Sound control in windy weather

Master Thesis

Jonas Buchholdt

Aalborg University
Electronic Engineering and IT

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Preface

This report is composed by Jonas Buchholdt during the 10th semester of Electronic Engineering and IT at Aalborg University. The general purpose of the report is *Signal Processing and Acoustics* .

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Aalborg University, May 1, 2019

Jonas Buchholdt
<Jbuchh13@student.aau.dk>

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Chapter 1

Introduction

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Part I

Problem Analysis and Requirements

Chapter 2

Analysis of sound propogation in outdoor venue

2.1 Live venue sound challenges

This section explores the challenges of producing sound in an outdoor environment. The challenge of producing a good sound experience for the audience highly depend on the calibration method and the atmosphere condition. It is well known that acoustically wave propagation is strongly affected by the inhomogeneous atmosphere doing the outdoor sound propagation. This inhomogeneous atmosphere shifts the calibration of the sound system which affects the intelligibility. In section 2.1.1 an overview of high Sound Pressure Level (SPL) Public Address System (PA) system is discussed.

2.1.1 Acoustics as live venue

An outdoor PA system is an essential sound reinforcement concept today. It is used to address information, music or just entertainment where the number of audiences is large, sometimes more than 10.000 audiences. The number of the audience makes it difficult to address the information to a large number of the audience without the reinforcement of the information. The reinforcement is nearly always done from a stage with a sizeable PA system and sometimes delay unit in the middle of the audience area. The stage lifts the artist while the PA system is designed to cover the audience area with sound. The optimal PA system covers the area with a linear frequency spectrum in the audible frequency range with a homogeneous SPL. Today, the used speaker is a line source array flown in both side of the stage and is therefore only close to the audience in front of the stage. The line source array is an array of small identically wide speakers attached to each other, to form a vertical line of speakers. An example of a line source array is shown in Figure 2.1

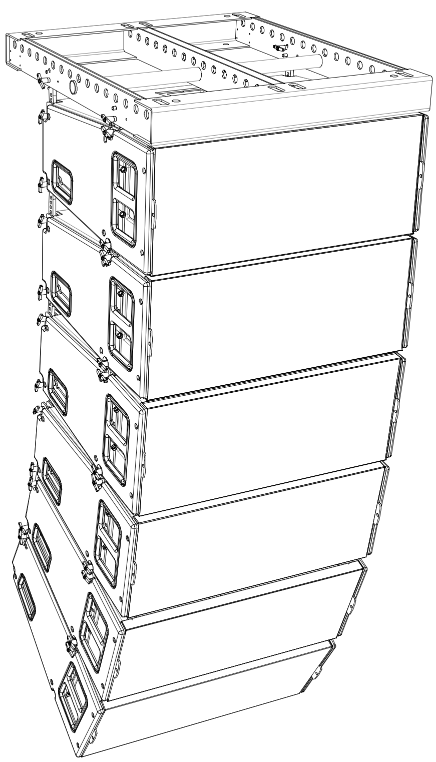


Figure 2.1: The figure shows an illustration of a KUDO line source array from L-Acoustics [?]

Every speaker or a small group of the line source array can be controlled individually, both in sound coverage area angle and SPL. The benefit of using the line source array design is that the coupling between the speaker makes a line acting source. With an optimised control system of the line source array, the audience area is covered with sound such that all audience can hear the information without damage the ear of the frontal audience. An optimised line source array has, for example, an optimised main lobe such that the lower part of the main lobe lays flat along the audience area. The following Figure 2.2 shows a graphical illustration of the outdoor PA venue concept.

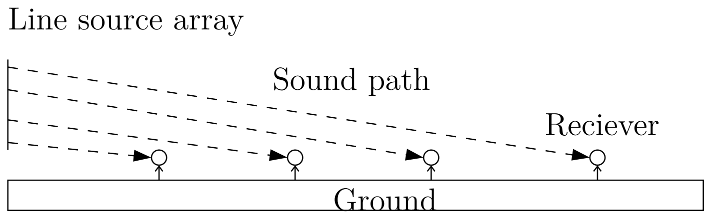


Figure 2.2: The figure illustrate the concept of outdoor PA venue

As shown in Figure 2.2, the distances from one element in the line source array to the receiving audience dependent on the audience position. The distance indicates that the signal to every line source element has to be set individually to cover the audience area wit ha homogeneous SPL. The individual control of the source is necessary because of the wave amplitude decay with distances. This phenomena is addressed in section 2.2. The adjustment is not as simple as just supply the upper speaker with more power. A sound wave is a mechanical movement of the particle in the air, which condensate and compression the air molecule, then low pressure and high pressure respectively. The movement of the molecule depends on the medium, and in this thesis, the medium is limited to air. The SPL is the pressure divination of the instantiates atmospheric pressure. The atmospheric pressure, therefore, set a lower bound on the condensation while very high pressure changes the speed of sound and distort the wave as it propagates. To ensuring that the information is communicated to the audience without distortion, the limitation is addressed in section 2.3.3. The medium in the air is not constant and varies over time regarding pressure, wind, humanity and temperature. The analysis starts with the experience for live concert of the author in section 2.1.2, next section 2.3 address the impact of homogeneous atmospheric effect on sound propagation. Then section 2.3 address the impact of inhomogeneous atmospheric effect on sound propagation.

2.1.2 Author experience of live concert

The Author of the thesis has experience with live concert both as an audience and as a sound engineer. The aspect of being the sound engineer and an audience to a live concert is very different. As a sound engineer, the area for controlling the sound is a secured area with a tent as protection. The tent roof often shadows for the high frequency, and the walls make standing waves of the low frequency because the distance between parallel tent walls fits with the wavelength for the low frequency. The sound engineer control area is defined as the Front Of House (FOH). The FOH is often equipped with an additional speaker, and the sound engineer does not fully know how it sounds outside the FOH, but base there mixes on experience ??. The aspect of being an audience depends on where the audience is regarding the stage. In close hand to the stage the SPL is high and often to high especially in the low frequency. The low frequency is often made as a vertical array at the ground or two end-fire arrays and shall be able to exhibit all audience by an audible low frequency spectrum typically from 25 Hz but one company extends down to 13 Hz. Therefore the SPL just in front of the subwoofer has a very high SPL. This position is not comfortable to be at in longer period, and the high SPL mask the higher frequency. The optimal audience position is in the centre of the stage and not as long from the stage as the delay towers. The average SPL is often less than 102 dB SPL since the sound engineer try to keep a maximum average SPL at 102 dB SPL just in front of the FOH. Moreover, it is the stereo sweet spot. This position is the only position where the stereo image is optimal. The stereo perspective problem is a hot topic nowadays, both L-Acoustics [**?**] and D&B Audiotechnik [**?**] have made there own solution to the problem. The idea is to fly many small line source array above the stage and assign every musician to there own line source array. The concept minimises the interference between two line source array playing the same mono signal. Near the delay towers or approximately 50 m from the main stage, the low frequency spectrum is still sharp and audible but something happens to the high frequency. Often the high frequency disappears for a few seconds and gets back. This phenomenon altering through the full concert. Behind the delay towers, the line source array in the delay tower reproduces the sound such that the audience in the back also gets the high frequency spectrum. The question is why does the high frequency disappear for a short period when the low frequency does not? This analysis focus on finding the atmospheric condition which cause the phenomena.

2.2 Ideal geometric spreading loss

When a line source generates a sound wave, the wave field exhibits two fundamental difference spatially directive regions, near-field and far-field. In near-field, the wave propagates as a cylindrical wave wherein the far-field the wave propagates as a spherical wave. When the wave propagates as a cylindrical wave, the wave propagates only in the horizontal plane, and therefore the attenuation is 3 dB SPL per doubling of distance. For a spherical wave propagation, the wave propagates in all direction. Therefore the attenuation is 6 dB SPL per doubling of distance. The near-field and far-field attenuation are based on non-absorption homogeneous atmospheric conditions. The border between the near-field and far-field depends on the hight of the array and the frequency. The distance can be calculated with Fresnel formula Equation 2.1, where the wavelength λ is approximated to $\frac{1}{3f}$ [?]

$$d_B = \frac{3}{2} f \cdot H^2 \sqrt{1 - \frac{1}{(3f \cdot H)}} \tag{2.1}$$

Where:

d_B	is is the distance from the array to the end of near field	[m]
f	is the frequency	[kHz]
H	is the hight of the array	[m]

In equation Equation 2.1 it can be calculated that less than 80 Hz radiate directly intro spherical wave on the exit of the speaker no matter the hight of the line source array. The following Figure 2.3 shows a horizontal cut of the near-field, far-field from a line source array.

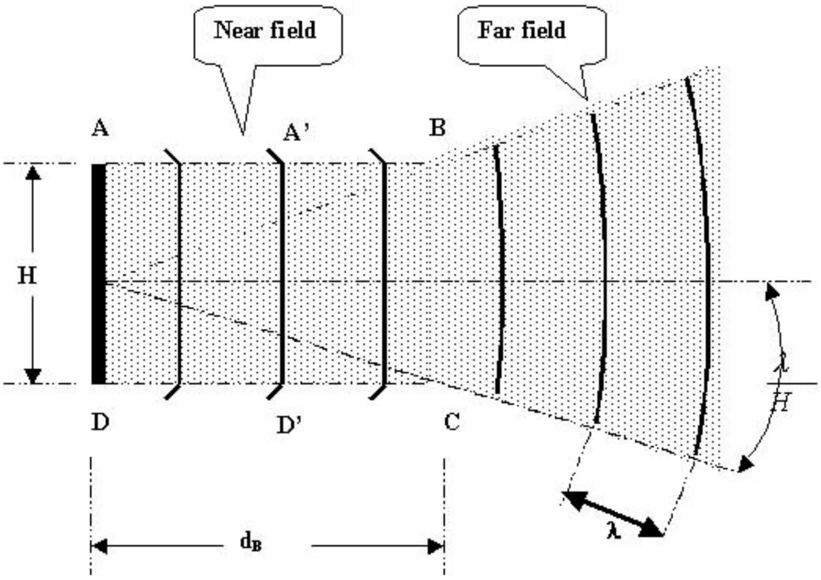


Figure 2.3: The figure shows horizontal cut of a SPL radiation pattern of a line source array [?].

As seen in Figure 2.3, the wave propagating as a plane cylindrical wave in the near-field, where the coverage area for every double of distance is twice as big. Since the coverage area is twice as big, the SPL is the half for the doubled distance. When the wave excites distance d_B, the wave propagates intro far-field where the coverage area is four times higher while travelling the double of distance and therefore the SPL is four times less. In far-field, the wave propagates as a spherical sound source

2.3 Homogeneous atmospheric conditions

This section aims to analyse the sound wave propagation in homogeneous atmospheric conditions. It is well known that the sound wave propagation is highly depending on the atmospheric conditions. The propagation depends on the atmospheric pressure, wind, temperature and humanity, where the two latter moreover is frequency dependent. The attenuation difference in frequency for temperature and humanity can be above 80 dB SPL [?]. The following sections introduce a brief discussion of homogeneous atmospheric conditions effect on sound propagation.

2.3.1 Humidity and temperature impact

The temperature and humidity have three impacts on wave propagation from a line source array, directionality of the speaker, the speed of sound and a lowpass effect. The following description starts with the latter.

Lowpass effect The effect of humidity and temperature on wave propagation act as a lowpass filter while the wave propagates. The low frequency remains without any additional attenuation where the high frequency highly depends on the atmospheric condition. In other words, attenuation in the high frequency range does not only depends on the spreading loss but also temperature and humanity. Therefore, for long distance, the atmospheric conditions have a high influence on the frequency spectrum delivered to the audience. Humanity and temperature attenuation are already well studied and standardised. Standard [**?**] gives an overview of calculating the SPL attenuation concerning the frequency, distance, temperature and humanity. The article [**?**] gives some examples of attenuation at a distance of 100 m. The article shows that if humanity increases proportionally to the temperature, the lowpass effect is small. If the change in temperature and humanity is the opposite of each other, for example, high temperature but dry, the attenuation in high frequency is significant. The following Figure 2.4 shows the worst-case scenario from [**?**].

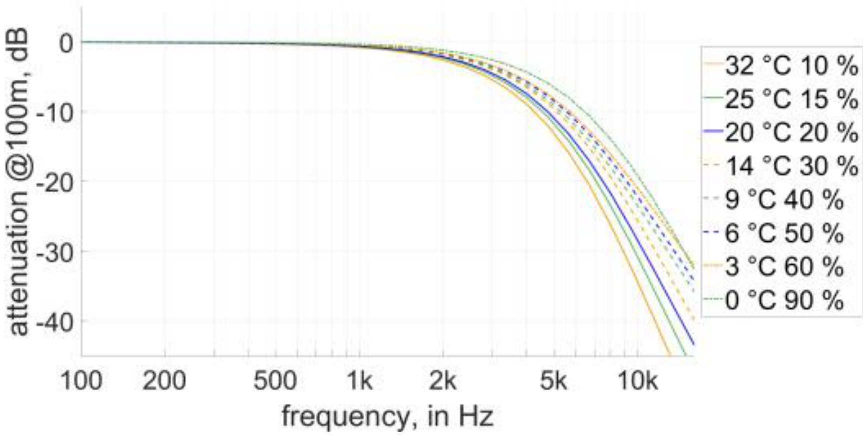


Figure 2.4: The graph shows the attenuation in dB with respect to frequency, humanity and temperature [**?**].

As shown in Figure 2.4 the attenuation in the high frequency is significant and excite 30 dB SPL within the audible frequency range. The attenuation is such markedly that applying more power does not cover the attenuation without an extreme high-pressure driver. That driver might be possible to design in theory but not in practice. Extreme high-pressure drivers introduce high distortion as is explained in section 2.3.3

Speed of sound The second consequence is the speed of sound. At temperature range from 0 °C to 40 °C the speed of sound with respect to humanity change is sparse and mostly only depend on temperature change. At 0 % humidity, the speed of sound increases with 0.6 m/s for every increasing degree °C. At humanity higher that 0 % the speed of sound increase with respect to humanity, depends on temperature. At

0 °C the speed of sound increases with approximately 0.8 m/s when the humidity raises from 0 % to 100 %. At 30 °C the speed of sound increases with approximately 2.7 m/s when the humidity raises from 0 % to 100 % [?] [?]. The wave propagation speed start at 331.5 m/s at 0 °C and 0 % humanity. The following Figure 2.5 shows the speed of sound with respect to humanity and temperature.

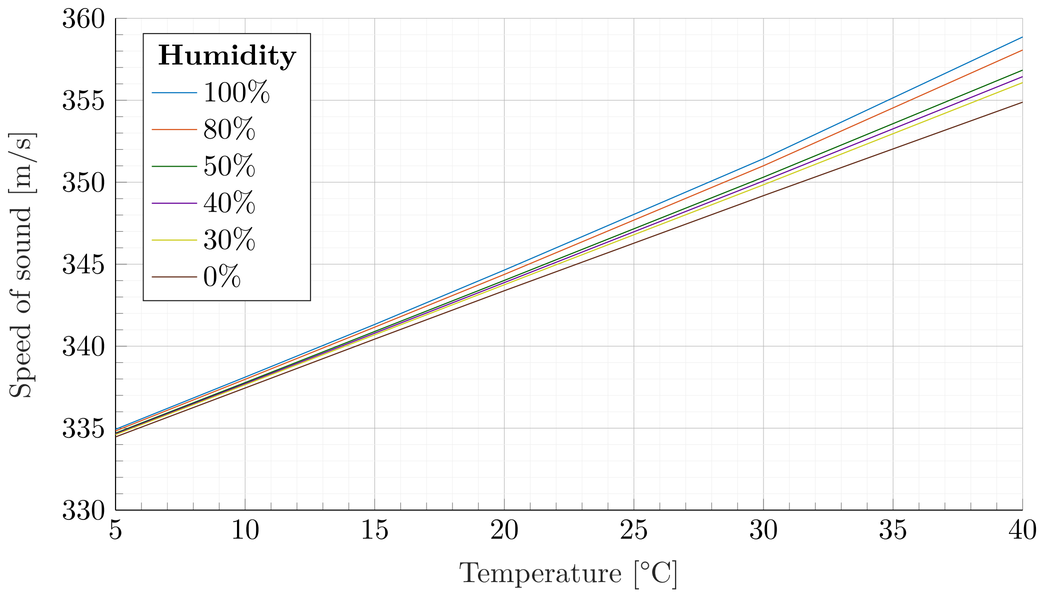


Figure 2.5: The figure shows the increase of sound speed with respect to humanity and temperature [?]

As seen in Figure 2.5, the effect of humidity is negligible compared to the effect of temperature changes, but as the temperature increases the humidity gets significant. At a temperature of 40 °C the speed of sound is changed 4 m/s from 0 % humidity to 100 %

Directivity The directivity of a line source array in the mid and high frequency is always controlled mechanically by a horn because the wavelength is short compared to the size of the speaker. At low frequency, the wavelength is too long to be controlled mechanically by a horn. Therefore the directional pattern is controlled via cancellation from a backwards pointing speaker. The directivity of both the low frequency and the high frequency driver sufferers from temperature increased. At the high frequency, the main lobe gets narrower when the mechanical horn gets warmer, and the effect is notable when the sun directly heats up the horn. When the surface of the horn heats up by the sun, the temperature can get much warmer in the horn that the air temperature. Therefore the surface of the horn affect the directivity of the high frequency by radiate warm air from the surface. The resend that main lope gets narrower is that the wavelength gets shorter at higher temperature [?]. The directivity of the low frequency is affected as in the high frequency with the temperature increase. The difference is not as significant as in the high frequency since there is no surface heat. The directivity is then not affected due to the sunlight, but only the temperature increasing and decreasing. As in the high frequency temperature differences change the wavelength, and then the length between the speaker in a cardioid low frequency does not match the optimised distance between the speaker more.

2.3.2 Wind impact

The wind influence is depending on the angle of the wind direction with respect to the direction of sound propagation. A homogeneous wind is a laminar wind flown with the same homogeneous speed. The following analysis assumes homogeneous laminar wind flow from one direction. The analysis is of both oblique wind and parallel wind with respect to the frontal direction of the line source array. The analysis starts with the latter.

Parallel to sound propagation When the wind flows in the same direction as the sound wave propagation, the wind flow in m/s is an addition to the speed of sound. When the wind flows in the opposite direction, it is a negative addition. In other cases, the influence is complicated since the wind deflect the sound waves.

oblique- and crosswind The effect of homogeneous oblique- and crosswind on sound propagation from a speaker is rarely studied, and the effect on high frequency seems to be unclear. One author has addressed the problem in a simulation of a low frequency source [?] where the author of [?] have practical experience with high power sound system and indicate that crosswind effect might be frequency dependent. The author indicates that the frequency dependency might be due to the directionality of the high frequency drivers. The author of [?] has simulated a homogeneous crosswind effect on an omnidirectional source at 100 Hz. The author of [?] implemented a ray tracing method with a vector based interpolation as shown in Figure 2.6.

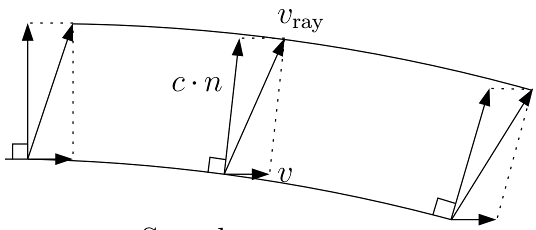


Figure 2.6: The figure shows a geometrical ray tracing calculation scheme of calculate the resulting wave direction at crosswind [**?**], [**?**]

Where:

c	is the speed of sound	[m/s]
n	is the normal unit vector	[m]
v	is the speed of wind	[m s]
v_{ray}	is the resulting sound ray	[m]

As seen in Figure 2.6, the ray vector v_{ray} is an addition of the sound speed vector $c \cdot n$ and the speed of wind v. The wave speed and wavelength, therefore, depend on the speed of the wind and the angle between the wind and the sound propagation. The following Equation 2.2 calculate the speed of sound in the v_{ray} direction with respect to the wind speed and angle.

$$c_r = c + ||v||_2 \cdot sin(\theta) = ||c \cdot n + v||_2 = ||v_{\mathrm{ray}}||_2 \tag{2.2}$$

Where:

θ	is the angle of the wave with respect to the wind	[°]
c_r	is the resulting speed of sound	[m/s]

As the wave propagating, the resulting v_{ray} increases in the direction of the wind. The article [**?**] simulates the effect of crosswind in a Finite-Difference Time-Domain (FDTD) simulation with a wind speed of 102.9 m/s. For the acceptable condition to a concert, the wind speed is less than 20 m/s. Otherwise, the audience is escorted from the stage to the exit, and the speaker system is taken down to ensure safety. The following Figure 2.7 shows a simulation result from [**?**], where the source is an omnidirectional 100 Hz spherical source while the wind has a constant uniform wind speed from left. The simulation is done in two dimensions.

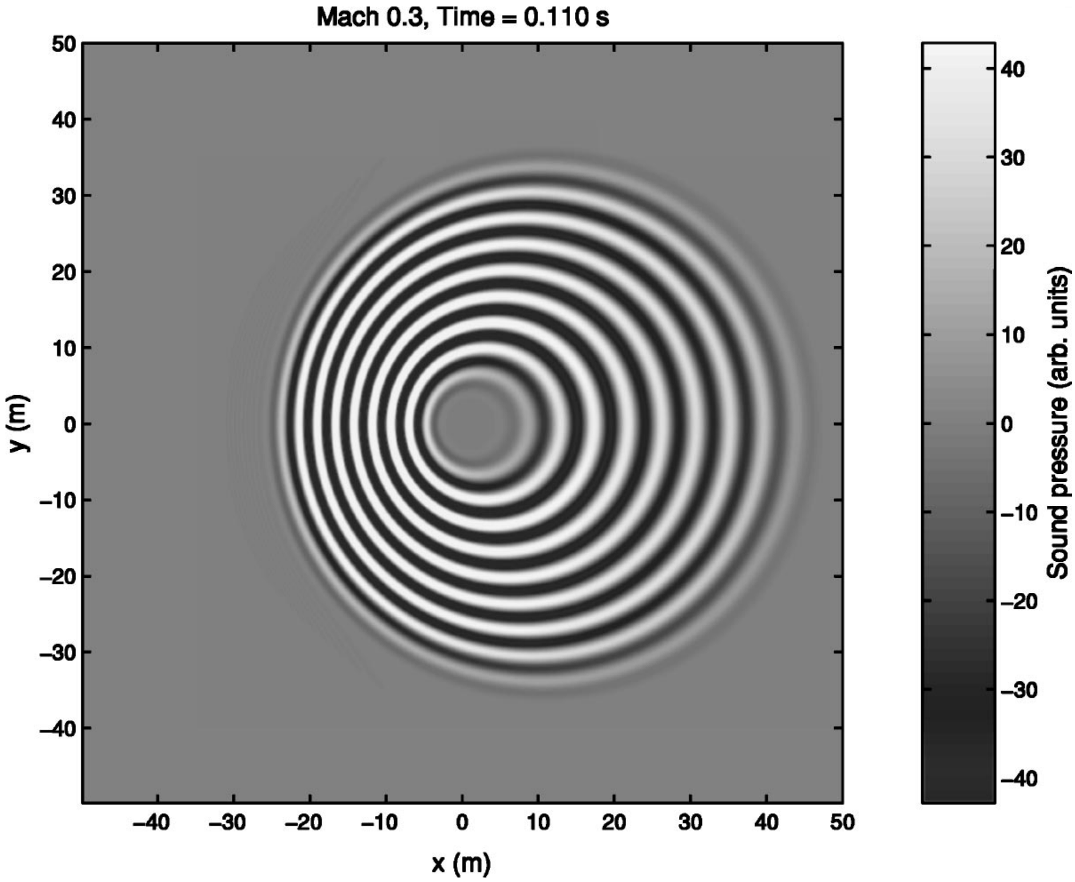


Figure 2.7: The figure shows a simulation of a 100 Hz omnidirectional source with a uniform constant wind speed from left with speed of 102.9 m/s [?].

As seen in Figure 2.7, the homogeneous crosswind does not affect the direction of the wave from a low frequency spherical source. It only affects the time of arrival to the audience.

2.3.3 Pressure impact

The influence of atmospheric pressure change is low compared to the effect of wind, humanity and temperature. The average atmospherical absorption from 4.0 kHz to 16.0 kHz with fixed temperature and variable humanity, increases with 2 dB SPL while going from 101.33 kPa to 54.02 kPa. The atmospheric pressure then only have a negligibility influence on sound propagation and is generally not frequency dependent.

Beside the small impact of pressure difference in the atmosphere, the high pressure generated by the speaker does have a tremendous influence on the sound propagation. There are three states in the propagation way that can produce distortion concerning the pressure. The design of the high frequency horn [?], the port design of the low frequency driver [?] and the influence of the sound path. The following description starts with the latte.

Sound path In the sound path, two factors distort the wave doing propagating in air. As described in section 2.1.1 a sound wave is condensation and compresses of the air particle. The air medium, therefore, has a lower limit that cannot be less than vacuum. The higher bound of SPL is then depending on the atmospheric pressure. As an example, at 54.02 kPa the highest SPL before distortion caused be vacuum is 188.6 dB SPL and at 101.33 kPa the highest SPL before distortion caused be vacuum is 194.1 dB SPL.

There is, therefore, a higher limit determined by the atmospheric pressure to vacuum, but distortion occurs much before the limit of vacuum. High pressure in the compression also distorts the sound because of the lack of linear dependency between the particle velocity and stiffness in the sound wave. The stiffness or density increases while the air particle is closer to each other. Therefore SPL increases more than the density of the sound wave which causing the compression of the sound wave to be stiffer and therefore propagate faster than in the condensation of the wave. This speed differences, therefore, produce harmonic distortion, and is even present in SPL less than 120 dB SPL [?]. The speed differences transform the sinusoid intro a sawtooth as it propagates which transfer energy to the harmonic of the propagation frequency. The distortion is not only SPL dependent, but also depend on the frequency. The higher the frequency is, the faster the sinusoid transformers intro a sawtooth, therefore, the distortion increases with frequency for constant SPL. The harmonic frequency is higher than the fundamental frequency and therefore, as explained in section 2.3.1, the harmonic has higher attunation with respect to the distance and viscosity. In most cases, the attention is not as high as the increase of the harmonic distortion, and therefore the distortion of the wave propagation is not fully compensated by the viscous losses in the air. [?]. The distortion made by air propagation is much less than the distortion in the mouth of the speaker which leads to the next distortion problem produced by high-pressure [?].

Driver throat and mouth design High pressure in both horn phase plug, sealed enclosures, vented enclosures and reflex enclosures for low frequency driver cabinet produces distortion as they act as nonlinear components. The latter produce distortion because high pressure makes air turbulence in the vent. In the optimal design, the distortion of air turbulent is low but is always present in high-pressure [?]. The air turbulence is not only caused in the vent of the low frequency driver, but it also occurs in the phase plug of the compression driver if the SPL is high [?]. The distortion depends on the moving mass, the stiffness and the viscous losses in the air on the diagram displacement and the SPL. As the air in the high frequency driver compress, it becomes heavier, stiffer and thicker which make nonlinear wave propagation. It typically occurs when the compression chamber exceeds approximately 170 dB SPL. At a higher level, the particle velocity resistance to the air flow increases and the laminar air flow turns intro turbulent air flow. The distortion is also depending on the length of the horn and the expansion rate of the horn flare. To keep the distortion as low as possible for the high frequency driver the displacement of the diaphragm should be kept significantly lower than the hight of the compression chamber [?]. Therefore, to keep the displacement of the high frequency driver as low as possible, the frequency range should be limited.

2.3.4 Ground absorption and reflection

In a concert area, ground absorption and reflection is complicated because there are two very different situations. Before the concert, the area is a local plan area often with mown grass and with ground reflection. An example of a frequency response over mown grass where the measuring hight of the microphone is in the hight of the ear is given in [?]. The measurement shows that the ground reflection affects the frequency response with high interference. A measurement in Appendix A and ?? is performed where the ground reflection clearly have a big influence on the received frequency response. In this measurement inhomogeneous airflow is present, but the interference is similar in homogenous airflow [?]. Doing the concert the interesting part is not such ground reflection effect but the audience reflection or absorption. The area along the concert is packed by the audience and therefore, the reflection is not easy to calculate. The absorption and reflection in an outside concert area with a group of audience is rarely studied, but absorption for the audience inside a concert hall is highly studied [?]. The absorption of the audience is founded to be high in all measured concert hall from 1.0 kHz octave band to 4.0 kHz octave band [?]. The average absorption a_{sabine} coefficient is calculated to be above 0.80. The method and result can be founded in [?]. The reflection in the high frequency in the audience area doing concert is therefore assumed to be low. At low frequency, the article [?] indicate that the absorption decay with frequency beneath 250 Hz, but The octave band for low frequency driver, which is 31.5 Hz, is not measured by [?]. The low frequency absorption at 31.5 Hz octave band is therefore assumed to low. The low frequency driver is mostly located in front of the stage on a line or in end-fire settings, often with a maximum distance of half the wavelength from acoustical centre to acoustical centre. The distance between the low frequency driver is determined by the half wavelength of the highest frequency, such that they radiate a plan wave [?]. A higher distance between the acoustical centre causes interference in the low frequency in the audience area.

2.4 Inhomogeneous atmospheric conditions

This section aims to analyse the sound wave propagation in inhomogeneous atmospheric conditions. In an inhomogeneous atmosphere, the pressure and speed is a function of position. By this fact, the modelling of a sound wave is very complex and depend on various variables such as temperature and wind speed. The following sections give a short introduction to the effect of inhomogeneous atmospheric conditions.

2.4.1 Atmospheric refraction

When the wind speed, the temperature and humanity is assumed to be homogeneous in the sound field, the sound is travelling in a straight plan wave. Often this is not true, the wind speed increases logarithmically with the hight from the ground to the geostrophic wind [?] in the free troposphere [?], and the temperature and humanity are inhomogeneous. The geostrophic wind in the free troposphere is located in a hight from approximately 1 km above the ground [?], [?]. The inhomogeneous atmospheric condition makes the speed of sound to depend on the hight from the ground. This inhomogeneous atmospheric condition results in a curved sound path and is defined as atmospheric refraction. For small distances, the atmospheric refraction has a spars effect on the sound travelling path, because the speed of sound is much faster than the speed of the wind and the temperature change. Generally distance up to 50 m is often assumed to have no significant refraction effect [?]. For distances larger than 50 m the refraction is assumed to have a significant influence, especially when the sound source and the receiver are close to the ground. Refraction is frequency and distance dependent and is measured in dB excess attenuation. The means of excess attenuation is that only the effect of wind or temperature is considered, all other atmospherical effect is excluded. A measurement is given in [?] for a point source where the wind speed is 5 m/s. At a distance of 110 m, it is observed that frequency above 400 Hz is refracting where frequency below is rarely effected of refraction. Moreover, at a distance of 615 m the refraction is present in the full measured frequency range from 50 Hz to 3.2 kHz. In the perspective of a live concert the interesting distance is the 110 m from the line source array to the audience rather than the 615 m. Both the downwards and upwards refraction is interesting. In the upwards refraction the audience might be in the shadow zone where for the downwards refraction the high frequency reflection from the ground is assumed to be low when the concert area is full of audience. Therefore the high frequency is refracted down intro the frontal audience, and only sparse reflection of the high frequency propagate to the back part of the audience. The following Figure 2.8 display the phenomena of upwards refraction.

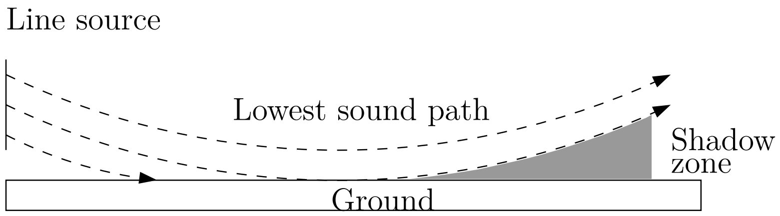


Figure 2.8: The figure illustrates that the shadow zone occurs from an upwards refraction. A line source speaker array contains of many couplet point sources. Every lowest sound path dashed line indicate the lower directional angle of one point source in the line source array.

The following description is based on the distance of 110 m and upwards refraction. As explained in [**?**] the refraction at a distance of 110 m is highly frequency dependent. At a frequency below 400 Hz the effect is sparse but above the effect is high and may result in 20 dB SPL attenuation at the audience. The reason that the refraction is frequency dependent is that the scale of the wind gradient and temperature gradient close to the ground is small compared to the wavelength of the low frequency [**?**]. This theory does not follow the shell's law of refraction. Shell's law describes the refraction as a layer change in the medium of propagation. Shell's law of refraction is defined as Equation 2.3

$$\frac{cos(a_1)}{c_1} = \frac{cos(a_2)}{c_2} \tag{2.3}$$

Where:

a_1	is the input angle in the horizontal plan	[°]
c_1	is the sound of speed in the medium of arrival	[m/s]
a_2	is the output angle in the horizontal plan	[°]
c_2	is the sound of speed in the medium of destination	[m/s]

As shown in shell's law Equation 2.3 the frequency dependency is not a factor and are therefore maybe only valid for a laminar wind flow profile. The article [**?**] only explorer frequency up to 3.2 kHz but since the refraction depends on the wavelength, the distance of refraction wave might be smaller for higher frequency. The attenuation with respect to refraction seems to have a saddle attenuation at 20 dB SPL. A measurement in [**?**] shows the attenuation for the center frequency of 1.2 kHz with 1/3 octave band filtered airplain noise over mown grass. The measurement is interesting with respect to a concert area and is therefore shown in Figure 2.9

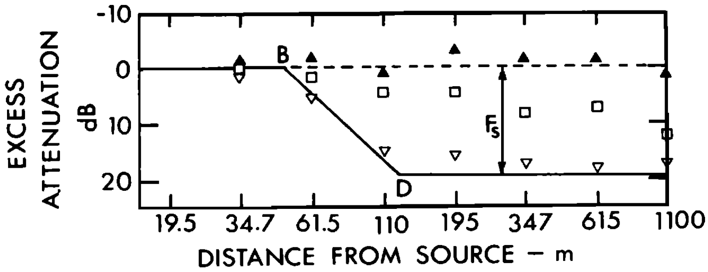


Figure 2.9: Excess attenuation measured for aircraft noise in the 1.2 kHz 1/3 octave band for the ground-to-ground configuration. The vector component of the wind velocity in the direction of propagation for ▲ is 5 m/s, □ is 0 m/s, and ▽ is −5 m/s. The temperature profile is neutral. F_s is the shielding factor, B is the shadow boundary [**?**]

The following two paragraphs explain the difference between wind refraction and temperature refraction.

Temperature Temperature decreases with respect to the hight at day time and increases at the night time. The increase or decrease is usually approximated as a logarithmic function. In the day time, the sun heats the ground even on a cloudy day, and the concert area is full of audience. Therefore, the eath and audience radiate warm air, which makes the temperature at a low hight warmer than the temperature at higher hight. These phenomena are named lapse where the opposite is defined as inversion. As explained in section 2.3.1, the speed of sound depends on the temperature. Therefore, at day time, the speed of sound in this situation decay with respect to hight. The speed change can be modelled as a change of layer for a plane wave. The output angle of the layer change follows the shell's law when the frequency dependency is excluded. Therefore when the temperature profile is logarithmic, the layer change is a function of hight and change the wave direction. The wave direction of the descript weather condition results in an upwards refraction. Since the temperature is a scalar quantity uniformly over a large area and a function of hight, an identical temperature profile is applicable all around an omnidirectional sound source. Therefore the upwards refraction is uniform all along the speaker in the horizontal plane. The following Figure 2.10 illustrate the phenomena where the temperature decay with respect to the hight and the line source array is omnidirectional in the horizontal plane. The omnidirectionality of the line source array is only present in the low frequency typically below 200 Hz.

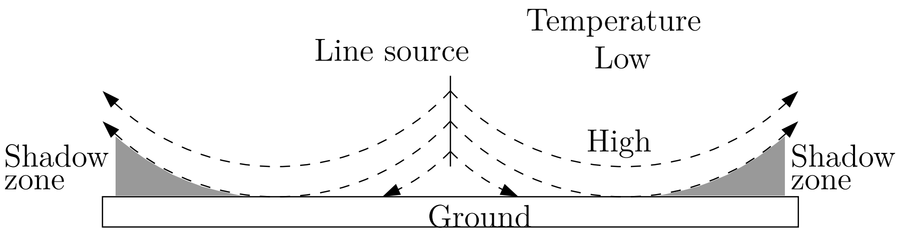


Figure 2.10: Wave refraction of a horizontal omnidirectional line source array in inhomogeneous temperature with lapse profile

When the temperature profile is reversed, the refraction is downwards.

Wind With respect to the wind speed, a concert area is often a protected area with for example barrier, stage and building. This blockage and the ground friction slows down the wind speed near the ground and cause turbulence. Moreover, from nature itself, the wind speed is often logarithmically increased with respect to the hight. When the wave is propagation in the same direction as the wind, the atmospheric refraction refracts the sound wave downwards. When the wave propagates against the wind, the atmospheric refraction refracts the sound wave upwards. The following Figure 2.11 illustrate the phenomena with a logarithmic increasing wind from left, and the line source array is omnidirectional in the horizontal plane.

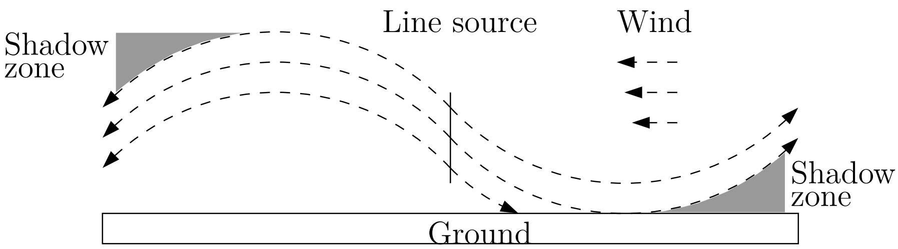


Figure 2.11: Wave refraction of a horizontal omnidirectional line source array in inhomogeneous logarithmically increasing wind profile where the wind gradient points towards left

As shown in Figure 2.8 the refraction is upwards when the wind flows in the opposite direction as the wave propagation. Behind the line array source, the refraction is downwards and is therefore different than for temperature refraction. The refraction of wind is the most dominant at a distance of 110 m. The following Figure 2.12 shows an excess attenuation plot of both inhomogeneous wind and lapse temperature profile.

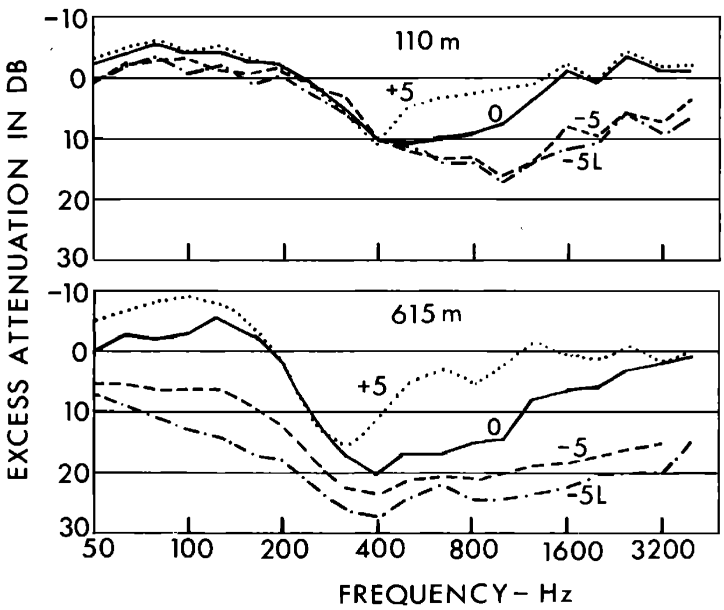


Figure 2.12: Observed attenuation of aircraft noise in a ground-to-ground configuration under a variety of weather conditions. Calculated losses from atmospheric absorption and spherical spreading have been subtracted from the attenuation measured in 1/3 octave bands for distances of 110 m and 615 m. The numbers on the curves indicate the vector component of the wind velocity in the direction of propagation in m/s. All curves are for neutral conditions of temperature except for those marked L, which are for lapse. [?]

It can be seen in Figure 2.12 that the refraction effect at a distance of 110 m starts at 400 Hz. The reason that sound enters the shadow zone is not fully understood, but one theory is that the shadow boundary wave is diffuse and therefore a significant amount of sound energy enters the shadow zone by turbulent air flow. In a non-turbulent atmosphere condition the SPL inside the shadow zone is attenuated well more than 30 dB SPL. Close to the ground, the atmosphere condition is always turbulent because of ground friction. The turbulence wind diffuses the sound wave and changes the direction of propagation. The wave that enters the shadow zone is considered as a creeping wave while turbulent air flow is present. The creeping wave will by them self also be refracted and therefore parallel to the other refraction waves. [?]

Oblique- and crosswind The effect of oblique- and crosswind on acoustical wave propagation in inhomogeneous atmospheric conditions are rarely studied. The author in [?] explain that the refraction is directly zero when only crosswind is present, and increase progressively as the direction of propagation deviate from the angle of crosswind. The author of [?] support this theory for inhomogeneous atmospheric condition.

Since the effect of oblique wind on a line source array speaker is rarely studied, a measurement in windy condition is performed. The measurement is performed over mown grass in a large open area used for football. The used measurement technique is done according to [**?**] where more than one impulse response is measured, and average by alining the impulse response. The wind was considered as healthy for an outdoor concert. The wind speed was measured to 14 m/s doing the full measurement. The measurement was done with a four element line source array 1.1 m above the ground. There were used two microphones, where both were situated 25 m from the speaker in the first measurements and 23 m from the speaker in the last measurements. While changing the distance, the angle to the speaker was changed. The frontal direction of the speaker was placed orthogonal to the wind direction, and the microphone was placed on both side of the speaker as shown in Figure 2.13

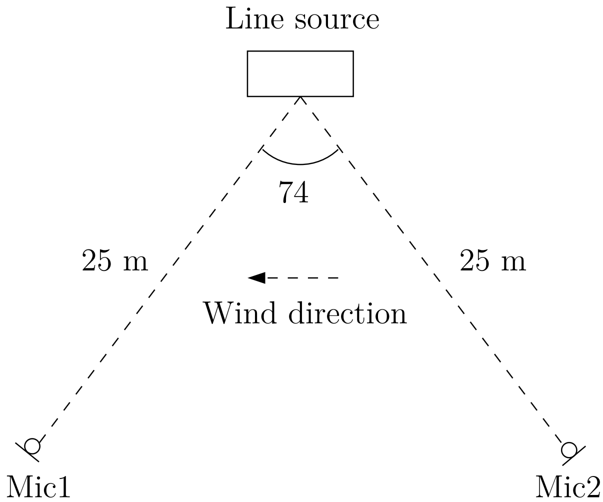


Figure 2.13: The figure shows the microphone position versus the position of the line source array angle of main lobe

The measurement was done with sine swept and according to the description in Appendix A. The measurement was performed with two microphone positions, two measurements where the microphone are within the speaker high frequency directional angle and three outside the speaker high frequency directional angle. The first measurement is shown in Figure 2.14. The other four measurement result can be seen in Appendix A. They show the same tendency, but the difference between the measurements are more drastically in the measurement where the microphone are situated outside the high frequency directional angle.

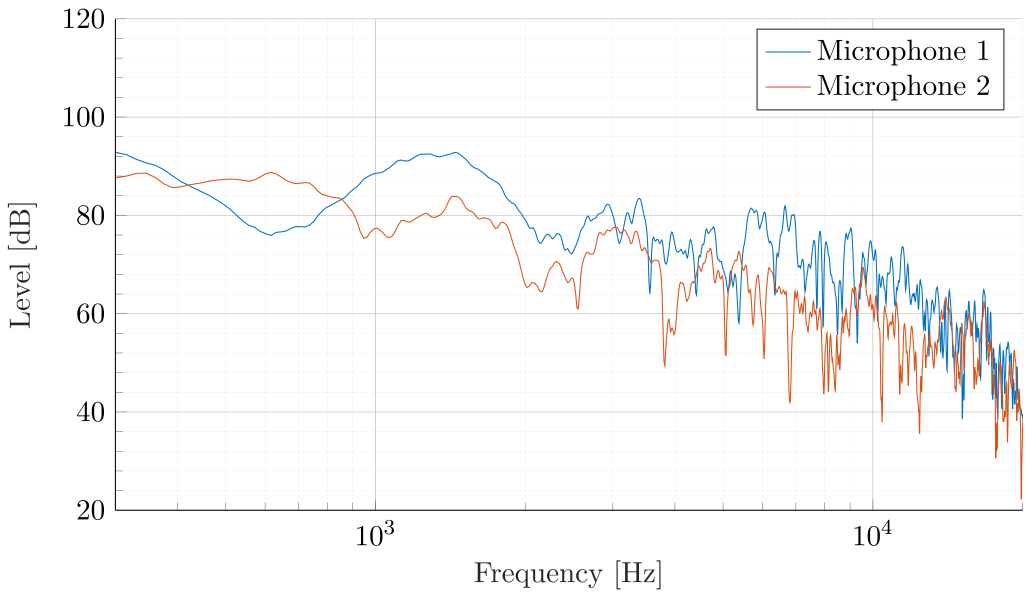


Figure 2.14: The graph shows the first transfer function measurement within the high frequency directional angle. The $L_{\text{eq},5}$ SPL different between the microphones is 4.41 dB SPL (IR_3)

It can be seen in Figure 2.14 that the general SPL is higher for microphone 1. Furthermore, microphone 1 also shows the typical downwards refraction ground reflection interference in the frequency response which is very similar to the calculated ground reflection interference in [**?**]. Microphone 2 does not have the same strong interference in the low frequency and the general SPL is lower than microphone 1. This difference indicates upwards refraction in the direction of microphone 2 with only a small amount of ground reflection. The resulting $L_{\text{eq},5}$ SPL difference for all measurement is shown in Table 2.1.

Table 2.1: The table shows the measured $L_{\text{eq},5}$ SPL for all measurement and the difference between the microphone

Measurement number	Mic 1 $L_{\text{eq},5}$	Mic 2 $L_{\text{eq},5}$	Difference
Measurement 1 Figure A.4	71.82 dB SPL	66.33 dB SPL	5.49 dB SPL
Measurement 2 Figure A.5	69.09 dB SPL	64.69 dB SPL	4.40 dB SPL
Measurement 3 Figure A.6	67.67 dB SPL	63.44 dB SPL	4.23 dB SPL
Measurement 4 Figure A.7	68.10 dB SPL	63.69 dB SPL	4.41 dB SPL
Measurement 5 Figure A.8	68.44 dB SPL	63.62 dB SPL	4.81 dB SPL
Average	69.02 dB SPL	64.35 dB SPL	4.67 dB SPL

As it is shown in Table 2.1, the $L_{\text{eq},5}$ SPL is higher for microphone 1 in all measurement. Moreover the average $L_{\text{eq},5}$ SPL difference is 4.67 dB SPL while for A-weighted $L_{\text{Aeq},5}$ SPL the average difference is 6.17 dB SPL.

With respect to the intelligibility frequency range, a weighting filter is designed to observe the SPL differences in the critical intelligibility frequency range. The filter is based on the founded intelligibility frequency range in [**?**]. It is shown in [**?**] that the critical intelligibility frequency range lays between 1.0 kHz and 4.0 kHz. The designed intelligibility weighting filter is an 8^{th} order bandpass filter with lower crossover frequency at 1.0 kHz and higher crossover frequency at 4.0 kHz. The resulting average difference is 7.88 dB SPL and the maximum difference is 9.95 dB SPL.

Turbulent Turbulence is an atmospheric condition where the wind eddies. It often starts with large eddies and progressively brakes down like a cascade effect to smaller and smaller eddies which only depend on the local region. When the eddies are as small as 1 mm the energy disappears in viscosity and thermal conduction. A statistical distribution of the eddies is defined as turbulence. The turbulence wind flow is, therefore, a chaotic and stochastic process by nature and is present all the time. It can occur because of change in landscape, stage and blockage, but can also be a process of flow speed increase in the wind, which make the wind to refract on itself. Turbulence is often high on a windy afternoon day and low under the inverse of lapse. Turbulence also often occurs near the ground because the ground surface slows down the speed of wind by the friction to the ground. The effect of turbulence on sound is known to make phase and amplitude fluctuation of pure tone. The fluctuation increases with distance until the standard divination of the phase fluctuation is comparable to 90° [**?**]. At this point the phase correlation for each sound path is uncorrelated

2.5 sound pressure level doing a concert

In Denmark there is no law limiting the SPL doing a concert. The only restriction there might be of SPL is area dependent. In a city the local komunity has limited the total SPL average over 15 min of any event. Out on the countryside, the sound engineer can decide by himself and the often used limit is A-weighted 102 dB SPL average over 15 min.

The standard ?? for long term exposior of high SPL limits the SPL for A-weighted 94 dB SPL average over maximum of 1 h, then the ear needs to have a brake to ensure no damage the the hering. A concert i often more than 1 h with A-weighted 102 dB SPL average. This is at least 8 dB SPL A-waighed more than the regulation recommend. It shall here be clearly understanded that the SPL measurement is done in the FOH and the actian exposed SPL is higher for the audience close the the stage.

Chapter 3

Summary of Problem Analysis

The analysis started addressing the generally used method for a live concert. It is founded that live concert today use line source array system to cover the audience area with sound. The line source array is flown above the audience at the main stage, and the delay speaker covers the back audience at a large concert. The line source array is constructed of many identical speakers attached to each other in a vertical line. Moreover, the distance from the speaker to the individual audience depends on the audience position. The analysis founded that a homogeneous SPL among all audience might not possible but the SPL among all audience can be optimised by knowledge of the condition of the atmosphere and gain up for the spreading lose. The author observes that the wind does have a frequency and distance-dependent effect on sound propagation, for example at high frequency the high frequency attenuate audibly in the crosswind. The high frequency blows away for periods and comes back again. The analysis of sound from a line source array started by the ideal geometric spreading loss. Here it is founded that the sound propagation of the line source array highly depends on the hight of the source. The line source array propagates differently with respect to frequency. At a certain hight of the line source array the propagation is a cylindrical propagation until a certain distance from the source where it starts propagating as a spherical source. In the cylindrical propagation, the sound field is defined as near-field while in the spherical propagation the sound field is defined as far-field. In the non-ideal scenario, the line source array propagates in inhomogeneous atmospherical condition. To cover the inhomogeneous atmospherical condition, the local homogeneous atmospherical condition is analysed. In the homogeneous atmospherical condition, it is founded that the temperature, humidity, pressure and wind influence the sound field. The effect of temperature and humidity is close coupled on sound propagation. When the temperature is high, and the humidity is low the air has a significant high frequency absorption whereas when the temperature and humidity follow each other, the absorption is less. The second effect the temperature and humidity have on sound propagation is the speed of sound. The higher the temperature is, the higher the sound of speed. The humidity affects the speed of sound the same way as the temparature, but the increase is negligible compared to the temperature. The effect of wind seems to have a sparse effect on the sound propagation when and only when the wind is homogeneous. It is founded that the speed of wind affects the speed of sound. If the wind moves in the direction of the sound propagation the wind speed is an addition to the speed of sound. In the opposite wind case, the speed of sound is lowered. In the case of oblique- or cross-wind, the effect seems to be unclear for high frequencies. One author has simulated a low frequency spherical source and founded that the only effect is the time of arrival to the audience. The impact of the atmospheric pressure is small, and the pressure close to the ground is so high that other limitations of wave propagation limit the SPL before the negative amplitude riches vacuum in the condensation. When the wave compresses the air, the wave travels faster such that the received wave at the audience is a sawtooth wave. The effect produces harmonic distortion where some of the harmonic energy is attuned be the viscous losses. The harmonic distortion is present in SPL lower than 120 dB SPL but is not as critical as the distortion created by the construction of the speaker enclosure. The audience area is assumed to have high absorption in frequency above 1.0 kHz, while frequency in octave band 31.5 Hz is assumed to have low absorption of the audience.

In the inhomogeneous atmospherical condition, it is founded that refraction of the sound wave is one of the biggest challenges for an outside sound concert. The refraction occurs because of inhomogeneous speed which is present in both inhomogeneous wind and temperature. It is further founted that the refraction is frequency dependent and distance dependent. The effect, however, is low at a distance lower than 50 m with a wind speed of 5 m/s. Depending on the atmospheric condition two kinds of refraction was founded, upwards and downwards. Upwards refraction produces a shadow zone where turbulent atmospheric condition makes creeping wave intro the shadow zone. For the case of oblique and crosswind the effect of high frequency, the refraction might be zero at direct crosswind but increases progressively as the direction of propagation deviate from crosswind. One measurement was done to research the effect of crosswind on a line source array. It was founded that the average $L_{\text{Aeq},5}$ SPL at microphone 1 was 6.17 dB SPL higher than microphone 2. Therefore it can be concluded that the crosswind with respect to the speaker coverage area does affect.

Chapter 4

Problem statement

Based on the knowledge founded in chapter 2 and the conclusion drawn in chapter 3 a problem statement can be made. For the rest of this theses, the following will be the focus.

Is it possible to control the speaker directivity such that the average SPL over the speaker coverage area is more homogeneous in cross- and obliquewind condition

4.1 Delimitation

The following delimitations are made for the rest of the project:

- It is chosen to work with mono line source array since the number of line source array element is limited to six pieces.
- Due to the amount of needed audience to the research, the homogeneous SPL is searched over mown grass without the audience.

Part II

Test Design

Chapter 5

Proposal solution

5.1 proposal of solution to the wind problem

This section aims to propose a solution to the problem founded in the crosswind measurement section 2.4.1 and the problem statement in chapter 4. To be able to find a solution to the problem, the optimal condition is defined in section 5.2. Then a proposed solution to the crosswind is defined in section 5.3 and lastly a proposed solution to the parallel wind is defined in section 5.4

5.2 Optimality condition

To be able to search for a solution and design a test to research if the proposed solution has the which effect on the coverage area, the optimal condition is defined in this section. The optimal condition is as simple as the SPL coverage in the coverage area of the speaker without wind. In other words, the line source array has a frontal horizontal directional angle defined as the -6 dB SPL limit of the main pressure lobe. The line source array main lobe is given in the horizontal degree as an addition of the main lobe from the frontal direction to both side and can both be symmetric and asymmetric, depending on the line source array element. The following Figure 5.1 shows an illustration of the main lobe.

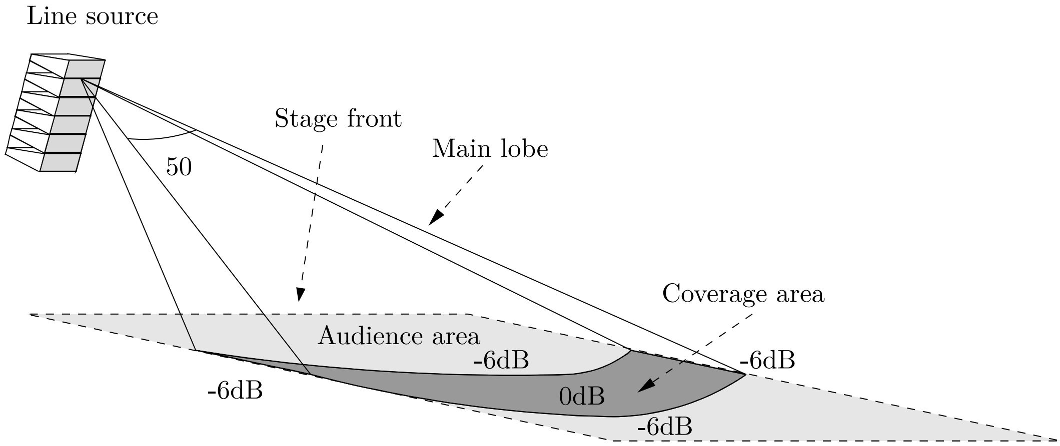


Figure 5.1: The figure shows the pressure limit which defined the main lobe of the line source array and the coverage area without wind

As illustrated in Figure 5.1 the coverage area is a parabolic surface which is limited as the -6 dB SPL coverage limit of the line source array. This is the coverage area with no wind effect and is the coverage area which is defined as the optimal condition. The solution to the crosswind and the parallel wind, therefore, is a way to be able to adjust the coverage area such that the line source array can eliminate the effect of the wind and cover the area as without wind. To be able to eliminate the wind effect at the audience area, audience area have to be defined To defined the audience area, a questioner is made among the large sound rental company in Denmark which ask for audience area to a concert. The questioner is founded in Appendix J. The gold of the questioner is to find the highest coverage distance from the line source array. The founded maximum distances before delay tower is approximate 60 m, and furthermore, the wind might be stronger than 5 m/s to a concert but wind above 10 m/s to 12 m/s will stops the concert. Therefore the defined coverage area as shown in Figure 5.1 is 60 m from the stage front.

5.3 Proposal solution to crosswind

The crosswind problem is shown in section 2.4.1 to highly modified the coverage area. Agents the wind the upwards refraction is shown to attenuate the sound more than 6 dB SPL A-weighted at a distance of only 25 m and a mean wind strange of 14 m/s. Furthermore, it is founded that the shadow zone SPL depends on the SPL in the sound path, because the wind eddies, eddies the sound into the shadow zone. It is then researched if adding more power into the upwards refraction direction also ads more power into the shadow zone by the wind eddies. The following Figure 5.2 illustrate the eddies theory in upwards refraction.

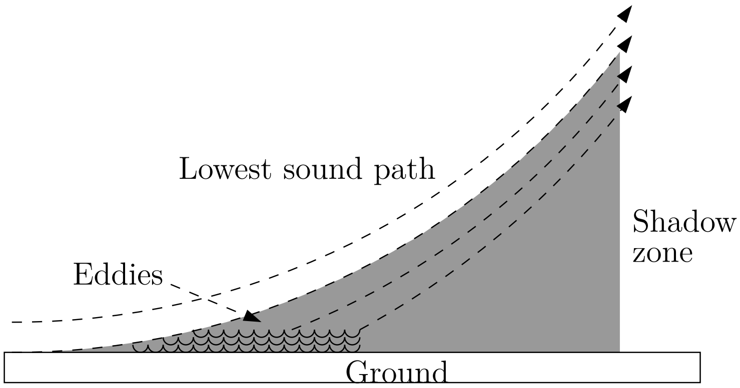


Figure 5.2: The figure shows the sound path above the zhadow zone and inside the shadow zone produced by the eddies

The proposed solution is then to steer more power into the direction of upwards refraction and less power into the front and in the direction of downwards refraction to research the eddies theory shown in Figure 5.2. The following Figure 5.3 shows a graphical illustration of the proposed solution to archive a more homogeneous SPL in the coverage area of the line source array with the wind.

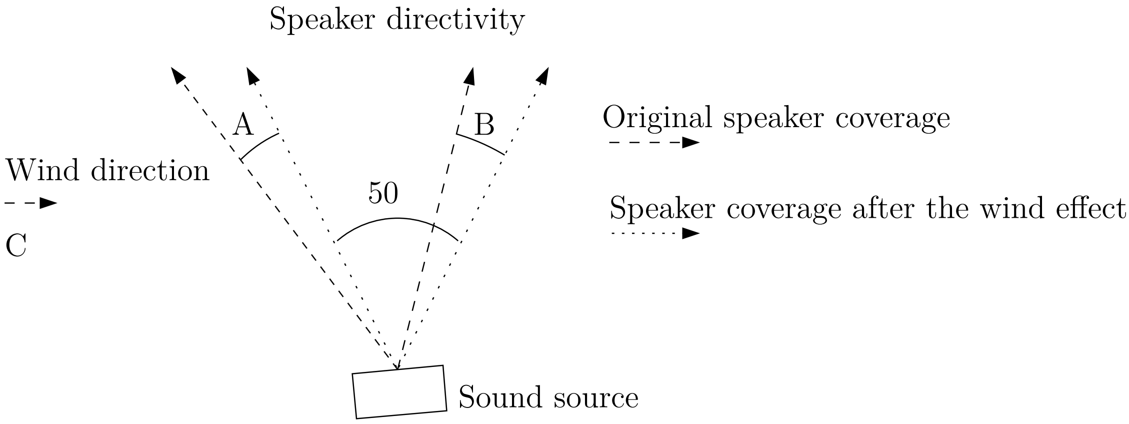


Figure 5.3: The figure shows the wanted direction of the sound coverage area after the effect of crosswind. C is the speed of wind in cross direction of the frontal direction of the speaker. A and B is the main lobe angle that needs to be founded. On the figure the angle are equal but that might not be true

The gold is then to search A° and B° based on wind speed C m/s and the optimal coverage area as shown in Figure 5.3 such that the SPL coverage differences is optimized. As founded in section 2.4.1 the refraction is frequency dependent, therefore, finding the optimal A° and B° might not be possible for all frequency. Furthermore the refraction in the low frequency is nearly zero for the distance present at the concert.

5.4 Proposal solution to parallel wind

The above proposal solution deals with the crosswind problem. When the wind direction change such that the wind comes from the back audiences to the stage, or in other words, is parallel with the frontal direction of the speaker, another theory is research than the eddies theory. The resand to search for another solution is that using the eddies theory in parallel wind require that the power from the line source array is raised. The proposed solution is then to move the shadow zone instead of raising the power in the shadow. To be able to move the shadow zone, the idea is to change the vertical angle of the main lobe, such that the upper speaker ether point more downwards or upwards for upwards refraction or downwards refraction respectively. Therefore if the upper speaker points more downwards the energy from the speaker might arrive at the ground where else if the line source element is pointing parallel to the ground, the energy never enters the ground surface. The following Figure 5.4 shows the proposal solution to parallel wind refraction.

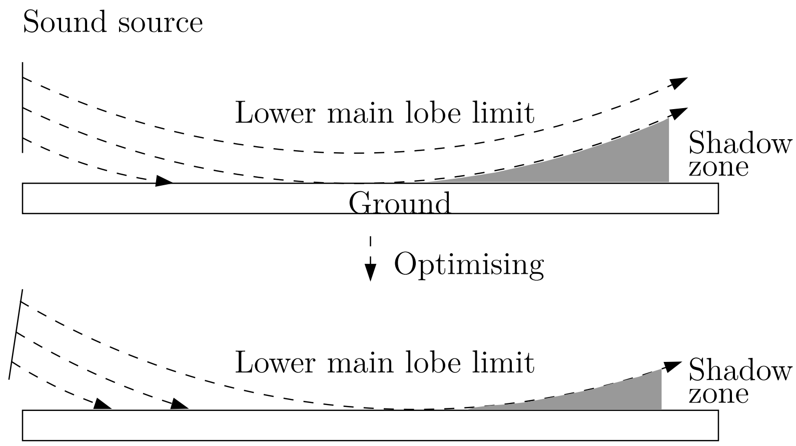


Figure 5.4: The figure shows the proposal solution to the upwards refraction. The upper part of the figure shows the lower vertical main lobe ray while the array is orthogonal to the ground where the lower part of the figure shows the lower vertical main lobe ray while the line array is angled more downwards

The shadow zone distance is depending on the hight of the line source array from the ground within the limited hight of flying points on the stage. As higher the line source array is flown, as higher the distance is before the shadow zone is present while the vertical angle is optimised to the audience area.

Chapter 6

Test of Proposal Solution

6.1 Test of proposal solution

This section aims to design a test setup for testing the proposed solution. The test setup will be based on an L-acoustics KUDO line source array which is described in section 6.2 without any modification. This chapter designs the measuring method and the needed windscreen for wind measurement. To be able to design the measuring, the used line source array is briefly described in section 6.2, then the measuring setup is designed in section 6.3 and in the end, the measuring method is designed in section 6.4.

6.2 Description of the used line source array

This section aims to explain the functionality for the line source array which is used to test the proposed solution. The description starts with a short introduction to the line source element, then the frequency response of the single element, the horizontal directionally control, and the vertical directionally control is explained.

The line source elements which is used to test the proposed solution is an L-Acoustics KUDO line source array. This line source array is a legacy long throw variable curvature speaker. The speaker is designed as the second option in the K series which today is renewed and renamed to L-acoustics K2. The speaker can be flown as a vertical line with a maximum of 21 elements. The maximum number of an element is due to the safety limit on the flying tools. One single element have a frequency response from 50 Hz to 18 kHz with a maximum deviation of ± 3 dB SPL and have a maximum SPL of 140 dB SPL at 1 m. The following Figure 6.1 shows the average frequency response over 40° horizontal angle of a single line source element.

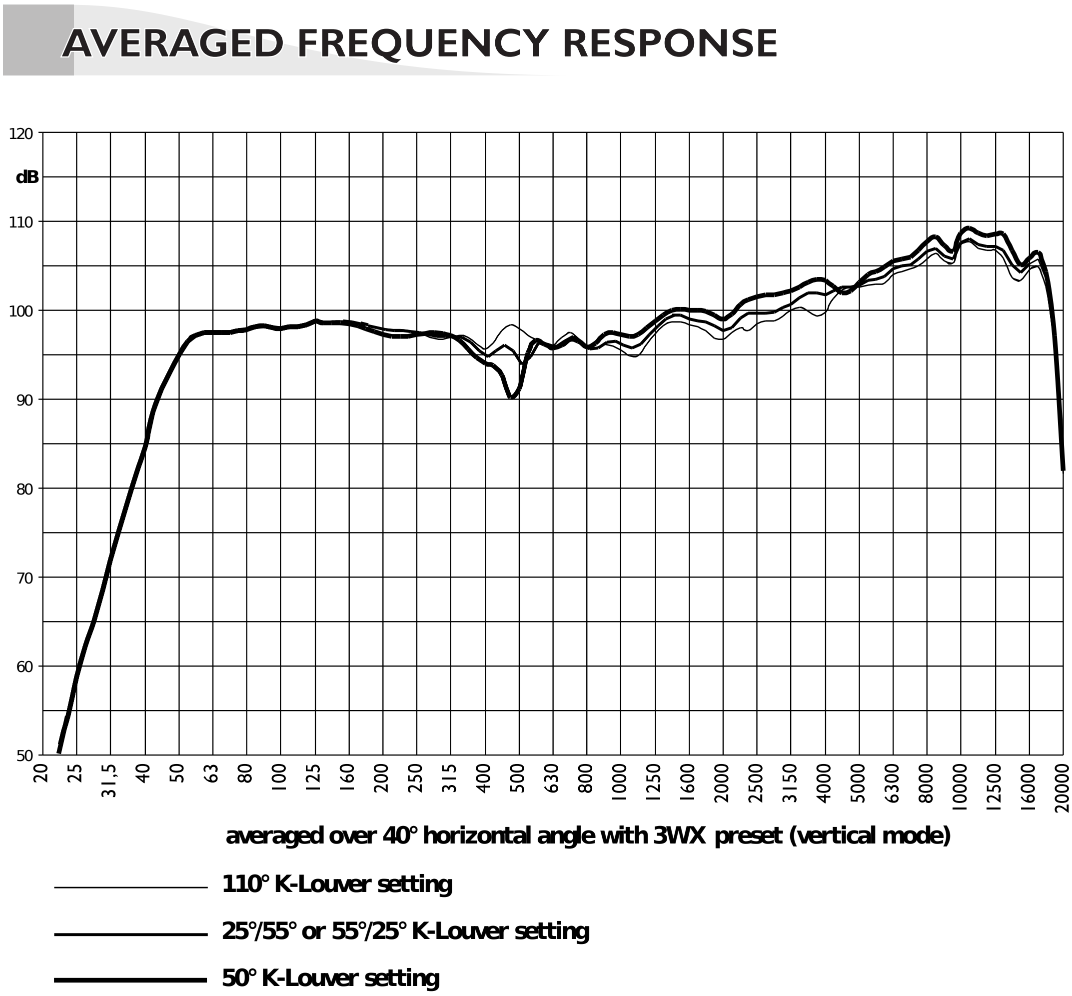


Figure 6.1: The graph shows the average frequency response over 40° horizontal angle of a single line source array at 1 W [?].

The horizontal coverage angle can be controlled individually on every line source element. The line source element allows both symmetric horizontal coverage and asymmetric coverage. The angle from the frontal direction to the outer main lobe −6 dB SPL is either 25° or 55°. By this two angle for both sides, four coverage angle of the speaker is possible, 110°, 50° and 80° ether to the left or the right. The following the Figure 6.2 shows both the wide and narrow symmetric main lobe option of the L-acoustics KUDO. The asymmetric coverage can be founded in [?]

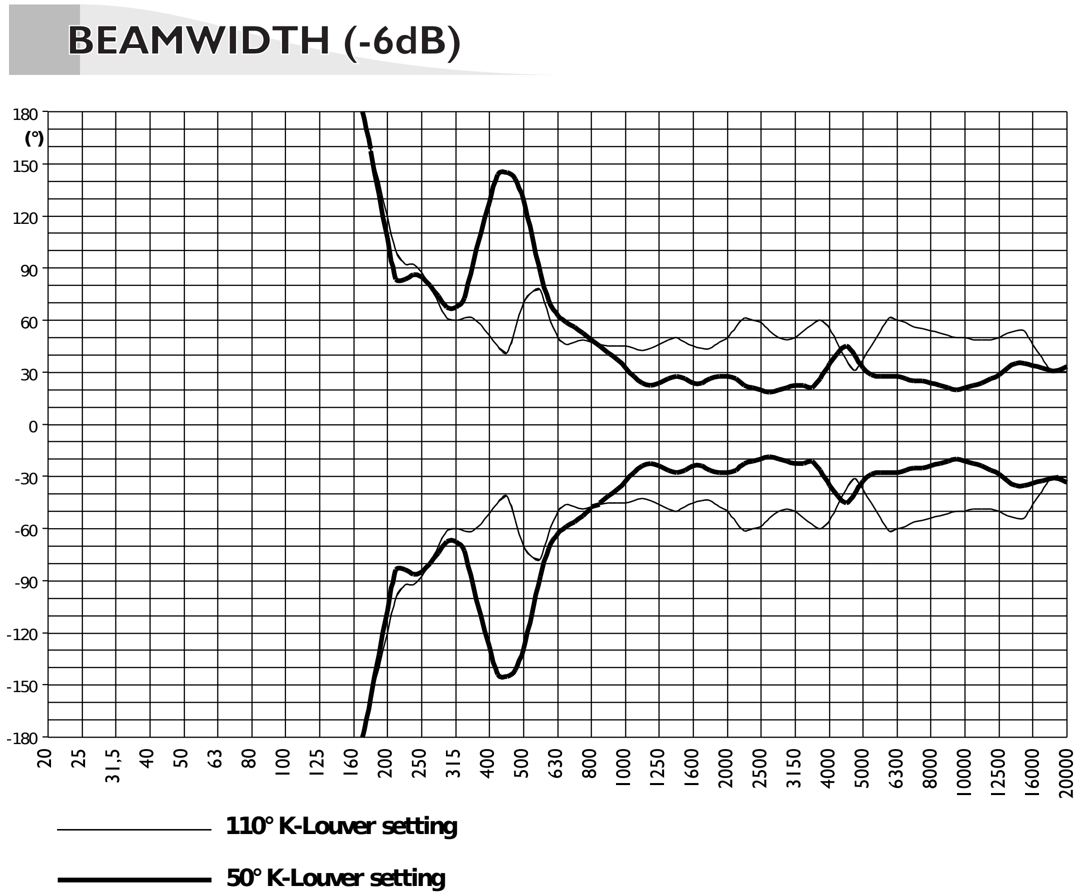


Figure 6.2: The graph shows the symmetric coverage area of the L-Acoustics KUDO line source array [**?**].

The mechanical coverage solution in the L-acoustics KUDO as well as other line source array element is not made for wind problems but for neighbouring disruptions and higher SPL in the main lobe of the high frequency. All solution used today is only possible to be changed by hand and is not electrically controlled. The method for changing the horizontal directivity in the L-acoustics KUDO line source element is two plexiglass plate fixed to the front grill. The fixing mechanism can be adjusted sidewise by realising two splits on both plexiglass plates. The plate can then be slid along the grill to change the mouth of the speaker output. The following Figure 6.3 illustrate the principle.

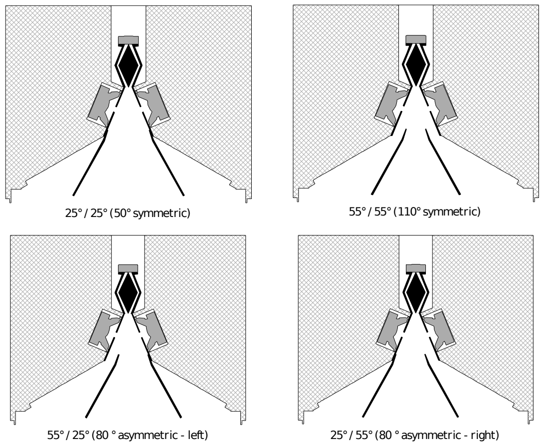


Figure 6.3: The figure shows how the horizontal directivity is controlled on a L-Acoustics KUDO line source array element [**?**].

The line source array vertically coverage area can also be controlled from 0° to 10° with 1° step size. To be able to control the vertical main lobe of the line source array, the mechanical solution is the angle between the line source element. This means that the vertical coverage control cannot be controlled on the individual line source element as the horizontal coverage. To be able to control the vertical coverage, the speaker is trapeze designed such that the high frequency horn throat stays together while the angle between the elements is adjusted in the back of the element for every line source element. The following Figure 6.4 shows how the line source element are angled vertically.

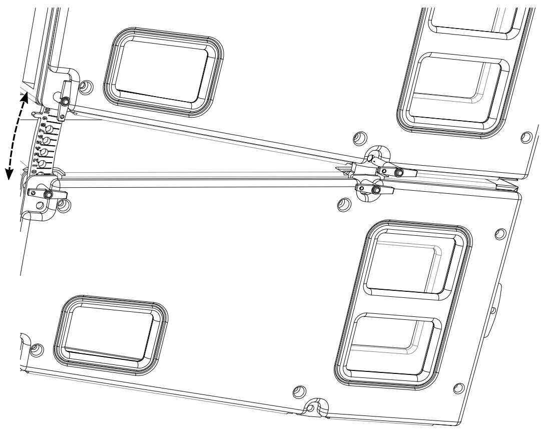


Figure 6.4: The figure shows how the vertical directivity is controlled on a L-Acoustics KUDO line source array element [**?**].

To be able to fix the vertical coverage on the L-acoustics KUDO, the upper left rigging pin shall only be placed into the line source element rig when the angle shown on the metal pease shows the desired vertical coverage between two line source element.

6.3 Measuring setup

This section aims to design the speaker setup such that the proposed solution can be tested without mechanical change of the line source array. The test setup, therefore, do not change the speaker directionality adaptive but mechanical by hands before every measurement. The amount of available line source array for the test is limited to six line source element and four belonging low frequency driver. The following Figure 6.5 shows the measuring setup which is used for both measurement.

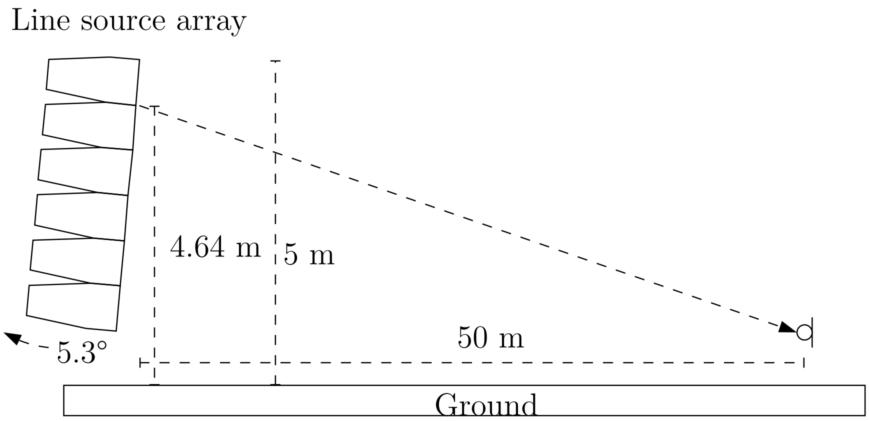


Figure 6.5: The figure shows test setup for both beasurement

Based on the founded maximum distances from the front of the stage to the delay tower, the line source array is rotated vertically as shown in Figure 6.5, such that the two upper line source element covers the measurement area without wind. The following two section section 6.3.2 and section 6.3.1 describe the line source settings which differs from each other with respect to the rotational procedure.

6.3.1 Crosswind line array settings

To test the crosswind proposal solution, the idea is to measure the SPL coverage while playing in the frontal direction and then measure the SPL while the line source array is horizontally rotated in the wind direction. To be able to ensure that the horizontal rotation of the line source array keeps the SPL in the downwards refraction direction as much as possible, this section starts designing the directional angle settings of the line source array. It is founded in section 2.4.1 that downwards refraction raises the SPL by downwards reflection but the amplification is much less than the attenuation in upwards refraction and is therefore assumed negligible for directionally chose. Therefore, to decide on the horizontal directionally settings the pressure versus angle has to be found for the used line source array. The −6 dB SPL directionally is already known from the datasheet, but this detail level is not high enough to decide on any prediction of the attenuation in the downwards refraction direction. Therefore a directionality measurement is made on the used line source array for all possible horizontal directionality settings. The measurement can be founded in Appendix G and the Figure 6.6 shows the resulting directionality of the L-acoustics KUDO with the 25° / 25° settings

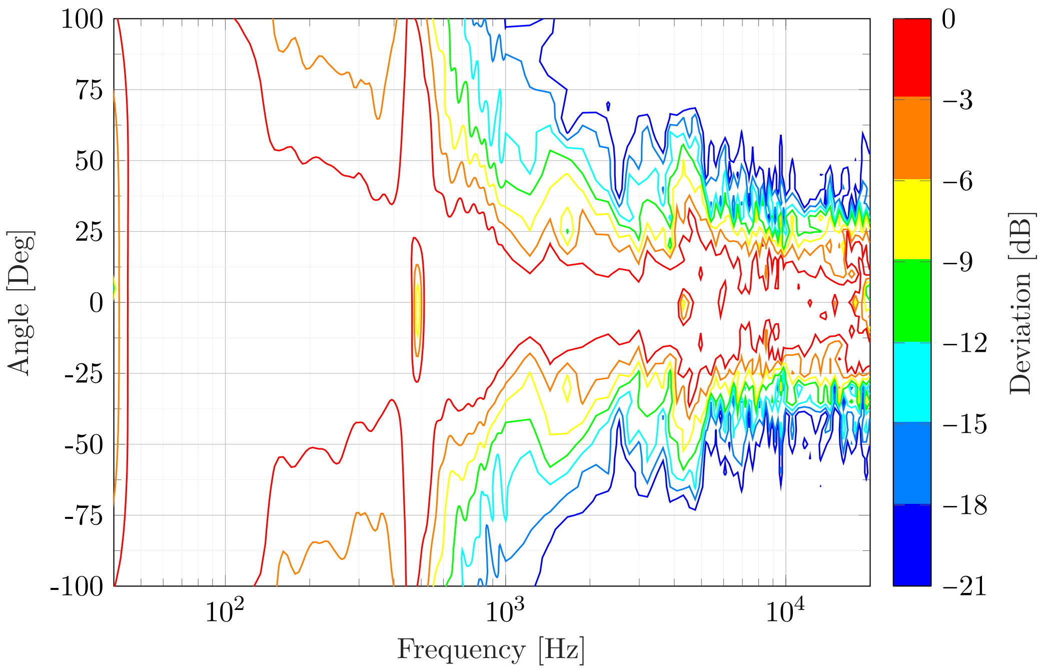


Figure 6.6: The graph shows a contour plot with 3 dB SPL step of the directionally of the L-acoustics KUDO with 25° / 25° settings. The lower black contour line indicate the dB directionality for the maximum rotation of the speaker

The following Figure 6.7 shows the resulting directionality of the L-acoustics KUDO with the 25° / 55° settings

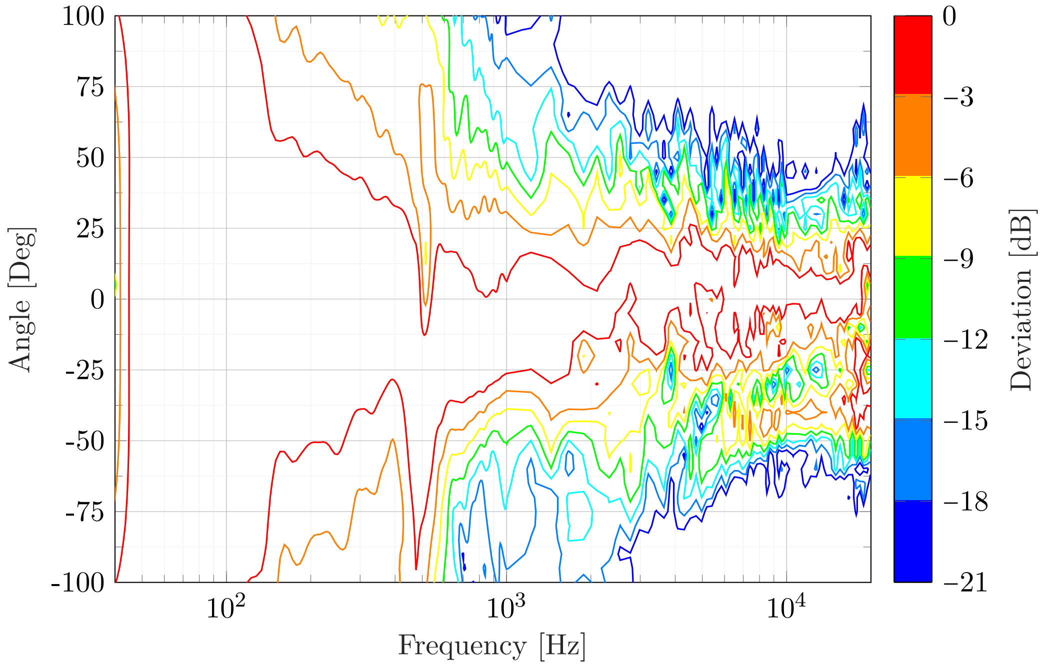


Figure 6.7: The graph shows a contour plot with 3 dB SPL step of the directionally of the L-acoustics KUDO with 25° / 55° settings. The lower black contour line indicate the dB directionality for the maximum rotation of the speaker

As seen in Figure 6.6, when the speaker is rotated 25° as the maximum allowed rotation in the proposal solution the SPL in the downwards direction is lowered from approximately −6 dB SPL to approximately −18 dB SPL which is an attenuation of 12 dB SPL. This 12 dB SPL attenuation might attenuate the SPL in the downwards refration too much. As seen in Figure 6.7, when the speaker is rotated 25° the attenuation SPL in the downwards direction is lowered from approximately −6 dB SPL to approximately −12 dB SPL which is an attenuation of 6 dB SPL. To decide on a mechanical solution an example is calculated based on the measurement above and the measurement in section 2.4.1. The example is based on an rotation of 20°, where the rotational power from a rotation of the L-acoustics KUDO is added to the measurement. The following explains and shows the example.

Example The example shows four cases from the line array, one case where the data from the datasheet is used, one case where the measurement in section 2.4.1 is used. Then two examples where the differences in SPL is calculated from an rotation of 20° for 25° / 55° settings and a rotation of 10° for 25° / 25° settings and added to the measurement. The following Figure 6.8 shows the example.

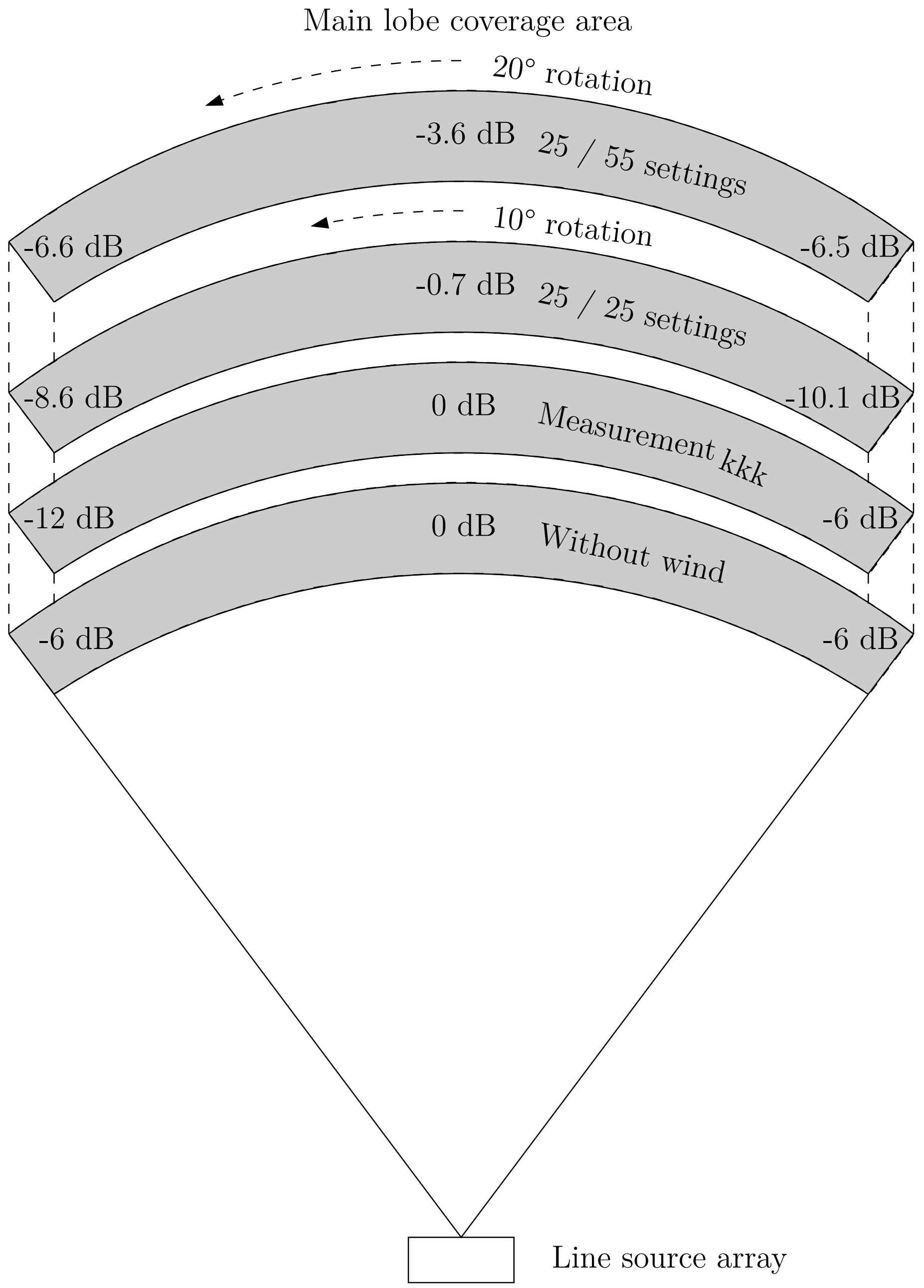


Figure 6.8: The figure shows the main lobe coverage area without rotation for the to lower and with maximum rotation for the to upper

The centre in the measurement was not measured doing the measurement, so the stated value is a prediction based on [**?**] which indicate that the energy addition at short distances because of downwards refraction is small compared to the energy loss with upwards refraction.

As seen in Figure 6.8 a rotational of 20° calculate a more homugenius SPL while the line source array is in 25° / 55° settings. The diviation from the frontal direction is approximatly 3 dB SPL. In the other case while the rotation is only 10° and the settings is 25° / 25° the SPL is also approximately evently spredt but the divination to the frontal direction is much higher and is at least −7.9 dB SPL. Based on the calculated example, the chosen directionally settings is 25° / 25° for the measurement. The following Figure 6.9 shows the measurement setup as a top wiev.

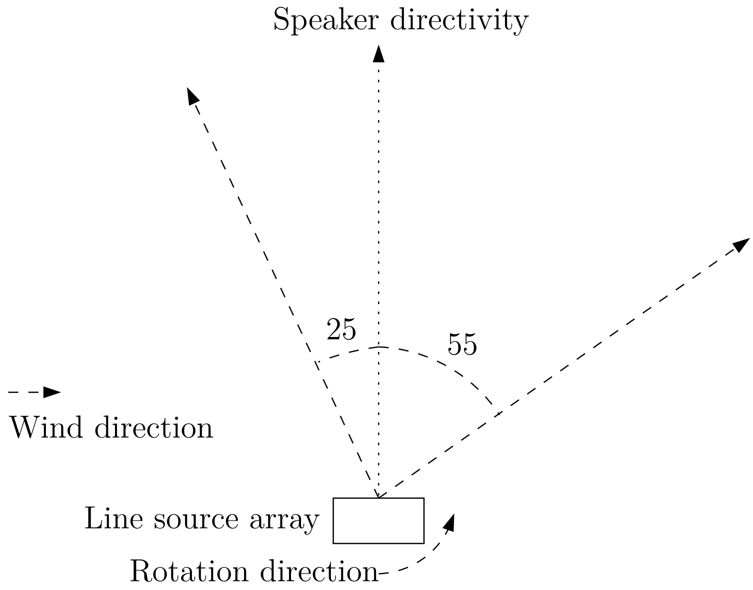


Figure 6.9: The figure shows the line source setup for the measurement.

The rotated array is rotated with few degrees for every measurement form 0° until 25°, which mean that the highest possible SPL points in the maximum angle.

The transfer function which shows the lowest SPL deviation between microphone position is the angle of the solution.

6.3.2 Parallel wind line array settings

For parallel wind, the idea is to have horizontal symmetric coverage while changing the vertical angle for every measurement. The array is tilted some degree until the optimal angle is measured. The optimal angle is the angle where the shadow zone is pushed as far away as possible concerning the wind speed. The following Figure 6.10 illustrate the line source array from the side.

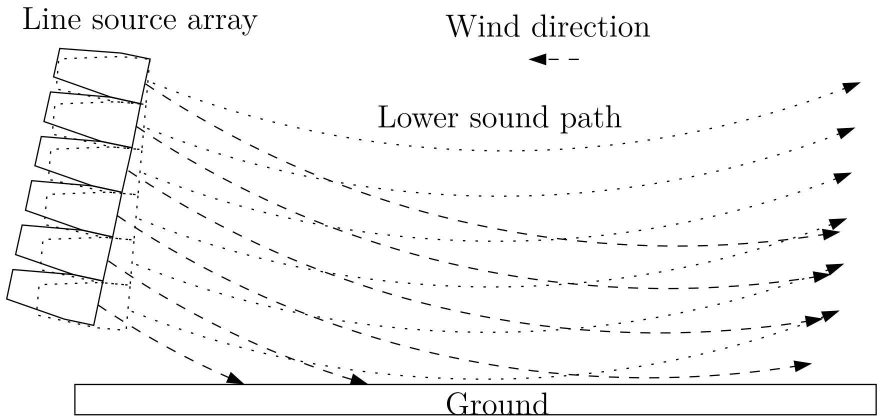


Figure 6.10: The figure shows the line source setup for the measurement. The line source array consist of six KUDO line source element attached with 0° vertical coverage angle

The Figure 6.10 illustrate that the wind makes refraction of sound upwards and therefore the main lobe is curved upwards.

6.4 Designing the measurement

This section aims to design a test on the non-modified line source array to test the proposed solution from section 5.1. To be able to test the proposed solution, a measurement system has to be designed. To be sure that the wind noise does not affect the measurement the part of the design takes care about finding the preferable microphone windscreen configuration, based on the available STOF in acoustics lab. The second part measure the frequency dependent attenuation of the chosen windscreen compare to the microphone response without a windscreen. The comparison is done to count for the attenuation of the windscreen such that the measured data reflect the SPL at the point as pleasant as possible. The third part takes care of designing the necessary data logging function. The last part designs the measuring signal playback and record method.

6.4.1 Microphone position

This section aims to design the position of the measuring microphone. The position of the microphone depends on the wind condition, for the crosswind the microphone be placed in the sound field were for the parallel wind, the microphone shall be placed in the shadow zone. The description starts with the former.

The microphone position highly depends on the coverage area of the line source element. Usually, the element which is flown highest cover the as far as possible where the line source element which is closest to the ground cover the frontal audience. Therefore, the distance from the speaker to the microphone has to be found based on the knowledge of coverage distance and the minimum distance before refraction.

The distances from the stage to the back audience depends on the size of the concert. For a small concert, the main stage covers the full area where for large concert delay tower helps the coverage. Delay tower is often used for a concert where the distances from the stage to the audience is above 75 m as Roskilde festival. Roskilde festival situate there delay tower at a distance no longer than 73 m from the main stage Appendix J. The general founded maximum distances from the main stage to the first delay tower was about 73 m for a huge concert, 50 m for a large concert and 30 m for small concert Appendix J. Base on the knowledge of the maximum distances founded in Appendix J and that Roskilde festival is a special case concerning the size and 50 m to 60 m is an often used maximum distance for large concert depending on the hight of the main stage line array. The maximum coverage distances are chosen to be 50 m for the test since the used line array flying tools is not able to fly the line array as high as the asked companies. The flying height of the top line source array element is about from 12 m to 16 m where the flying height of the used test setup is only up to 7 m. Furthermore it was shown in Table 2.1 that refraction occur at a distance of 25 m with 13 m/s.

Another factor which plays a specific role for the microphone distance from the sound source is the hight of the microphone. There are pros and cons for placing the microphone both at the ground or above the ground. The pros of placing the microphone on the ground are that the ground reflection is eliminated, but the cons are that the shadow zone might be closer to the line source array that above the ground, as it can be seen in section 2.4.1. Relating it to the concert situation, the ground reflection in the high frequency is assumed to be low where the ground reflection in the low frequency range is assumed to be higher. Moreover, the hight of the audience ear is not at the ground but in a hight of approximately 1.70 m therefore the most realistic scenario without the audience in the high frequency is at the ear height. Since the critical frequency range is in the high ineligibility frequency range, the low frequency reflection has an only sparse effect on the measuring result.

Since the measuring distance is chosen to be 50 m the angel of the speaker is chosen to be the lowest possible angle. This choice is taken because the coverage area at that distance is high at the narrow-angle and is a realistic chose from the company and concert area point of view. The following Figure 6.11 shows the microphone position with respect to the speaker position.

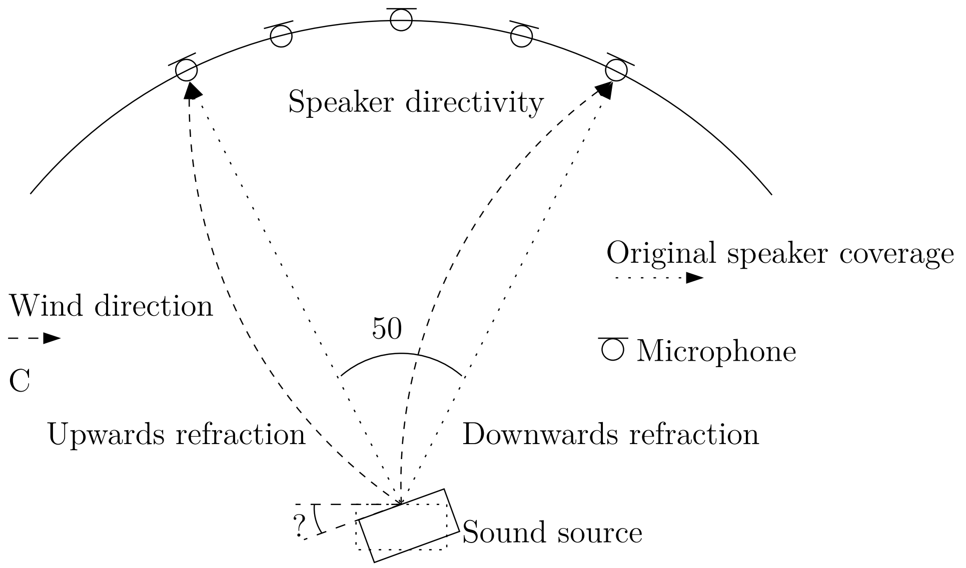


Figure 6.11: The figure shows the measurement setup

The microphone position of the vertical refraction measurement depends on the shadow zone position, and it is wanted to measure in the shadow zone to explore if it is possible to move the shadow zone backwards by tilting the line array. Therefore the shadow zone has to be found by measurement before the microphone position can be specified.

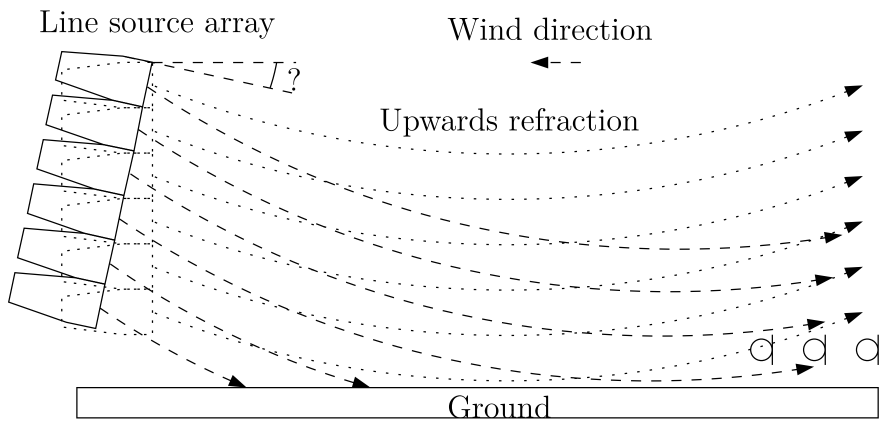


Figure 6.12: The figure shows the measurement setup

6.4.2 Design of windscreen

The aim of this section is to be able find a windscreen to the measuring microphone such that the wind noise is low compare to the measuring signal. There is to aspect in this windscreen configuration, firs the the wind noise cannot be filtered electronical between the microphone and the preamp.Therefore the strength of the wind noise shall be as low as the preamp do not overload. Secondly the measurement shall be measurement of the signal and not the wind noise, therefore the signal shall be sufficiant higher that the wind noise such that the measurement is trustable and represent the SPL procused of the speaker at the measuring point.

The idea of an additional wind screen is use the original windscreen to the microphone and then try to stop the wind in just at the microphone with a blocking surface. The surface shall therefore be able to lower the windspeed at the microphone and have as less reflection as possible. The original windscreen is cept on the microphone in the wind stop area to attenuate the wind noise that passes the blockaga and attenuate the turbulence prodused by the wind stopper.

The first two windscreen concept is very identical but just with difference size of material. The idea for the first windscreen is to seal the microphone with foam all around except at the frontal direction. The frontal direction include both 180° angle in the vertical direction and 90° in the horizontal direction. The resan to chose that high degree vertical opening is that no sound is from ground reflection is effected and no sound from upwards is stopped by the foam. The reson to have a narrow horizontal opening is to be able to get sound inside the opening but still have a wind stopping effect. The following Figure 6.13 illustrate both windscreen configuration one and windscreen configuration two, just with one size foam wedge.

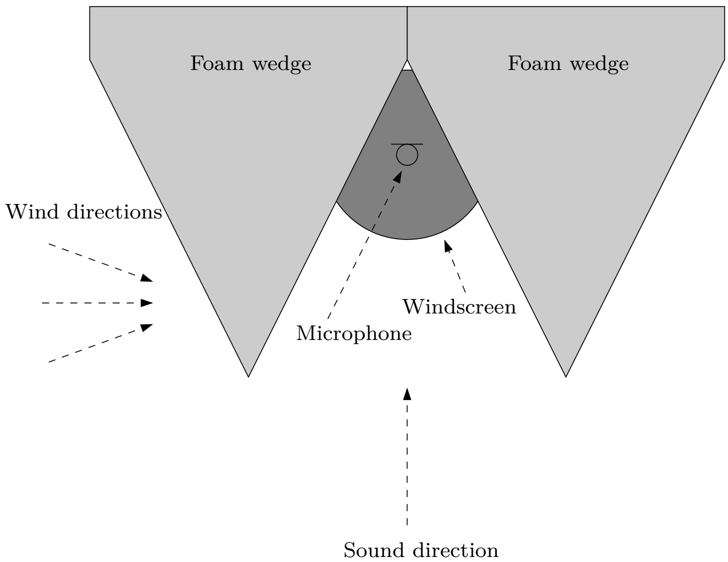


Figure 6.13: The figure shows the foam wedge concept. The concept is is covering over to differend foam wedge, ether two small or two large. The small concept is defined as windscreen configuration one, where the large concept is defined as windscreen configuration two.

The next concept build on the concept in Figure 6.13 just with plan surfaces rockwool plates. The opening is also 180° angle in the vertical direction and 90° in the horizontal direction. The concept is defined as windscreen configuration three. The following Figure 6.14 illustrate the concept.

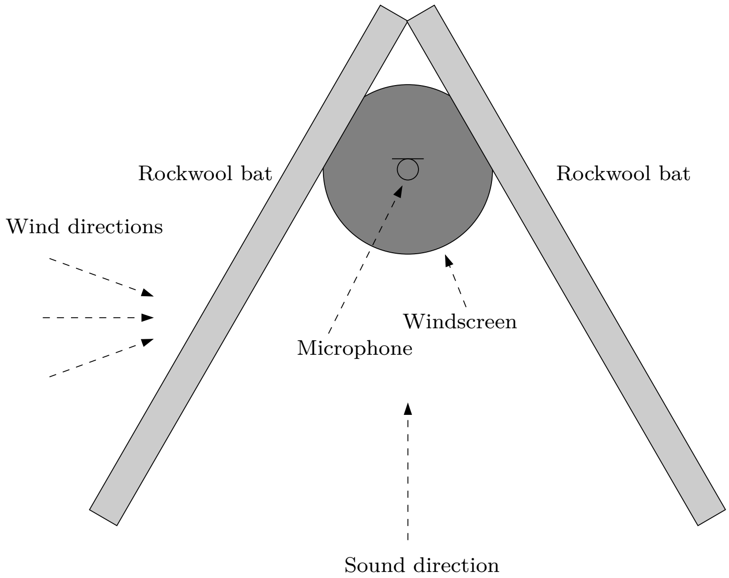


Figure 6.14: The figure shows the rockwool concept. This concept is defined as windscreen configuration three.

The next concept build on minimizing the reflection from the additional windscreen by only placing the microphone close agenst one surface, which cover for the wind noise. The concept is definde as windscreen configuration four. The following Figure 6.15 illustrate the concept.

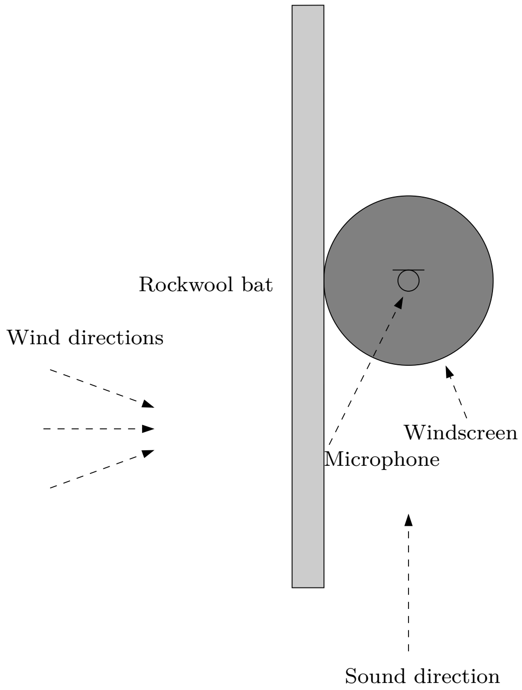


Figure 6.15: The figure shows the single rockwool concept. This concept is defined as windscreen configuration four.

A combination of windscreen configuration two and windscreen configuration five is also tested. The combination is defined as windscreen configuration five. The following Figure 6.16 illustrate the concept.

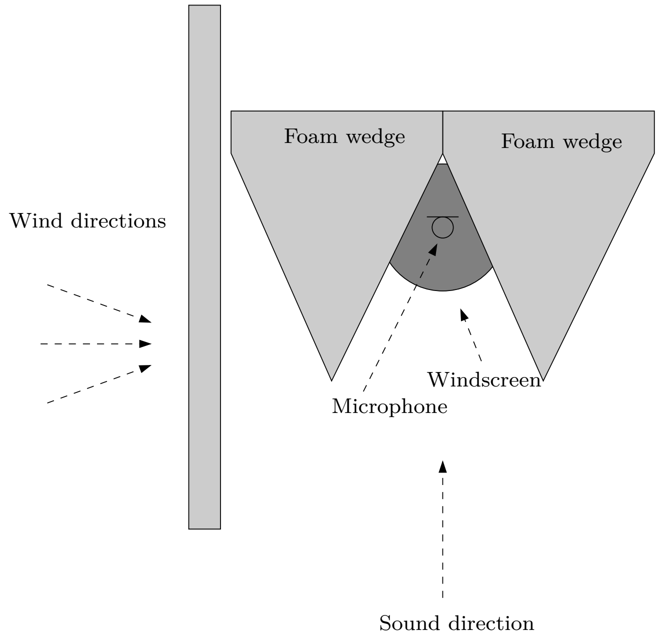


Figure 6.16: The figure shows the single rockwool concept. This concept is defined as windscreen configuration five.

Before the optimal windscreen configuration is founded, an optimality creteria is defined and a test is designed. The optimal creteria for the windscreen is as low as possible wind noise at the microphone and low reflection and direct sound attenuation from the windscreen. To find the windscreen configuration which meets the creteria best, three test is made on the windscreen configuration. First the wind speed attenuation of the windscreen configuration is measured to ensure that the windscreen configuration concept does have an effect on the wind speed. The measurement of the windspeed attenuation can be founded in **??**. Secondly the frequency response of the windscreen have to be founded to ensure that the windscreen configuration does not have a large influence on the frequency measurement response of the speaker. To test this criteria, the frequency response of a speaker is measured in the anechoic chamber without any windscreen configuration and without the original windscreen. This measurement is compared with the frequency response of the speaker with the windscreen configuration. The measurement is founded in **??**. Finally the wind noise is measured. To measure the wind noise two low speed and low noise fan is generating 2.5 m/s at the microphone position. The wind noise is measured without any windscreen configuration and the original windscreen and compared with the wind noise in the microphone position in the windscreen configuration. To ensure that the background noise is identically on the wind noise measurement with and without the windscreen configuration two microphone are used and recorded simultanius. Both the time signal and the frequency content is analysed. The measurement is founded in **??**. The result for all configuration is as following.

Configuration one is the one with the smallest foam wedge and size of the wedge is measured to have the worst wind attenuation. The wind attenuation shows that the wind speed is lowered from 8 m/s to 2 m/s. But the directional turbulence in the wind is more stabile in this configuration compare the configuration three and above. The frequency response of the windscreen configuration is the one that have the lowest effect. At low frequency upto 100 Hz the windscreen does not effect the measurement. Frequency above the frequency response gets off with about 2 dB SPL compare with only the original windscreen. The measured wind noise attenuation is equal zero. In the measurement the wind noise is actualy a bit worse compare to only the original windscreen. The attenuation is both approximatly 10 dB SPL for both with only the original windscreen and the windscreen configuration in the low frequency below 10 Hz, but at some frequency the attenuation is lower that 5 dB SPL for the windscreen configuration. for frequency above 10 Hz the windscreen configuration have no effect.

Configuration two is the one with the largest foam wedge and size of the wedge is measured to have one of the best wind attenuation. The wind attenuation shows that the wind speed is lowered from 8 m/s to 1 m/s and have less peek in the wind speed compare to the windscreen with rockwool. The directional turbulence in the wind is more stabile in this configuration compare the configuration three and above but little less stabil compare to configuration one. The frequency response of the windscreen configuration have an amplification in the low frequency range from 80 Hz to 600 Hz of 2 dB SPL. From 1.0 kHz and above the frequency response is very similar compare to only the original windscreen. At low frequency upto 80 Hz the windscreen does not effect the measurement much. The windscreen attenuate the wind noise 10 dB SPL more than only the original windscreen from 30 Hz and downwards to the measured limit at 2 Hz. The frequency range between 30 Hz and 600 Hz have the same attenuation as the original windscreen and the frequency above have further 10 dB SPL more attenuation than the original windscreen.

Configuration three is the one with two rockwool bat formed as an arrow and is measured to have wind attenuation between the small wedge and large wedge. The frequency response of the windscreen configuration is the worst. It alternate between ±6 dB SPL. At the low frequency range from 80 Hz to 600 Hz the amplification goes from 2 dB SPL at 80 Hz to 6.2 dB SPL at 250 Hz and then back to 0 dB SPL at 700 Hz. At 1.0 kHz the attunuation is at 6 dB SPL and above the frequency response alternate around the frequency response of the original windscreen. The windscreen attenuate the wind noise 10 dB SPL more than only the original windscreen from 30 Hz and downwards to the measured limit at 2 Hz. The frequency range between 30 Hz and 600 Hz have the same attenuation as the original windscreen and the frequency above have further 5 dB SPL to 10 dB SPL more attenuation than the original windscreen. Based on that the frequency response and the wind wind noise attenuation is worse than configuration two, the configuration is excluded from the rest of the test and is not used.

Configuration four is the one with only one rockwool bat where the microphone is siturated close to the side of the windscreen and is measured to have one of the best wind attenuation. The wind attenuation shows that the mean wind speed is lowered from 8 m/s to 1 m/s, but the directional and wind speed turbulence is less stabile compare to the configuration the windscreen with foam wedge. The wind speed turbulence circulate from 0 m/s to 2 m/s. The frequency response of the windscreen configuration is does not change more than ±2 dB SPL in the low and high frequency range. At frequency from 600 Hz to 300 Hz the windscreen have an attenuation of 4 dB SPL. MISSING NOISE ATTENUATION

Configuration five is the one with only one rockwool bat and the two large wedge where the microphone is siturated as in configuration two and is measured to have the best wind attenuation. The wind attenuation shows that the mean wind speed is lowered from 8 m/s to 0.8 m/s, but the directional turbulence is less stabile compare to the configuration the windscreen with only foam wedge. The frequency response of the windscreen configuration is as configuration two but with little closer fit to without windscreen in the high frequency. MISSING NOISE ATTENUATION

6.4.3 Optimization of the chosen windscreen

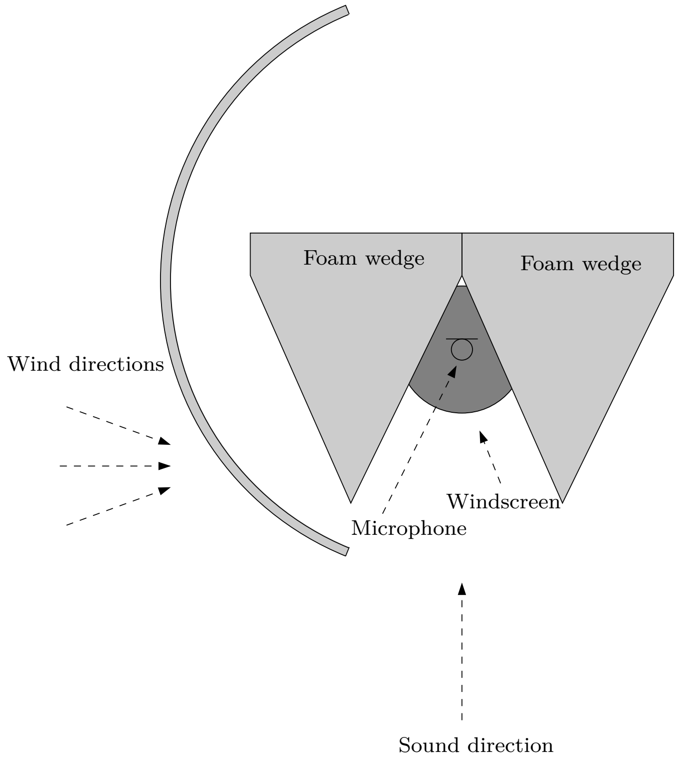


Figure 6.17: The figure shows the single rockwool concept

There is made a preliminary test test to measure the effect in a fast measuring setup before a field measurement. The measurement was done in acoustics lab with two fan and a wind speed of 2.5 m/s. The result is founded in **??**

As seen in the measurement, the original wind screen does have a huge wind noise attenuation but as shown in ... measurement, the wind noise can be further attenuated by the developed wind stopping concept.

6.4.4 Attenuation of the windscreen

The aim of this section is to analyse the inflyence of the selected windscreen.

Free field

6.5 Angeling of the line source array

The aim of this section is to design the turning method for the line source array and ensure that the speaker point in the desired angle.

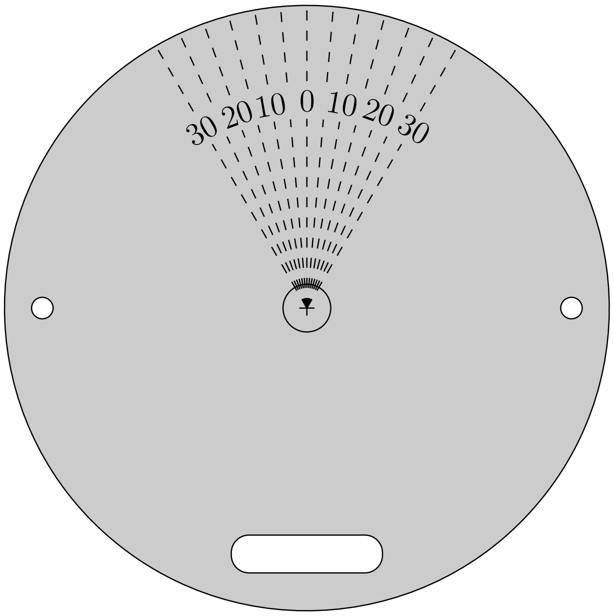


Figure 6.18: The figure shows the angle plate

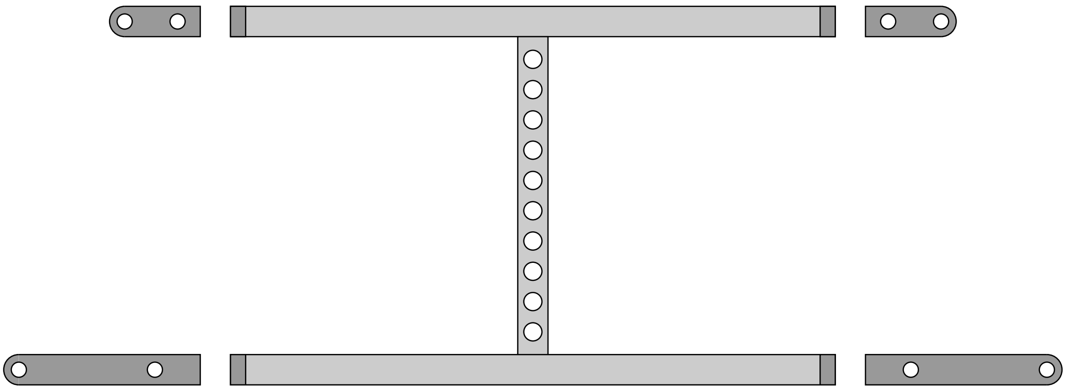


Figure 6.19: The figure shows the laser holder

6.6 Data logging system

To be able to use the information of the wind spees temparature and humidity, the data logging of the atmospherical condition have to be syncronius with the measurement.

The speaker is chosen to be adjusted to the narrow main lobe because it is assumed that the distance from the audience to the speaker is so large that the wide angle goes beyond the audience area.

Because of limitation, the speaker is flown in a hight of 6 m.

The humidity and temperature have to be measured.

To measure the SPL coverage of the speaker a flat area with mown grass is chosen to be used. The optimal area area without any building or trees might not be posible, therefore blockage or sound reflaction surface other than the ground is only allowed to be present in the double of distance compare to the distance from the speaker to the microphone. Based on the refraction effect versus distance founded in section 2.4.1. The distance from the speaker to the microphone array is chosen to be 50 m. The distance is based on the experience of the author described in section 2.1.2 and the founded refraction effect in section 2.4.1. It was founded that the refraction effect should be minimal at a distance of 50 m when the speed of wind is 5 m/s.

To keep the wind speed realistic for measurement and for concert, but still having wind pressent, the wind speed doing the measurement is limited in the range for average 5 m/s to 10 m/s. Less average wind speed than 5 m/s is avoided to ensure measureble effect of the wind on sound propagation. The higher limit of the 10 m/s is chosen to ensure that the speaker tower is safe at the hight at 6 m. The limited size of the setup makes the setup wind (følsom) because it is not puttet up as a cube but only as a surface.

where the refraction at 110 m already starts at 400 Hz.

The area is without The resend to use this

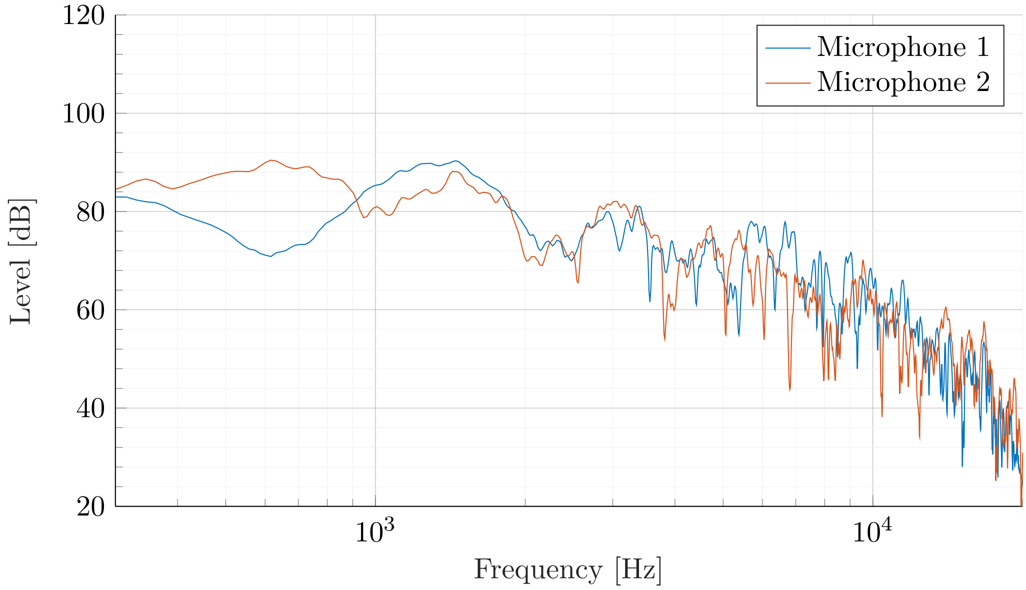


Figure 6.20: The graph shows the first transfer function measurement within the high frequency directional angle. The $L_{\text{Aeq},5}$ SPL different between the microphones is 6.77 dB SPL (IR_3) The graph is normed to contain the same $L_{\text{Aeq},5}$ SPL

6.7 Measuring program

Chapter 7

Test of measuring design

7.1 Test of measuring design

This section aims to test the design measurement in a windy day, to outsource problem and error in the measuring design. The test is done in full scale with all six line source array element and in the designed hight. The test is intended to both test the crosswind test design and the parallel wind design, but the weather condition only allowed for crosswind test. After the crosswind test design was tested the wind speed dropped to beneath 1 m/s. In the crosswind test design, three problems and one code error are observed. The code error is a data save bug, where only the direction of the wind at the speaker tower is saved. The wind direction at the microphone position is not saved doing the test measurement. The code bug is fixed for the final measurement. The following three section 7.1.1, section 7.1.2 and section 7.1.3 explain the observed difficulties or design failure and propose a solution to them. The first difficulty which is discovered is high peaks and deeps in the measurement due to ground reflections. The second problem is the non-equal frequency response of the designed windscreen, where the third error is the angle of the line source array while measuring in the hight of the ear. The explanations is based on the measurement as seen in Figure 7.1 which shows the frequency response on all three microphones at 0° rotation.

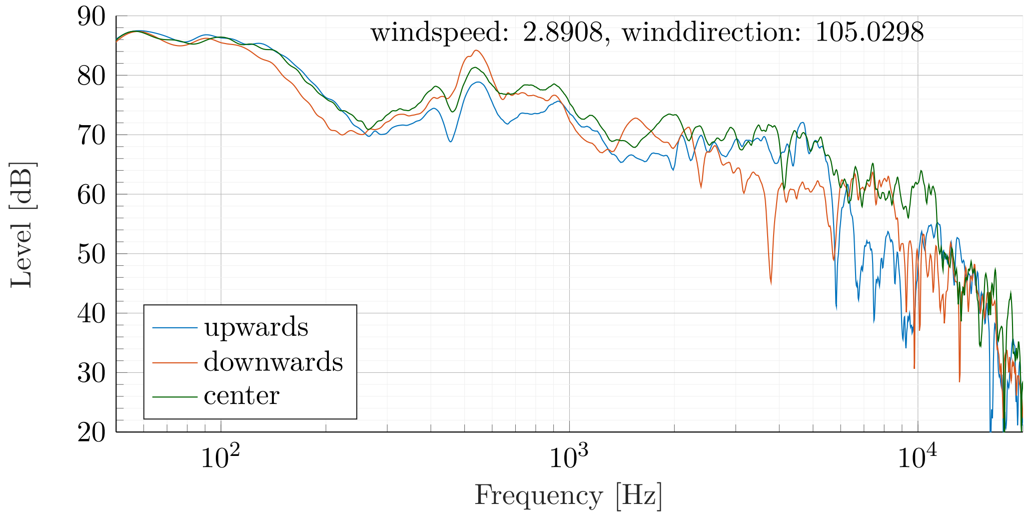


Figure 7.1: The graph shows the frequency response for all three microphone while the speaker is not rotated

All three microphone measurement in Figure 7.1 is the mean of 10 measurement. The mean is calculated in time domain by aligning the impulse response with help of cross correlation. The wind speed and wind direction is also the mean of the 10 measurement.

7.1.1 Ground reflections

The design of the windscreen is intended to block for the ground reflection of the sound and lower the wind noise. The noise floor is founded to the lower than 50 dB SPL in the frequency of interest and with a wind speed of 2.2 m/s. The measurement is done with 75 dB SPL which is more than 25 dB SPL of headroom. The headroom is not measured with higher wind speed since the wind speed dropped beneath 1 m/s at the end of the measurement, and the headroom measurement in the start was not saved. The headroom was only checked visually by some pre-measurement.

The second outcome of the windscreen is to block for the ground reflection, such that the measuring position can be in the hight of the ear as explained in **??**. This part of the windscreen works, but as the frequency grows the effect seems to be worse since higher peaks and depth are present in the measurement compare to the directionality measurement of the speaker in Figure 6.7. The centre microphone does not have that high peaks and deeps, but the side microphone seems to suffer from ground reflection, which makes the comparison between the microphone position difficult. One thing that might cause the reflection is the position different from 0° horizontal of the windscreen in the air to the ground. If the plate is tilted forward, there might be some reflection reaching the microphone. A calculation of the reflection might have helped to justify the ground reflection theory, but since the source is a line source array and not a point source the ground reflection is not as easy to calculate since there might be thousands of sound path from the line source to the microphone. the following Figure 7.2 illustrate the path calculation difficulties.

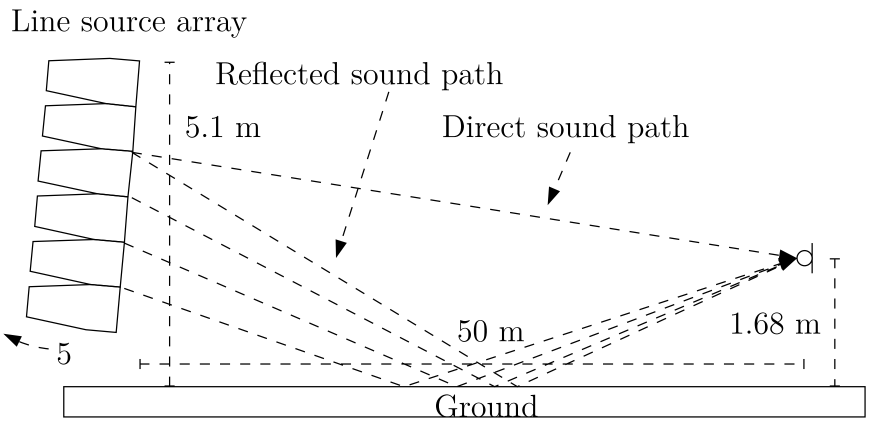


Figure 7.2: The figure shows an illustration of the measured setup and some soundpath

To be able to make a qualified considering to decide if the peaks and depth are due to ground reflection, the measurement is compared to a measurement where the microphone windscreen is situated on the ground and the frequency characteristics in the measuring direction in Figure 6.7. It have to be noted that the speaker mount, mounted on the windscreen does that the windscreen cannot lay flat on the ground but is tilted forward. The following Figure 7.3 shows a frequency response on at all three microphones position of the line source array, where the windscreen is placed on the ground and with a non-rotation line source array.

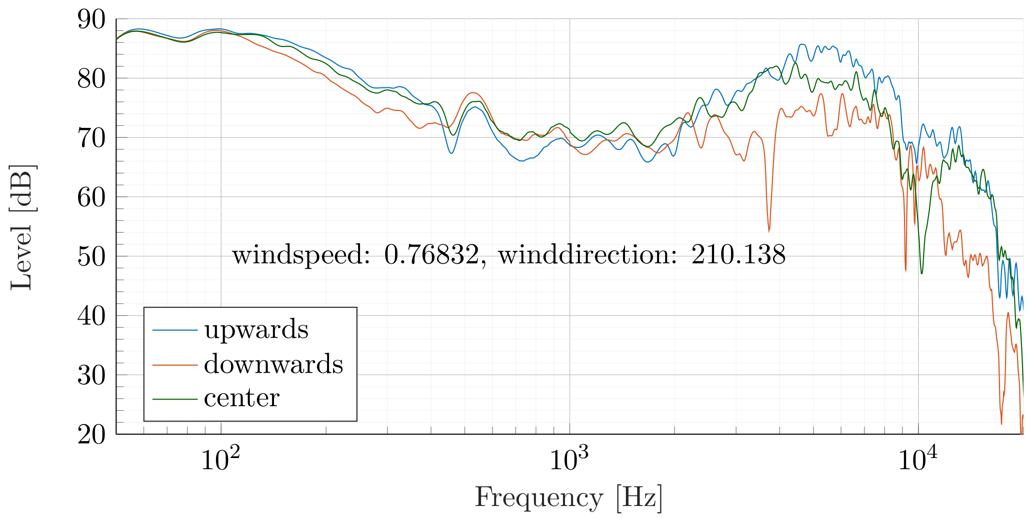


Figure 7.3: The graph shows the measuring result along all three microphone, while the microphone is in the windscreen and the windscreen is on the ground. The graph is a mean of three measurement for the upwards and downwards microphone and one measurement for the center position. The mean calculation is done in time domain where the all impulse responses is allined with cross-correlation

The depth seen at 3.7 kHz in the downwards direction is due to the directionality characteristics of the speaker and can also be seen in Figure 6.7. The same applies to 9.2 kHz and 10 kHz

Comparing the measurement where the windscreen is lifted 168 cm from the ground in Figure 7.1 and the measurement in Figure 7.3, it is seen that the first ground reflection comes around 250 Hz depending on the microphone. This ground reflection is dB wise even for all microphone position and might, therefore, be due to sound wave travels through the windscreen bottom plate. The following arriving ground reflection depends highly on the microphone position and then might be due to the horizontal angle of the plate. For the upwards refraction microphone, there seems to be highly reflections in the frequency area from 5.5 kHz to 9.5 kHz where comb filtering is present. Comparing the line source frequency characteristics, it also shows depth in that frequency range, but as high depth as present in the measurement. The measurement has more that 15 dB SPL attenuation where the line source frequency characteristics only have 6 dB SPL attenuation.

Another common response on all microphone while the windscreen lays on the ground is depth around 10 kHz. This depth might be due to the lift of the microphone while it is within the modified original windscreen or that the windscreen is tilted forward and does not lay flat on the ground.

Based on the founding in the measurement, ground reflection occur in the measurement.

7.1.2 Frequency boost

It is discovered doing the measurement that the frequency boost within 2.0 kHz and 10 kHz produced be the windscreen might be non-equal between the microphone. A comparison measurement is done, where the windscreen is removed from the centre microphone since the wind stopped at the end of the measurement. The following Figure 7.4 shows the measurement for all three microphones with windscreen and one measurement where the windscreen is removed from the centre microphone. The original modified windscreen covering the microphone is also removed.

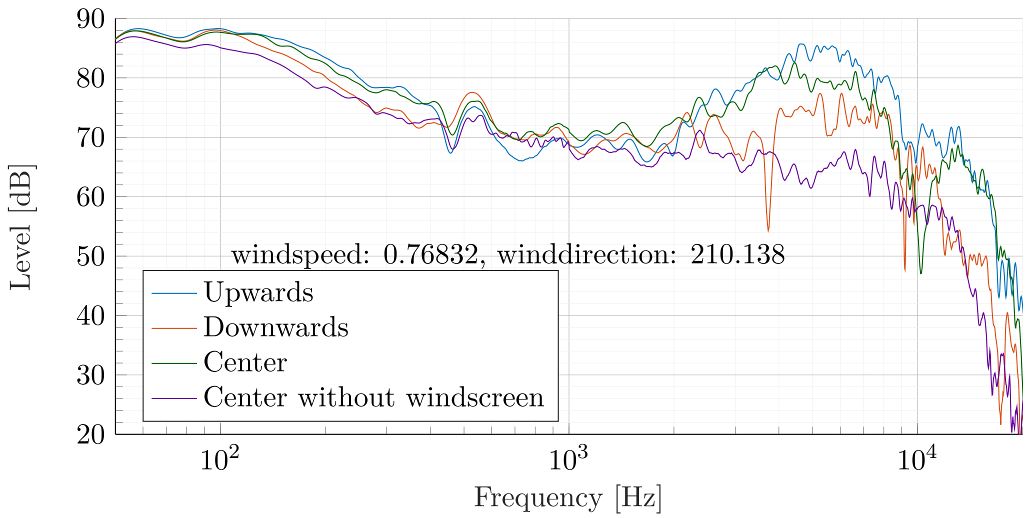


Figure 7.4: The graph shows the measuring result along all three microphone, where one of the measurement for the center microphone is done with designed windscreen setup and one measurement is done without the design windscreen setup but with the modified original windscreen. While the microphone is in the designed windscreen, the windscreen is on the ground. While the microphone is outside the design windscreen, the microphone lays in the grass. The graph is a mean of three measurement for the upwards and downwards microphone and one measurement for the center position for both center measurement. The mean calculation is done in time domain where the all impulse responses is allined with cross-correlation

It is seen in Figure 7.4 that the depth at 10 kHz is gone but also that the windscreen amplify the measurement up to 20 dB SPL. The amplification is not a problem in itself, and the arising problem is that the amplification is not even along with the windscreens. The measurement is done with less than 1 m/s of wind speed, and therefore refraction can be excluded as a factor of differences. This means that differences on the windscreen setup might influences highly on the frequency boost. Three mechanical differences are observed on the windscreen setup doing the measurement. The first is the horizontal angle of the windscreen, which was different along with all windscreens doing the measurement. Secondly, the vertical angle of the windscreen was also different along with all microphones. The vertical angle is defined to be 0° to the speaker while the speaker points directly into the centre of the windscreen opening as shown in Figure 7.5.

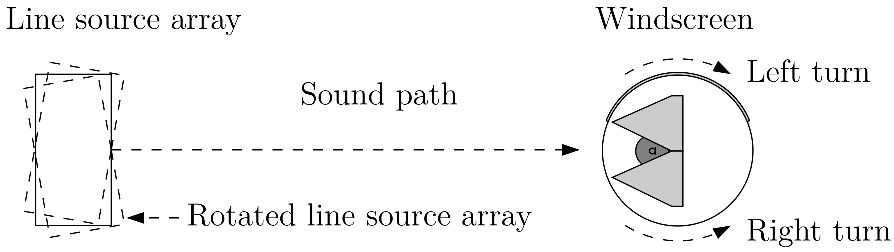


Figure 7.5: The figure shows an illustration the windscreen vertical 0° angle

As illustrated in Figure 7.5 no matter the tournament of the speaker, the opening shall point directly to the speaker. In the measurement the opening was not pointing directly to the speaker as shown in Figure 7.5, The vertical angle was adjusted with depends on the wind witch mean that the windscreen at downwards direction was turned left, where the windscreen at upwards direction was turned right.

The last differences were the placement of the foam on the plate. In the downwards direction the foam was placed more inwards in the plate while the other to foam on the windscreen was placed as shown in Figure 7.5

7.1.3 Speaker angle

The angle of the line source array was calculated based on the ground and not from the hight of the microphone hight. The following Figure 7.6 shows the microphone position calculation.

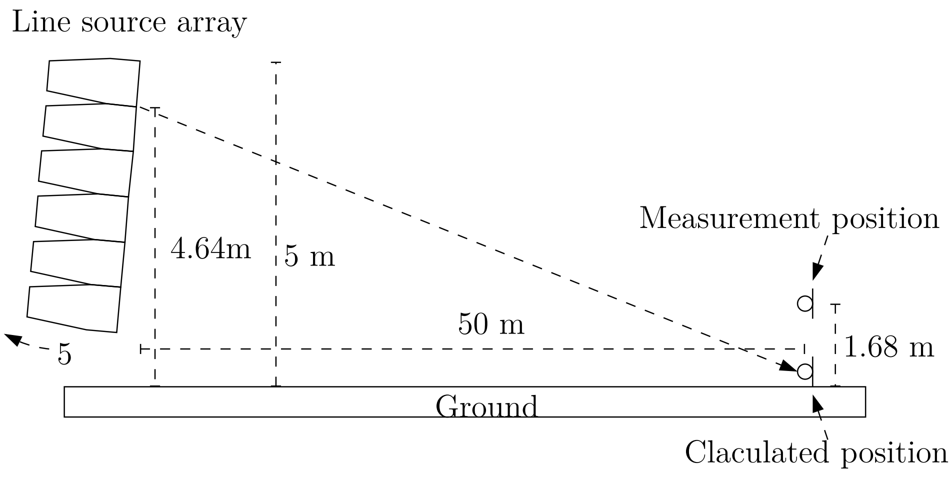


Figure 7.6: The figure shows the microphone position doing the calculation of the line source array tilting and the measurement

While using the tilt angle doing the measurement, which was 5°, the highest hight before the microphone exit above the nearfield of the speaker coverage main lobe is calculated to be 62 cm. Therefore, since the microphone was placed 1.68 m above the ground the microphone is fare above the nearfield main lobe.Comparing the Figure 7.1 and Figure 7.3 it is clearly seen that the microphone was outside the nearfield main lobe of the high frequency. Above 2.0 kHz the SPL is more than 10 dB SPL lower in the ear measuring hight compare to the ground position with the same distances to the speaker.

7.1.4 Measuring result

While all error and difficulties are described and disturb the measurement, the measurement indicates that raising the power in the main lobe also raising the power in the shadow zone. Comparing the microphone agents each other is difficult since the differences in the boost between the microphone as described in section 7.1.2. Therefore, to extract useful data, the frequency response on the same microphone is compared for 0° of tournament, 10° of tournament and 20° of tournament. The first microphone which is compared is the microphone in the upwards direction. The following Figure 7.7 shows the frequency response for every tournament.

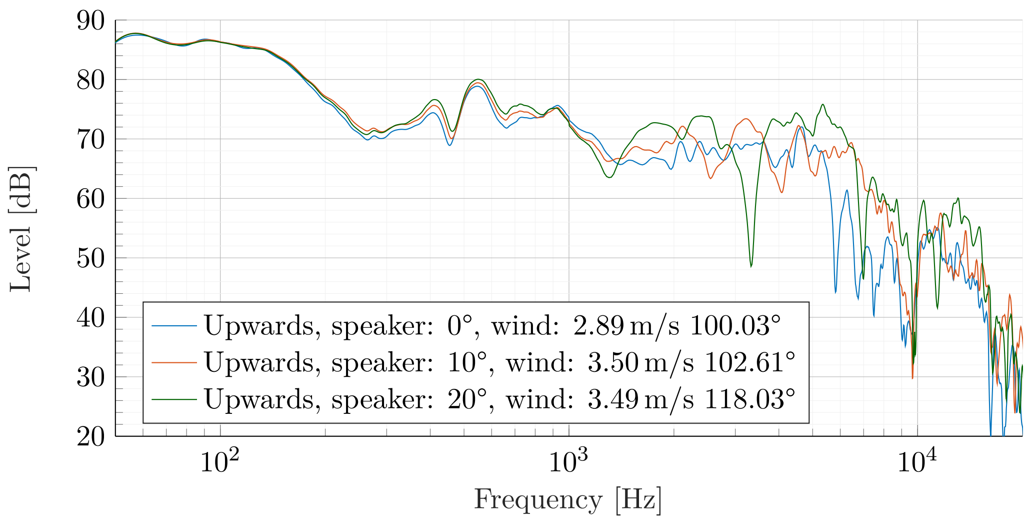


Figure 7.7: The graph shows the measuring result for the upwards microphone in three speaker angle, while the microphone is in the windscreen and the windscreen is in the hight of the ear. The graph is a mean of 10 measurement in all three angle. The mean calculation is done in time domain where the all impulse responses is allined with cross-correlation

As seen in Figure 7.7, while the speaker is turned, the SPL seems to raise. The peaks and depth are not at the same frequency which makes the visually evaluation difficult, but it visually shows that turning the speaker raises the SPL in some frequency area especially above 1.0 kHz. The following Table 7.1 shows the single number SPL both non weighted and A-weighted.

Table 7.1: The table shows the measured $L_{\mathrm{eq},5}$ and $L_{\mathrm{A_{eq},5}}$ SPL for the upwards microphone

Speaker angle	0°	10°	20°
$L_{\mathrm{eq},5}$	59.65 dB SPL	60.47 dB SPL	61.71 dB SPL
$L_{\mathrm{A_{eq},5}}$	56.91 dB SPL	58.20 dB SPL	60.28 dB SPL

The second microphone which is compared is the microphone in the downwards direction. The following Figure 7.8 shows the frequency response for every tournament.

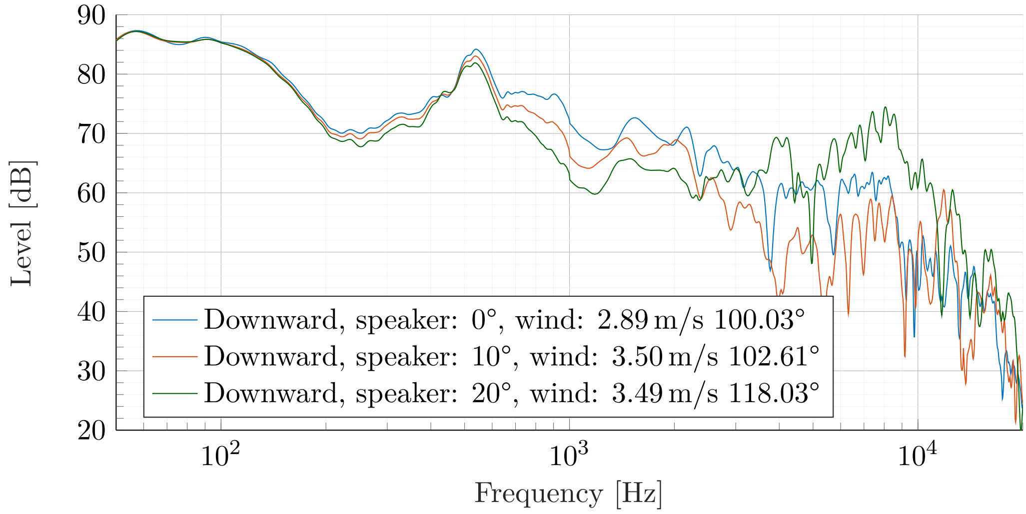


Figure 7.8: The graph shows the measuring result for the downwards microphone in three speaker angle, while the microphone is in the windscreen and the windscreen is in the hight of the ear. The graph is a mean of 10 measurement in all three angle. The mean calculation is done in time domain where the all impulse responses is allined with cross-correlation

As seen in Figure 7.8, while the speaker is turned, the SPL seems to fall unless the 20° above 2.5 kHz. The raise in power comes from the directionality charatstics of the line source array as seen in Figure 6.7. The peaks and depth is ether not at the same frequency which make the visually justment difficult but it generally that turning the speaker lower the SPL from 0° to 10° above 650 Hz. The following Table 7.2 shows the single number SPL both non weighted and A-weighted.

Table 7.2: The table shows the measured $L_{\mathrm{eq},5}$ and $L_{\mathrm{A}_{\mathrm{eq},5}}$ SPL for the downwards microphone

Speaker angle	0°	10°	20°
$L_{\mathrm{eq},5}$	59.87 dB SPL	58.47 dB SPL	60.13 dB SPL
$L_{\mathrm{A}_{\mathrm{eq},5}}$	57.25 dB SPL	54.60 dB SPL	57.37 dB SPL

The third microphone which is compared is the microphone in the center direction. The following Figure 7.9 shows the frequency response for every tournament.

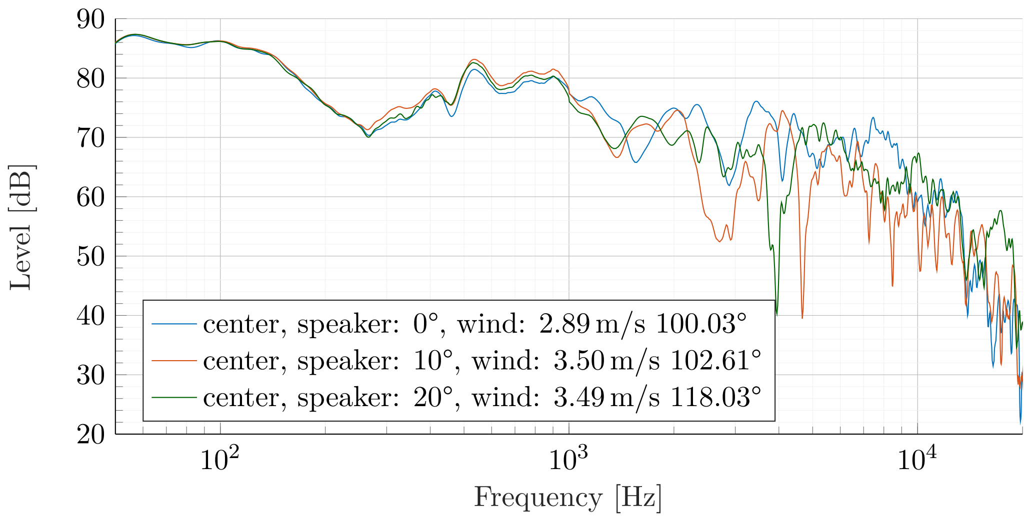


Figure 7.9: The graph shows the measuring result for the center microphone in three speaker angle, while the microphone is in the windscreen and the windscreen is in the hight of the ear. The graph is a mean of 10 measurement in all three angle. The mean calculation is done in time domain where the all impulse responses is allined with cross-correlation

As seen in Figure 7.9, the frequency response does not ether raise or fall markebly while the speaker is turned. The large depth between 2.0 kHz and 5.0 kHz comes from the frequency charatestic of the line source array as seen in Figure 6.7. The following Table 7.3 shows the single number SPL both non weighted and A-weighted.

Table 7.3: The table shows the measured $L_{\mathrm{eq},5}$ and $L_{\mathrm{A}_{\mathrm{eq},5}}$ SPL for the center microphone

Speaker angle	0°	10°	20°
$L_{\mathrm{eq},5}$	62.73 dB SPL	61.80 dB SPL	61.78 dB SPL
$L_{\mathrm{A}_{\mathrm{eq},5}}$	61.65 dB SPL	60.08 dB SPL	60.01 dB SPL

7.2 Update to the final test

Part III

Results

Chapter 8

Results

Chapter 9

Discussion and conclusion

9.1 Conclusion

Part IV

Appendix

Appendix A

cross wind effect on line source array

A measurement was made to measure the transfer function differences in two point in cross wind. The used speaker have a horizontal dispersion pattern of 100°.

Materials and setup

To measure the transfer function in a cross wind situation, the following materials are used:

Table A.1: Equipment list

Description	Model	Serial-no	AAU-no
PC	Macbook	W89242W966H	-
Audio interface	RME Fireface UCX	23811948	108230
Microphone	GRAS 26CC	78189	75583
Preamp	GRAS 40 AZ	100268	75551
Microphone	GRAS 26CC	78186	75582
Preamp	GRAS 40 AZ	100267	75550
dB technologies	DVA T4	-	-
Wind measurement tools	Drahtlose Wetterstation	-	2157-45
flying tools	-	-	-

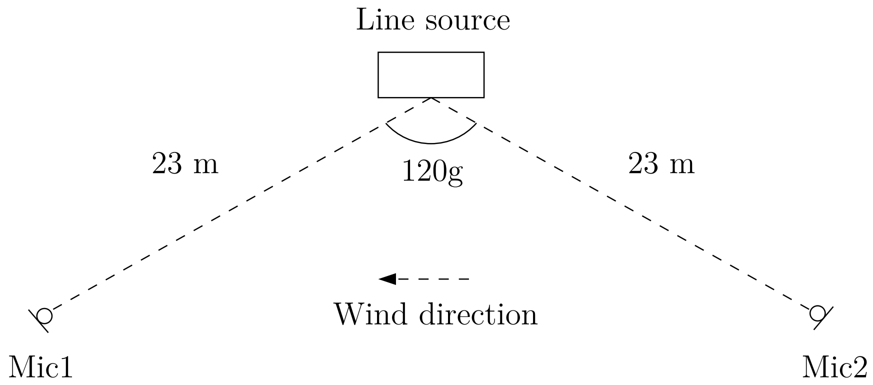


Figure A.1: The figure shows the microphone position versus the position of the line source

(a) The picture shows the speaker setup

(b) The figure shows the wind direction

Figure A.2: The figures shows the measurement set up for Appendix A and **??**

Test procedure

1. the microphone i calibrated.
2. The wind direction is measured.
3. The materials are set up as in Figure H.1 where the speaker is placed in cross wind direction, such that the frontal wave direction is orthogonal the the wind. The microphone and speaker is connected to the audio interface.
4. The speaker and microphone is placed 1.1 m above the ground
5. the wind direction goes from microphone 2 to microphone 1.
6. 10 sine sweep is performed with a length of 5 s each.
7. The impulse response is calculated and filtered with a 4th order highpass filter at 300 Hz to exclude wind noise.
8. The correlation is calculated for each impulse response to the first impulse response for time alignment [**?**] of both microphone channel.

9. The mean impulse response is calculated for the 10 measurement of both microphone.
10. The transfer function is calculated with a 40 sample moving mean filter.
11. The measurement is repeated three times.
12. The measurement i repeated with an angle of 74° and a distance of 25 m for both microphone.

Measurement area

To be able to measure in a windy area, the football stadium at Fredrick Alfred Nobels Vej 7, 9220 Aalborg is used. The following Figure A.3 shows a picture of the area and the approximate position of the speaker and microphone.

Figure A.3: The picture illustrate the area, where the wind flow is measured

Results

The wind speed was 14 m/s for each measurement and the temperature was 5°. The humidity was not measured.

The following measurement shows the result for 120°

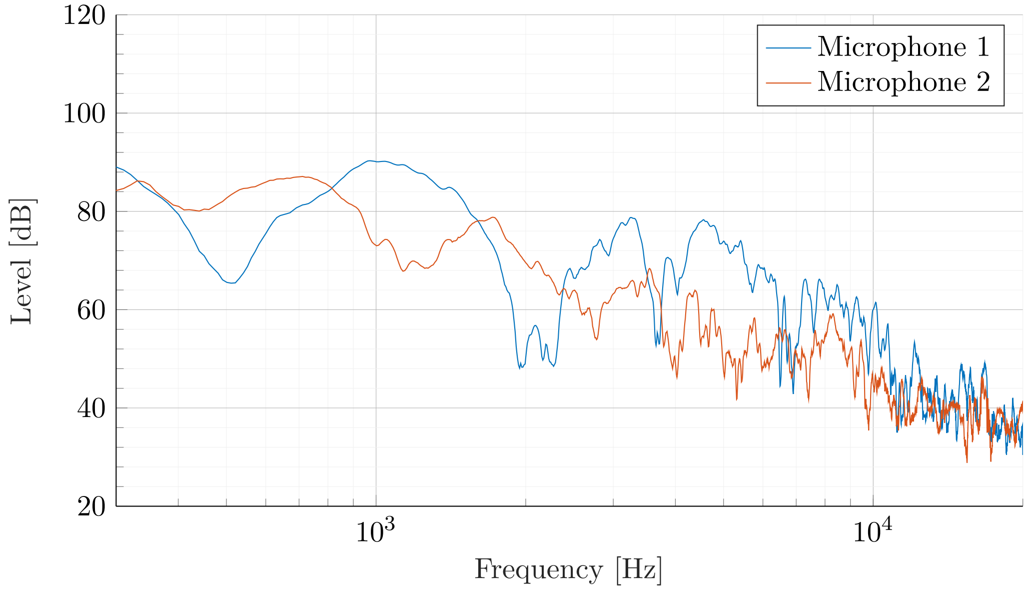


Figure A.4: The graph shows the first transfer function measurement. The $L_{\mathrm{eq},5}$ Sound Pressure Level (SPL) different between the microphones is 5.49 dB SPL (IR_6)

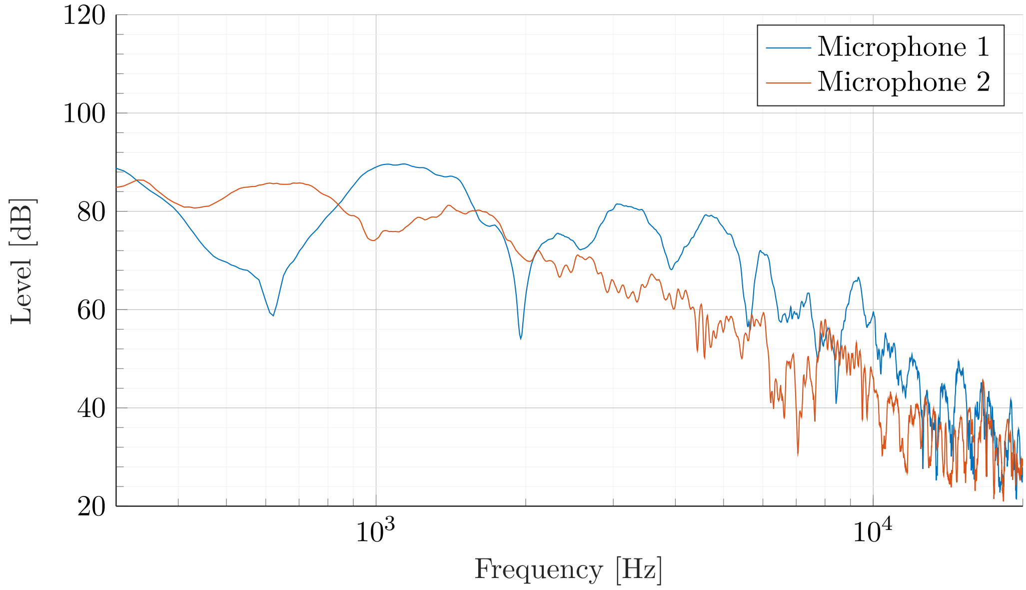


Figure A.5: The graph shows the second transfer function measurement. The $L_{\mathrm{eq},5}$ SPL different between the microphones is 4.40 dB SPL (IR_7)

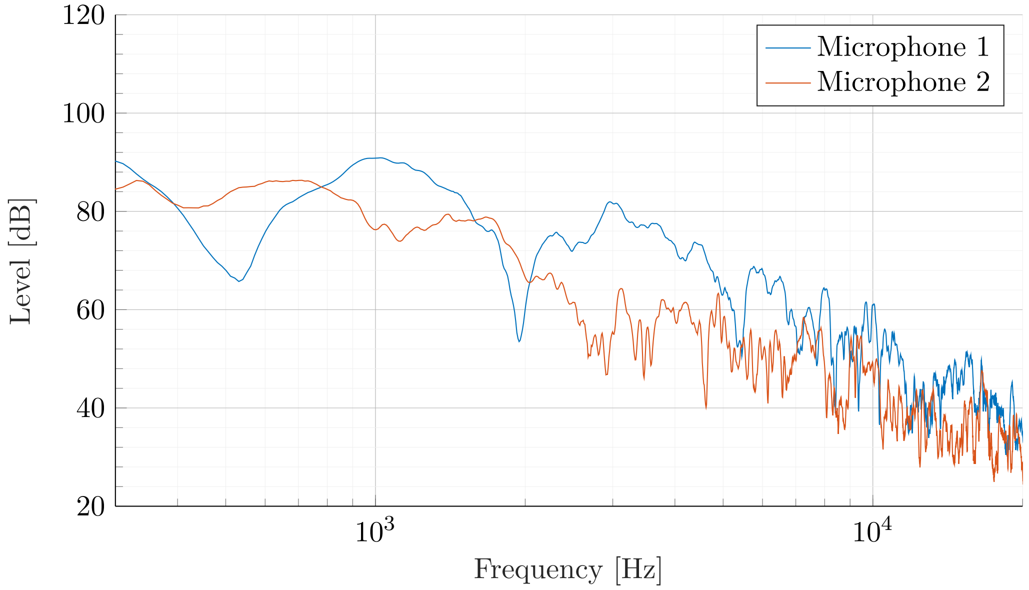


Figure A.6: The graph shows the third transfer function measurement. The $L_{\mathrm{eq},5}$ SPL different between the microphones is 4.23 dB SPL (IR_8)

On Figure A.4, Figure A.5 and Figure A.6 it is seen that the general pressure is higher for microphone 1. It is also seen that ground reflection effect is much higher on microphone 1 than microphone 2. This support the theory about upwards refraction of sound wave on microphone 2 and downwards refraction on microphone 1

The following measurement shows the result for 74°

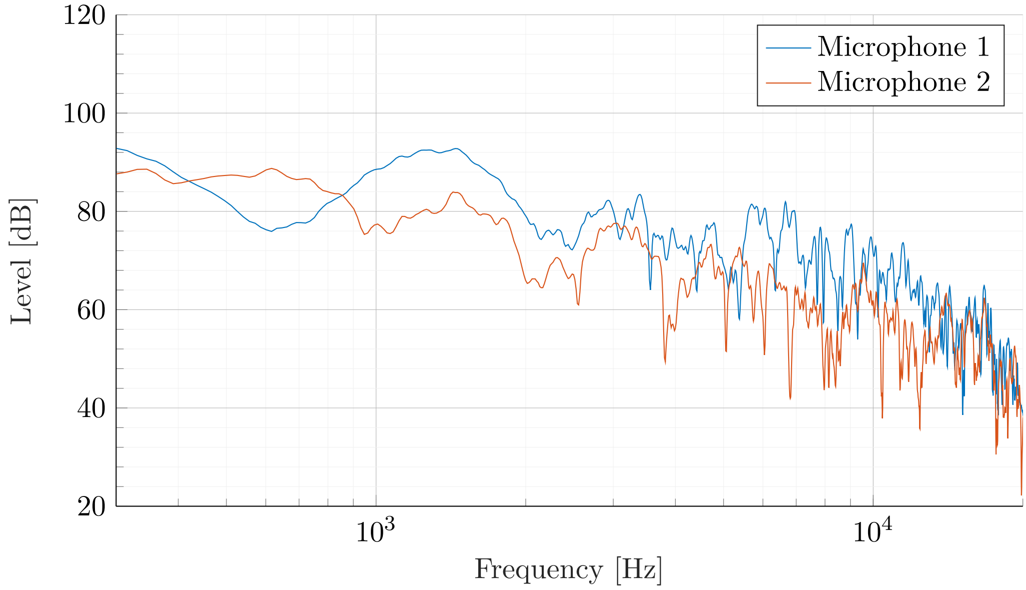


Figure A.7: The graph shows the first transfer function measurement. The $L_{\mathrm{eq},5}$ SPL different between the microphones is 4.41 dB SPL (IR_3)

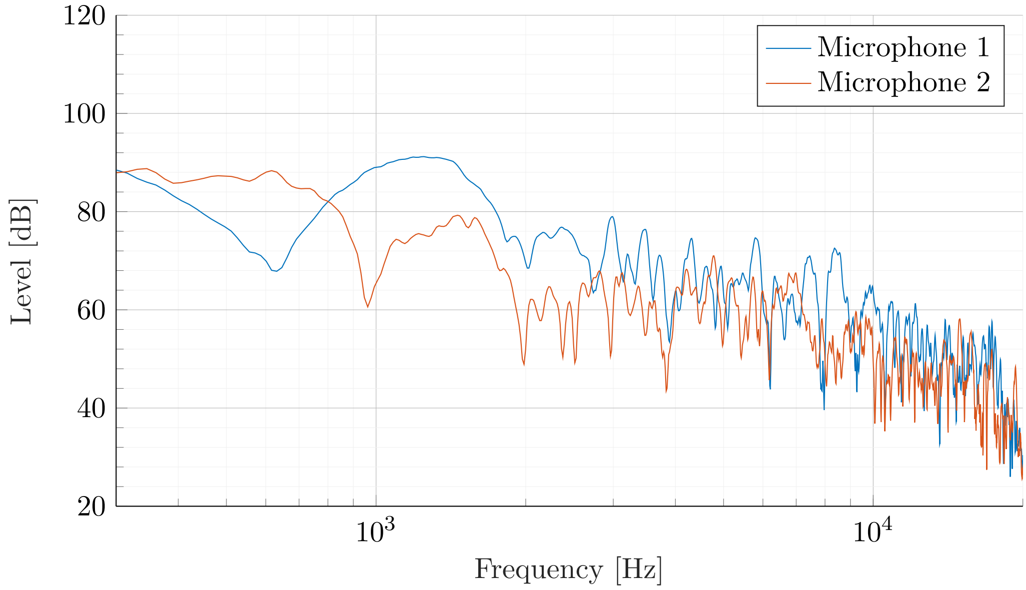


Figure A.8: The graph shows the second transfer function measurement. The $L_{\mathrm{eq},5}$ SPL different between the microphones is 4.81 dB SPL (IR_5)

On Figure A.7 and Figure A.8 it is seen that the general pressure is higher for microphone 1. It is also seen that ground reflection effect is much higher on microphone 1 than microphone 2. This support the theory about upwards refraction of sound wave on microphone 2 and downwards refraction on microphone 1

Summary

Appendix B

Windscreen concept measurement

A measurement was made to measure the wind attenuation of difference windscreen configuration. All configuration include the GRAS AM0069 windscreen with an additionally wind stopper surface all around the microphone except of the frontal direction. The measurement is done as a preliminary test with low wind speed, to test the concept before a field test with wind speed as in the speaker measurement. The measurement is done to ensure that the measured wind does not overload the preamp of the microphone at the specified wind speed.

Materials and setup

To measure the wind attenuation of the windscreen configuration the following materials are used:

Table B.1: Equipment list

Description	Model	Serial-no	AAU-no
PC	Macbook	W89242W966H	-
Audio interface	RME Fireface UCX	23811948	108230
Microphone	GRAS 26CC	78189	75583
Preamp	GRAS 40 AZ	100268	75551
Microphone	GRAS 26CC	78186	75582
Preamp	GRAS 40 AZ	100267	75550

Table B.2: Equipment list

Description	Model	Serial-no	AAU-no
Fan	IMPEGA	-	-
Fan	IMPEGA	-	-
Windscreen	GRAS AM0069	-	-
Windscreen	GRAS AM0069	-	-
large foam wedge	-	-	-
Small foam wedge	-	-	-
Rockwool bat	-	-	-

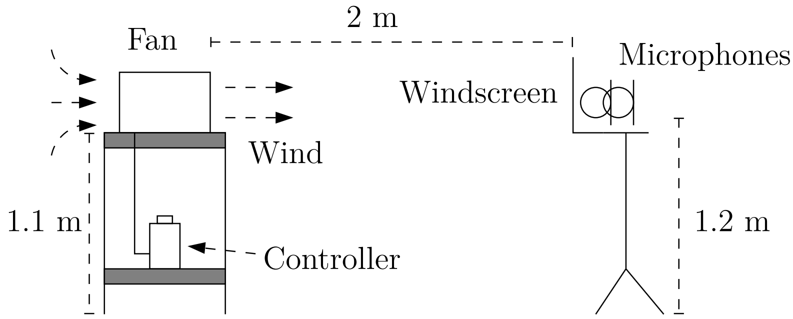


Figure B.1: The figure shows the measurement setup for the wind noise measurement in the microphone position and outside the windscreen. The two microphone seems to lay onto each other but it shall show that one is inside the windscreen and one is outside the windscreen

Figure B.2: The picture shows the measurement set up

Test procedure

1. The materials are set up as in Figure B.1 where the two microphone connected to the audio interface.

2. Both microphone is calibrated.
3. Both fan is activated
4. A 7 s time signal is measured three times synchronise on both microphone.
5. The frequency content is calculated by `fft` on all six measured time signal.
6. The average of the frequency response for each microphone is calculated
7. The difference between the microphone is calculated to find the attenuation of the windscreen configuration
8. The procedure is done for all windscreen configuration and one where no additionally wind stopper is added around the microphone. This last configuration is defined as reference configuration.
9. A no wind measurement is measured the same way just without the fan activated and only with GRAS AM0069 windscreen in the end.
10. The wind speed is measured.

Measurement area

To be able to generate a controlled wind flow, the hall way in Fredrick Bajers vej 7B5, 9200 Aalborg is used. The following Figure B.3 shows a drawing of the area ans the position of the fan and windscreen.

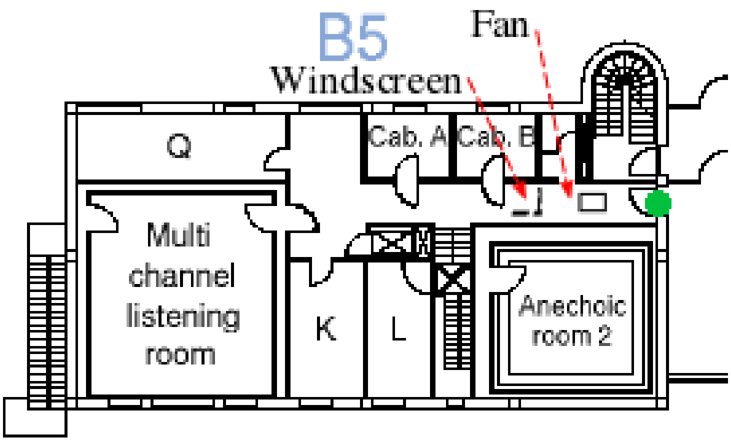


Figure B.3: The picture illustrate the area, where the wind flow is measured

Results

The following graphs shows the result of the measurement. The wind speed is measured to be 2.5 m/s.

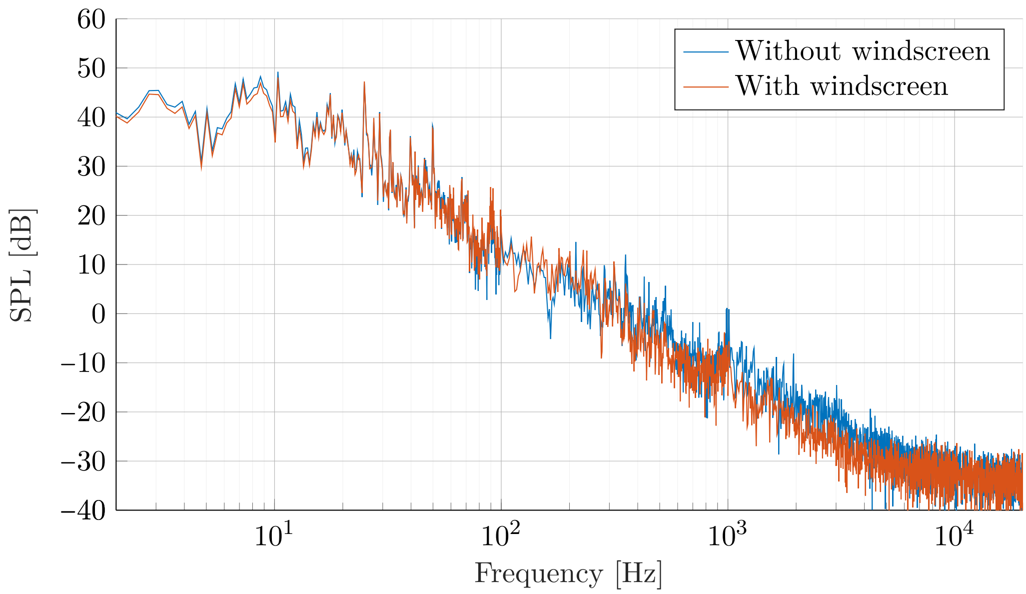


Figure B.4: The graph shows the frequency content without the fan activated

The Figure B.4 shows the frequency content in the measuring area without the fan activated for both microphone and the reference windscreen configuration.

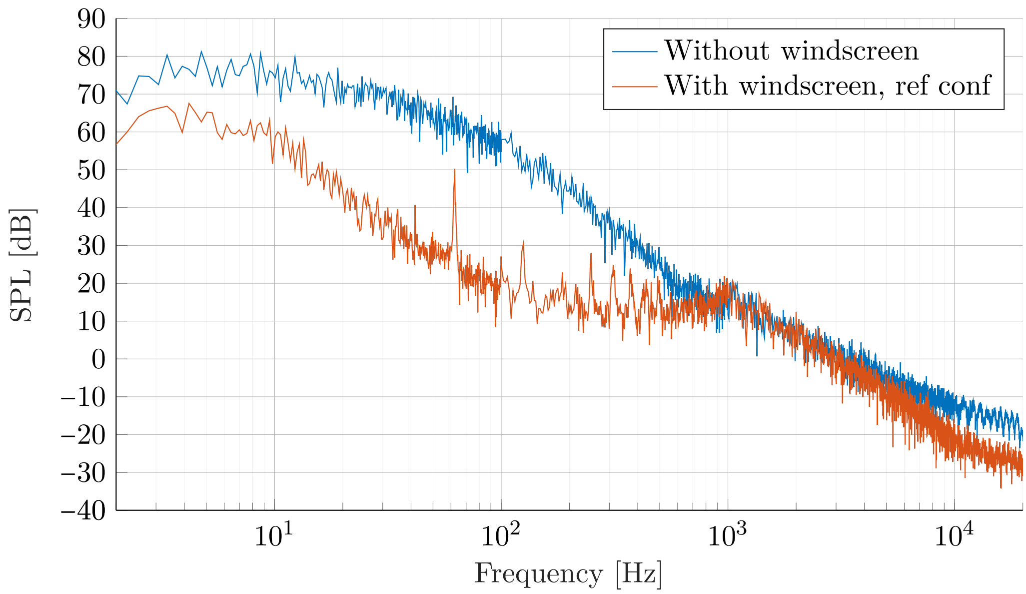


Figure B.5: The graph shows the frequency content with the fan activated

The Figure B.5 shows the frequency content in the measuring area with the fan activated for both microphone and the reference windscreen configuration. It is seen that the highest attenuation is at 900 Hz but the general attenuation is

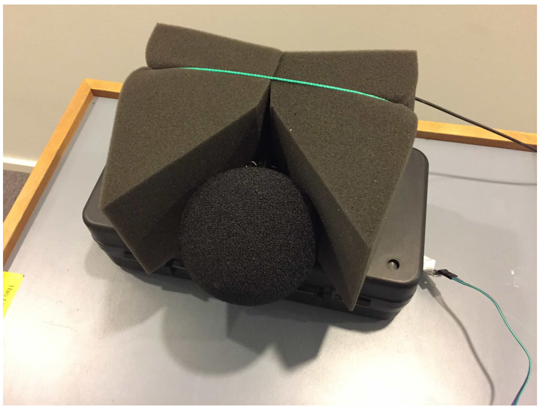


Figure B.6: The picture shows the microphone covered with windscreen and the Small foam wedge windscreen configuration. This configuration is defined as configuration one

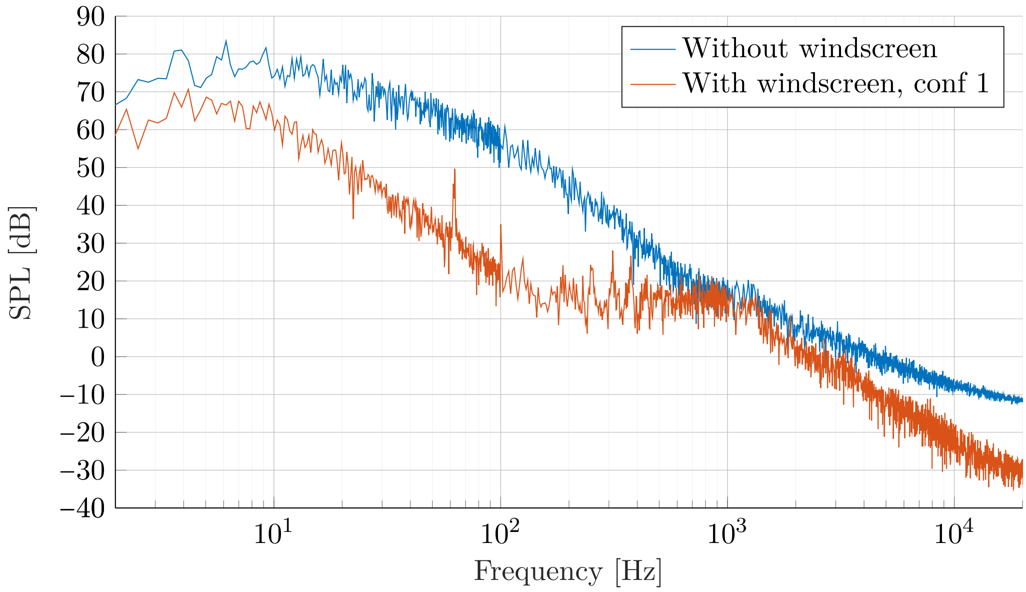


Figure B.7: The graph shows the frequency content of the measurement with configuration one

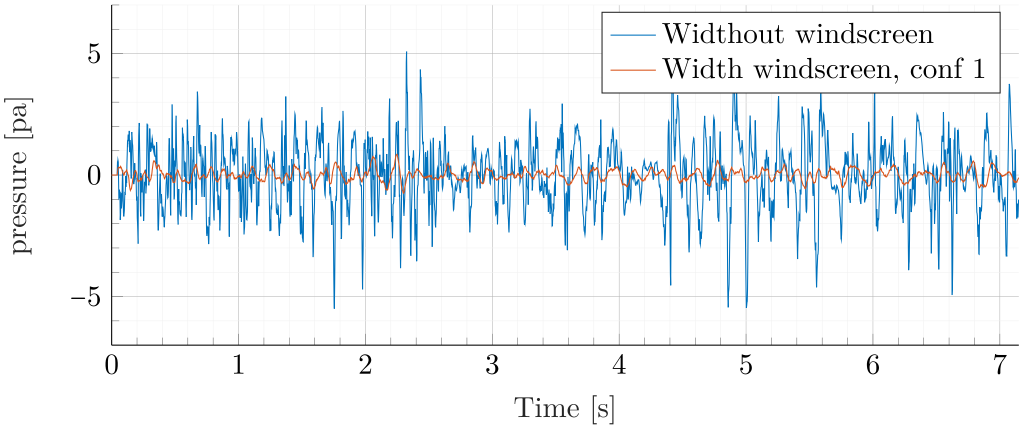


Figure B.8: The graph shows one of the time measurement with configuration one

The Figure B.7 and Figure B.8 shows the measurement with configuration one in frequency and time domain respectively. It can be seen that the general windscreen attenuation is not lowered compared to the reference windscreen measurement.

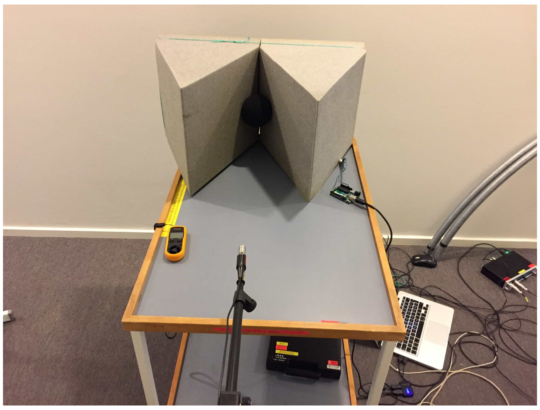


Figure B.9: The picture shows the microphone covered with windscreen and the large foam wedge windscreen. This configuration is defined as configuration two.

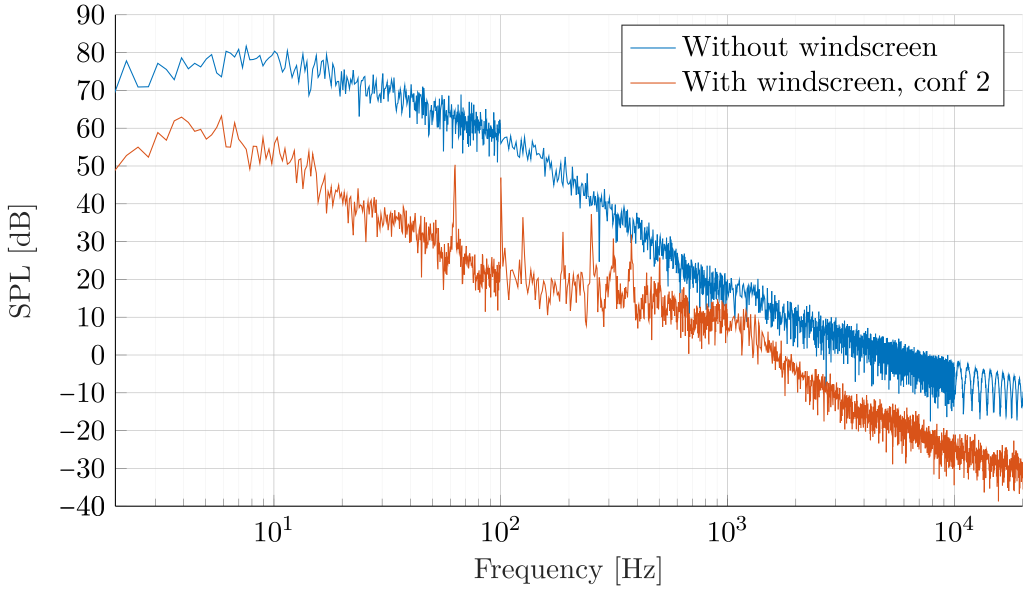


Figure B.10: The graph shows the frequency content of the measurement with configuration two

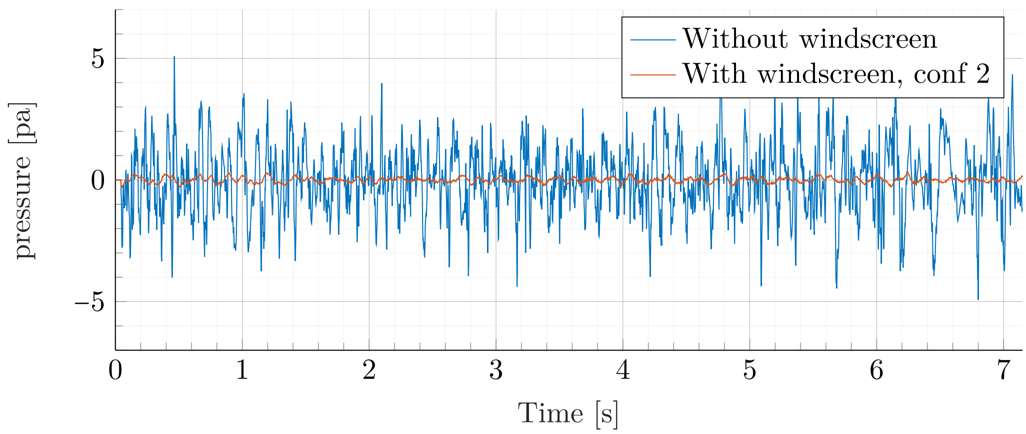


Figure B.11: The graph shows one of the time measurement with configuration two

The Figure B.10 and Figure B.11 shows the measurement with configuration two in frequency and time domain respectively. The measurement shows that the windscreen attenuation does have an effect compare to the reference windscreen measurement. The attenuation is nearly greater for all frequency especially in the low and high frequency range.

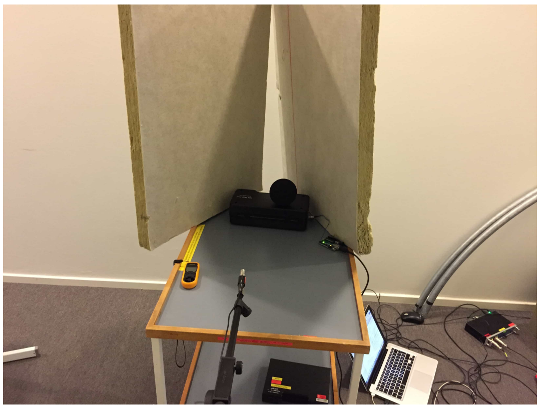


Figure B.12: The picture shows the microphone covered with windscreen and the rockwool wind stopper. This configuration is defined as configuration three.

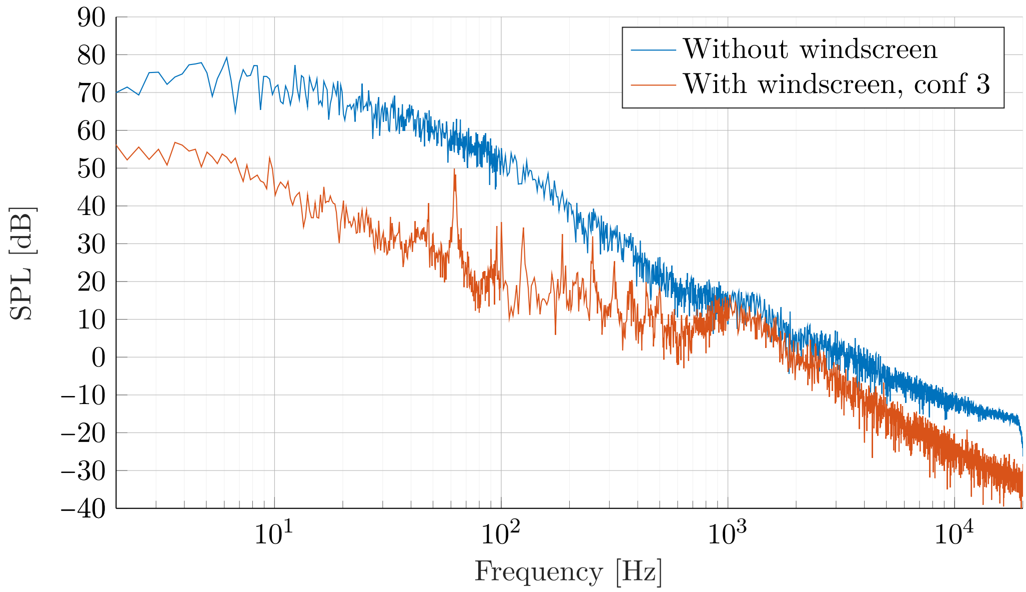


Figure B.13: The graph shows the frequency content of the measurement with configuration three

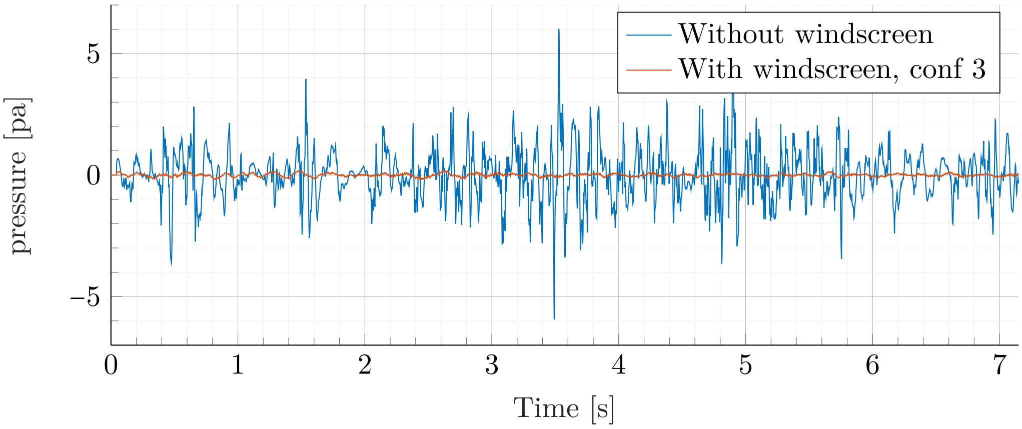


Figure B.14: The graph shows one of the time measurement with configuration three

The Figure B.13 and Figure B.14 shows the measurement with configuration three in frequency and time domain respectively. The measurement shows that the windscreen attenuation does have an effect compare to the reference windscreen measurement but the attenuation is not as good as in configuration two. There is better attenuation is the low frequency compare to the reference windscreen measurement but in the high frequency the attenuation i worse than the reference windscreen measurement, that might be bue to reflection on the surface of the rockwool filt.

Summary

It is shown that the additionally wind stopper hypotheses does attenuate the wind noise at a wind speed of 2.5 m/s

Appendix C

Windscreen attenuation measurement

A measurement was made to measure the frequency attenuation of difference windscreen configuration. All configuration include the GRAS AM0069 windscreen with an additionally wind stopper surface all around the microphone except of the frontal direction. The measurement is done in the anechoic chamber. The measurement is done to analyse the effect of the windscreen in the frequency domain to ensure that the chosen windscreen do not add to much reflect to the measurement. The optimal criteria is therefore as low difference as possible.

Materials and setup

To measure the frequency response of the windscreen configuration the following materials are used:

Table C.1: Equipment list

Description	Model	Serial-no	AAU-no
PC	Macbook	W89242W966H	-
Audio interface	RME Fireface UCX	23811948	108230
Microphone	GRAS 26CC	78189	75583
Preamp	GRAS 40 AZ	100268	75551
Windscreen	GRAS AM0069	-	-
large foam wedge	-	-	-
Small foam wedge	-	-	-
Rockwool bat	-	-	-

Table C.2: Equipment list

Description	Model	Serial-no	AAU-no
Speaker stand	-	-	-
Speaker stand	-	-	-
Speaker	B&W	03508438	1441-0

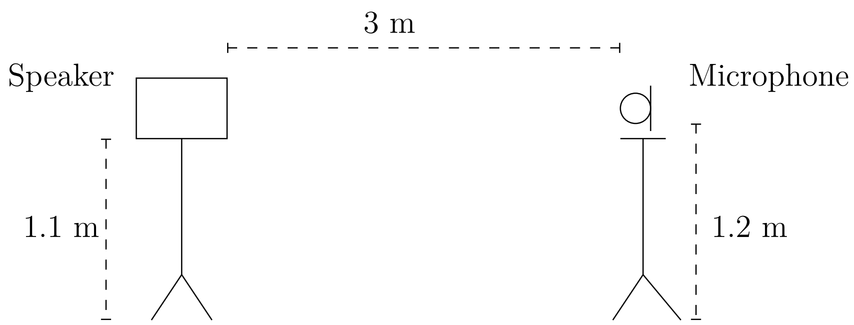


Figure C.1: The figure shows the measurement setup in the anechoic chamber

Test procedure

1. The materials are set up as in Figure C.1 where the microphone is connected to the audio interface.
2. The microphone is calibrated
3. The speaker is placed 3 m from the microphone and pointing in the direction of the microphone.
4. The windscreen configuration is placed such that the microphone have approximately the same position as without the windscreen.
5. The transfer function is measured
6. The procedure is started over until all windscreen is measured.
7. The transfer function is calculated and plotted versus the transfer function without windscreen MATLAB®.
8. The difference between the transfer function wit and without windscreen is calculated and plotted in MATLAB®.

Measurement area

To be able to measure the windscreen frequency response, the anechoic chamber in Fredrick Bajers vej 7B4, 9220 Aalborg is used. The following Figure C.2 shows a drawing of the area ans the position of the fan and windscreen.

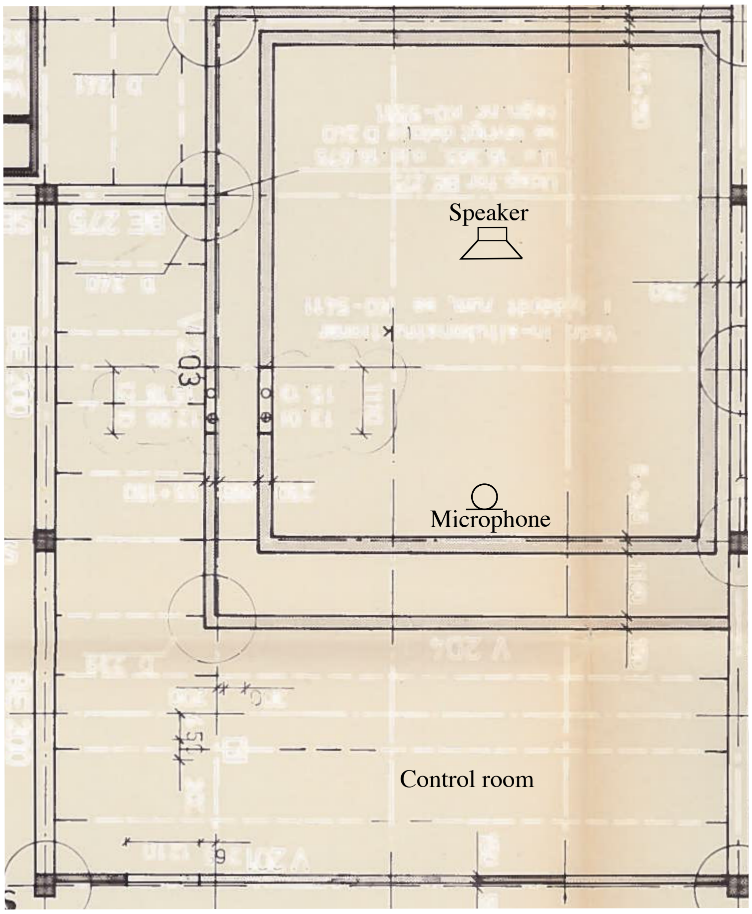


Figure C.2: The picture illustrate the area, where the wind flow is measured

Results

The following Figure C.3 shows the speaker.

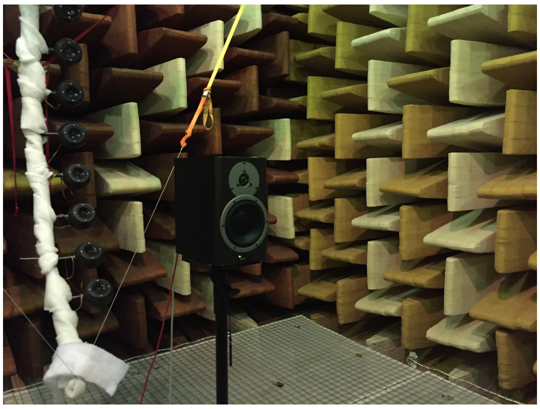


Figure C.3: The picture shows the used speaker

The following graphs shows the result of the measurement.

Figure C.4: The picture shows the measurement microphone with the original windscreen

The measurement shown in Figure C.5 shows frequency response of the speaker with and without the windscreen. The Figure C.4 shows the microphone position with the windscreen. The position is not changed for the measurement without windscreen.

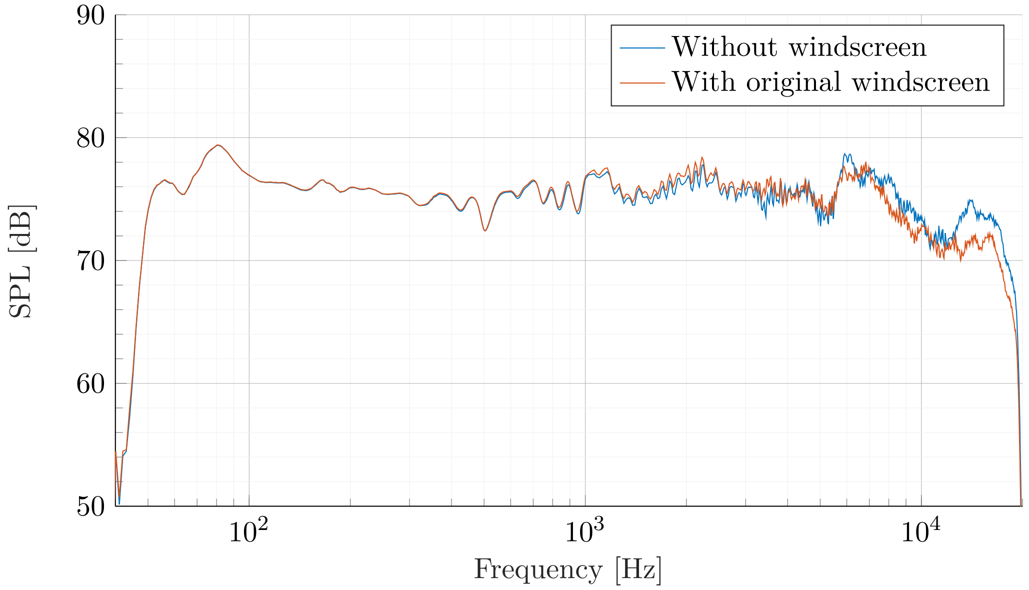


Figure C.5: The graph shows frequency response of the speaker measured without windscreen and with the original windscreen

The measurement shown in Figure C.6 shows the difference SPL between the measurement with and without original windscreen.

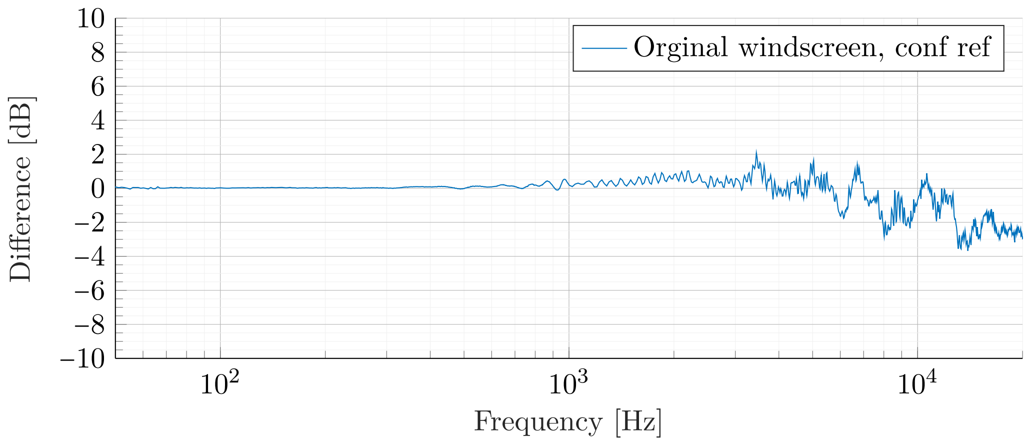


Figure C.6: The graph shows frequency response of the windscreen, where the frequency characteristic for the speaker is out calculated.

Figure C.7: The picture shows the measurement microphone with the large foam wedge configuration two

The measurement shown in Figure C.8 shows frequency response of the speaker without the windscreen and with the windscreen configuration one. The Figure C.7 shows the measured set up.

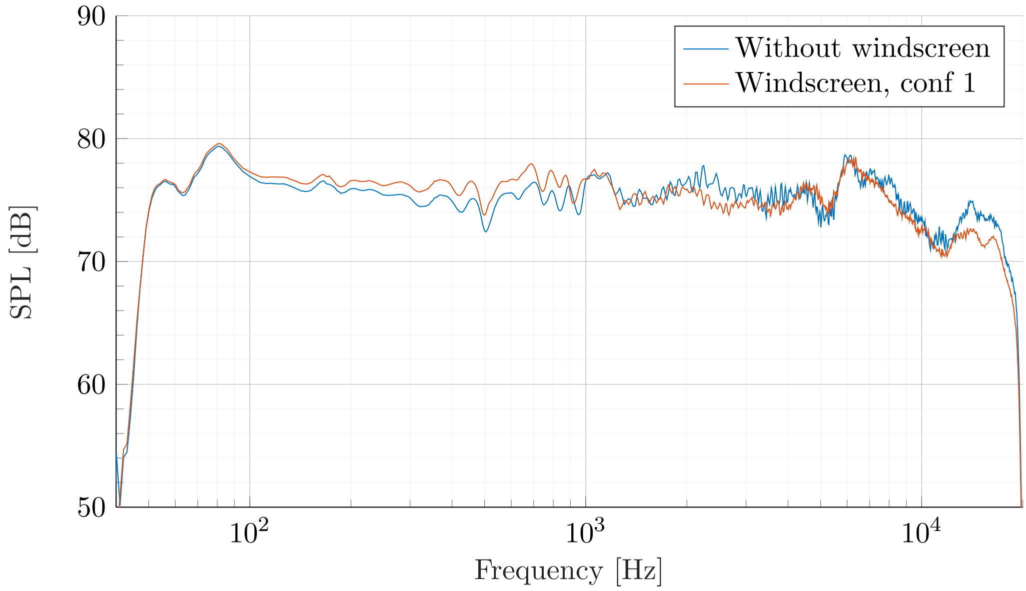


Figure C.8: The graph shows frequency response of the speaker measured without windscreen and with windscreen configuration one

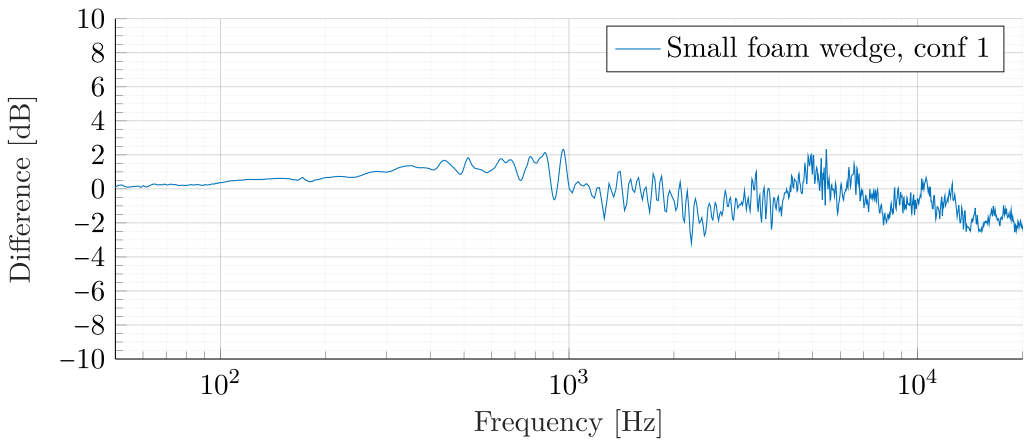


Figure C.9: The graph shows frequency response of the windscreen, where the frequency characteristic for the speaker is out calculated.

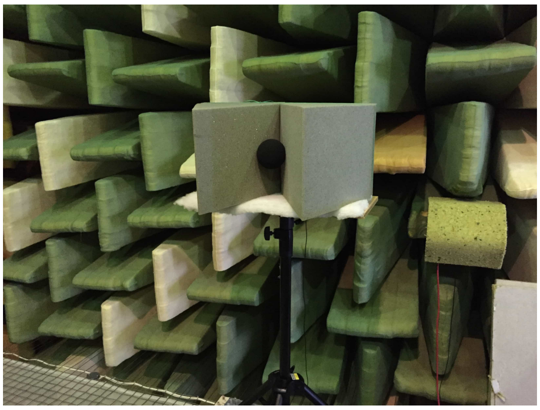


Figure C.10: The picture shows the measurement microphone with the large foam wedge configuration two

The measurement shown in Figure C.11 shows frequency response of the speaker without the windscreen and with the windscreen configuration two. The Figure C.10 shows the measured set up.

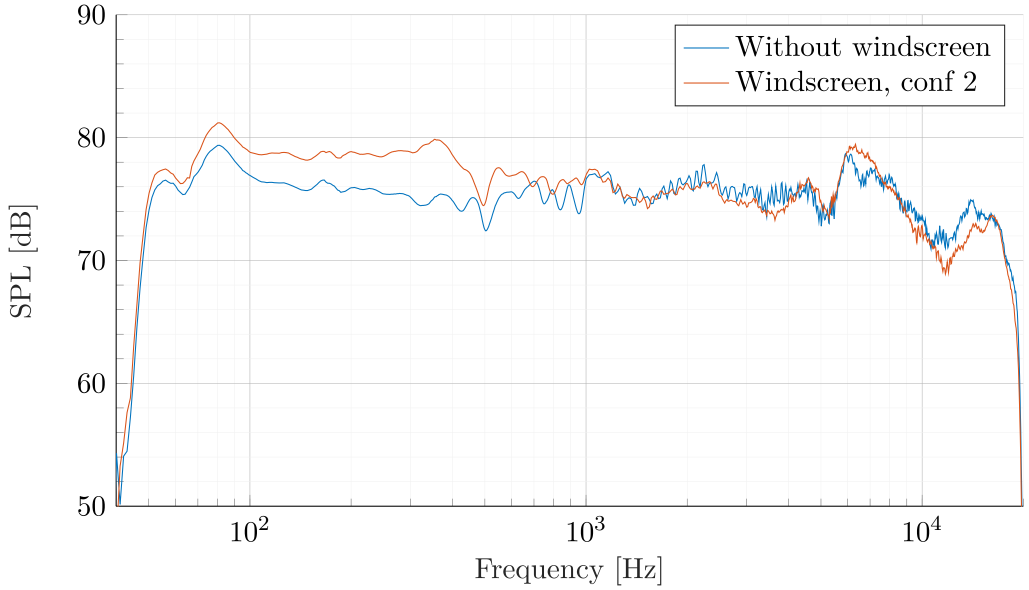


Figure C.11: The graph shows frequency response of the speaker measured without windscreen and with windscreen configuration two

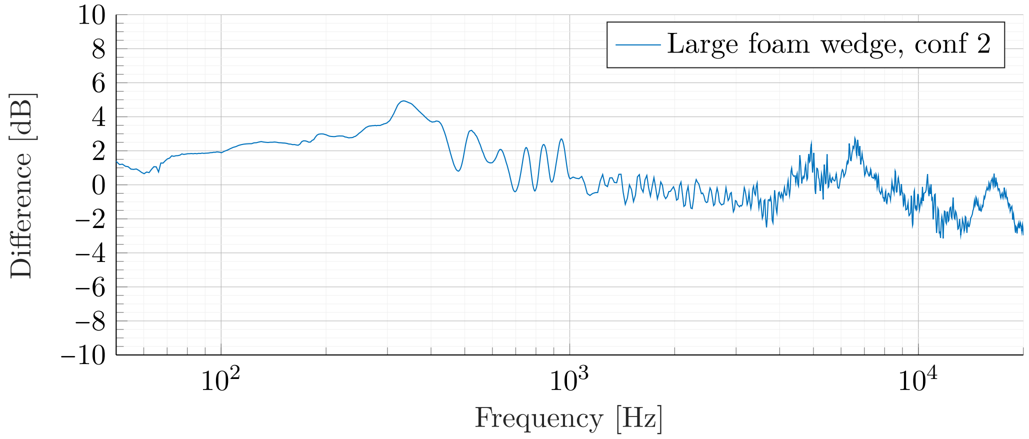


Figure C.12: The graph shows frequency response of the windscreen, where the frequency characteristic for the speaker is out calculated.

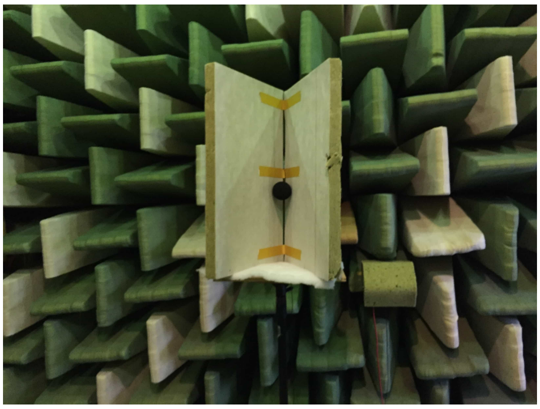


Figure C.13: The picture shows the measurement microphone with the rockwool bat configuration three

The measurement shown in Figure C.14 shows frequency response of the speaker without the windscreen and with the windscreen configuration three. The Figure C.13 shows the measured set up. In this measurement both the speaker and the microphone is lifted 20 cm

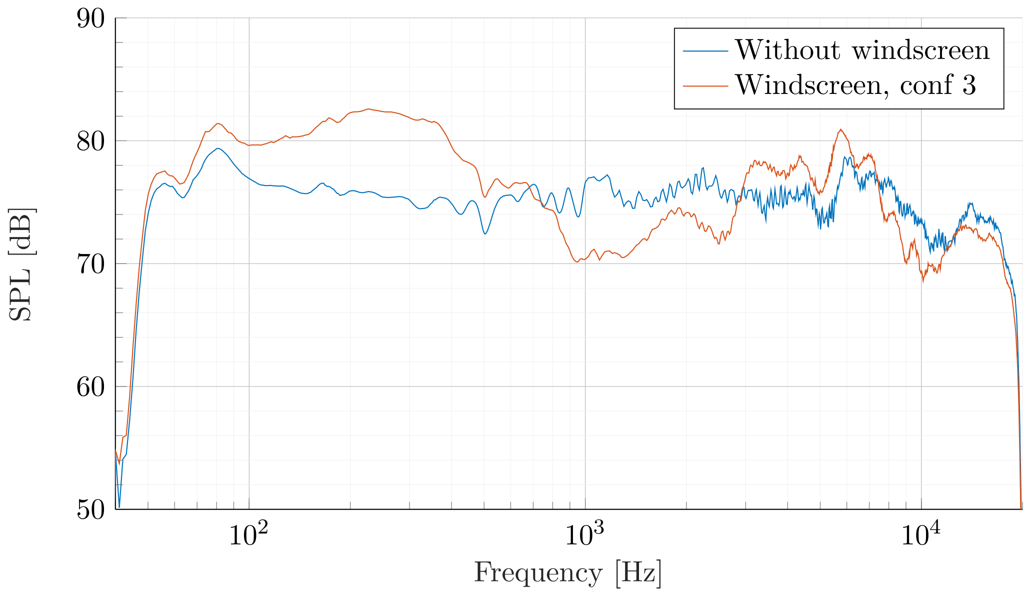


Figure C.14: The graph shows frequency response of the speaker measured without windscreen and with windscreen configuration three

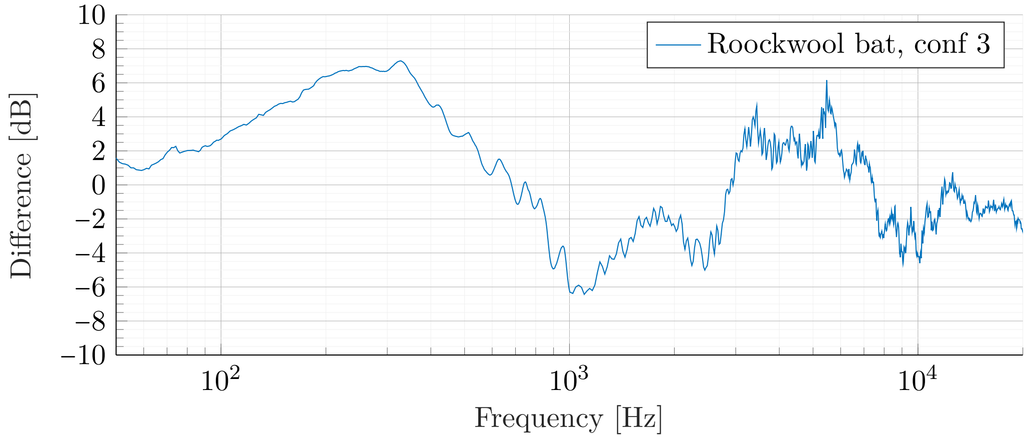


Figure C.15: The graph shows frequency response of the windscreen, where the frequency characteristic for the speaker is out calculated.

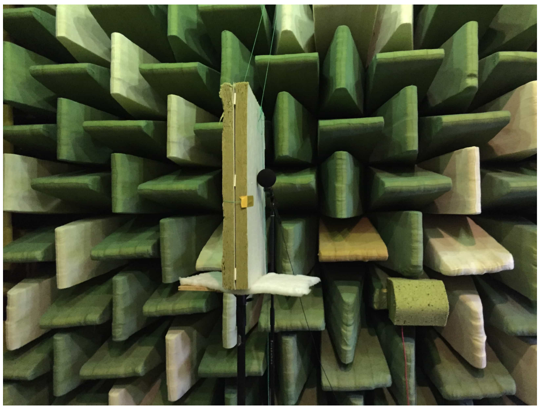


Figure C.16: The picture shows the measurement microphone with the rockwool bat configuration four

The measurement shown in Figure C.17 shows frequency response of the speaker without the windscreen and with the windscreen configuration four. The Figure C.16 shows the measured set up. In this measurement both the speaker and the microphone is lifted 30 cm

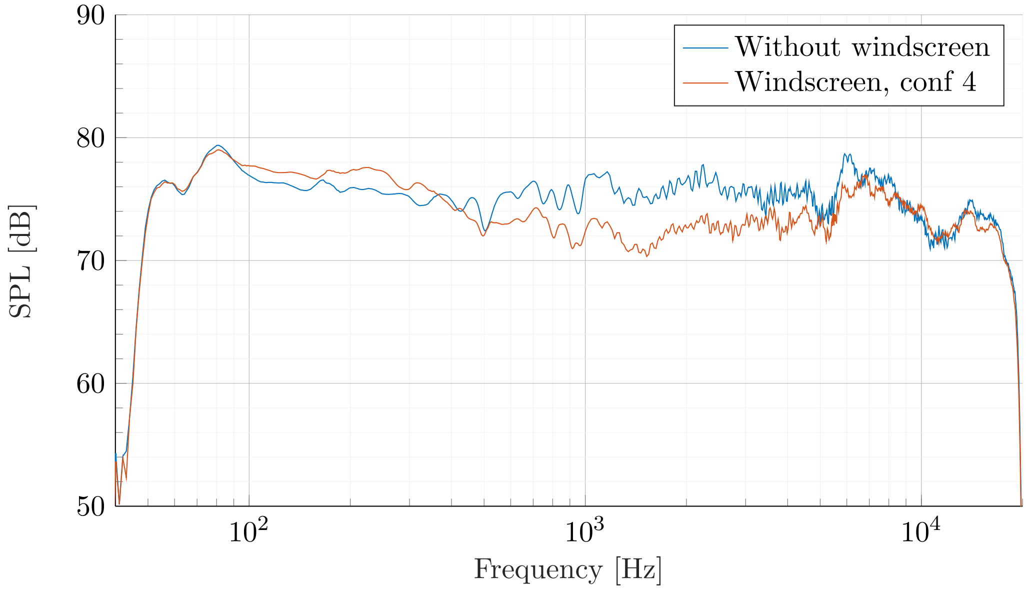


Figure C.17: The graph shows frequency response of the speaker measured without windscreen and with windscreen configuration four

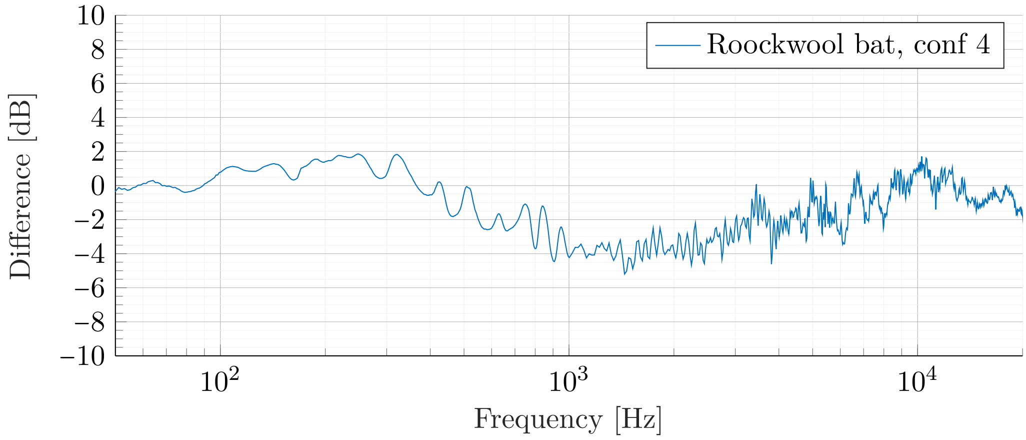


Figure C.18: The graph shows frequency response of the windscreen, where the frequency characteristic for the speaker is out calculated.

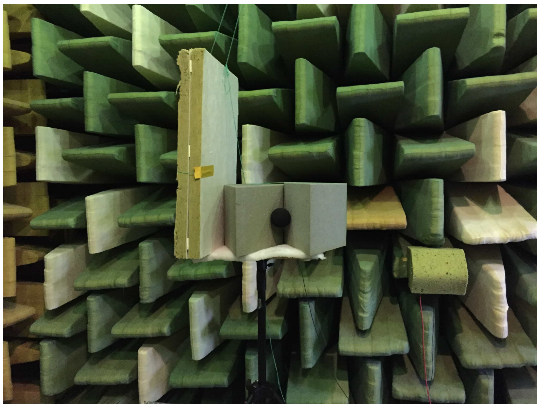


Figure C.19: The picture shows the measurement microphone with the rockwool bat configuration four

The measurement shown in Figure C.20 shows frequency response of the speaker without the windscreen and with the windscreen configuration four. The Figure C.19 shows the measured set up.

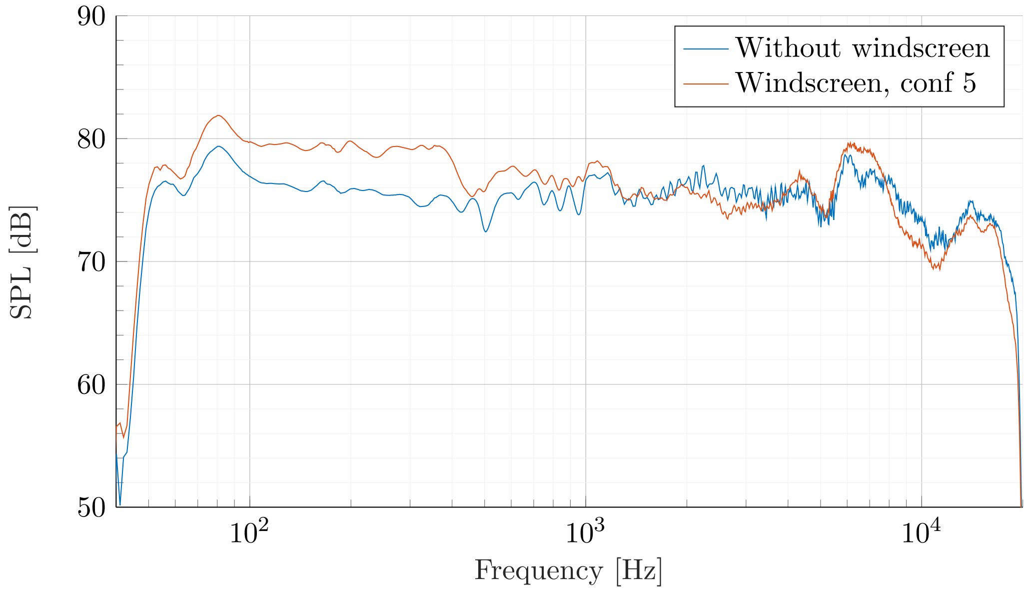


Figure C.20: The graph shows frequency response of the speaker measured without windscreen and with windscreen configuration five

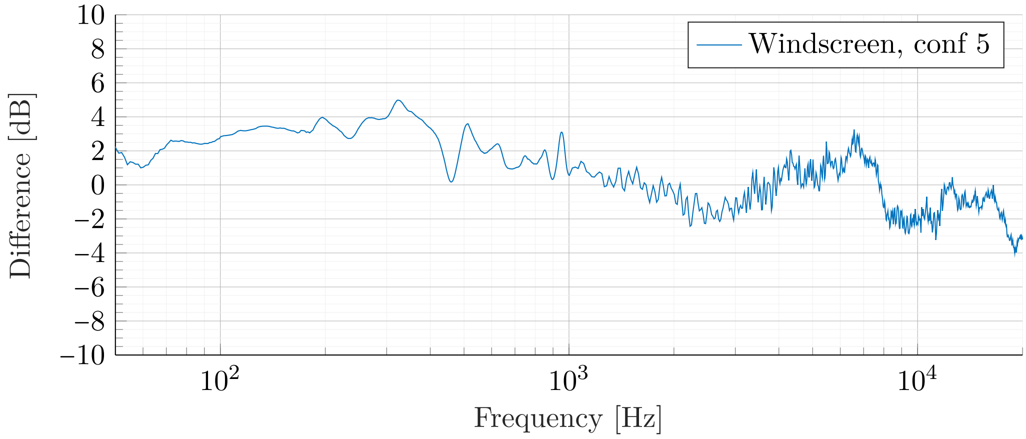


Figure C.21: The graph shows frequency response of the windscreen, where the frequency characteristic for the speaker is out calculated.

Summary

It is seen in the graph that all windscreen configuration change the frequency response of the microphone, even with only the original windscreen. Without taking the reference configuration intro account, the one with the lowest frequency change in the low frequency response is configuration one and four. The one with the lowest frequency change in the middle frequency is configuration one and two. The one with lowest frequency change in the high frequency is configuration four. The mix between configuration two and four shows a low frequency change in the middle and high frequency.

Appendix D

Windscreen attenuation measurement

A measurement was made to measure the wind attenuation in the microphone position with the difference windscreen configuration.

Materials and setup

To measure the wind attenuation of the windscreen configuration the following materials are used:

Table D.1: Equipment list

Description	Model	Serial-no	AAU-no
PC	Macbook	W89242W966H	-
large foam wedge	-	-	-
Small foam wedge	-	-	-
Rockwool bat	-	-	-
Fast fan	-	-	-
Fan control	tranformator	-	60398
Windspeed tools	FT technologies FT742-D-SM	9002-922	105634

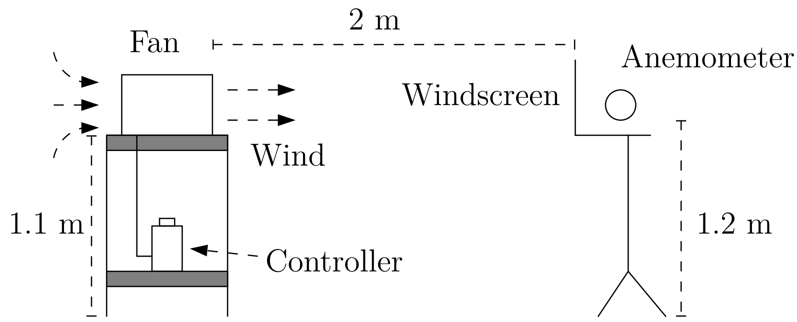


Figure D.1: The figure shows the measurement setup for the wind speed measurement in the microphone position

Test procedure

1. The materials are set up as in Figure D.1.
2. The fan is placed such that is produce directly crosswind.
3. The fan is activated
4. The anemometer is activated to record.
5. The anemometer is placed in the wind for approximately 10 s.
6. Then the anemometer is moved intro the microphone position for approximately 10 s.
7. Then the anemometer is moved back in the wind for approximately 10 s.
8. Then the anemometer is moved intro the microphone position for approximately 10 s again.
9. The wind speed measurement is plotted in MATLAB®with a moving mean filter with 2 sample and as m/s versus s.
10. Next the wind direction measurement is plotted in MATLAB®with a moving mean filter with 2 sample and as ° versus s.
11. The measurement is done for all windscreen configuration the same way.

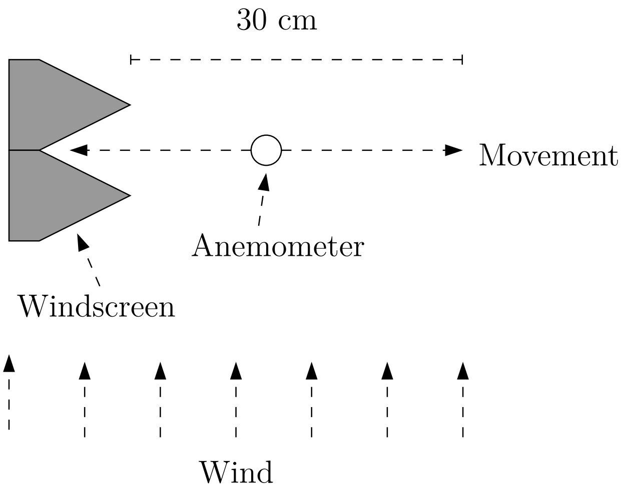


Figure D.2: The figure shows the movement of the anemometer doing the measurement

The following Figure D.3 shows the anemometer used for the measurement.

Figure D.3: The picture shows anemometer used for the measurement

Measurement area

To be able to generate a controlled wind flow, the hall way in Fredrick Bajers vej 7B5, 9220 Aalborg is used. The following Figure B.3 shows a drawing of the area ans the position of the fan and windscreen.

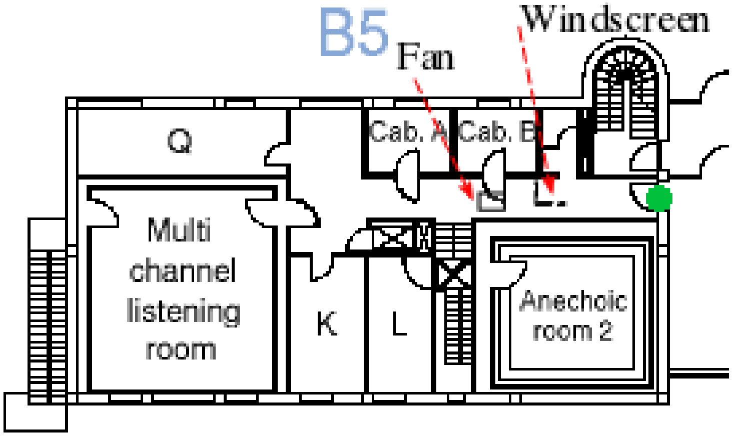


Figure D.4: The picture illustrate the area, where the wind flow is measured

Results

The following graphs shows the result of the measurement.

Figure D.5 shows the measurement setup of the foam wedge, where the Figure D.6 shows the result.

Figure D.5: The picture shows the measurement setup with the small wedge, configuration one

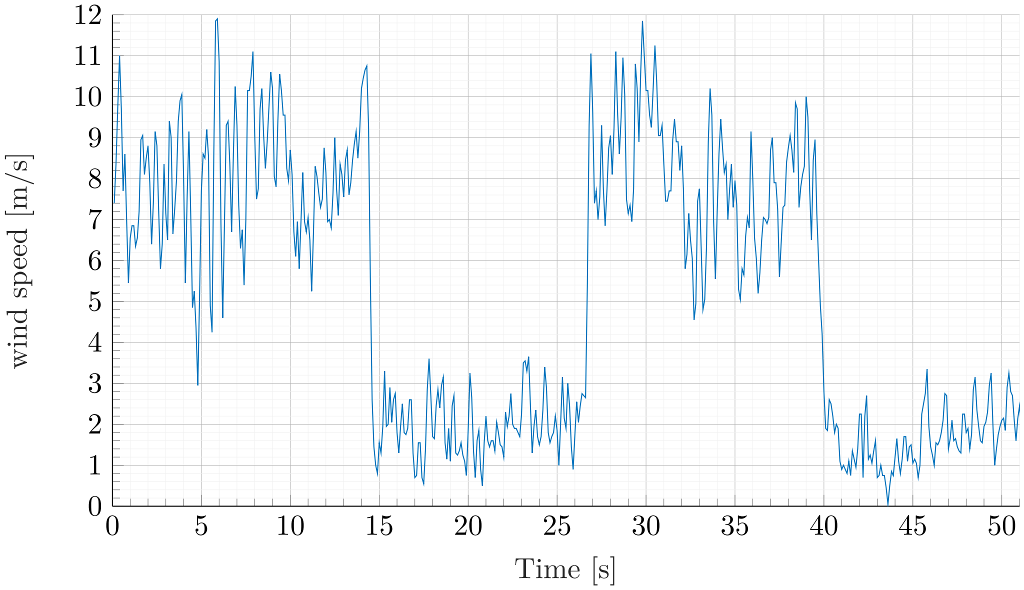


Figure D.6: The graph shows the wind speed versus time for configuration one. The grape have a high speed period and a low speed period, in the high speed period, the anemometer is in the wind approximatly 30 cm from the windscreen where in the low speed period, the anemometer is inside the windscreen.

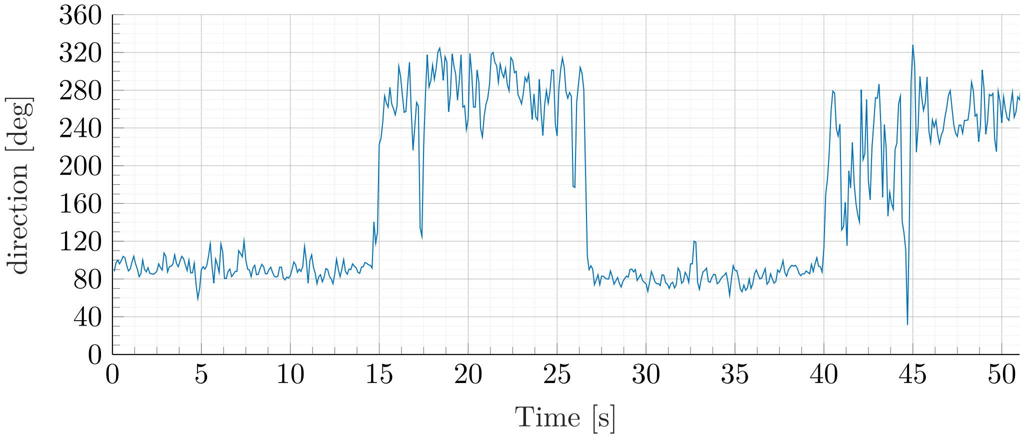


Figure D.7: The graph shows the synchronous direction of the wind with respect the the wind speed in Figure D.6

It is seen in Figure D.6 that the wind speed is lowered from approximately 8 m/s to 2 m/s. It is seen in Figure D.7 that the windscreen produces turbulence in the windscreen and the direction of the wind change approximately 180°.

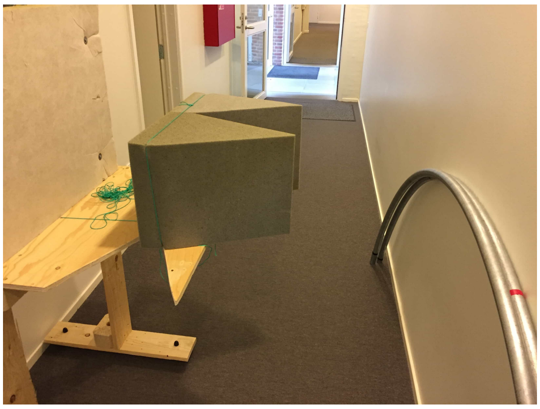


Figure D.8: The picture shows the measurement setup for the large wedge, configuration two

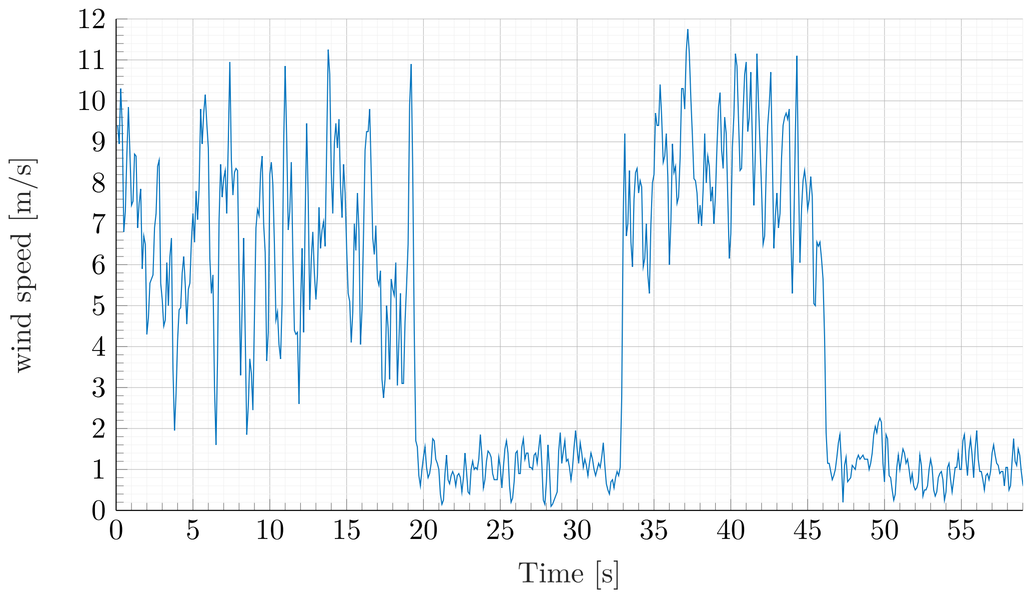


Figure D.9: The graph shows the wind speed versus time for configuration two. The grape have a high speed period and a low speed period, in the high speed period, the anemometer is in the wind approximatly 30 cm from the windscreen where in the low speed period, the anemometer is inside the windscreen.

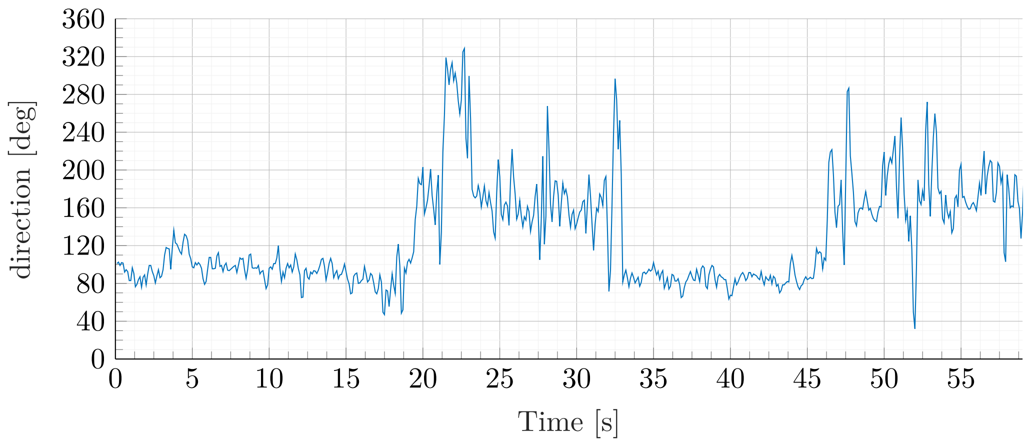


Figure D.10: The graph shows the synchronous direction of the wind with respect the the wind speed in Figure D.9

It is seen in Figure D.9 that the wind speed is lowered from approximately 7.5 m/s to 1 m/s. It is seen in Figure D.10 that the windscreen produces turbulence as high as with the small foam wedge in the windscreen and the direction of the wind change approximately 70°.

Figure D.11: The picture shows the measurement setup for the single rockwool bat, configuration four

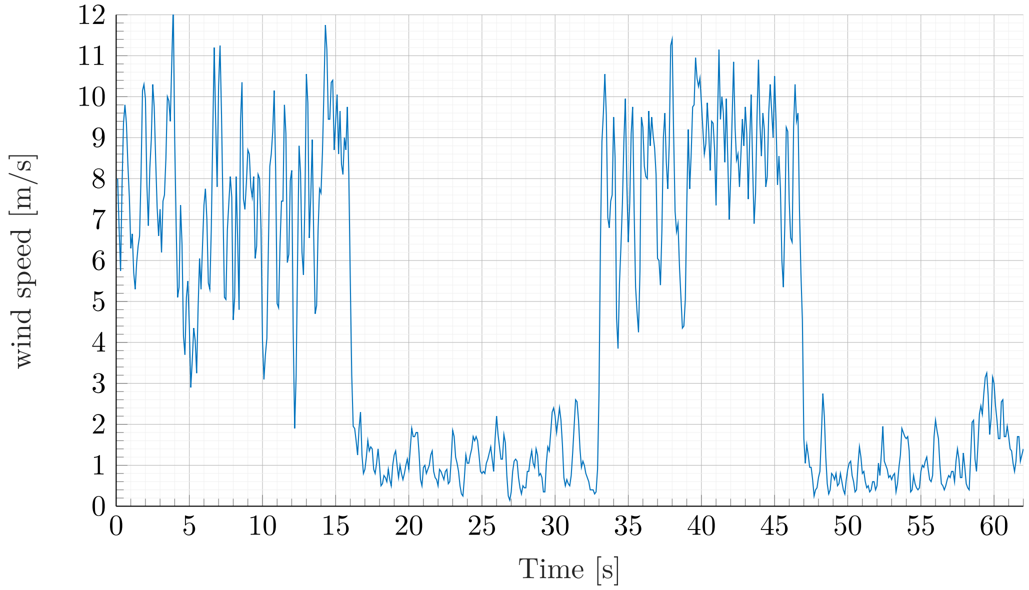


Figure D.12: The graph shows the wind speed versus time for configuration four. The grape have a high speed period and a low speed period, in the high speed period, the anemometer is in the wind approximatly 30 cm from the windscreen where in the low speed period, the anemometer is inside the windscreen.

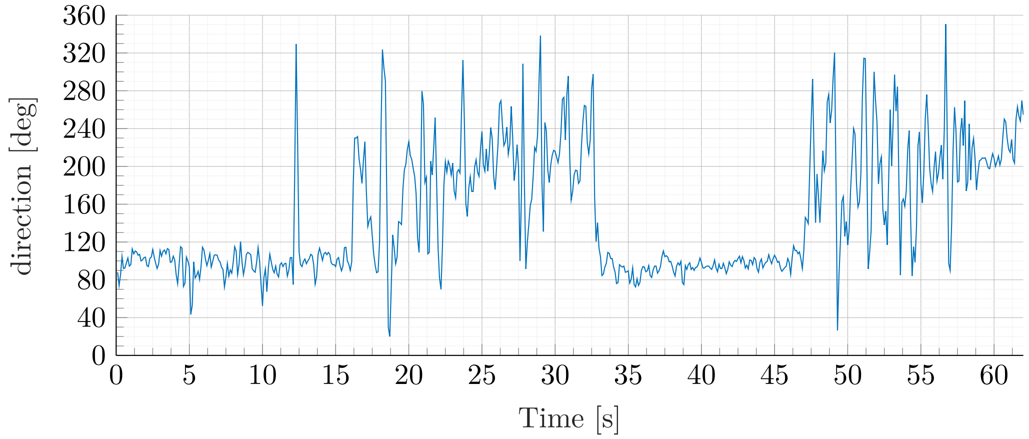


Figure D.13: The graph shows the synchronous direction of the wind with respect the the wind speed in Figure D.12

It is seen in Figure D.12 that the wind speed is lowered from approximately 8 m/s to 1 m/s. It is seen in Figure D.13 that the windscreen produces higher turbulence compare to the foam wedge windscreen. The direction of the wind change is approximately 100°.

Figure D.14: The picture shows the measurement with the large wedge and single rockwool bat, configuration five.

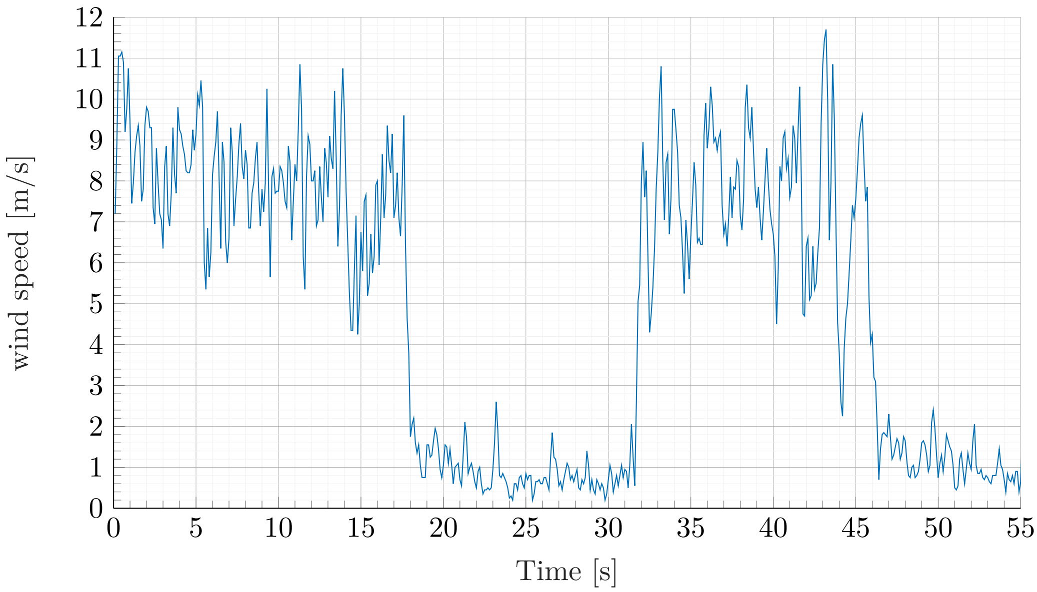


Figure D.15: The graph shows the wind speed versus time for configuration five. The grape have a high speed period and a low speed period, in the high speed period, the anemometer is in the wind approximatly 30 cm from the windscreen where in the low speed period, the anemometer is inside the windscreen.

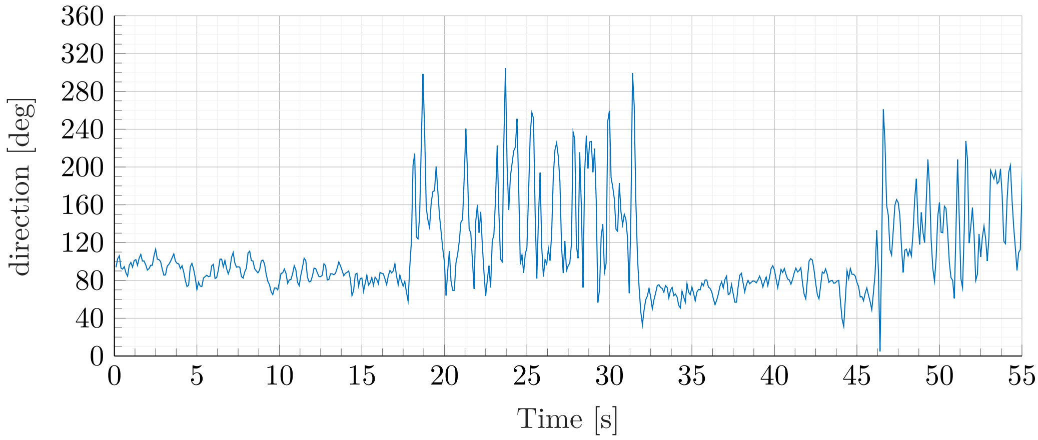


Figure D.16: The graph shows the synchronous direction of the wind with respect the the wind speed in Figure D.15

It is seen in Figure D.15 that the wind speed is lowered from approximately 8 m/s to 0.8 m/s. It is seen in Figure D.16 that this windscreen produces the highest turbulence of all windscreen. The direction of the wind change approximately 60°.

Summary

It is measured that all windscreen configuration lower the wind speed in the microphone position. The lowest attenuation of the wind is of the small foam wedge where the highest wind speed attenuation is attained by configuration five. Generally all windscreen yield turbulence wind direction, but the single rockwool bat configuration four seems to be the worst with respect to directional turbulence.

Appendix E

Windscreen attenuation measurement

A measurement was made to measure the wind attenuation in the microphone position with the difference windscreen configuration.

Materials and setup

To measure the wind attenuation of the windscreen configuration the following materials are used:

Table E.1: Equipment list

Description	Model	Serial-no	AAU-no
PC	Macbook	W89242W966H	-
Optimised windscreen	-	-	-
Fast fan	-	-	-
Fan control	tranformator	-	60398
Windspeed tools	FT technologies FT742-D-SM	9002-922	105634

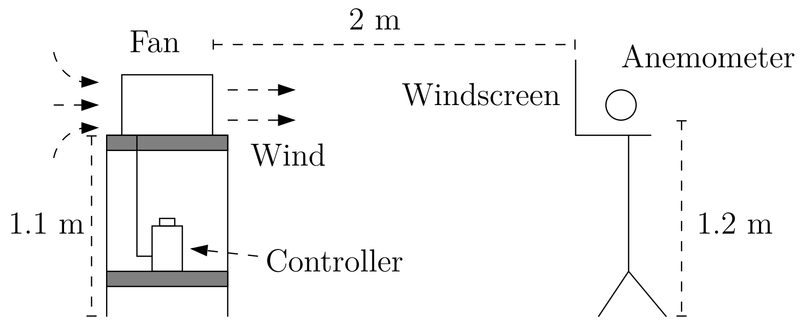


Figure E.1: The figure shows the measurement setup for the wind speed measurement in the microphone position

Test procedure

1. The materials are set up as in Figure E.1.
2. The fan is placed such that is produce directly crosswind.
3. The fan is activated
4. The anemometer is activated to record.
5. The anemometer is placed in the wind for approximately 10 s.
6. Then the anemometer is moved intro the microphone position for approximately 10 s.
7. Then the anemometer is moved back in the wind for approximately 10 s.
8. Then the anemometer is moved intro the microphone position for approximately 10 s again.
9. The wind speed measurement is plotted in MATLAB®with a moving mean filter with 2 sample and as m/s versus s.
10. Next the wind direction measurement is plotted in MATLAB®with a moving mean filter with 2 sample and as ° versus s.

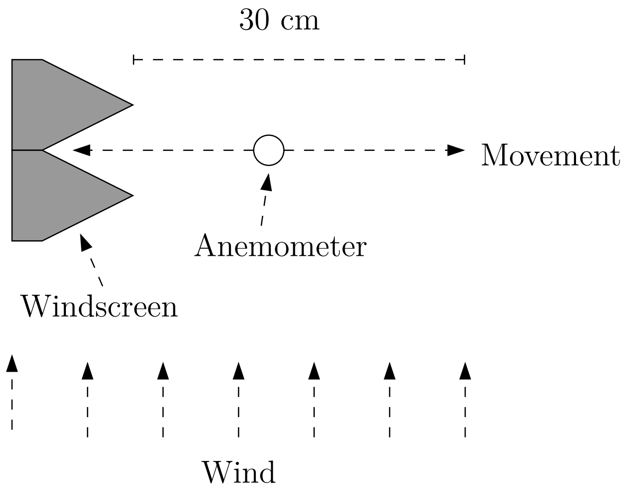


Figure E.2: The figure shows the movement of the anemometer doing the measurement

Measurement area

To be able to generate a controlled wind flow, the hall way in Fredrick Bajers vej 7B5, 9220 Aalborg is used. The following **??** shows a drawing of the area ans the position of the fan and windscreen.

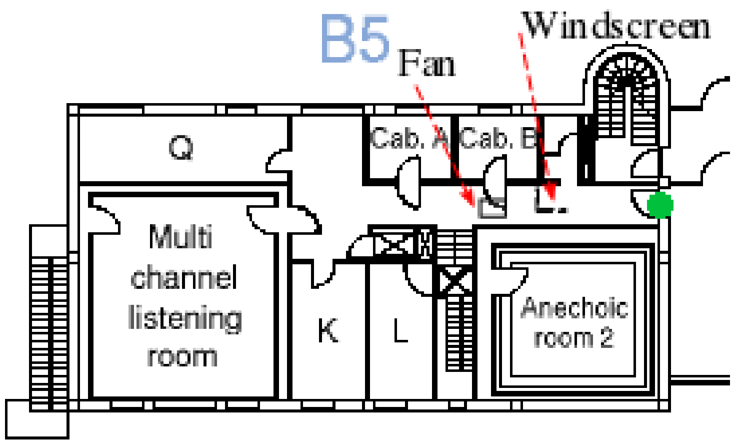
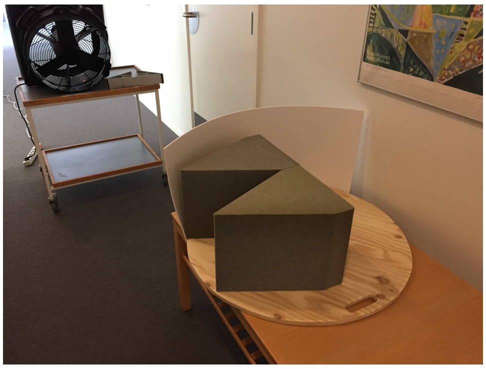


Figure E.3: The picture illustrate the area, where the wind flow is measured

Results

The following graphs shows the result of the measurement.

Figure D.5 shows the measurement setup of the foam wedge, where the **??** shows the result.



(a) The picture shows the measurement setup for the optimised windscreen configuration five from back

(b) The picture shows the measurement setup for the optimised windscreen configuration five in front

Figure E.4: ap:wind:large_opt_pic

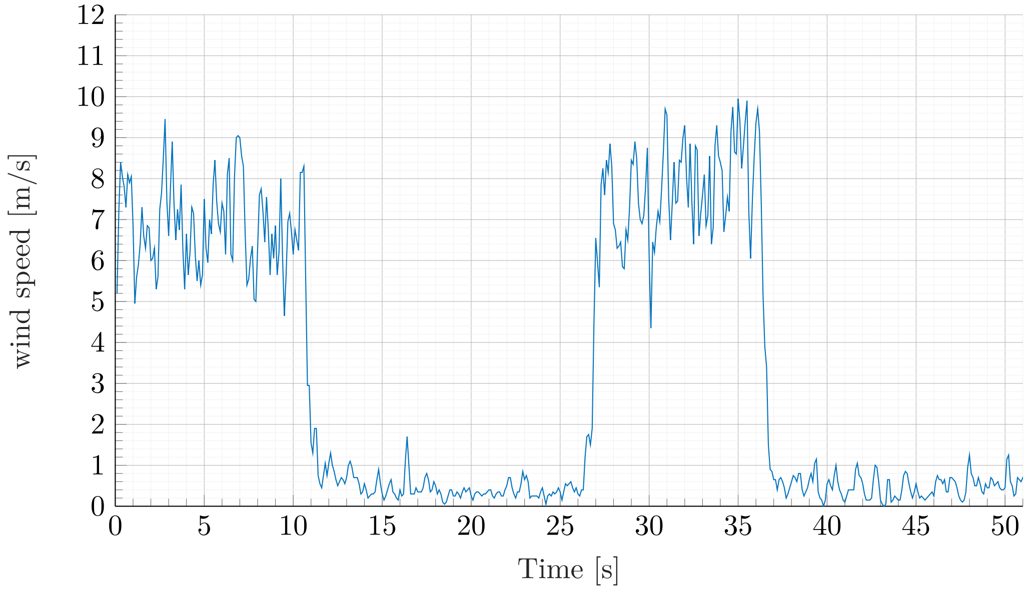


Figure E.5: The graph shows the wind speed versus time for the optimised configuration five. The grape have a high speed period and a low speed period. In the high speed period, the anemometer is in the wind approximately 30 cm from the windscreen where in the low speed period, the anemometer is inside the windscreen.

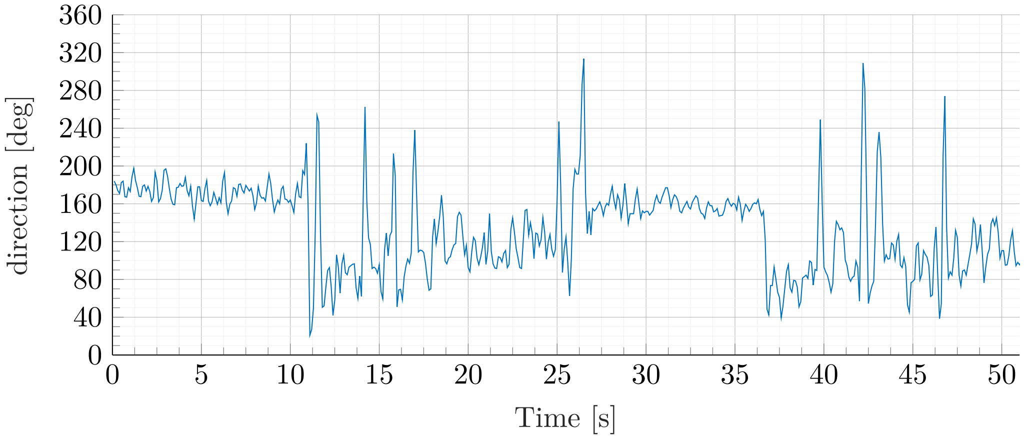


Figure E.6: The graph shows the synchronous direction of the wind with respect the the wind speed in Figure E.5

It is seen in Figure E.5 that the wind speed is lowered from approximately 8 m/s to 0.5 m/s. It is seen in Figure E.6 that the windscreen produces turbulence in the windscreen and the direction of the wind change approximately −100°. The reason that the angle is negative in this measurement is that the anemometer is turned 180° in the vertical plan for practical reason.

Summary

It is measured that the optimised windscreen lower the wind turbulence by optimising the aerodynamics with curved surface and small thin edges.

Appendix F

Outline ET 250-3D turntable control

In this appendix the control of an Outline ET 250-3D turntable will be described. The turntable can be controlled in three ways. the first which are Transistor–Transistor Logic (TTL) commands through a jack connector. Secondly, it can be controlled by specific Dynamic Link library (DLL) commands through Ethernet. All of those are included in a software pack available from the manufacturer. Thirdly the turntable can be controlled by User Datagram Protocol (UDP) commands through Ethernet. For controlling the turntable by MATLAB, the second Ethernet based control method is the easiest to do because MATLAB supports Ethernet access. The DLL method requires a complicated scripts which might only work on Windows operation systems. UDP can be sent and received by the vast majority of operating systems, which support IPv4 Ethernet connection. The usage of UDP leads to short and simple scripts, where the script opens a UDP channel as a file, and e.g. the script shall only edit the file in the right position to move the turntable. Because of the advantages in implementation, UDP is chosen over the other controlling methods for the turntable.

Materials and setup

The following materials are used:

Table F.1: Equipment list

Description	Model	Serial-no	AAU-no
PC	Macbook	W89242W966H	-
Turntable	Outline ET 250-3D	REIBO012	-
PC - software	MATLAB 2018b	-	-

The UDP setup of the computer

To establish connection between the turntable and computer, both have to run at the same SUBNET MASK. The turntable comes with a factory setting for Ethernet connection which is as follows:

Table F.2: Turntable network address

Internet Protocol (IP)	192.168.1.34
SUBNET MASK	255.255.255.0
DEFAULT GATEWAY	192.168.1.250
BROADCAST IP	192.168.1.255

Turntable control command

The software is implemented as a function, where the user can retrieve the turntable position, specify a position and stop the turntable. The function name is:

Code snippet F.1: The turntable control function | ET250_3D.m

```
function [angle] = ET250_3D(cmd, angle)
```

The function is made as a switch case with input variable "cmd", and an angle input. The following command can be send to the "cmd" of the function:

Table F.3: Function commands

cmd = 'udp_start'	Which start a connection on port 7000
cmd = 'set'	Which move the turntable to the specified angle
cmd = 'get'	Which get the position of the turntable
cmd = 'stop'	Which stop the turntable from moving
cmd = 'udp_stop'	Which stop the connection on port 7000

The MATLAB function

Code snippet F.2: The turntable control function | ET250_3D.m

```
function [angle] = ET250_3D(cmd, angle)


switch cmd

    case 'udp_start'
        echoudp('on',7000)
        u = udp('192.168.1.34',7000);
        fopen(u)


```

```
    case 'set'
        %request current position
        fwrite(u,hex2dec(['04';'00';'00';'04']))
        x = fread(u,7);
        angle_current = (x(4)*256+x(5))/10;

        %calc shortest way
        angle_delta = angle-angle_current;
        if angle_delta > 180
            angle_delta = angle_delta - 360;
        end
        if angle_delta < -180
            angle_delta = angle_delta + 360;
        end

        cmd(1) = uint8( 1.5-sign(angle_delta)/2 );  %1st byte = direction
        cmd(2) = uint8( floor(abs(angle_delta*10)/256) );  %angle in degree*10
        cmd(3) = uint8( mod(floor(abs(angle_delta*10)),256) );  %angle in degree*10
        cmd(4) = 0;

        fwrite(u,cmd)
        x = dec2hex(fread(u,2));                          %receive ACK


    case 'get'
        %request current position
        fwrite(u,hex2dec(['04';'00';'00';'04']))
        x = fread(u,7);
        angle = (x(4)*256+x(5))/10;

    case 'stop'
        fwrite(u,hex2dec(['03';'00';'00';'03']))          %send stop stop
        x = dec2hex(fread(u,2));

    case 'udp_stop'
        echoudp('off')
        fclose(u)
end


end
```

Appendix G

The directionality of L-acoustics KUDO

A measurement was made to measure the directionality of a L-acoustics KUDO. The goal of this appendix is to measure the polar response and calculate transfer functions of the line source array element in a free field environment with calibrated measuring equipment. During the measurement of the polar response, impulse responses are measured with a specified degree step size all around the speaker. E.g. if the step is one degree, the loudspeaker will be turned 1 degree for every impulse response measurement, until 360° is achieved.

Materials and setup

To measure the directionality of the line source element, the following materials are used:

Table G.1: Equipment list

Description	Model	Serial-no	AAU-no
PC	Macbook	W89242W966H	-
Audio interface	RME Fireface UCX	23811948	108230
Microphone	GRAS 26CC	78189	75583
Preamp	GRAS 40 AZ	100268	75551
Speaker	L-acoustics KUDO	7733	-
Turntable	Outline ET 250-3D	REIBO012	-
PC - software	MATLAB 2018b	-	-

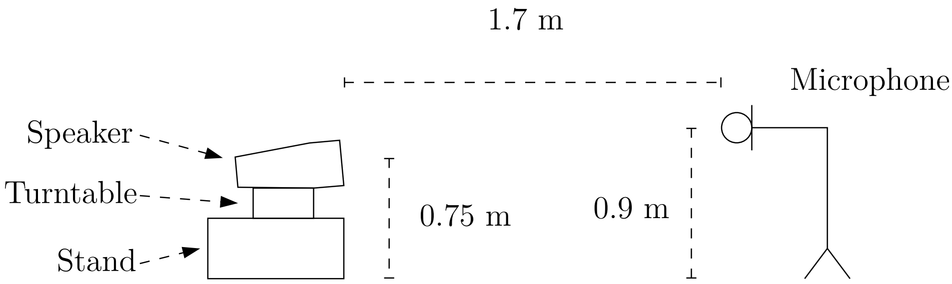


Figure G.1: The figure shows the measurement setup in the anechoic chamber

Test procedure

1. The materials are set up as in Figure G.1 where the microphone is connected to the audio interface.
2. The microphone is calibrated
3. The speaker is placed 1.7 m from the microphone and pointing in the direction of the microphone.
4. The impulse response is measured for every 5°
5. The −3 dB SPL step contour is calculated until −21 dB SPL and plotted.

Measurement area

To be able to measure the windscreen frequency response, the anechoic chamber in Fredrick Bajers vej 7B4, 9220 Aalborg is used. The following Figure G.2 shows a drawing of the area ans the position of the fan and windscreen.

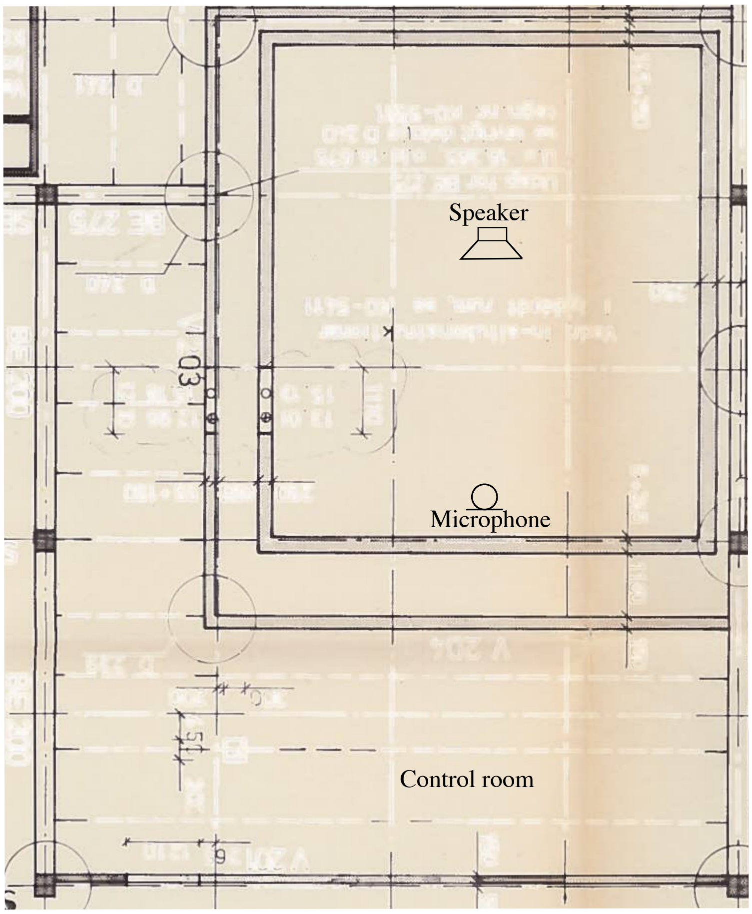


Figure G.2: The picture illustrate the position of the microphone and the speaker

Results

The following Figure G.3 shows the measurement setup.

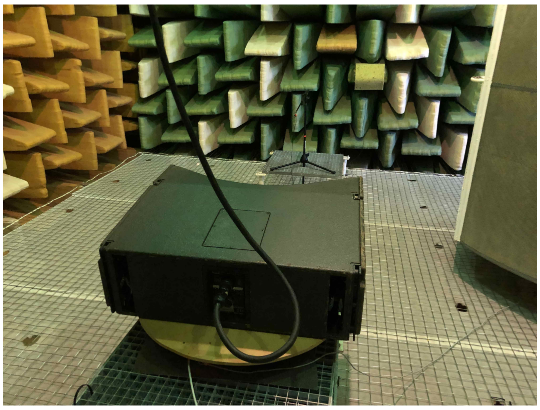


Figure G.3: The picture shows the measurement setup

The following graphs shows the result of the measurement.

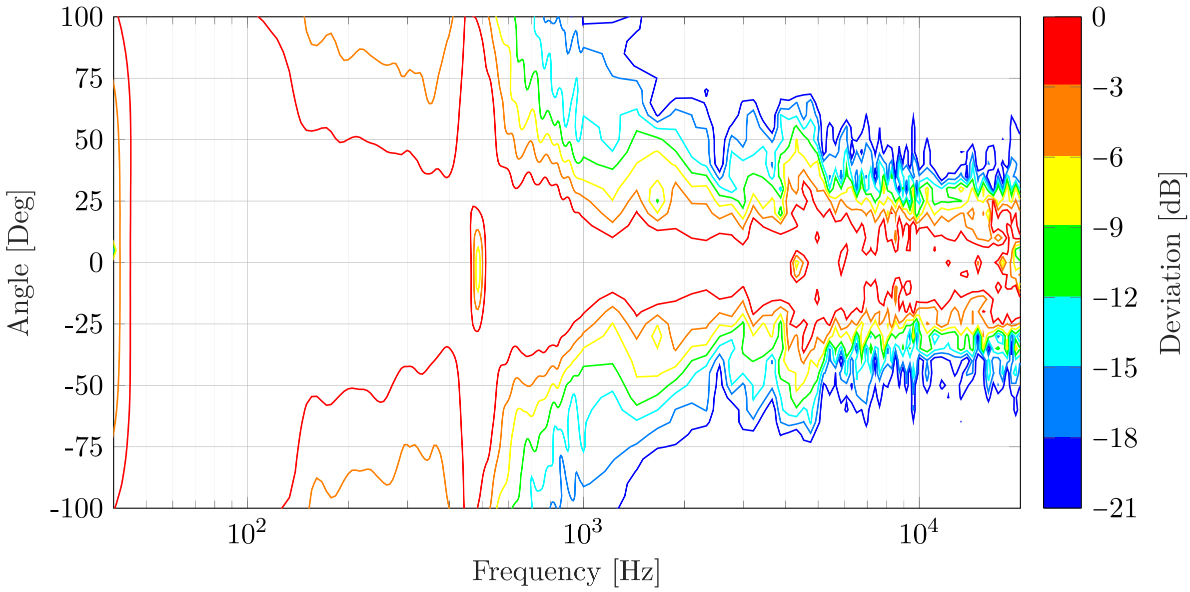


Figure G.4: The graph shows

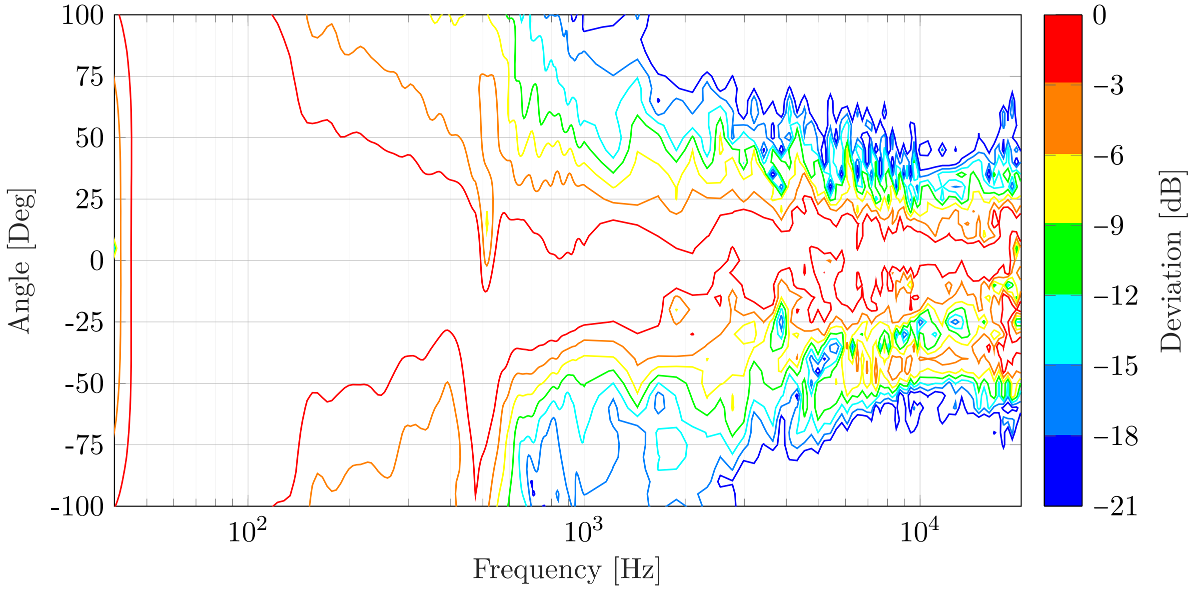


Figure G.5: The graph shows

Summary

Appendix H

cross wind effect on line source array

A measurement was made to measure the transfer function differences in three point in crosswind. One microphone situated in downwards direction, one microphone situated in upwards direction and one microphone situated in center, which is between the other two microphone. The used speaker have a horizontal dispersion pattern of 80°, but it is the 50° angle which is used as explained in **??**

Materials and setup

To measure the transfer function in a crosswind situation, the following materials are used:

Table H.1: Equipment list

Description	Model	Serial-no	AAU-no
PC	Macbook	W89242W966H	-
Audio interface	RME Fireface UCX	23811948	108230
Microphone	GRAS 26CC	78189	75583
Preamp	GRAS 40 AZ	100268	75551
Microphone	GRAS 26CC	78186	75582
Preamp	GRAS 40 AZ	100267	75550
Microphone	GRAS 26CC	??	
Preamp	GRAS 40 AZ		
3 Windscreen	Author design	-	-
Line source element	L-acoustics KUDO		
Line source element	L-acoustics KUDO		
Line source element	L-acoustics KUDO		
Line source element	L-acoustics KUDO		
Line source element	L-acoustics KUDO		
Amplifier	Lab PLM10000Q		
Amplifier	Lab PLM10000Q		
Mixer	Yamaha LS9		
Wind measurement tools	Davis	-	
Angling tools	Author design	-	
flying tools	-	-	-

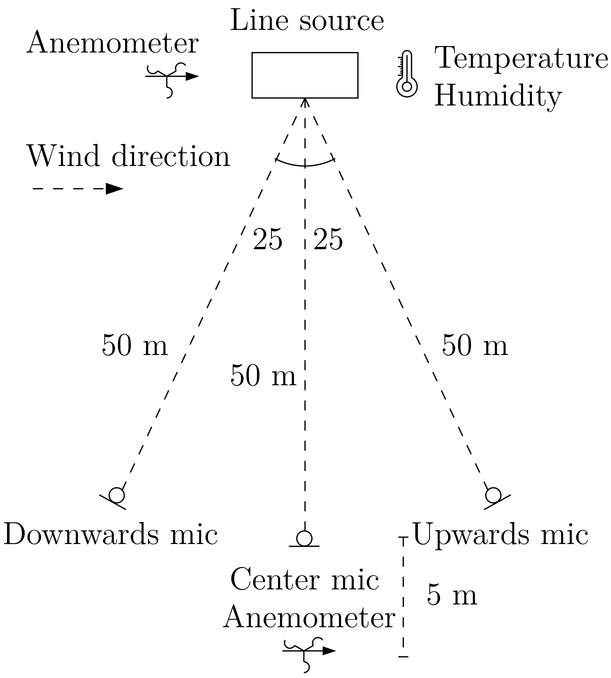


Figure H.1: The figure shows the microphone position versus the position of the line source, while the array is 0° horizontal turned

(a) The picture shows the speaker setup

(b) The figure shows the wind direction

Figure H.2: The figures shows the measurement set up for Appendix A and **??**

Test procedure

1. The microphone i calibrated.
2. The wind direction is measured.
3. The materials are set up as in Figure H.1 where the speaker is placed in cross wind direction, such that the frontal wave direction is orthogonal the the wind. The microphone and speaker is connected to the audio interface.
4. The speaker is placed 2.92 m above the ground.
5. The speaker is tilted 5° pointing down agents the ground.
6. The microphone is placed 1.68 m above the ground, 50 m from the speaker. One 25° to the left of the speaker, one 25° to the right of the speaker and one in center between the to other microphone.
7. The anemometer at the speakers is situated on the speaker tower in the same side as shown on the setup and a hight of 4.64 m
8. The anemometer at the microphone position is lifted 1.68 m above the ground.
9. The wind direction goes from the upwards microphone to the downwards microphone.
10. The humidity and temperature is measured at the speaker position.
11. 10 sine sweep is performed with a length of 5 s each while the wind direction and speed is measured.
12. The impulse response is calculated and filtered with a 4th order highpass filter at 20 Hz.
13. The correlation is calculated for each impulse response to the first impulse response for time alignment [**?**] of all microphone channels.
14. The mean impulse response is calculated for the 10 measurement of all three microphone.
15. The transfer function is calculated with a 10 sample moving mean filter.

16. The the transfer function is down sampled to fit the plotting program.
17. The transfer function is calculated with a 5 sample moving mean filter.
18. The wind measurement is synchronised to the transfer function in time.
19. The measurement is repeated 6 times with different horizontal speaker angle from 0° to 30° in step of 5°

Measurement area

To be able to measure in a windy area, parking lot at Tryvej 13, 9320 Hjallerup is used. The following Figure H.3 shows a picture of the area and the approximate position of the speaker and microphone.

Figure H.3: The picture illustrate the area, where the wind flow is measured

Results

All measuring result is not shown here, the rest can be founded in the attached file. One synchronised measurement is shown for the upwards microphone where the speaker is turned 0°, 10°, 20° and 30°. The shown measurement result is for one measurement and is not a mean from 10. This shows the time synchronised result. The average result can be founded in **??**.

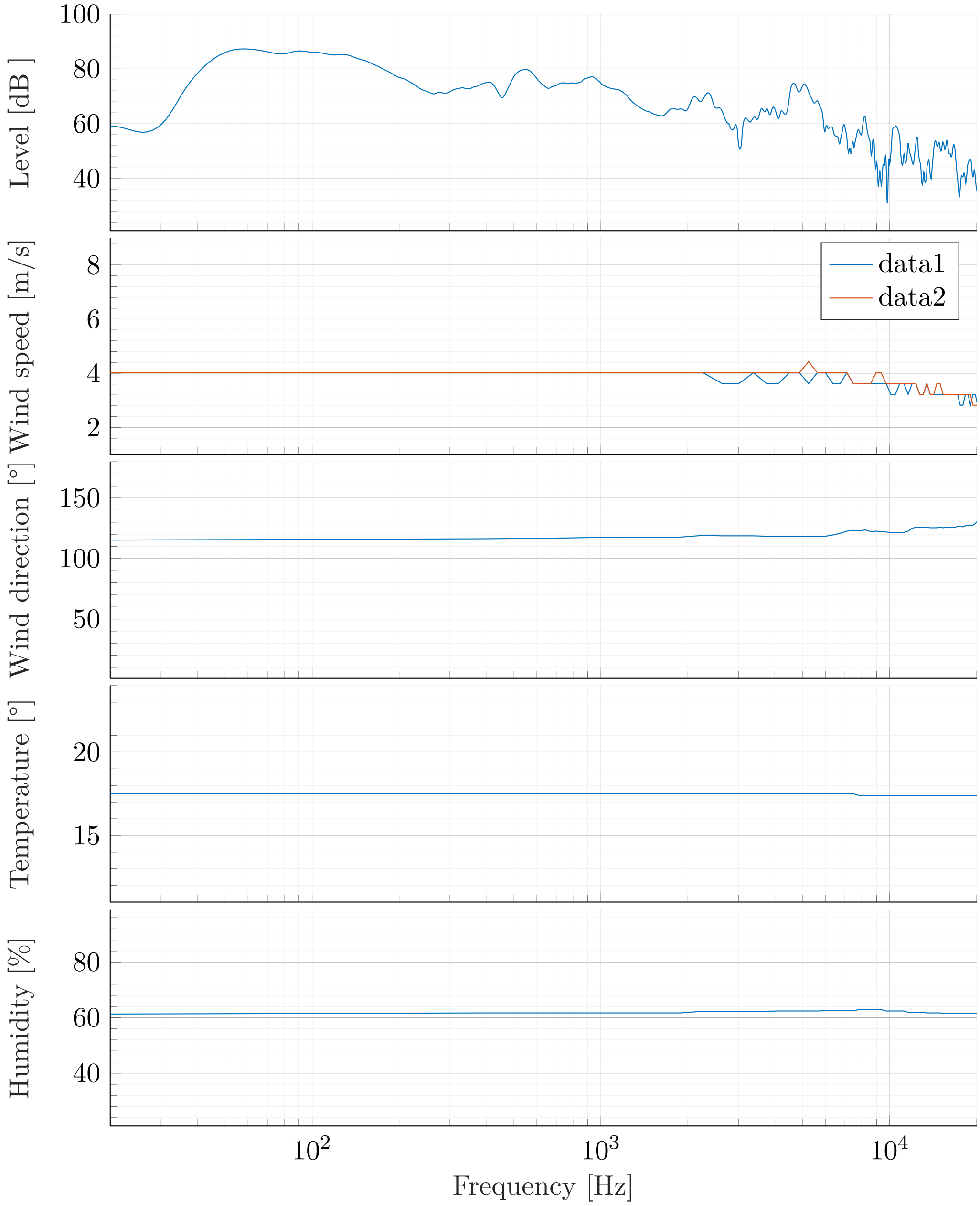


Figure H.4: the graph shows

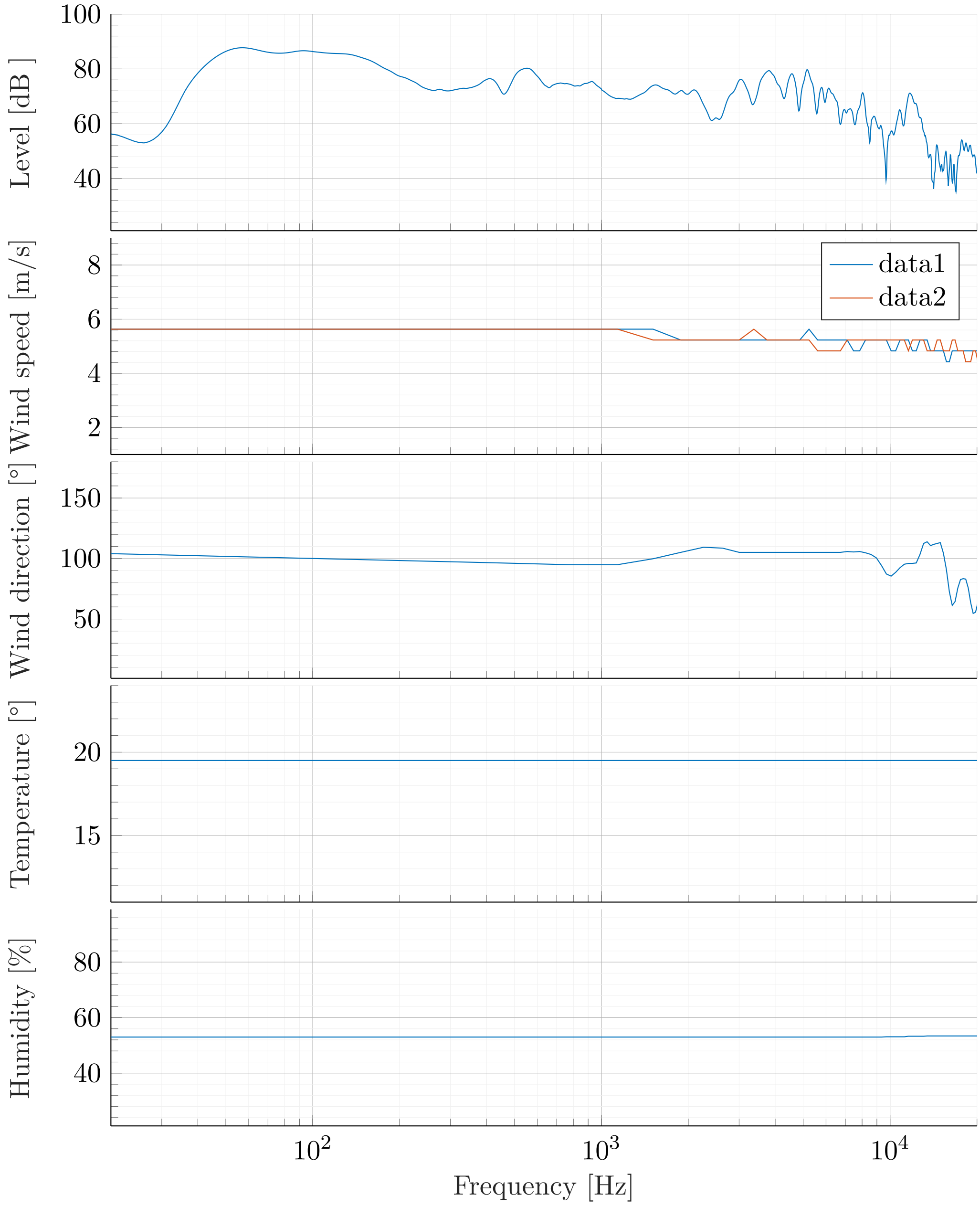


Figure H.5: the graph shows

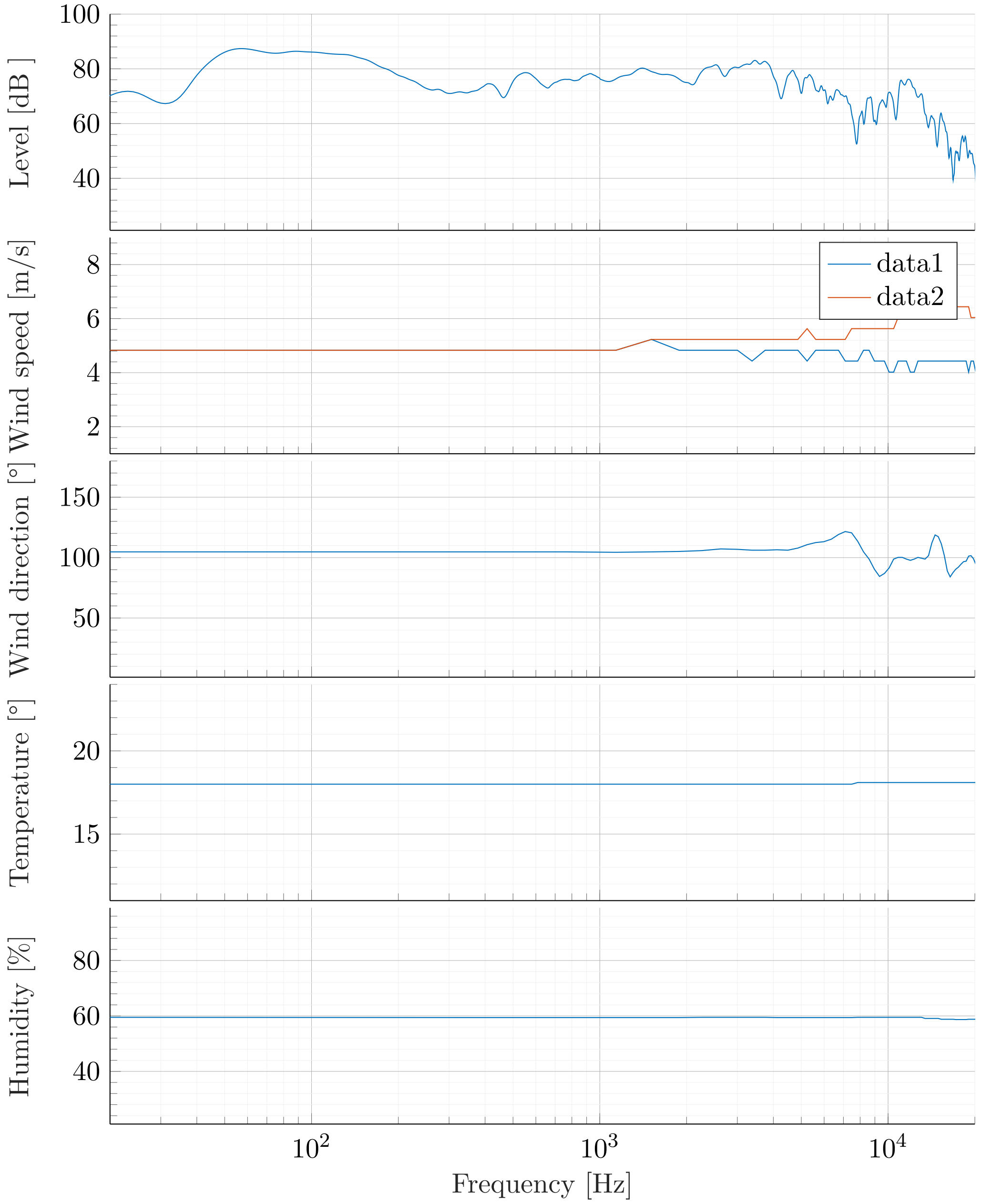


Figure H.6: the graph shows

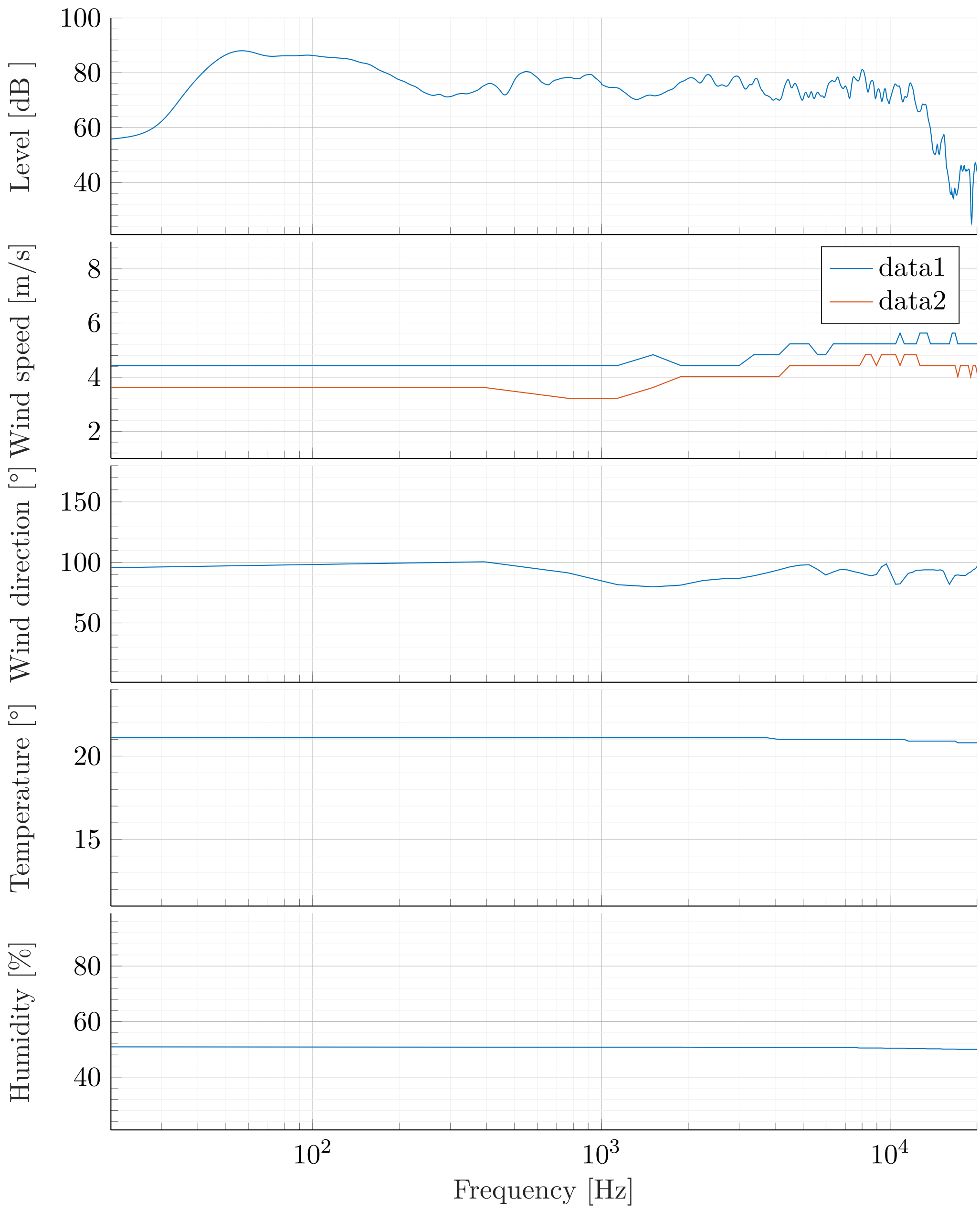


Figure H.7: the graph shows

Summary

Appendix I

Line source array angle measuring design

To measure the angle of the speaker, a angle plate with coloured laser indicator is designed.

Materials and setup

The following martial is used

Table I.1: Equipment list

Description	Model	Serial-no	AAU-no
Laser pen	Red	-	-
Laser pen	Green	-	-
Angle plate	-	-	-
Laser pen holder	-	-	-

(a) The picture shows the angle finder plate

(b) The figure shows the lasers and the laser holder

Figure I.1: The figures shows the angle finder martial for the used line source array

Adjusting the line source horizontal angle

1. The materials are set up as in Figure I.1.
2. The line source array is turned until the laser pointers light is on the drawn line of the angle or with the same distance to the line.

Appendix J

Questionnaire

A questionnaire was made to find the maximum coverage distance for a line array.

Question	Answer	Unit
Company	Profox	
What is the distances between the main stage to the first delay tower at large concert	50	[m]
How many audience attempt to a large concert you produces	15000+	Number
What is the distances between the main stage to the first delay tower at medium concert	30	[m]
How many audience attempt to a medium concert you produces	5000+	Number
Flying hight of top speaker	12 - 16	[m]
Does the wind direction and speed influence on the distance decision	No	
Do you want to know the end result	Yes	

Question	Answer	Unit
Company	Nordic sales	
What is the distances between the main stage to the first delay tower at large concert	75	[m]
How many audience attempt to a large concert you produces	100000+	Number
What is the distances between the main stage to the first delay tower at medium concert	50	[m]
How many audience attempt to a medium concert you produces	30000	Number
Flying hight of top speaker	?	[m]
Does the wind direction and speed influence on the distance decision	Yes	
Do you want to know the end result	No	

Question	Answer	Unit
Company	Moto rental	
What is the distances between the main stage to the first delay tower at large concert	50-60	[m]
How many audience attempt to a large concert you produces	20000+	Number
What is the distances between the main stage to the first delay tower at medium concert	50	[m]
How many audience attempt to a medium concert you produces	5000+	Number
Flying hight of top speaker	12 - 16	[m]
Does the wind direction and speed influence on the distance decision	Yes	
Do you want to know the end result	Yes	

Question	Answer	Unit
Company	Roskilde festival	
What is the distances between the main stage to the first delay tower at large concert	73	[m]
How many audience attempt to a large concert you produces	100000+	Number
What is the distances between the main stage to the first delay tower at medium concert	?	[m]
How many audience attempt to a medium concert you produces	?	Number
Flying hight of top speaker	?	[m]
Does the wind direction and speed influence on the distance decision	No	
Do you want to know the end result	No	

Question	Answer	Unit
Company	AV-center Aalborg	
What is the distances between the main stage to the first delay tower at large concert	60	[m]
How many audience attempt to a large concert you produces	20000+	Number
What is the distances between the main stage to the first delay tower at medium concert	30	[m]
How many audience attempt to a medium concert you produces	5000+	Number
Flying hight of top speaker	14 - 16	[m]
Does the wind direction and speed influence on the distance decision	Yes	
Do you want to know the end result	Yes	
comment: The biggest problem lays in the frequency range from 1.0 kHz to 7.0 kHz where the understanding of the music despisers and the music sound muddy	-	

Question	Answer	Unit
Company	Kinovox	
What is the distances between the main stage to the first delay tower at large concert	60	[m]
How many audience attempt to a large concert you produces	20000+	Number
What is the distances between the main stage to the first delay tower at medium concert	30	[m]
How many audience attempt to a medium concert you produces	10000	Number
Flying hight of top speaker	14 - 16	[m]
Does the wind direction and speed influence on the distance decision	Yes	
Do you want to know the end result	No	
comment: He use to rotate the line array agents the wind if he know that the wind is crosswind and the wind will continue along the concert time. Moreover he would stop the concert if the wind speed is above 10 m/s	-	