Evaluation of piezoelectric speaker drivers for loudspeaker and array speaker design

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Abstract

This thesis is an evaluation of piezoelectric speaker drivers for loudspeaker design. It goes through the electrical and sound generating properties of the speaker drivers and general theory of piezoelectric speaker drivers. A prototype single speaker driver loudspeaker has been realized and evaluated by measuring the frequency response and sound pressure level (SPL), total harmonic distortion (THD), directivity and power consumption.

Further two types of array speakers (linear and square) has been realized and evaluated for the same properties as the single speaker driver loudspeaker. The different ways of connecting the speaker drivers together, parallel or series connection, has been compared in regards to SPL, power consumption and THD.

The speaker drivers generates approximately 60 dB SPL at one meter distance away from the speaker between 300 Hz to 20 kHz. Parallel connections proved to increase the overall measured SPL (over 70 dB SPL at one meter for four speaker drivers), however the amplifier was placed under more stress due to it needing to supply more power. Series connections proved to be less problematic for the amplifier, however there was no increase in SPL compared to a single speaker driver. The THD measurements showed that series connections resulted in less THD compared to parallel connection and single speaker drivers. The directivity measurements showed that the square array became more directional in both the vertical and horizontal planes compared to the individual speaker driver. The line array became even more directional compare to the square array, but only for one plane and the other plane showed a similar directivity pattern as the individual speaker driver.

They perform similarly to electromagnetic speaker drivers with comparable dimensions in area of speaker diaphragm and could be used in applications where thickness/size is limited.

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Acronyms and concepts

SPL: Sound pressure level
THD: Total Harmonic Distortion
AC: Alternating Current
PZT: lead zirconate titanat
dB: decibel
eq: equation

Speaker driver: In electronics a driver often refers to a type of circuit or component that controls another component or circuit, but for speakers this is not the same. A speaker driver refers to the actual motor and structure responsible for generating sound.

Omnidirectional: The same for all directions

Contents

1 Introduction

The general trend over the years has been to make electronic products smaller and light weight in an effort to make them more portable. Sound systems have been able to be decreased but the traditional electromagnetic speakers require a rather large foot print that limits their ability to reduce in size. There are alternatives that are starting to replace them, one of the alternatives are piezoelectric speakers. They show great potential, not only in the reduced size but also promises lower power consumption and scalability possibilities. Piezoelectric speakers are starting to appear in in-ear applications such as headphones and hearing aids, and historically been found useful as tweeters and buzzers. But it has only been in the last decade that the industry has starting to present full-range piezoelectric speaker options and mostly for in-ear applications.

They could prove to be good alternatives for applications where size is limited, other properties such as lower power consumption and the low weight could also prove useful. There are few piezoelectric loudspeakers available on the market as of today but the interest of piezoelectric speakers seems to be increasing, most for in-ear applications.

During this project a prototype piezoelectric speaker will be realized that will be evaluate in order to map out any obstacles that may arise when designing a speaker utilizing these types of speaker drivers. One part if the evaluation will be to measure some metrics of the speaker drivers. From discussions with the engineers at Axis there are a few measurements that are of interest when measuring speaker drivers of which the sound pressure level, total harmonic distortion and directivity will be measured to give a good idea of their performance.

What are the different properties of piezoelectric speakers and what has to be taken into account when designing with these types of speaker drivers. What are their shortcomings and strengths, what can be done to solve these shortcomings or improve their performance?

The small size allows more speakers to be placed together and realizing an array configuration, will there be any increases in overall performance or any detrimental effects from placing speakers in an array configuration? Is there a more efficient way of connecting the speakers, evaluating parallel and series connections between the drivers and comparing linear versus square array, what are the gains and losses between the two?

This thesis aims to increase knowledge of piezoelectric speaker in general on Axis and simultaneously measure and evaluate available speaker drivers to help them decide if they are viable options for future products.

2 Background

2.1 Electronics

To make this document easier to follow this sections aims to refresh or introduce the reader to some fundamental electronic concepts.

2.1.1 Impedance

Resistance relates to current and voltage as described by Ohms law, see eq. 1.

$$U = IR \tag{1}$$

Where U is the voltage in Volts, I is the current in Ampere and R is the resistance in Ohm. Reactive circuit elements such as capacitors (C) and inductors (L) are frequency dependant and work differently to resistors. Ideally they are reactive, this the imaginary counterpart of resistance. A real component will be plgued with parasitic resistive and reactive properties and the total impedance of a component is described in eq. 2.

$$\overline{Z} = R + jX = |Z|e^{arg(Z)} \tag{2}$$

Where Z is the impedance, R is the real part knows as the resistance, X is the reactance and j is the imaginary unit. The impedance of a pure resistor is given in 3, capacitor in eq. 4 and of an inductor in eq. 5.

$$Z_R = R \tag{3}$$

$$\overline{Z_C} = \frac{1}{j\omega C} \tag{4}$$

$$\overline{Z_L} = j\omega L \tag{5}$$

ω is the angular frequency, L is the inductance in Henry and C is the capacitance in Farad. This shows that the impedance of capacitors and inductors is changing with frequency, see figure 1.
When either a capacitive or inductive impedance is applied to Ohms law the equation needs to be rewritten, see eq. 6 and 7.

$$I = C\frac{dV}{dt} \tag{6}$$

$$V = L\frac{dI}{dt} \tag{7}$$

This shows that for capacitances the current is proportional to the real-time change of the voltage and for inductors the voltage is proportional to the real-time change in current. Both capacitances and inductors thereby cause a phase shift between the applied AC current and the resulting AC voltage. In an ideal capacitor this phase shift is -90 degrees and for a real one its is in between 0 to -90 degrees. Meanwhile an ideal inductor has a phase shift of 90 degrees and a real inductor has a phase shift of anywhere between 0 to 90 degrees phase shift.

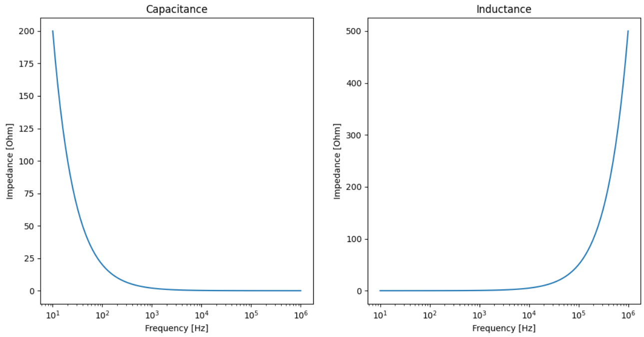


Figure 1: Impedance of capacitor and inductor over frequency.

2.1.2 Parallel and series connections

When electrical components are placed in series the impedance is increased as the path the current travels through becomes more resistive compared to in parallel where the current gets more paths to travel through, effectively lowering the impedance. The resulting impedance of three components with impedance $\overline{Z_1}$, $\overline{Z_2}$ and $\overline{Z_3}$ when put in series can be calculated with eq. 8 and when put in parallel in eq. 9.

$$\overline{Z_T} = \overline{Z_1} + \overline{Z_2} + \overline{Z_3} \tag{8}$$

$$\frac{1}{\overline{Z_T}} = \frac{1}{\overline{Z_1}} + \frac{1}{\overline{Z_2}} + \frac{1}{\overline{Z_3}} \tag{9}$$

Following equation 8 and 9 with equations 3, 4 and 5 the total resistance, inductance and capacitance of a circuit of multiple components of the same type can be calculated. The resistive and inductive load, refers to the total resistance or inductance of a circuit, will be decrease when put in parallel and increased when put in series. The capacitive load, refers to the total capacitance of a circuit, on the other hand will increase for when put in parallel and decreased when put in series.

2.1.3 Power [W]

Power, expressed in Watts is the energy consumed by electrical components, power is either described as apparent power or actual power (also called true power). The apparent power is calculated with eq. 10 and the actual power is calculated with eq. 11. In fully resistive elements the apparent power is equal to the actual power since a fully resistive component do not apply a phase shift between the voltage and current. Due to the phase shift that occurs with reactive elements the equation is no longer true and needs some altercation.

$$P_{Apparent} = VI \tag{10}$$

$$P_{actual} = VIcos(\varphi) \tag{11}$$

From equation 11 the theoretical actual power of an ideal capacitor and an ideal inductor is zero. In reality these are often plagued with parasitic elements that causes the phase shift to be less than 90 degrees and thus the power is more than zero.

2.1.4 Voltage division

If a voltage is applied over two loads that are placed in series it is possible to calculate the voltage between the loads. In the circuit drawn in figure 2 there are two loads connected to the same voltage. The voltage in node V_{out} can be calculated with equation 12.

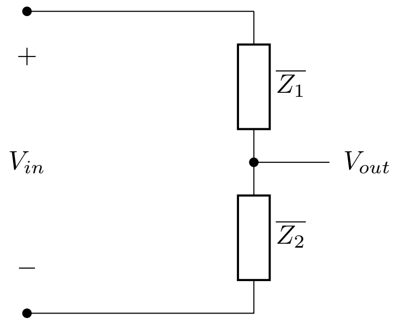


Figure 2: Two loads connected in series with the node between them marked as V_{out}

$$V_{out} = \frac{\overline{Z_2}}{\overline{Z_1} + \overline{Z_2}} V_{in} \tag{12}$$

2.1.5 Low-pass filters

It is possible to filter out some frequencies when handeling AC signals with the use of low-, band- or high- pass filters. But for this report we will only go over low-pass filters.

One way to realize a low-pass filter is shown in figure 3, here a capacitor is connected in series to a resistor. By connecting something to the node between the resistor and the capacitor only lower frequency signals would be seen by what ever component was connected there. As the frequency increases the impedance of the capacitor would decrease and allowing high frequency signals a low impedance path to ground, low frequency signals would see the capacitor as a high impedance path. This allows the system to filter out the high frequencies by letting them pass through the capacitor. The signal strength at the node between the resistor and capacitor will gradually decrease, but when the signal is -3 dB (half) of that of the original value the frequency of the signal is equal to the cut-off frequency of the filter.The cut-off frequency can be calulated with eq. 13 where R is the resistance and C is the capacitance.

$$f_{-3dB} = \frac{1}{2\pi RC} \tag{13}$$

The voltage in the node between the resistor and capacitor can be calculated with eq. 12. For lower frequency signals the capacitor will display a higher impedance and the voltage in this node would be high, as the frequency increases and the impedance of the capacitor decreases the voltage in this node would decrease.

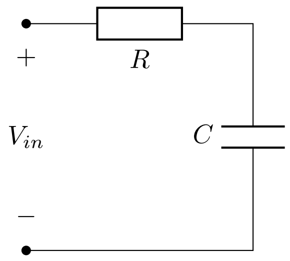


Figure 3: Low-pass filter realized with a resistor R and a capacitance C.

2.2 Sound and sound waves

According to [1], "sound, a mechanical disturbance from a state of equilibrium that propagates through an elastic material medium". To better understand sound the properties of waves needs to be explored, in particular sound waves. Sound waves are longitudinal waves, which means that the mechanical vibrations occurs along the axis of propagation, these waves are also called compression waves.

Most often when talking about sound it is sound in air, as this is the most common way humans perceive sound. It can be helpful to think of air as multiple layers stacked against each other. As a sound wave propagates through the air the wave will "push" these layers together or "pull" the layers apart creating regions of high and low air pressure. These high and low density regions will move through the air in the direction of the wave. The air layers will therefore move back and forth but not move along with the wave. After the sound waves have passed these layers will then return to their original place. This means that the air particles are only moving in place and the energy is traveling trough the air.

Since sound can be described as waves they can be expressed mathematically as seen in eq. 14 as a sine function.

$$y = Asin(2\pi(\frac{x}{\lambda} - \frac{t}{T})) \tag{14}$$

This is the equation describes the wave in one direction. The x-axis, over time t and x denotes the distance along the axis of propagation. A is the amplitude of the wave, λ is the wavelength, T it the period of the wave and y is the amplitude of that point in space and time. If the equations describes an AC signal propagating through a wire it could tell what voltage or current is at 1 meter into the wire after 6 seconds.

There are several parameters that are used to describe different properties of the wave. Since waves have a cyclical behavior the motion of a sound wave will be repeated. The period, T, is the time it takes a wave to repeat its motion. This property is used to define another property of waves, frequency f. Frequency is defined as how many reputations occur per second, see eq. 15.

$$f \equiv \frac{1}{T} \tag{15}$$

How long a wave is in space is called the wavelength and is a measurement of how far the wave travels during one period. In eq. 16.

$$\lambda = \frac{c}{f} \tag{16}$$

c is the speed of the wave in the medium it travels. The speed of sound varies depending on the medium it travels through and temperature. In air at $15°C$ the speed of sound is approximately $340m/s$ whilst at $0°C$ the speed of sound measures at approximately $330m/s$ [2]. The speed of sound can be calculated with the use of the following equation, 17 according to [3]. Where $gamma$ is the ratio of specific heats (=1.4 for air at STP), T is the temperature in kelvin and R is the ideal gas constant (286 $m^2/s^2/°K$ for air).

$$c = \sqrt{\gamma R T} \tag{17}$$

2.2.1 Creation of sound

Sound is created by mechanical vibrations being introduced in a elastic material. The most common type of speakers are electromagnetic speakers, more about them in section 2.6. These will create sound due to their piston-like movement, back and forth, which creates a compression wave in the surrounding air. There are other ways to create sound but the fundamental concept is that a mechanical movement causes high and low air pressure regions that propagate through the air.

2.2.2 Interference of multiple waves

When waves meet they will interact with each other. The waves can pass through each other an will not be affected after their interaction. However they will during their interaction with each other combine and create a wave. The resulting wave will be the sum of the waves amplitudes in each point in space. This can lead to the amplitude being increased (constructive interference) or decreased (destructive interference) depending on the frequency, amplitude and phase difference between the two waves in each point [4].

See figure 4 for a illustrative description of interference. In the figure, the top most graph shows three waves with the same amplitude. The first wave (blue) has no phase shift and frequency x Hz, the second wave (red) has double the frequency (2x Hz) and a $\pi/2$ phase shift and the third wave (black) has the same frequency as the first wave but a π phase shift. The bottom graph shows the interference between three identical waves (blue, total constructive interference), two waves with different frequency and phase and the last shows the interference that occurs for two signals that are identical except they are completely out of phase with each other (total destructive interference).

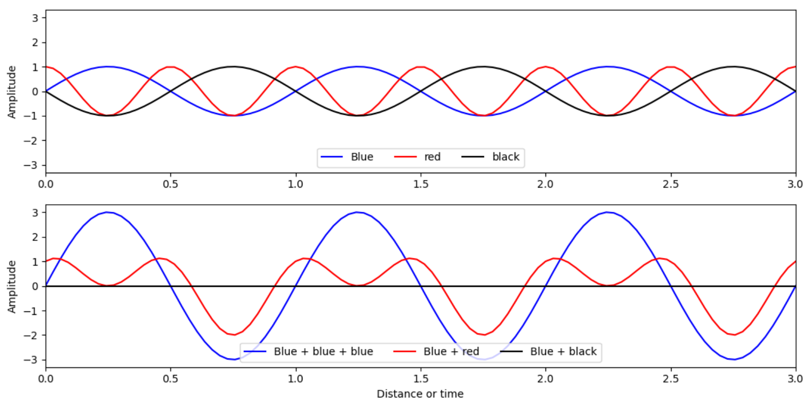


Figure 4: In the top graph three waves are presented, in the bottom graph are the interference between three combinations of the waves in the top graph.

2.3 Properties of sound waves

In audio electronics there are a nuumber of parameters that is being measured and a concepts that need to be introduced. The section aims to enrich the reader in these concepts and give a beter understanding for audio electronics.

2.3.1 Acoustical impedance

Acoustical impedance is a measurement of the resistance to motion at a given point within a material. There is no single definition for acoustical impedance but the most common one is the specific acoustic impedance which is defined as the ratio of the sound pressure to the associated particle velocity, see eq. 18.

$$z \equiv \frac{p}{u} = c\rho \tag{18}$$

Where z is the specific acoustic impedance, p is the sound pressure and u is the particle velocity, ρ is the mass density and c is the velocity of sound. With this equation it is possible to calculate the characteristic impedance, in air at room temperature it is approximately $412mks$ rayls [5].

2.3.2 Sound intensity, I

Sound intensity is the measurement of the amount of energy that propagate through a given area during a given time and can be calculated for a planar wave with eq. 19. The normal of the area that is measured is parallel to the direction of propagation.

$$I(\theta) = \frac{Ecos(\theta)}{TS} = \frac{Wcos(\theta)}{S} \tag{19}$$

Where E is the energy of a sound wave (Nm/s), W is sound power, $I(\theta)$ is the intensity passing through the area in the direction of angle θ. S is the area that is being measured, T is the period of the wave and θ is the angle between the normal to the surface and the direction of propagation. We get the maximum intensity when the angle $\theta = 0$, see eq. 20 [5].

$$I = \frac{W}{S} \Rightarrow \frac{I}{I_0} = \frac{\frac{W}{W_0}}{\frac{S}{S_0}} \tag{20}$$

Where W_0 is the reference power ($10^{-12}W$), S is the measurement area, S_0 is $1m^2$. The reference power relates to the lowest pressure variations that humans are able to hear. Sound power is the sound energy, the mechanical work of the wave, that the source emits over a period. The sound energy is given by eq. 21 and the sound power is given by eq. 22

$$E = pSdx \tag{21}$$

p is the rms (root-mean-square) acoustic pressure and with this we can express the sound power as the energy over time, see eq. 22

$$W = \frac{pSdx}{dt} = pSu \tag{22}$$

In eq. 22 u denotes the particle velocity of the medium for which the wave is traveling through.By using eq. 22 we can rewrite eq. 20 and what follows is that the intensity is correlated to the pressure.

$$I = pu \tag{23}$$

The intensity is directly measurable using a sound level meter, it is also proportional to the power which allows multiple sounds to be calculated by adding the separate intensities together.

2.3.3 Decibel notation

The lower limit of Intensity that humans can hear is $10^{-12}W/m^2$ and the sound intensity produced by a jet aircraft could be up towards $1W/m^2$. With more than 12 orders of magnitudes difference, decibel notation is used for simplification, see eq. 24 and 25.

$$L_I = 10log_{10}\frac{I}{I_0} \tag{24}$$

$$L_W = 10log_{10}\frac{W}{W_0} \tag{25}$$

Eq. 20 can then be rewritten as shown in eq. 26

$$L_I = L_W - 10log_{10}\frac{S}{S_0} \tag{26}$$

2.3.4 Sound pressure Level, SPL [dB]

When measuring sound it is often the pressure variations that are measured. The scale is in decibel and the air pressure of the sound wave is compared to the sound pressure level, which is $20\mu Pa$ which relates to the lower limit of intensity mentioned in section 2.3.3. This pressure is used as a reference since it is close to the lowest air pressure that humans can detect for a sound wave with a frequency of $1kHz$ [6]. In table 1 below are some examples of common sound in everyday life and their SPL.
It is possible to calculate the SPL at one point with eq. 27 using the measured pressure at that point and the sound pressure level.

$$L_p = 10log_{10}\frac{p^2}{p_{ref}^2} = 20log_{10}\frac{p}{p_{ref}} \tag{27}$$

L_p is the SPL in dB, p_{ref} is the reference pressure also refereed to as the sound pressure level and p is the root-mean square of the measured pressure level. As we showed in eq. 23 it became apparent that the intensity of the wave is correlated to the pressure. It is then possible to calculate the sound pressure level with the use of the sound intensity.

$$L_p = L_I + 10log_{10}(\frac{\rho_0 c_0}{400}) \tag{28}$$

The correction that is being added is due to the specific acoustic impedance. In air this only adds ≈ 0.13 dB and is therefore often left out of calculations.

Table 1: Common sounds and their respective volume presented in dBSPL [7] [8].

Type of sound	Avrage SPL [dB]
Normal breathing	10
Ticking watch	20
Soft whisper	30
Refrigerator hum	40
Normal conversation	60
Vacuum cleaner	70
Traffic	80
Chainsaw	90
Clubs/concerts	110
Human pain threshold	120
Eardrums break	180

2.3.5 Sound power level, [W]

Sound power level is given in the unit Watts and is used when talking about the strength of a sound source. To calculate the sound power level eq. 20 is used.

We can rewrite eq. 26 with the use of eq. 28 to get eq. 29.

$$L_I = L_W + 10log_{10}S \Leftrightarrow L_p = L_W + 10log_{10}S + 10log_{10}\frac{\rho_0 c_0}{400} \tag{29}$$

Note that L_W is the sound power level in dB.

2.3.6 Sound attenuation over distance

As sound travels away from its source the SPL decrease. If the sound source is considered a point source and the sound travels in free-field, without reflections or any other influences, in all directions i.e. spherically. The SPL can be calculated using a slightly modified eq. 29. As the surface will be the mantel area of a sphere we can write it as shown in eq. 30.

$$L_p = L_W + 10log_{10}\frac{1}{4\pi r^2} + 10log_{10}\frac{\rho_0 c_0}{400} \tag{30}$$

Where r is the distance away from the source. Using this equation it is possible to calculate the change in SPL between two point, see eq.31. This equations is derived by using eq. 30.

$$\Delta L_p = 20log_{10}\frac{r_2}{r_1} \tag{31}$$

r_1 and r_2 are the distances in meters to the source from points 1 and 2. An altercation to this formula is shown in eq.32 which calculates the SPL at another distance provided the SPL at one point.

$$L_{p2} = L_{p1} + 20log_{10}\frac{r_2}{r_1} \tag{32}$$

This formula only works for far field and should not be used for near field distances [5]. In the near field the resulting wave is typically a complex pattern due to interference caused by the size and aperture of the sound source. The length of what is considered near field can be calculated with eq.33 for a sound source with circular aperture. Interference is related to wavelength and frequency and therefore the distance that governs what is near and far field distance changes with frequency [9].

$$NearfieldLength = \frac{D^2}{4\lambda} = \frac{D^2 f}{4c} \tag{33}$$

Where D is the diameter of the circular aperture and c is the speed of the sound in the material. For frequencies between $20Hz$ to $20kZ$ the near field distances in air for a speaker with 10 cm diameter aperture would be between $0.15mm$ and $15cm$.

2.3.7 Efficiency of speaker drivers

Efficiency or sensitivity of a speaker is a measurement of how loud a speaker is. The most common measurement configuration for speaker efficiency is to measure the SPL at one meters distance when driving the speaker at one Watt of power. This measurement is intended for electromagnetic speakers as the measurement does not translate well for piezoelectric speaker drivers since piezoelectric speakers rarely require one watt of power. But generally piezoelectric speakers are measured for 10 cm and for a specified peak-peak voltage, but this may also be because they are intended for in-ear applications and not loudspeaker applications.

Electromagnetic speakers have roughly the same impedance over the frequency range whilst piezoelectric speakers do not. Electromagnetic speakers have either 4 or 8Ω impedance and for the 8Ω speakers $1V_{rms}$ signal is $\approx 1W$. Meanwhile $1V_{rms}$ on a piezoelectric speaker would result in a lower wattage. This is because the impedance of piezoelectric speaker drivers varying over frequency since they are mainly capacitive. Depending on the speaker driver and the piezoelectric material used the capacitive load differ. For a speaker of $500nF$ at lower frequencies the wattage would be around a few milliwatts and for higher frequencies it would still be below half a watt.

2.3.8 Total harmonic distortion, THD

Total harmonic distortion is a measurement of how non-linear a system is. First lets explain distortion, simply put distortion is when the signal has been altered in some way. If a clean sine wave is feed to the speaker driver the speaker will create sound with the same frequency as the signal and an amplitude that correlates with the amplitude of the signal. If the signal is subjected to clipping somewhere in the system the signal will lose the appearance of a sine wave and become more square. Distortion can also be due to harmonics. Harmonics are overtones of the signal, if a $1kHz$ signal is being played and it occurs a $2kHz$ signal as well as the $1kHz$ signal the $2kHz$ signal is the second degree harmonic of the signal. Distortion caused by harmonics is called harmonic distortion. The total harmonic distortion is a measurement of all harmonics of a system [10].

To calculate the THD each amplitude of the harmonic frequencies are squared and then summed, then the sum of the harmonic frequencies are presented as a fraction of the square to the input signals amplitude. Eq. 34 is used to calculate the THD.

$$THD = \frac{\sqrt{V_2^2 + V_3^2 + V_4^2 + V_5^2 + ...}}{V_F} \tag{34}$$

Where V_F is the fundamental frequency, e.i. the frequency that is being measured. V_n is the n:th harmonic of the fundamental frequency. As the THD is often described as a percentage the value from eq. 34 needs to be multiplied by 100% [11].

2.3.9 Directionality and directivity

It is common that the sound pressure level for a given distance to the source will depend on the direction of the point to the source. This property is called directionality and when expressed in changes in sound pressure levels it is refereed to as directivity. Directivity is often shown in polar plots, one horizontal and one vertical plot.

Directivity is given as the intensity in one direction compared to the average intensity of an omnidirectional source and is denoted as Q, see eq. 35

$$Q(\theta, \phi) = \frac{I(\theta, \phi)}{I_{average}} \tag{35}$$

Where the average intensity can be calculated with the use of eq. 20 and using the mantle area of a sphere as the surface S, as shown below:

$$I_{average} = \frac{W}{4r\pi r^2} \tag{36}$$

Two speakers could produce the same sound pressure levels but their sound fields could be very different. It can be very useful to know how directional a speaker is. Directivity index is defined as the difference in decibels between the sound pressure level from a real source that is measured for given directions and then compared to the average sound pressure level from an omnidirectional source. See eq. 37 for a mathematical definition.

$$D(\theta, \phi) \equiv L_p(\theta, \phi) - \overline{L_p} \tag{37}$$

$D(\theta, \phi)$ is the directivity index (gain) for a given direction in dB, $L_p(\theta, \phi)$ is the SPL for a given direction in dB, $\bar{L_p}$ is the average SPL for all angles in dB and (θ, ϕ) denotes the specified angle.

The directivity index can be expressed with the directivity Q as shown in eq. 38 below:

$$D(\theta, \phi) = 10log_{10}Q(\theta, \phi) \tag{38}$$

Within the audio industry the Q of a speaker most often refers to the on-axis directivity, i.e. $Q(0, 0)$ or Q_0 [5]. However a more intuitive way of presenting how directional a speaker driver is, is to measure the sound output for multiple angles an present them in a polar plot. This will show for what angles the speaker drivers have the highest outputs and if there are any angles where it has lower output.

2.3.10 Increasing the power feed to the speaker driver

The SPL that is produced by a speaker driver is related to the amount of power that is provided. Speakers drivers are essentially a transformer that converts electrical energy to mechanical work that causes these pressure variations in the air that humans interpret as sound. Some power will be lost in passive and parasitic components but most of it will be transferred into *sound*. By increasing the power the SPL will not increase at the same rate since SPL is measured in decibels and power linear. If the power is doubled for a speaker driver the increase in SPL will only be $+3dB$ [12].

2.3.11 Multiple sources and mutual coupling

If two sound sources are placed in close vicinity of each other the total sound power output will be a combination of the two sources power output. If the signal to the sources are uncorrelated the total output is equal to the sum of the two sources power output. If the signals are correlated a phenomena called mutual coupling occurs, it describes the interaction between multiple sources that radiate the same sound at the same time.

The two sources sound waves will interfere with each other for every point in space. The length between each point and a sound source is called path length and when the path length differ by a multiple of a wavelength there will occur maximum constructive interference. If the path length differ by a multiple of half a wave length there will occur total destructive interference and the sound power in these point will be zero. Between the speakers a plane (perpendicular to the speaker faces) that consists of points that have equal path length to both speakers, this means that for speakers playing the same signal in phase along this plane there will always be total constructive interference [13].

From equation 27, if the pressure is increased by a factor of two the SPL should increase by 6 dBSPL. This seems contradictory, imagine two speakers are playing at one Watt the increase in SPL should only be 3 dB since the total power used is two Watt. This becomes even more confusing when looking at the radiated power in all directions. If the distance between the speakers is smaller than half a wavelength there will not occur any points where the path length is different by any multiple of half a wavelength, i.e. no total destructive interference. This will result in the radiated sound power being increased in all directions and from this the total radiated power to be increased by a factor of four. Essentially 1 + 1 = 4, and two speakers that are placed close to each other effectively doubles the power of both speakers. As the distance between the speaker drivers increase and/or the wavelength decreases there will occur angles for which the radiated sound power is decreased due to destructive interference and the total radiated sound power will be reduced to approximately twice that of a single speaker driver, i.e. 1 + 1 = 2. The increase in radiated sound power in respect to wavelength and distance between speaker drivers can be calculated using formula 39 [13].

$$\frac{W_2}{W_1} = 1 + \frac{sin(kR)}{kR} \tag{39}$$

Where W_2 is power radiated from one speaker driver when in the nearby vicinity of an another speaker, W_1 is the radiated power from one lone speaker driver, $k = 2\pi/\lambda$ is the wave number and R is the distance that separate the two speaker drivers.

2.4 Human sound perception

Measuring sound will present the objectively measured sound pressure level and distortion, but what are humans able to hear and how do we interpret this is not the same as the data presents itself. When doing measurements in a laboratory the resolution relies on the hardware that is used to measure, for example a good microphone will give more accurate results. The human ear has its own limitations and the brain will decode the information collected by the ear. Human hearing is subjective and for ever person it will be different but in the following sub-sections some general information will be presented of how humans perceive sound.

Since the range of hearing is based on the lowest detectable pressure variations at 1 kHz the range for human hearing starts at 0 dB SPL. There is no upper limit for what we humans can detect, eventually higher SPL will become painfull and damage our ears. The pain threshold is at 120-130 dB, sounds of 110 dB will cause discomfort, however chronic hearing damage can be caused by hearing sound at levels of 90 dB over longer periods of time. These are more general limits, the limits for discomfort and pain are of course subjective.

The subjective increase in loudness is not linear with the increase in dBSPL. An increase of 10 dB is commonly accepted as perceived double the loudness whilst double the power is equal to an increase of 3 dB. Meanwhile, 3dB is just enough difference for humans to detect any subjective difference in loudness [14].

Frequency is an important parameter when talking about sound, as humans perceives different frequencies as different tones. The most general frequency span for what humans can hear is the frequencies between $20Hz$ to $20kHz$. However, the upper frequency limit decreases with age and the frequency span of an adult is around $15 - 17kHz$ and could be even lower if they have damaged their ears [15].

2.4.1 Phon

Phon is a unit of loudness level, humans do not perceive all frequencies the same and this unit is a subjective measure of loudness [16]. A 1 kHz tone will appear louder than a 200 Hz tone when both are playing at 60 dBSPL.

The unit is referenced to 1 kHz tone of a said dBSPL, 60 dBSPL at 1 kHz is defined as 60 phons. The measurement is done by playing a 1 kHz tone at a fixed SPL, for example 60 dBSPL. The listner will then listen to a 2 kHz tone, the SPL will be changed until the listener deems that the two tones are equally loud. Then whatever dBSPL the 2 kHz tone ends up on will be equal to 60 phons for a 2 kHz tone.

In figure 5 different phons are displayed as a function of actual dBSPL over frequency. From the graph it is possible to conclude that the human hearing is most sensitive between 1 to 6 kHz and peaks around 3.5 kHz.

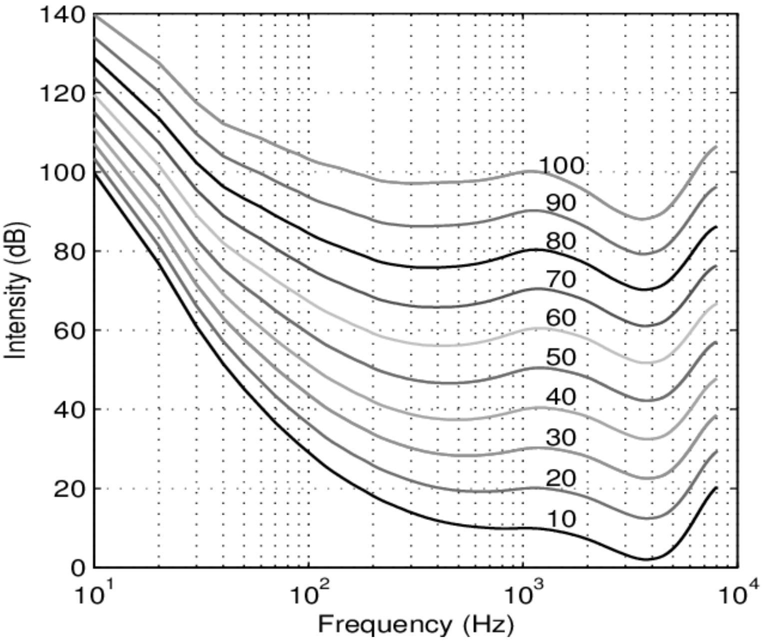


Figure 5: Here 10 to 100 phons are shown to their true dBSPL [17].

2.4.2 Psychoacoustic bass extension

The human perception is easily ”tricked” and this can be exploited to achieve perceive low frequency tones without them being played.

The *virtual pitch theory* states that human auditory system creates a pitch f_0 if presented with harmonic series that has a common fundametnal frequency. This means that if a speaker is unable or has to low output for the frequency 60 Hz, by playing 120 Hz, 180 Hz, 240 Hz... the human brain recreates the fundamental frequency from the harmonic series and makes us believe that the 60 Hz tone is played.

This algorithm has its draw-backs as it work well for speakers whose cut of frequency is at approximately 200 Hz. However, when the cut-off frequency is increased some of the harmonic frequencies are below the cut of frequency of the speaker and it can not create these harmonic series. This results in that the virtual pitch theory can not fully compensate for low-frequency response for speakers with higher cut off frequency [18].

2.5 Piezoelectricity

In 1880 the Curie brothers discovered the piezoelectric effect. They discovered that some crystals will generate electrical voltage on its surfaces if exposed to a mechanical deformation. The reverse is also true, when an electrical signal that is applied to the crystals it mechanically deform. Materials that are piezoelectric are refereed to as piezoelectric materials/crystals or piezoceramics [19]. In figure 6 a quartz unit cell can be seen under two external forces and what happens to the charges in the unit cell.

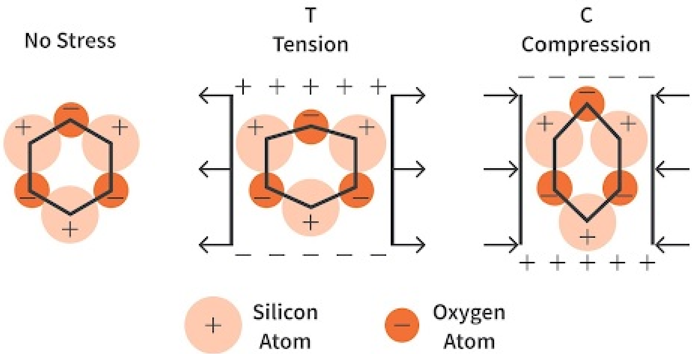


Figure 6: This image shows the piezoelectric effect in quartz, when there is no external force applied on the crystal the charges within the materal is balanced. When an external force is applied the atoms inside the crystal are moved and the charges become unbalanced and polarizes the crystal [20].

2.5.1 Piezoelectric effect

The piezoelectric effect is divided into two separate effects, the direct piezoelectric effect is when the crystal generate an electrical charge when exerted to a mechanical force. The converse piezoelectric effect is when an applied electrical signal causes a mechanical deformation. The effects can be explained with eq. 40 direct effect and 41 converse effect. Piezoelectric speakers are exploiting the converse piezoelectric effect as it causes movement from electrical signals which then can be used to create sound waves.

$$D = dT + \epsilon E \tag{40}$$

$$X = sT + dE \tag{41}$$

D is the electrical displacement, d the piezoelectric coefficient, T the stress, ϵ is the permittivity of the material, E is the electric field, X is the strain and s is the mechanical compliance [21]. Strain describes the deformation of a crystal under stress [22] and the definition of strain is shown in eq. 42 [23].

$$X = \frac{\Delta L}{L_0} \tag{42}$$

Where ΔL is the change in length caused by some force/stress and L_0 is the original length of the material. By combining equation 42 and 41 we can write it as 43 below.

$$\frac{\Delta L}{L_0} = sT + dE \tag{43}$$

This equation can then be used to calculate how much the crystal will grow or shrink in one dimension. However, depending on the type of crystal and direction of the crystal lattice and the electrical field the mechanical deformation that is caused by the piezoelectric effect is not always the same as displayed in figure 7. For some of the piezoceramics the deformation of the crystal appears along one direction and elongates the crystal, for some it may be a two dimensional elongation. The deformation can also manifest itself as a shearing, flexing or twisting of the crystal.

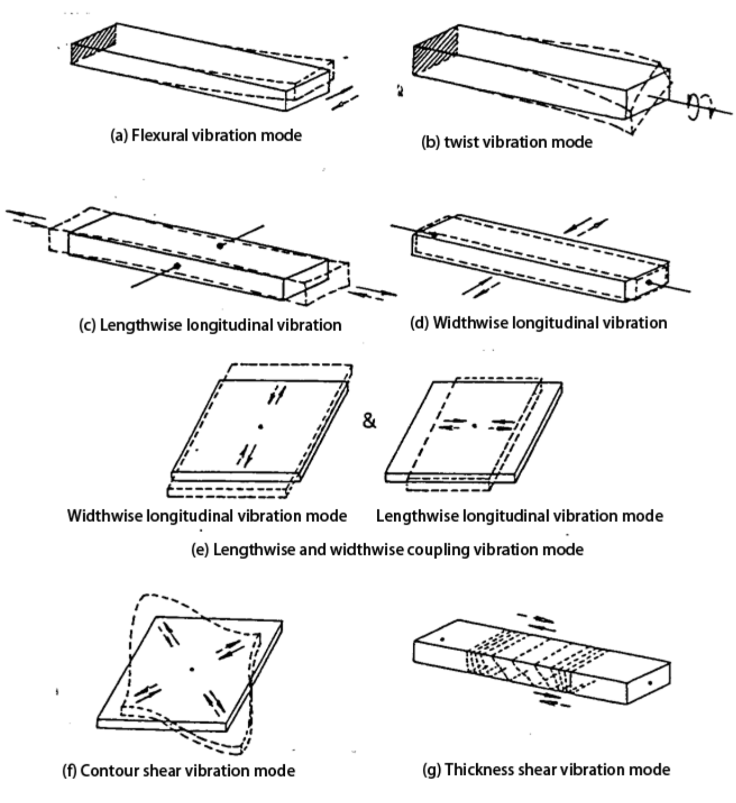


Figure 7: Different types of deformation that may arise in piezoelectric materials [24].

The piezoelectric coefficients differs depending on what directions the forces and electric fields are applied and are separated into two sub coefficients denoted as d_{31} and d_{33}. d_{33} applies when the force is applied to the same face as the charge collects at and d_{31} applies when the charges collect at the same surface as for d_{33} but the force is applied at an angle perpendicular to the polarization [25].

2.5.2 Piezoelectric materials

Piezoelectricity is not found in all crystal as there are some specific requirements on the crystal lattice [26]. The piezoelectric effect is described as the asymmetric shift of charges, this property can not be found in every crystal. This essentially means that the crystal lacks a center of symmetry in regards of the charges within the crystal lattice. They material also needs to be dielectric, poor conductors.

A common example of a piezoelectric crystal is quartz and it is used in many applications for its piezoelectric properties. To name a few, it can be used to keep the rate of a clock or in lighters the spark is being created with quartz crystals. However the most commonly used type of piezoelectric material is lead zirconate titanate (PZT, Pb[Zr(x)Ti(1-x)]O3) with greater sensitivity and higher operating temperature than its predecessor [21].

2.6 Speaker drivers

As discussed in section 2.2.1 sound can be created by movement of an object. In speakers a membrane is moved back and forth to create sound waves. The motion is caused by an electrical signal that is being feed to the speaker driver. This means that speakers are a form of two-step transformer, first converting an electrical signal to motion and then motion into pressure waves.

There are a few different ways to realise speaker drivers, the most common speaker drivers are electromagnetic. These can be used in many different applications and can be found in almost all sound producing electronics. A less common speaker driver is the piezoelectric speaker driver, which you probably have listened to many times without knowing it. These are mostly used as ”buzzers” to produce one continuous tone or as ”tweeters” used to enhance the higher range of frequencies of a loudspeaker system.

2.6.1 Electromagnetic speakers

Electromagnetic or magnetic speaker driver are both names for one type of speaker driver. It generates sound by exploiting the force caused by the interaction of a magnetic field from a magnet and the magnetic field generated by an electric current in the voice coil. In figure 8 the cross section of a speaker driver is shown and the components are named and pointed out [27].

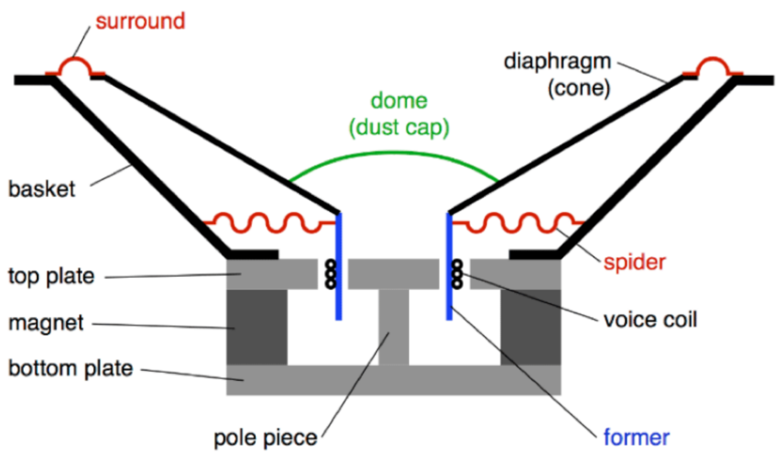


Figure 8: Cross section of an electromagnetic speaker driver with all the essential components named and pointed out [27].

When a wire conducts an electrical current it gives rise to a magnetic field that curls around the wire. When a wire is made into several loops and creates a coil the resulting magnetic field has the appearance of a magnetic dipole [28]. In figure 9 is a schematic of the magnetic field created by a magnetic dipole and a coil conducting electricity.

According to Ampere’s - Maxwell’s law the polarity of the dipole moment that is generated by a coil will be changed depending on the direction of the current. Thus, an alternating

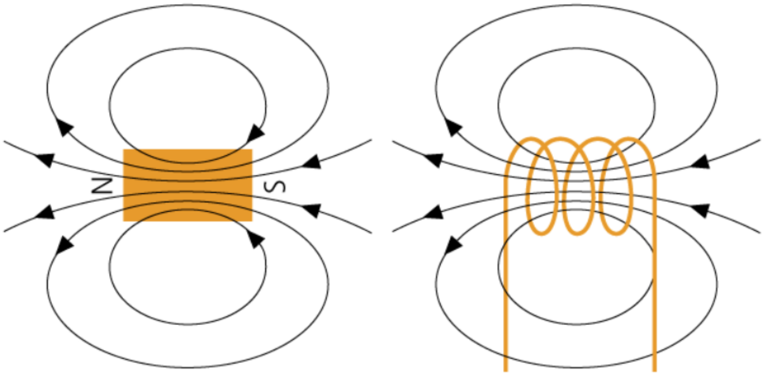


Figure 9: On the left is a magnet with its two poles noted as north (N) and south (S) and the magnetic field lines drawn out. On the right a coil that conducts electricity and its resulting magnetic field [28].

current will cause the coil to generate a magnetic field with alternating polarity [29].

In figure 8 the cross section shows that the magnet is hollow. This allows the voice coil to be placed inside the magnetic field of the magnet. The magnetic field of the magnet and the voice coil will thus be placed in the same place in space. The voice coil is mounted on the diaphragm (sometimes called membrane) of the speaker, which is the part responsible for moving the air. As an alternating current is feed through the coil the magnetic field of the coil will interact with the magnets magnetic field and either be attracted or repelled. This causes the diaphragm to move up and down [27].

The spider and surround helps to keep the diaphragm in place and connected to the basket. The basket is a rigid structure that holds the speaker and the dust cap helps keep dust and foreign objects out of the speaker that may interfere with its performance.

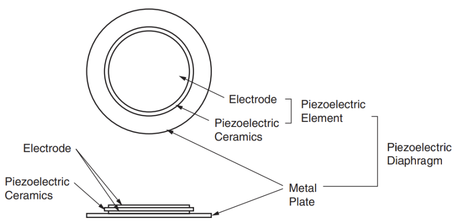
An important parameter of a speaker driver is the *excursion*, it is a measurement of the linear displacement of the diaphragm in the axis parallel to the normal of the diaphragm. The maximum linear excursion, denoted X_{max}, is the limit for when the diaphragm stops moving linearly. This can be due to several factors, when the voice coil leaves the magnetic field of the magnet the force between the magnet and the magnetic field of the voice coil will decrees. This reduction in force will result in non-linear movement of the diaphragm. It should be noted that the surrounds also could affect the X_{max}, as the diaphragm moves out the surrounds will be stretched and create more resistance to the moving coil. The bottom plate is placed to close to the voice coil the bottom plate could also hinder the movement of the diaphragm and therefore affect the excursion limit [30]. The excursion relates directly to the amount of air that will be displaced during operation and therefor the pressure variations created by the speakers.

2.6.2 Piezoelectric speakers

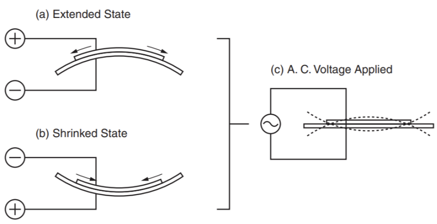
In comparison to electromagnetic speakers piezoelectric drivers do not rely on magnetism for the creation on sound. They exploit the properties of piezoelectric materials to cause movement from electrical signals that generates this disturbance in air pressure.

As stated in section 2.5, piezoelectric materials will perform mechanical movement due to the deformation that the electrical signal causes. This deformation can be realized in a different ways, this means that the mechanical movement can be created in different ways as well.

One way to create a piezoelectric speaker is to attach a circular piezoelectric material to a flexible plate, this plate will act as the speaker diaphragm. When the piezoelectric is exposed to as signal its area will start to grow and shrink which will cause the flexible plate to bend, this is called a bi-laminar plate. See figure 10a for a schematic of the speaker anatomy and figure 10b for how the speaker operates for different applied signals. In this configuration the ends of the flexible plate will move freely, by anchoring the flexible plate to some supporting structure the middle of the plate to move back and forth. Given the total bend in the material is the same the plate will have a grater out of plane movement, effectively increasing the speaker excursion. see figure 11 for comparison between attached ends and free ends. This also help lessen acoustical short circuiting that occurs when the front and back are not isolated from each other which should increase the sound output.



(a) Anatomy [31].



(b) Operation [31].

Figure 10: These figures show the anatomy (a) and operation (b) of on type of piezoelectric speaker [31].

There are other ways to realize a piezoelectric speaker, by using piezoelectric that move like springboards that are connected one one side to a support and on the other end to a loose structure that will act as the diaphragm [32]. Another way is to use piezoelectric materials that bend like the springboard type but are not attached to anything at the other end. The piezoelectric itself acts as the diaphragm [33].

According to theory, if a piezoelectric speaker were to be modeled as a single-resonance system the diaphragm resonance frequency would be the border between two separate frequency regions. The mass controlled region and the stiffness controlled region. In the stiffness controlled region the sound pressure is proportional to the frequency squared

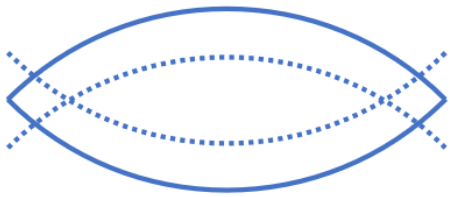


Figure 11: Comparison between bilaminar speaker with free and attached ends. The dotted lines show a speaker not attached at the ends where the ends of the speaker driver can move freely and the full lines shows when the same piezo speaker is attached to a support structure.

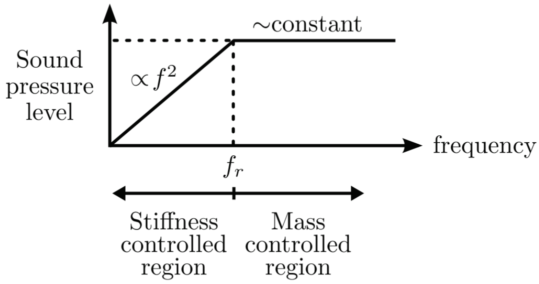


Figure 12: Theoretical frequency response if a piezoelectric loud speaker modeled as a single-resonance system. [18]

whilst the mass controlled region sound pressure is approximately constant regardless of frequency, see figure 12 [18].

3 Set-up

3.1 Circuit

The speaker needs an electrical signal to play sounds and the components that supplies the speaker with power as well as signals, a power supply, a signal generator and an amplifier will be needed and will be shown in figure 13. There is a distinction between speaker which refers to the whole system, meaning the amplifier included while the speaker driver only refers to the component that creates sound, in this case the piezoelectric speaker driver.

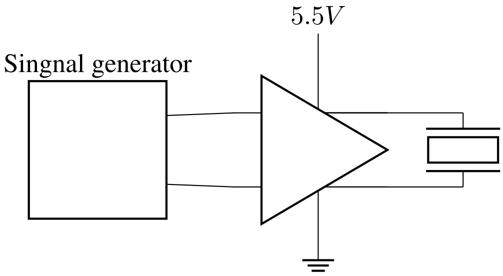


Figure 13: Block diagram of the curicuit that supports the speakers. Analog signals are feed to the audio amplifier which is powered by a voltage supply and then supply the speaker with the amplified audio signals.

3.1.1 Amplifier

The amplifier that will be used is the PAA-MAX9788-01, which is pcb-circuit supporting the Maxim MAX9788 amplifier.

The selection of amplifier was more challenging than expected. Compared to traditional electro dynamic speakers which are inductive in nature and have fairly low impedance. The piezoelectric speakers are capacitive and for lower frequencies the speaker drivers can be up towards $3k\Omega$ impedance. This is quite a lot compared to electrodynamic speakers which display only 4Ω or 8Ω impedance. The fact that piezoelectric speakers are capacitive in nature means that the amplifier need to be able to handle capacitive loads. There are some audio amplifiers that are intended for piezoelectric speakers on the market. Most of them are for pcb circuitry and only a handful can be used directly without any other development. Due to few alternatives and the time frame of the project the best of the available amplifiers was chosen.

The amplifier needs a $2.5 - 5.5V$ supply voltage and can amplify signals from $0 - 1Vpp$ to $0.3 - 19Vpp$. During the measurements the supply voltage will always be set to $5.5V$ since this will reduce the THD and noise produced by the amplifier. The supply voltage also governs the maximum amplification and increases the limit of the output signal before clipping occurs.

The amplifier has also been equiped with protection resistors at the outputs. Since the impedance changes rather much with frequency the high frequency have a tendency to draw

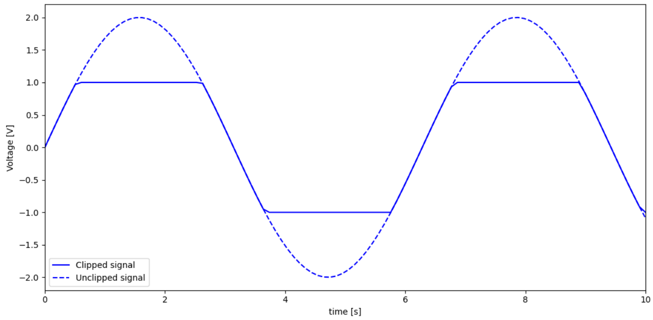


Figure 14: Here an un-clipped and a clipped signal is being showed, in this case an amplifier is trying to create a signal of $4V_{pp}$ but can not support voltages over $2V_{pp}$.

large currents which could damage the amplifier. Therefore, by placing one 10 Ω resistors at each of the outputs of the amplifier protects the amplifier from these high currents.

Clipping occurs when the maximum voltage that the amplifier can produce is lower than the amplitude of the amplified signal is. This could alter a sine wave to have more of a square appearance which essentially means that the signal has been distorted, see figure 14. Clipping can also occur when the amplifier fails to supply enough current for the load that is attached to the output.

3.1.2 Power supply

The voltage source needs to be able to supply at least $5.5V$ and currents of at least $200mA$. During measurements a TT*i* CPX400A or a TT*i* MX180TP power supply unit will used. Both units have been measured to validate that the voltage is the same between the two.

3.1.3 Signal generator

To play sound the speaker needs an electrical audio signal, this has been done in two ways. First a signal generator that can generates single frequency tones with adjustable voltage peak-peak. It allow for example 1 kHz signals to be played or a 2 kHz square wave. The speakers have also been connected to a computer to play music or continuous frequency sweeps.

For the audio measurements the speaker will be measured inside an anechoic chamber which is equipped with *Audio Precision* software and hardware, which has built-in frequency generators. The signal generator will be useful when measuring the electrical properties of the speakers and amplifier, it allows the measurements to be done out-side of the audio laboratory.

3.1.4 Speaker driver

The speaker drivers that will be evaluated is the Sonitron SPS-6555-03. It can handle signals of voltage signals of $60V_{pp}$ and has a capacitance of $480nF$. The amplifier can not supply signals with high enough peak-peak voltages that is would be outside the range for which the speaker can handle.

It was chosen due to its compatibility with the ”best” amplifier alternative, it also showed promising specifications of reasonable frequency responses and larger frequency range compared to other alternatives.

3.2 Mounting

The speakers can be mounted on virtually any surface, however a more structurally rigid material should be used to reduce the mounting surface to vibrate. Therefore a medium-density fibreboard (MDF) will be used, it is widely used in sound systems. It is very dense and resists vibrations which help reduce distortion otherwise caused by the speakers.

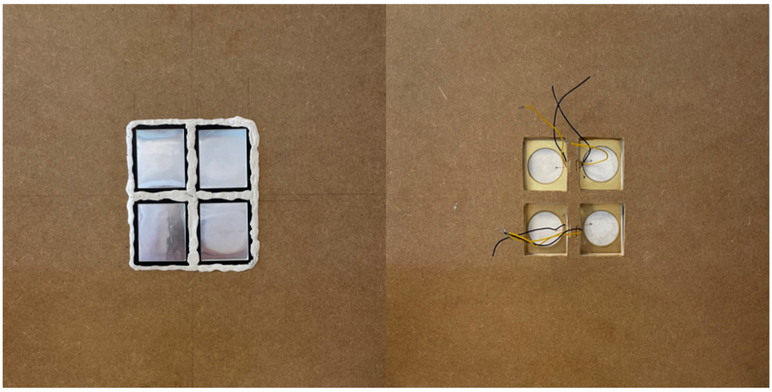


Figure 15: To the left the front of the speakers are shown and they are mounted flush to the surface using daub. On the right the backside of the speakers are shown, the material behind the speaker has been removed to reduce back reflection.

The speaker diaphragm need to be able to move freely, therefor holes will be made where the speakers are placed to reduce back-reflections of the speaker, see figure 15. In the application guide from the manufactuer of the speaker drivers there are a few different examples on how to mount the driver. After conversation with the engineers at Axis a good way to attach the speakers is to use daub, it will hold the driver in place and reduce any unwanted vibrations.

The speaker will then be mounted on a wall that is suspended in the anechoic chamber. The dimensions off the wall ensure that the back and front of the speakers are separated acoustically and removes the possibility for acoustic short circuiting. Due to the holes in the wall any sound that is being emitted backwards will propaget to the walls of the anechoic chamber where it will be absorbed. Therefore, there should not be any noticeable back reflections of the speakers. The set up in the anechoi chamber is shown in figure 16. The microphone has been measured to be 1 meter away from the center of the speaker or center of the array. The MDF wall is suspended in the ceiling of the chamber and has been placed to be as level as possible. There is a sligth tilt to it, approximately 1.4 degrees. The tilt was unavoidable, attempts to reduce it were done unsuccessful.

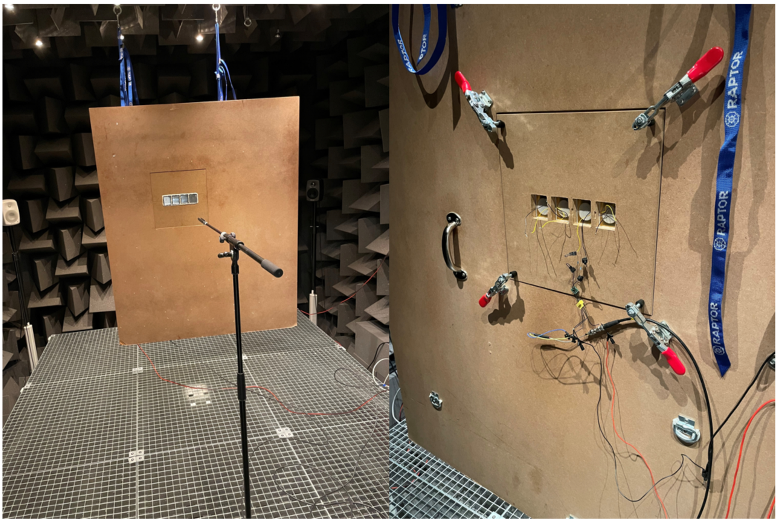


Figure 16: This is how the measurement was set up in the audio chamber. To the left, the front of the speakers and the microphone placement. To the right shows the back side and how the speaker panel has been attached to the wall. With clamps with rubber knobs, in the seams there is isolating material of a rubber/styrofoam like material to reduce vibrations between the two panels.

3.3 Connections

When using only one speaker driver the connection is simple, each output of the amplifier is connected to each end of the speaker. However when there are more than one speaker, how the speakers are connected becomes important. From section 3.1.4 the speakers are showing a capacitive behavior within the audio frequency range ($100Hz - 20kHz$). It is possible to connect the speakers in either series or in parallel with each other.

3.3.1 Two speakers

There are two possible way to connect two speakers to the two outputs of the amplifier, either in parallel or in series. Both ways works but from an electronic point of view it differs.

Series
When connected in series the total impedance over the speakers is increased compared to one speaker, see eq. 8 however the capacitive load is reduced. This is confirmed by combining eq. 4 and 8 as shown below in eq. 44

$$\frac{1}{C_T} = \frac{1}{C_1} + \frac{1}{C_2} + ... + \frac{1}{C_n} \tag{44}$$

The total capacitance of two capacitive loads that are equally capacitive will have a total capacitance of half of one of the load separately.

It should be stated that the voltage over each induvidual driver will be decreased when placed in series. From eq. 12 the voltage between two loads can be calculated. If the loads are equal the voltage is divided equally. This means that if the amplifier emitts a signal of $10V_{pp}$ over two drivers in series each driver will only have a $5V_{pp}$ voltage signal over them. From section 2.5.1, eq. 43 entails that the voltage correlated with the deformation of the crystal and thereby the excursion of the drivers' membrane. The excursion itself is related to the sound output of the driver, this means that each speaker will create less sound pressure compared to one speaker that is played with the same signal. From section 2.3.11 and 2.3.10 it is known that the SPL is expected to drop by 3dB SPL for half the signal stregth, however each additional speaker should add some amount of SPL to the overall measured SPL. If measured from the axis of symmetry this increase in db SPL should be 6 dB for two identical signals that are played in phase of each other. Therefore it is expected that when the drivers are connected in series they should produce the same SPL as one driver when feed the same signal, if measured directly in front of the speakers.

Parallel
Parallel connection between the drivers should behave differently from series connected drivers. The overall impedance of the speakers is decreased, eq. 9. The capacitive load is increased, using eq. 4 and 9 the total capacitance of the circuit can be calculated with the following, eq. 45

$$C_T = C_1 + C_2 + ... + C_n \tag{45}$$

In comparison to series connection, parallel connected drivers has the same voltage over them as the amplifier supplies, there occurs no voltage division. However, due to the decreased impedance the amplifier will need to supply higher currents to achieve the voltage. This should increase overall power consumption and increase the overall sound output.

3.3.2 Multiple speaker systems

During the project there will be measurements done on a total of four speakers, these four speakers can be connected in different configurations. The different connection configurations that will be measured are one driver; two, three and four drivers in parallel; two, three and four drivers in series; and when two sets of series connected drivers that have been connected in parallel, see figure 17.

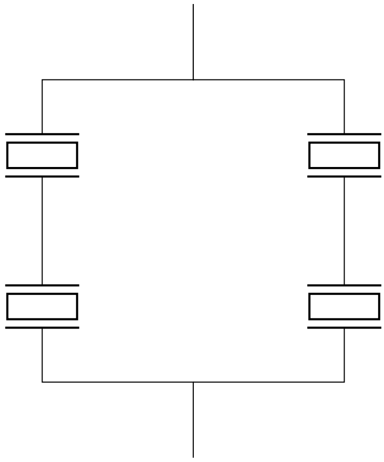


Figure 17: Circuit of two sets of two series connected speakers that have been placed in parallel. This configuration, according to theory have the same impedance as one speaker.

The idea of placing them in this last configuration is to see whether the increase in drivers has an increase in power consumption and sound output for a system that electronically appears the same.

3.4 Measurement equipment

3.4.1 Electrical measurement equipment

The measurement of the electrical signals that will be done will be measured with an oscilloscope, power meter and an impedance analyzer.

Oscilloscope
The oscilloscope can be used with differential probes which allows for measurements of the voltage over the outputs over the amplifier. Balanced probes to measure voltages at specific nodes, both of these measurements will help in understanding of the circuit.

Power meter
The power meter used is a Rohde & Schwarz programmable power meter. A power meter measures the voltage and current at the same time as well as the phase difference between the two. This allows the power meter to calculate both the apparent and actual power.

3.4.2 Acoustical measurement equipment

For the acoustic measurements are based on the measurement techniques presented by Angelo Farina, this technique is considered the industry standard and is employed by most if the available measurement products. Briefly, it measures an exponentially-swept sine signal and allows for measurements of impulse response and distortion simultaneously [34].

Anechoic Chamber
The measurements will be conducted in an anechoic chamber with a measured noise floor of 8.3 dB and has been certified by IAC acoustics and equipped with hardware and software from Audio Precision.

Microphone
The microphone that will be used to measure the speakers is the GRAS 46AE 1/2” CCP free-field standard microphone from GRAS. It has a sensibility of 50 mV/Pa and a dynamic range of 17dB to 138 dB.

4 Electrical measurements

4.1 Measurement protocol

4.1.1 Amplifier evaluation

The amplification of the amplifier need to be characterize, this will be done by measuring the input signal from the signal generator and compare it to the audio signal at the output of the amplifier. The amplifier will be feed with varying supply voltage in the range $(2.5 - 5.5V)$ and a varying input signal $(0 - 1V_{pp})$ when feed with a $1000kHz$ signal.

The set-up for this test can be seen in figure 18. The input signal will be generated by a signal generator and the supply voltage will be provided by a power supply unit. An oscilloscope, equipped with differential probes will be connected to both outputs of the amplifier.

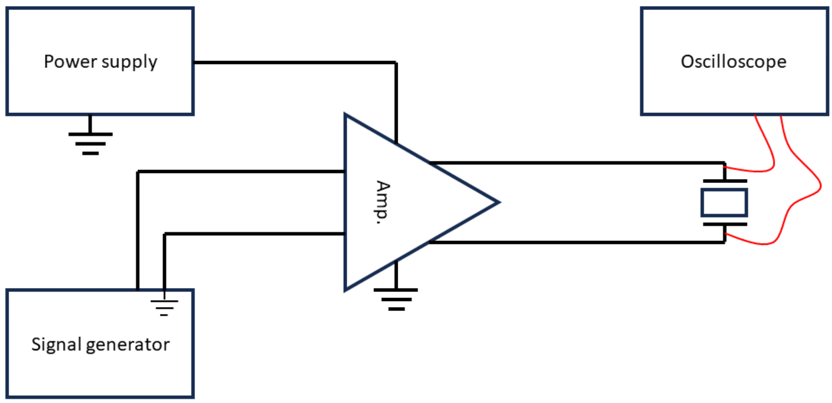


Figure 18: Set up for measuring the amplification of the amplifier

To measure the different connection configurations they will be connected as seen in figure 18, the image only depicts one speaker driver. This measurement will give some information about the voltage over the total load that the different speaker configurations display.

4.1.2 Impedance of the speaker drivers

To measure the impedance of the drivers an impedance analyzer will been used. Measuring the impedance and the phase of the drivers for different connection configurations over frequency. This measurements will be done in order to calculate the power consumption of the speakers and get a better understanding for what happens when more speakers are placed connected together.

4.1.3 Power consumption

Power measurements can be done in two ways, either by measuring the power of the whole system, i.e. the power that is supplied by the power supply or by measuring the power that the speaker drivers use.

To conduct either of these measurement a power meter will be used, it measures the phase shift between the current and voltage and can then calculate the actual power by using eq. 11. When measuring the power of the whole system the power meter is connected in series between the power supply and the amplifier, if the power over the driver is to be calculated the power meter is connected in series with the outputs of the amplifier and the driver contacts. It is also possible to calculate the power over the load with the use of eq. 11 and rewrite it with the help of eq. 1 as seen below in eq. 46.

$$P = \frac{V^2}{Z} cos(\varphi) \tag{46}$$

But to do these calculations the impedance and phase shift needs to be know. These calculations can be done after the impedance of the speakers have been measured and the voltage on the outputs.

4.2 Results

4.2.1 Amplifier evaluation

First, different supply voltages was measured against different input signal. This measurement only showed that an increase in supply voltage increased the limit until the amplifier started clipping. A high supply voltage would increase the range of voltages that the amplifier remained linear. The following measurements have after this measurement used a maximum specified supply voltage of $5.5V$.

When characterising the amplifier the input signals larger than $100mVpp$ were closing in on the amplifier starting to clipp the signal. Therefor for measurements following this measurements the ranges of amplifier input voltages will be in the ranges $0 - 100mVpp$. If the signals start clipping, this could give rise to unwanted distortions. If the distortion of the speakers are to be measured, distortion caused by the amplifier should be minimized if possible.

From measurements done on the amplifier of the output voltage signal was found to be altering slightly between for the different connection types, see figure 19.

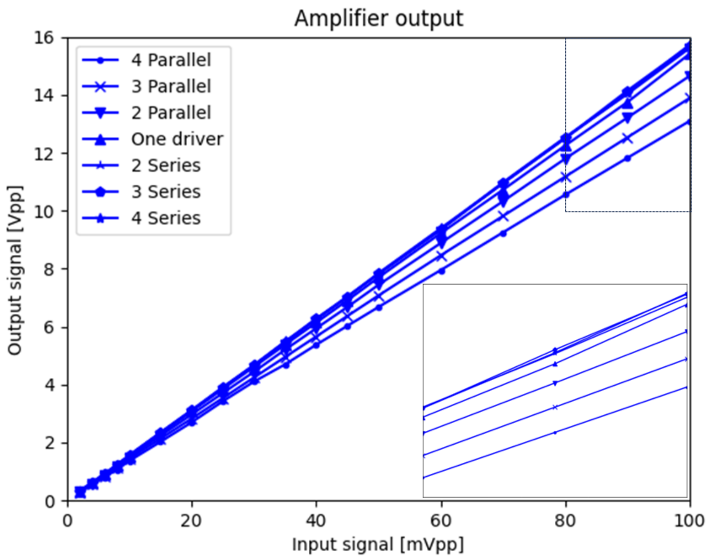


Figure 19: Output voltage [Vpp] of the amplifier for 1 kHz signals with varying input signal [mVppp].

It appears that the more speakers are placed in parallel the lesser the gain of the amplifier and the more in series the better the gain, but this seems to only increase slightly.

4.2.2 Impedance of the speaker drivers

To confirm that the impeadance for multiple speakers that are connected to each other follow what the theory suggests, measurements were done.

When measuring the impedance of the speakers drivers in the frequency span of $100Hz$ to $30kHz$ it displayed mainly capacitive properties. The speakers were connected to an impedance analyser with wires short enough that the it would not affect the measurement and the speaker was resting on the table freely. Within almost the full frequency span they displayed a 90 degree negative phase shift which is congruent with capacitors and the impedance decreases with increasing frequency. It showed less phase shift for frequencies around 300 Hz which is believed to have occurred due to internal vibrations of the speaker driver. When the speaker was pushed down into the table and had less opportunity to vibrate freely these abnormalities were no longer present. To overcome this irregular behavior it will be important that when mounting the speakers they need to have good contact with the frame/mounting that they will be place in.

The results will be presented in figure 20, the figure only shows the impedance of a few frequencies. For the full measurement data see figures 39, 40, 41 and 42 in the appendix. The real and imaginary part of the impedance was also measured, these result can be found in the appendix, see figures 43, 44, 45 and 46.

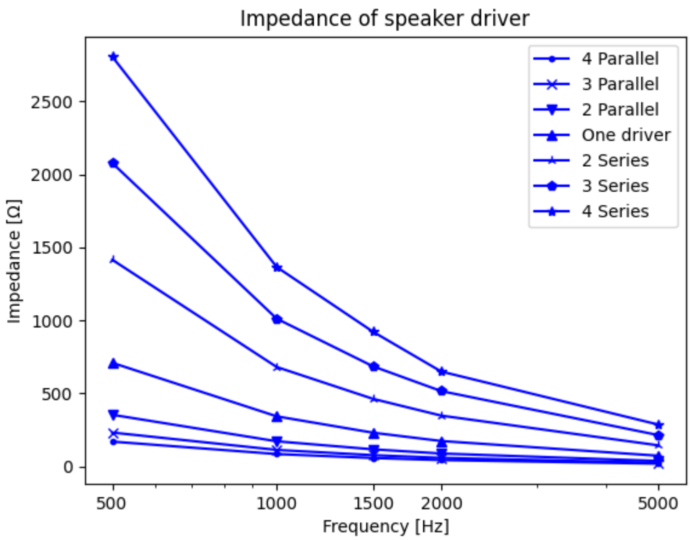


Figure 20: Displays the difference of impedance magnitude between the different connection types over frequency, at frequency 500Hz, 1, 1.5, 2, 5 kHz.

To confirm that when four speaker drivers are connected as shown in figure 17 display the same impedance as one single driver the impedance of this configuration was also measured. Results are shown in figure 21

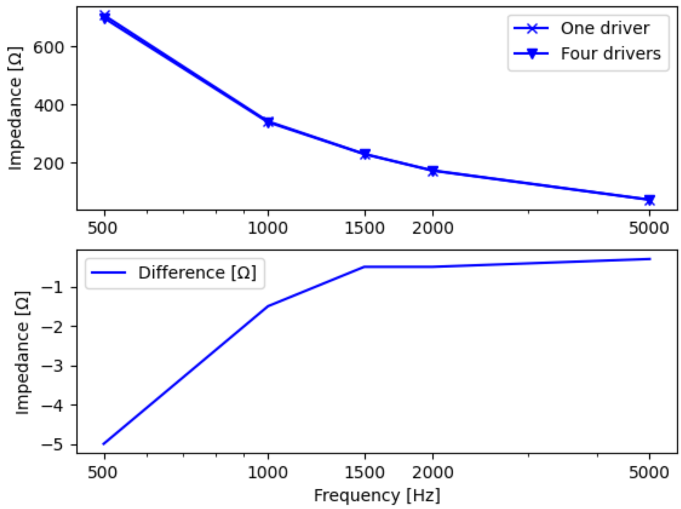


Figure 21: The top graph shows the measured values of the impedance of the speaker drivers for the one driver and four drivers connected shown in figure 17. The bottom graph shows the difference in magnitude between the two configurations.

From figure 21 the difference in impedance that separates the two configurations is very small. Close enough to claim that the impedance is the same between the two connection configurations and that they can be assumed to behave the same electronically. This can be further controlled by looking at the measured phase, the real and imaginary part of each impedance. The phase for the different connection configurations are shown in figure 22.

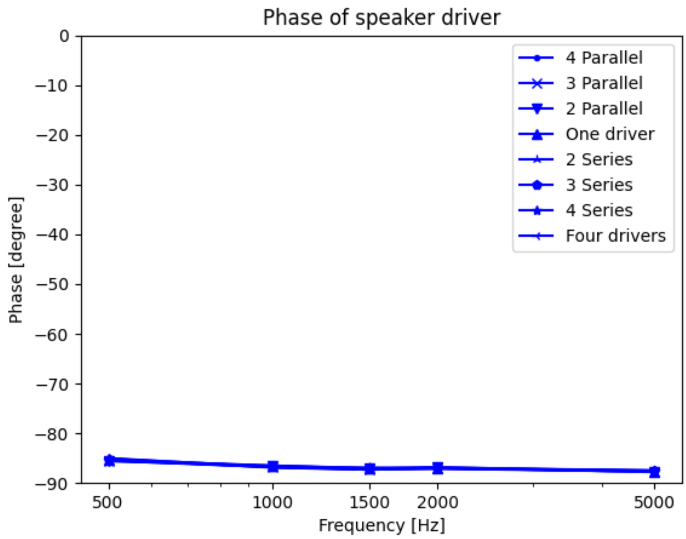


Figure 22: The phase of all different connection configurations are shown in the graph.

See figures 39, 40, 41 and 42 in the appendix for full measurement results. The real and imaginary parts of the impedance can also be found in the appendix, figures 47 and 48.

4.2.3 Power consumption

When measuring the power of the speaker drivers the could not be done with the power meter due to its low resolution. To overcome this the voltage and current that passed through the speakers was recorded and the apparent power was calculated. With the use of the measured phase shift of the speakers from the impedance measurements was used to calculate the actual power using eq 11, the results are shown in figure 23. Four speaker drivers in series appears to be zero for the whole range, this is not true and this measurement error is also caused by low resolution of the measurement equipment.

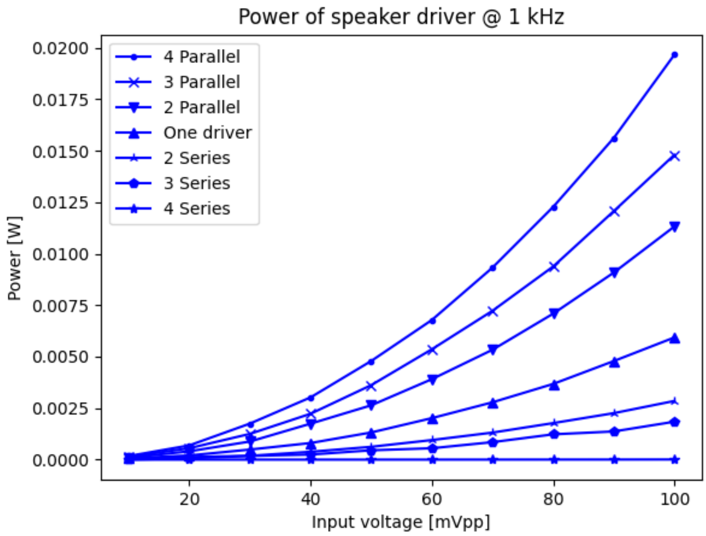


Figure 23: Calculated power consumption of different speaker driver configuration at 1 kHz at different input voltages.

For the power measurements of the whole system the apparent power and the true power were only deviating by a few milliwatts or not at all. By knowing this and knowing the limitations of the resolution of the power meter equipment the apparent power will be presented in figure 24. This still gives a good insight into the power of the speaker systems power consumption as the amplifier seems to be mostly restive and consumes much more power compared to the speaker drivers alone. Here it is visable that the apparent power for four speaker drivers in series is only slightly lower than three in series.

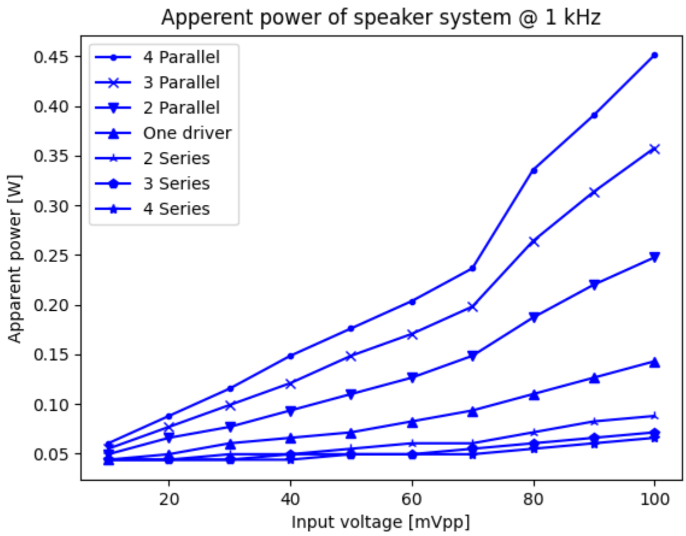


Figure 24: Apparent power of the power supply while playing different speaker driver configurations at different signal strengths at 1 kHz.

5 Audio measurements

5.1 Measurement protocol

There are a many different properties of speaker drivers that can be measured to characterize them. In this project the main three that will be measured are frequency response, THD and directivity.

5.1.1 Frequency response

By measuring the SPL for a fixed distance over the frequency span while playing a frequency sweep of the full audible frequency bandwidth. The measurement returns information of how well the sound output is for different frequencies. Industry standard to measure the SPL of loudspeaker is to place the recording device at one meters distance.

5.1.2 THD

Total harmonic distortion is a measurement that displays the difference between the input and output signal. By playing a sweep it is possible to measure eventual harmonics of the played sound, this allows the THD measurement to be done simultaneously as the frequency response measurement. The measurement set up and hardware used allows for up to the fiftieth harmonic to be recorded. The value will then been calculated with eq. 34 and presented in a graph.

5.1.3 Increasing power supplied to speaker driver

The theory states that a doubling of power should increase the measured SPL by 3 dBSPL, however does this relationship stay true for piezoelectric speakers? This is easily done by comparing the frequency response between two measurements and comparing the SPL. This can easily be done by comparing the frequency response for two measurements with different power usage.

5.1.4 Multiple drivers

When connecting multiple drivers to the same system the electrical properties will be altered as seen in section 4.2.2. How will these changes in electrical properties change the sound output of the speaker drivers?

Series
When the speakers are placed in series, the speaker drivers need to share the voltage between them. There will occur voltage division over each element and the voltage supplied by the amplifier will be divided equally over each driver. It was confirmed that each driver have, within a slim margin, the same impedance from the impedance measurements. How ever, due to the total increase in impedance due to the series connection the current that is supplied by the amplifier is much lower. This decrease in current results in lower power consumed for the system, which can be seen from the power measurements presented in figure 23 and 24. From the power measurement the power consumed by two speakers in series is roughly half of the power consumed by one lone speaker driver. This means that it can be expected

that the two series connected speakers should be playing at roughly $-3dB$ in comparison to one driver.

The total capacitive load of the speakers will be decreased when they are placed in series. This will decrease the capacitance to half for two speaker drivers in series and to a quarter of the capacitance for four drivers in series compared to the capacitance of one speaker. This will affect the cut of frequency of low-pass filtered that is caused by the resistors that are placed at the output of the amplifier and the capacative drivers. This could lead to increased frequency response for higher frequencies, which should increase for when even more speakers are placed in series as the cut-of frequency should increase as the total capacitance decreases.

Parallel
The inverse could be expected when placed in parallel, instead of sharing the voltage all the speakers are subjected to the voltage supplied by the amplifier. This will however increase overall power which was confirmed in the power measurements, see figures 23 and 24. Approximately the twice the power is used for two speakers playing in parallel compared to a lone speaker driver. This should then result in an increase in SPL of $+3dB$.

Instead of the total capacitance decreasing with increased number of series connected speakers it should increase as speakers are placed in parallel. This should decrease the cut of frequency of the low-pass filter that is realized due to the output resistors and the capacitance of the speakers.
Since increased frequency decreases the impedance of the speaker driver it may not be possible to play the parallel connected speakers at higher frequencies. Frequencies above a certain level could lower the impedance to become so low that the current would increase to a level that the amplifier could not supply. This will have to be tested carefully to find a frequency span that allows the speaker driver configurations to be tested without destroying the amplifier or that the amplifier starts clipping due to current limitations.

5.1.5 Directivity

The directivity of the speakers is measured by measuring the SPL for different angles to the speaker diaphragm normal in the vertical and horizontal directions. In figure 25, a schematic image of the measurement is displayed. The distance to the speaker shall be fixed for all directions and the signal and supply power shall be set to a fixed level.

Since the speakers are mounted on a wall it will be easier to mount the recording microphone on moving stand. The stand is essentially a one meter long arm that moves around the speaker. It will make a full frequency sweep of every fifth degree between -90 to 90 degrees where 0 degrees is in front of the speakers, i.e. in the direction of the normal of the wall. Two measurements will need to be done, one measurement for the vertical and on for the horizontal direction. The directivity of one speaker driver will be measured and for a linear array of four drivers and one square 2 by 2 array. For the array measurements the speakers will be connected as seen in figure 17.

The arm that will hold the recording microphone will move vertically (from the floor to

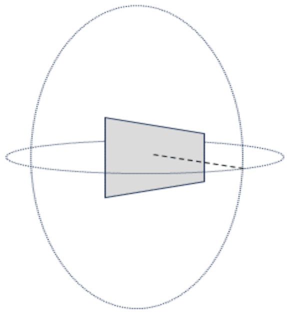
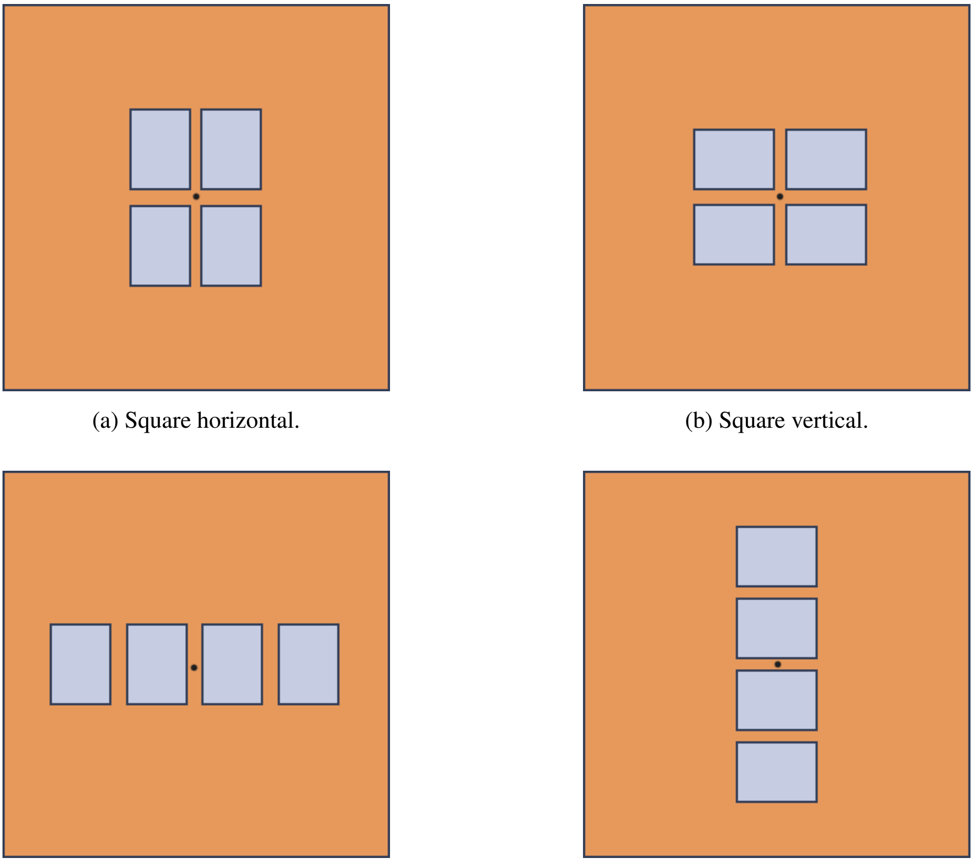


Figure 25: Here is the principle behind the directivity measurement presented, the gray square is the speakers, the dashed line represents the normal to the speaker and the dotted lines shows for which the measurements will be done. However the measurements will not measure any angles behind the speakers, i.e. no back-radiation will be measured.

the ceiling), to measure both the vertical and the horizontal orientation of the speakers the speaker will be remounted on the wall after the first measurement and the rotated 90 degrees. In the figure 26 below vertical and horizontal orientation for linear array and square array are shown. For the directivity measurements for one speaker, the speaker will be mounted in the same direction as the individual speakers are placed in figure 26. Horizontal is when the longer side of the rectangle is parallel to the direction the microphone is moving and vertical the speaker will be when the speakers have been rotated 90 degrees.

From the measurements of just one speaker driver, information of how the general behavior of the speaker drivers directionality will be found. Since they are rectangular the directionality is expected to be different between the two orientations. The array configurations are expected to show signs of interference between the sound waves. For the line array, the vertical measurement should have more prominent side-lobes due to interference between the individual drivers. The horizontal measurement should not be as subjected to interference since the measurement will be done along the axis of symmetry of the array. Regarding the square array there should be some signs of interference taking place. However, if the speaker driver were square and had the same directionality for both vertical and horizontal measurements a square array should display the same properties. It is expected that the square arrays vertical and horizontal measurement should be more alike than the same measurements done on the line array. Interference depends heavily on wavelength and therefore the largest differences should be seen for higher frequencies.



(a) Square horizontal.

(b) Square vertical.

(c) Linear horizontal

(d) Linear vertical

Figure 26: 26a and 26b show the orientation of the square array during measurements, 26c and 26d show the orientation of the linear array during measurement. The black dots show the center of the array from which the symmetry axis is placed.

5.2 Results

During the audio measurements a lot of data was collected most of the data will be presented in graphs which can be found in the following sections. However all graphs will not be presented in this section and any other information that has not been presented here can be found in the appendix.

5.2.1 Frequency response

Frequency response of one speaker driver is seen in figure 27, the frequency response of the different connection configurations can be found in the appendix, see figures 49, 50, 51, 52, 53, 54 and 55.

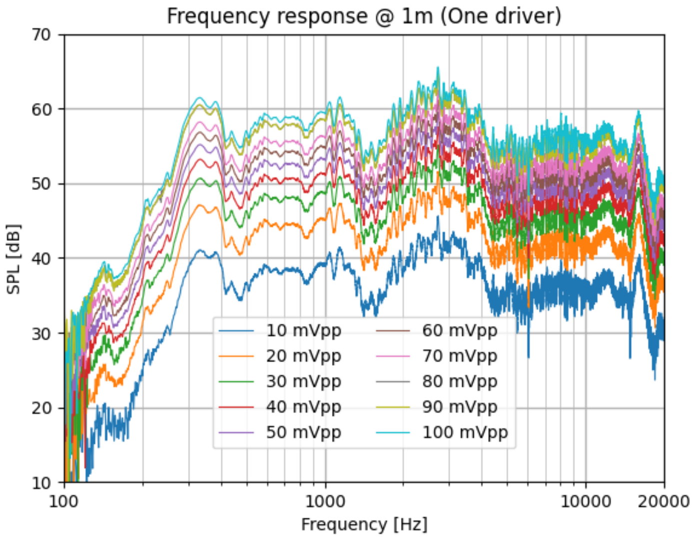


Figure 27: Measured frequency response of the speaker driver for different amplifier input signals strengths.

5.2.2 THD

The THD of each individual speaker playing at 100 mVpp input signal can be seen in figure 28. The measured THD from the series, parallel and the two types of array configurations are shown in figure 29, 30 and 31.

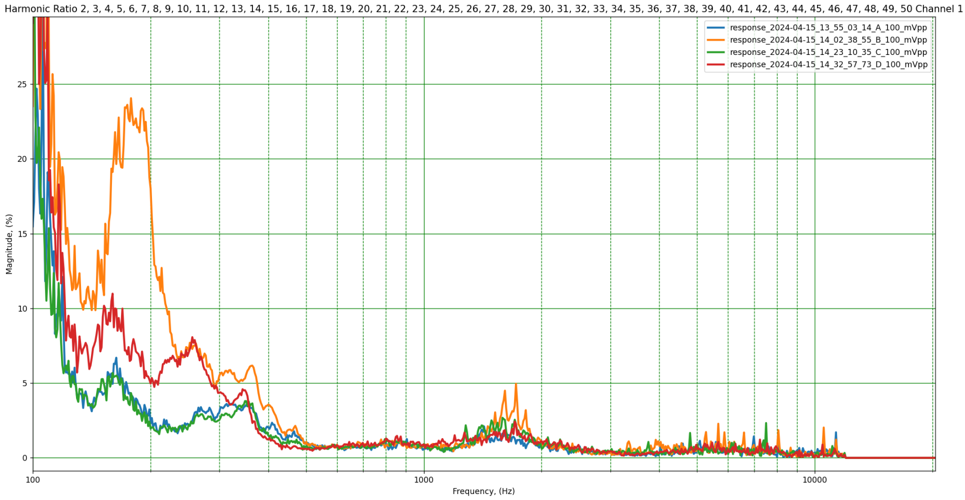


Figure 28: THD measurements from each individual speaker driver when played at 100 mVpp.

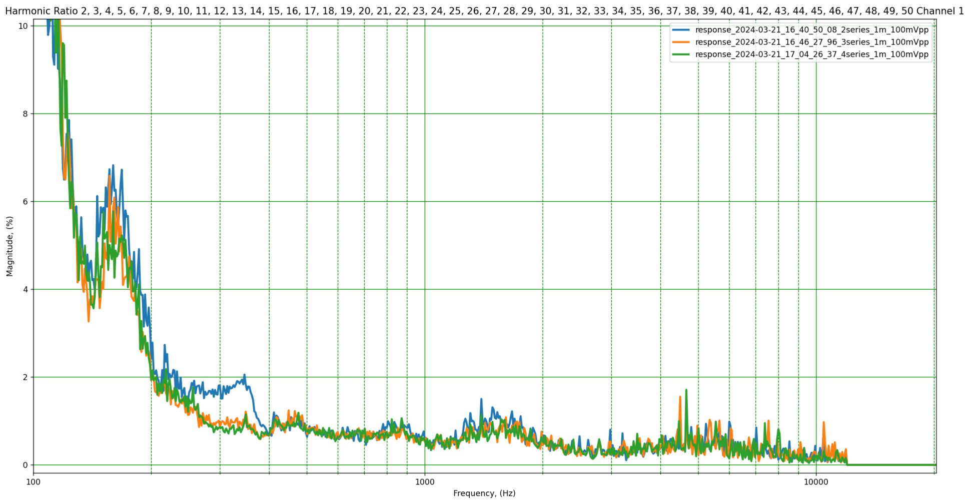


Figure 29: THD measurements from the series connections when played at 100 mVpp. Blue: 2 series, Orange: 3 series, Green: 4 series.

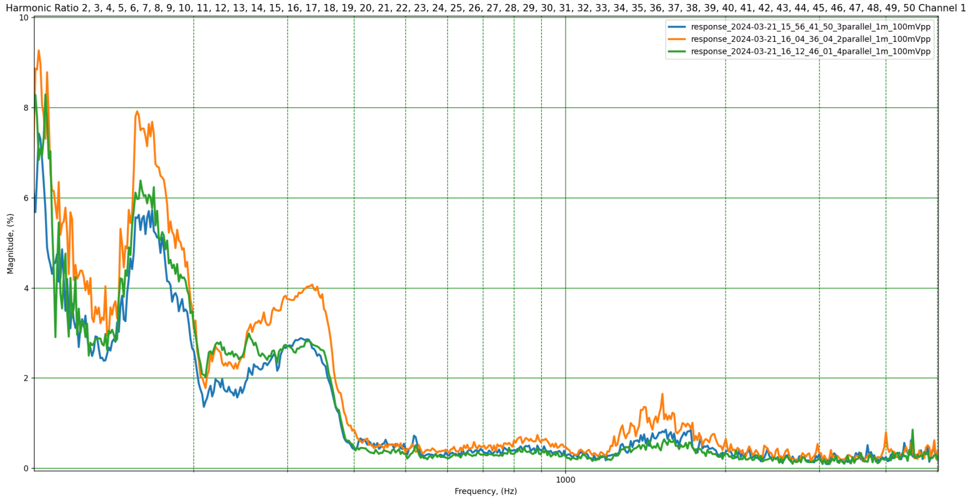


Figure 30: THD measurements from the parallel connections when played at 100 mVpp. Orange: 2 parallel, Blue: 3 parallel, Green: 4 parallel.

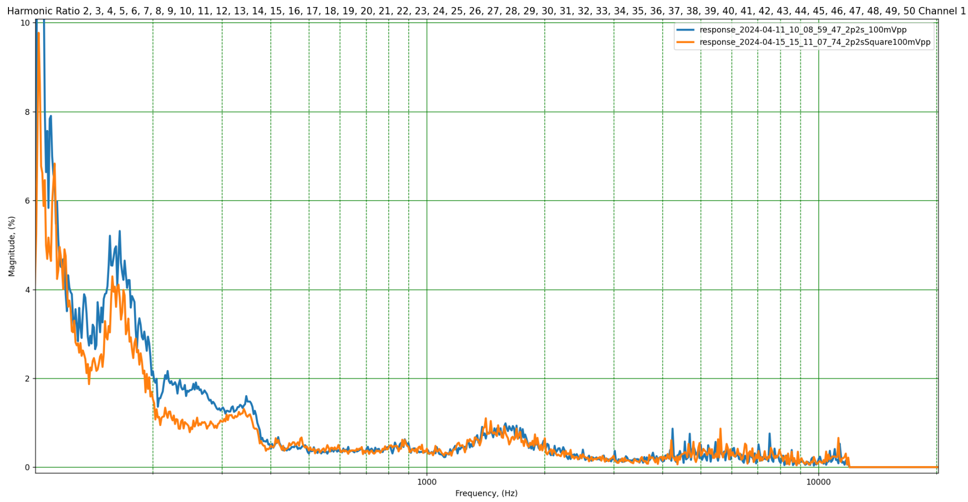


Figure 31: THD measurements from the two types of four speaker arrays connected as figure 17 shows when played at 100 mVpp. Blue shows the square array, orange shows the linear array.

5.2.3 Increasing power supplied to speaker driver

In figure 32 the SPL for one speaker driver playing at 50 mVpp input signal is compared to the SPL for one speaker playing at 100 mVpp. In the top graph the measured frequency responses are show and in the lower graph the difference between the measurements are presented and a simple averaging function is applied to smooth out the graph.

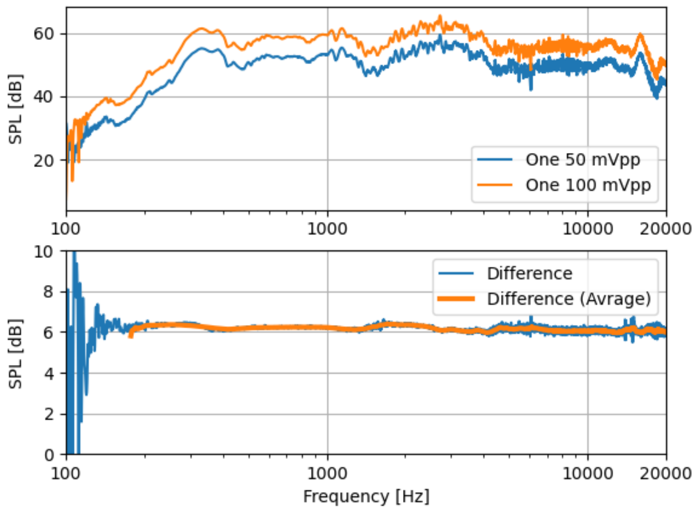


Figure 32: The frequency response of one speaker being played for a 50 mVpp and a 100 mVpp signal is compared. In the upper graph is the measured data displayed and in the lower graph is the calculated difference between them and a simple averaging function to smooth out the data.

5.2.4 Multiple drivers

In figure 33 the SPL for different speaker configurations playing a $1kHz$ signal for changing input voltages. Here we can see that the parallel connection increases sound output and series connection performers roughly the same as one speaker. From the power measurements it is apparent that parallel connections consume more power which explains the increase in sound output. The power is however less for series connections compared to one speaker although the sound output appears to be the same for both.

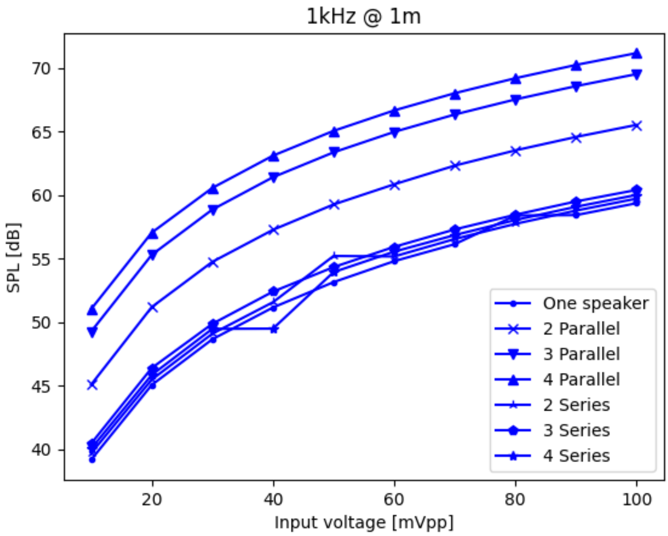


Figure 33: SPL in dB at different driving voltages for different connection configurations.

Figure 34 shows the SPL output of one speaker compared to four speakers connected as shown in figure 17 and power supplied by the power supply. In the upper graph of figure 35 the frequency response of one speaker and four speakers connected as in figure 17 are compared and in the lower graph the difference between them is presented.

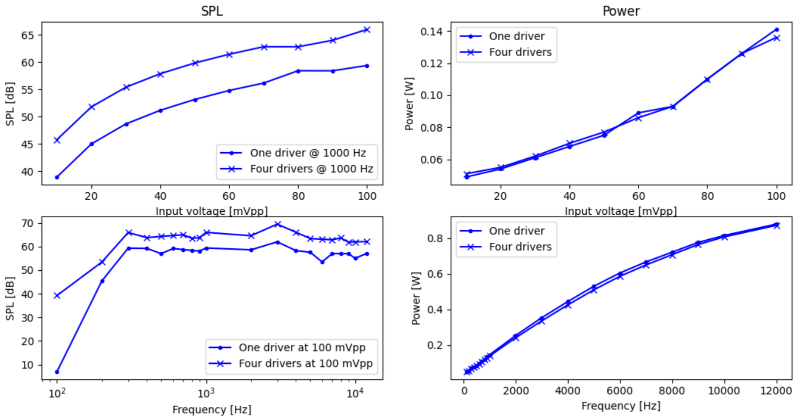


Figure 34: Comparison between one speaker driver and four speaker driver connected as shown in figure 17

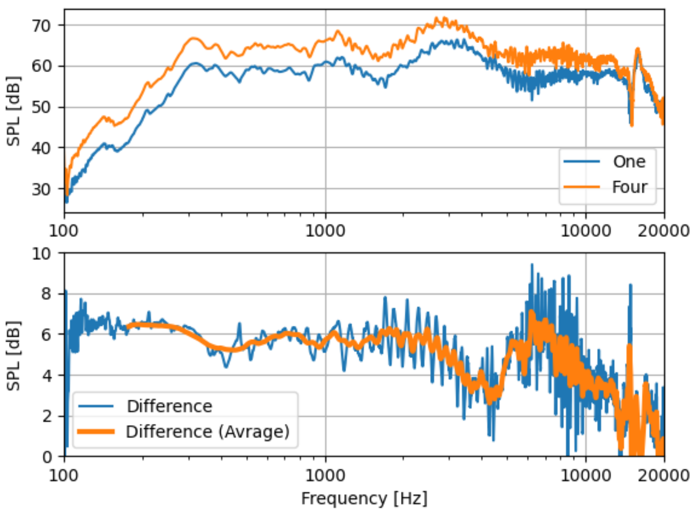


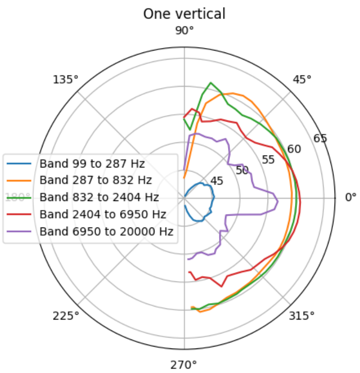
Figure 35: The average frequency response of every individual speakers and four speakers connected as in figure 17 are compared with the measured values in the upper graph and the calculated difference in the lower graph and a simple averaging function to smooth out the data.

5.2.5 Directivity

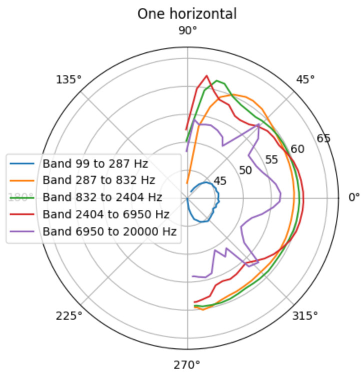
The results from the directivity measurements will be presented as the average SPL for specified frequency bands. This smooths out the graphs and gives a better understanding of the speaker drivers directivity for different frequency bands. If singular frequencies were to be shown there would be visible main and side lobes. However, due to the measurements being 5 degrees in between each iduvidual measurement these graphs lack the resolution to present any detailed information for induvidual frequencies. Furthermore, the average SPL of a frequency band is more representative if the speaker drivers are to be used to play sounds such as speech and/or music. If they where to play singular frequencies, e.i. being used as a buzzer the frequencies of interests could be measured, but this was not done.

In figure 36 the results from one speaker driver is presented. Since the speaker drivers are not completely square it was expected that the directionalty pattern of the speaker drivers should be different from the vertical and horizontal measurement. The highest frequency band show some side lobes at around $\pm 45°$ from the horizontal measurement. The vertical measurement appears to have less prominent side lobes. For lower frequencies the speaker drivers appear fairly omnidirectional. The lowest frequency band is much smaller, but this has to do with that the frequency response of the speaker drivers at low frequencies is lower.

For all measurements there seems that for angles around $+90°$ the speakers seems to loose a lot in SPL. This is most likely to a flaw in the set-up. The wall that the speaker drivers were mounted on had a slight tilt to it. This tilt could be responsible for the uneven response at the ends ($\pm 90°$) of the measurements.



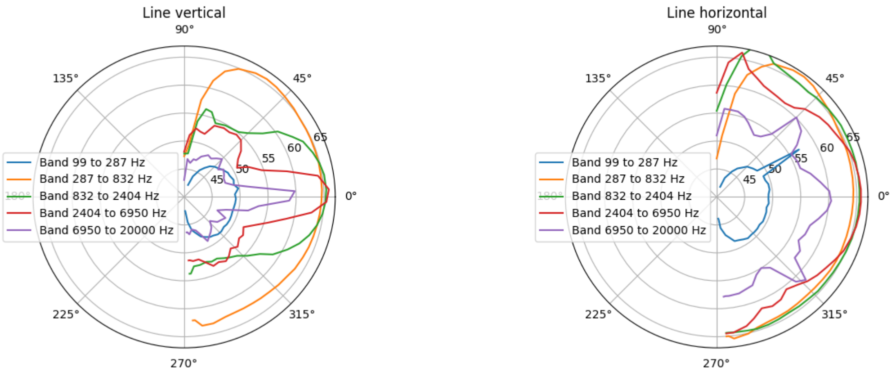
(a) One speaker, vertical



(b) One speaker, horizontal

Figure 36: Polar plots of directivity of one speaker driver

Results from the directivity measurement of the line array are presented in figure 37. For lower frequencies, below $800Hz$ the array appears to be omnidirectional. Since the speaker drivers are placed close to one another this is expexted as the ratio between the wavelengths and the distance between drivers is to small. The horizontal measurement has the same appearance as one speaker driver, except that the SPL is approximately $+6dB$ higher. The lowest frequency band has one point that stands out from the rest, this is most likely a measurement error. The vertical measurement is much more directional. The two lowest frequency bands have to long wavelengths for any destructive interference to occur but for the higher frequency bands the main lobe are very prominent and there are small side lobes.

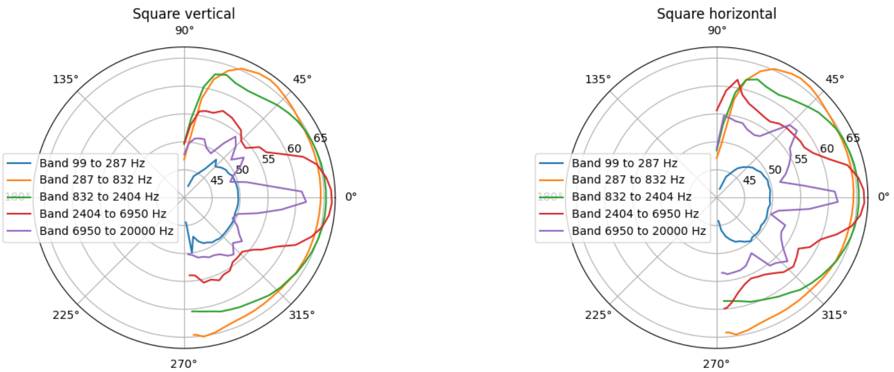


(a) Line array, vertical

(b) Line array, horizontal

Figure 37: Polar plots of directivity of line array with four speaker drivers

The square array directionality measurements are presented in figure 38 and the two measurements are fairly similar. This is completely unexpected since the dimensions of the array is more similar for both vertical and horizontal compared to the linear array.



(a) Square array, vertical

(b) Square array, horizontal

Figure 38: Polar plots of directivity of square array with four speaker drivers

6 Discussion

6.1 Resonance frequency of the speaker driver

If we recollect what was stated in section 2.6.2, if a piezoelectric speaker were modeled as a single-resonance system it would be frequency dependent up until its own resonance frequency. For frequencies greater than the resonance frequency the sound output will no longer be correlated to the frequency. From the measurements of the speaker drivers frequency response the speaker sound output follow the same figure as the theoretical case shown in figure 12 and the resonance frequency appears to be around 300-400 Hz.

This is further confirmed by the impedance measurements. When the speaker drivers were not anchored to any rigid structure their electrical properties were altered at around 300-400 Hz. At the resonance frequency of the speaker it would start to self-resonate, this would cause vibration and bending of the piezoelectric crystal. This bending that the speaker is subjected to while self-resonating could trigger the direct piezoelectric effect and thus affect the electric properties of the drivers.

6.2 Decrease in SPL at higher frequencies

After the resonance frequency (300-400 Hz) the speaker drivers frequency response is fairly flat and when played at 100 mVpp signals the SPL is comparable to that of normal conversations, 60 dB. After ≈ 12 kHz the sound output appears to gradually decrease. It is possible that the speaker drivers have a lower frequency response after this point. It could also be the effects of the low-pass filter that is being seen. The cut-off frequency of the low pass filter was calculated to be around 17 kHz using eq. 13.

It is easier to see the effect of the low-pass filter by comparing the frequency responses of one speaker to the four speakers in series, see figure 54. The calculated cut off frequency of the series connected speaker is ≈ 6 kHz. By comparing the two measurements it is possible to see that the SPL decrease in comparison for one speaker driver compared to four in series somewhere after 10 kHz. The cut-off frequency is equal to when there has occurred a drop of 3 dB in signal strength, so that the effects of the low-pass filter appears for frequencies lower than the cut-off frequency is according to theory.

The four parallel connected speaker drivers have a calculated cut-off frequency of ≈ 4 kHz. Since the measurements were only done for 100 Hz to 5 kHz the effect of the low-pass filter becomes a little harder to see. Nevertheless, the effects of the low-pass filter can be seen when comparing the four parallel connected speakers to the two parallel connected speakers, see figures 51 and 49.

6.3 Multiple speaker drivers

When connecting multiple speakers together there needed to be a few considerations before measuring them. The capacitance and thereby the impedance would be altered of the speaker drivers, this in turn would increase/decrease the voltage division over the speaker drivers and the protection resistors. The low-pass filter that is realized by the speaker drivers and the protection resistors would get different cut-off frequencies. How will this affect the overall power of the system? And how would the system react to any increases/decreases in voltage and/or current that would arise due to the altered impedance?

The speaker drivers have been measured individually and they display similar frequency responses, but there are some small variations, this will change the overall appearance of the frequency response from the measurements of multiple drivers. The changes are generally small, but will result in some small variations over the frequency range. By looking at figure 32 and 35 when comparing the frequency response of one speaker at different voltages the difference is relatively flat compared to when comparing one to four speakers for which the difference is much more uneven. But this uneven difference between is also affected by interference so the differences that are seen are not solely due to the variations of the speaker drivers.

6.3.1 Parallel and series connecting multiple speaker drivers

The parallel connections drastically decreased the impedance and the power consumption increased. To not damage the amplifier the speakers where only measured for a maximum of 5 kHz, after that the impedance became to low and the amplifier started clipping. Any measurements above 5 kHz would show more of the inabilities of the amplifier and not the speaker drivers. As the capacitance would increase when connecting the speakers in parallel the capacitive load was outside the range of the rated capacitive load of the amplifier. The amplifier has been tested to still behave as intended for frequencies below 5 kHz for even four parallel speakers but beyond this point the measurements would be plagued with non-linearities and odd behavior caused by the amplifier. A better amplifier, designed for these capacitive loads and currents could not be found when selecting components. The increased capacitance, as discussed in section 6.2, lead to the cut-off frequency to be lowered and the effects of the low-pass filter could be observed. Furthermore, when placed in parallel the measured SPL were higher compared one speaker, but that can in part be explained by the increased power consumed. Further, when the speaker drivers are connected in parallel the voltage is placed on all the speaker drivers and the total voltage over the speakers is the same as the voltage supplied by the amplifier. This triggers a larger deformation of the piezoelectric materials as seen from the equation that describes the converse piezoelectric effect, eq. 41.

For series connected speakers the capacitance will be decreased for each additional speaker driver. When the speakers were connected in parallel the total capacitive load of the speaker driver was larger than the rated capacitive load of the amplifier. However, in series the total capacitive load realized by the speaker dirvers was still within the range of the capacitive loads that the amplifier was specified for. Higher impedance results in a lower current since the voltage was the same as it was for one speaker.
When the speakers were connected in series the impedance of the speaker drivers became

only larger as more speakers were connected. Since there occurs a voltage division between the protection resistors and the speaker drivers this increased impedance will increase the voltage placed over the total load of the speaker drivers, this can be calculated with eq. 12. From this relation it is possible to deduce that if the impedance of the second load becomes larger, the voltage in the node between the two components will increase. It can however never increase to be larger than the voltage that is supplied, in this case by the amplifier. The ratio between the total load seen by the amplifier (protection resistors and speaker drivers) and the speaker drivers total load gives governs the ratio of the total voltage that can be measured in the node between the protection resistor and speaker drivers. In the case of a single speaker driver the ratio is already such that most of the voltage is placed on the speaker driver and not wasted on the protection resistors. However, when more speakers are connected in series the total load of the speaker drivers is increased and this ratio changes, but only minimally as it also increases the total impedance seen by the amplifier outputs. This results in a minimal increase in the voltage place on the load and decreases the voltage wasted in the protection resistors. This effectively increases the total voltage placed over the speakers, from eq. 41 the deformation in the piezoelectric crystal is governed by the voltage placed upon the material. This slightly increases the power efficiency of the series connected speaker drivers compared to when connected in parallel as less power is consumed by the protection resistors. However, since they are placed in series the voltage will be divided over each speaker meaning that each speaker will get half, a third or a quarter of the total voltage depending on the number of speakers connected. This resulted in no increase in SPL as they performed approximately the same as one speaker driver, it did however alleviate the system putting less strain on the amplifier.

When designing loudspeakers with piezoelectric drivers the protection resistors placed on the outputs of the amplifier should be as conductive as possible to increase the voltage over the load, but resistive enough to still protect the amplifier.

6.3.2 Increased power efficiency of multiple speaker drivers

From the power measurements the power used by four speakers in parallel and in series seems to be the same for a given SPL, although parallel connections can produce overall higher SPL and utilize more power they appear to be equally power efficient. Parallel connections proved to be problematic for the system, a lower cut of frequency of the low-pass filter, higher currents that caused clipping in the amplifier and could damage the electronics if pushed far enough. Series connecting the speakers will not increase the SPL but it increases the cut-off frequency of the low-pass filter and draws less currents which reduces the risk of damaging the amplifier. By connecting the speakers as seen in figure 17 resulted in a speaker configuration that had the same impedance as one speaker but produced higher SPL without the strain on the system that the parallel connections caused, but did not generate higher SPL than the parallel connections.

From figure 34 and 35 the SPL difference between one and four speakers, connected as seen in figure 17, are shown and was measured directly in front of the speaker drivers along the symmetry axis, see dots marked in figure 26). A 6 dB difference has been measured for almost every input signal strength and generally the whole frequency range. What is interesting is that the power used by the speakers is the same. Since the measurement has been directly in front of the speaker there should be totally constructive interference. Each speaker driver of the four speaker driver speaker gets a quarter of the power each, essentially lowering the generated sound output by 6 dB. Since there occurs total constructive interference the pressure will increase by times the number of speakers, in this case 4. Each doubling of pressure increases the SPL by 6 dB, four would then be 12 dB increase, 12-6 = +6 dB. So by utilizing multiple speakers and interference the power efficiency can be increased. The measurement does not display a 6 dB increase for all frequencies, this could occur due to many factors. If it is caused by a measurement error or by destructive interference or any other effect has not been confirmed.

Measurements from directly in front of the speakers should not be confused with the increased SPL gained due to mutual coupling since mutual coupling measures the total radiated power of the speakers and point measurements do not give information of the total radiated energy. This increase has to do with interference between the sound waves. An increase of 3 dB SPL due to mutual coupling is only when the distance between two separate speakers are shorter than half a wavelength, if the distance between the speakers is increased the increase in SPL gained from mutual coupling is gradually decreased until the there is no additional increase. The approximate frequency for which mutual coupling stops to occur can be derived from eq. 39. The distance between the speaker driver edges is approximately 1 cm which correlates to a frequency of 5.4 kHz. Since the speaker drivers are not omnidirectional this decrease and the center of the speaker drivers are even farther apart it can expected that the effects of mutual coupling should decrease at even lower frequencies. Since the effect of mutual coupling is reduced as side lobes become visible in the polar plots, mutual coupling can be expected to diminish when the speaker becomes less and less omnidirectional. From the measurements the diminishing of mutual coupling appears to start at around 3 kHz and then there is a gradual reduction in radiated energy until it stabilizes at around 19 kHz where the radiated energy is roughly the same for both speakers. 3 kHz relates to a distance between the speaker of $\approx$ 3 cm which is closer to the distance between the center of both speakers. From 3 to 19 kHz there is a gradual decrease, this has essentially increased the radiated energy of the speaker within most of the human hearing range. However, from the calculations this should occur at 9 kHz, the measurements where done at 5 degree intervals which is fairly low angular resolution, the speaker drivers are not ideal omnidirectional sound sources. Both of these factors complicates these calculations and are most likely responsible for the mismatch between theory and my measurements.

6.3.3 THD of multiple speaker drivers

The results are within reasonable values for frequencies above 400 Hz. One reason to why the THD is high for frequencies lower than 400 Hz is main due to the SPL being drastically lower. The THD is calculated with eq. 34 and from this equation we know that the THD is a ratio compared to the response of the fundamental frequency. Since the frequency response of the speaker drivers are lower for these frequencies the THD will become larger.

There appears to be more THD when the speakers are placed in parallel even though the increase in radiated SPL. As stated before the parallel connected speakers are require more power and more current. The amplifier gives rise to more THD even before it starts clipping and this could be the cause of the increased THD for the speaker drivers when connected in parallel. The series connected speaker configurations are more lenient on the amplifier and display less THD even compared to the individual speaker drivers.

6.4 Directivity

When the directivity measurements were done the speakers were connected as shown in figure 17, this was done since this configuration did not unnecessarily strain the system and electronically was closest comparable to one speaker. The speaker drivers appear to be omnidirectional for lower frequencies but as the frequency increases a main lobe and side lobes become more apparent. This is believed to have to do with the fact that the speakers are rectangular and not round, the side lobes are more prominent for when the speakers have been measured horizontally as seen in figure 36b. This measures along the longer side and there occur interference by the sound that is created by the far ends of the speaker. The same tendencies can be identified for the vertical measurement, see figure 36a, but since the speakers is less wide in this direction the effects from the interference along this axis is less obvious.

The line array did for the horizontal measurement look similar to that of one speaker, this measures essentially the same axis and the multiple element do not interfere with each other along this axis. From the vertical measurement there is a clear difference between the line array compared to the measurements of one speaker driver. For frequencies below 1 kHz there still is quite the omnidirectional pattern, but above 1 kHz there is a prominent main lobe that becomes less wide as frequency increases giving a highly directive sound profile.

The square array is less directive compared to the vertical line array measurement but slightly more directive compared to a single speaker driver. It still holds some similarity to the directionality patterns that the individual speakers display but the pattern will become more directive if more speakers are added. Since the speakers are feed signals at the same time they are playing in phase with each other there will always occur constructive interference along the axis that is placed in the center of the array parallel to the speaker diaphragm normal.

6.5 Materials

As stated in section 2.5.2 the most common type of piezoelectric material that is being used is PZT, which contains lead. The speakers that have been used during the project are believed to not contain lead. The manufacturer of the speaker drivers do not disclose what materials that the speakers are made out of, there is however a section in the data sheets where the fabricator mentions which materials are not used in their products and there it is specified that the speaker driver should not contain lead. It does not however specify if it contains PZT. Presumably since PZT contains lead the piezoelectric materials should not be PZT but they do not specify this any further [35].

In the EU and other parts of the world lead is banned for use in electrical products, in 2011 a new directive on hazardous substances was presented 2011/65/EU. It restricts the use of certain hazardous substances in electrical equipment including lead. However PZT had an exemption for the use as piezoelectic materials titled 7(c)1, which supposedly should expire in July 2016 [36]. The exemption was renewed and extended and would expire in july 2021. At the time the data sheet was published it could have been in compliance with the EUs' RoHS directive. The exemption was renewed again and extended and the new expiry date is in July 2024 [37].

There are efforts in trying to find a replacement for PZT that is in compliance with the EUs' RoHS directive [38] [39].

6.6 Outlook

Today there is great efforts in making piezoelectric speaker drivers intended for in-ear applications (examples: headphones, hearing aid). The general direction of the market and research reports appears to be towards in-ear applications. As for the future in piezoelectric loudspeaker with applications in intercom and audio systems is still to be determined. Piezoelectric speakers are already being used for high frequency applications such as tweeters . However, full range piezoelectric speaker alternatives are few and do not outperform the traditional electromagnetic speaker drivers.

The industry seems to be more focused on in-ear applications, this is partly due to piezoelectric speakers are easier to make smaller compared to electromagnetic speakers which require a permanent magnet which results in a larger foot print. The distances for which in-ear applications need to radiate sound is much shorter and it is possible to achieve higher SPL for lower frequencies. Also as the acoustic properties of the ear canal differ greatly compared to that of free-space. Piezoelectric speakers are decreasing in size and perform rather high SPL's, as of 2021 one article shows simulated results of a 64 mm^2 footprint speaker that produces 80 dB SPL play a 1 kHz signal at 10 cm distance [40].

Other than acoustical performance aspects such as manufacturability and energy efficiency needs to be taken into account. Electromagnetic micro speakers have supposedly low energy efficiency and complex manufacturing processes, the potential seen in piezoelectric speakers is that they could replace them eventually. As it stands today their sound output appears to be lesser that of the micro electromagnetic speakers. There are progress in creating speaker designs that simplify the manufacturing processes and also perform adequate SPL [33]. There are some examples of piezoelectric speaker drivers intended for in ear use that have reached the market [41], [42], [43].

For loudspeaker there seems to be less interest from the industry, but there are examples of full-range piezoelectric speakers that are avaliable on the market [44]. It shows promising specifications with high SPL and low power consumption. The company claims to be the worlds first full range piezoelectric speaker.

As of today with the products that are available piezoelectric loudspeakers could replace electro-magnetic speakers in smaller sound systems such as speakers in computers, phones and desktop speakers where a lower maximum SPL satisfies the need. In loudspeaker systems that need to create higher SPL in noisier environments or when the listner is farther away, for example in subway systems, alarms and concerts electromagnetic speakers are better. Generally in these larger sound system the quality of the sound and the maximum SPL is more important compared to the size and energy efficiency of the speakers and therefor electromagnetic speakers are a better option. In my opinion piezoelectric speaker will not replace electromagnetic speakers completely but rather be an additional option for new speaker innovation.

6.7 Future research

Piezoelectric speakers have the ability to be used as microphones as well as speakers. Simply put, the low weight of the speakers allows the speaker membrane to move due to incoming sound waves. If the membrane starts moving it will cause bending of the piezoelectric material, the direct piezoelectric effect generates electrical signals from the deformation that is caused by the sound waves. In this project the main focus was to evaluate the speaker properties of piezoelectric speakers but as a future project to evaluate the microphone properties of piezoelectric speakers would be interesting. This could lead to products that require both speakers and microphones to only need one component that satisies the need for both instead for two seperate components.

During the project I have stumbled upon articles that evaluate the energy harvesting properties of piezoelectric speakers, for example [45]. The idea is that background sound and noise triggers the direct piezoelectric effect and the electrical signal is then used to charge a battery for example. This is possible but to what amount of energies that can be collected and how to increase the energy collected is still researched. How good would piezoelectric speakers be at harvesting energy compared to these energy harvesting piezoelectric actuators?

As stated in section 3.1.1 it was explained that the amplifier used for this project was sub-optimal and it would have been interesting to try the speaker drivers with a better and more optimal amplifier. This project has still had the possibility to research multiple factors but has potentially been limited by due to the lesser components available. There are a few amplifiers out there that could be used, the time frame of the project was fairly short it was not viable to wait for three to four months until the speakers had arrived. Comparing more types of piezoelectric speakers could also be useful to see if there are any common behaviors between different types of speaker drivers and some behaviors that do not translate to other speaker drivers.

References

[1] R. E. Berg, *Sound*, 2024. **url**: %5Curl%7Bhttps://www.britannica.com/science/sound-physics%7D.%20(2024/02/12).

[2] T. E. o. E. Britannica, *Speed of sound*. **url**: %5Curl%7Bhttps://www.britannica.com/science/speed-of-sound-physics%7D%20(2024/02/12).

[3] T. Benson, *Speed of sound*. **url**: %5Curl%7Bhttps://www.grc.nasa.gov/www/k-12/BGP/sound.html%7D.

[4] T. E. o. E. Britannica, *Interference*. **url**: %5Curl%7Bhttps://www.britannica.com/science/interference-physics%7D%20(2024/04/4).

[5] M. Long, "2 - fundamentals of acoustics," **in***Architectural Acoustics (Second Edition)* M. Long, **editor**, Second Edition, Boston: Academic Press, 2014, **pages** 39–79, ISBN: 978-0-12-398258-2. DOI: https://doi.org/10.1016/B978-0-12-398258-2.00002-7. **url**: https://www.sciencedirect.com/science/article/pii/B9780123982582000027.

[6] N. R. C. (C. on Low-Frequency Sound **and** M. Mammals, "Appendix a, comparison of sound-pressure reference levels in air and water," **in***Low-Frequency Sound and Marine Mammals: Current Knowledge and Research Needs* National Academies Press (US), 1994. **url**: %5Curl%7Bhttps://www.ncbi.nlm.nih.gov/books/NBK236684/%7D%20(2024/02/12).

[7] U. N. Stockholm, *What is spl?* **url**: %5Curl%7Bhttps://www.uns.nu/spl-what-is.html%7D%20(2024/02/12).

[8] N. C. for Environmental Health, *What noises cause hearing loss?* **url**: %5Curl%7Bhttps://www.cdc.gov/nceh/hearing%5C_loss/what%5C_noises%5C_cause%5C_hearing%5C_loss.html%7D%20(2024/02/12).

[9] I. S. University, *Radiated fields of ultrasonic transducers*. **url**: %5Curl%7Bhttps://www.nde-ed.org/NDETechniques/Ultrasonics/EquipmentTrans/radiatedfields.xhtml%7D%20(2024/02/12).

[10] K. Wouk, *What is audio distortion, and what causes it?* **url**: %5Curl%7Bhttps://www.howtogeek.com/821586/what-is-audio-distortion/%7D.

[11] D. Berners, *Total harmonic distortion*. **url**: %5Curl%7Bhttps://www.uaudio.com/blog/total-harmonic-distortion/%7D%20(2024/04/5).

[12] E. Stahlhammer, *Spl and watts: Evaluating loudspeaker power ratings to more accurately predict performance*. **url**: %5Curl%7Bhttps://pro.harman.com/insights/av/spl-and-watts-evaluating-loudspeaker-power-ratings-to-more-accurately-predict-performance/%7D%20(2024/03/04).

[13] J. Borwick, "3 - loudspeaker and headphone handbook," **in***Loudspeaker and Headphone handbook* Focal press, 2012.

[14] E2S, *Db (decibel) ratings*. **url**: %5Curl%7Bhttps://www.e2s.com/references-and-guidelines/db-decibel-ratings%7D%20(2024/05/2).

[15] F. D. Purves D Augustine GJ, *Neuroscience. 2nd edition*. Sinauer Associates, 2001. **url**: %5Curl%7Bhttps://www.ncbi.nlm.nih.gov/books/NBK10924/%7D%20(2024/02/12).

[16] T. E. o. E. Britannica, *Phon*. **url**: %5Curl%7Bhttps://www.britannica.com/science/phon%7D%20(2024/05/2).

[17] S. F. on ResearchGate, *Equal-loudness curves*. **url**: %5Curl%7Bhttps://www.researchgate.net/figure/Equal-loudness-curves-from-Robinson55-Each-line-is-at-the-constant-phon-value%5C_fig7%5C_34029037%7D%20(2024/05/03).

[18] L. K. Chiu, D. V. Anderson **and** B. Hoomes, "Audio output enhancement algorithms for piezoelectric loudspeakers," **in***2011 Digital Signal Processing and Signal Processing Education Meeting (DSP/SPE)* 2011, **pages** 317–320. DOI: 10.1109/DSP-SPE.2011.5739232.

[19] Nationalencyklopedin, *Piezoelektricitet*. **url**: %5Curl%7Bhttps://www.ne.se/uppslagsverk/encyklopedi/l%5C%C3%5C%A5ng/piezoelektricitet%7D%20(2024/05/15).

[20] O. Team, *Piezoelectric effect*. **url**: %5Curl%7Bhttps://optimizeias.com/piezoelectric-effect/%7D.

[21] N. Sezer **and** M. Koç, "A comprehensive review on the state-of-the-art of piezoelectric energy harvesting," *Nano Energy*, **jourvol** 80, **page** 105 567, 2021, ISSN: 2211-2855. DOI: https://doi.org/10.1016/j.nanoen.2020.105567. **url**: https://www.sciencedirect.com/science/article/pii/S2211285520311411.

[22] N. Nakata, *Basic physics for siesmology*. **url**: %5Curl%7Bhttps://www.mit.edu/~nnakata/page/Teaching%5C_files/GEOPHYS130/GEOPHYS130%5C_notes%5C_all.pdf%7D%20(2024/03/12).

[23] saVRee, *Strain and strain types*. **url**: %5Curl%7Bhttps://www.savree.com/en/encyclopedia/strain-and-strain-types%7D%20(2024/03/12).

[24] Q. C. I. A. of Japan, *Quartz crystal devices*. **url**: %5Curl%7Bhttps://www.qiaj.jp/pages/frame20/page01-e.html%7D.

[25] J.-F. Li, "Fundamentals of piezoelectricity," **in***Lead-free piezoelectric materials* Wiley-VCH GmbH, 2021.

[26] A. Kholkin, N. Pertsev **and** A. Goltsev, "Piezoelectricity and crystal symmetry," **injanuary** 2008 **pages** 17–38, ISBN: 978-0-387-76538-9. DOI: 10.1007/978-0-387-76540-2_2.

[27] Stetron, *A loudspeaker driver breakdown*. **url**: %5Curl%7Bhttps://www.stetron.com/loudspeakerdriver-breakdown/%7D%20(2024/02/14).

[28] Nationalencyklopedin, *Magnetism*. **url**: %5Curl%7Bhttp://www.ne.se/uppslagsverk/encyklopedi/l%C3%A5ng/magnetism%7D%20(2024/02/14).

[29] FOSCO., *A plain explanation of maxwell's equations*. **url**: %5Curl%7Bhttps://www.fiberoptics4sale.com/blogs/electromagnetic-optics/a-plain-explanation-of-maxwells-equations%7D%20(2024/02/15).

[30] Eminence, *Understanding loudspeaker data*. **url**: %5Curl%7Bhttps://eminence.com/pages/support%5C_%5C_understanding-loudspeaker-data%7D%20(2024/05/15).

[31] Murata, *Piezoelectric sound components application manual*. **url**: %5C%7Bhttps://www.murata.com/~/media/webrenewal/support/library/catalog/products/sound/p15e.ashx%7D%20(2024/03/20).

[32] R. Liechti, S. Durand, T. Hilt **andothers**, "A piezoelectric mems loudspeaker lumped and fem models," **in***2021 22nd International Conference on Thermal, Mechanical and Multi-Physics Simulation and Experiments in Microelectronics and Microsystems (EuroSimE)* 2021, **pages** 1–8. DOI: 10.1109/EuroSimE52062.2021.9410865.

[33] F. Stoppel, C. Eisermann, S. Gu-Stoppel, D. Kaden, T. Giese **and** B. Wagner, "Novel membrane-less two-way mems loudspeaker based on piezoelectric dual-concentric actuators," **in***2017 19th International Conference on Solid-State Sensors, Actuators and Microsystems (TRANSDUCERS)* 2017, **pages** 2047–2050. DOI: 10.1109/TRANSDUCERS.2017.7994475.

[34] A. Farina, "Simultaneous measurement of impulse response and distortion with a swept-sine techniquem," **in***AES Convention (February 2000)*.

[35] Sonitron, *Catalogue 2014*, 2014. **url**: %5Curl%7Bhttps://www.keiconn.com/assets/pdf/Sonitron%5C_eCatalog.pdf%7D%20(2024/05/15).

[36] Ö.-i. e.V., *Clarification questionnaire exemption no. 7(c)-i*. **url**: %5Curl%7Bhttps://rohs.exemptions.oeko.info/fileadmin/user%5C_upload/RoHS%5C_Pack%5C_22/Exemptions/7c%5C_I/Ex%5C_7c-I%5C_Applicant%5C_2%5C_COCIR%5C_response%5C_Clarification%5C_Questions%5C_final28012021.pdf%7D%20(2024/05/14).

[37] SiliconExpert, *Rohs exemptions report*. **url**: %5Curl%7Bhttps://www.siliconexpert.com/blog/rohs-exemptions-report/%7D%20(2024/05/14).

[38] S. Zhang, R. Xia **and** T. R. Shrout, "Lead-free piezoelectric ceramics vs. pzt?" **in***2006 15th ieee international symposium on the applications of ferroelectrics* 2006, **pages** 201–207. DOI: 10.1109/ISAF.2006.4387860.

[39] K. Ichihashi, K. Tsukamura, T. Kimura, T. Kasashima **and** M. Yamazaki, "Design of (k, na)nbo3-based lead-free piezoelectric sounder by utilizing different material properties from pb(zr, ti)o3," *IEEE Transactions on Ultrasonics, Ferroelectrics, and Frequency Control*, **jourvol** 71, **number** 1, **pages** 46–55, 2024. DOI: 10.1109/TUFFC.2023.3311442.

[40] R. Liechti, S. Durand, T. Hilt **andothers**, "A piezoelectric mems loudspeaker lumped and fem models," **in***2021 22nd International Conference on Thermal, Mechanical and Multi-Physics Simulation and Experiments in Microelectronics and Microsystems (EuroSimE)* 2021, **pages** 1–8. DOI: 10.1109/EuroSimE52062.2021.9410865.

[41] XMEMS, *Montara*. **url**: `%5Curl%7Bhttps://xmems.com/products/#cowell%7D%20(2024/05/15)`.

[42] XMEMS., *Cowell*. **url**: `%5Curl%7Bhttps://xmems.com/products/#cowell%7D%20(2024/05/15)`.

[43] USOUND, *Ganymede series*. **url**: `%5Curl%7Bhttps://usound.com/product/ganymede/%7D%20(2024/05/15)`.

[44] Flora, *Dragonfly*. **url**: `%5Curl%7Bhttps://flora.tech/dragonfly-piezo-speaker/%7D%20(2024/05/15)`.

[45] P. Mangaiyarkarasi **and** P. Lakshmi, "Design of piezoelectric energy harvester using intelligent optimization techniques," **in***2018 IEEE International Conference on Power Electronics, Drives and Energy Systems (PEDES)* 2018, **pages** 1–6. DOI: `10.1109/PEDES.2018.8707773`.

Appendix

Figure 39: On the left, impedance measurements of one speaker driver. On the right, impedance measurements of four speaker driver connected as a parallel connection between two sets of series connections. Yellow line shows the magnitude of the impedance and blue shows the phase.

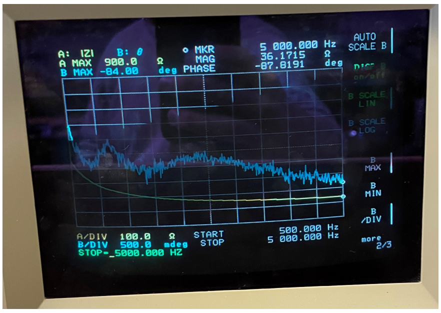


Figure 40: On the left, impedance measurements of two speaker drivers in parallel. On the right, impedance measurements of two speaker drivers in series. Yellow line shows the magnitude of the impedance and blue shows the phase.

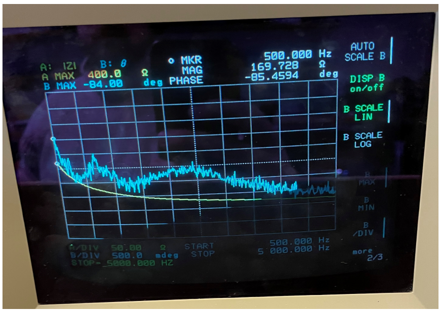


Figure 41: On the left, impedance measurements of three speaker drivers in parallel. On the right, impedance measurements of three speaker drivers in series. Yellow line shows the magnitude of the impedance and blue shows the phase.

Figure 42: On the left, impedance measurements of four speaker drivers in parallel. On the right, impedance measurements of four speaker drivers in series. Yellow line shows the magnitude of the impedance and blue shows the phase.

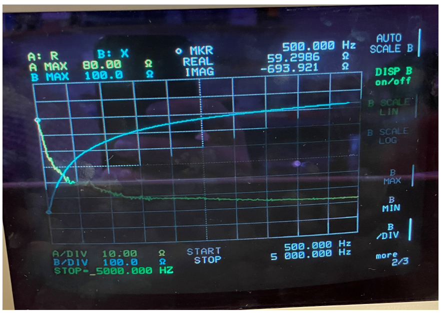


Figure 43: On the left, impedance measurements of one speaker driver. On the right, impedance measurements of four speaker drivers connected as a parallel connection between two sets of series connections. Yellow line shows the real part of the impedance and blue line shows the imaginary part of the impedance.

Figure 44: On the left, impedance measurements of two speaker drivers in parallel. On the right, impedance measurements of two speaker drivers in series. Yellow line shows the real part of the impedance and blue line shows the imaginary part of the impedance.

Figure 45: On the left, impedance measurements of three speaker drivers in parallel. On the right, impedance measurements of three speaker drivers in series. Yellow line shows the real part of the impedance and blue line shows the imaginary part of the impedance.

Figure 46: On the left, impedance measurements of four speaker drivers in parallel. On the right, impedance measurements of four speaker drivers in series. Yellow line shows the real part of the impedance and blue line shows the imaginary part of the impedance.

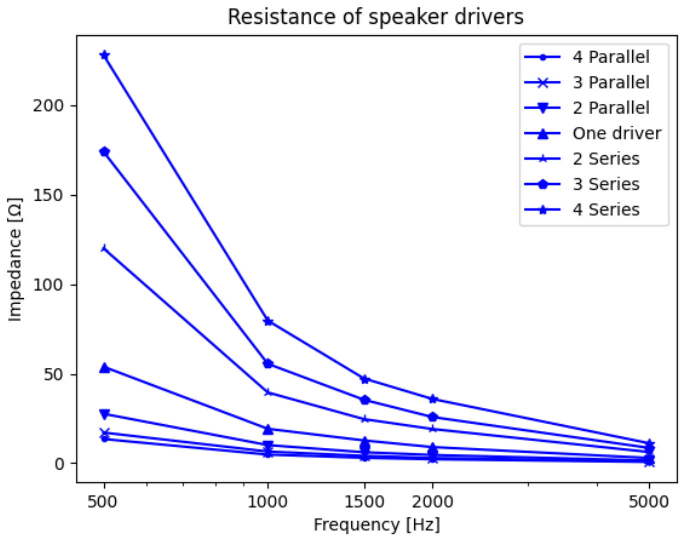


Figure 47: Resistance (real part of impedance) of 500Hz 1,1.5,2 and 5kHz. Data from measurements shown in figures 43, 44, 45 and 46

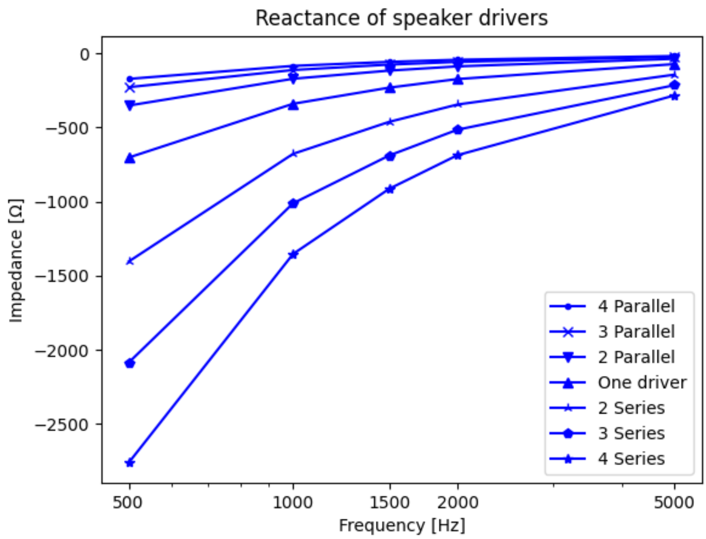


Figure 48: Reactance (imaginary part of impedance) of 500Hz 1,1.5,2 and 5kHz. Data from measurements shown in figures 43, 44, 45 and 46

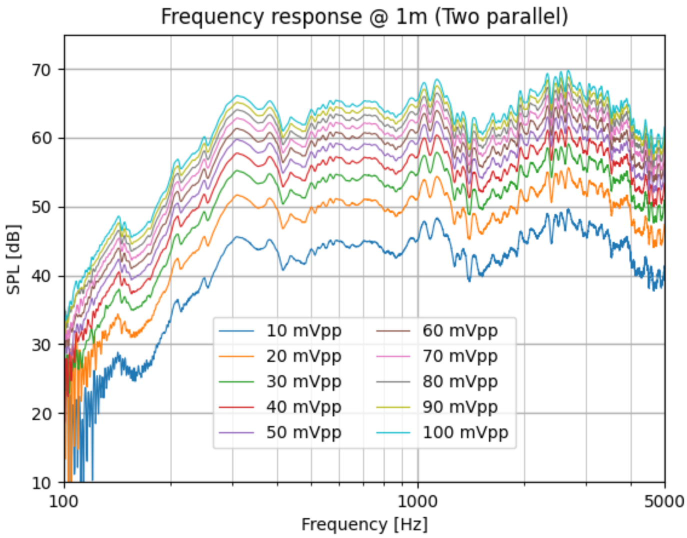


Figure 49: Frequency response for two parallel

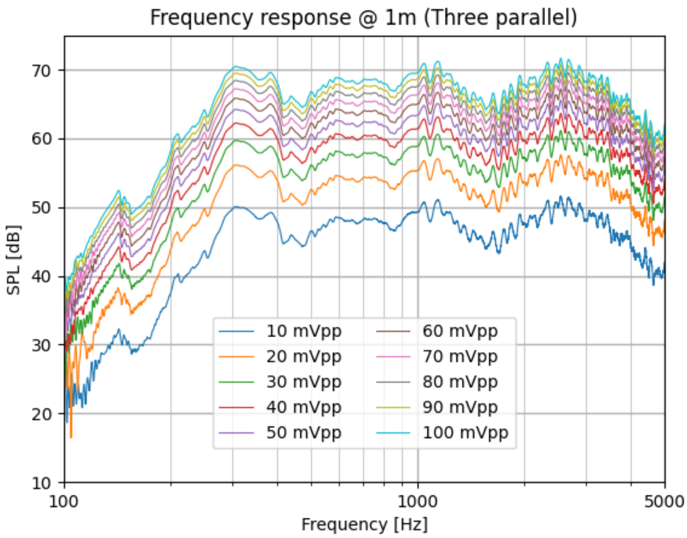


Figure 50: Frequency response for three parallel

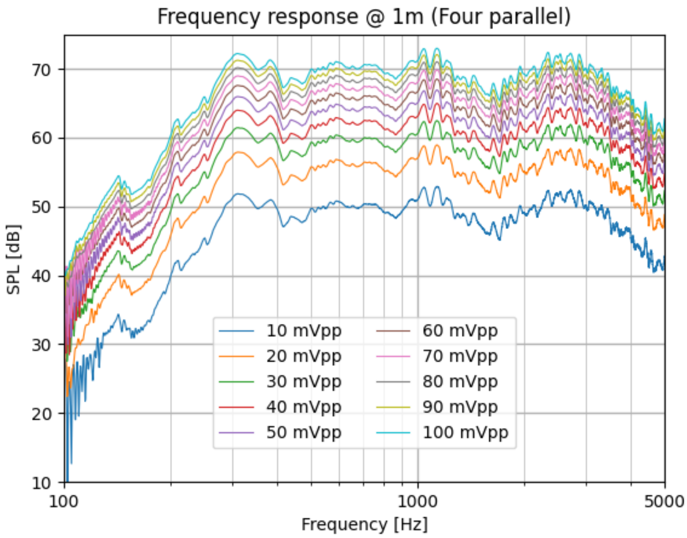


Figure 51: Frequency response for four parallel

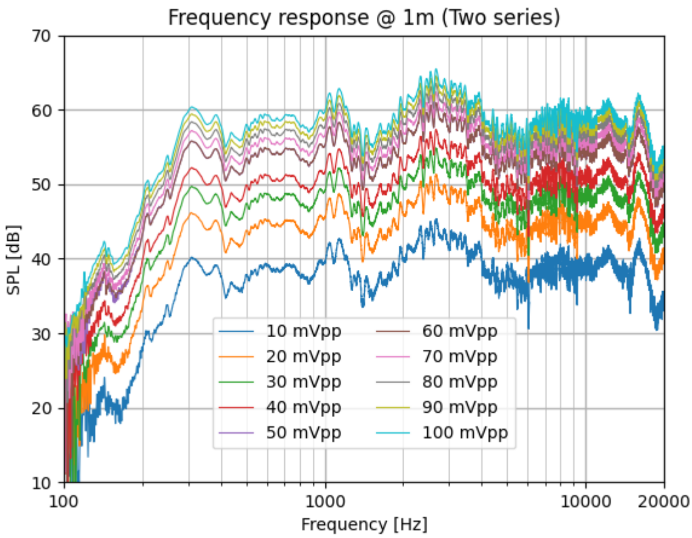


Figure 52: Frequency response for two Series

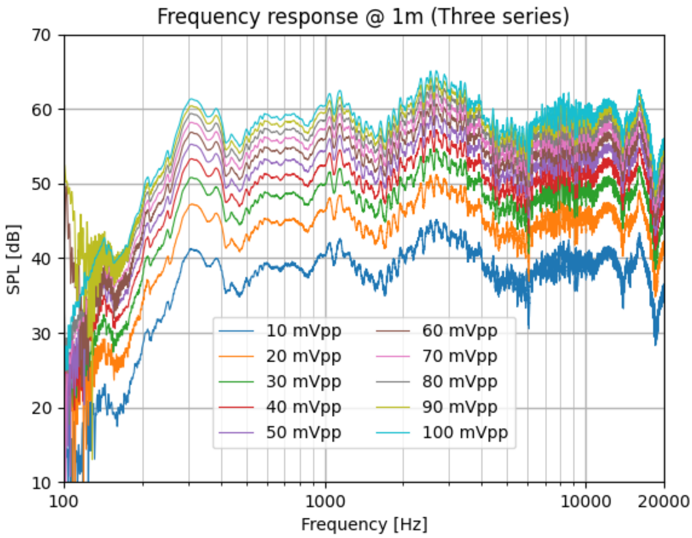


Figure 53: Frequency response for three Series

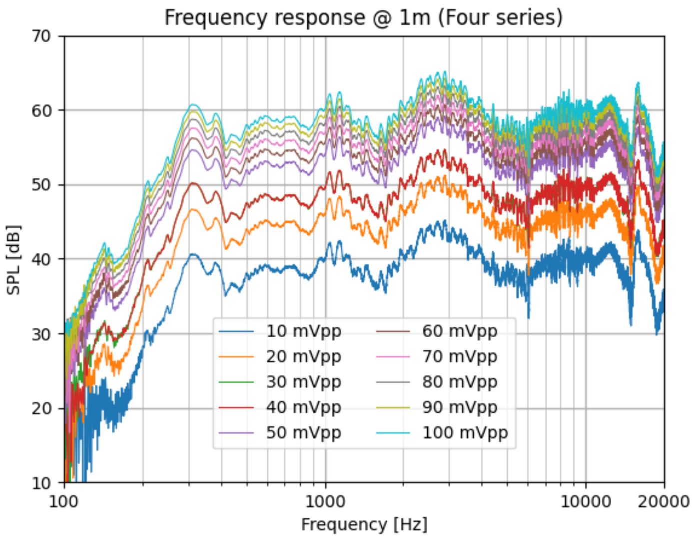


Figure 54: Frequency response for four Series

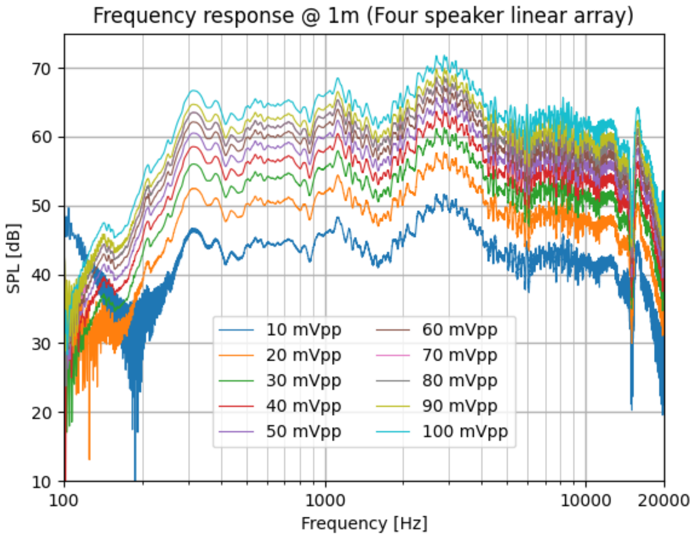


Figure 55: Frequency response for four speakers connected as shown in figure 17 in the line array.

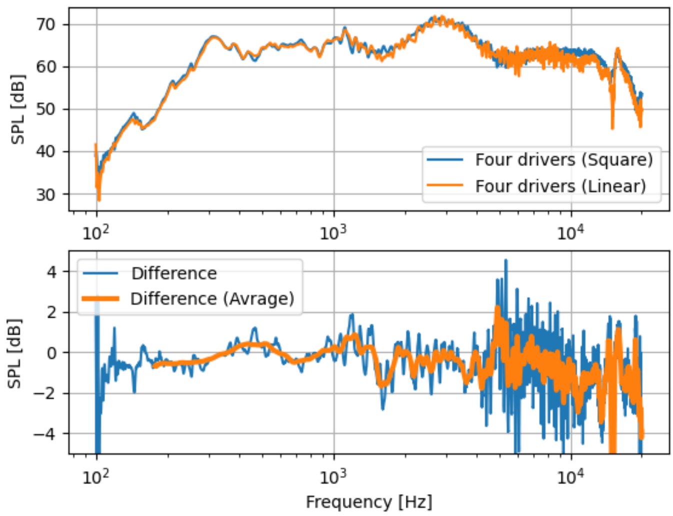


Figure 56: Comparison between linear and square array frequency response measured along the symmetry axis at 100 mVpp input signal.