VŠB – Technical University of Ostrava

Faculty of Mechanical Engineering

Department of Applied Mechanics

Design and Optimization of High Class Loudspeaker Cabinet

Návrh a Optimalizace Reproduktorové Soustavy Vyšší Třídy

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VŠB - Technical University of Ostrava
Faculty of Mechanical Engineering
Department of Applied Mechanics

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2. Choice of appropriate layout for a particular case.
3. Design of laudspeaker box, simulation of response properties.
4. Measurement on the prototype of the laudspeaker box, design of the crossover.

References:

[1] Cocker, J. M.. Handbook of Noise and Vibration Control. USA: John Wiley & Sons, Inc. 2007. 1038 s. ISBN 0-4713-9599-4
[2] Randall, F. B.. Industrial Noise Control and Acoustics. Louisiana: Marcel Dekker, Inc.2003. 534 s.ISBN 0-8247-0701-1
[3] METZLER, Bob. Audio Measurement Handbook. Beaverton: Audio Presision, Inc., 1993

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ANNOTATION OF DIPLOMA‘S THESIS

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This thesis deals with the design process of a loudspeaker system, what should be considered when designing the loudspeaker system, the inside parts the loudspeaker system and the features of these parts, and the path followed by the integration of all these components. After all the theory and descriptions, system is simulated and after that prototype is obtained. Subsequently, the results which are obtained from the simulation and the prototype are compared to decide about the success of designed loudspeaker.

ANOTACE DIPLOMOVÉ PRÁCE

SARI, F. *Návrh a optimalizace reproduktorové soustavy vyšší třídy: Diplomová práce.* Ostrava: VŠB –Technická univerzita Ostrava, Fakulta strojní, Katedra mechaniky, 2019, 66 s. Vedoucí diplomové práce: Weisz, M.

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The last but not the least, especially, thanks to my family, my greatest chance in this life.

TABLE OF CONTENTS

List of Used Signs And Symbols

Symbol	Name of Quantity	Unit
A	Amplitude	[m]
L_e	Voice coil inductance	[mH]
R_e	DC resistance	[ohms]
SPL	Sound Pressure Level	[dB]
S_{xx}	Spectral power density	[-]
t	time	[s]
ω	Angular frequency	[rad·s^{-1}
φ	Phase	[rad]
σ	Standard deviation	[-]
$\delta(\tau)$	Dirac pulse	[-]
τ	Time	[s]
Abbreviation	**Full name**	
BNC	Bayonet Neill–Concelman (Coupling Connector Type)	
B&C	B&C Speakers - The Name of The Speaker Company	
B&K	Brüel & Kjær - Danish Measuring Equipment Manufacturer	
CAD	Computer Aided Design	
CD	Compact Disc	
CPB	Constant Percentage Band	
DC	Direct Current	
DFT	Discrete Fourier Transform	

Abbreviation	Full name
FRF	Frequency Response Function
LAN	Local Area Network
PC	Personal Computer
TEDS	Transducer Electronic Data Sheet

INTRODUCTION

Loudspeakers are attractive devices especially for music lovers. Speaker design and development is an interesting field that combines knowledge of acoustics, electronics, and carpentry with the ever-subjective target of making things sound good. A person who is interested in this area, can create own products of similar or better quality with wide range of speaker companies.

The main purpose of this thesis is to give information how to properly design an enclosure for a speaker. Before the explanation of design and measurements, it is important to understand the background and theory of loudspeakers. In order to design a loudspeaker system, it is necessary to have sufficient knowledge of loudspeaker systems.

The aim of this thesis is to provide comprehensive information about loudspeaker design and optimization as well. When designing and optimizing a loudspeaker system, engineers must do considerations. Some of the major considerations are choosing appropriate enclosure, which drivers to use, how to make the crossover network, and methods for measurement.

In this master thesis, firstly it is planned to do investigation of issue of the loudspeaker system. As it is said, a sufficient background about the issue will help to design and to understand how loudspeakers and measurements are made. Most of the common enclosure types are mentioned. In addition, their working principle, advantages and disadvantages for each one is explained. Then, the choice of appropriate layout is made. Thereafter, a three-way loudspeaker which has one high frequency range driver, one mid frequency range driver and two low frequency drivers is designed.

When measuring the characteristics or the properties of the loudspeaker, it is important to make simulation in order to compare the results and to have an idea about success of the designed and produced prototype loudspeaker. For this reason, simulation of response properties is done in this thesis.

And finally, after the prototype is completed, the frequency response, directivity and sensitivity properties of prototype of the loudspeaker are measured. According to comparison of simulation and measurement, it can be said that our production is successful.

1. THEORY AND BACKGROUND

1.1. Loudspeaker Systems

First it is important to know what sound is. Sound is a vibration or disturbance that travel through any medium by transferring energy from one particle to other and can be heard when it reaches a person's or animal's ear. [1]

Sound consists of longitudinal or compression waves that move through air or other materials. The various characteristics of sound waves are wavelength, frequency, velocity and amplitude. Wavelength is the distance from one crest to another of a wave. Since sound is a compression wave, the wavelength is the distance between maximum compressions. The frequency of sound is the rate at which the waves pass a given point. The distance travel by a periodic motion per unit time is called velocity of wave. The sound waveform moves at approximately 344 meters/second, 1130 feet/sec. or 770 miles per hour at room temperature of 20^{o}C (70^{o}F). Since sound is a compression wave, its amplitude corresponds to how much the wave is compressed, as compared to areas of little compression. Thus, it is sometimes called pressure amplitude. [2]

Sound moves in pressure waves. When air particles are compressed and rarified fast enough, we hear it as sound. The faster the air pressure changes, the higher the “frequency” of the sound we hear. When a speaker moves back and forth it pushes on air particles which changes the air pressure and creates sound waves. [3]

A loudspeaker is a device that transform an electrical signal into sound waves. The loudspeaker uses a coil which can slide backwards and forwards over the central pole of a circular permanent magnet. The coil is joined by the brown bars to a paper cone. The wire from the amplifier carries an alternating current which makes the coil (and the paper cone) move backwards and forwards at the same frequency as the changing current. The paper cone then moves the air backwards and forwards which creates the sound. [4]

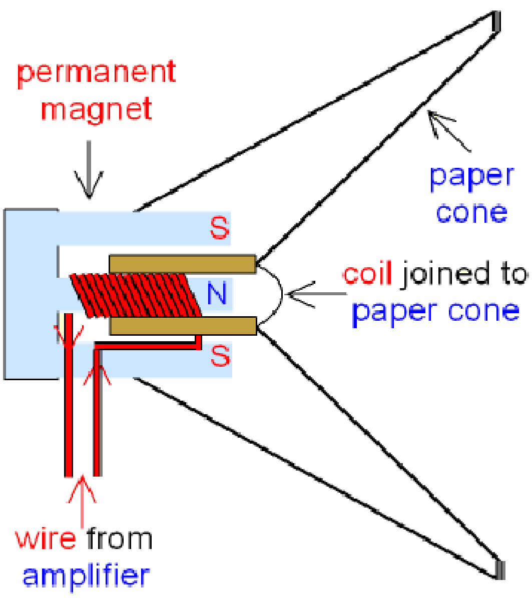


Figure 1.1 - Basic schema of a speaker [4]

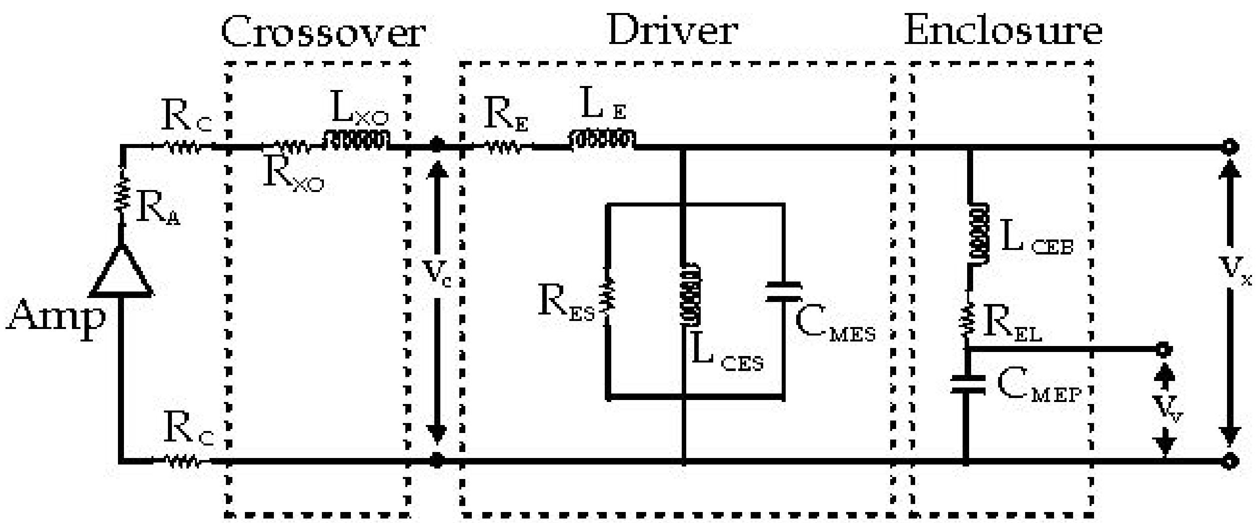
Speakers are generally placed in a speaker enclosure or speaker cabinet which is frequently a rectangular or square box made of wood or sometimes plastic. The enclosure's materials and design act a significant role in the quality of the sound. The most generally utilized kind of speaker is the dynamic speaker. There is obviously a very wide range in the quality of loudspeakers. Cost size, and many other factors contribute to the overall quality. [5] All speakers are either passive or powered. Most speakers that are commercially available are passive. Passive speakers do not have a built-in amplifier and must be connected to an amplifier using a regular speaker wire. Powered speakers are also known as self-powered or active speakers. They feature built-in amplifiers for high and/or low frequencies. Powered speakers often require heavy enclosures, making them less mobile and functional as passive speakers. They are sometimes less reliable since they have built-in electronic components that require external power sources. [6]

Loudspeakers consist of three fundamental parts;

1 – Speaker Drive Units

2 – Crossover Network

3 – Enclosure



R_A = Amp output resistance

R_C = Speaker cable resistance

R_{XO} = Crossover inductor resistance

Figure 1.2 - Equivalent circuit diagram of the speaker system [7]

1.2. Parts of A Speaker

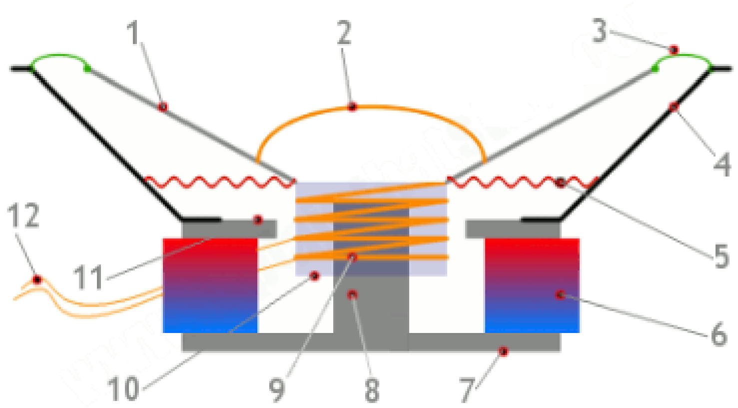


Figure 1.3 - Speaker parts [8]

1.Diaphragm: Moves back and forth allowing the air in the surroundings to move; creating sound.

2.Dust Cap: Protects the Voice Coil from dust, keep these out of the reach of children as these are just asking to be pushed in.

3.Surround: A lining that connects the basket with the diaphragm.

4.Basket: A metal frame which holds the speaker together, which in turn is held by the speaker case.

5.Spider: A type of suspension that keeps tension on the voice coil but allowing it to move, acts similar to the struts on a car that reduce unnecessary vibration.

6.Magnet: Often a rare earth neodymium.

7.Bottom Plate: Holds the pole piece and magnet.

8.Pole Piece: Directs the voice coil magnetic field.

9.Voice Coil: Moves the diaphragm via magnetic field created by current in the wire.

10. Former: Material in which coil is wrapped around.

11. Top plate: Typically made of iron.

12. Cables: The wires that connect to the voice coil from whatever the input source is. [8]

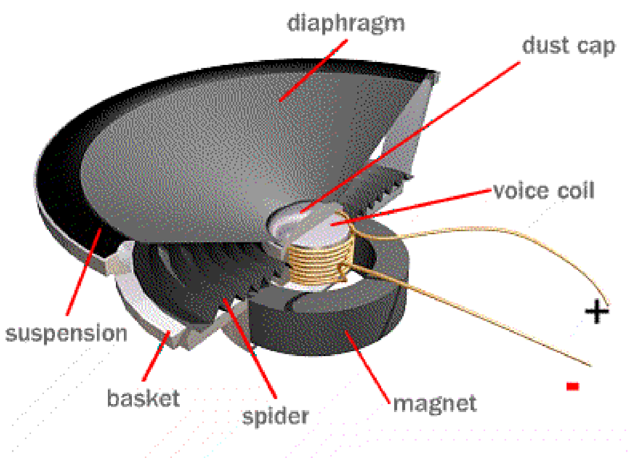


Figure 1.4 - Parts of a speaker [8]

1.3. Types of Loudspeakers

Speakers are categorized depending on the number of characteristics, including the types of drivers and enclosure used in their construction.

1.3.1. Dynamic

This is a device that uses an electromagnetic coil and diaphragm to create sound. [9] The most common type of speaker, these devices are typically passive speakers. They generally have one or more woofer driver to produce low-frequency sound, which is also known as bass. One or more tweeter drivers in dynamic drivers produce high-frequency sound, or treble.

Professional audio dynamic speakers that offer higher performance may also have drivers on the rear of the speaker enclosure to further amplify sound. [6]

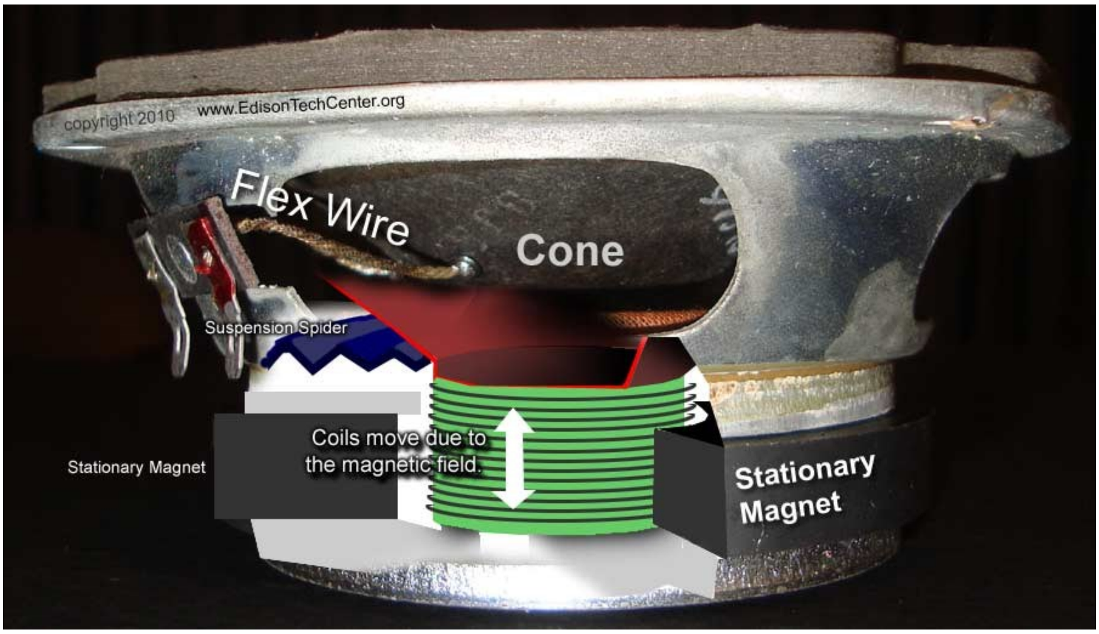


Figure 1.5 - A dynamic speaker [9]

There is a number of basic components common to all moving coil dynamic loudspeakers, but since the magnetic assembly has so much influence upon the amount and quality of the sound reproduced, it must be considered the heart of the loudspeaker. All loudspeaker magnetic assemblies have the same essential parts: a set of concentric pole pieces, a magnet, and a surrounding iron pot to carry the magnetism from the magnet to the pole pieces.

The most critical element in the design of a magnet assembly is the voice coil gap. To complete the magnetic circuit, magnetic energy must jump across the gap between pole pieces. The strength of the magnetic field in the voice coil gap varies inversely with the size of this precision-machined air gap. As the gap flux density becomes stronger, a more efficient motor action of the voice coil operating within the magnet assembly will be realized.

The amount of push exerted upon a voice coil and speaker cone in response to an input signal current depends not only upon the flux density within the voice coil gap, but also the amount of voice coil wire that is immersed in the voice coil gap at any one time. The voice coil must be kept perfectly aligned mechanically within the magnetic gap during its vibration cycle.

If the vibrations are non-linear, the voice coil may scrape against the magnet's walls, causing distortion and eventually, shorted turns in a burned-out voice coil.

The leads from the voice coil and the bobbin are attached to the diaphragm or cone constructed of paper, cloth, or aluminum. As the voice coil reacts to an input signal by repelling itself in and out of the magnet voice coil gap, the mechanical energy introduced to the attached diaphragm is translated into acoustic energy. The diaphragm vibrates at a certain frequency, and an audible tone is generated. This energy conversion should provide the greatest amount of acoustic power output for a given amount of electrical input, with a minimum of distortion. [10]

1.3.2. Electrostatic

Along with ribbon or planar-magnetic speakers, electrostatic speakers are a type of flat-panel loudspeaker or diaphragm speaker. They feature one driver and a thin membrane over two conductive, stationary panels. Electrostatic speakers generally have an outside power source and are plugged into an electrical outlet. The membrane receives an alternating current from an amplifier, which produces sound. Sound delivered using electrostatic speakers is crisp and detailed. They are often used for high frequencies and are not ideal speakers for bass or low-frequency sound since the thin membrane moves very little.

1.3.3. Planar-Magnetic

These speakers feature a tall, thin, and narrow metal ribbon instead of the wide diaphragms found on electrostatic speakers. Rather than charged metal panels like those found on electrostatic speakers, the ribbon in a planar-magnetic speaker is suspended between powerful magnets. A current passes through the metal ribbon, which resonates towards or away from the magnets in order to generate sound waves. This type of speaker does not need to be connected to an electrical system to operate. [6]

1.4. Types of Speaker Drivers

Multiple loudspeaker transducers are frequently placed in the same enclosure, where high fidelity reproduction of sound is required. In such a case the individual speakers are referred to as drivers. Drivers supply their best performance within a specific frequency range.

1.4.1. Full-Range

Designed for a wide range of frequencies, these drivers are typically smaller devices. This type of driver is commonly used for public address or public announcement (PA) systems, televisions, radios and computers, earphones, and some toys.

1.4.2. Mid-Range

This type of driver is designed to handle mid-range frequencies. They are typically used for frequencies between 200 Hz and 2000 Hz.

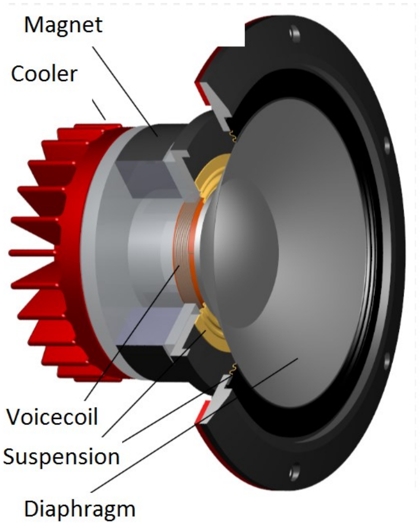


Figure 1.6 – A mid-range driver

1.4.3. Woofer

Handles low-frequency or bass ranges under 200 Hz for consumer systems and below 100 Hz for professional audio systems. Some woofer drivers are also used for mid-range frequencies when they are combined with a tweeter driver or another woofer driver.

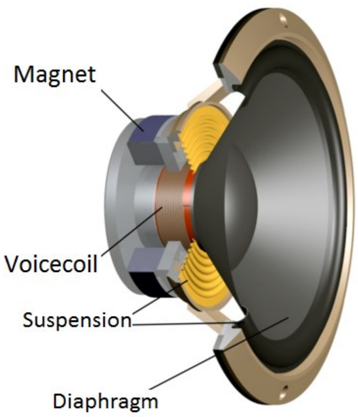


Figure 1.7 – A woofer

1.4.4. Tweeter

Designed to handle high-frequency ranges, or treble. Typically, these drivers are used for frequencies above 2000 Hz. Tweeter drivers are designed to handle different power and output levels on various types of speakers, including ribbon or planar-magnetic speakers. These drivers are generally found in home stereo systems, as well as professional sound systems.

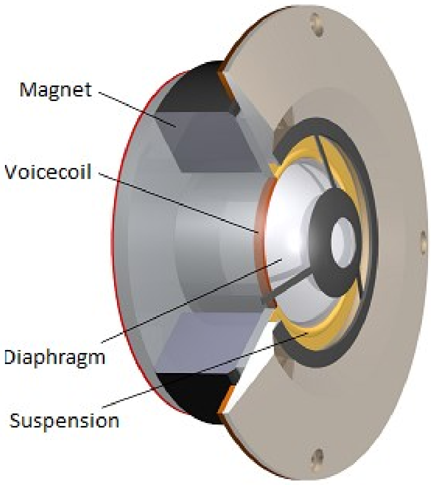


Figure 1.8 – A tweeter

1.5. Crossover Network

A crossover network is a device which divides the full frequency range electrical signals into the appropriate sections or bandwidths to be delivered to loudspeakers which are to cover a specialized bandwidth in a multi-way system. By doing so, a crossover network serves two important functions;

- Firstly, it allows optimum system sound quality by ensuring that each loudspeaker only operates within its intended bandwidth.
- Second, it prevents damage which would otherwise be caused by low frequencies being fed to Midrange and High Frequency Drivers. [11]

There are three basic ways to “crossover” or divide frequencies.

1- **High Pass Crossover:** Allows frequencies above the chosen cut off frequency to pass through to a speaker or group of speakers.

2- **Low Pass Crossover:** Allows for frequencies below the chosen cut off frequency to pass through to a speaker or group of speakers.

3- **Band Pass Crossover:** Uses a combination of a High Pass Crossover and a Low Pass Crossover to allow a range of frequencies above and below two chosen crossover frequencies (one High Pass and one Low Pass) to pass through to a speaker or group of speakers. [12]

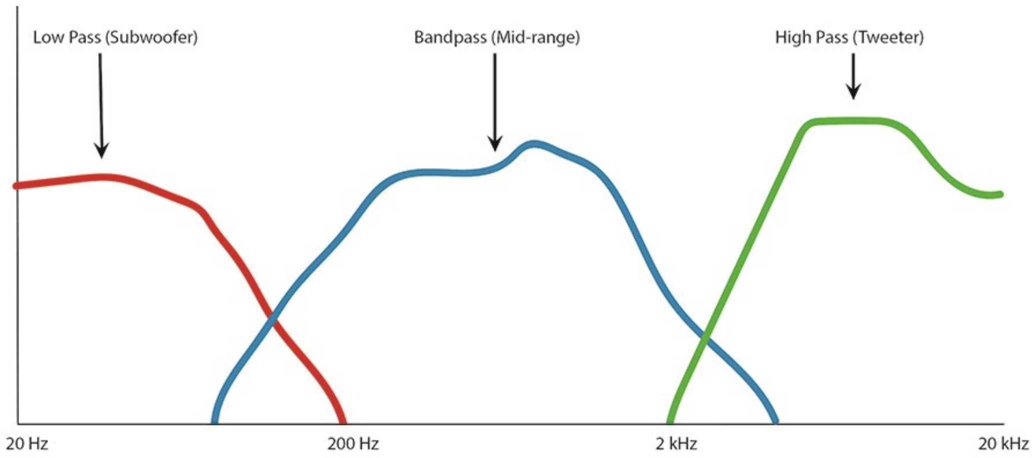


Figure 1.9 - Separation of frequencies [12]

There are mainly two types of crossovers;

1 – Passive Crossover Network

2 – Active Crossover Network

Passive crossovers are used for obvious reasons; they are simple to build (although not that simple to design), relatively low cost. And active crossover sound better than passive crossovers because they eliminate all of the disadvantages of passive crossovers. [13] In addition, passive crossovers don't need power to filter the signal as desired. Active crossovers require power and ground connections but give you much more flexibility and fine-tuning control over your music. [14]

1.5.1. Passive Crossover Network

The term 'passive' means that for the circuitry there is no additional power source required. Passive crossover networks use capacitors, inductors, and resistors to split the audio signal so high frequency information goes to the high-frequency driver and low frequency information goes to the woofer. Passive crossovers are the most cost-effective and allow the loudspeaker to be powered with one amplifier channel. While they work well in most cases, passive crossovers are not as precise as active crossovers. [15]

Pros of passive crossovers;

- Simple (plug and play)
- Single speaker cable (per speaker)
- Single stereo amp
- Usually less expensive

Cons of passive crossovers;

- Crossover interference (back EMF) with amplifier signal
- Loss of speaker damping and amp's direct control of driver especially near the crossover frequency
- Higher loading on amplifier with greater losses requiring higher wattage
- Uneven phase shifting between drivers with different impedances
- Variable and nonlinear responses with changes in power and temperature
- Greater probability of amplifier clipping and driver damage due to higher power and complex impedance loads. [13]

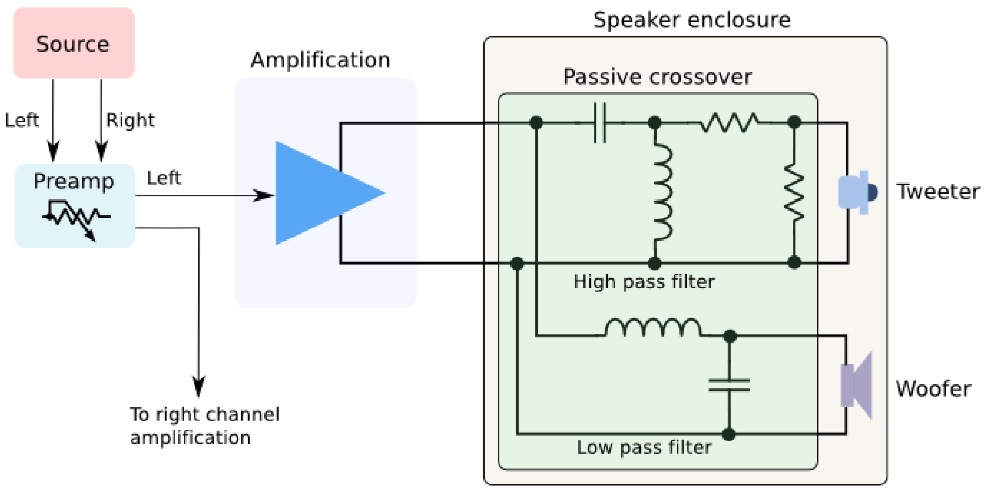


Figure 1.10 – A passive crossover network [16]

1.5.2. Active Crossover Network

Active crossovers require external power to operate. They are connected between the output of a mixer (or preamplifier) and a power amplifier. One of the advantages of active crossover is that you can often change characteristics such as crossover frequency and the rate of transition. [15]

Pros of active crossovers;

- Direct control of each driver by each amp channel
- Simple and easier impedance loading on amp
- No parasitic power losses
- No loss of damping (driver control)
- Less likelihood of clipping with clipping limited to single driver
- Consistent crossover behavior regardless of power level or signal content
- Less loading on each amp with load divided among multiple amps
- Less distortion
- Highly flexible and adaptable especially with DSP (digital signal processing) technology

Cons of active crossovers;

- Inherently more complex
- Potentially increased noise
- At least twice the number of cables
- Requires multiples amps or amps with multiple channels
- Usually more expensive [13]

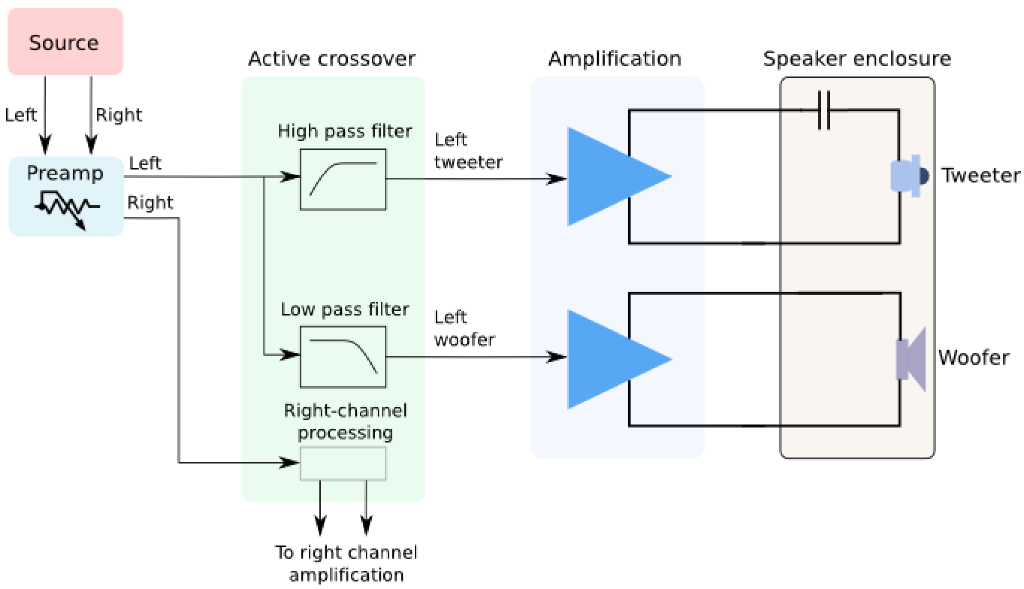


Figure 1.11 – An active crossover network [16]

1.5.3. What Is an N-Way Crossover or N-Way Loudspeaker?

Loudspeakers are frequently categorized as "N-way", where N is the number of drivers in the system. For example, a speaker with a woofer and a tweeter is two-way. An N-way speaker generally has an N-way crossover to divide the signal among the drivers.

Combinations of capacitors, inductors, and resistors can direct high frequencies to the tweeter and low frequencies to the woofer. This amounts to filter action. A two-way crossover network divides the frequency range between two speakers. A three-way crossover network divides the frequency range between three speakers.

1.6. Enclosure (Speaker Cabinet)

A loudspeaker cabinet is an enclosure (frequently box-shaped) in which speaker drivers (e.g., loudspeakers and tweeters) and associated electronic hardware, such as crossover circuits and, in some cases, power amplifiers, are placed. The role of a speaker cabinet is the separate the waves produced by the front of the speaker, from the waves produced by the back of the speaker. If they meet, they will cancel each other, since they are out of phase, and will lead to poor bass response. This happens especially to the lower frequencies (bass). [17]

An ideal enclosure would be infinitely rigid. It would secure the drive units in a fixed position and be acoustically transparent.

1.6.1. Sealed (Infinite Baffle) Enclosures

This type of enclosure has the back of the baffles completely housed in a box of an appropriate size to prevent sounds from being radiated from the back of drivers, confining the back sound within the box. Thus, only sound radiated from the frontal side of drivers will reach the listeners. [18] One problem with this design, however is that the enclosure itself can create unwanted resonances that color the sound. These resonances, however, can be kept in check by the use of low-resonance materials (such as aluminum and wood) for the enclosure as well as insulating materials such as sheep’s wool inside the enclosure.

Pros of sealed enclosures;

- Simple (easy to design and build),

- Space-saving construction,

- Precise playback of even the deepest frequencies,

- High power handling,

- Excellent transient response,

- No wind noise (as with ported enclosures),

- Cheaper to manufacture.

Cons of sealed enclosures;

- Require more power to overcome the internal air pressure (less efficient) [19]

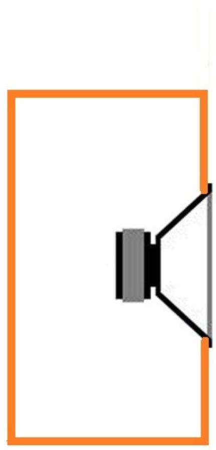


Figure 1.12 – Sealed enclosure [17]

1.6.2. Bass Reflex (Vented) Enclosures

Unlike sealed enclosures, bass reflex (also known as “vented” or “ported” enclosures) include at least one tube-like opening or vent connecting the interior of the enclosure to the outside. [19] A bass reflex speaker is designed so that the back wave of a speaker cone is routed through an open port (called a vent or tube) in the enclosure in order to reinforce overall bass output. These ports are generally located on the front or rear of the speaker cabinet and can vary in depth and diameter (even wide enough to reach your hand through). Channeling the speaker cone's rear sound wave through such a port can often be an effective way to increase output volume, reduce distortion, and improve bass response and extension (versus sealed enclosure speakers). [20]

Pros of bass reflex enclosures;

- Higher volumes,

- Stronger bass,

- Compact enclosure size,

- Greater efficiency than with sealed enclosures.

Cons of bass reflex enclosures;

- More difficult to achieve a clean and accurate frequency response,

- Less precise and “boomier” bass (which some may actually see as an advantage),

- Bass reflex vents or tubes that are not properly designed can lead to “wind noise”. [19]

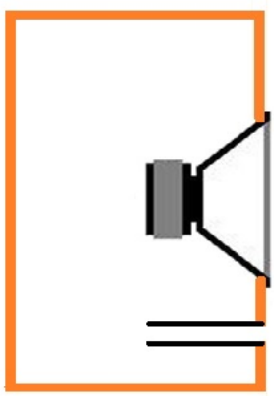


Figure 1.13 – Bass reflex enclosure [17]

1.6.3. Horn Enclosures

The basic principle on which the horn relies on is impedance matching. The speaker is a mechanical system, which has a high impedance, versus the air, which has a low impedance. When a wave propagating in a tube, meets an abrupt change in acoustic impedance, part of its energy will be reflected back. Horns are not tubes. They have a certain taper. The duct or tube is progressively increasing in its cross section. Because of this taper or flare, the horns act like an impedance transformer (aka coupler). They make a smooth transition from the high impedance of the cone, to the low impedance of the air.

It is explained the principle on how the horn works, let’s enumerate the reasons why you would want to couple a horn to your speaker:

- Highly improved efficiency.

- Eliminating the resonance introduced by speaker boxes.
- Increased directivity. The sound doesn't spread as much, like with normal radiating speakers. Depending on how the horn is designed, the sound is directed into certain areas. This can be a good or a bad thing, depending on the application.
- Reduction of speaker generated nonlinear distortion.

The horn enclosure is a popular choice for outdoors or for very large rooms. The main reason is because they have very high efficiency and the designer can control the directivity of the sound. The size of the enclosure is not much of a problem, since they are for outside use. However, home audio is no stranger to horn enclosures.

The horn is composed of 3 main parts:

- The Throat: Which is the part that is connected to the speaker.

- The Neck: Which describes the length of the horn.

- The Mouth or The Bell: Which describes the end part of the horn, "connected" to the air.

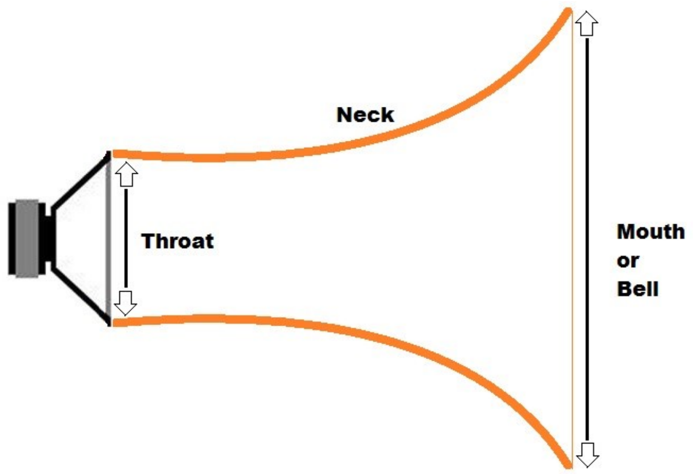


Figure 1.14 – Parts of horn enclosure [21]

A horn will start expanding, starting from the throat and end at the mouth. The speaker will be connected at the throat of the horn, and radiate sound at the mouth of the horn. All of these parts influence how will the horn affect the overall sound. The flare and mouth design, the phase and direction of the particle velocity at the mouth, will all have an impact on the sound quality and directivity of the horn.

One of the main characteristics of the horn is its shape. The horn has a certain taper, which is determined by the cross-section expansion rate. The cross-section area is determined by a function of distance, from the throat of the horn along its axis. This function will give the neck of the horn a certain shape. This means that the neck can have various shapes.

Here are some common horn profiles:

- Parabolic: Easy to design and construct, but poor impedance conversion.

- Conical: Easy to design and construct, but poor impedance conversion.

- Exponential: Good wide band impedance conversion, but some nonlinearity.

- Hyperbolic: Very good and high impedance conversion, but relative nonlinearity.

- Stepped: High impedance conversion. Nonlinearity depends on step resolution. This shape is not like the others. The horn is not growing in a smooth fashion, but in abrupt square steps (imagine a cube, then a larger cube, and so on. The speaker plays through these cubes) [21]

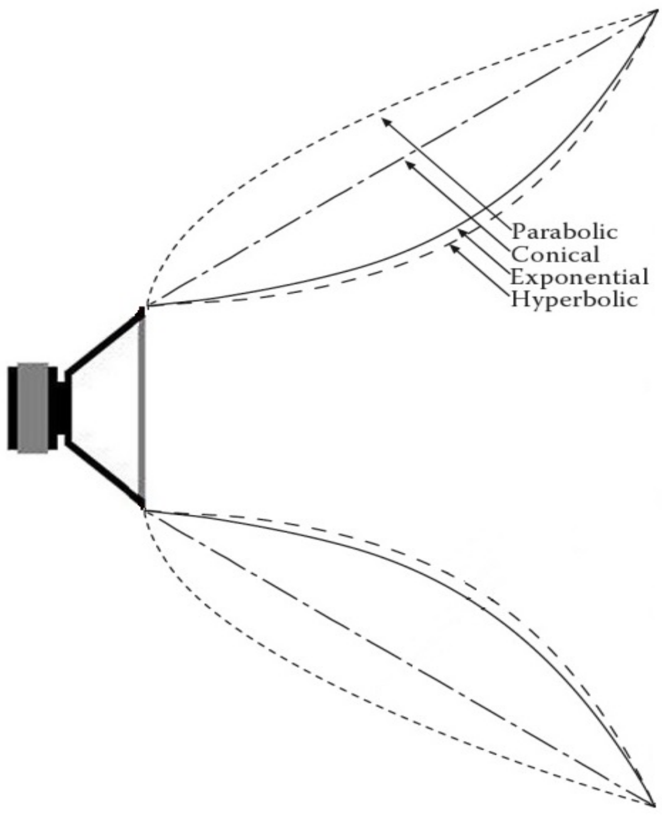


Figure 1.15 – Horn profile types [21]

2. DESIGN

2.1. Choice of Appropriate Layout

The construction of loudspeaker consists of 4 number of “drivers” (1 Tweeter, 1 Mid-Range Speaker,2 Woofers) mounted on a flat surface called “a baffle” and placed within “a Bass Reflex (Vented) Enclosure”. Each driver consists of a flexible cone attached to a coil of wire that is mounted so that it can move freely, within limits, inside of a fixed magnet. This coil is referred to as the “voice coil.” Electrical currents passing through the coil create a varying magnetic field, which reacts with the fixed field and produces mechanical fluctuations of the coil. The cone then moves in turn and sets a column of air in motion both in front of and behind the cone. In this way, an electrical signal is converted into a sound pressure wave. Ideal loudspeakers produce acoustic waves that are a linear transformation of the electrical input (excitation) signal. This fact is the underlying assumption in traditional, “frequency response” analysis of electro-acoustic systems.

2.2. Design of Loudspeaker Enclosure

In this thesis, it is designed that a three-way loudspeaker. As it is said in section 2.1, the construction of loudspeaker consists of four number of “drivers” (1 Tweeter, 1 Mid-Range Speaker,2 Woofers) mounted on a flat surface called “a baffle” and placed within “a Bass Reflex (Vented) Enclosure”. The tweeter placed at the top. Mid-range driver is placed just below the tweeter. And the woofers are placed at the bottom side.

It is shown that the front view of the loudspeaker. (see figure 2.1)

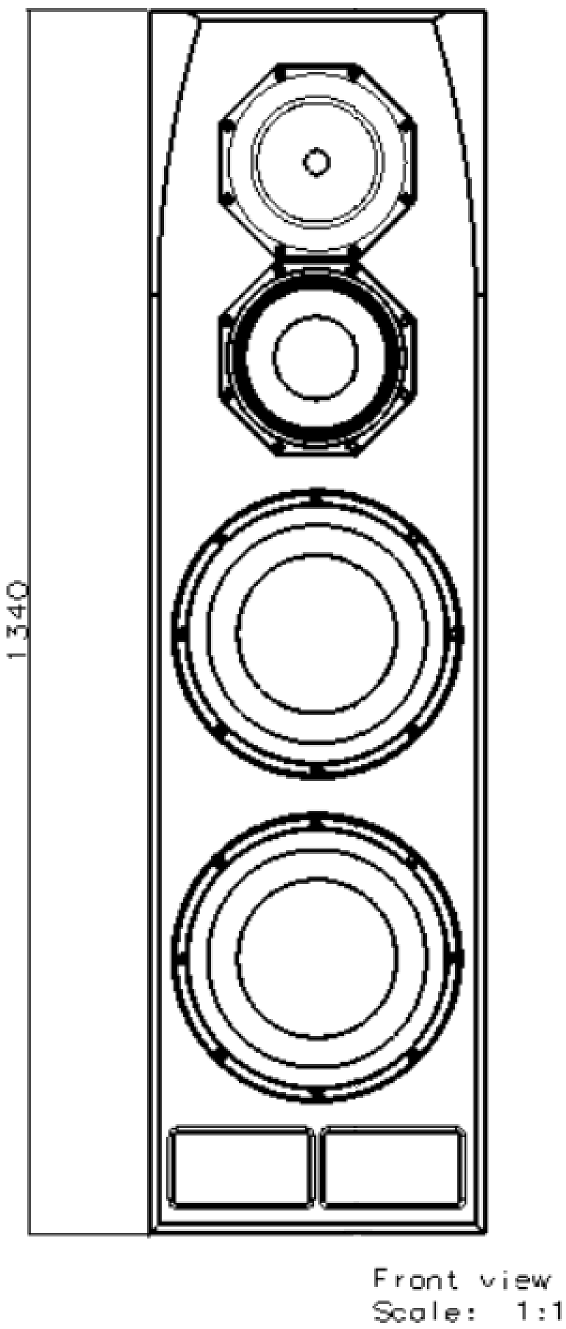


Figure 2.1 – Front view of designed loudspeaker

We can see the top view of the loudspeaker below. (see figure 2.2)

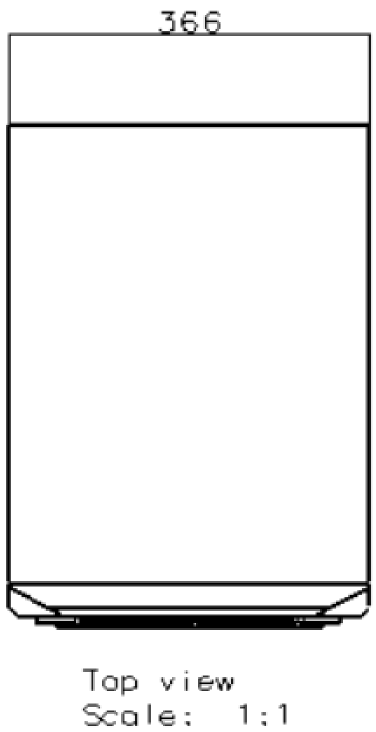


Figure 2.2 – Top view of designed loudspeaker

Left side view of the loudspeaker is shown below. (see figure 2.3)

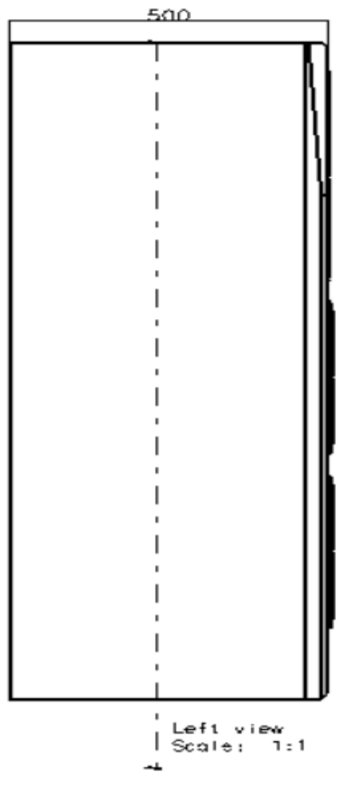


Figure 2.3 – Left side view of the loudspeaker

A cutaway view of the loudspeaker can be seen below. (see figure 2.4) We can see how designed the bass reflex pipe in this figure. In addition, it is important to desing the section of tweeter as closed box. The reason of this, to avoid the pressure effect of the woofers.

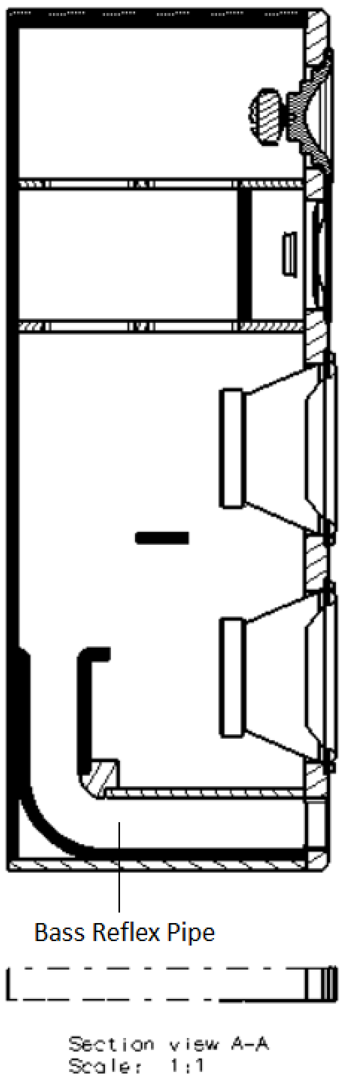


Figure 2.4 – Cutaway view of the designed loudspeaker

And finally, we can see the isometric view of the designed loudspeaker. (see figure 2.5) In order to design this loudspeaker, a CAD software which is named CATIA V5 is used.

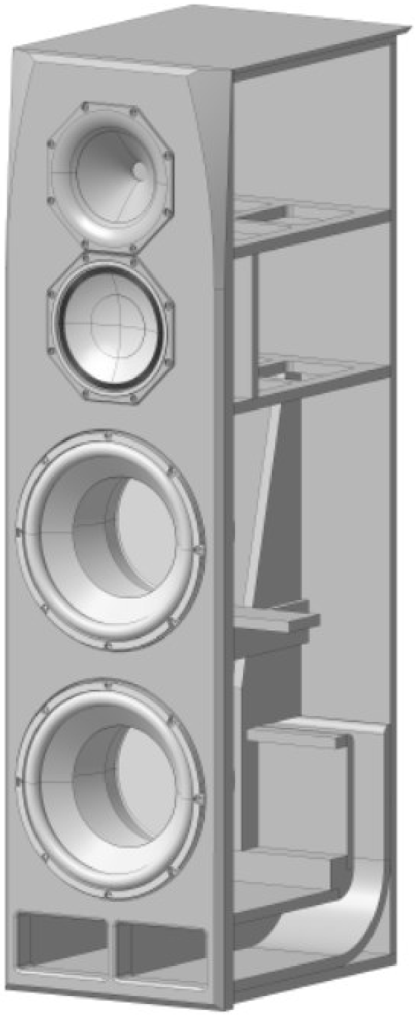


Figure 2.5 – Isometric view of designed loudspeaker

2.3. Design of Crossover Network

For this designed three-way loudspeaker, we used both passive and active crossovers. Between high frequency driver and mid frequency driver, there is series connection. And there is parallel connnection between two low frequency drivers. Active crossover is used between signal source and amplifiers. And passive crossover is used between high frequency driver and mid frequency driver. We can see designed crossover network circuit below. (see figure 2.6)

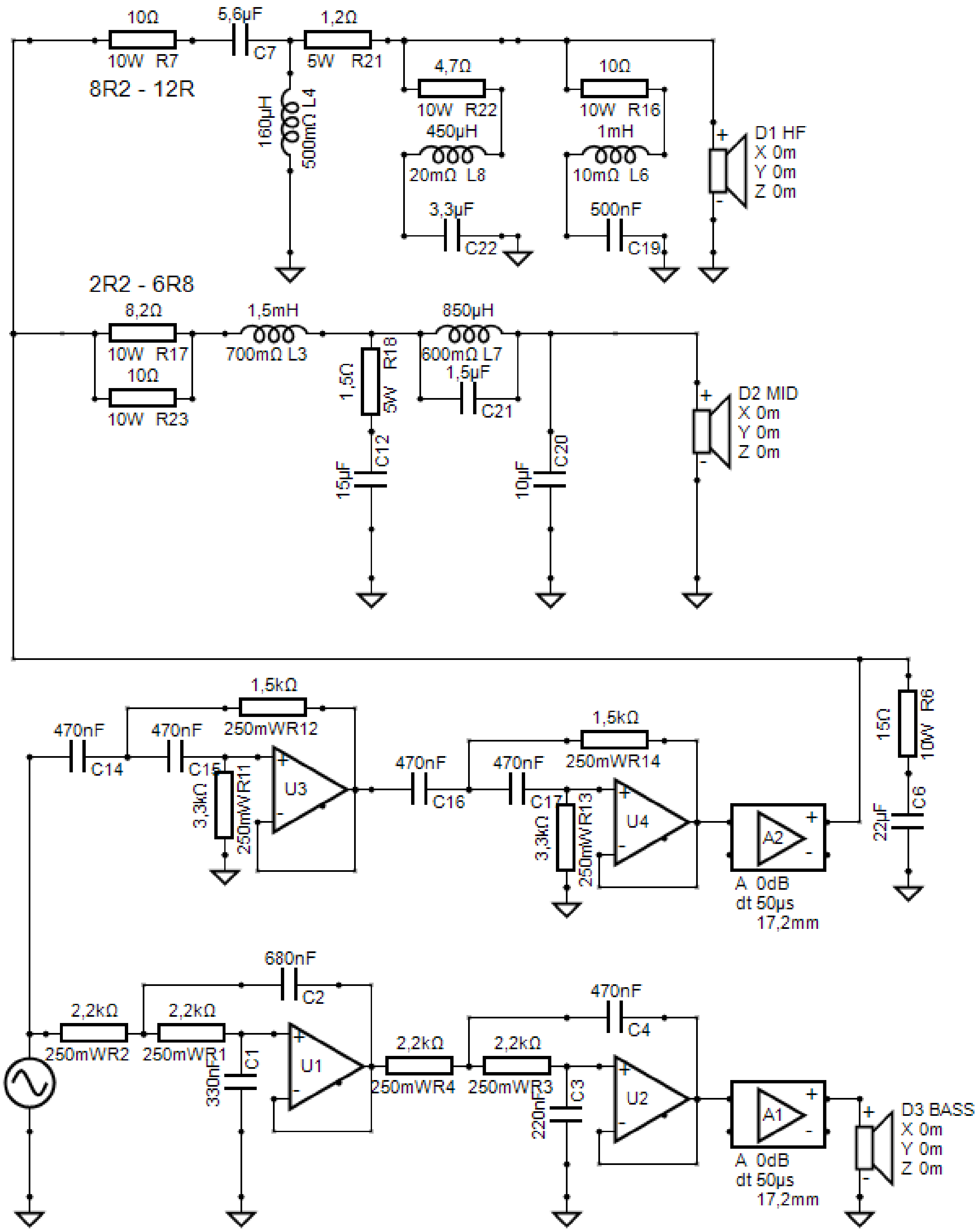


Figure 2.6 – Crossover Network Circuit

2.4. Speakers

Our loudspeaker system has 4 drivers. There is 1 tweeter (B&C Speakers DE250 8/ohm series), whose data sheet is included in the appendix (Appendix A), 1 mid-range speaker (B&C Speakers 8MBX51-8 series), whose data sheet is included in the appendix (Appendix B) and 2 woofers (Eminence LAB12 series), whose data sheet is included in the appendix(Appendix C). Here we can see the figures of speakers below.

Figure 2.7 - B&C Speakers DE250 8/ohm

Figure 2.8 - B&C Speakers 8MBX51-8

Figure 2.9 - Eminence LAB12

3. CHARACTERISTICS OF SPEAKER SYSTEMS

The aim of all calculations and designers of professional loudspeaker systems is to construct an acoustic device that is capable of so-called true reproduction. This term can be defined as achieving an identical auditory perception when listening to an input signal while listening to an output signal. For example, in everyday life, this could mean equivalence of live performance and reproduction. Unfortunately, in technical practice this is a utopia, as achieving such a state is basically physical and thus technically impossible. So, we are only able to make out through the hearing whether the musician is playing directly in front of us or the music recorded by him is spreading out of the speaker.

A lot of attribute is involved in the above loss of loyalty to the reproduced signal. Whether it is the previously described material properties of the loudspeaker membrane (stiffness, compliance) and its own vibrations or also the acoustic properties of the used loudspeaker. Furthermore, this electroacoustic string also distorts the amplifier used.

For this reason, all loudspeaker systems are subject to a certain degree of distortion in the output signal, which is reflected in the quality of the resulting acoustic sensation, which is essential for end users of these products. How large is this distortion can be determined by sophisticated measurement of the different response characteristics of the speaker systems, which will be our task in the following chapters.

3.1. Frequency Response Function

Frequency response function, often also referred to as frequency response, is undoubtedly one of the most important physical data for most acoustic sources that act as a transmission system. Speaking of the system's frequency response function, as is sometimes called the FRF, we can already guess from the name itself what this function says.

Frequency response characteristics exist in more than one kind. Most often, we can meet the amplitude frequency characteristic, which is basically a graphical representation of the functional dependence of the characteristic sensitivity on the excitation frequency on the logarithmic scale. This characteristic is used not only to describe acoustic sources, but also to determine the parameters of various technical devices, which can be amplifiers, preamps or CD players. We can also rarely encounter a description of the response characteristics of a space using the FRF.

The subject of this master thesis was mainly to use the frequency response function for declaring the properties of loudspeaker systems, where it is considered one of the determining parameters of the quality of the electroacoustic device. However, when interpreting this parameter, we can also encounter the infamous deception of potential loudspeaker customers, where the manufacturer deliberately sets this characteristic in a graph with an excessive scale width in the Y-axis direction. This fact, together with the logarithmic scale of the vertical axis, results in the resulting frequency response seemingly smooth. It means that not all frequency drops are visible, and the product can appear outwardly superior to what it is.

Obviously, the ideal situation for FRF description would be if the described characteristic was a constant waveform and would appear on the graph as a parallel line with the longitudinal axis. The FRF would then have a flat and smooth course. This would mean that the entire speaker system is linear with zero distortion. However, on the basis of electro-acoustic paradigms, it is clear that this situation cannot be achieved in technical practice.

On the other hand, there is a human auditory organ, which is also inherently a non-linear receiver whose frequency sensitivity should not be neglected in the detection of FRF loudspeaker systems, since these systems are primarily for our sensory perception and hearing. However, it is not the purpose of this work to deal with psychoacoustics of man and his sound perception, because this area is a separate and very extensive issue of acoustic science. However, we can at least marginally approach the non-linearity of the human ear, which is aptly described by the so-called constant volume curves, which are actually the signal lines of individual frequencies perceived with the same volume. [22] Their course is shown in the following figure. (see figure 3.1)

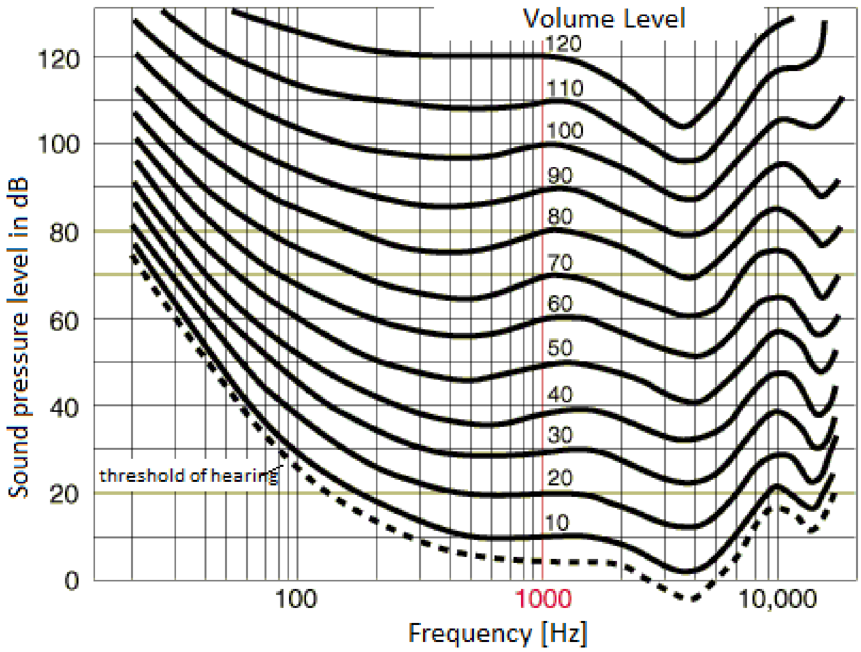


Figure 3.1 - Constant volume curves [23]

3.1.1. Rules for Measuring the Frequency Response Function

When measuring the FRF, it is advisable not to limit itself to measuring this function in the loudspeaker reference axis, but it is recommended to include results from measurements for off-axis positions (e.g. 30 °, 60 ° from the reference axis). This further measurement gives us at least an approximate idea of the directionality of the speaker. For the requirements for professional output data from measurements, we must also verify the reverberation conditions of the acoustic environment in which we measure the loudspeaker systems. Due to we have semi-anechoic room, this aspect will be reflected in the measurement.

However, it is sometimes advantageous to measure the loudspeaker systems directly in a real room where these systems are most often located. The system is then measured in 3 positions of the area and the data from them is averaged. In connection with this measurement, the frequency characteristic of the room must also be determined and then superimposed with the measured characteristics of the speaker. Obviously, reflections and standing waves in the room will cause large fluctuations at low frequencies, which will greatly distort FRF results in this area, although in the range of about 400 Hz, the resulting curves of the 3 measurement positions become almost identical.

This work will only include measurements in a semi-anechoic room where we will try to eliminate the effects of floor reflected waves that could affect the resulting FRF using various measurement procedures.

3.2. Frequency Response

The frequency response is used to describe the audible frequency range that a loudspeaker can reproduce. Audio frequencies are measured in Hertz (Hz) and the theoretical range of human hearing is generally regarded as being from about 20 Hz through 20 kHz. [24] Frequency Response is a curve that shows how strongly an audio system reproduces different parts of the frequency range. Frequencies are measured along a graph's x-axis, with sound pressure level (SPL) measured in decibels (dB) along the graph's y-axis. The curve will be higher on the graph at frequencies where the system plays louder, and lower (perhaps showing only varying background noise) at frequencies where the system plays weaker or not at all. A perfect frequency response curve would look like a flat line over all frequencies. A typical one, though, will have variations of from 3 to 30 decibels ("dB") over most of the frequency range, and often dropping off at very low bass frequencies. [25]

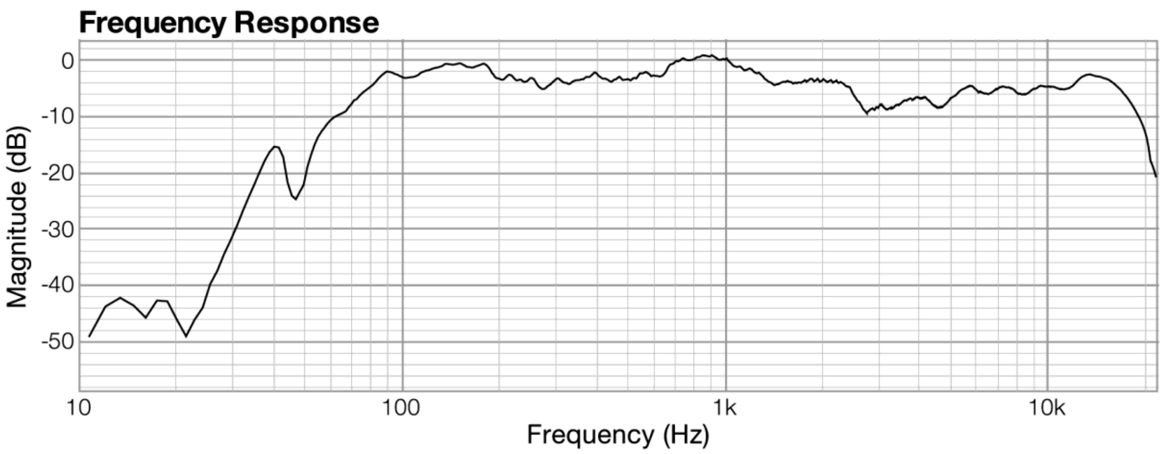


Figure 3.2 - Frequency Response Graph [26]

A reliable measurement of the frequency response of loudspeakers is basically only possible under professional conditions, since several factors can influence the measurement. These factors include the microphone used as the measuring device (the frequency response of the microphone must also be taken into account), the position of the microphone and the acoustic characteristics of the room, among other variables. The test system would normally be placed in an anechoic chamber (a room designed to completely absorb sound reflections). The sound reflections that occur in any living space can have a great influence on the measurement of the amplitude. [27]

3.3. Directivity

Directivity is the term used to describe the way a speaker's frequency response changes at off axis angles. [28] In other words, how effective the speaker is at taking the sound it produces and sending it in one particular direction instead of all directions. [29] Directivity is a measure of the directional characteristic of a sound source. It is often expressed as a Directivity Index in decibels, or as a dimensionless value of Q. [30] The Q of a loudspeaker is the ratio of sound pressure-squared at one angle (a single point in space) from the loudspeaker to the average sound pressure-squared radiated from the loudspeaker. Since it only considers one angle (a single direction), many Q ratings are required to fully characterize the radiation from a device. A useful simplification is to consider the "axial" Q – the directivity factor for the "on-axis" position of the loudspeaker. This is usually the point of highest sound pressure deviation, and the optimal listener position. Q can be expressed in decibels by taking its base 10 logarithm and multiplying by 10. This is the loudspeaker's directivity index and describes the loudness advantage at a particular angle over the same total acoustic energy radiated omnidirectionally. [31]

A loudspeaker that is a high directivity device is commonly called a "long throw" device. A speaker with low directivity is a "short throw" device. [29] The directivity of all traditional loudspeaker devices is fundamentally limited by nothing more than the size of the source compared to the wavelengths it is generating. A large loudspeaker will be more directive than a small loudspeaker, or a loudspeaker specified at higher frequency (smaller wavelength) will also have more directivity. No amount of phasing, shading, focusing, or other method can overcome this fundamental limit; in fact, any of these methods will always reduce directivity. To make any loudspeaker more directional, one can only make the loudspeaker physically larger, either by creating a large active surface, or, as an equivalent, by using a large number of small speakers, driven in aggregate to form a large radiating surface. [32]

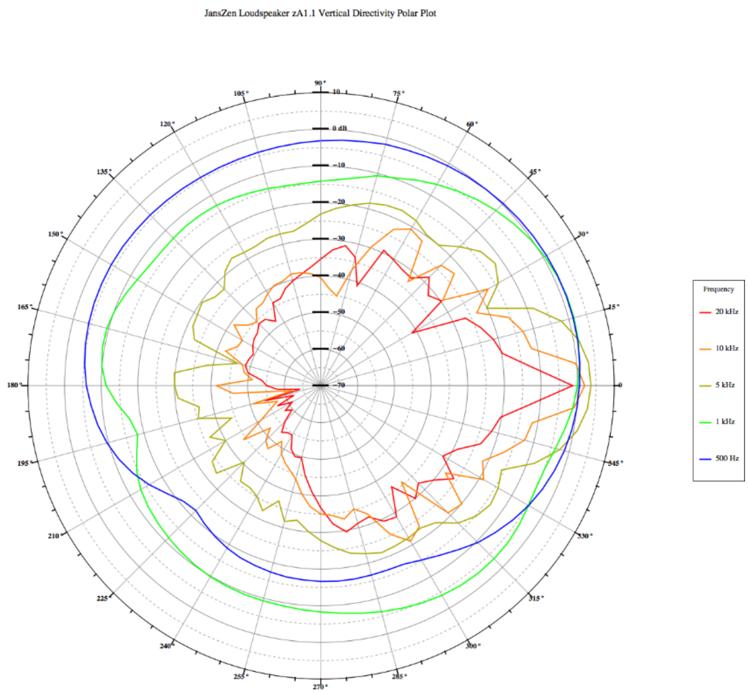


Figure 3.3 - Directivity Graph [33]

3.4. Sensitivity

The sensitivity specification of a loudspeaker is one of the more useful specifications. It will give you a general idea of how loud the speakers can play and how much power will be required to achieve desired listening levels. Speaker sensitivity is the measurement of the amount of sound pressure that the loudspeaker will produce (i.e. how loud it will play), at a distance of one meter, when only one Watt of power is applied from an amplifier. Sound pressure is typically measured in Decibels (dB). [24]

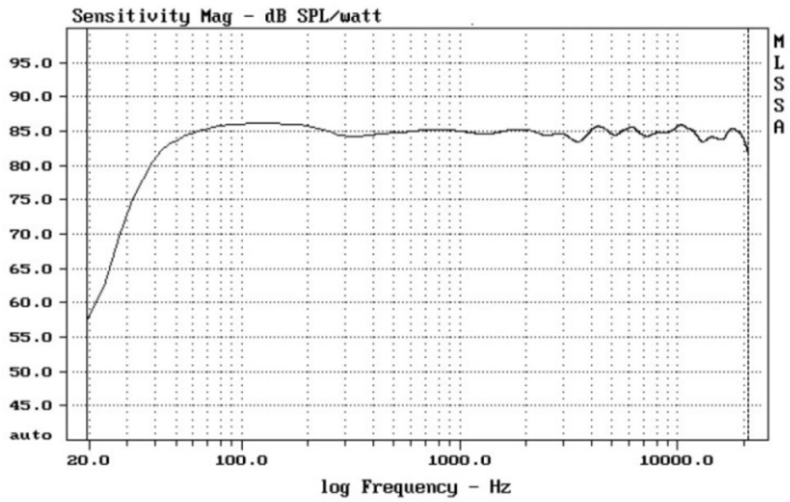


Figure 3.4 - Sensitivity Graph [34]

3.5. Impedance

Impedance is the speakers' resistance to power or impeding the flow of power. Generally speaking, the majority of today's loudspeakers are rated at an Impedance of 8 ohms. And this is the standard specification for which many amplifiers are also rated—meaning their power delivered into 8-ohm loads. [35] It is important that the impedance of a loudspeaker be matched with its amplifier. Too high impedance will not allow the amplifier to transfer energy into them. Too low impedance will draw too much power from an amplifier without properly transferring the electrical signal, which will cause the amplifier to prematurely clip and create harmonic distortion that is annoying and damaging. [36]

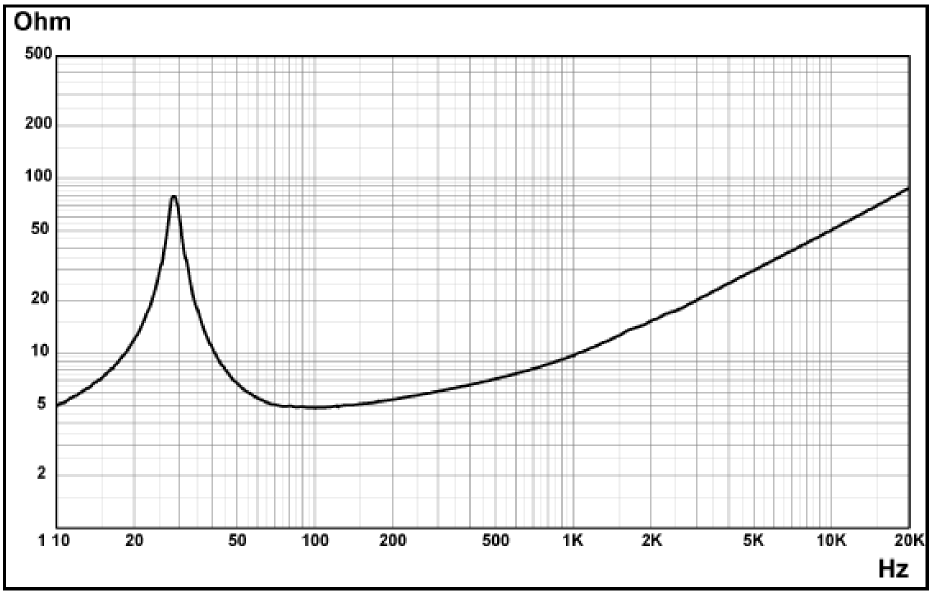


Figure 3.5 - Impedance Graph [37]

3.6. Distortion

Distortion is a measurement of inaccuracy exhibited when a signal of single frequency is presented. Usually measured as a percentage, it gives an indication of how much of a signal heard is the result of speaker imperfections.

Several things can cause distortion, and most can be avoided through careful speaker system design. Most speaker system elements are made using an electromagnetic or electrostatic linear motor assembly connected to a cone. In this arrangement, the motor should be carefully designed to ensure that it moves in a controlled fashion, and that it moves exactly with the signal presented to it. The cone attached to the motor may not be rigid enough and may begin to vibrate at harmonic frequencies when presented with certain signals. Further, the mounting of the cone requires a suspension, which may prevent the cone from free movement and may cause the cone to twist or limit its motion more in one direction than the other. Therefore, even though the motor assembly is simple, it is important to engineer and build the motors carefully, and to choose only those that are of high quality. [36]

The distortion can be divided into two principal parts:

1. Linear distortion
2. Nonlinear distortion

The term *linear distortion*, which might sound rather confusing, implies that the output signal has the same frequency content as the input signal. In this distortion, it is the amplitude and/or phase of the output signal that is distorted. In contrast, the term *nonlinear distortion* suggests that the output signal contains frequency components that are absent in the input signal. This means that the energy is transferred from one frequency at the input to several frequencies at the output. [38]

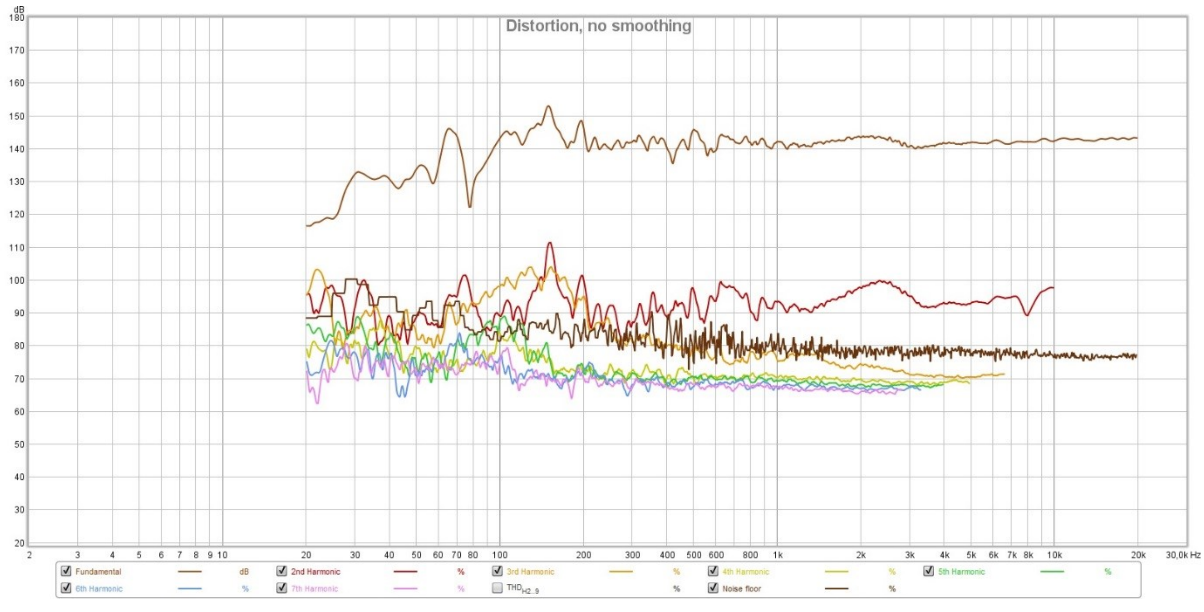


Figure 3.6 - Distortion Graph [39]

3.7. Power Handling

It is a measure of how much power can be presented to a speaker drive without causing it to be damaged. Often times, cone travel is impaired by the speaker drives suspension at power levels much lower than its rated maximum, so that fact must be taken into consideration. It is generally safest to assume that distortion rises dramatically between 50% and 70% of a speaker drive's maximum rated power handling capacity. [36] Power handling is measured in watts.

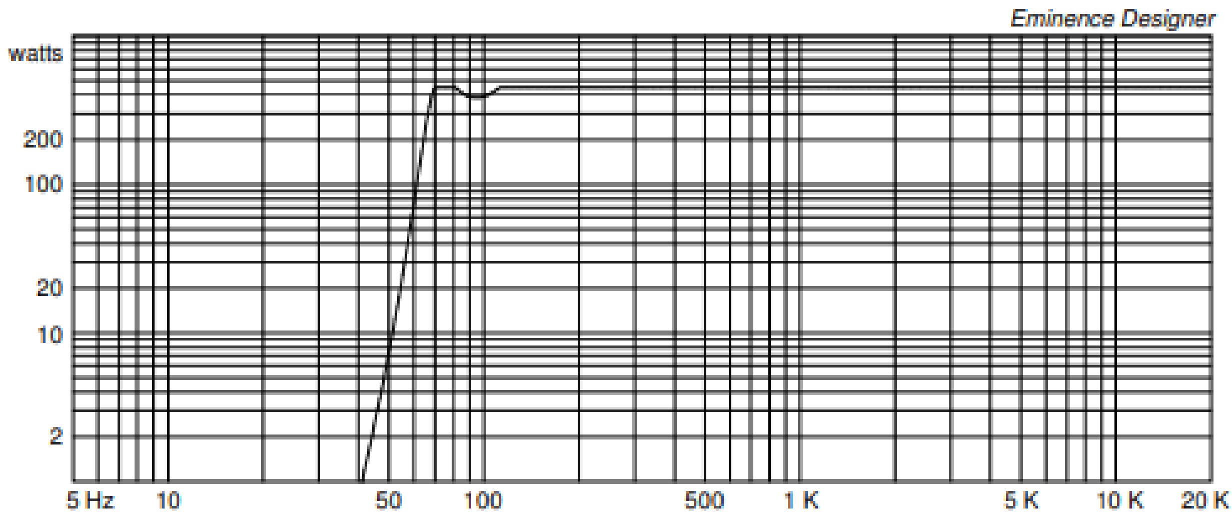


Figure 3.7 - Power Handling [40]

3.8. Phase Shifts

Phase shifts are differences in time between two components of an audio signal. For example, when a snare drum is played, two distinct sound components are created at the same time – a low frequency sound that is the result of the resonation of the drum and a higher frequency sound that is caused by the impact of the drumhead and the vibrations of the snares. If the two signals are separated and sent to different speaker systems elements, and if one of the elements is much closer to the listener than the other, then one component of the sound will reach the listener sooner than the other.

Phase shifts are also introduced to a small degree by crossover components and by the electrical characteristic of the driver motors themselves. However, these shifts are much less than one cycle and are negligible. The only problem that can arise from these small phase shifts is the possibility of two speaker drives receiving a portion of a signal, and then being shifted near 180 degrees so that the sound output of the individual drivers partially cancels or modifies the other. [36]

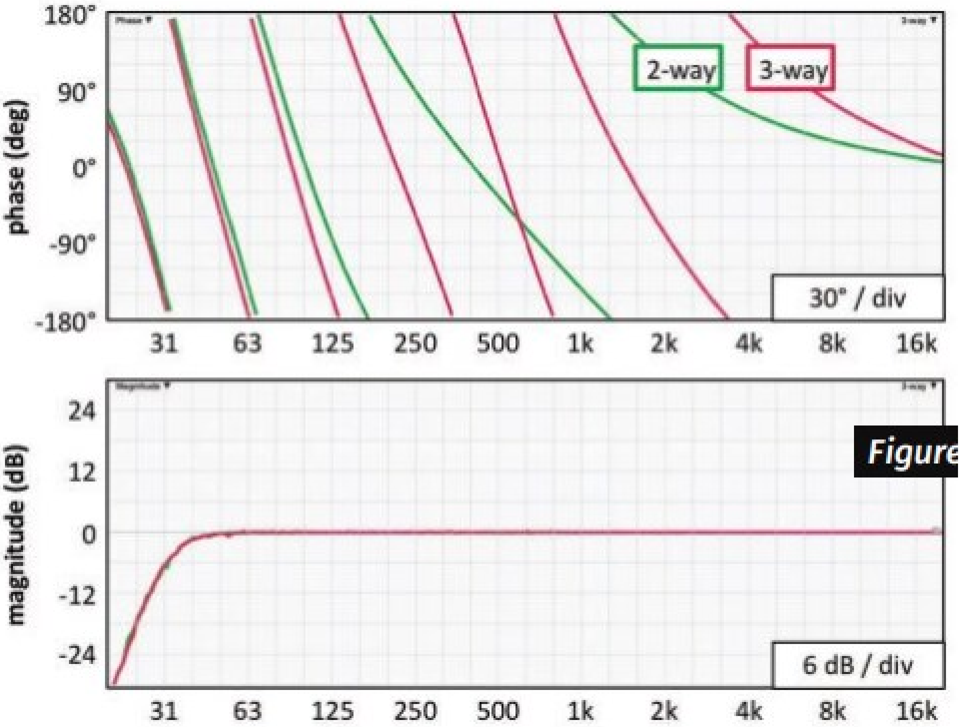


Figure 3.8 - The typical phase shift for a regular 2-way or 3-way loudspeaker [41]

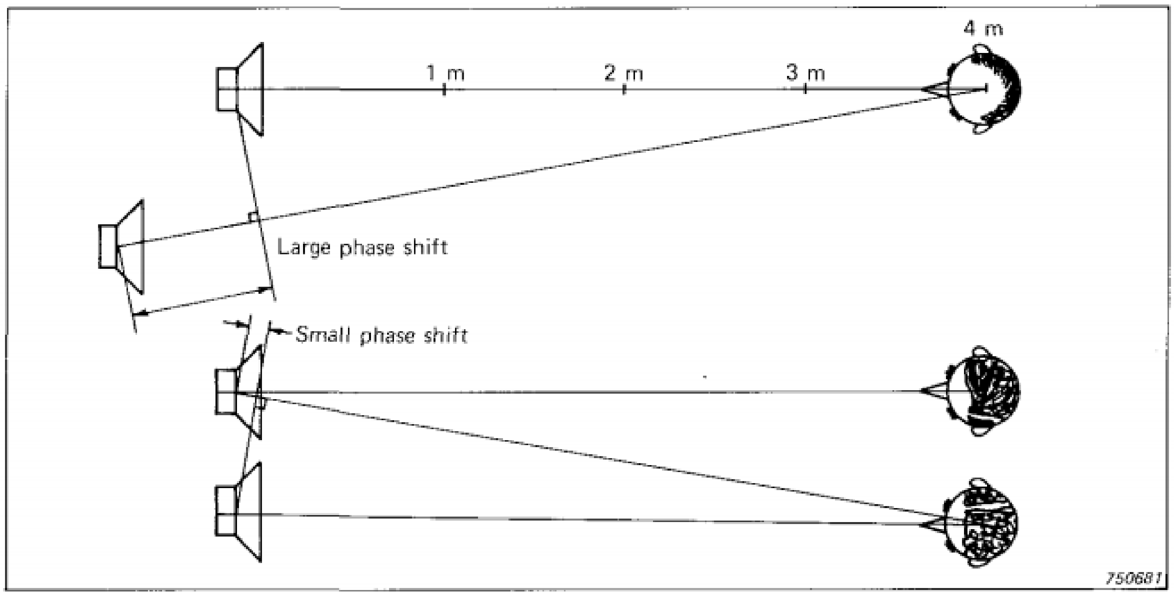


Figure 3.9 - Axial movement of the speakers introduces considerably more phase shift than even greater movements of the listener away from the axis [42]

4. SIMULATIONS

Simulations are made by means of VituixCAD software. The measured datas which are used for simulations are obtained during measurements via LabShop Pulse software. Firstly, we measure the SPL. These measurements are made for individual drivers of loudspeakers. After we obtain the SPL measurement results, we use them to measure frequency response and directivity characteristics. The SPL results are measured not only from the reference axis (right in front of loudspeaker) but also the different angles which are starting from 0(reference axis) to 90 degrees in 10 degrees increment. The microphone has the same height as the twitter and the distance between them is 1m.

For the frequency response and directivity, we use SPL vs. frequency. The following graphs shows us the simulated frequency response and directivity of the loudspeaker based on real data from measurements.

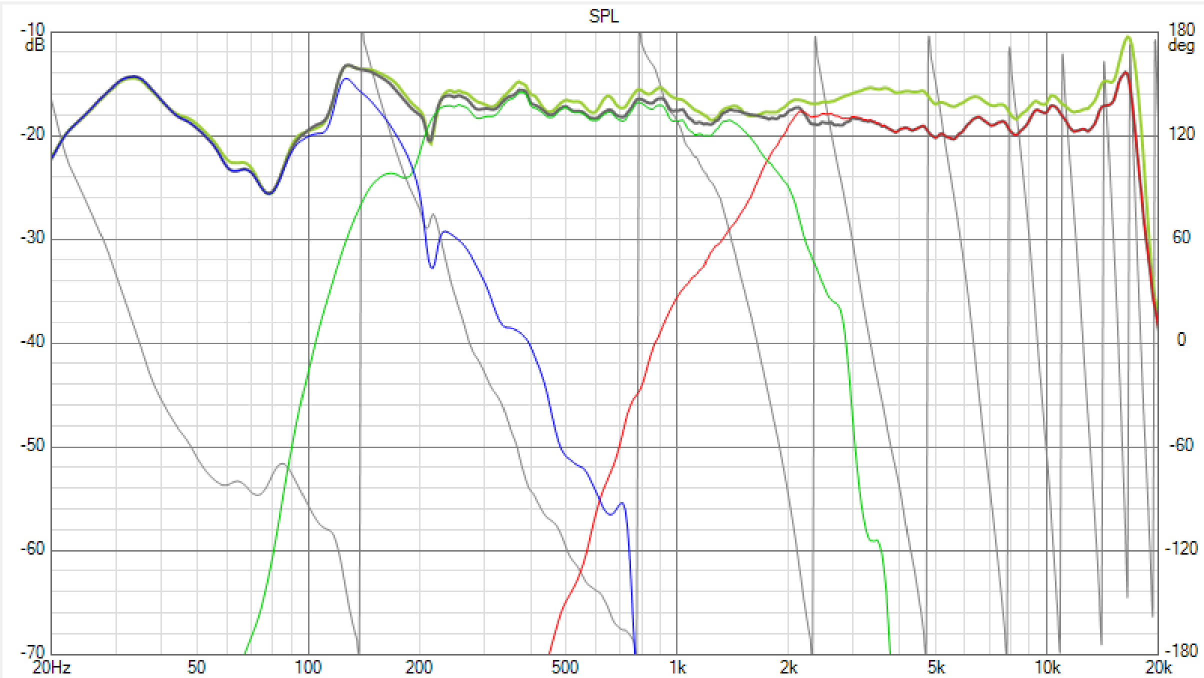


Figure 4.1 – Frequency response graph

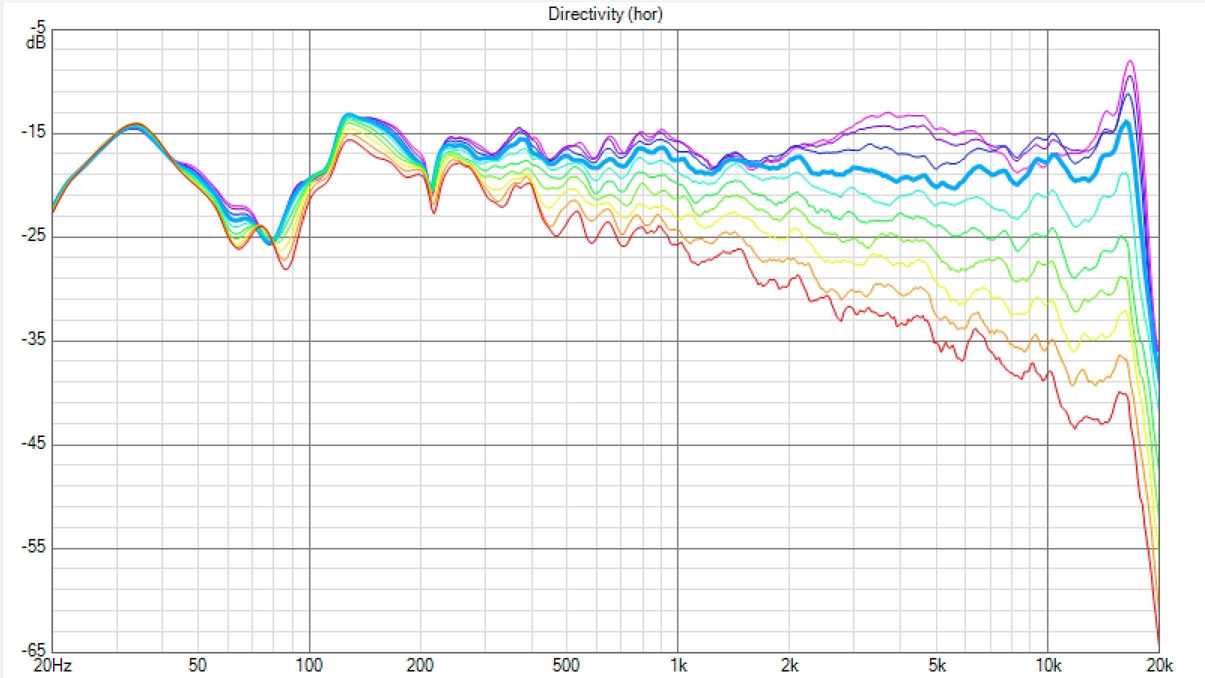


Figure 4.2 – Directivity graph

For the sensitivity, we measure the output voltage and current from amplifier and SPL from loudspeaker. And we must know the impedance of the loudspeaker. We can see the impedance graph below.

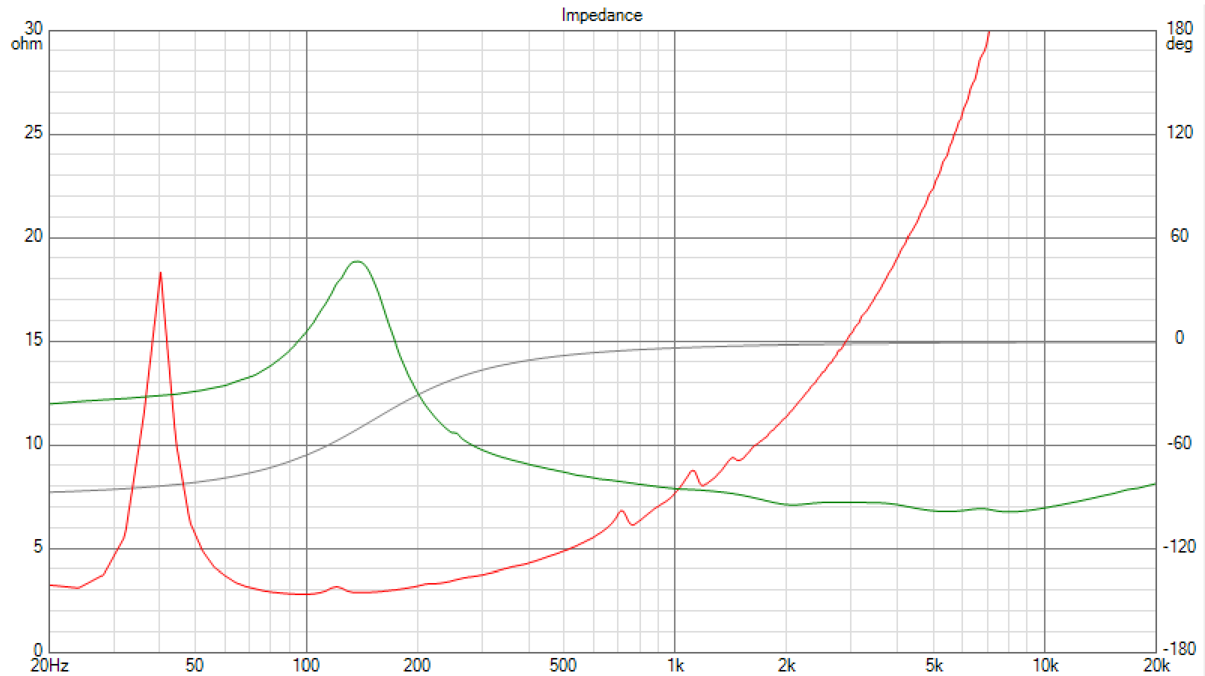


Figure 4.3 – Impedance graph

5. METHODS FOR FRF MEASUREMENT AND AMPLITUDE FREQUENCY CHARACTERISTICS

5.1. Steady Harmonic Signal Excitation (Pure Tones)

The first method we used to obtain the amplitude frequency characteristic of the loudspeaker system being measured was the method of excitation by steady harmonic signal or pure tones.

This simplest procedure is also implemented in the professional software package PULSE, originating from the Danish company Brüel & Kjær. Specifically, it is the "Loudspeaker Test" module, which generates pure tones in ascending order corresponding to the middle frequencies of third-octave bands, while real-time data from the measuring microphone is analyzed and subsequently graphically marked. This method works with discrete frequencies that in subsequent steps cover the entire frequency range in which the amplitude frequency response is to be evaluated.

However, the simplicity and straightforwardness of this method is dearly purchased by its time-consuming and economical inefficiency, as it is necessary to measure over the entire range of the frequency domain and it is usually also a prerequisite to own the appropriate software modules. Due to the fact that the measurement was evaluated only in thirty frequencies of the third-octave spectrum and the resulting curve of the amplitude frequency characteristic had to be interpolated, its below-described process (Fig. 5.1) can be considered rather indicative.

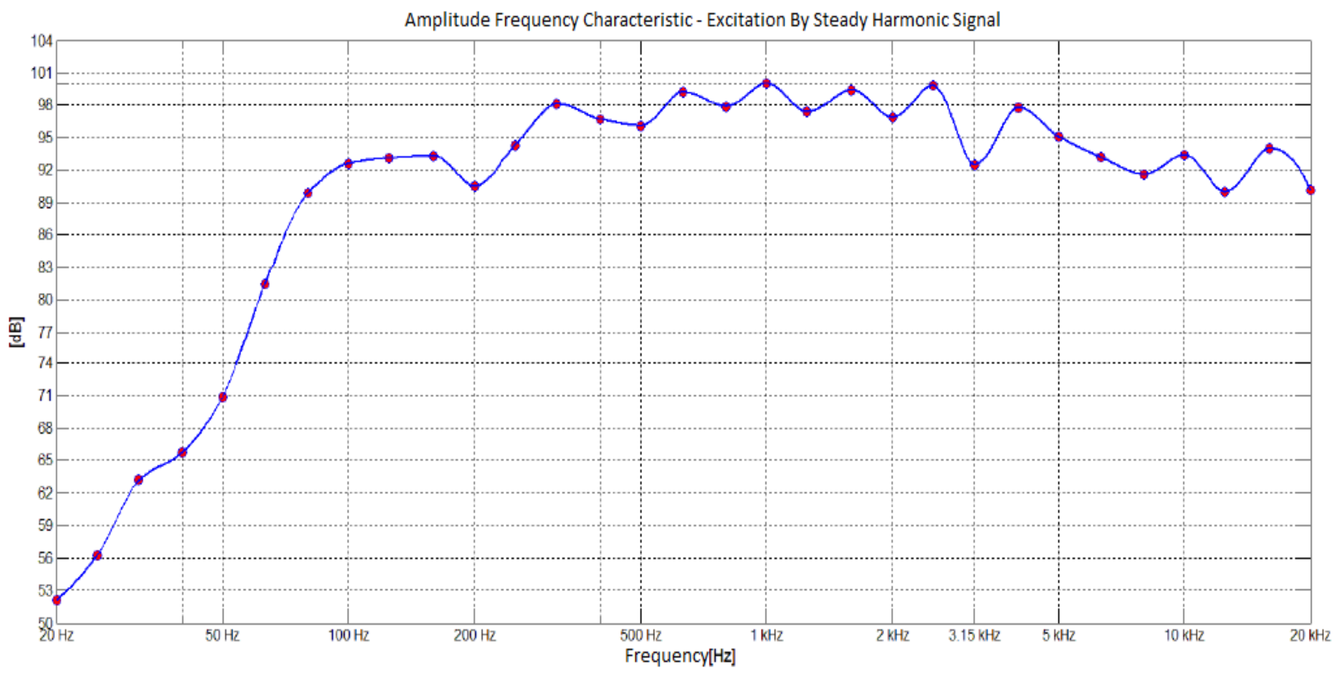


Figure 5.1 - Amplitude frequency characteristic curve – excitation by pure tones

Table 5.1 - Basic measurement parameters for pure tone excitation

Method Name	Excitation by steady harmonic signal
Frequency Measuring Range	20 Hz – 20 kHz
Measurement Time	180 s
Number of Freq. Characteristics	30
Measuring Distance (Microphone-Speaker)	1 m
Temperature and Relative Air Humidity	20 ° C; 48%

Mathematical Notation:

$$x(t) = \sin(\omega \cdot t) \qquad (5.1)$$

$$y(t) = A \cdot \sin(\omega \cdot t + \varphi) \qquad (5.2)$$

Excitation by the harmonic input signal $x(t)$ causes the speaker system response, which is the output signal $y(t)$ characterized by the same frequency. However, the output harmonic signal may have a different amplitude and may also be phase shifted relative to the input signal.

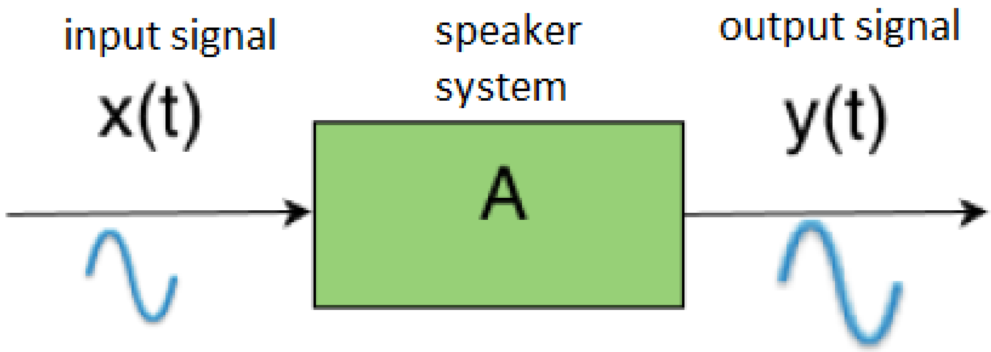


Figure 5.2 - Schema of the method complementing the mathematical notation

5.2. Method of Evaluation Using Random Noise

The frequency response function (FRF) of the linear dynamic system can also be measured by comparing the amplitude and phase of the harmonic signal at the input and output of the measured system. In our case it was the input voltage signal $x(t)$ measured by the amplifier and the output signal $y(t)$, the sound pressure level measured by the microphone. A linear dynamic system for us was a loudspeaker system, whose linearity ensures that the individual frequency components do not change when passing through.

The speaker parameters are often measured with a sine wave harmonic signal. However, loudspeaker systems are primarily intended to reproduce a music signal that is composed of different types of signal (basic tone, harmonic tone, and noise). Thus, a loudspeaker measured only by a harmonic signal can exhibit a different reproduction behavior if it does not have full control over the membrane mass and its proper movement.

A more convenient method was chosen, at which we will provide a test signal containing all the frequency components from the frequency range for which FRF is detected. Specifically, the test signal used white noise, which falls between the signals of a wide range and shows the frequency of amplitude of its components with a tolerance of 3 dB.

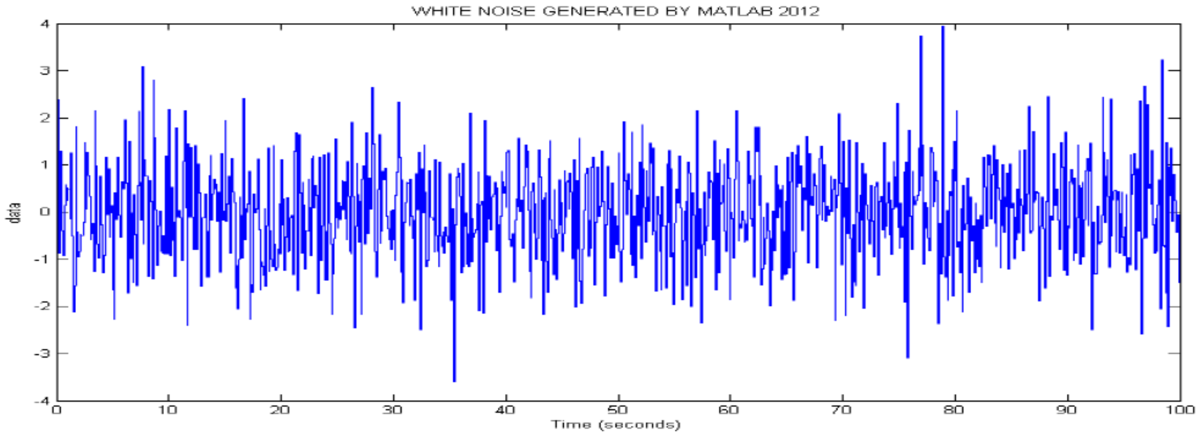


Figure 5.3 - White noise signal generated by MATLAB

White noise can be characterized as a random signal with constant power spectral density for all frequencies (5.3). It is a time-stabilized signal containing harmonic frequencies throughout the audible spectrum, with the same power in all constant percentage bands (CPB). Its name stems from an analogy with white light, which also contains all the light frequency components. Obviously, if the noise at all frequencies had a nonzero power, its total power would have to be infinite. In fact, we call it white if its spectrum is flat at a defined frequency range, mostly within the audible range.

$$S_{XX}(\omega) = \int_0^\infty \sigma^2 \cdot \delta(\tau) e^{-j\omega\tau} d\tau = \omega^2 \qquad (5.3)$$

A white noise signal is generated by the random number generator in mathematical modeling. In analogue devices, the amplification of the noise elements of the electronic elements that are generated by thermal stress is used for its generation.

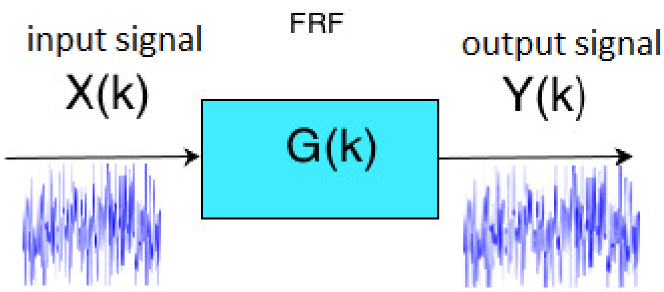


Figure 5.4 - FRF calculation using random noise

The practical FRF calculation by this method can be divided into the following steps according to [43]:

- Evaluation of the DFT from the time record of the N-length input $X(k)$ and the output $Y(k)$ of the tested loudspeaker system
- Share of two complex numbers $Y(k) = |Y(k)| \cdot \exp(j\beta)$ and $X(k) = |X(k)| \cdot \exp(j\alpha)$ by equation (5.4)
- Frequency transfer function is obtained by repeating the calculation of the ratio (5.4) in the range of index k according to (5.5)

$$G(k) = \frac{Y(k)}{X(k)} = \frac{|Y(k)|}{|X(k)|} \cdot \exp\big(j(\beta - \alpha)\big) = A(k) \cdot exp\big(j\varphi(k)\big) \qquad (5.4)$$

$$k = 0,1,2,\ldots,N/2 \qquad (5.5)$$

$A(k)$ is the frequency response amplitude and $\varphi(k)$ its phase shift.

The random noise assessment also utilized the extensive functionality of the PULSE analyzer from Brüel & Kjær and the B&K Pulse - Labshop software, which again served as both generator and analyzer. The amplifier of the input and output signal was then numerically compared in the analyzer. The evaluation of this record was repeatedly performed due to the requirement to eliminate random noise, which was achieved by averaging the obtained waveform.

As a result, the FRF was already detailed, which can be considered as a precise description of the measured dynamic system based on the chosen method (Fig. 5.5).

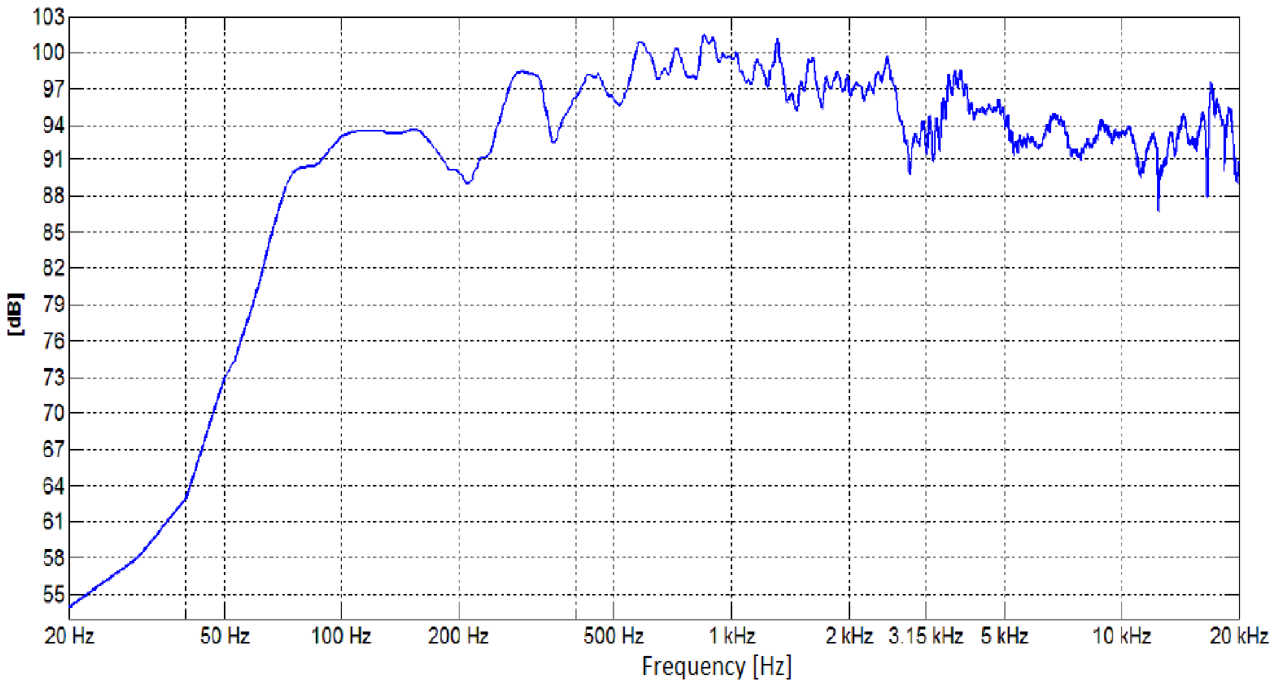


Figure 5.5 - FRF waveform obtained by random noise (white noise)

However, the disadvantage of this method is the length of measurement due to the requirement of averaging, which may lead to the inclusion of errors which are stemmed from the reflection of sound waves from the floor of the semi-anechoic room. This error can partially affect the resulting FRF.

Table 5.2 - Basic measurement parameters for random noise evaluation

Method Name	Evaluation by Random Noise
Frequency Measurement Range	20 Hz - 20 kHz
Measurement Time	10 s
Measuring Distance	1 m
Temperature and Relative Humidity	20 C; %45

In this project it is wanted to get linearity in 20Hz and 20kHz range. Speaker with big cone helps to get a good linearity. And if we want to have good linear low frequencies, we need to use big surface of cone on low frequencies.

6. DESIGN OF MEASUREMENT METHOD

Multiple methods and approaches are commonly used to measure the transfer characteristics of loudspeaker systems. The choice of the method depends mainly on the desired accuracy of the result, which also determines other assumptions.

To measure the characteristics, the enthusiasts who make the loudspeaker systems in amateur conditions will also need their PCs with an external sound card and the necessary software. When measuring the response characteristics directly by the manufacturer, obviously high demands are made on the accuracy of the output data and at the same time all methodology and labeling must comply with the national technical standards. For experimental purposes, a research experimental noise laboratory of VŠB-TUO was available, together with all its technical equipment, which felt top measuring microphones, amplifiers and analyzers.

6.1. Design of Measurement Chain

For the laboratory measurement of the frequency transfer function, a measurement chain was designed, which is illustrated in the following diagram. (see figure 6.1.)

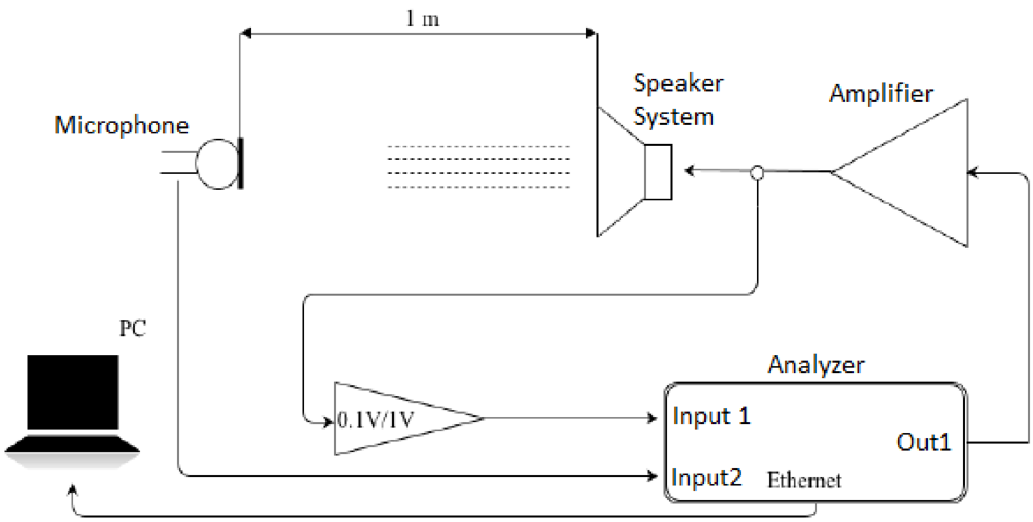


Figure 6.1 - FRF measurement scheme and amplitude frequency characteristics of loudspeaker system

The measurement chain for all the methods used below is always based on this scheme, with the sequence of each step being as follows.

The analyzer, which also serves as a generator, is the source of the amplifier signal, where it will be linearly amplified. Subsequently, this voltage signal is fed to a loudspeaker system in which an acoustic-sound wave is converted to a signal by means of electroacoustic transducers. It spreads through the acoustic environment, in our case air, at a distance of one meter, where an electroacoustic converter - a condenser microphone - is placed. This converts the acoustic signal back to electrical, which is then routed through the coaxial cable back to the beginning of the string, i.e. to the analyzer. Here is an analogue digital converter that allows you to work with the measured data further. Finally, the data is transmitted in digital form via an Ethernet cable (LAN interface) to the PC, where the final processing and evaluation of the measured results (post-processing) takes place.

6.1.1. Analyzer

For the purposes of this master thesis, a portable analyzer PULSE type 3560-C by Brüel & Kjær was chosen. It is a 4-channel data collection unit that is designed for highly professional measurement of acoustic or vibration quantities. The advantage of this analyzer is also the built-in TEDS technology, which is used for automatic recognition of connected sensors and subsequent loading of the necessary parameters.

Figure 6.2 – PULSE signal analyzer from Brüel & Kjær - type 3560 C

The analyzer does not contain any controls as such, so it is necessary to connect the operating PC with the LabShop Pulse software from B&K, where the analyzer settings can be made together with the recording and evaluation of the measured data. The analyzer specifications are shown in the following table. (table 6.1)

Table 6.1 - Basic parameters of PULSE 3560-C analyzer

Number of inputs	4
Number of generator outputs	2
Sampling frequency	65 536 Hz
Frequency measuring range	50 Hz – 25,6 kHz
Distinction	16 bits

6.1.2. Microphone

To measure the sound pressure level, a 1/2″ condenser, omnidirectional microphone from B&K of the type 4189-A-021 is selected, which is designed for free field measurement. This professional microphone is also equipped with TEDS automatic identification technology, which is fully compatible with our analyzer. The interconnection of these two components is accomplished by a coaxial cable provided with BNC connectors.

Unless directionality information is desired, all testing should be performed with the microphone directly on the axis of the speaker. In this thesis, measurements were made in 10 degrees increments.

Figure 6.3 - Measuring microphone B&K type 4189

Table 6.2 - Basic microphone parameters

Sensitivity	50 mV/Pa
Frequency range	20 Hz ÷ 20 kHz
Dynamic range	16.5 – 134 dB
Temperature range	-20 ÷ +80°C
Nonlinearity	±2 dB

6.1.3. Amplifier

To amplify the generated signal, we had a Korean firm TIRA of the BAA 60 type (figure 6.4) available. This universal single-channel linear amplifier is designed for laboratory use in particular for amplifying vibration exciters or generated acoustic signals. It has a digital output current and voltage indicator on the customs side. On the back side there is an analog output of an internal voltmeter, which we used to connect directly to the analyzer using a coaxial cable provided with BNC connectors. This connection between the amplifier and the analyzer was needed to record the progress of the input signal, which was still used to obtain the FRF by pulse response.

Figure 6.4 - Linear amplifier TIRA BAA 60

Table 6.3 - Parameters of TIRA BAA 60 amplifier

Max. Output Power	60 VA
Frequency Range	DC ÷ 20 kHz
Max. Output Voltage	16 V
Max. Output Current	3,9 A
Harmonic Distortion	< 0,1 %

6.1.4. Speaker System

Our loudspeaker system has 4 drivers. There are 1 tweeter (B&C Speakers DE250 8/ohm series), 1 mid-range speaker (B&C Speakers 8MBX51-8 series) and 2 woofers (Eminence LAB12 series).

6.1.5. Semi - Anechoic Room

Of course, the environment into which the measuring chain itself was placed also played a very important role in the measurement. Due to the available options, a semi- anechoic room was used for the measurement, which is part of the experimental noise laboratory of Technical University of Ostrava. This certified semi-anechoic measuring instrument allows to perform standardized measurements on objects not exceeding the dimensions given in the table below. (table 6.4)

The basic element of this room is the wedges of absorbent material, which are subsequently provided with a resilient coating stabilizing their shape and preventing the loss of mineral fibers. The shape and dimensions of these wedges were calculated by the construction company based on the required chamber size and determine the frequency characteristics of the room. The semi-anechoic room was designed according to the CSN EN ISO 3745 series of standards.

Table 6.4 - Parameters of the semi- anechoic room of VŠB-TUO

Room Dimensions [Width; Length; Height]	9,5 m; 8,5 m; 2,6 m
Frequency Attenuation	from 100 Hz
Absorbent Material	Porous mineral fiber material

To determine the transfer characteristics correctly, it is necessary to know the properties of the given acoustic environment in which we want to measure these functions, as mentioned earlier in chapter 3.1.1. For these purposes, a significant acoustic quantity serves as a reverberation time, which is the most widely used parameter when determining the sound decay in a particular acoustic space. The reverberation time is then called the time after which the sound pressure level is reduced by 60 dB, which is a decrease of 10^6 times its original value. [44]

Figure 6.5 - Photograph from measurements taken in the semi-anechoic room of VŠB-TUO

The reverberation time is uniform for the entire room volume (in the case of an ideally diffuse acoustic field) and depends on the absorption properties of the material used therein. Measurement of this quantity does not fall within the scope of this work, but we must not forget its significance in the subsequent measurement of frequency characteristics.

7. RESULTS OF MEASUREMENTS

In this section, it is shown that the measurement on the prototype of the loudspeaker. After that, the results which are obtained from simulation and prototype are compared. Before the results, we can see the prototype of the loudspeaker in the testing room. (see figure 7.1)

Figure 7.1 – Prototype of loudspeaker during testing

7.1. Results of Measurement On Prototype

By means of the measurement on prototype, we obtained directivity, frequency response and impedance charts. These charts are shown in the following figures.

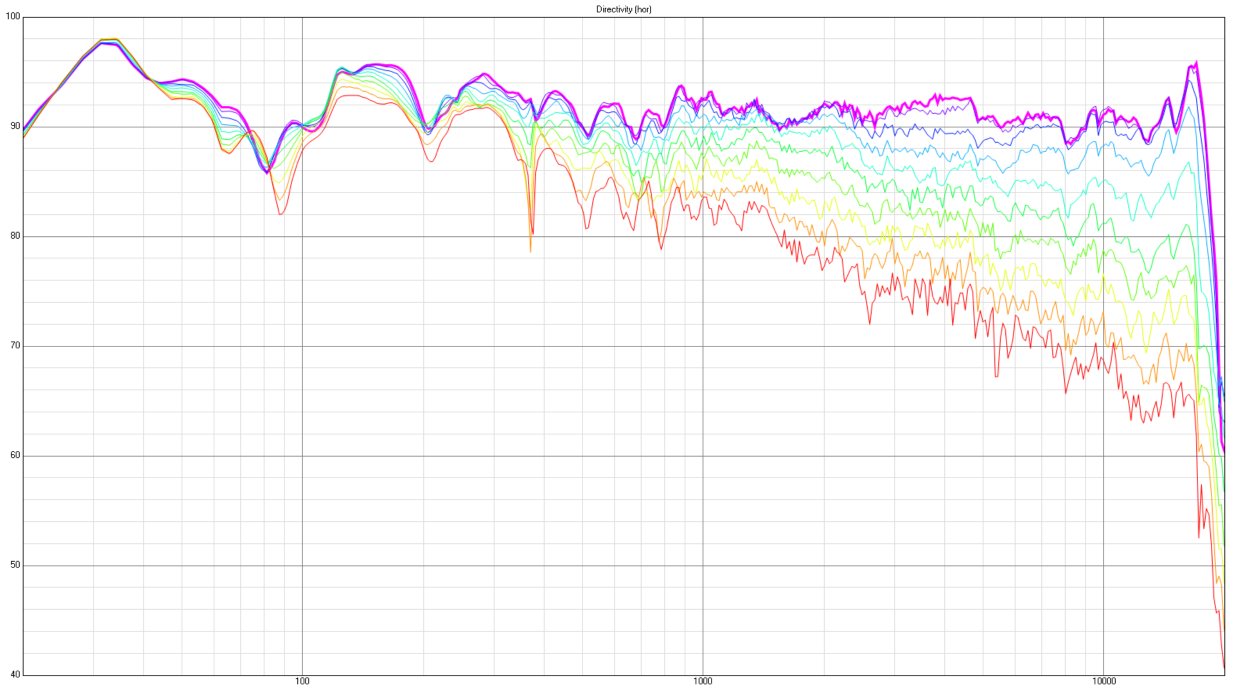


Figure 7.2 – Directivity graph of the prototype

The pink line on the frequency response graph, shows the average SPL - sensitivty. It is measured 92 dB/W/m.

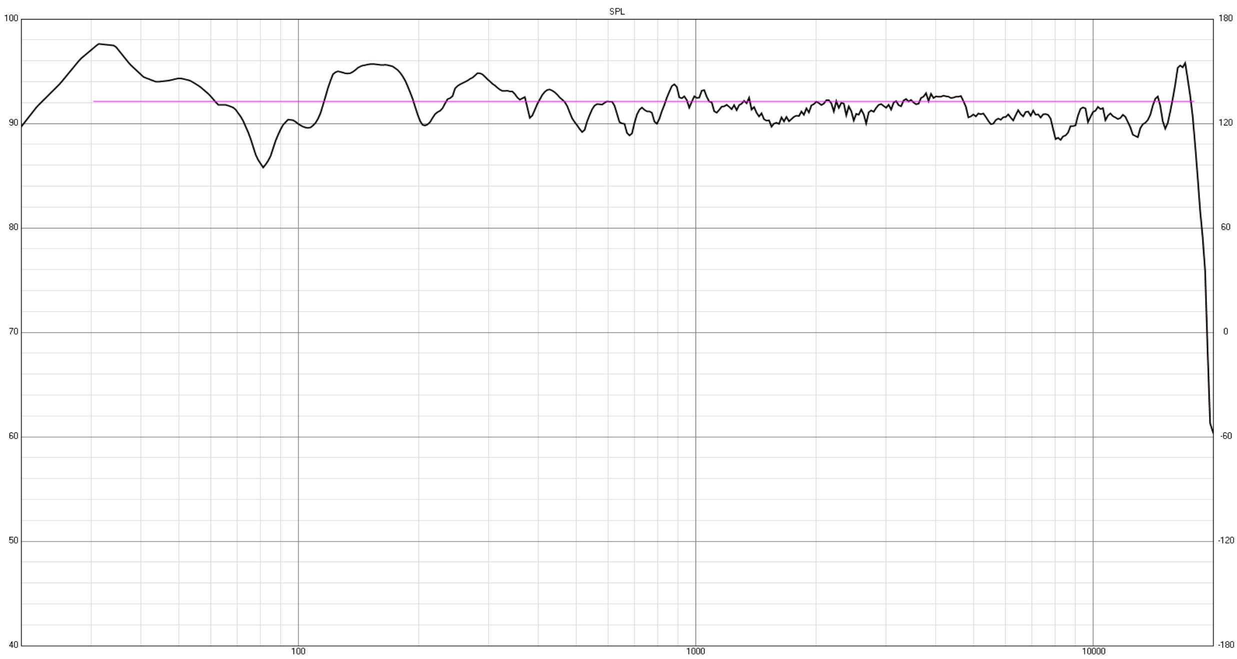


Figure 7.3 – Frequency response graph of the prototype

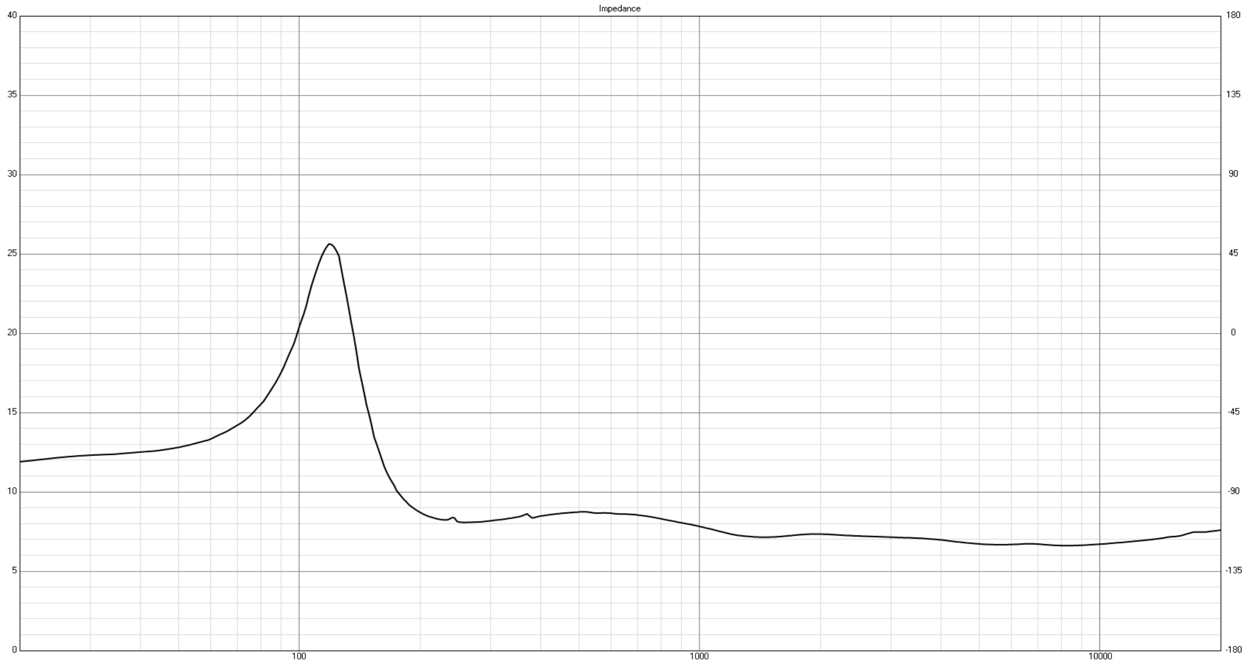


Figure 7.4 – Impedance graph of the prototype

7.2. Comparison of Simulation and Prototype Results

In this section, it is compared that the results of simulation and prototype. Firstly we see the comparison of sensitivity results of prototype(see figure 7.5) and simulation (see7.6). The results have semi-linear shapes. They are very similar and it means that it is successful.

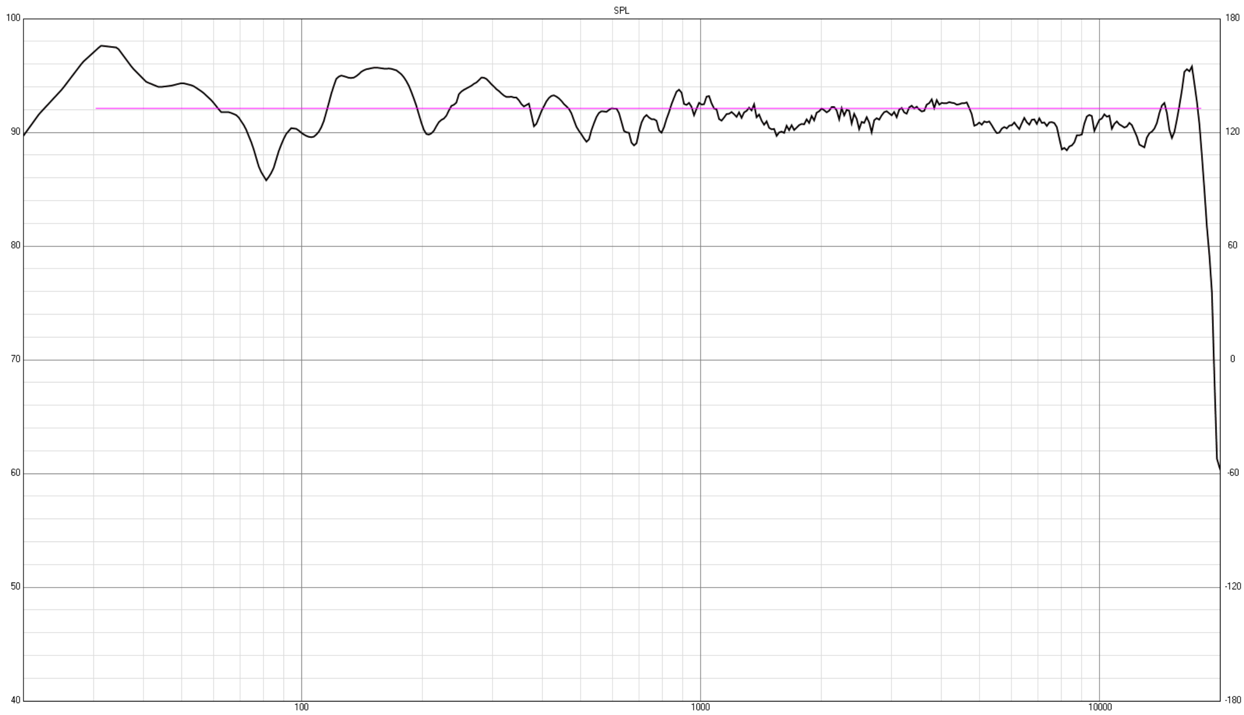


Figure 7.5 - Frequency response graph of the prototype (used for sensitivity comparison)

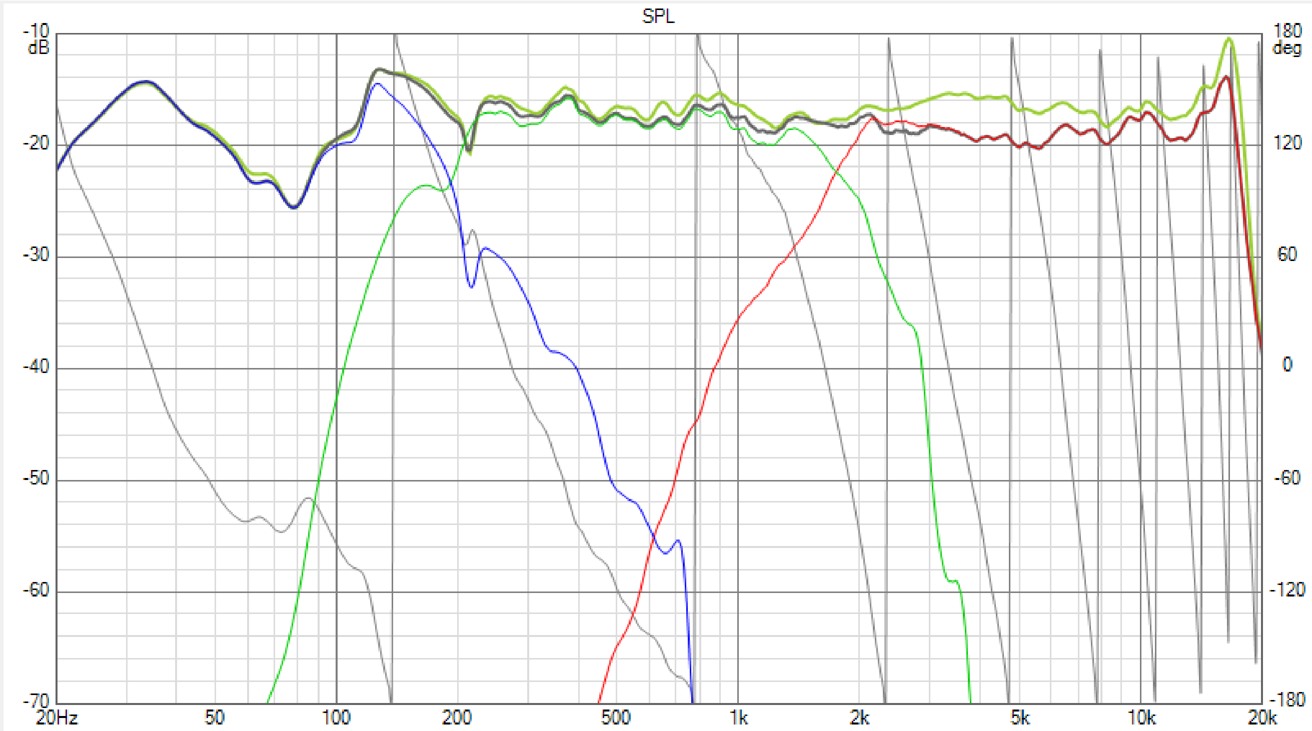


Figure 7.6 – Frequency response graph of the simulation (used for sensitivity comparison)

Then the comparison of directivity is shown above. Their directivitiy shapes are almost same and it means that it is successful. We see when the angle is increasing, on the contrary the SPL is decreasing. (higher angle – lower SPL)

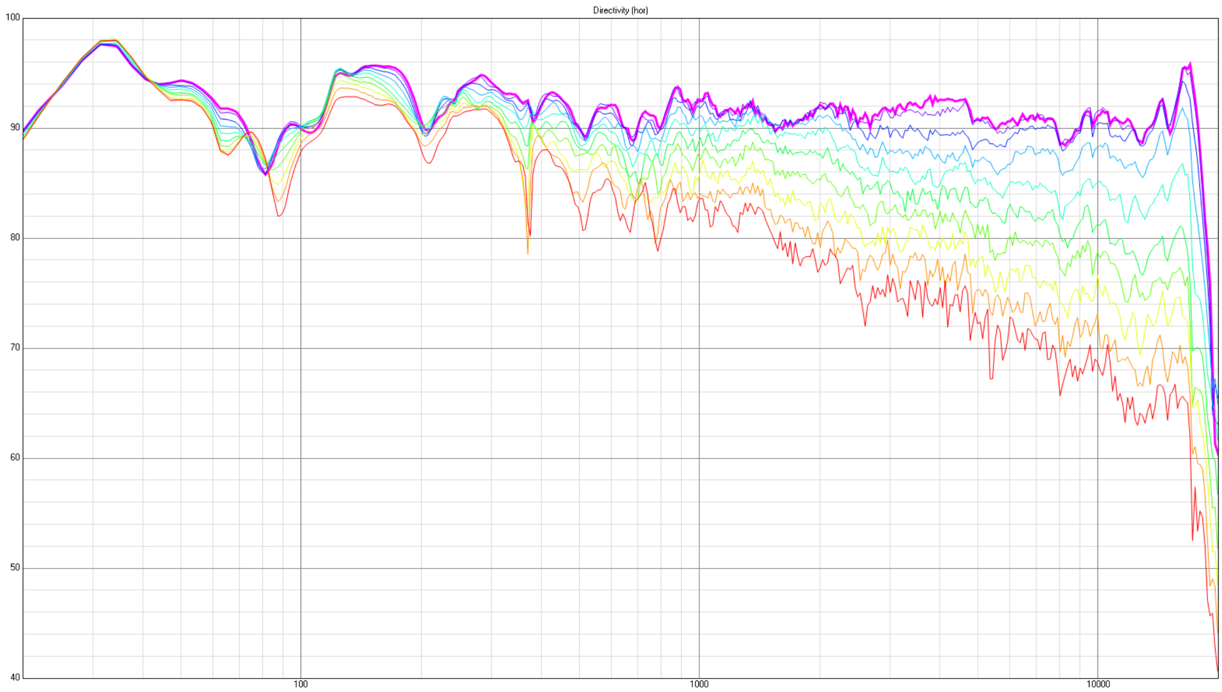


Figure 7.7 – Directivity graph of prototype (used for directivity comparison)

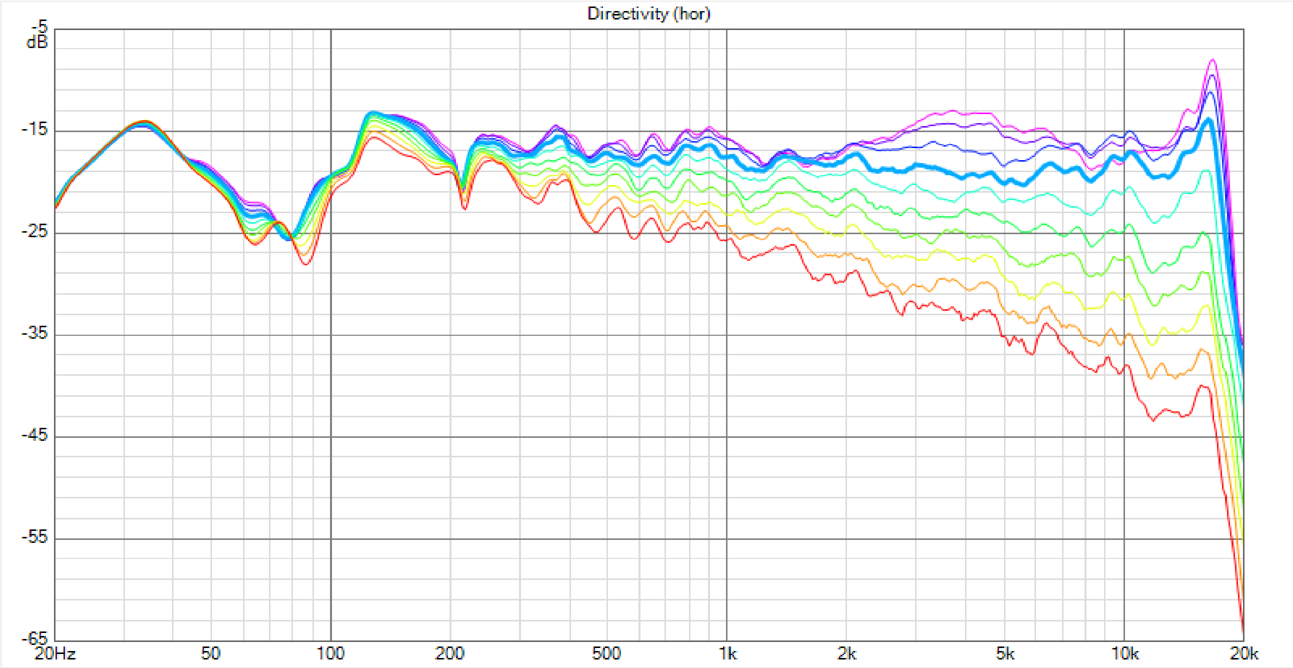


Figure 7.8 - Directivity graph of simulation (used for directivity comparison)

And finally, after all the design procedures and measurements, we obtain our loudspeaker. It is shown that the last version of loudspeaker below. (see figure 7.9 and 7.10)

Figure 7.10 - Front view of the last version of loudspeaker

Figure 7.9 - Cross view of the last version of loudspeaker

8. CONCLUSION

The aim of this work is to give ideas how to design and optimize a loudspeaker. This document has essentially composed of what is required in the speaker cabinet or enclosure design and analysis.

The theoretical introduction of the master thesis clarified the relationship between sound and loudspeaker system, parts of the loudspeaker system and types of loudspeaker systems.

Thereafter, in the design chapter, it is mentioned that appropriate layout of the loudspeaker system. In addition, designed enclosure and crossover network are shown.

Then, characteristics of the loudspeaker is explained. Some of those characteristics are used for measurements. And those characteristics allows us to understand whether the measurement results obtained from the simulation and the prototype correspond to each other.

The main purpose of doing simulation is to obtain the real design of crossover network. Simulation is completed by means of VituixCAD software, and results are obtained. In connection with this, experimental measurement of prototype of the loudspeaker system is carried out via LabShop Pulse software from B&K and our measurement chain using proposed technique which is named 'Method of Evaluation Using Random Noise' in the semi-anechoic room of VŠB-TUO.

Finally, based on the comparison of the obtained results, with respect to their directivity and sensitivity, it is understood that the compared results are corresponding to each other. And based on these results, a well-designed crossover network is achieved. Consequently, it is understood that a successful speaker is obtained. In other words, we get a high-quality system from the point of view of SPL linearity in listenable frequency range 20Hz – 20 kHz.

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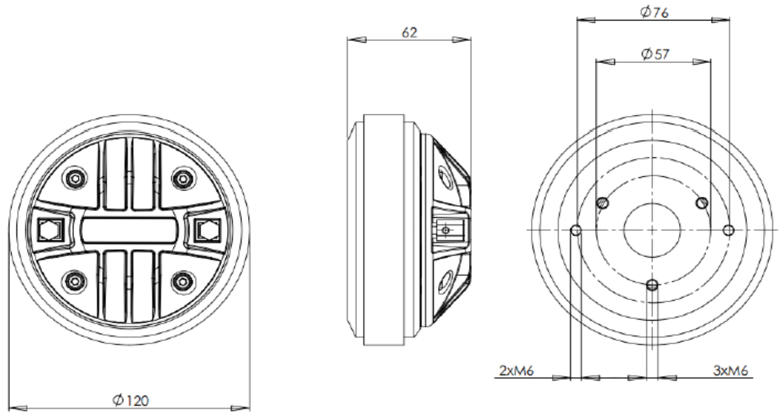
APPENDICES

Appendix A: Tweeter speaker catalog sheet B&C Speakers DE250 8/ohm series

DE250 8Ω

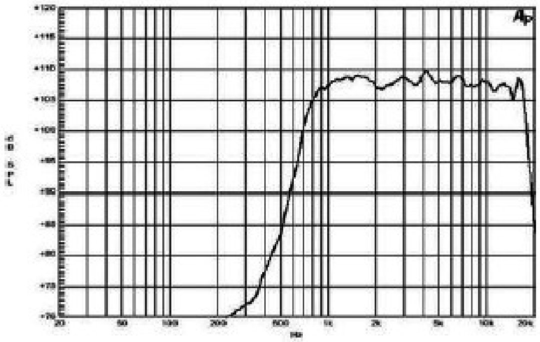
HF Drivers - 1.0 Inches

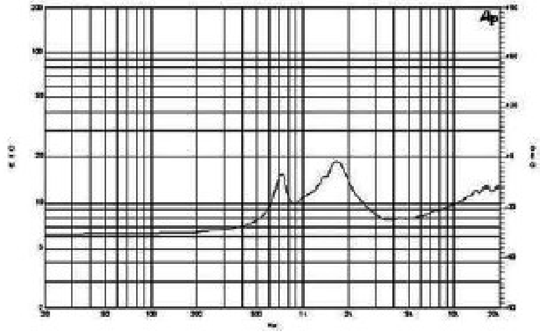


- 120 W continuous program power capacity
- 1' horn throat diameter
- 44 mm (1.7 in) aluminium voice coil
- Polyimide diaphragm
- 1000 - 18000 Hz response
- 108.5 dB sensitivity

DE250

HF Drivers- 1.0 Inches





SPECIFICATIONS[1]

Throat Diameter	25 mm (1.0 in)
Nominal Impedance	8 Ω
Minimum Impedance	7.8 Ω
Nominal Power Handling[2]	60 W
Continuous Power Handling[3]	120 W
Sensitivity[4]	108.5 dB
Frequency Range	1.0 - 18.0 kHz
Recommended Crossover[5]	1.6 kHz
Voice Coil Diameter	44 mm (1.7 in)
Winding Material	Aluminium
Inductance	0.11 mH
Diaphragm Material	Polyimide
Flux Density	1.85 T
Magnet Material	Ferrite

MOUNTING AND SHIPPING INFO

Two M6 holes 180° on 76 mm (3 in) diameter	
Three M6 holes 120° on 57 mm (2.2 in) diameter	
Overall Diameter	120 mm (4.7 in)
Depth	62 mm (2.4 in)
Net Weight	2.1 kg (4.6 lb)
Shipping Units	4
Shipping Weight	8.85 kg (19.51 lb)
Shipping Box	265x135x170 mm (10.43x5.31x6.69 in)

REPLACEMENT DIAPHRAGM

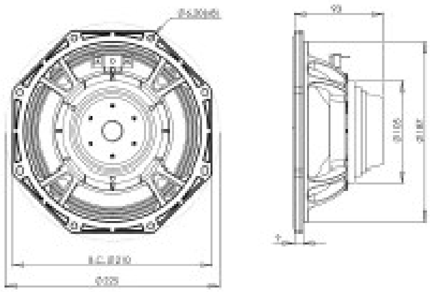
MMDDE2508

1. Driver mounted on B&C ME 45 horn.
2. 2 hour test made with continuous pink noise signal within the range from the recommended crossover frequency to 20 kHz. Power calculated on rated nominal impedance.
3. Power on Continuous Program is defined as 3 dB greater than the Nominal rating.
4. Applied RMS Voltage is set to 2.83 V for 8 ohms Nominal Impedance.
5. 12 dB/oct. or higher slope high-pass filter.

Appendix B: Mid-range speaker catalog sheet B&C Speakers 8MBX51-8 series

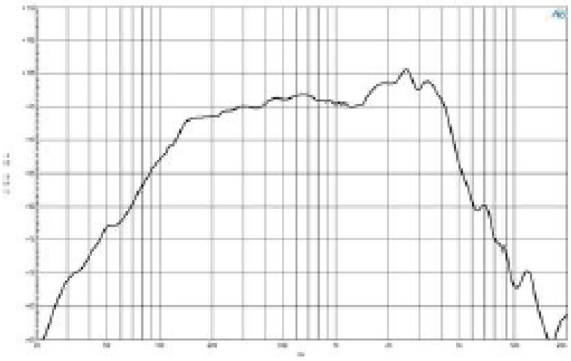
8MBX51 8Ω

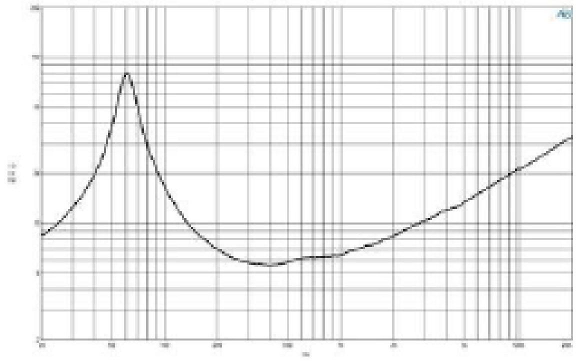
LF Drivers - 8.0 Inches



- Product Preview - Coming Soon....
- 400 W continuous program power capacity
- 50 mm (2 in) copper voice coil
- 60 - 4000 Hz response
- 96.5 dB sensitivity
- Neodymium ring magnet assembly
- Aluminium ring allows a very low distortion figure
- Ventilated voice coil gap for reduced power compression

The MBX series mid-bass woofers from B&C Speakers offer acoustic designers a new range of high efficiency, wide bandwidth alternatives that are not currently available in the B&C range. These full-featured transducers incorporate lightweight neodymium motors, inside/outside wound copper clad aluminum wire voice coils, weatherproof impregnated paper cones, and a symmetrical inductance profile. The MBX Series is especially well suited for two-way loudspeaker enclosures.





SPECIFICATIONS

Nominal Diameter	200 mm (8.0 in)
Nominal Impedance	8 Ω
Minimum Impedance	5.9 Ω
Nominal Power Handling[1]	200 W
Continuous Power Handling[2]	400 W
Sensitivity[3]	96.5 dB
Frequency Range	60 - 4000 Hz
Voice Coil Diameter	51 mm (2.0 in)
Winding Material	Aluminium
Former Material	Glass Fibre
Winding Depth	15.0 mm (0.59 in)
Magnetic Gap Depth	7.0 mm (0.28 in)
Flux Density	1.3 T

DESIGN

Surround Shape	Triple Roll
Cone Shape	Curvilinear
Magnet Material	Neodymium Ring
Spider	Single
Pole Design	T-Pole
Woofer Cone Treatment	Waterproof Impregnated Cone
Recommended Enclosure	19.0 dm^3 (0.67 ft^3)
Recommended Tuning	63 Hz

PARAMETERS[4]

Resonance Frequency	60 Hz
Re	4.9 Ω
Qes	0.31
Qms	5.6
Qts	0.29
Vas	23.0 dm^3 (0.81 ft^3)
Sd	220.0 cm^2 (34.1 in^2)
η_0	1.7 %
Xmax	6.0 mm
Xvar	8.0 mm
Mms	20.0 g
Bl	11.4 Txm
Le	0.4 mH
EBP	193 Hz

MOUNTING AND SHIPPING INFO

Overall Diameter	225 mm (8.86 in)
Bolt Circle Diameter	210 mm (8.27 in)
Baffle Cutout Diameter	187.0 mm (7.36 in)
Depth	93 mm (3.66 in)
Flange and Gasket Thickness	9 mm (0.35 in)
Air Volume Occupied by Driver	1.1 dm^3 (0.04 ft^3)
Net Weight	1.8 kg (3.97 lb)
Shipping Units	1
Shipping Weight	2.25 kg (4.96 lb)
Shipping Box	255x255x150 mm (10.04x10.04x5.91 in)

SERVICE KIT

RCK008MBX518

Appendix C: Bass speaker catalog sheet Eminence LAB12 series

EMINENCE®

PROFESSIONAL SERIES

LAB12

Recommended for vented, sealed, and horn loaded, professional audio enclosures as a subwoofer. Also great as an automotive sub.

- [ ] Midrange
- [ ] Midbass
- [ ] Woofer
- [x] Subwoofer
- [ ] Bass Guitar
- [x] Sealed Box
- [x] Vented Box
- [ ] Scoop Loading
- [x] Horn Loading

SPECIFICATION

Nominal Basket Diameter	12", 305 mm
Nominal Impedance*	6 Ω
Power Rating**	
Watts	400 W
Music Program	800 W
Resonance	22 Hz
Usable Frequency Range	25 Hz – 0.1 kHz
Sensitivity***	89.2 dB
Magnet Weight	160 oz.
Gap Height	0.375", 9.5 mm
Voice Coil Diameter	2.5", 64 mm

THIELE & SMALL PARAMETERS*

Fs	22 Hz
Re	4.29 Ω
Le	1.48 mH
Qms	13.32
Qes	0.39
Qts	0.38
Vas	4.42 cu.ft., 125.2 liters
Vd	659 cc
Cms	0.35 mm/N
BL	15 T-M
Mms	146 grams
EBP	56
Xmax	13 mm
Sd	506.7 cm2
Xlim	22 mm

MOUNTING INFORMATION

Recommended Enclosure Volume	
Sealed	22.7–28.3 liters, 0.8–1 cu.ft.
Vented	45.3–101.9 liters, 1.6–3.6 cu.ft.
Driver Volume Displaced	0.109 cu.ft., 3.09 liters
Overall Diameter	12.32", 312.9 mm
Baffle Hole Diameter	11.04", 280.4 mm
Front Sealing Gasket	Yes
Rear Sealing Gasket	N/A
Mounting Holes Diameter	0.26", 6.6 mm
Mounting Holes B.C.D.	11.77", 299 mm
Depth	6.44", 163.6 mm
Net Weight	22 lbs , 9.98 kg
Shipping Weight	23.8 lbs , 10.8 kg

MATERIALS OF CONSTRUCTION

- Copper voice coil
- Polyimide former
- Double stacked 80 oz. ferrite magnets
- Vented and extended core
- 12-spoke die-cast aluminum basket
- Kevlar-reinforced paper cone
- Foam cone edge
- Dual inverted dust caps

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FREQUENCY RESPONSE & IMPEDANCE CURVE*