Sound control in windy weather

Master Thesis

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Electronic Engineering and IT
Aalborg University
http://www.aau.dk

AALBORG UNIVERSITY
STUDENT REPORT

Title:
Sound control in windy weather

Theme:
Signal Processing and Acoustics

Project Period:
MSc, 10th Semester 2019

Project Group:
Jonas Buchholdt

Participants:
Jonas Buchholdt

Supervisor:
Sofus Birkedal Nielsen

Number of Pages: 51

Date of Completion:
?th June 2019

Abstract:

Will come later

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* .

For citations, the report employs the Harvard method. If citations are not present by figures or tables, these have been made by the authors of the report. Units are indicated according to the SI standard.

Aalborg University, March 12, 2019

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Contents

Glossary

FDTD Finite-Difference Time-Domain. 15

FOH Front Of House. 9, 24, 25

PA Public Address System. 7, 8

SPL Sound Pressure Level. 7, 8, 9, 10, 11, 16, 17, 19, 20, 22, 24, 25

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 explore the challenges of producing sound in an outdoor environmental. 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 affect 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 important 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 the large number of audience without reinforcement of the information. The reinforcement is nearly always done from a stage with a large 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 can be seen in Figure 2.1

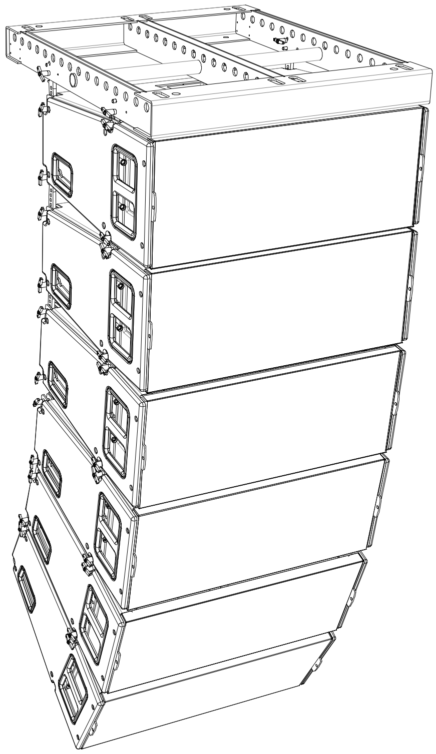

Figure 2.1: The figure shows an illustration of a KUDO line source array from L-Acoustics [L-Acoustics]

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 the line source array module design is that the coupling between the line source element can be controlled. With an optimized control system of the line source array, the audience area can be covered with sound such that all audience can hear the information. An optimized line source array has, for example, an optimized 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.

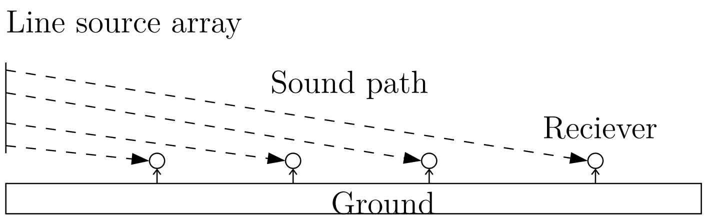

Figure 2.2: The figure illustrate the concept of outdoor PA venue

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. This indicates that the signal to every line source element has to be adjusted with respect to the coverage area. This is necessary because the wave amplitude decay with distances and 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 local pressure divination of the instantiates atmospheric pressure. The atmospheric pressure, therefore, set a lower bound on the condensation while very high pressure change the speed of sound and distord the wave as it propagate. To communicate the information without introducing distortion by the lower limit, the maximum amplification is therefore limited by the lower bound of the atmospheric pressure. Luckily the pressure near the ground is typically 101.325 kPa or a maximum sound pressure of 194 dB SPL. The movement of the particle in the air depends on the medium in the air. The medium in the air is not constant and varies over time with respect to 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 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 mix on experience. The aspect of being an audience depends on where the audience is with respect to 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. Between the delay line towers, the low frequency spectrum is still good and audible but something happens to the high frequency. Often the high frequency disappears for few second 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 thesis will focus on answer this question.

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.

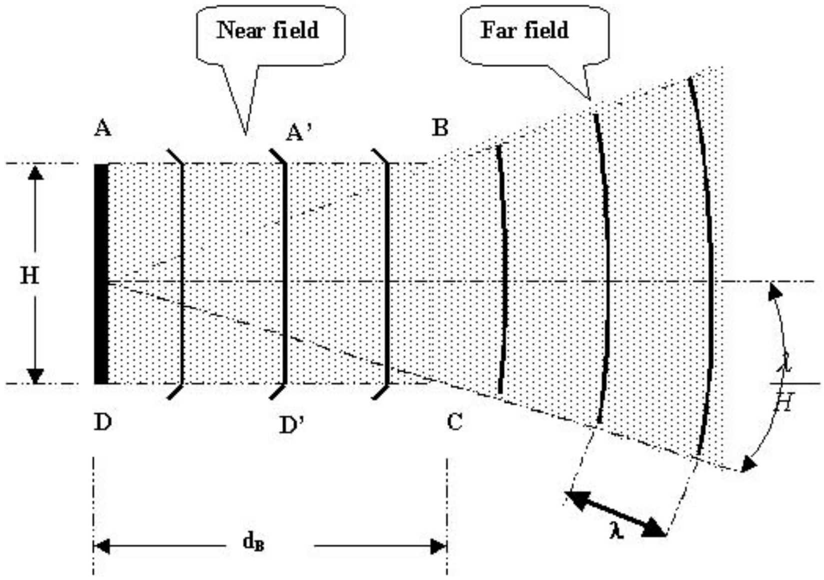

Figure 2.3: The figure shows horizontal cut of a SPL radiation pattern of a line source array [Bauman et al., 2001].

As it can be seen in Figure 2.3 as the wave travels in the near-field the wave is considered as plan but as the wave is propagates intro far-field the propagation is only plan in a limited hight. In the far field the SPL coverage area is four times higher when the wave have travelled the doubled distance. Therefore the SPL is four times less at the double distance in the far-field.

2.3 Homogeneous atmospheric conditions

The aim of this section is 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 Humanity and temperature impact

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

Lowpass effect The effect of humanity and temperature act as a lowpass filter, where the low frequency remains without any additional attenuation. In other words, attenuation in the high frequency range per doubling of distance depends not only on the spreading loss but also on 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 frequency attenuation with respect to the distance, temperature and humanity. The article [Corteel et al., 2017] gives some examples of attenuation at a distance of 100 m. The article shows, if humanity increases proportional to the temperature, the lowpass effect is small. If the change in temperature and humanity is the opposite of each other, for example, the 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].

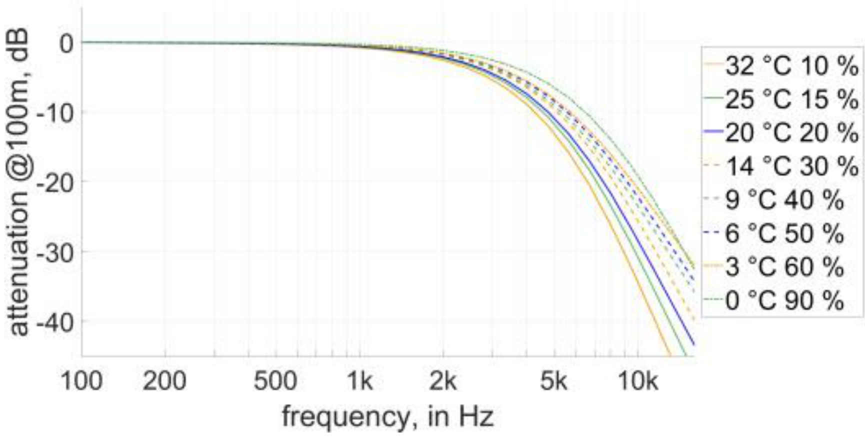

Figure 2.4: The graph shows the attenuation in dB with respect to frequency, humanity and temperature [Corteel et al., 2017].

As shown in Figure 2.4 the attenuation in the high frequency is significant. The attenuation is so markedly that turning up for the high frequency power is not an option, since that will require extreme high pressure driver. Those 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.

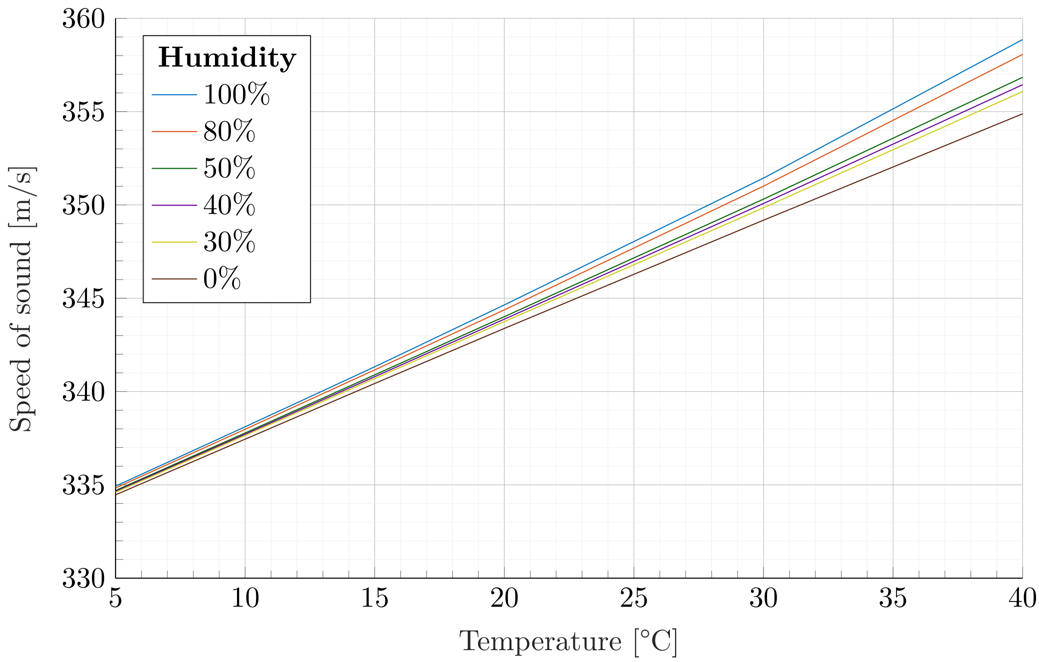

Figure 2.5: The figure shows the increase of sound speed with respect to humanity and temperature [Bohn, 1987]

As it is explained and shown in Figure 2.5 the effect of humidity is neglible compare 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 had changed 4 m/s from 0 % humidity to 100 %

Directivity The directivity in a line source array in the mid and high frequency is always controlled mechanical by a horn because the wave length is short compare to the size of the speaker. At low frequency the wave length is to long to controlled mechanical by a horn and therefore the directional pattern is done via cancellation from a backwards pointing speaker. Both directivity sufferers from the temperature increasment. At the high frequency the main lobe gets narrower when the mechanical horn gets warmer, and the effect is notable when the sun directly heat up the horn. The resend that the temperature of the mechanical horn affect the directivity of the high frequency is that the warm horn surface heat up the air at the mouth of the horn.

This increase of temperature speed up the speed of sound and the wavelength gets shorter. Because the wavelength gets shorter the main lobe gets narrower [Levine et al., 2018]. The directivity of the low frequency is affected as in the high frequency with the temperature increasment. The difference is not as effective as in the high frequency since there is no up heat of surface. The directivity is then not affected due to the sunlight but only the temperature increasment and decreasment. As in the high frequency temperature differences change the wave length and then the length between the speaker in a cardioid subwoofer does not match the optimized wavelength any more.

2.3.2 Wind impact

The wind impact is depending on the angle of the wind direction with respect to the direction of sound propagation. A homogeneous wind is a laminar wind flow with the same homogeneous speed. The following analysis assume homogeneous laminar wind flow from one direction. The analysis is done for both oblique wind and parallel wind with respect to the frontal direction of the speaker driver. 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 complex since the wind deflect the sound waves.

oblique- and crosswind The effect of oblique- and crosswind on sound wave propagation is rarely studied, and the effect seems to be unclear. Few author have addressed the problem in a simulation of traffic noise and by practical experience [de Oliveira, 2012], [Hornikx and Renterghem, 2017], [Ballou, 2008]. They found that the crosswind effect might refracts the wave in the wind direction. Furthermore, they found that the effect is not linear in frequency. The author of [Ballou, 2008] 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 the homogeneous cross wind effect on a omnidirectional source. The author of [Prospathopoulos and Voutsinas, 2007] implemented a ray tracing method with a vector based interpolation as shown in Figure 2.6.

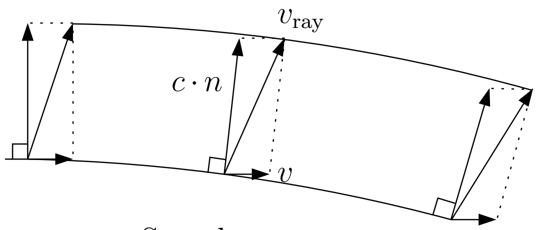

Figure 2.6: The figure shows a geometrical calculation scheme of calculate the resulting wave direction at crosswind [Prospathopoulos and Voutsinas, 2007], [Ostashev et al., 2005]

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 it is 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 wave length therefore depend on the speed of wind and the angle between the wind and the sound propagation. The following Equation 2.2 calculate the speed of sound with respect to the angle and wind

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

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 acceptable condition to a concert the wind speed is less than 20 m/s otherwise the concert is stopped for safety. The following simulation Figure 2.7 shows the simulation result from [Ostashev et al., 2005]. The source is a omnidirectional 100 Hz spherical source while the wind have a constant uniform wind speed from left. The simulation is done in two dimensions.

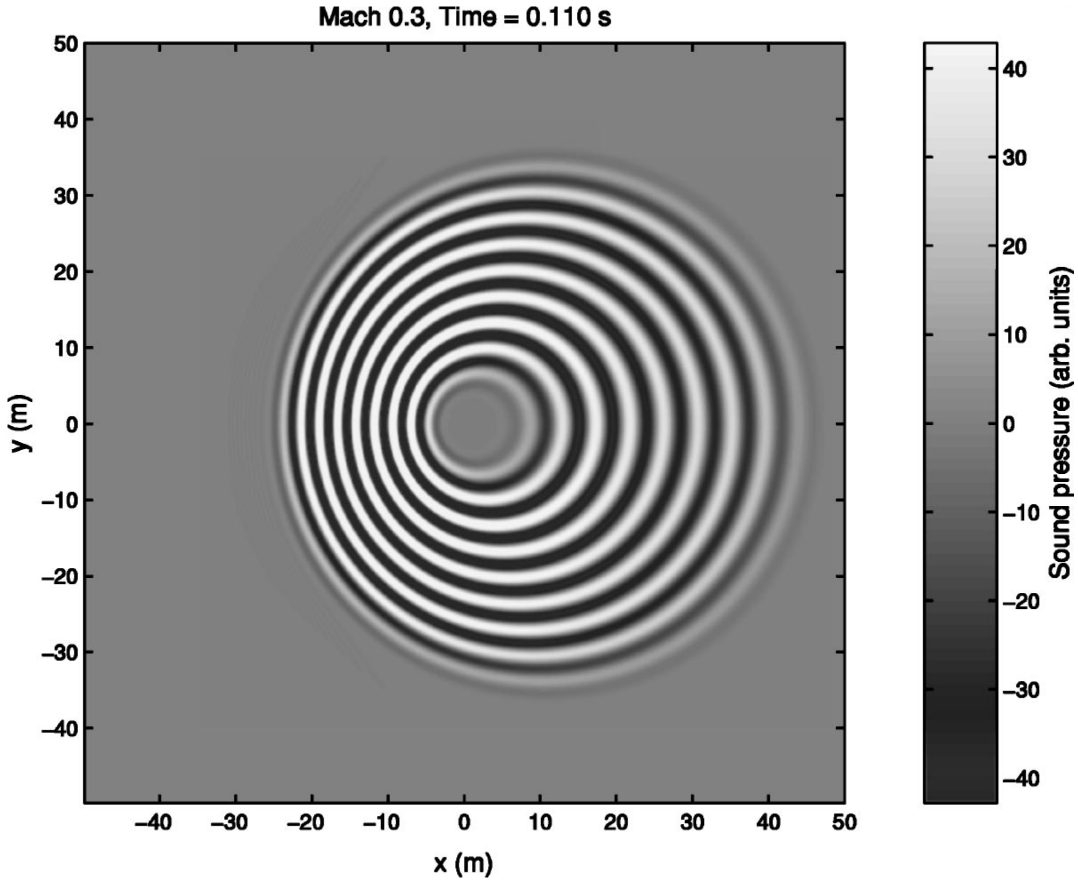

Figure 2.7: The figure shows a simulation of a 100 Hz omnidirectional source with a uniform constant wind speed from left with speed of 102.9 m/s [Ostashev et al., 2005].

It can be seen in Figure 2.7 that the cross wind do not effect the direction of the wave, it only effect 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 attenuation from 4.0 kHz to 16.0 kHz with fixed temperature was 2 dB SPL while going from 54.02 kPa to 101.33 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 pressure generated by the speaker does have a high influence on the sound propagation. The pressure impact often cause distortion of the sound wave. There is two places in the propagation way that can produce distortion with respect to the pressure, at the driver mouth design of the horn [Czerwinski et al., 1999] and the port design of the low frequency driver [Vanderkooy, 1998] and in the sound path. The following description starts with the latte.

Sound path In the sound path there is two main factor that cause distortion in the propagating sound. As descried the sound is condensation and comparation and the condensation can not be less than vacuum. Therefore the the higher bound of SPL is 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 determide by the atmospheric pressure to vacuum, but this is not the only limit for distortion. Very high pressure in the comparation also distorte the sound. At the comparation seriusly signal deterioration orccore if the amplitude is high. The distortion of the comparation is explained by the lack of linear dependency between the particle velocity and the SPL in a sound wave. The SPL increases more that the dencity of the sound wave which causeing the condensation of the sound wave to be stiffer and therefore propagate faster than in the condencation of the wave. This effect cause that the speed of sound to travle faster in the comparation and slower in the condencation. The effect is athat the sine wave transformes to a sawtoorh as it propagate because the harmonics transform energy to the hither hamonic of the sound wave. The distortion made by air propagation is much less than the distortion in the mouth of the speaker which leeds to the next distortion problem produced by high pressure [Czerwinski et al., 1999].

Driver throught and mouth design High pressure in both horn phase plug, seald enclosures and vented enclosures or reflex enclosures for low frequency driver cabinet produce distortion as they act as nonlinear components. The latter produce distortion because high pressure makes air turbulence in the vent. By the optimal design the distortin of turbulent flow can be kept low [Roozen et al., 1998]. The turbulence phenomena does not only cause in the mouth of the low frequency driver, it also occore in the phase plug of the compression driver if the SPL is high [Czerwinski et al., 1999]. The distortion depend on the air's moving mass, the stiffness and the viscus losses on the diagram displasement and the SPL. As the air in the hifh frequency driver comprrssion it become heavier, stiffer and thicker which make nonlinear wave propagation. It typical occore when the compression chamber exceeds approximatly 170 dB SPL . At higher level the particle velocity resistance to teh iar flow increases and the laminar air flow turns intro turbulent air flow. The distortion is also dependning on the length of the horn and the expantion 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 significant 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 limmited as high as possible since the displacement gets lower as the frequency increases.

2.3.4 Ground absorption

In a concert area, ground absorption is complex because there is two very different factors. Before the concert the area is a locally plan area with ground reflection, which can be modeled as explained in [Piercy et al., 1977]. The interesting part is along the concert, but also the complex part, the area along the concert is packed by audience, and is therefore not easy to calculate. The reflection of the high frequency is assumed to be low, because the audience stands close to each other and therefore forms a anechoic acting surface for high frequency. The low frequency driver, also called subwoofer is posisioned in front of the stage on a line or in end fire settings, often with a maximum distance of 2.8 m from acoustical center to acoustical center. The distance between the low frequency driver is determinde by the half wavelength of the highest frequency, such that they radiate a plan wave [Bauman et al., 2001]. Higher distance between acoustical center will cause interference in the low frequency in the audience area. The absorption from the audience in the low frequency is assumed to be low since the size of the audience is much smaller than the wave length.

2.3.5 Homogeneous speed equation

The following Equation 2.3 calculate the speed of sound based on homogeneous temperature and wind speed.

$$c = c_0\sqrt{1 + t/t_0} + u \cdot sin(\theta) \tag{2.3}$$

Where:

c	is the resulting speed of sound	[m/s]
u	is the speed of wind	[m/s]
c_0	is the speed of sound at 0 °C	[m/s]
t	is the temperature	[°C]
t_0	is the temperature at 0 °C (273.15)	[K]
θ	is the angle of wind with respect to the wave propagation	[°]

2.4 Inhomogeneous atmospheric conditions

The aim of this section is 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 path. 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 results in a curved sound path and is called 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 rearly 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 live concert the intersting 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 intersting. 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 asumed 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.

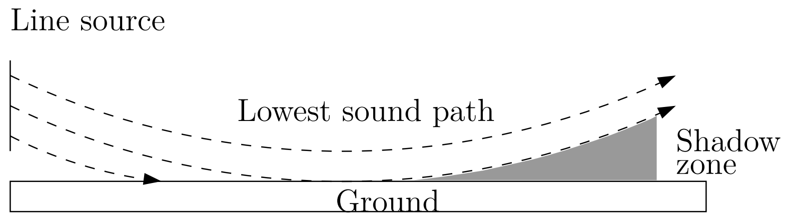

Figure 2.8: The figure illustrate the shadow zone occure from a upwards refraction. A line source speaker array contains of many couplet point sources. Every lowest sound path dashed line indicate the lower directional angle of one poins source in the line source array.

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 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 compare to the wavelength of the low frequency [Piercy et al., 1977]. This theory does not follows the shell's law. Shell's law describe the refraction as a layer change in the medium of propagation. Shell's law of refraction is defined as Equation 2.4

$$\frac{cos(a_1)}{c_1} = \frac{cos(a_2)}{c_2} \tag{2.4}$$

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 it can be seen in shell's law Equation 2.4 the frequency dependency is not a factor and ether shell's law is only a approximation or the frequency dependency does not apper fron only laminar wind flow profile. The article [Piercy et al., 1977] only explorer frequency upto 3.2 kHz but since the refraction depend on the wavelength, the distance of refraction wave might be much smaller for high 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. The measurement is interesting with respect to a concert area and is therefore shown in Figure 2.9

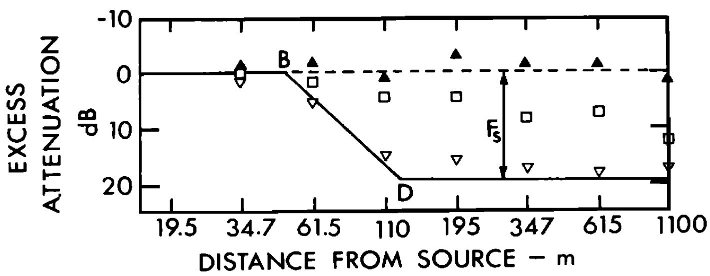

Figure 2.9: Excess attenuation measured for aircraft noise in the 1.2 kHz 1/3 octave band for the ground-to-ground configuration. The vector component of the wind velocity in the direction of propagation for ▲ is 5 m/s, □ a is 0 m/s, and ▽ is −5 m/s. The temperature profile is neutral. F_s is the shielding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

The following two paragraph explains the difference between wind refraction and temperature refraction.

Temperature Temperature decresses with respect to the hight at day time and increases at the night time. The increase or decrease can usually be approximated as a logarithmic function. In the day time, the sun heats the ground even at 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. This phenomena is named lapse where the uppersite 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 moddled as a change of layer for a plan wave. The output angle of the layer change follows the shell's law. Therefore when the temperature profile is logatrimic the layer change is a function of hight and change the wave direction. The wave direction of the descript weather condition result in an upwards refraction. Since the temperature is a scalar quantity uniformly over large area and a function of hight, an identical temperature profile is aplicable all around the 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.

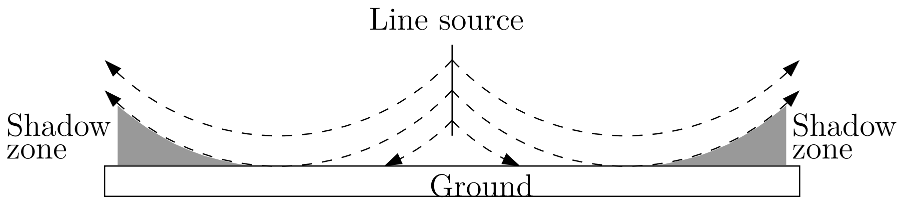

Figure 2.10: Wave refraction in inhomogeneous temperature with lapse profile

When the temperature profile is reversed, the refraction will be 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. 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 shows the phenomena when the wave propagates against the wind.

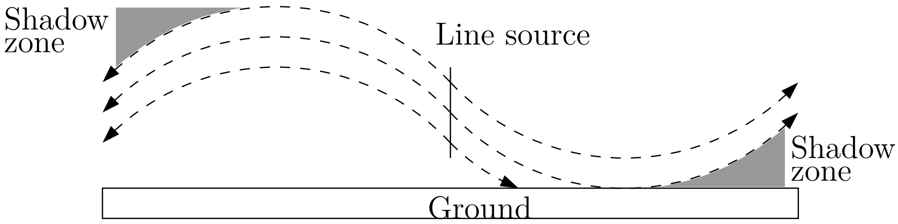

Figure 2.11: Wave refraction in inhomogeneous logarithmically increasing wind profile where the wind gradient points towards left

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 rafraction 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.

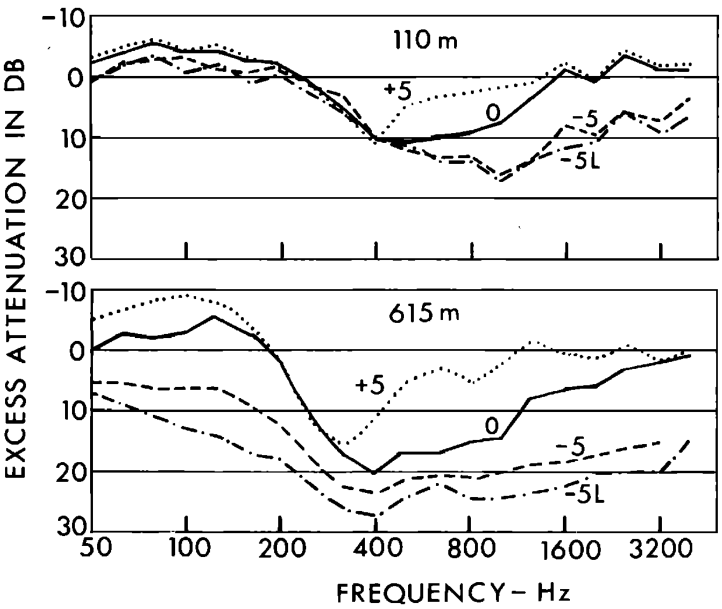

Figure 2.12: Observed attenuation of aircraft noise in a ground-to-ground configuration under a variety of weather conditions. Calculated losses from atmospheric absorption and spherical spreading have been subtracted from the attenuation measured in 1/3 octave bands for distances of 110 m and 615 m. 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 significant amount of sound energy enters the shadow zone in turbulent weather. 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 change the direction of propagation. The wave that enter the shadow zone can be consitered as creeping wave in turbulent senario. The creeping wave will by them self also refract and therefore be parallel to the the other refraction waves. [Embleton, 1996]

Oblique- and crosswind The effect of oblique- and crosswind on acoustical wave propagation in inhomogeneous atmospheric conditions are rarly studied. The author in [Piercy et al., 1977] explain that the refraction is directly zero when crosswind is present, and increase progrative as the direction of propagation deviate from the angle of crosswind.

Since the effect of crosswind on a line source speaker is rarely studied, a measurement in windy condition is preformed. The measurement is done is a large open area used for football and according to [Gunness, 2001]. The wind was considered as strong for 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 one meter above the ground. There was used two microphone, where both was situated 23 m from the speaker, beyond the limiting high frequency directional angle of the speaker. The speaker was placed to propagate in direct crosswind and the microphone was placed on both side of the speaker as shown in Figure 2.13

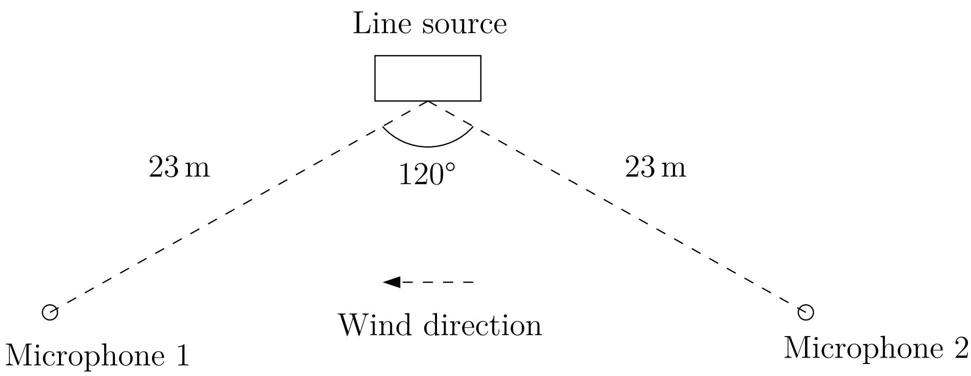

Figure 2.13: The figure shows the microphone position versus the position of the line source

The measurement was done according to the description in Appendix A. The measurement was preformed three times where the first measurement is shown in Figure 2.14. The other two measurement result can be seen in Appendix A and they show same tendency.

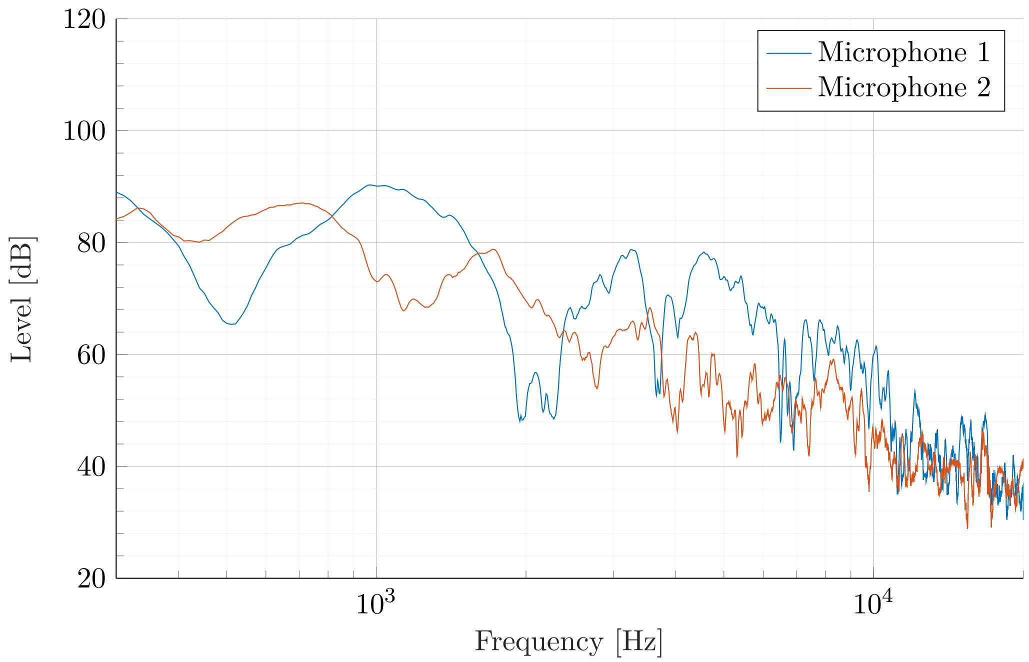

Figure 2.14: The graph shows the average of the first transfer function measurement routine

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 [Embleton, 1996]. Microphone 2 does not have the same strong interference in the low frequency and the general SPL is lower than microphone 1. This indicate upwards refraction.

Turbulent Turbulence is a atmospheric condition where the wind eddies. It often starts with large eddies and prograsively brakes down as a cascade effect to smaller and smaller eddies which only depend on the local region. When the eddies is as small as 1 mm the energy dissepears in viscosity and thermal conduction. A statistically distribution of the eddies is defined as turbulence. The turbulence wind flow is therefore a chaotic and stochastic process by the nature and is pressent 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 at a windy afternoon day and low under the inverse of lapse. Turbulence also often occore near the ground because the ground surface slow down the speed of wind by the resistance 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 comparible to 90° [Piercy et al., 1977]. At this point the phase correlation for each sound path is uncorrelated

2.5 Calibration of sound system

This section analyses the calibration method, which is used by a selection of some Danish sound company. By experience of the author, the hypothesis is that the sound system is calibrated in one point and the microphone is placed just in front of the FOH. The FOH is often a little tent, where the sound engineer controls the sound system. The tent is only open in the direction of the stage and reflection might occur from the tent ceiling to the calibration microphone.

2.6 Sound pressure level measurement doing the concert

2.7 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

Three effect of atmospheric conditions have been observed on the analysis, pure attenuation, lowpass effect and refraction effect

Chapter 4

Problem statement

Based on the knowledge founded in chapter 2 and the concluction drawn from **??** a problem statement can be made. For the rest of the project the following will be the focus.

research the effect of oblique- and crosswind on wave propagation.

4.1 Deimitation

The following delemitation is made for the search for a solution of .

Part II

Test Design

Chapter 5

Design

Part III

Results

Chapter 6

Results

Chapter 7

Discussion and conclusion

7.1 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 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	T4	-	-
Wind measurement tools	Drahtlose Wetterstation	-	2157-45
flying tools	-	-	-

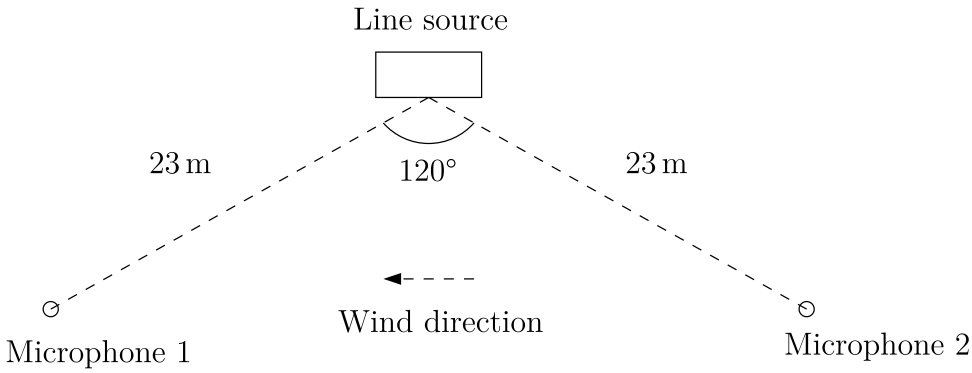

Figure A.1: The figure shows the microphone position versus the position of the line source

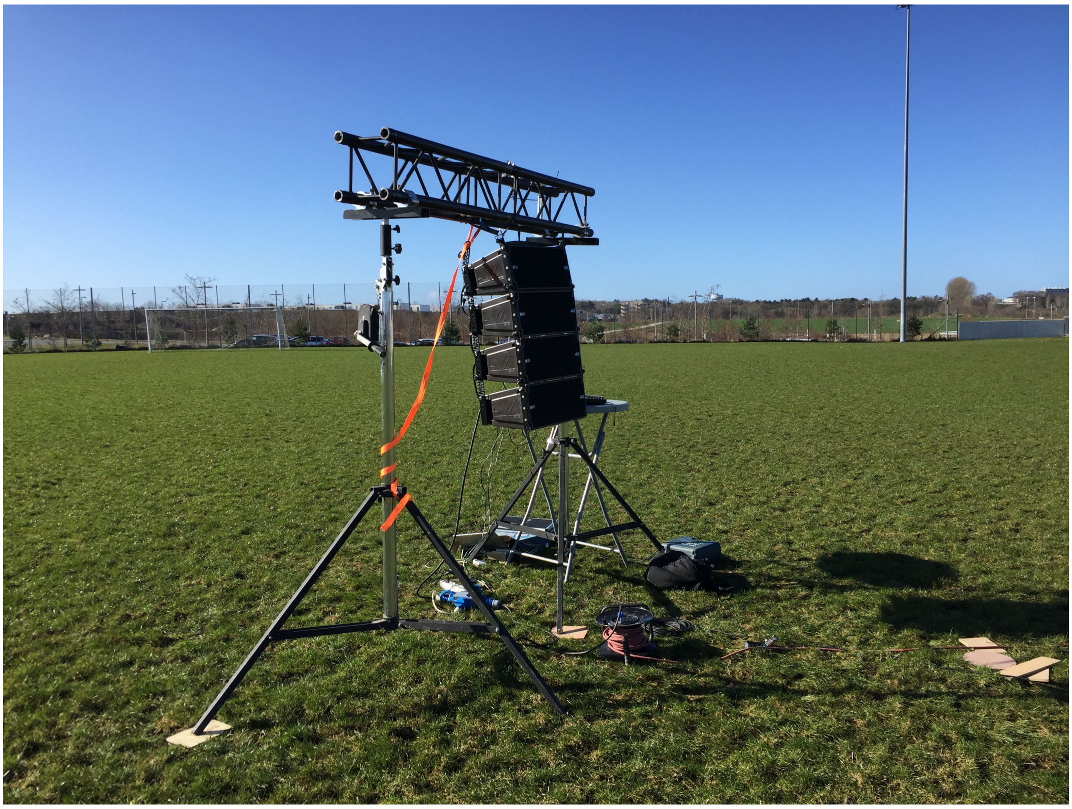

Figure A.2: The picture shows the speaker setup

Figure A.3: The figure shows the wind direction

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 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.

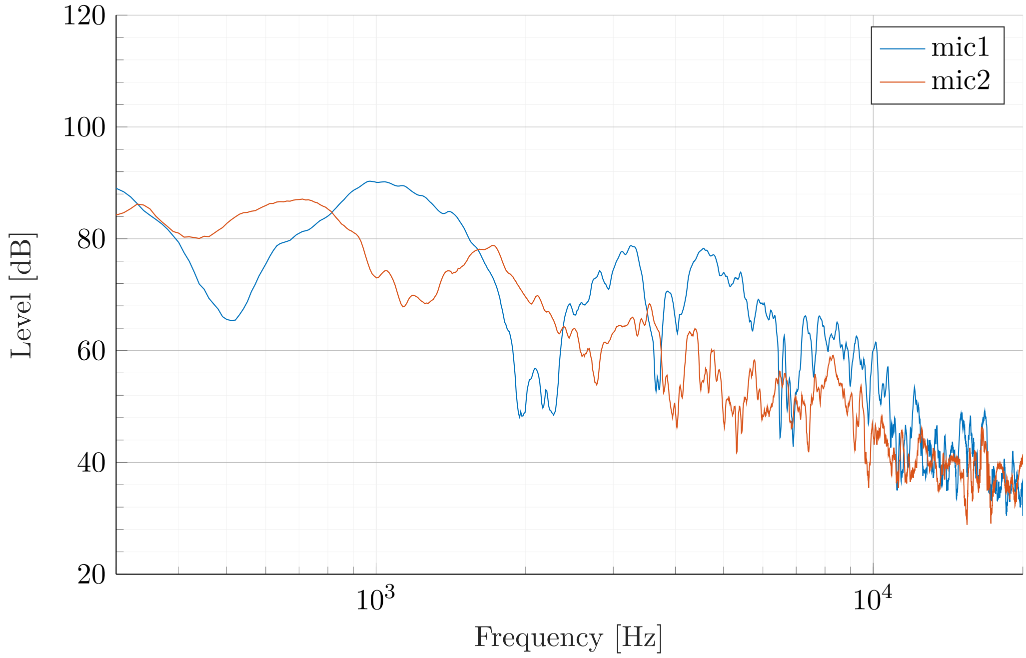

Figure A.4: The graph shows the first transfer function measurement

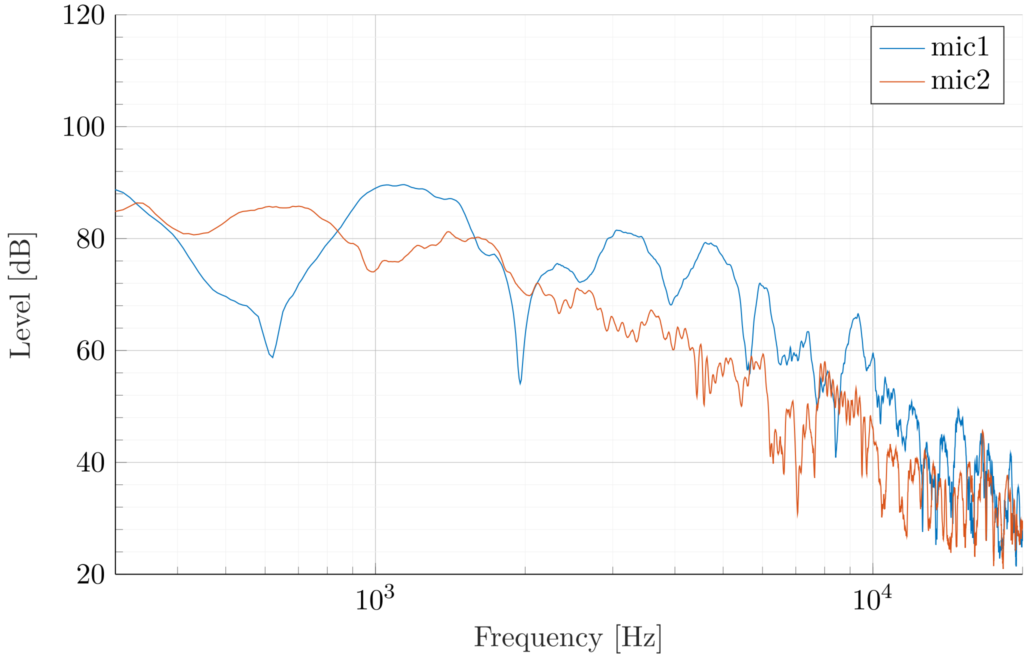

Figure A.5: The graph shows the second transfer function measurement

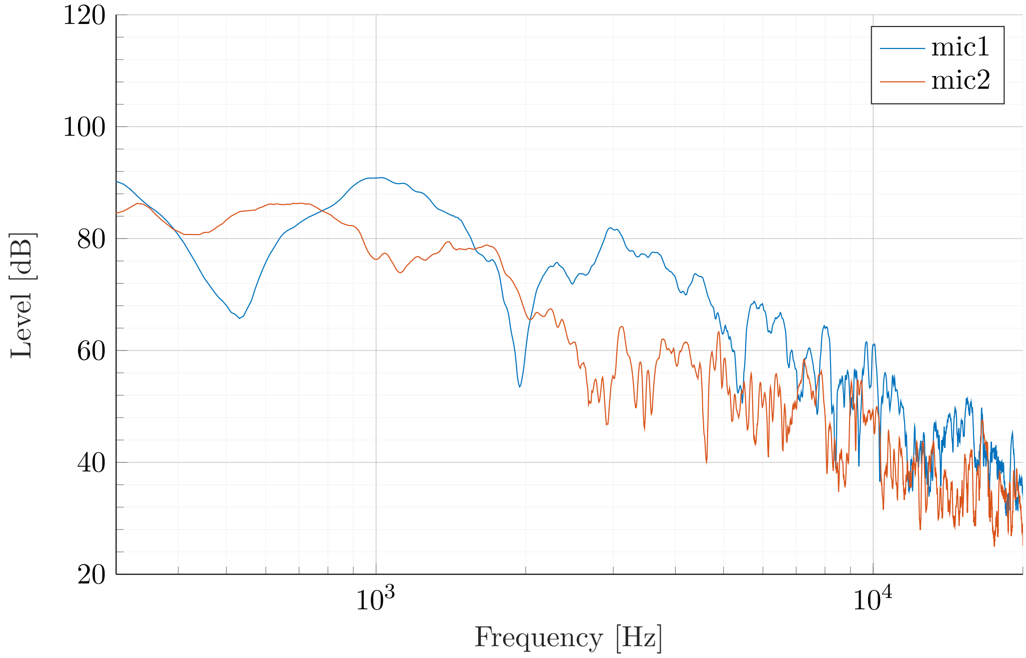

Figure A.6: The graph shows the third transfer function measurement

On Figure A.4, Figure A.4 and Figure A.4 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

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