other, making a total of 1 in each point in the listening or measurement spaces respectively. The height of the microphone capsule was two centimeters from the floor. This was done to reduce the number of reflections that would have negatively colored our required response target during measurements.

3.4.2 Capturing the Data

The measurement microphone used in taking measurements was the Behringer ECM8000 flat frequency condenser microphone, with an omnidirectional polar pattern. This means the response of a system can be picked up at all angles about the microphone without favoring any specific region of the frequency range. Below shows the specification sheet of the microphone.

MEASUREMENT CONDENSER MICROPHONE
ECM8000 Technical Specifications
Ultra-Linear Measurement Condenser Microphone

FEATURES

- Ultra-linear condenser microphone for measurement and recording application
- Perfectly suited for room equalization application plus high-resolution studio recording and live applications
- Exceptionally flat frequency response and ultra-high sound resolution
- Evenly weighted, true omnidirectional pattern
- Works with phantom power from +15 to +48 V
- Perfect for use with the BEHRINGER ULTRACURVE or any other analyzer
- Ultra-low noise transformerless FET input eliminates low-frequency distortion
- Gold-plated 3-pin XLR connector for highest signal integrity
- Swivel stand mount and transport case included
- High-quality components and exceptionally rugged construction ensure long life
- Conceived and designed by BEHRINGER Germany

SPECIFICATIONS

Type	electr. condenser
Polar Pattern	omnidirectional
Impedance	200 Ohms
Sensitivity	70 dB
Freq. Response	20-20000 Hz
Connector	gold-plated XLR
Phantom Power	+15 to +48 V
Weight	136 g

behringer.com

Figure 3.14: the specification sheet showing the frequency response of the measurement microphone and the polar response respectively.

Each microphone position represents a full acoustic measurement, so a detailed explanation which shows the positions of the microphones relative to the loudspeaker and each other respectively was drafted.

3.4.3 On-site Measurements

3.4.3.1 Generation of Test Signals or Stimuli

The test signals employed include:

- Pink noise resonating from 20Hz-20KHz was generated within the measurement software which is SMAART(System Measurement And Acoustic Real Time tool).
- Music tracks of different genre.

These test signals are used specifically because of their Low frequency content which is the desired signal needed to excite our system to vibrate.

The parameters of the test signals are as shown as follows:

- music: CD version, frequency response:20Hz-20KHz; sample rate of 44.1KHz, 16 Bit
- Pink noise: frequency response: 20Hz-20KHz; gain:-12bB

3.4.3.2 Transfer function settings

The transfer function measurements were carried out using the following settings:

- reference signal: pink noise
- Measurement signal: the measurement microphone
- averaging:4 seconds
- Phase smooth: 1/24 octave
- Magnitude smooth:1/48 octave

- room temperature: 23 degree celsius

The output signal or device under test was the signal being amplified and made louder by the loudspeaker, which travels through the air and eventually hits the measurement microphone at its measurement position.

The input signal or reference signal was the test signal generated by measurement software, which was an electrical signal in the electrical/electronic domain.

Below shows a detailed pictorial description of the measurement process.

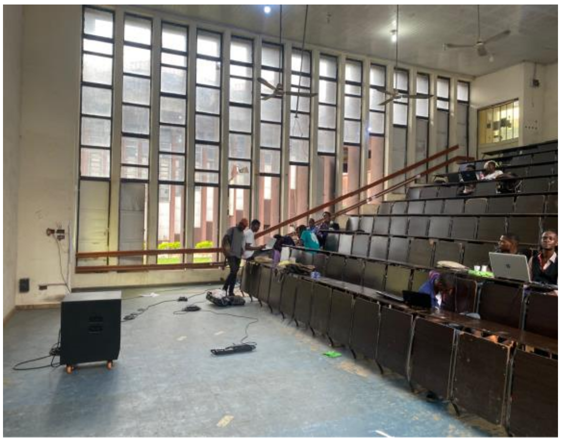
The measurement microphone was placed on a case to prevent the uneven movement where it may roll over from its designated position, and also for protection from walking persons within the area since we could not isolate the room from other users.

A closed loop configuration was used for the reference signal which was fed into one of the input channels of the Analog-to-digital converter from the output port on the A/D. This was a transfer function measurement on its own which confirmed that the A/D was not interfering with the characteristics of the test signal but just allowing the signal to pass through.

Figure 3.15: measurement microphone suspended at a height of 2 centimeters from the floor.

Figure 3.16: setup of measurement software on the laptop device.

The setup process included a complete installation of the measurement system, specifically resting upon the transfer function measurement software which had to be installed from the company’s website. A demo version of the software was installed which came with its own limitations (which have been discussed in the concluding part of the project in Chapter five).



(a)

(b)

(c)

Figure 3.17: (a) direct axis measurement at 1 meter from cabinet; (b) discussion by team members during measurement process;(c) device under test centrally located in the space.

The loudspeaker cabinet was centrally position within the room in order for us to get an even response as the microphone was moved and in turn, the changes in measurement positions.

Another measurement position was taken seven(7) meters away from the system under test.

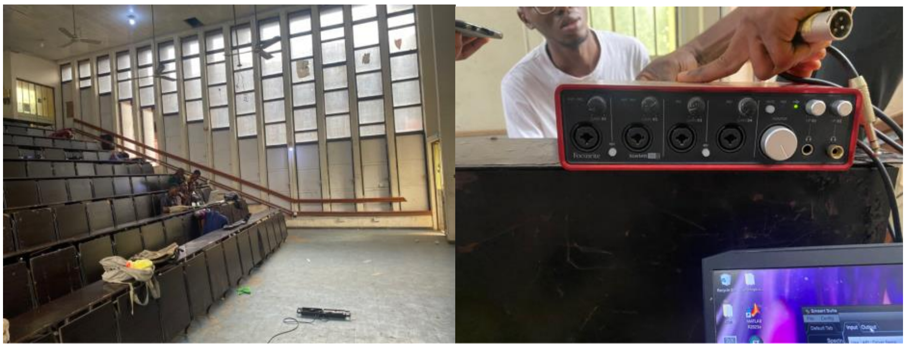


Figure 3.18: pictorial views of measurement procedure

CHAPTER FOUR

RESULTS AND ANALYSIS

We took measurements in various positions within the space to understand how the response sound of our system behaves. Specifically, we measured in two different locations:

1. EEE Final Year Lecture Theatre
2. An Open Field Area, not related to the lecture hall

To do this, we used a special microphone called the BEHRINGER ECM4000 electret condenser microphone, and we also used a microphone similar to the one in mobile phones, like the APPLE IPHONE 11 microphone. For reference, we measured sound in the university's main auditorium, which has less sound bouncing around and is easier to work with.

4.1 RESULTS OF DATA COLLECTION

The data from all positions/points are shown below:

Figure 4.1: measurement done at 1 meter from source.


Smaart Suite
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Level (%)
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project 2022-5.59meters ▼
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Real Time
Impulse
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0.0
dB FS Peak
Max 0.0
Transfer Function
Averaging Depth: 1
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project 2022
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Find Track
+ TF Engine
View
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Signal Generator
Pink Noise
-22 dB
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Output
Data Bar
Capture
Capture All
Reset Avg
Spectrum View
TF View
Clear All dB
dB +
dB -
Ctrl Bar


Smaart Suite
File Config Options View Command Window Help
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Frequency (Hz)
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None
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dB FS Peak
Max 0.0
Transfer Function
Averaging Depth: 1
Track
project 2022
15.52
Find Track
+ TF Engine
View
Timer
Signal Generator
Pink Noise
-22 dB
On
Output
Data Bar
Capture
Capture All
Reset Avg
Spectrum View
TF View
Clear All dB
dB +
dB -
Ctrl Bar

Figure 4.2: Data from EEE 500 Level Lecture Theatre at multiple positions

4.2 DISCUSSION ON DATA GOTTEN

The reference measurement, taken one meter from the sound source, showed some changes in the low-frequency part of the sound. The way the room affected the sound didn't change much, even though there were echoes. This is because the sounds we played were much louder than the background noise.

First, we measured sound in the EEE Final Year Lecture Theatre, which is a bit like a live concert venue. This place has wooden seats, concrete walls, and tile floors, making it tricky to get a good sound measurement. We measured the sound in five different spots in this room.

After measuring in the room, we noticed that the low-frequency sound built up a bit, especially at the lower end (between 45Hz-125Hz), because low sounds have long wavelengths. The differences we saw in the 45-125Hz range in all the places were because of how far the microphone was from the sound source and the angle of the microphone. Even though there were changes in the sound due to echoes, background noise, and how the sound spreads, the general shape of the sound measurements in the reference room looked similar to the measurements we took in other positions, although they were less noisy and didn't get as much interference from the room's corners and walls.

After we did some filtering on the reference measurements, they looked smoother and less bumpy compared to the measurements we got in the main position which was 1 meter away from the loudspeaker.

The main goal of this project was to design and make a system that can make low-frequency sounds between 45Hz and 125Hz louder, and we wanted to see if this worked in different places, not just in one position within the main class theatre. We also looked at how the distance between the microphone and the sound source, and the angle of the microphone, affected our measurements. We put the microphone flat on the floor and at an angle of 45 degrees, which made the measurements cleaner because we got fewer reflections from the floor.

Figure 4.3: microphone position in front of device under test.

4.3 EFFECT OF DISTANCE ON MEASUREMENTS

From the captured responses gotten, it is clear that as the distance of the measurement microphone increases from the loudspeaker(Device under test), the sound pressure level drops. There is a clear drop in amplitude of about 6dB SPL as distance increases.This conforms to and confirms the inverse square law.

We used spatial averaging on top of the usual smoothing in frequency. Both of these methods helped make our measurements more consistent, especially for specific frequency ranges. The most noticeable differences were at low frequencies, where we saw the biggest changes between a single measurement at one spot and an average of measurements from many spots close to the microphone.

4.4 AVERAGING MEASUREMENTS

When we averaged measurements, we noticed that the sound was a bit quieter compared to a single measurement at one spot. This happened because averaging made the local sound effects less noticeable, especially after we smoothed the measurements in frequency. Smoothing in frequency means we combined measurements from different times to make the data smoother. When we averaged multiple measurements, it was like doing this smoothing operation twice, which made the sound even smoother.

4.5 SOUND SYSTEM'S FREQUENCY RESPONSE

The graph below shows how our sound system responds to different frequencies after we did many measurements.

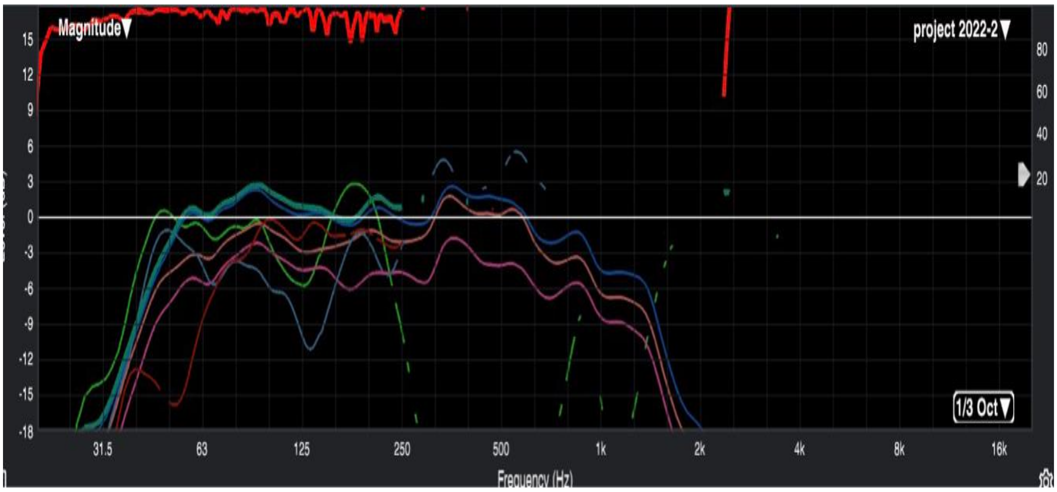

Figure 4.4: frequency response of the loudspeaker system.

CHAPTER FIVE

CONCLUSION AND RECOMMENDATION

5.1 CONCLUSION

This thesis project focused on estimating the pseudo (artificial) anechoic transfer function/impulse response of an 18” sub-low frequency loudspeaker through multiple measurements in a general room. We experimented with various methods throughout the project, but the final and most effective approach was maximum likelihood estimation with the system operating independently before post-filtering using a DBX DSP OPTIMIZATION SYSTEM. This method made use of automatically calculated information such as filtering out reflected echoes, determining the number of reflected echoes, identifying echo time delays, and assessing the spatial frequency response in different rooms or spaces. We applied a post-filter using a parametric equalizer employing a Linkwitz-Riley filter with a slope of 24dB/octave.

In the context of live sound reinforcement and amplification, loudness-based production relies heavily on precise monitoring for assessing balance, spectrum, speech intelligibility, and audio quality. Loudness, as perceived when audio is reproduced acoustically, is a complex, nonlinear function influenced by amplitude, frequency, and time. Given this complexity, achieving a controlled in-room spectral response is essential for meaningful level calibration, as required by global and regional loudness standards.

With these considerations in mind, we can make the following conclusions:

1. The frequency response of our designed sub-low frequency amplification system, consisting of an 18” low frequency transducer in a band-pass configuration and a matched

2000 Watt power amplifier, demonstrated consistency from 45 Hz to 125 Hz with minimal variation as the frequency increased.

2. Transfer function measurements proved to be a reliable choice for assessing the frequency response of various listening spaces. Using FFT techniques with sweep signals as excitation signals was the preferred method for most transfer function measurements, particularly when capturing room impulse responses for aural purposes.
3. Sub-low frequency amplification systems can be manufactured in Africa using locally-sourced materials while achieving responses comparable to those of products manufactured anywhere else in the world, meeting international standards and practices.

5.2 CHALLENGES

Despite the successes of this project, we encountered several challenges:

- The absence of an anechoic chamber, a space designed to have no sound reflections, made it difficult to achieve a response identical to our desired one.
- Spatial averaging, which we used to create a mean response from multiple measurement positions, may not significantly improve equalization outcomes and can risk compromising results at the primary listening position.
- Limited by the cost of a licensed version of SMAART, measurements were restricted to distances within fifty (50) feet, preventing us from taking measurements at multiple positions within the listening areas, which could have provided a better response.
- The direct wave's behavior varied slightly with small changes in microphone position but significantly with large changes in distance, raising questions about the true impulse

response of the loudspeaker.

- Conducting multiple measurements to achieve the required response consumed extra time and resources, as a single direct measurement was insufficient.
- Differences in voltage levels, especially in the three hundred level class venue, resulted in a slight hum in the power amplifier. While negligible during testing due to the amplitude of the test signal, it was still a challenge.
- The presence of people in the spaces during measurements introduced deviations from the desired response, as voices interfered with the measurement software, evident in the coherence charts.
- Variances in seating arrangements within each listening space compared to venues with standard seating caused multiple reflections in the impulse response (IR) charts, resulting in coloration of the direct output at those positions.
- The response of our system exceeded our expected response due to the lack if depth of the cabinets design. A longer depth would have given a response having a low pass centered beneath the 300Hz range compared to the actual response gotten post measurement.

5.3 RECOMMENDATIONS

Based on our findings and challenges faced, we propose the following recommendations:

- Seek access to an anechoic chamber for accurate measurements rather than relying on pseudo spaces with varying levels of reflections and absorptions.
- Prior to measurements, carefully select spatial averaging positions based on the room's intended equalization and acoustic characteristics.

- Consider employing a multi-microphone setup to expedite the measurement process.
- Conduct measurements in spaces free from human noise or discussions to achieve more accurate responses during measurements.

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APPENDIX I

LOUDSPEAKER'S SPECIFICATIONS

MODEL NAME	**CS1822**
System type	passive
Configuration	18" subwoofer
frequency response +/- 3dB	35-220Hz
sensitivity @ 1 meter	99dB SPL
system rated impedance	4 Ohm
low frequency transducer size	220mm/18"
voice coils	4"
LF power max.	4000W
system continuous power	2000W
system program power	1000W
enclosure color option	black urea paint
Hardware	4 transport trolleys
Weight	34.5Kg