HELMUT SCHMIDT UNIVERSITÄT
Universität der Bundeswehr Hamburg

Acoustic Communication Between Two Robots Based on the NAO Robot System

Bachelor Thesis

by

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Abstract

The aim of this thesis is to research the basics for a communication over acoustic signals between NAO robots. This communication system needs to fulfil the requirements which are laid down by the rules and environment of the Standard Platform League (SPL) of the RoboCup. In this league, two teams of five NAO robots play soccer against each other. This thesis will give the reader an overview of the different leagues of the RoboCup and what the recent developments in the SPL are concerning acoustic methods of communication and interaction.
There will be a thorough investigation of the characteristics of the microphones, loudspeakers and noise produced by the robot itself. The data that is acquired will help to understand the capabilities of the robot system.
Finally, some aspects will be discussed that are common in a communication system. Different methods will be introduced and it will be evaluated how well they might be suited when implementing a system.

Statement

Hereby I, Florian Bergmann, do state that this work has been prepared by myself and with the help which is referred within this thesis.

Hamburg, 1st of January 2015

Foreword

This bachelor thesis was written and researched at the Helmut Schmidt University. I would like to thank Prof. Dr.-Ing. habil. Udo Zölzer for the opportunity to use the laboratory and his interest in my topic. In addition, I am thankful for Dr.-Ing Martin Holters who supervised me and helped me with guidance and advice for my thesis.

Furthermore, I would like to thank Prof. Dr.-Ing. Gerhard Bauch for agreeing to be my supervisor at the Hamburg University of Technology.

My thanks also goes to Dipl.-Ing Sebastian Kraft, Dipl.-Ing Marco Fink and M. Sc. Felix Eichas for their help and counsel when I had questions about the topic or the equipment of the laboratory.

Special thanks is due to M.Sc. Patrick Göttsch who helped me to take the necessary steps that allowed me to write a bachelor thesis about the NAO Robot System and the Roboting@TUHH e.V association which let me use a NAO robot for the duration of my thesis.

Finally, I would like to thank all research assistants from the faculty of Allgemeine Nachrichtentechnik and the members of the NAO team HULKs for their support and interest in my topic.

Hamburg, 2015

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Chapter 1

Introduction

The communication system that this thesis will investigate needs to be compatible to the rules of the RoboCup. In order to be able to understand what the conditions are, there will be a short outline of what the RoboCup and the Standard Platform League (SPL) entails. In addition, the motivations for this thesis and its relevance to robotic soccer will be presented. This includes an explanation why soccer is of interest when it comes to robotics and the reason for dealing with acoustic communications for the SPL. Finally, a few projects that already use acoustics in the RoboCup will be described briefly. These use acoustic methods for communicating or localizing of its own position.

1.1 RoboCup

The RoboCup is a competition that was founded in 1997. The competition is for robots and artificial intelligence research where teams from all over the world compete in different leagues. Even though RoboCup stands for “Robot Soccer World Cup”, there are competitions that deal with logistical problems (RoboCup Logistics League), rescue missions (Rescue Robot League), human interactions at home (RoboCup@Home) and more.

There are several different soccer leagues. Next to a purely simulation league, there are leagues for different sizes of robots that are built by the teams (kid-size, adult-size and more), as well as the Standard Platform League which is the focus of this thesis. The common goal of the RoboCup can be found on its website [Fed15] and states:

"By the middle of the 21st century, a team of fully autonomous humanoid robot soccer players shall win a soccer game, complying with the official rules of FIFA, against the winner of the most recent World Cup."

1.1.1 Standard Platform League

The Standard Platform League is one of the major leagues of the Robocup. In the SPL, the hardware is given and the teams only program the robots. Therefore, the software of the robots is the main aspect in this league and the team with the better algorithms and technique will win. Every team uses the humanoid NAO robot system that is developed by the French company Aldebaran.

The rule set of the SPL for the year 2014 can be found under [Com14]. The rules for 2015 have not yet been release and as of the 19th of January 2015, the rule for Acoustic Communications states:

"There are no restrictions on communication between the robots using a microphone or a speaker. The only exception is that the coaching robot may not communicate acoustically via its microphone or speaker."

This is subject to change and this thesis will therefore only take into account the rule that is stated above. The coaching robot is not playing in the game but sits on the side of the field and gives directions to its team members. Since the only restriction is about the coaching robot and not any players, there are no limitations on what the playing robots can do.

1.2 Motivation

The research and development of building robots has increased in the last few decades and their presence becomes more common every year. Next to industrial robots that are used for production, there are already robots that people use at home. These are, for example, robots that vacuum the floor or cut the grass. Robots have many use cases like making a coffee in the morning or lifting heavy materials that a human can not lift. We are still at the early beginning of discovering the capabilities of robots and developing better software and hardware that allows robots to be able to handle more and more complex tasks. One aim is to make robots less static by improving their ability to adapt to their tasks and react to unknown situations.

1.2.1 From Chess Computer to Robots Playing Soccer

It has always been the challenge of mankind to make machines that can compete with humans. This began with the development of chess computers in the mid-70s. Chess was an optimal game for a computer because there are a finite number of possible movements. Furthermore, chess is a game that is turn-based and a computer has time to calculate its next step. The early chess computers were not able to defeat a human due to limited processing power. However, due to the rapid development of computers it became more and more easy to calculate the effects of every single movement. In May 1997 the chess computer *Deep Blue* was able to defeat the world champion Garry Kasparov. For more information about

Figure 1.1: *Deep Blue [IBM15]*

Deep Blue, see [IBM15]. Nowadays, a computer is able to defeat most humans easily because our capabilities, to calculate that many solutions, are limited. Due to its nature, chess is a game that machines can play easily. A different type of challenge had to be found to measure the capabilities of machines.

Scientists tried to find a task that is easy for humans but would potentially be harder for a computer. This lead to soccer played by robots. Any robot can understand the rules of the game as easily as humans but what is not easy for a machine is to react to a situation that can not be anticipated. In soccer, the players have to react quickly to many different situations. Unlike chess, soccer is played in real time and a robot does not have a lot of time to react. A lot of situations that occur can not be known beforehand and in order to win against humans, the robots need to have an artificial intelligence that can react to arbitrary situations.

1.2.2 Why Acoustic Communication?

Robots usually communicate over wireless or wired network. Network Communication has a high bit rate with little distortion and a low latency. Whereas acoustic communication has a limited bandwidth, is exposed to a lot of noise and the signals have a higher delay. This brings up the question why it is worth to consider an acoustic communication over a communication through a wireless network.

There are two main aspects that favour the use of acoustic communication for robots. The first one is the unavailability of a network. This could be due to physical limitations, as well as, temporary outages of an existing system. Experience has shown that the wireless network at championships of the RoboCup might not be available during a game due to technical difficulties. In such an event, the robots are not able to communicate with each other and the result is that every robot tries to get the ball. Communication over acoustic signal can be used as a fall-back method to prevent chaos on the field.

The second aspect of acoustic communication is the human side of the challenge. It is desired to make the rules for the games more and more like real soccer for humans. Therefore, the robots have to be able to interact and play like humans. Due to this, the final goal would be to not allow the robots to use anything that a human can not. Since we communicate by acoustic means, the robots should be able as well. Recent changes in the RoboCup have shown that acoustics are now becoming more and more important for the robotic soccer leagues. For some time, the games have been started by blowing into a whistle. So far, there was no advantage for the robots to listen to it because they got a wireless message at the same time, that tells them that the game starts. The whistle was only for the crowd that watched the game. However, starting 2015 there will be a time delay between the whistle blow and the wireless signal. A team that is able to detect the whistle has an advantage over teams that are not.

1.3 Recent Projects with Acoustic Methods

There have only been a few projects that deal with the use of acoustic signals and sound processing in the RoboCup. However, the recent rule changes will probably lead to a rise of development towards acoustics. Here I will present some of the projects of other teams that have been worked on recently.

1.3.1 Sound Recognition Challenge 2014

During the WorldCup 2014 in Brazil, one of the *technical challenges* was to react to a given acoustic signal and a kick-off whistle blown by a human. For the first part of the challenge, four robots were placed in different positions from the loudspeaker and a known signal was played. Every time the robots detected the signal, they were supposed to lift up one of their arms. Due to the well known signal and little noise, it was fairly easy to detect it and a simple approach was to perform a Power Spectral Density (PSD) analysis and look for a rise in the power at the frequencies which relate to the audio signal that was given.
The second part of the challenge was a little more difficult because a whistle blow is never the same and the rise in the power can occur at slightly different frequencies. However, the former approach can still be used with a higher tolerance. While there were teams that could not detect the signals or had false positives, most of the participating teams were able to solve the challenge correctly.

1.3.2 Acoustic Communcation

The Austrian-Kangaroos team from the Vienna University of Technology and the University of Applied Sciences Technikum Vienna presented the status of their acoustic communication during the Open Challenge of the RoboCup 2014. The Open Challenge allows teams to show anything that they researched that will benefit the SPL. For this challenge, they had one NAO robot facing a ball and three robots with their back to the ball. The ball was moved to one side of the robot and the robot then communicated its position to the three listeners over sound signals. In order to show that they understood where the ball was, they lifted their respective arm.

According to their published description paper [Kan14], a Frequency Shift Keying with error correction was used. This allowed a data rate of 8 bps during the demonstration.

1.3.3 Localisation over Acoustics

In 2008 David Becker and Max Risler from the TU Darmstadt published a paper about autonomous robots using an acoustic pattern recognition that is based on the principle of CDMA to calculate the distances between the robots. The robot platform that was used is the well known Aibo robot dog by Sony. For further information please see their paper [BR08].

Chapter 2

Hardware

This thesis focuses specifically on the NAO robot system that is now used in the SPL. Since 2008 it replaced the well known AIBO robot from Sony that used to play in the SPL. The NAO robot is a humanoid robot system that is mainly build for research and development purposes. In this chapter, the hardware of the robot will be described. Due to the different available versions of the NAO robot, this will mainly focus on the hardware of the NAO V4 and the NAO V5 - Evolution.

2.1 The Robot

The NAO robot system is developed and produced by a French company called Aldebaran. It is used in different fields for research purpose and for interactions with humans. Next to many universities, it is also used to teach autistic children in the Autism Solution for Kids (ASK) program which was created by Aldebaran. The robot has a weight of about 5.2 kg, a height of 573 mm and a shoulder width of 275 mm.

The first version was released in 2008 and the hardware has continuously been upgraded since then. The latest versions feature a variety of sensors, like two HD cameras that are used to find objects and navigate on the field. It has multiple integrated tactile and pressure sensors, an ultrasonic sensor, two infra-red emitters and receivers, accelerometer and a gyroscope. The CPU is different depending on the version that is used but the latest NAO (V5 Evolution) features an Intel ATOM 1.6 GHz that runs a customised Linux operating system. It has a second CPU that is inside the torso to control the motors. Depending on the version, the robot has up to 26 degrees of freedom. For this thesis, the two loudspeakers and the four microphones are of main interest.

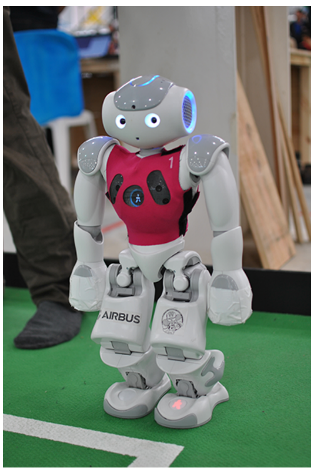

Figure 2.1: *NAO [HUL15]*

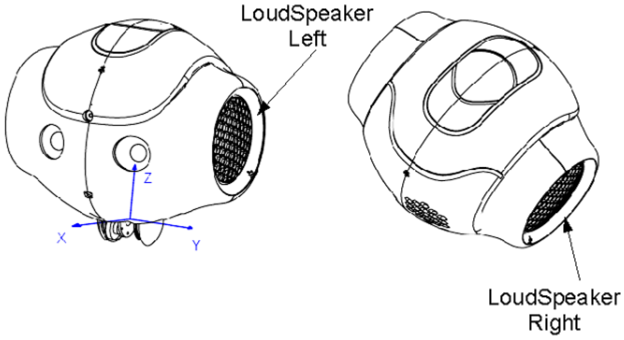

Figure 2.2: *The location of the two loudspeakers on the NAO head [Ald15a]*

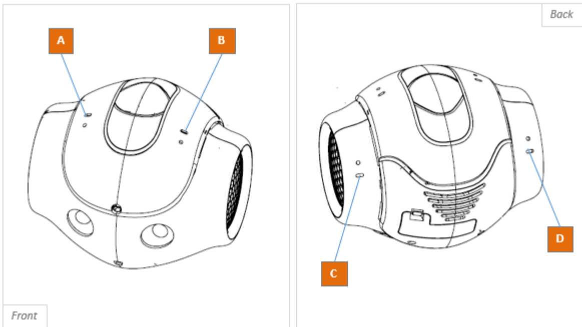

Figure 2.3: *The location of the four microphones (V5 - Evolution) [Ald15b]*

2.2 Loudspeakers

The NAO robot has two loudspeakers that are on the sides of the head which allows stereo playback. This is shown in Figure 2.2 for the NAO V5 - Evolution. The recommended sampling rate is 48000 Hz but others are available as well. According to data sheets, the loudspeakers of most models have a frequency range of up to 20 kHz. The exact coordinates are give on the Aldebaran Website [Ald15a].

2.3 Microphones

There are four microphones located in the head of the NAO. Depending on the version, they can be at different positions but they are chosen so that the localisation of the direction of sound is possible. As an example, the positions of the NAO V5 - Evolution can be seen in Figure 2.3. This is interesting for other acoustic projects where the direction of the sound is helpful but for a basic acoustic communication it is not needed. The microphones of the available versions of the NAO robot system have slightly different specifications. The version that is mainly used for this thesis will be a NAO V4 but due to the different specifications the NAO V5 - Evolution might be used as a comparison for measurements. On the official documentation it is stated that the NAO V4 has a frequency range of 300 Hz - 8 kHz and the range of the NAO V5 - Evolution is 150 Hz to 12 kHz. In this thesis, all four channels with a sampling rate of 48000 Hz will be used for sound processing. The exact positions of the microphones can be found on the Aldebaran Website [Ald15b].

Chapter 3

Acoustic Measurements

In this chapter, the capabilities of the hardware will be measured. The knowledge of this will help to construct an appropriate communication model. If the properties of the communication channel are known, an adequate modulation scheme can be chosen to achieve an efficient and error-free data transfer without collisions.

For all measurements the configurations have been chosen as follows. The sampling rate for the microphone and loudspeakers were set to 48000 Hz. The internal amplifier of the loudspeakers and the microphones were both set to -4.5 dB. This is because a signal that was played from the loudspeakers with a gain of 0 dB was felt to be unpleasantly loud. Whereas, a signal with the chosen gain was judged to be acceptable for an audience. The gain of the microphones was chosen as an arbitrary reference value but not deterministically evaluated.

3.1 Impulse Response of the Loudspeakers

In a system with loudspeakers and microphones, the capabilities of the loudspeakers will be the deciding factor for the frequency range that can be used to transfer data. In addition, measuring of the loudspeakers provides more comprehensible data for an evaluation. The impulse response of the loudspeakers gives precise information about its linearity and its invariance over time.

The impulse response only describes a system under the conditions that were present while it was measured. Therefore, it is desired to measure under conditions that come as close as possible to a perfect system. To reduce background noises, the measurement of the impulse response was executed in a sound box. The robot was placed on the ground and standing still. A microphone was closely positioned next to the robot and pointed at one of its speakers.
To find the impulse response an exponential sinus sweep according to [Far00] was created in Matlab. The chosen values for the angular frequency were $\omega_1 = 0.0026$ and $\omega_2 = 2.618$ to cover a frequency range of 20 - 20000 Hz. For the amount of data points L, a value of 480000 was chosen to reflect a signal of 10 seconds at a sample rate of 48000. The signal was then played from one of the loudspeakers and recorded.

The spectrogram of the recorded signal can be seen in Figure 3.1. There is some harmonic distortion but the energy of it is low compared to the energy of the exponential sinus sweep. Furthermore, there is a noticeable amount of distortion in the lower frequencies that is due to vibrations of the shell that occurred during the playing of the signal.

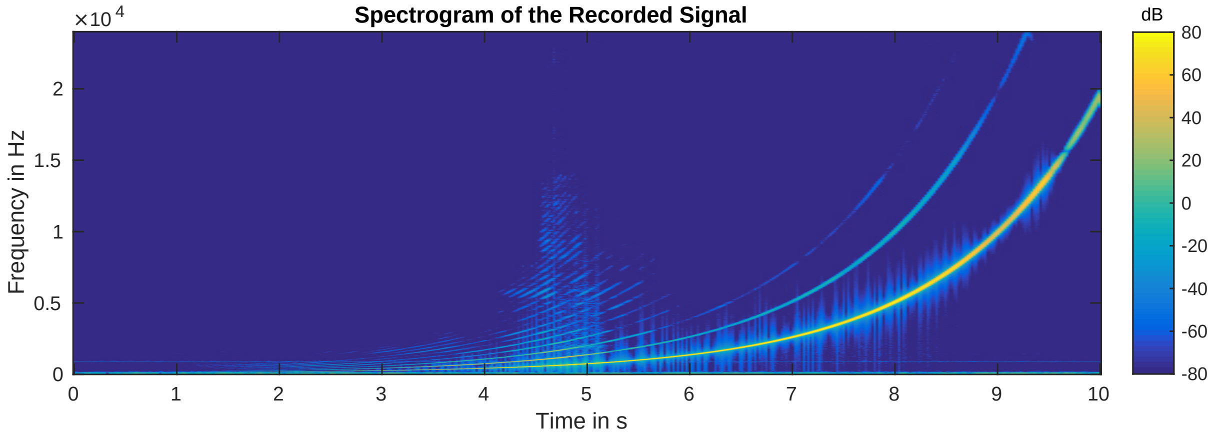

Figure 3.1: *The spectrogram of the exponential sinus sweep recorded by the NAO.*

In the graph, one can see that the middle and high frequency range of the exponential sinus sweep has higher energy than the lower frequency. This shows that the loudspeakers were not built for very low frequencies and reach their limits there. Only in the very high frequencies (above 15000 Hz), is the signal weak again.

The recorded signal (Figure 3.2) and the inverse of the exponential sinus sweep signal were convolved.

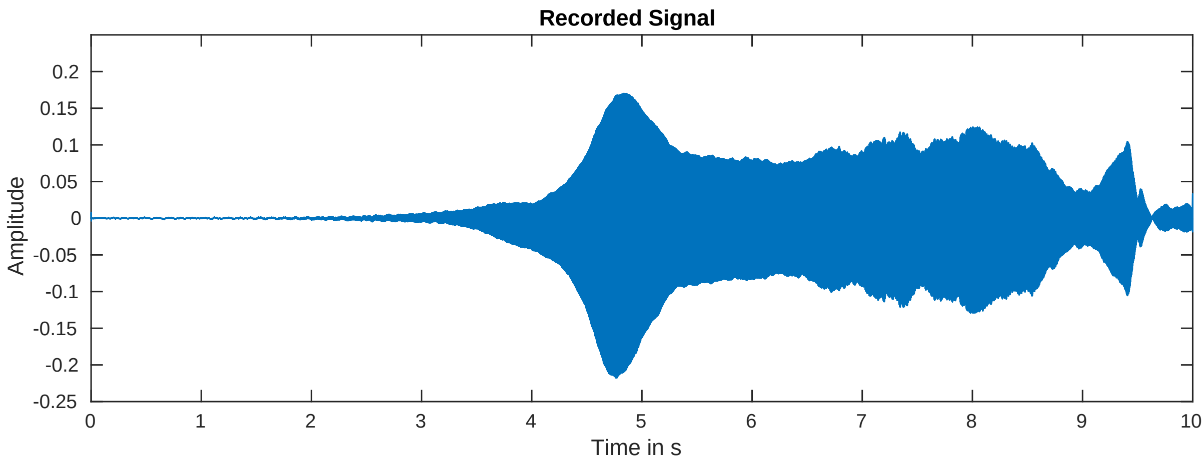

Figure 3.2: *The exponential sinus sweep recorded by the NAO.*

The result was then divided by the scaling factor that is given in [HCZ09] and defined as:

$$C = \frac{\pi L(\frac{\omega_1}{\omega_2} - 1)}{2(\omega_2 - \omega_1 \log(\frac{\omega_1}{\omega_2}))} \tag{3.1}$$

This yielded in the impulse response, which can be seen in Figure 3.3. This impulse response can now be used to describe the system under the specific circumstances that were present while recoding the exponential sinus sweep.

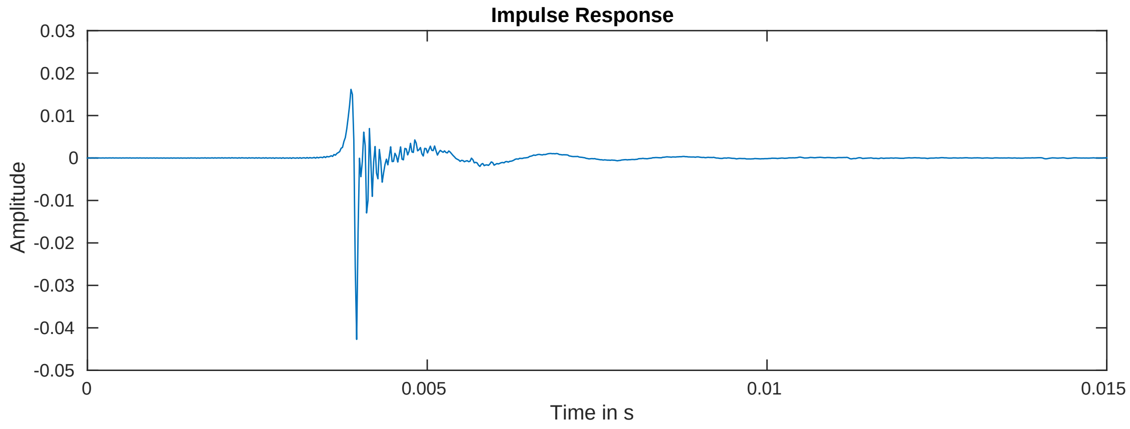

Figure 3.3: *The impulse response of the loudspeakers that is given by the convolution of the inverse signal and the recorded signal.*

3.1.1 Frequency Response

The frequency response allows to determine how a specific system is able to handle different frequencies. It can be calculated by taking the absolute value of the result that is gained by taking the FFT (Fast Fourier Transform) of the impulse response. The plot of the frequency response is shown in figure 3.4. It can be seen that in the lower frequencies the loudspeakers have trouble to create a signal. The energy of the signal is low and rises until it reaches its peak at about 600 Hz. Therefore, the capabilities of the hardware do not allow to send signals over frequencies that are much lower than about 500 Hz. Between 600 Hz and 7000 Hz the signal is constant with small fluctuations which makes this range adequate for transmitting the data. After that, there is a small gap again with a low around 10000 Hz where the energy falls about 10 dB. The energy rises again up to 13000 Hz and after that it drops significantly and becomes weak.

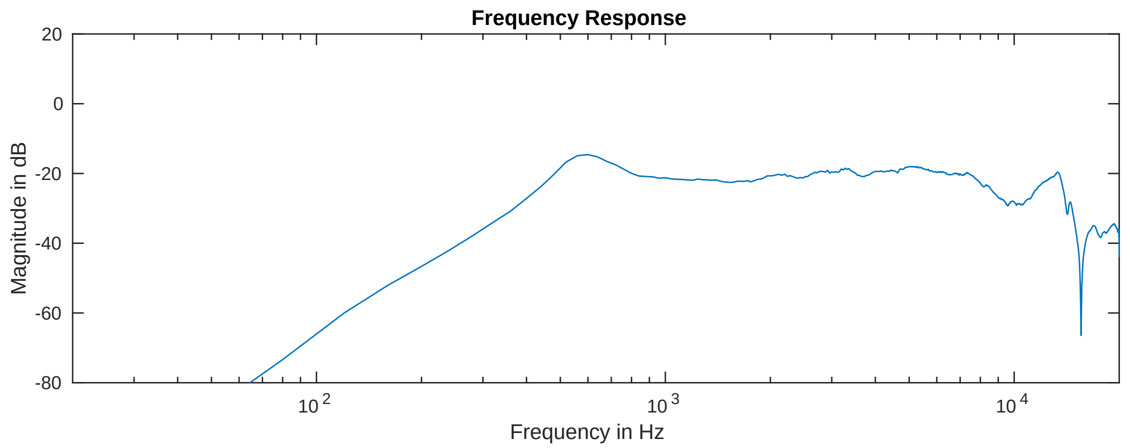

Figure 3.4: *The frequency response of the loudspeakers.*

3.2 Characteristics of the microphones

Comparing the quality of microphones objectively is much harder than doing the same for loudspeakers. In order to measure loudspeakers, one needs a good quality microphone that has linear characteristics. However, there are not many loudspeakers that have a linear energy output. In addition, it is very complicated to get exact measurements and one would need to use a reference microphone in order to compute the reflections due to the environment. In general, one can assume that the quality of the microphones is good, although the exact behaviour was not tested in this thesis because of the complicated procedure.

The specifications for the microphones state that the electrical bandpass of the microphone's way is 300 to 8000 Hz. To test this, white Gaussian noise was played from an external loudspeaker and then recorded with the microphones of the robot. The recorded spectrogram is shown in Figure 3.5. Even though there are frequencies above the given 8000 Hz, it becomes clear that the energy of the signals are weak compared to those below. When scaling the axis, the lower limit that is at about 300 Hz is noticeable as well.

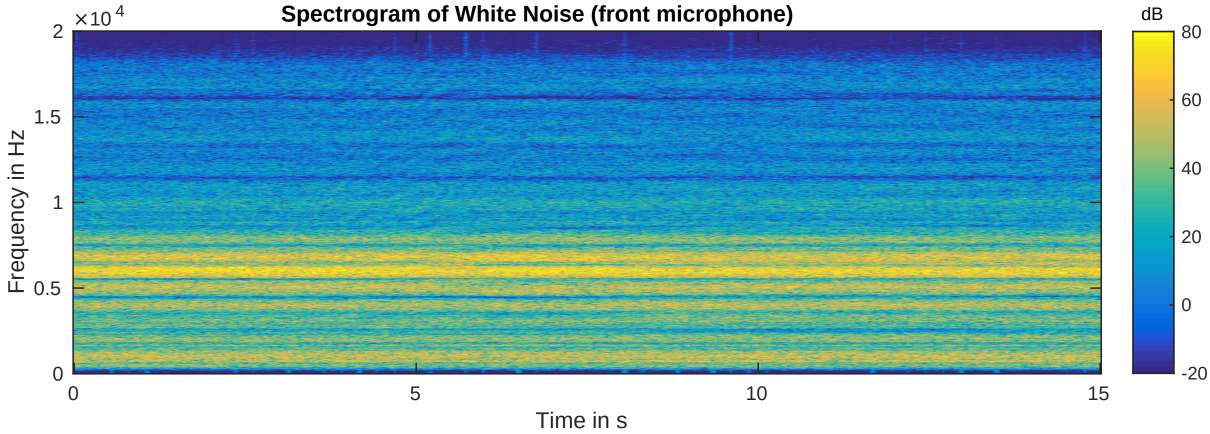

Figure 3.5: *The frequency response of the loudspeakers.*

Overall, the measurements agree with the specifications that were given by the manufacturer. This was to be expected, but it is valuable to be able to affirm the correctness of the data sheet.

3.3 Disturbances

In an environment that is free of background noise, there are two main sources of noise that comes from the robot itself. The first one is the fan which is located in the head. Hot air gets blown out at the back of the head. The openings for the air are right beneath the back microphone.
Besides the fan, the robot itself makes noise when it moves. Especially noticeable is the sound that is produced when the robot walks and the feet hit the ground.[1] The recordings have been

[1]This has been improved in the NAO V5 - Evolution which has feet that dampen the noise.

made with the microphones of a NAO V4 that is limited to a frequency range of 300 Hz to 8000 Hz.

3.3.1 Fan-Noise

In order to evaluate how much the fan-noise will interfere with a signal that is sent over the channel, an experiment was executed. A schematic representation of the set-up for the experiment can be seen in Figure 3.6.

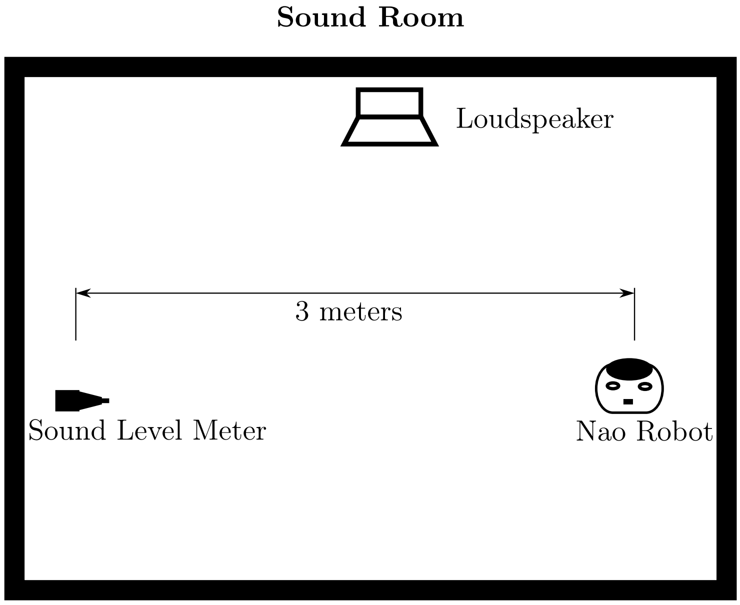

Figure 3.6: *The set-up of the experiment for the fan-noise.*

The idea was to have one robot play a signal while another robot records it and then compare the energy of the signal to the noise. A sinus signal with a frequency of 1000 Hz was used as the reference signal for this experiment. Due to the limitation that only one robot was available during the writing of this thesis, it was not possible to record the signal with one robot and have another one play it. To work around this problem, the signal was played from the robot and measured with a sound level meter at a fixed distance. The distance was chosen as 3 meters due to the size of the sound room where the measurements were executed. Furthermore, the field size in the SPL is 6 x 9 meters, and the used distance of 3 meters, is sufficient to expect to reliably detect a signal. Then the signal was played by an external loudspeaker with settings where the energy that reached the robot was equal to the measured energy by the sound level meter.

All four microphones were used to record. The recorded sound files included the signal which a robot at a distance of 3 meters from the sender will hear and the noise of its own fan. The spectrogram of the sound that was recorded by the microphone located in the back of the head can be seen in Figure 3.7.

Surprisingly, the spectrograms of the four microphones showed that the fan noise was most present in the recording of the front microphone (Figure 3.8) and not of the the back microphone as one would expect. The microphones on the side of the head were influenced the least.

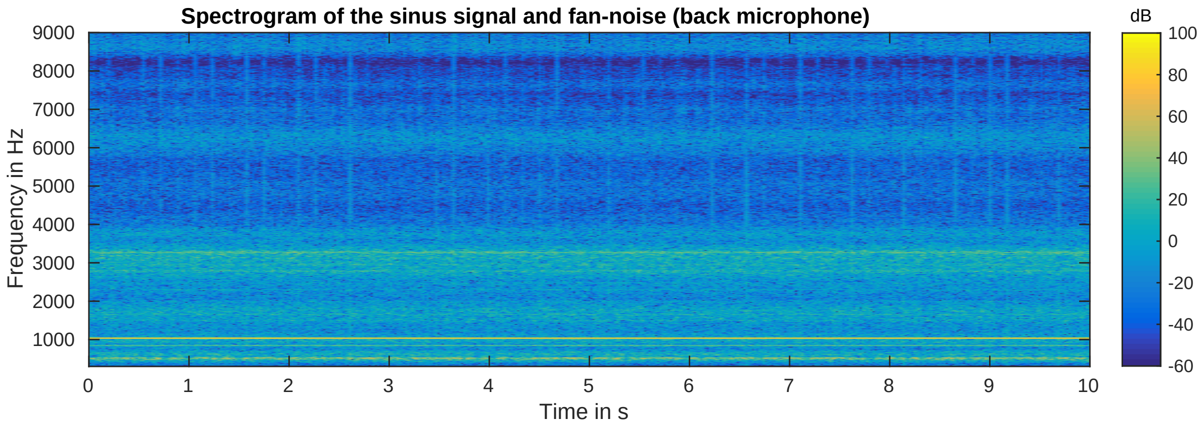

Figure 3.7: *The spectrogram of the sinus signal at 1 kHZ and the fan noise recorded with the back microphone of the NAO robot.*

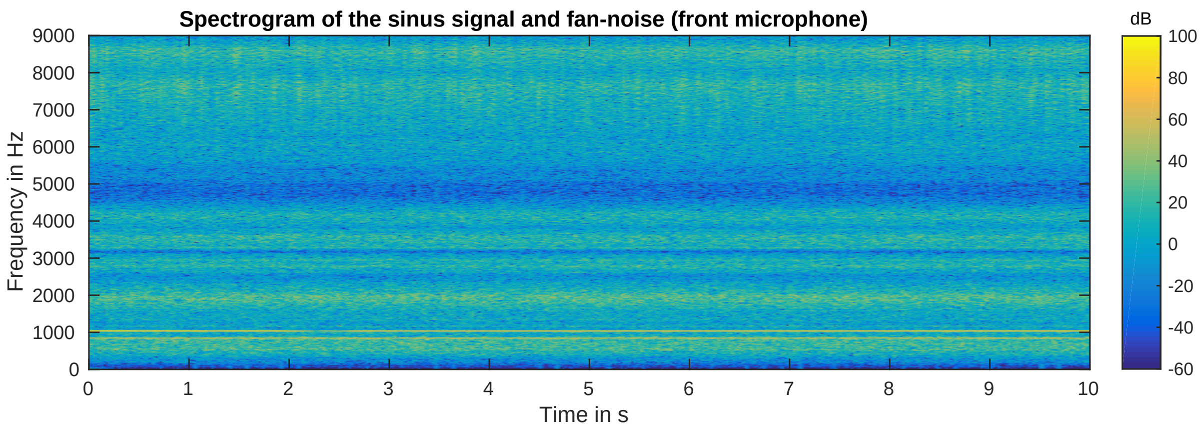

Figure 3.8: *The spectrogram of the sinus signal at 1 kHZ and the fan noise recorded with the front microphone of the NAO robot.*

The sinus signal with a frequency of 1000 Hz can be seen in the spectrogram for the front microphone when scaling the y-axis (Figure 3.9).

The energy of the fan-noise is lower than the energy of the sinus signal that would reach the robot at about 3 meters. When taking a look at the frequency range of the recording (Figure 3.10), the difference between the noise and the signal is about 30 dB or more in most cases. There are a few frequencies that have spikes and only the spike at about 800 Hz comes close to the 1000 Hz signal. When taking a look at the spectrogram, there is a constant energy registered at about 800 Hz during the 10 seconds. It is unclear where this disturbance comes from, but it appeared in other recordings as well. In the spectrogram of the recorded exponential sinus sweep (Figure 3.1), it can be seen again around 800 Hz, but is weaker. It can be assumed that it is noise by the fan since the sound room was designed to be noiseless. It could be produced by the rotating blades of the fan. When choosing the carrier frequency of the communication system, frequencies close to 800 Hz should be avoided and the data of all microphones should be processed at the same time due to their different characteristics.

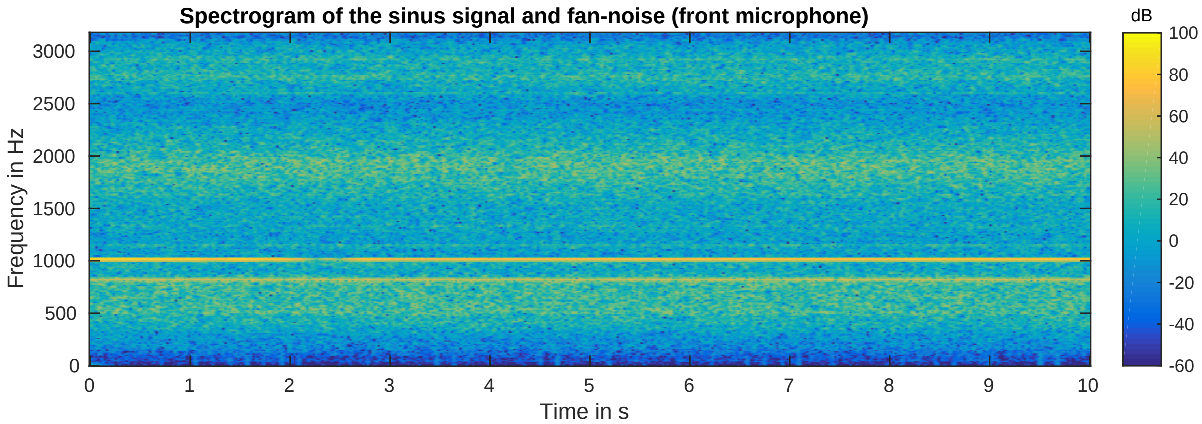

Figure 3.9: *The scaled spectrogram of the sinus signal at 1 kHZ and the fan noise recorded with the front microphone of the NAO robot.*

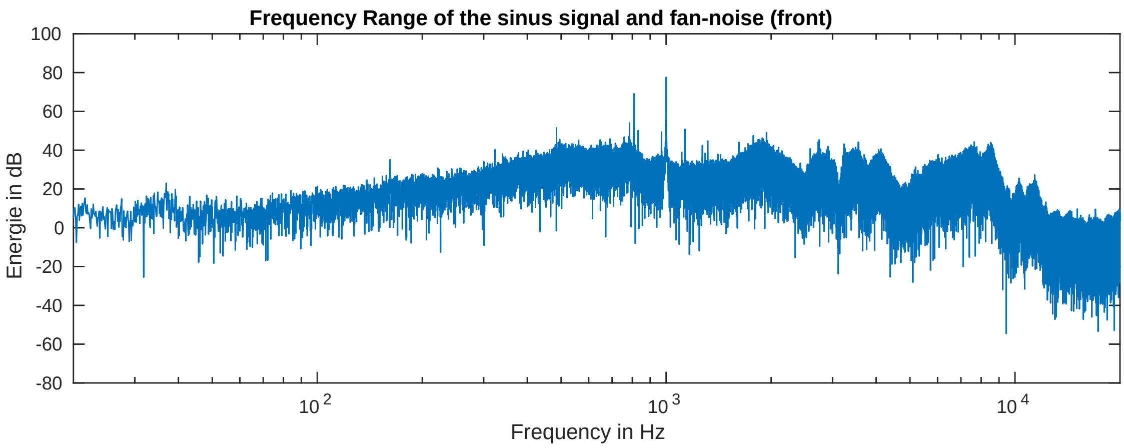

Figure 3.10: *The spectrogram of the sinus signal at 1 kHZ and the fan noise recorded with the front microphone of the NAO robot.*

3.3.2 Walking

When the robots walk they make a lot of mechanical noise. This noise is only present during the walking, but it is very noticeable to the human ear. In order to find out how much the walking interferes with a transmitted signal, the microphones recorded for 5 seconds while the robot moved forward. Since the fan can not be turned off, the recordings of the walking were compared to ones where the robot was standing still. This time, the left and right microphones were used because they are less influenced by the noise due to the fan. In figure 3.11 the spectrograms of the left and right microphones are shown. The upper one is with walking while the lower one is while standing still.

The disturbances due to the fan only influence the frequencies below about 4000 Hz of the microphones that are located on the side of the head. The walking, on the other side, produces some disturbance in the whole bandwidth. The energy due to the walking itself is low but together with the noise due to the fan and possible background noise, it will have a stronger effect on the channel. Fortunately, the walking-noise is only present during intervals and not constantly.

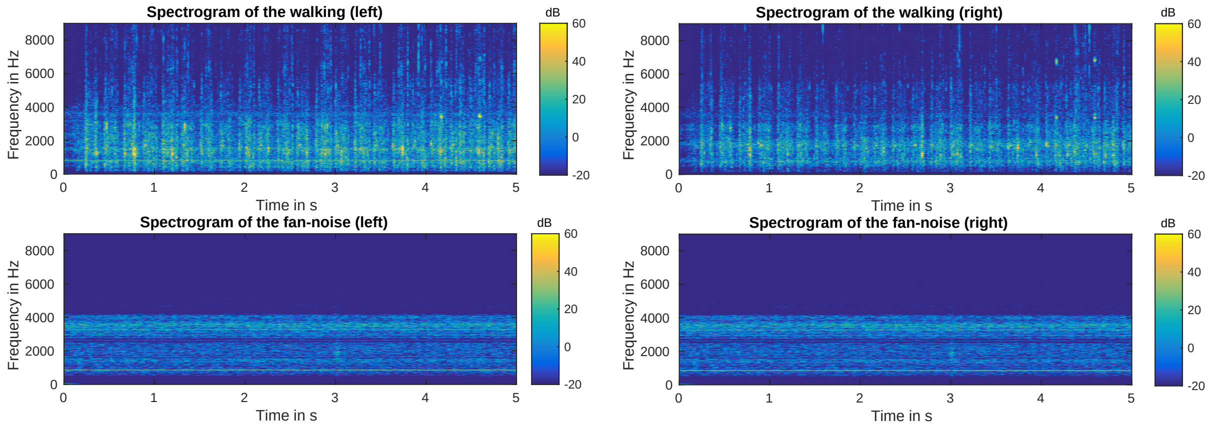

Figure 3.11: *The spectrogram of the robot walking compared to the fan-noise. Bot were recorded by the left and right microphone.*

3.4 Bandwidth

The bandwidth (B) of a frequency spectrum is the difference between the upper and lower frequency of the available range. The bandwidth gives information about how much data can be transferred over a channel. In order to get a reliable data transfer, a range has to be chosen where the frequency spectrum has almost linear properties. When taking a look at the frequency spectrum of the robot in Figure 3.4, it appears that for best results, the minimum frequency $f_{min} = 800$ Hz should be chosen. The maximum frequency f_{max} is limited by the capabilities of the microphone, which has its upper limit at 8000 Hz.[2] This results in a bandwidth of:

$$B = f_{max} - f_{min} \tag{3.2}$$

which yields:

$$B = 7200 \text{ Hz}$$

At this point, it is worth noting that during a soccer game there are spectators and especially some high frequencies are unpleasant to the human ear. In consideration for the people that are watching these should be avoided and the bandwidth might become even smaller.

3.5 Signal to Noise Ratio

The signal-to-noise ratio (SNR) indicates how much a signal is changed by noise. It compares the level of the chosen signal power to the power of the noise. Therefore, the SNR shows how much background noise will corrupt the signal that is being send over the channel. In order to calculate the SNR, one has to use the average power within the same system bandwidth. To calculate the SNR for the chosen bandwidth, the general equation is:

[2]The upper limit of the NAO V5 - Evolution is 12000 Hz and will be compatible with an implementation of a communication system that was developed for the NAO V4.

$$SNR = \frac{P_{Signal}}{P_{Noise}} \tag{3.3}$$

Since the data from audio files that have been recorded is used, the SNR will be expressed using the logarithmic decibel scale. This is defined as:

$$SNR_{dB} = 10 \log_{10} \left(\frac{P_{Signal}}{P_{Noise}} \right) \tag{3.4}$$

which is equivalent to:

$$SNR_{dB} = P_{Signal,dB} - P_{Noise,dB} \tag{3.5}$$

To calculate the power of the signal, it is assumed that the magnitude of the power of all signals in the bandwidth corresponds to the power of the sinus signal with a frequency of 1000 Hz. The sinus signal that was used for the experiment when dealing with the disturbances had a power of about 80 dB. This means that the power in decibel of the signal over the whole bandwidth is equal to:

$$P_{Signal,dB} = 80 \text{ dB}$$

The power of the noise can be calculated by taking the sum of all energies and then divided by the bandwidth:

$$P_{Noise} = \sum_{f=f_{min}}^{f_{max}} P(f) \tag{3.6}$$

Where $P(f)$ is the power at the frequency f.

In an environment where there is no background noise and the noise due to the movement of the robot is negligible small, only the disturbances due to the fan remain. Therefore a recording of the back microphone that only included the fan-noise was used. Using the data from the sound file and inserting it into equation 3.6 yields:

$$P_{Noise} = 2230200 \text{ W}$$

To get the power of the noise in decibel from P_{Noise} can be calculated by using the equation:

$$P_{Noise,db} = 10 \log_{10} P_{Noise} \tag{3.7}$$

$$P_{Noise,dB} = 63.48 \text{ dB}$$

Finally, to calculate the SNR for the chosen bandwidth, we have to take the difference of P_{Signal} and P_{Noise} (equation 3.5) which results in:

$$SNR_{dB} = 16.52 \text{ dB}$$

The conversion from logarithmic decibel scale to power ratio is:

$$SNR = 44.84$$

3.6 Bit-Rate

In order to find out how much data can be send over the chosen bandwidth the Shannon-Hartley theorem can be used. It allows to calculate the maximum possible data rate of a specific bandwidth depending on how much noise is present. The channel capacity (C) according to the theorem is:

$$C = B \log_2 \left(1 + \frac{P_{Signal}}{P_{Noise}} \right) \tag{3.8}$$

It is assumed that the environment is the same as for the calculation of the SNR. This means that the equation 3.8 can be rewritten to:

$$C = B \log_2 (1 + SNR) \tag{3.9}$$

The maximal possible channel capacity for the back microphone under the influence of the fan is:

$$C = 39733 \text{ bit/s}$$

with B = 7200 Hz and SNR = 44.84.

3.7 Doppler Effect

During a soccer game, it is very unlikely for the robots to stand still. Therefore, the transmitter and receivers are all moving, which will result in the Doppler Effect. The Doppler Effect is a change in frequency of a wave for an object that moves. The typical example for this is a moving car that is driving towards and then away from a bystander. It can be noticed that the sounds made by the car are higher when it is approaching.

The change of the frequency due to the Doppler Effect can be calculated for two moving objects, the source (s) and the observer (o). The general equation for the observed frequency is defined in [HRW01] as:

$$f_o = f_s \left(\frac{v_{sound} \pm v_o}{v_{sound} \mp v_s} \right) \tag{3.10}$$

The NAO robots have a maximum speed of about 0.08 $\frac{m}{s}$. The two worst cases are when two robots run towards or away from each other with maximum speed. In the case that they move towards each other, the relative speeds in (3.10) becomes positive in the numerator and negative in the enumerator. In the second case, where they move away from each other, the numerator becomes negative and the enumerator positive. With the speed of sound $v_{sound} = 343\ \frac{m}{s}$ the equation yields for the first case:

$$f_0 \approx 0.9995 f_s$$

and respectively for the second:

$$f_0 \approx 1.0005 S f_s$$

In both cases, the change is approximately 0.05% and therefore the Doppler Effect has a very small impact on the received frequency. In conclusion, even though sound travels much slower than light, the maximum possible speed of the robot is too slow for it to be noticeable.

3.8 Summary

In this chapter, the impulse response and its corresponding frequency response of the loudspeakers were determined for an environment with no external background noise. This allowed to evaluate the capabilities of the loudspeakers and its behaviour for different frequencies.
Furthermore, the disturbances due to the robot were analysed. In order to determine how big the influences of the fan-noises are on a transmitted signal, an experiment was carried out that allowed to compare the magnitudes of a send signal that a robot will receive in three meters to the experienced noise. From the results of the experiment, it became clear that each microphone is affected differently. The microphones on the side are not much influenced. The noise of the fan is definitely recognizable in the back microphone and the front microphone which appears to be affected the strongest. After that, recordings that included noise due to walking were compared with the recoding of the fan noise. The comparison showed that the walking noise itself is not very strong but will add to the sum of all noises which will disturb the signals that are sent.
In order to give an overview of the possibilities that can be achieved when implementing a communication system, some characteristics were calculated. These include the bandwidth of the system, the signal-to-noise ratio in an ideal environment, the maximal expected bit-rate and the impact of the Doppler Effect.
The results reflect the capabilities of a perfect system that is not comparable with a realistic environment. Usually, there is much more noise and that will increase the signal-to-noise ratio. Nevertheless, the available bandwidth is wide enough to allow a good bit-rate, even if there is more noise present.

Chapter 4

Acoustic Transmission System

In this chapter, parts of an acoustic transmission system for communication are proposed that comply with the requirements that are given by the hardware, the environment and desired features. First, a list of requirements and what information needs to be transferred will be defined. The information has to be encoded to a binary sequence that can then be used to create a digital signal. Furthermore, error control will be discussed to be able to detect when the message is lost or distorted.
After that, some protocols will be presented that regulate who has access to the channel. Usually, there are more than two participants during a soccer game and they all want to broadcast their information to the others. The advantages and disadvantages of the protocols will be evaluated and how suited they are for the SPL.

Different possible concepts and methods will be introduced that could be part of the transmission system. There will, however, not be a final concept that proposes the best system due to the different possible types of communication and it would go beyond the scope of this thesis to compare every method. Depending on the aim of a transmission system, different methods will be more appropriate than others. This thesis will give an idea on how to implement a well working transmission but not actually implement one.

4.1 Requirements

The requirements for an acoustic transmission system were chosen from experience and the boundaries of the environment.

- **Distance:**
  The field size is 6 x 9 meters and therefore the maximum distance between two robots is less than 11 meters ($\sqrt{6^2 + 9^2} \approx 10.8$ meter). It can not be expected to have a reliable data transfer with acoustic signals over that distance. The desired distance for an errorless communication should be feasible for about 3 meters. That distance is close enough for a strong signal on the receivers side and far enough to react appropriately.
  The measurements have shown that this is given in an environment with minimal background noise. An implemented system should be tested in a real environment to test if this is still the case.

- **Amount of data**
  The goal of a working acoustic communication is to be able to use it as a fall-back method when wireless network communication is not available. Therefore, it is desired to limit the amount of data that will be sent to the absolute minimum required data to prevent chaos on the field. This includes the own player number, the position of the robot, the position of the ball relative to the robot and when the ball was seen there. Furthermore, it can be helpful in many situations to send if the robot has fallen or is still upright. Depending on the transmission system, one should include the length of the whole message. This information can be transmitted under 160 Bit (20 Byte).

- **Data-rate**
  Due to the fact that soccer is a fast game and changes happen quickly and in real-time, a fast data-rate is desired. Usually, there are 5 robots on the field and if the transmission method would be chosen, so that they can not send messages at the same time, a data rate of 800 Bit/s would be needed for every robot to receive all messages after 1 second. The bit-rate that was calculated in 3.6 is well above the requirement and even if more noise is present, the bandwidth still allows for a sufficient rate.

- **Error-control**
  When sending data over an acoustic channel, noise is to be excepted. It is required to know when data that was received has errors due to disturbances. If possible, the encoding should allow to correct errors when decoding the message.

- **Robustness**
  During a soccer game, there is a crowd that anticipates and watches the game. This results in a lot of background noises that reach from yelling, to cell phones that are ringing. This noise can not be prevented and should be filtered out. The transmission system of the robots should be robust and reliable. A working system needs to be examined under the circumstances of a game with spectators.

4.2 Transmitted Information

Before it can be decided how data is transmitted, the information that will be sent has to be defined and how a message looks like. Two different types of messages will be shown and in what situations they are beneficial.

4.2.1 Messages

A message that is transmitted should consist of a meaning such as the position of a robot and the actual data. There are different ways to identify the data that is sent and where it belongs. Two types that are easy to implement and are common will be introduced.

One could first send a key that allows the receiver to assign the data to its purpose. This means, for example, that if a robot wants to send a message with its own position, it first transmits a code that is unique and then the data for the coordinates. The receiver would have to know what the code means and can then process the message. An example for this is shown in figure 4.1. This method allows to send data at an arbitrary time, for example, when a specific event occurs. The receivers would only need to know when the message begins, how long the it is and be able too look up the meaning of its key. However, this method produces a lot of overhead due to the keys that are included with each message.

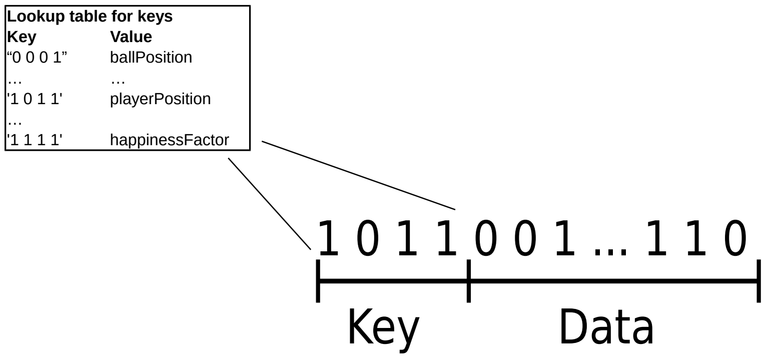

Figure 4.1: *An example for a transmitted message using keys*

Another common method is to send one message that holds all the information (Figure 4.2). Due to the knowledge of which bytes belong to which value, the receiver can separate the data and associate it to its meaning. This message requires a predefined standard and the knowledge when the message started. The message would be smaller compared to the messages in the first method for the same amount of transmitted information. However, one would have to wait until the whole message is transferred before sending the next one. This could lead to the same data being transmitted multiple times and a longer time delay.

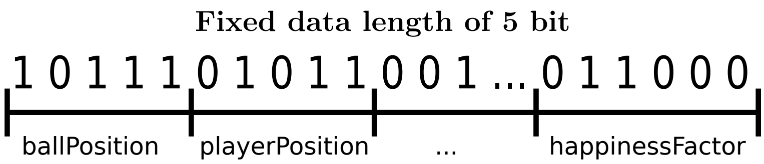

Figure 4.2: *An second for a transmitted message using the known length*

For the wireless communication, the rules of the SPL [Com14] define a standard for messages. The so called SPL standard message allows a maximum size of 800 bytes for arbitrary data and a fixed amount of predefined data that holds the following information:

Type	**Name**	**Possible Values**
char[4]	header	"SPL "
uint8_t	version	The current version (5 as of 2015-01-23)
uint8_t	playerNum	1-5
uint8_t	teamColor	1 for red and 0 for blue
uint8_t	fallen	1 for fallen and 0 else
float[3]	pose	Absolute x, y and theta coordinates on the field of the robot
float[2]	walkingTo	Absolute x and y coordinates of the robots target
float[2]	shootingTo	Absolute x and y coordinates of the target for the next shot
int32_t	ballAge	The time when the ball was last seen in milliseconds
float[2]	ball	Relative x and y coordinates of the ball to the robot
float[2]	ballVel	Relative x and y values for the velocity of the ball
unit16_t	intention	Describes what the robot plans to do: 0 - nothing particular (default) 1 - wants to be keeper 2 - wants to play defense 3 - wants to play the ball 4 - robot is lost Comment: The second byte is a padding byte
uint16_t	numOfDataBytes	How many bytes are actually used out of the a allowed 800 bytes for data
uint8_t[800]	data	Arbitrary data

While a wireless network is present, all robots have the data available that is given in the table above. In case of an outage of the wireless network when the communication is switched over to acoustic communication the transmitted messages should be reduced to the most important data.

Therefore, if one would chose to send messages depending on specific events, the quantity of how often an event occurs should be kept to a minimum to prevent collisions. This means that broadcasting the own position of the robot every time that it moves, should be avoided. Instead, a robot could, for example, claim ownership of the ball when it is within a specific range.
If instead the position of the robot is of interest, as well as, other regular updating information like the ball position, a standard message should be defined. In order to prevent collisions, the access has to be regulated so that the robots can not broadcast their information at the same time.

4.2.2 Encoding and Decoding

The messages that will be transmitted hold arbitrary data like numbers and symbols. Encoding changes the message to a code that is gained by following a defined algorithm. The reasons why encoding is used are that the generated code is designed to allow an easier and faster transmission. Especially when transmitting numbers, many encoding methods also compress the message that has to be send. It is simply not possible to transfer a symbol like "A" without mapping it to some kind of code. Furthermore, it is not efficient to send a high number like

11 by "beeping" 11 times. Decoding is then used to extract the meaning of the code. Furthermore, only the sender and receiver know what code is used to encrypt and decipher the message, which adds a security functionality to the message.

For encoding numbers, the easiest method is binary code where the number is represented as a sequence of zeros and ones. There are encoding methods that also allow to encode alphabetical symbols like ASCII code. ASCII code encodes every letter by itself which increases the symbol length but one could also use a method that maps whole words or sentences to a code if the amount of needed words is small. Another well known method is Morse Code which was designed to have a shorter length for characters that appear frequently in the English language. There are many more codes, like Huffman code, that are known and have been evaluated in many papers.

In the end, it is up to the developer that implements the transmission system, to decide which code should be used. In the case that keys will be sent within each message, one should decide on an efficient algorithm and map the meaning to a code and use a different method for the encoding of the data. Whereas, if only data in form of numbers is transmitted, binary code should be sufficient.

4.2.3 Error Protection

Error protection aims to detect errors in a received message and if possible to fix them without the need to retransmit the message. There are several reasons how errors could arise. It is possible that the signal strength was not sufficient enough or that disturbances of the channel altered the signal. Therefore a good error protection has two functionalities:

- Error detection
- Error correction

If only one bit was wrongly received, it is called a *bit error* and a *burst error* when multiple bits that are next to each other were wrong [Fre02]. Every kind of method for a better error protection will need more data to be transferred. However, the additional data is small and well worth the benefits.

An easy way of detecting an error is the use of a parity bit. At the end of every code word, another bit is added. The bit will be set to one or zero, depending on if the total sum is even or odd. Two examples are shown in Figure 4.3. With a parity bit, one can detect that an error occurred, but not the location of it. In addition, it only works if there is an uneven amount of errors in the code word. There are a few other methods that allow to detect if an error occurred like repetition codes and checksums.

A method that allows to detect and correct an error is the block coding. The code is divided into blocks of the same length as seen in Figure 4.5 and then the parity bits for each column and row are calculated. If an error occurs, the parity bit for that column and row would both be different than expected and the exact position where the error is can be determined. In order to fix it, the bit at that position has to be inverted. Block coding methods first transmit the information bits followed by the parity bits. A few examples for block coding are Hamming

Parity Bit

d1	d2	d3	d4	y
1	0	1	1	1
0	1	1	0	0

Figure 4.3: *Two codes of length four with their parity bit (y).*

codes, Reed-Salmon codes and Expander codes.
Besides block coding, another family of methods are the convolution codes. Unlike block coding, the convolution codes do not transmit the information followed by the parity bits but a new code that is gained from the information and mathematical operations. These codes are often used in mobile networks and satellite communications. The new code is generated with shift registers and every bit that holds information results in multiple new code segments. When decoding the message, an error can be detected and corrected if it appears in one segment due to the information that the other segments hold.

Block Code

d1	d2	d3	d4	y
1	0	1	1	1
0	1	1	0	0
0	0	1	0	1
1	1	1	0	1
0	0	0	1	

Figure 4.4: *Four codes with a length of four as block code.*

4.3 Access Control

In a communication system, it is important to regulate the access for each participant of the channel. A soccer team with multiple players that need to communicate with each other corresponds to a network with broadcast link (for more information see [KR13]). In a broadcast link, there are multiple sending and receiving member (also known as *nodes*) connected to the same channel. There are multiple ways to control the access when it comes to transmitting in such a network.

There are already many protocols that regulate the access in a network. In general, the available protocols can be divided into three groups [KR13, p. 447]. These are channel partitioning protocols, random access protocol and taking-turn protocol.

4.3.1 Channel Partition Protocols

There are three techniques that are common to partition a channel. One is called time-division multiplexing (TDM), where the time is divided into time frames. The second one is frequency-division multiplexing (FDM) and divides the channel into different frequency ranges. The last technique is code division multiple access (CDMA). In CDMA, each sender has a code that is used to encode the data. Channel partition protocols are designed for systems where the channel will be used frequently for example in the case that messages will be send periodically.

When using TDM, every participant has to be synchronised so that they don't send at the same time. TDM is convenient when sending the same data regularly, because when used properly, there is no collision and every member gets to send the same amount of data.

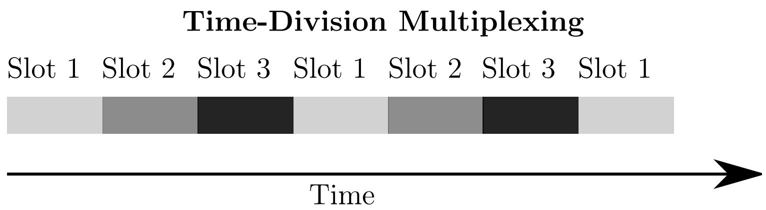

Figure 4.5: *TDM divides the time into slots and each participant is assigned one.*

FDM allows participants to send whenever they want. There is also no chance of collision because the signals will never be send over the same frequencies.

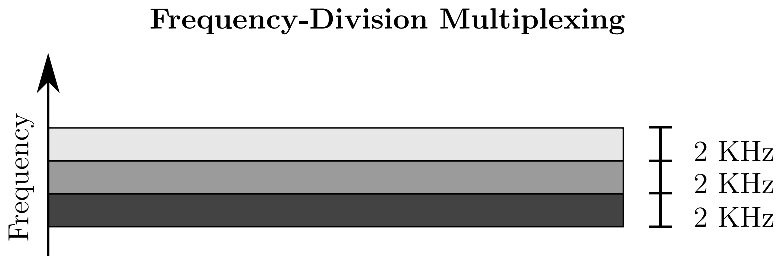

Figure 4.6: *FDM divides the bandwidth into frequency ranges for each participant.*

Both, TDM and FDM have similar advantages and drawbacks but have different implementations. CDMA is unlike the other two techniques. When the codes that are used for encoding are chosen carefully, it allows multiple sender to transmit at the same time. However, it is quite complex to implement and needs a lot of processing power. Most mobile phones use CDMA and have chips that implement the calculations that are needed in hardware.

If the frequency bandwidth would be divided into five parts, each robot would still have enough for a sufficient bit-rate since there are only five robots in one team It is easy to implement and the robots could send data whenever they want. However, there are some difficulties when dividing the bandwidth that make the process more complicated. A robot that sends data would have to listen for transmitted data at the same time. This increases the amount of processing power that is needed. In the worst case, a sending robot would have to process the signals of four other robots simultaneously. Furthermore, the noise is differently distributed over the bandwidth and this could lead to an unfair partition of the bandwidth. It can be expected that if more teams want to use acoustic communications at the same time, the rules for the SPL will be adapted and each team could be assigned a specific bandwidth or they have to agree on using different frequencies. If the developed system would already assign each robot its own frequency range, it could lead to complications. It would be better to assign each robot a time slot for transmitting. The only difficulty would be that they have to be synchronised but this could be done over an existing wireless network or at the beginning of the game when they are all in the same half and in reach for a strong acoustic signal. The drawbacks, however, are that the messages would be sent with a small delay and when one robot would be taken out of the game its time slot would be unused. Furthermore, when a robot would be out-of-sync, collision could occur. The drawback are acceptable and TDM will be better suited due to its easier implementation than CDMA. To sum it up, for messages that are send periodically, like the SPL standard message, the time-division multiplexer technique is preferable. It will be easy to adapt in case of future rule changes. Its implementation is straightforward and the needed processing power is reasonable.

4.3.2 Random Access Protocols

Random access protocols use the full channel to transmit data at the moment that data needs to be sent. In the case that collision occurs, the nodes that transmitted at that time wait for a random amount of time and then retransmit their data. Due to the unlikeness that nodes randomly wait for the same delay, the chance for another collision is small the next time that a node transmits. Random access protocols are commonly used when data has to be transferred irregularly. There are many protocols that follow this principle but the most used ones are the ALOHA protocols and the carrier sense multiple access (CSMA) protocols.

There are two different ALOHA protocols which are pure and slotted ALOHA. In the pure ALOHA protocol the sender just transmits data when it has some. It then then monitors the channel and when collision was detected, the data is retransmitted after a random amount of time. In slotted ALOHA, it works the same but the time is divided in slots and instead of transmitting right away, the sender picks the next available slot.

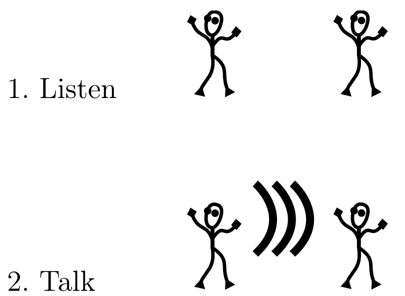

Figure 4.7: *Stickmen using CSMA*

Carrier sense multiple access protocols, on the other hand, first listen to the channel and only when it is free, they send their data. This is well known as the listen-before-talking principle (Figure 4.7) that is often used in a human conversation. In case that the channels is sensed as busy, they wait a random delay before trying to transmit again. Due to this, there are less collisions possible then there are in ALOHA protocols. Nevertheless, there is still the chance of collision due to delays or in wireless networks due to nodes that are out of reach for a sending note but in reach of the receiving node.[1] Again, there are two popular protocols that are based on CSMA. These are CSMA with collision detection (CSMA/CD) and CSMA with collision avoidance (CSMA/CA).

[1] This is known as the *hidden node problem.*

CSMA/CD is widely used in wired networks and nodes that transmit data listen at the same time to the channel in order to detect collisions. When a collision occurred, the node waits a random delay that increases every time that the data collides. If the channel is free again, the data is resent.

For wireless local area networks (WLAN), the standard that is used is CSMA with collision avoidance (CSMA/CA). Unlike for wired networks, in WLAN it is not possible to listen to the channel at the same time that data is being transferred. Due to that, the protocol aims to avoid collisions by first checking if the channel is free and then waiting for a time window (called DIFS) before checking again. If the channel is still free, then it is allowed to transmit. In order to know if a collision occurred, the listener has to send back an acknowledgement when it has received the data. If the sender does not get an acknowledgement back, it will retransmit.

Random access protocols are used when data needs to be send as soon as it is available. This includes messages that are transmitted when specific events happen. There are only five robots on the field so the amount of members in a network is limited. Due to this, the probability of collisions is small but still possible depending on how what events will trigger a message and it is recommended to use a CSMA protocol. The robots are able to process sound at the same time that they transmit a signal. Therefore, the CSMA/CD protocol can and should be used due to the less delay compared to CSMA/CA.

4.3.3 Taking-Turns Protocols

Like random access protocols there are many variations of taking-turns protocols. To get an idea of how taking-turn protocols work, two of them will be described. The first one is known as the polling protocol. The protocol regulates the traffic with the help of a master node. The master node *polls* other participants when they are allowed to send data. The other nodes receives a message from the master node when it is their turn to transmit and how much data they can sent. When the master node senses that the sender is done, it can allow the next one to transmit. There are no collisions possible like it is in random access protocols. Furthermore, there are no empty slots possible that are unused in the two groups of protocols above. There is a small delay due to the polling but it is a minor drawback. However, the whole system depends on the master node and if it fails the system becomes unusable.

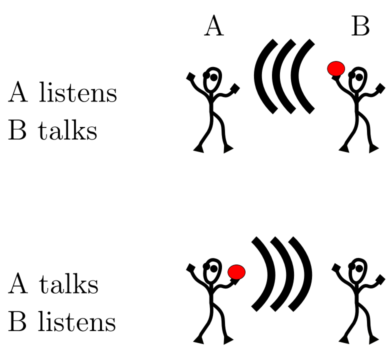

Figure 4.8: *Stickmen using a token*

In order to eliminate the dependency on one node, the token-passing protocol was designed. It uses a token that allows the owner to transmit a fixed amount of data. When the owner of that token is finished, the token gets passed onto the next node in the order. If there is no data to send, the token is handed over immediately. This is similar to a group of people using an object that allows the owner to talk as seen in Figure 4.8. The protocol is efficient and less dependent on one specific node. If the token is not passed onto the next node due to some failure, the whole system is stuck. It is possible to retrieve the node and start over, but it costs time and the other nodes have to know which node failed and is not available any more.

While taking-turns, protocols are very efficient and collision-free, the dangers are high. Especially in a soccer game with robots, one has to expect that robots will be taken out of the game due to problems. If this happens without that, the other players are aware of it, the whole system could break down. In addition, when using acoustic signals to hand over the token, it could get lost due to the limited coverage.

4.3.4 Conclusion

Depending on what message will be sent, different protocols are more suitable than others. If the messages are event based and irregular, one of the random access protocols would be preferable. Due to being able to listen while sending, a protocol that follows the CSMA/CD principle is adequate.
However, if messages would be send periodically with a fixed length like the SPL standard messages, it would be better to use a channel partitioning protocol. Collision would be prevented and each robot would get the same bit-rate to transfer data. Due to the points that were discussed in 4.3.1, one should chose a time over a frequency partitioning.
Taking-turn protocols can be disregarded since the risk of failure is too high for a soccer game and implementing a reliable and error resistant system with a token is difficult.

4.4 Transmitted Signal

Before one can transmit data, the binary code has to be represented as a digital signal. Then the digital signal has to be modified by a modulation with a carrier signal. In this chapter, different modulation methods will be discussed that allow us to transfer our digital signal of binary data using the available bandwidth.

4.4.1 Carrier Signal

For an analogue modulation, the most common type of carrier signal is a sine-wave. This wave can easily be modified by adjusting its amplitude, frequency and phase to hold the information of the digital signal. In case of an digital modulation, a pulse sequence is used as a carrier. This pulse is represented as a square-wave signal.

4.4.2 Modulation Methods

The purpose of a modulation method is to create a signal that will optimally use the available bandwidth. The original information will be modulated with the carrier signal and then transmitted. On the receiver side, a demodulator will convert the signal back to its original state. There are many different modulation methods for digital signals. The three most basic ones are amplitude-shift keying (ASK), frequency-shift keying (FSK) and phase-shift keying (PSK). Most other methods are a variance of one of these three or a combination of them.

When modulating a digital signal, the changes for the amplitude, frequency and phase are discrete and not continuous. This is refereed to as shift keying. Figure 4.9 shows how the signal is changed by the three basic methods.

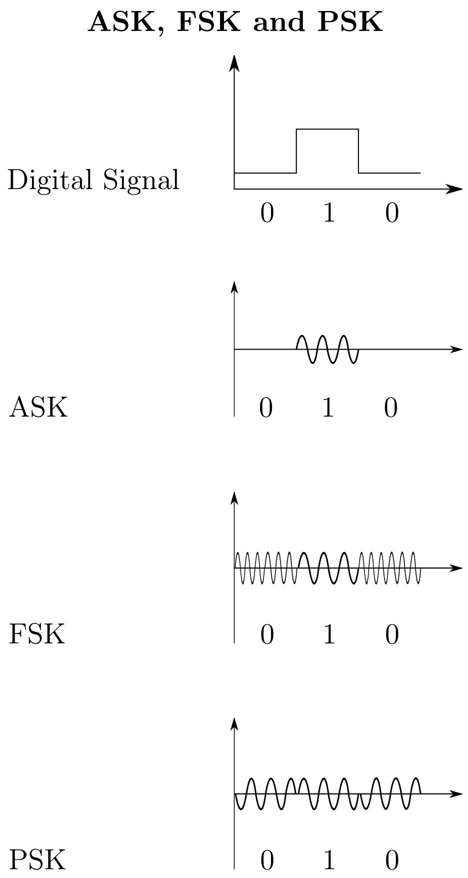

Figure 4.9: *Different digital modulation methods*

A high amplitude of a signal can easily be detected by looking at the energy of a signal for a specific frequency.
If an ASK modulation would be used to transmit data, one would have to choose the carrier frequency carefully to avoid as much disturbance due to noise. Furthermore, it is difficult to have more than one Bit during one time interval.
With FSK, it is easier because more than one Bit could be assigned to a specific frequency. This would increase the throughput and allows a better use of the available bandwidth. In order to demodulate an FSK signal, more calculations are needed than for ASK but the gain would be a higher data rate.
For PSK, the change of the phase for one bit would be 180°. More Bits can be transferred when changing the phase to 90° or even less. This is then called a 4-PSK (other possible variations are 8-PSK, 16-PSK and 32-PSK). Higher than 32-PSK is not possible due to the phases becoming too low to differentiate between them.
Another method worth mentioning is an combination of ASK and PSK to be able to transmit more Bits at once. This method is called QAM and there are again, different variances of it.

There are many more, and it would go beyond the constraints of this thesis to introduce all of them. An implementation of a transmission system should allow to change the modulation and demodulation method without that other parts have to be rebuild. At this point, the recommendation would be to start with a simple method like FSK and if desired, implement more complex methods at a later point. Different implementations would have to be tested and evaluated to see which method is the best combination of data throughput, robustness and complexity of implementation.

Chapter 5

Summary and Outlook

The main focus of this thesis was the investigation of the capabilities of the NAO robot system for an acoustic transmission system. The first chapter acquired knowledge about the environment that the transmission system will be exposed to. Furthermore, the significance of an acoustic communication was represented for a team of robots that play soccer.
In the second chapter, the details about the hardware were listed to get a better understanding of the robot that is the subject of this thesis. Due to the spare data that was available, own measurements were taken.
Valuable information was gained by measuring the capabilities of the given hardware of the robot. A closer look was taken, in chapter three, at the microphones and the loudspeakers of the robot. In addition, the effect of disturbances were examined and their influence on a transmitted signal. Some more data was calculated that helped to evaluate the limits of a prospective communication system. It became clear, that the robots are very well able to transfer data over acoustic means.
In the fourth chapter, several aspects of a transmission system were introduced. Different methods were compared and their signification for an acoustic communication by the NAO robots was discussed. Due to the unique conditions for robots that participate in a soccer game, different requirements and challenges have to be considered compared to usual communication networks.

The obtained information and ideas from this thesis will assist to develop and implement a well working acoustic transmission system. The next step is now to decide on the methods that will be used and implement those. After that, the system has to be tested under real conditions in a soccer game of the SPL. The data that will be gathered can then be used to evaluate the overall performance of the system. Different methods can be implemented and compared to each other more advanced techniques can be investigated.

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