Sound control in windy weather

Master Thesis

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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, March 22, 2019

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Glossary

FDTD Finite-Difference Time-Domain. 15

FOH Front Of House. 9, 26

PA Public Address System. 7, 8

SPL Sound Pressure Level. 7, 8, 9, 10, 11, 12, 16, 17, 20, 23, 24, 25, 26, 27, 28, 29, 33, 34, 35, 36, 38, 39, 53, 58

Chapter 1

Introduction

Coming later

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

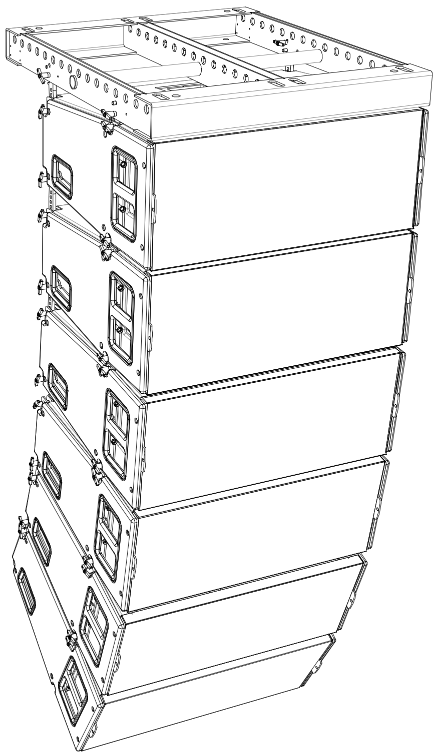


Figure 2.1: The figure shows an illustration of a KUDO line source array from L-Acoustics [L-Acoustics, b]

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 can 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.

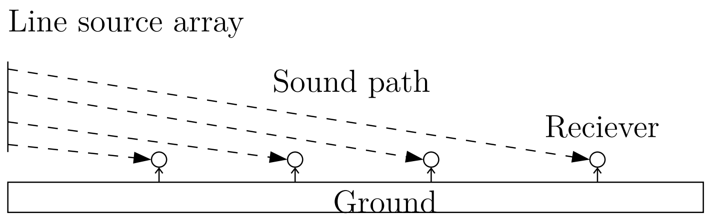


Figure 2.2: The figure illustrate the concept of outdoor PA venue

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 [L-Acoustics, 2019] and D&B Audiotechnik [d&b audiotechnik, 2019] 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 $\frac{1}{3f}$ [Bauman et al., 2001]

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

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.

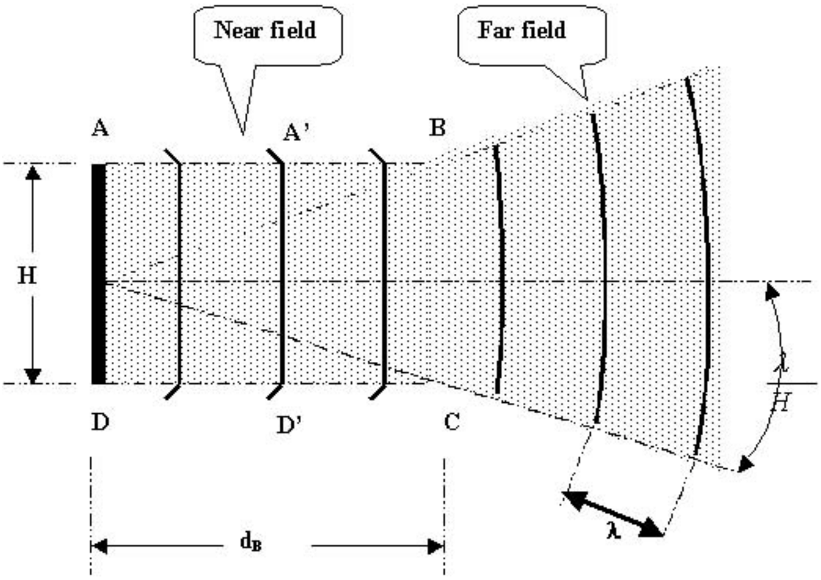


Figure 2.3: The figure shows horizontal cut of a SPL radiation pattern of a line source array [Bauman et al., 2001].

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 [Corteel et al., 2017]. 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 [ISO 9613-1:1993] gives an overview of calculating the SPL attenuation concerning the frequency, distance, temperature and humanity. The article [Corteel et al., 2017] 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 [Corteel et al., 2017].

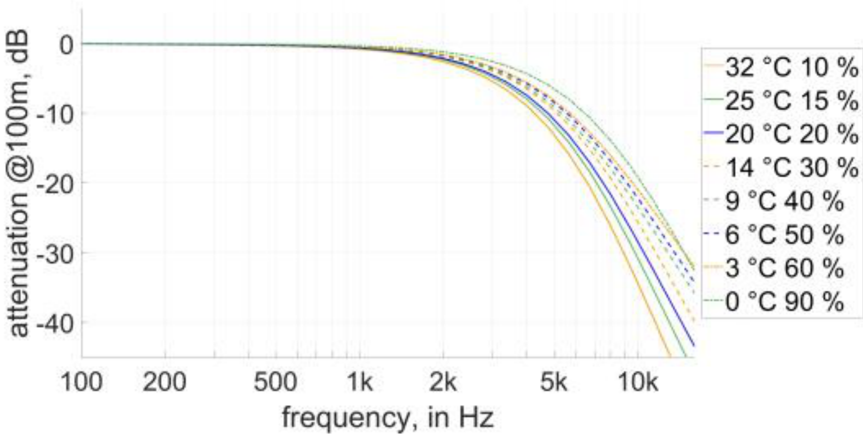


Figure 2.4: The graph shows the attenuation in dB with respect to frequency, humanity and temperature [Corteel et al., 2017].

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 % [Wong and Embleton, 1985] [Bohn, 1987]. 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.

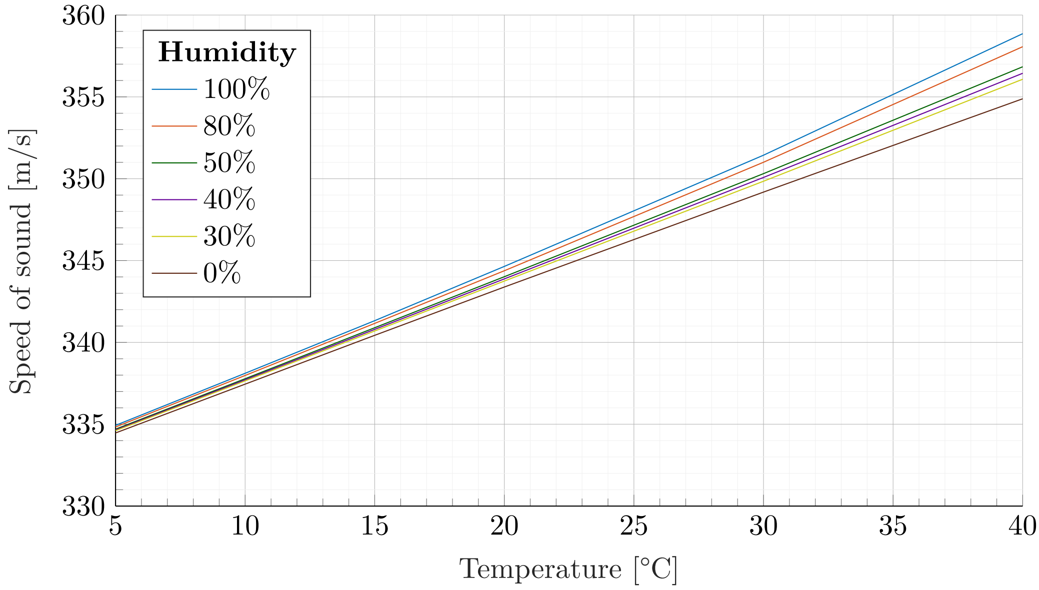


Figure 2.5: The figure shows the increase of sound speed with respect to humanity and temperature [Bohn, 1987]

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 [Levine et al., 2018]. 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 [Ostashev et al., 2005] where the author of [Ballou, 2008] 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 [Ostashev et al., 2005] has simulated a homogeneous crosswind effect on an omnidirectional source at 100 Hz. The author of [Prospathopoulos and Voutsinas, 2007] implemented a ray tracing method with a vector based interpolation as shown in Figure 2.6.

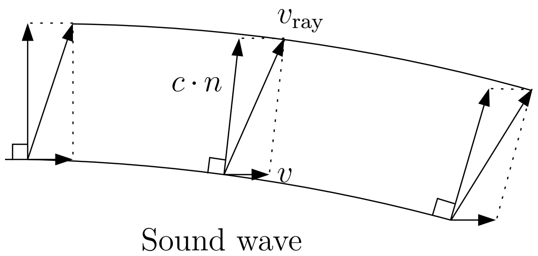


Figure 2.6: The figure shows a geometrical ray tracing calculation scheme of calculate the resulting wave direction at crosswind [Prospathopoulos and Voutsinas, 2007], [Ostashev et al., 2005]

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 [Ostashev et al., 2005] 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 [Ostashev et al., 2005], 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.

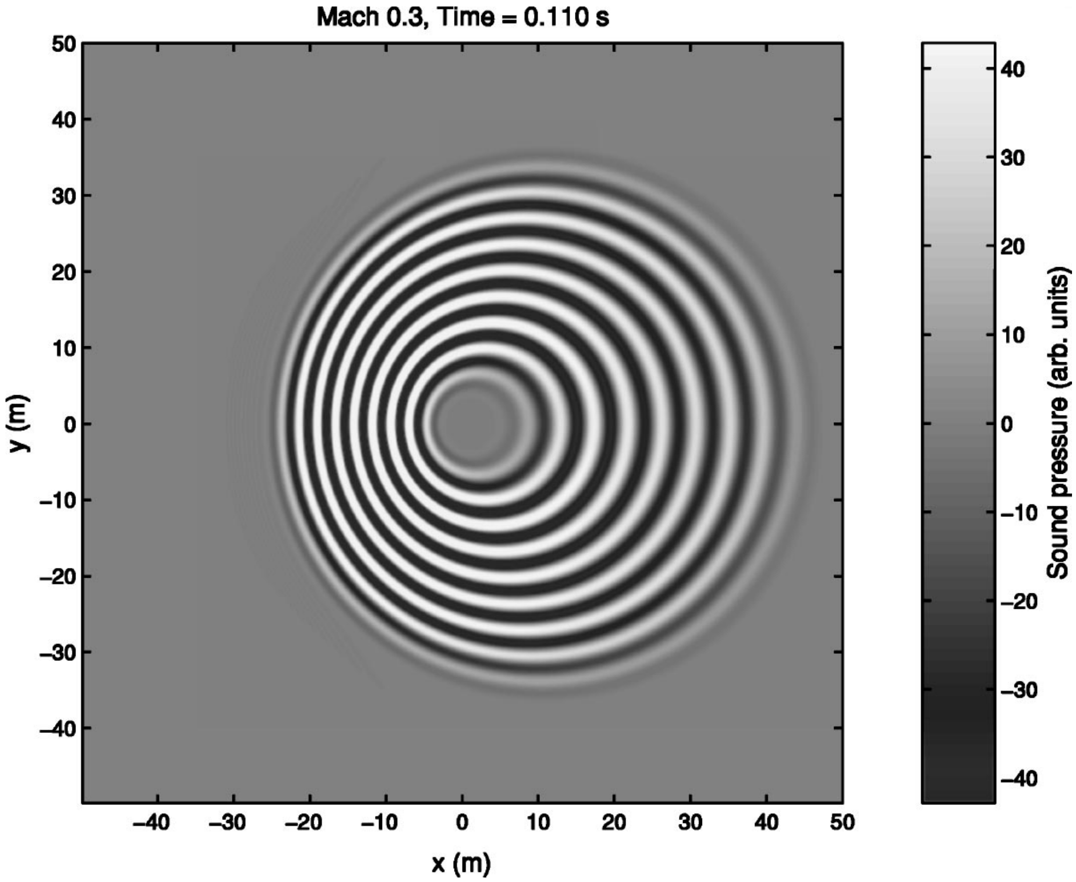


Figure 2.7: The figure shows a simulation of a 100 Hz omnidirectional source with a uniform constant wind speed from left with speed of 102.9 m/s [Ostashev et al., 2005].

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 [Czerwinski et al., 1999], the port design of the low frequency driver [Vanderkooy, 1998] 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 [Czerwinski et al., 1999]. 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. [Czerwinski et al., 1999]. 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 [Czerwinski et al., 1999].

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 [Roozen et al., 1998]. 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 [Czerwinski et al., 1999]. 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 [Voishvillo, 2004]. 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 [Piercy et al., 1977]. The measurement shows that the ground reflection affects the frequency response with high interference. A measurement in Appendix A and Appendix B 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 [Piercy et al., 1977]. 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 [Beranek, 2006]. 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 [Beranek, 2006]. The average absorption a_{sabine} coefficient is calculated to be above 0.80. The method and result can be founded in [Beranek, 2006]. The reflection in the high frequency in the audience area doing concert is therefore assumed to be low. At low frequency, the article [Beranek, 2006] 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 [Beranek, 2006]. 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 [Bauman et al., 2001]. 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 [Yang, 2016] in the free troposphere [Rossing, 2014], 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 [Rossing, 2014], [Association, 2003]. 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 [de Oliveira, 2012]. 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 [Piercy et al., 1977] 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 no reflection of the high frequency propagate to the back part of the audience. The following Figure 2.8 display the phenomena of upwards refraction.

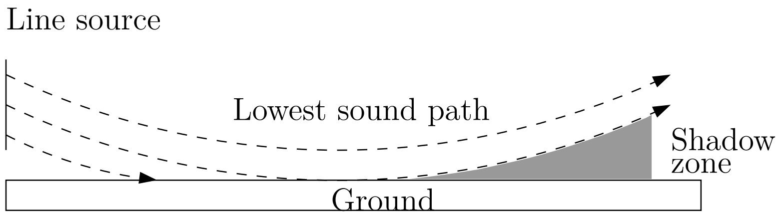


Figure 2.8: The figure illustrates that the shadow zone occurs from an upwards refraction. A line source speaker array contains of many couplet point sources. Every lowest sound path dashed line indicate the lower directional angle of one point source in the line source array.

The following description is based on the distance of 110 m and upwards refraction. As explained in [Piercy et al., 1977] 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 [Piercy et al., 1977]. 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 [Piercy et al., 1977] 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 [Piercy et al., 1977] 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

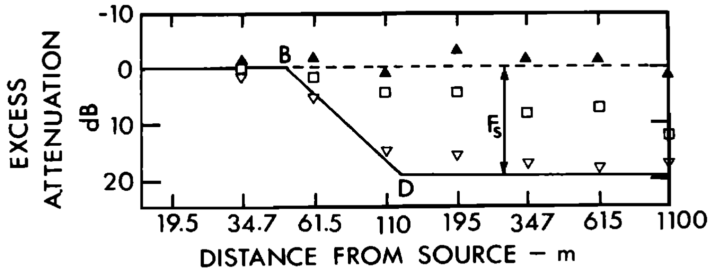


Figure 2.9: Excess attenuation measured for aircraft noise in the 1.2 kHz 1/3 octave band for the ground-to-ground configuration. The vector component of the wind velocity in the direction of propagation for ▲ is 5 m/s, □ a is 0 m/s, and ▽ is −5 m/s. The temperature profile is neutral. F_s is the shielding factor, B is the shadow boundary [Piercy et al., 1977]

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.

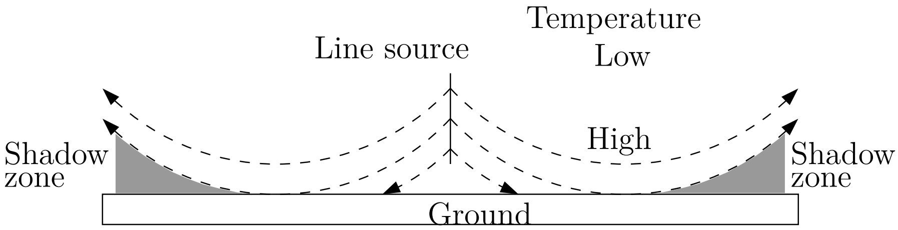


Figure 2.10: Wave refraction of a horizontal omnidirectional line source array in inhomogeneous temperature with lapse profile

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.

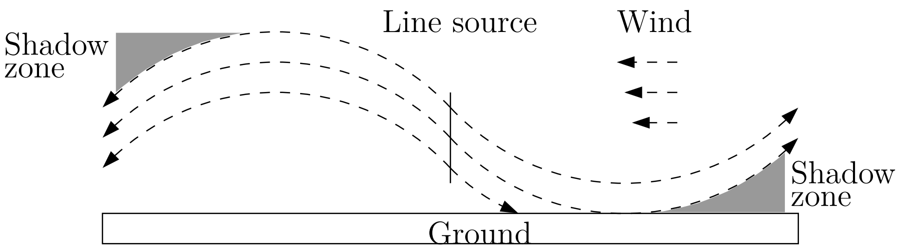


Figure 2.11: Wave refraction of a horizontal omnidirectional line source array in inhomogeneous logarithmically increasing wind profile where the wind gradient points towards left

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.

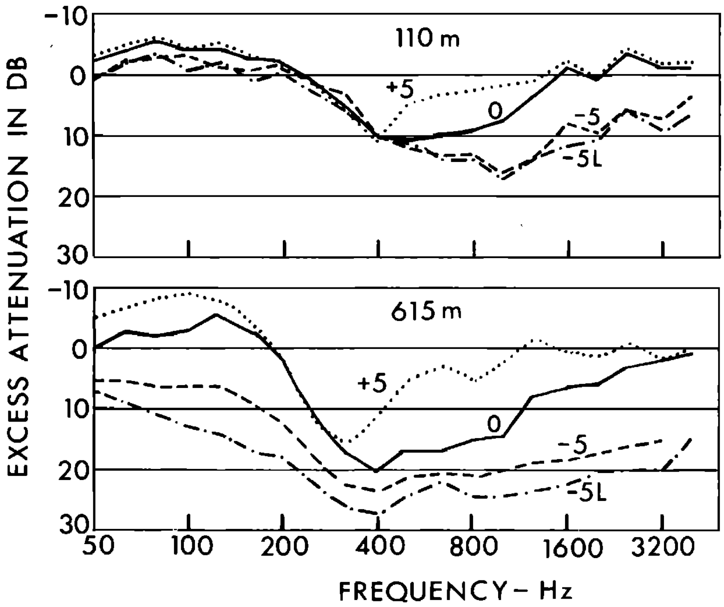


Figure 2.12: Observed attenuation of aircraft noise in a ground-to-ground configuration under a variety of weather conditions. Calculated losses from atmospheric absorption and spherical spreading have been subtracted from the attenuation measured in 1/3 octave bands for distances of 110 m and 615 m. 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. [Piercy et al., 1977]

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. [Embleton, 1996]

Oblique- and crosswind The effect of oblique- and crosswind on acoustical wave propagation in inhomogeneous atmospheric conditions are rarely studied. The author in [Piercy et al., 1977] 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.

Since the effect of crosswind 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 [Gunness, 2001] 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

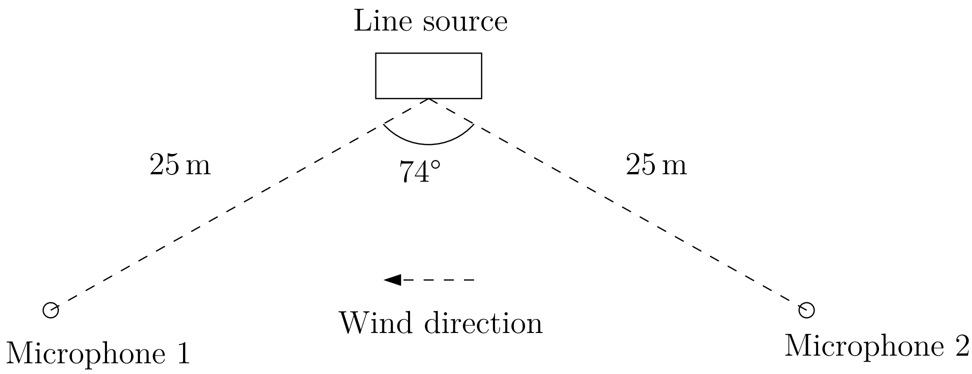


Figure 2.13: The figure shows the microphone position versus the position of the line source array angle of main lobe

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 and Appendix B. 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.

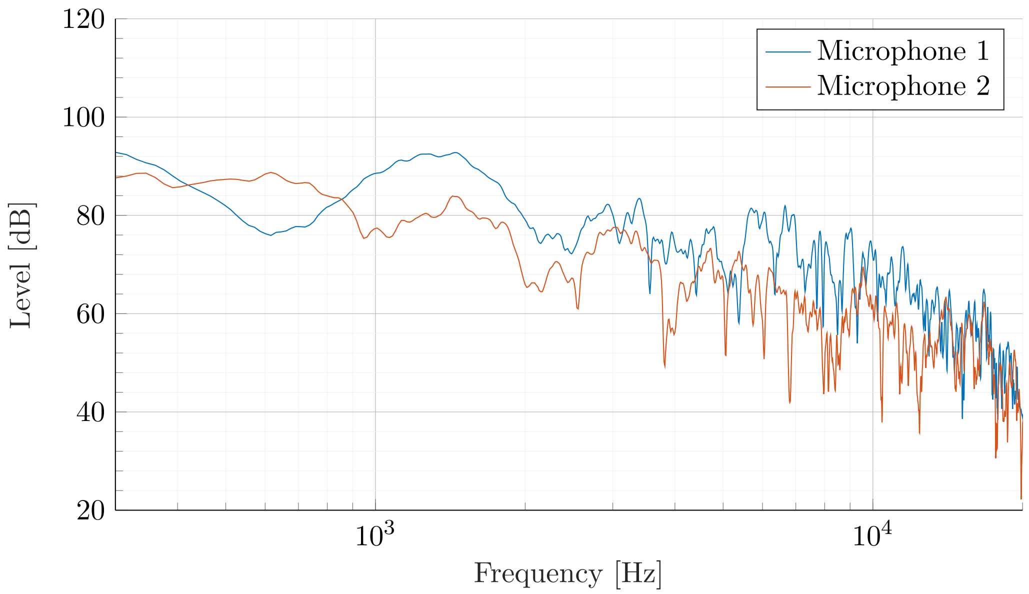


Figure 2.14: The graph shows the first transfer function measurement within the high frequency directional angle. The $L_{\text{eq},5}$ SPL different between the microphones is 4.41 dB SPL (IR_3)

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 [Piercy et al., 1977]. 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.3	71.82 dB SPL	66.33 dB SPL	5.49 dB SPL
Measurement 2 Figure A.4	69.09 dB SPL	64.69 dB SPL	4.40 dB SPL
Measurement 3 Figure A.5	67.67 dB SPL	63.44 dB SPL	4.23 dB SPL
Measurement 4 Figure B.2	68.10 dB SPL	63.69 dB SPL	4.41 dB SPL
Measurement 5 Figure B.3	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 [Letowski and Scharine, 2017]. It is shown in [Letowski and Scharine, 2017] 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° [Piercy et al., 1977]. 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 crosswind, 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 $L_{\mathrm{Aeq},5}$ 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 Deimitation

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.
- Only a solution to the crosswind direction is research.
- 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 cross wind problem

The aim of this section is to propose a solution to the problem founded in the cross wind measurement in section 2.4.1. The solution is based on the problem statement in chapter 4. The solution is a more homogeneous SPL coverage in the coverage area of the speaker without wind. In other word, the line source array has a frontal horizontal directional angle defined as the $-6\,\mathrm{dB}$ SPL limit of main pressure lobe. A line source array main lobe is given in the horizontal degree as an addition of the main lobe from the frontal direction to the side and can both be symmetric and asymmetric, depending on the line source array element.

The proposal solution is to be able to steer the main lobe horizontal direction of the line source array electronical. As beeing able to steer the main lobe horizontal direction of the line source array, the main lobe can be steered more up agents the direction of the coverage area where the wind attenuate. The crosswind problem is not as drastical close to the speaker, so the line source array which shall be able to be controlled is the coverage area is the element which cover the audience in back. The solution is based on a changeable main lobe which is as narrow as possible to archive as high SPL and the audience and as low neighbog desterbines as possible. The following Figure 5.1 shows a graphical illustrate of the proposal solution to archive a more homogeneous SPL coverage area in the frontal direction of the speaker without wind.

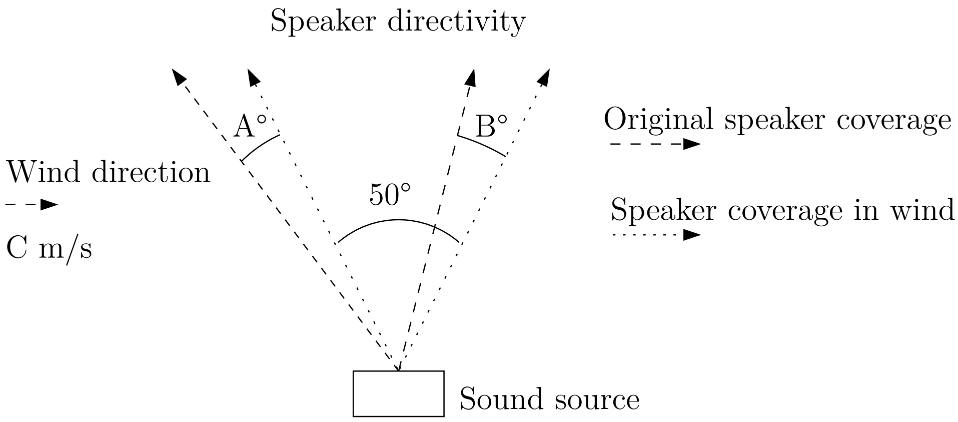


Figure 5.1: The figure shows the wanted direction of the sound coverage area after the effect of crosswind. C is the speed of wind in cross direction of the frontal direction of the speaker. A and B is the angle that needs to be founded. On the figure the angle are equal but that might not be true

The gold is then to search A° and B° based on wind speed C m/s as shown in Figure 5.1 such that the SPL coverage differences is minimized. The angle of A and B in the figure is equal, this might not be true in for the solution.

A filter is needed since the effect is frequency dependent.

5.2 Description of the used line source array

The speaker which will be used to design the solution is a L-Acoustics KUDO line source array where the main lobe coverage can be controlled mechanical. It is both possible to make the main lobe symmetric and asymmetric on this line source element. The following the Figure 5.2 shows both the wide and narrow symmetric main lobe option of the KUDO. The asymmetric coverage can be founded in [L-Acoustics, a]

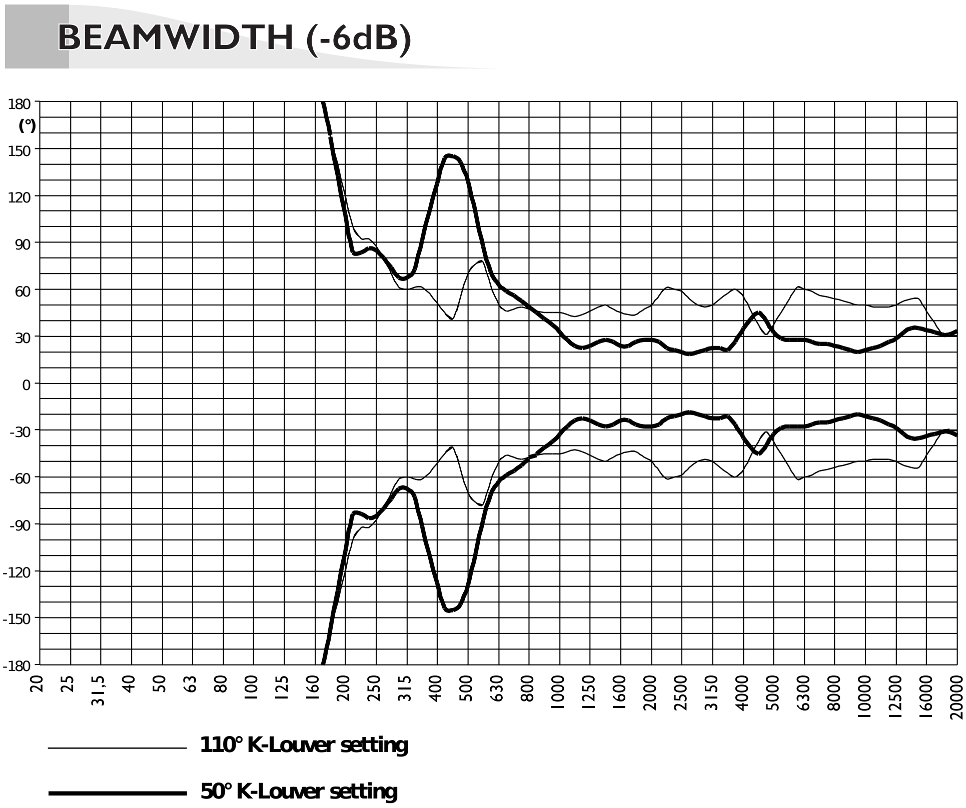


Figure 5.2: The graph shows the symmetric coverage area of the L-Acoustics KUDO line source array [L-Acoustics, a].

The mechanical coverage solution in the KUDO as well as other line source array element is not made for wind problems but for neighbog desterbines and higher SPL in the main lobe of the high frequency. All solution used today is only posible to change by hand and is not electrical controlled.

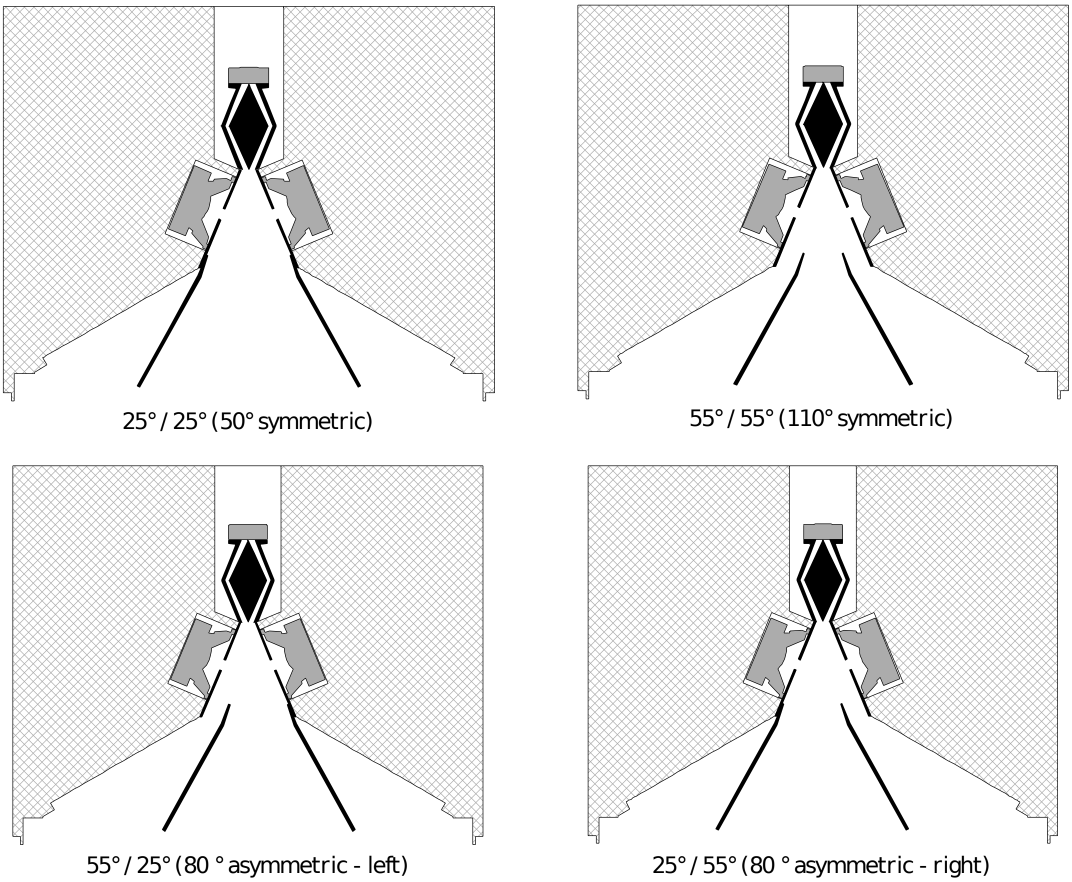


Figure 5.3: The figure shows how the directivity is controlled on a L-Acoustics KUDO line source array element [L-Acoustics, b].

5.3 Designing the measurement

The aim of this section is to design a test on a non modified line source array to test the proposal solution from section 5.1. To be able to test the proposal solution, a measurement system have to be designed. To be sure that the wind noise do not effect the measurement the part of the design takes care about finding the perferable microphone wind screen configuration, based on the aviable STOF in acoustics lab. The second part measure the frequency dependent attenuation of the chosen windscreen compare to the microphone response without windscreen. The comparisen is done to cont for the attenuation of the windscreen such that the measured data reflect the SPL at the point as good as possible. The third part takes care about designing the nesesarly data logging function. The last part designs the measuring signal playback and record method.

5.3.1 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.

There is to wind stopping concept that will be tested, a plan surface of rockwool as shown in **??** and a foam wedge solution as shown in **??**

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 **??**

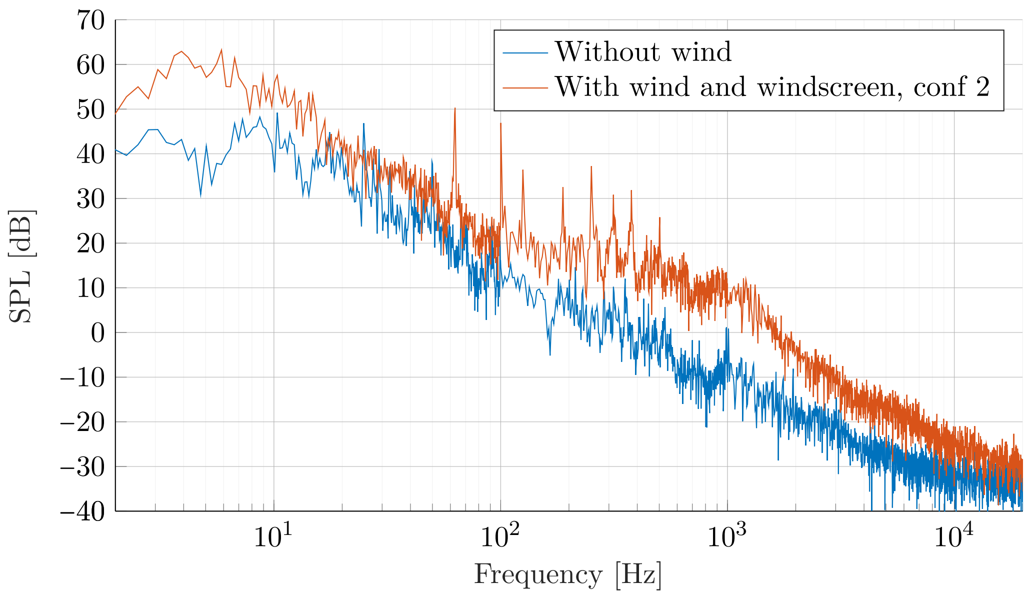


Figure 5.4: The graph shows one of the time measurement with configuration three

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.

5.3.2 Attenuation of the windscreen

5.3.3 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.

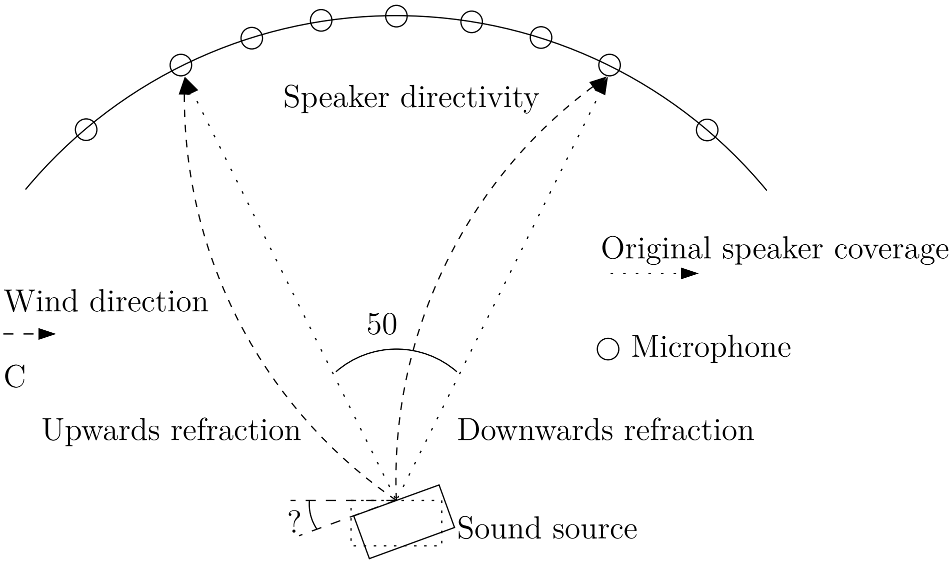


Figure 5.5: The figure shows the measurement setup

where the refraction at 110 m already starts at 400 Hz.
The area is without The resend to use this

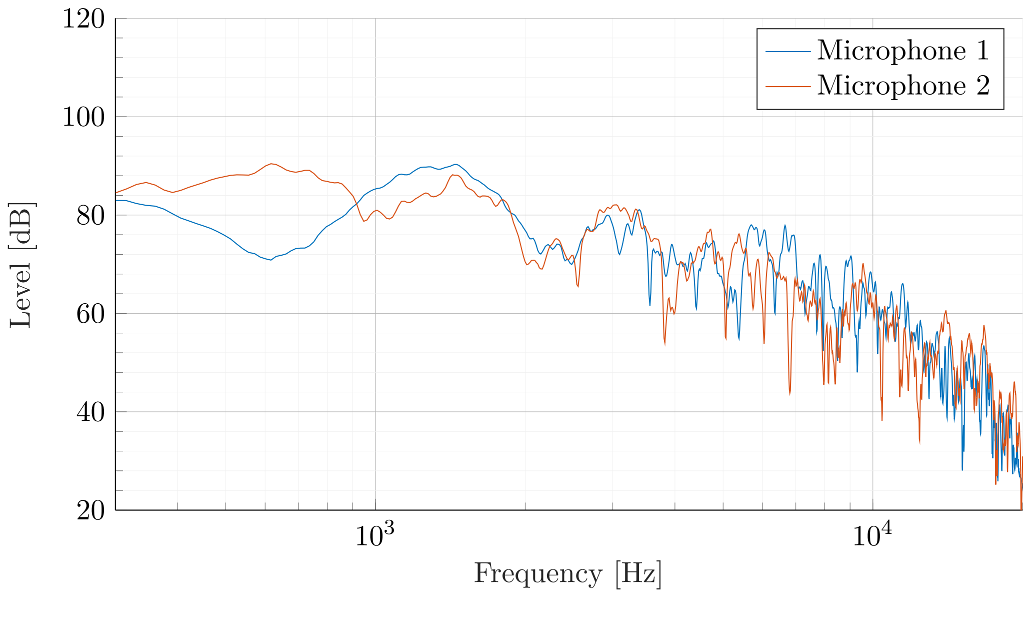


Figure 5.6: The graph shows the first transfer function measurement within the high frequency directional angle. The $L_{\mathrm{Aeq},5}$ SPL different between the microphones is 6.77 dB SPL (IR_3) The graph is normed to contain the same $L_{\mathrm{Aeq},5}$ SPL

5.4 Measuring program

5.5 Result

Chapter 6

Product design

6.1 Technical solution

The proposal solution is therefore a electronical controlled angle and width of the main based on the strangth of the wind and the coverage distance to the audience.

Part III

Results

Chapter 7

Results

Chapter 8

Discussion and conclusion

8.1 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	-	-	-

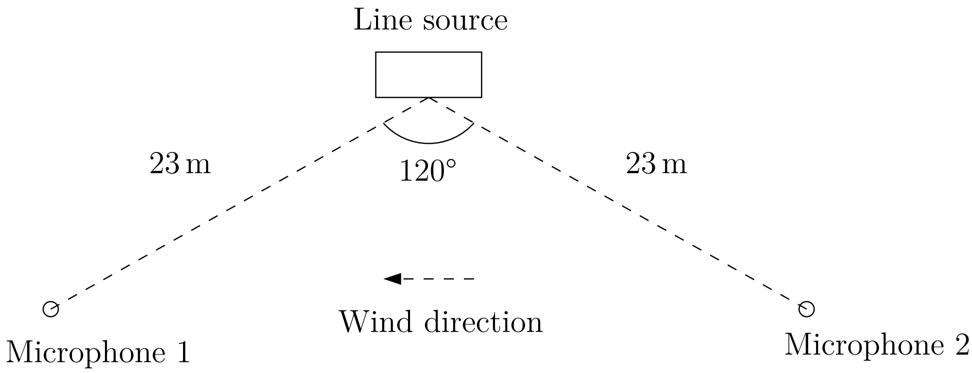


Figure A.1: The figure shows the microphone position versus the position of the line source

(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 Appendix B

Test procedure

1. the microphone i calibrated.
2. The wind direction is measured.
3. The materials are set up as in Figure A.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 [Gunness, 2001] 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.

Results

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

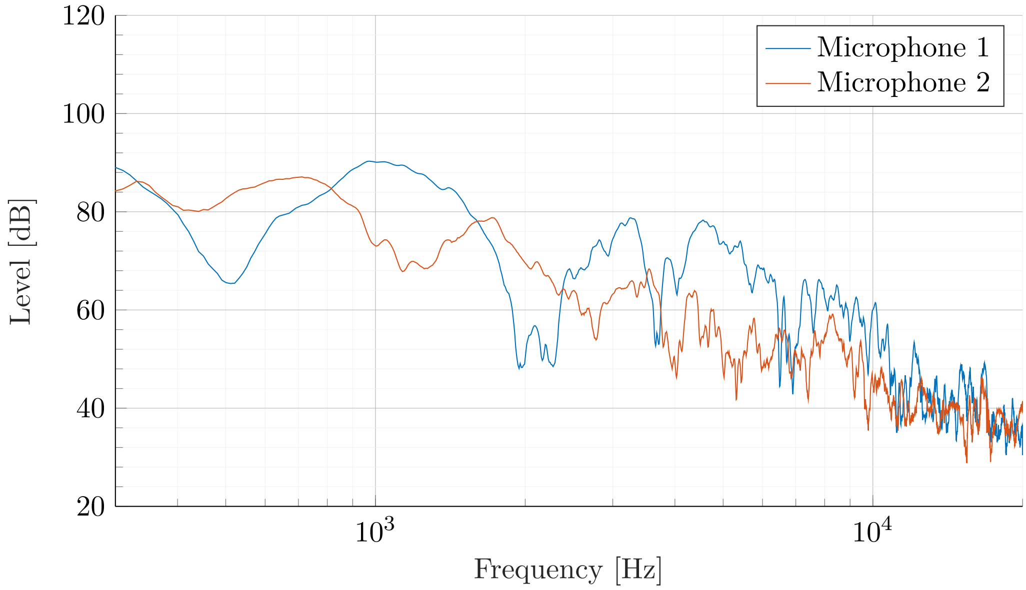


Figure A.3: The graph shows the first transfer function measurement. The $L_{\mathrm{eq},5}$ Sound Pressure Level (SPL) different between the microphones is 5.49 dB SPL (IR_6)

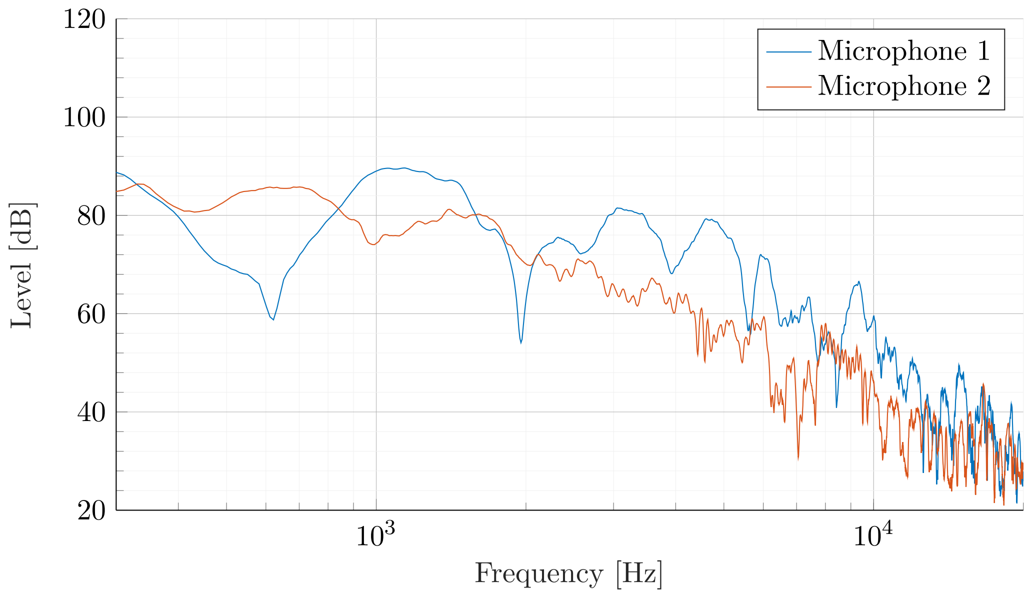


Figure A.4: The graph shows the second transfer function measurement. The $L_{\mathrm{eq},5}$ SPL different between the microphones is 4.40 dB SPL (IR_7)

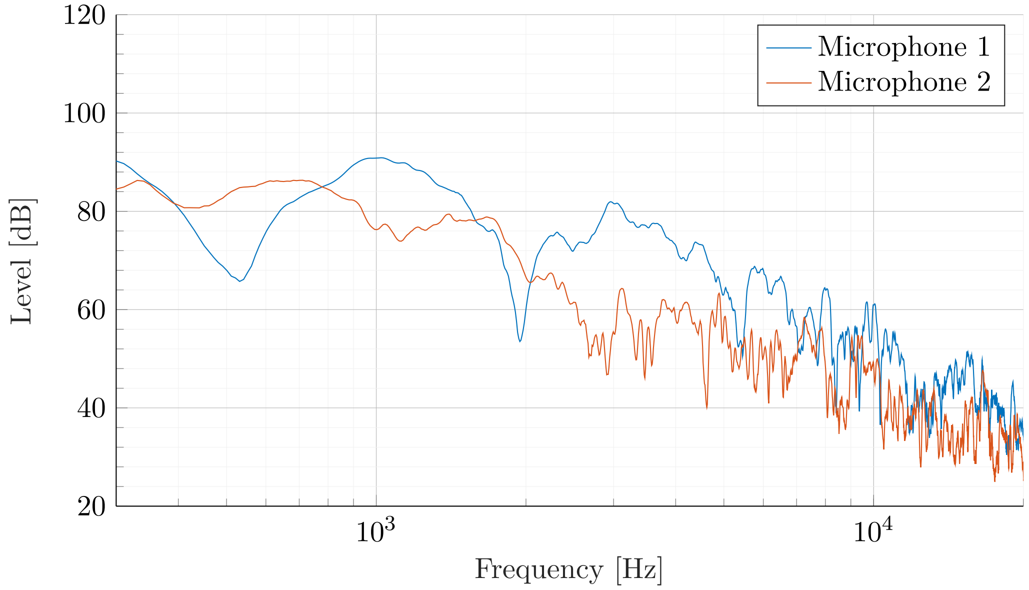


Figure A.5: The graph shows the third transfer function measurement. The $L_{\mathrm{eq},5}$ SPL different between the microphones is 4.23 dB SPL (IR_8)

On Figure A.3, Figure A.4 and Figure A.5 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

Appendix B

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 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
dB technologies	DVA T4	-	-
Wind measurement tools	Drahtlose Wetterstation	-	2157-45
flying tools	-	-	-

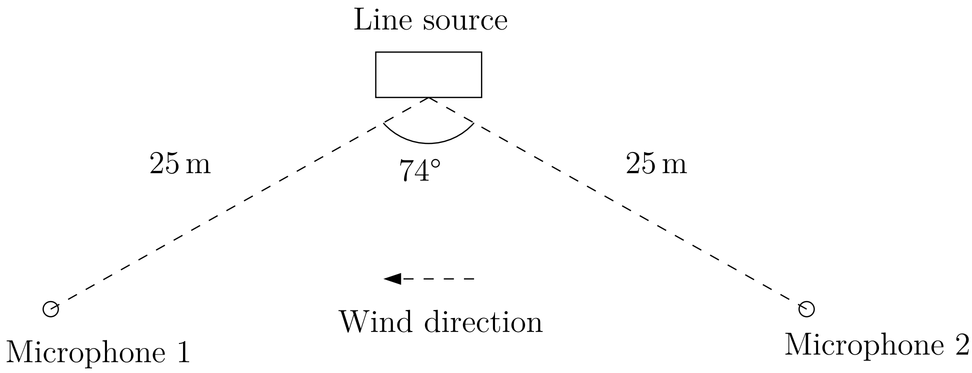


Figure B.1: The figure shows the microphone position versus the position of the line source

Test procedure

1. the microphone i calibrated.
2. The wind direction is measured.
3. The materials are set up as in Figure B.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 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 [Gunness, 2001] 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 two times.

Results

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

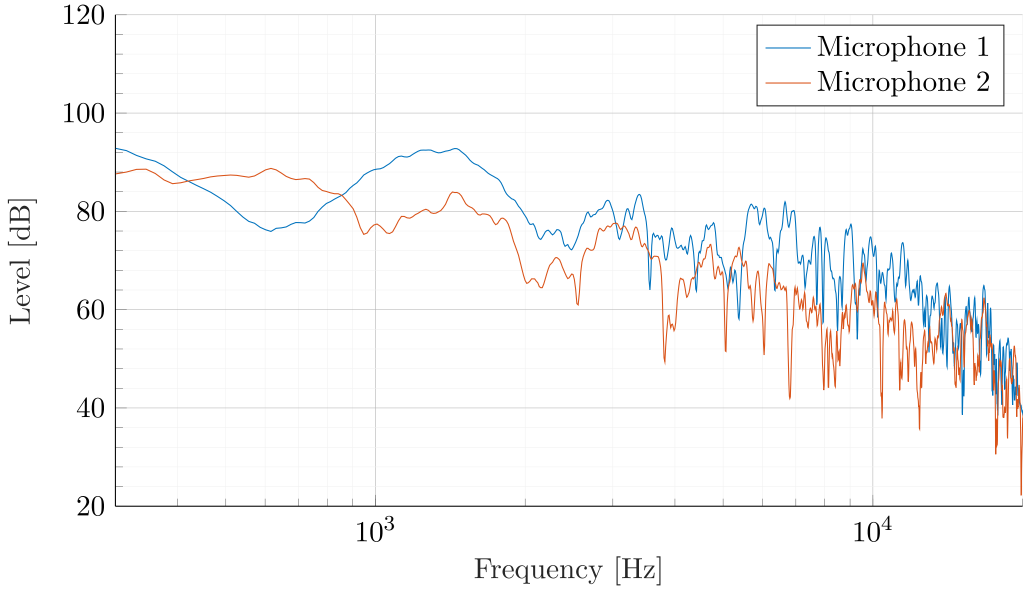


Figure B.2: The graph shows the first transfer function measurement. The $L_{\mathrm{eq},5}$ SPL different between the microphones is 4.41 dB SPL (IR_3)

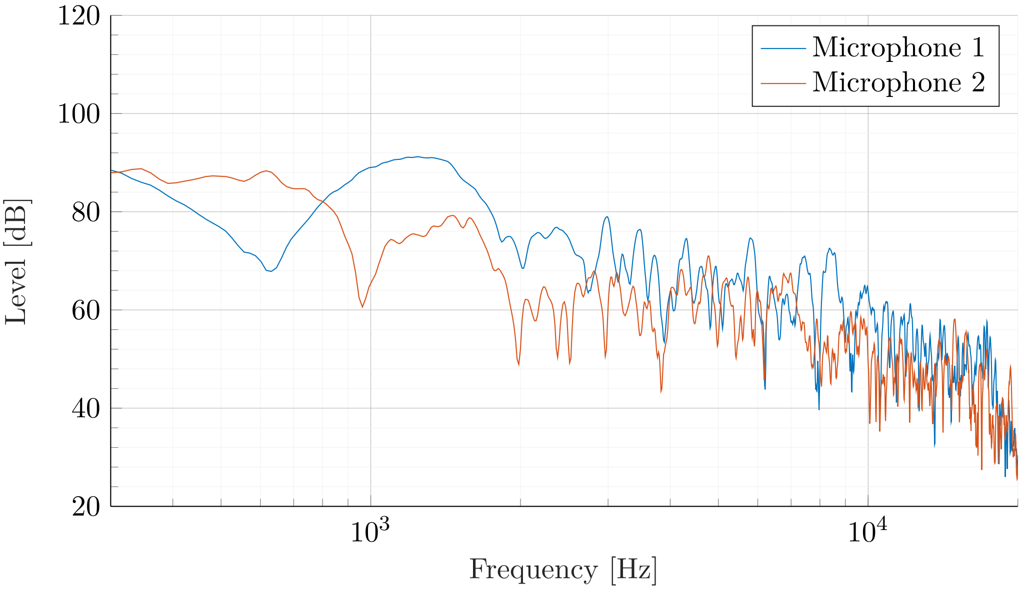


Figure B.3: The graph shows the second transfer function measurement. The $L_{\mathrm{eq},5}$ SPL different between the microphones is 4.81 dB SPL (IR_5)

On Figure B.2 and Figure B.3 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

Appendix C

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:

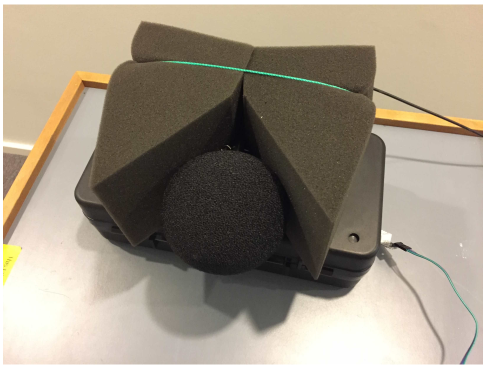
Table C.1: Equipment list

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
Microphone	GRAS 26CC	78186	75582
Preamp	GRAS 40 AZ	100267	75550
Microphone	GRAS 26CC	78186	75582
Preamp	GRAS 40 AZ	100267	75550

Table C.2: Equipment list

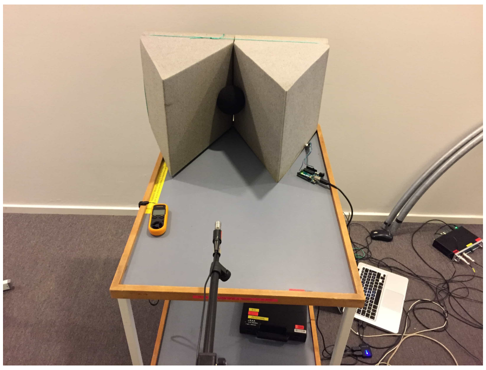
Description	Model	Serial-no	AAU-no
Fan	IMPEGA	-	-
Fan	IMPEGA	-	-
Windscreen	GRAS AM0069	-	-
Windscreen	GRAS AM0069	-	-
large foam wedge	-	-	-
Small foam wedge	-	-	-
Rockwool bat	-	-	-

(a) The picture shows the measurement set up

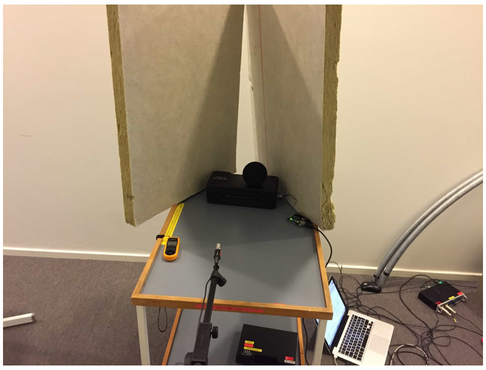


(b) The picture shows the microphone covered with windscreen and the Small foam wedge wind stopper configuration. This configuration is defined as configuration one

Figure C.1: The figures shows the measurement set up and wind screen configuration one



(a) The picture shows the microphone covered with windscreen and the large foam wedge wind stopper. This configuration is defined as configuration two.



(b) The picture shows the microphone covered with windscreen and the rockwool wind stopper. This configuration is defined as configuration three.

Figure C.2: The picture shows the windscreen configuration two and three

Test procedure

1. The materials are set up as in Figure C.1a 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.

Results

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

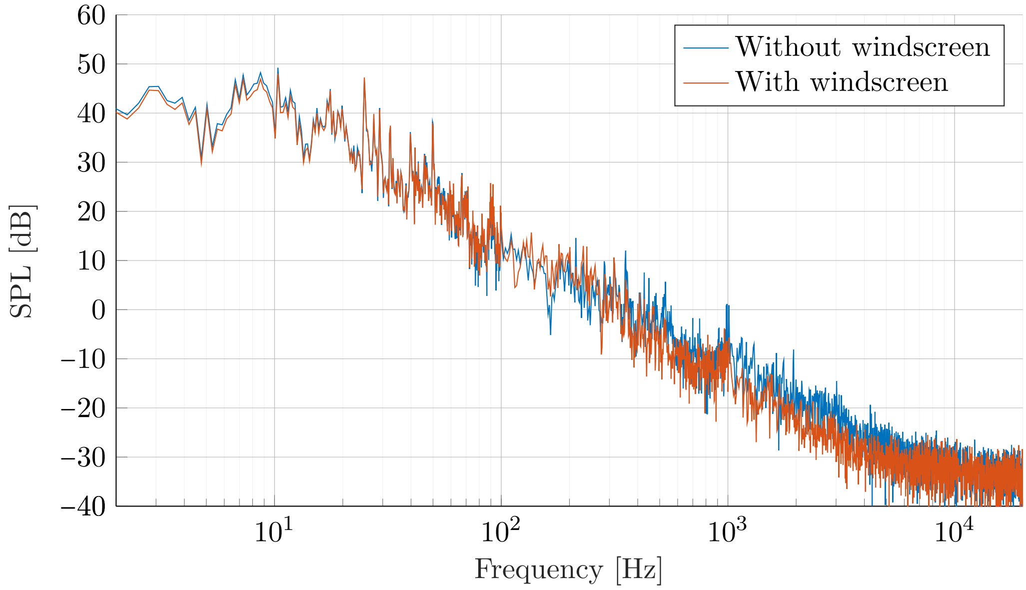


Figure C.3: The graph shows the frequency content without the fan activated

The Figure C.3 shows the frequency content in the measuring area without the fan activated for both microphone and the reference windscreen configuration.

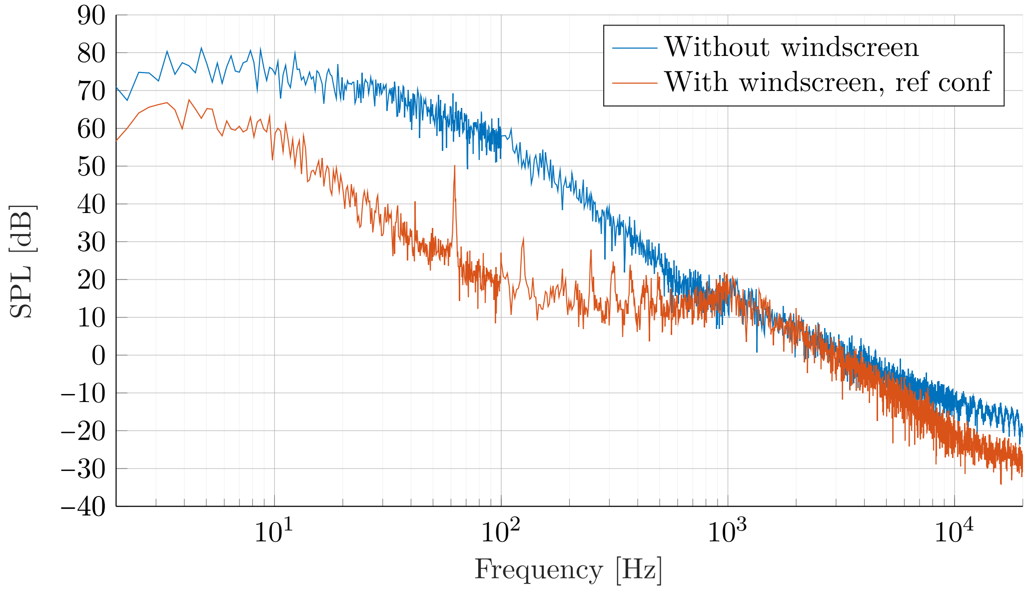


Figure C.4: The graph shows the frequency content with the fan activated

The Figure C.4 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

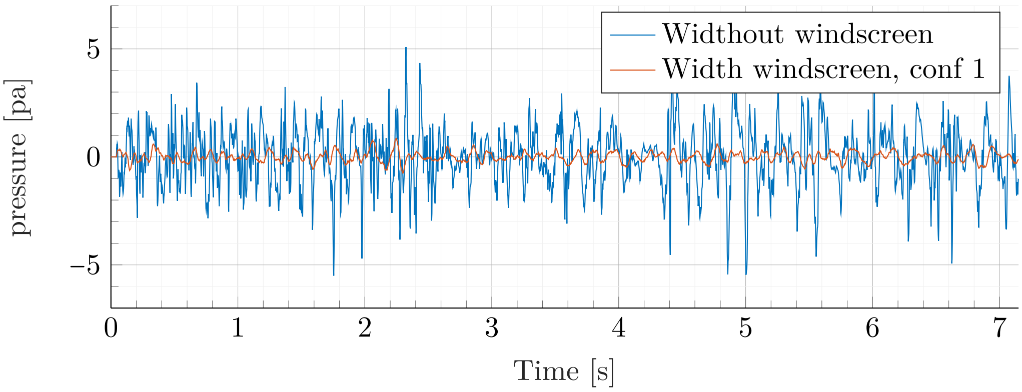


Figure C.5: The graph shows one of the time measurement with configuration one

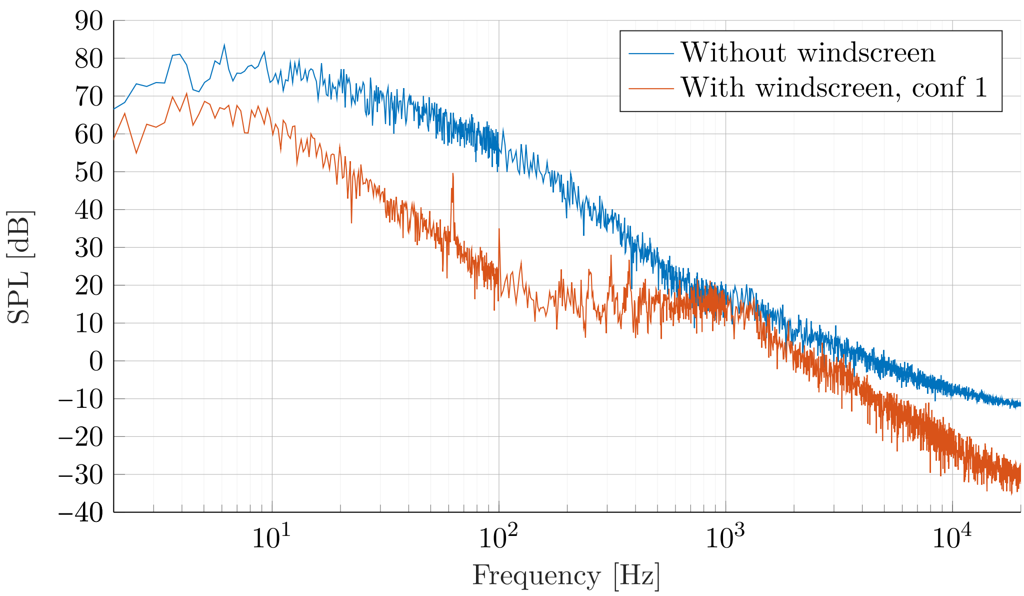


Figure C.6: The graph shows the frequency content of the measurement with configuration one

The Figure C.6 and Figure C.5 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.

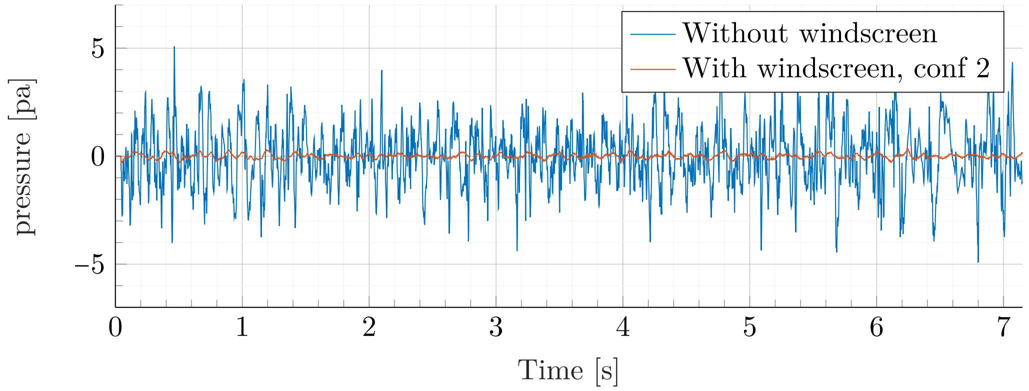


Figure C.7: The graph shows one of the time measurement with configuration two

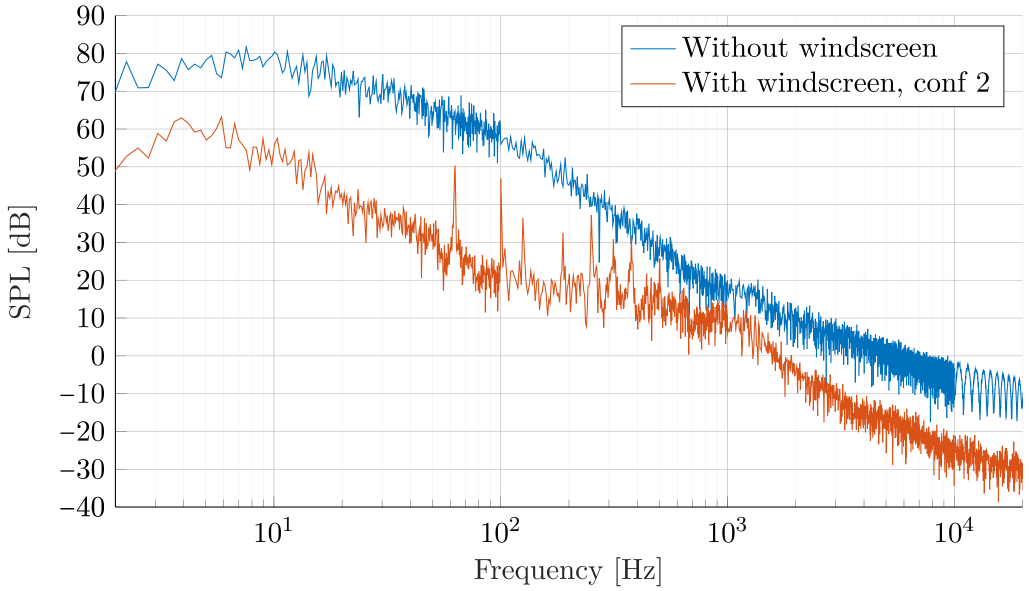


Figure C.8: The graph shows the frequency content of the measurement with configuration two

The Figure C.8 and Figure C.7 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.

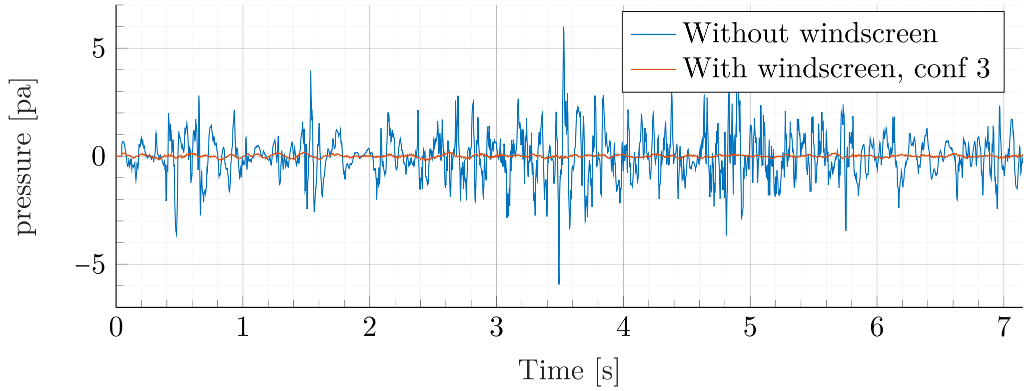


Figure C.9: The graph shows one of the time measurement with configuration three

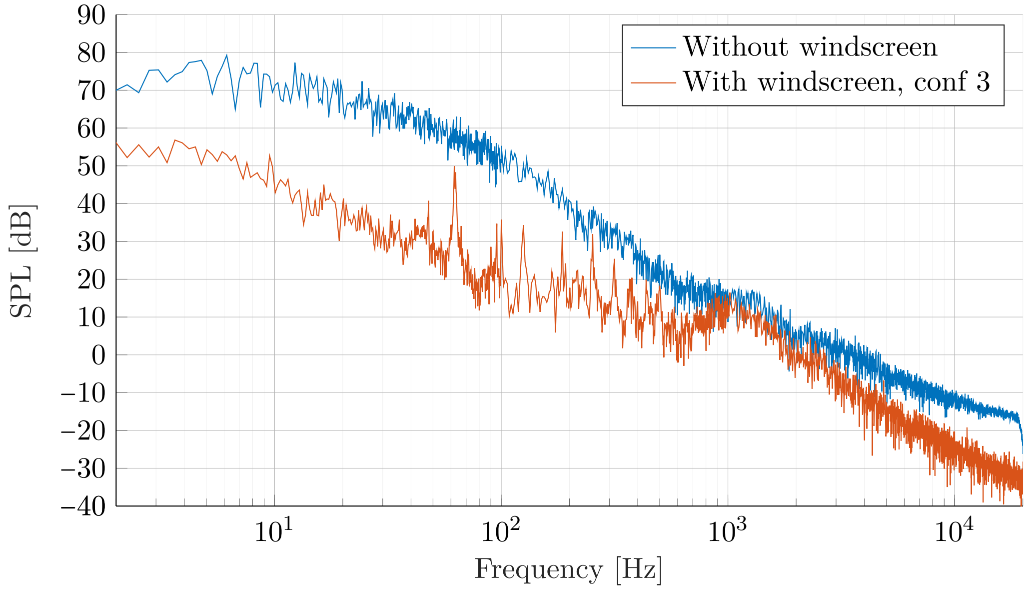


Figure C.10: The graph shows the frequency content of the measurement with configuration three

The Figure C.10 and Figure C.9 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

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