

THE BUG ZAPPER



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LIST OF ABBREVIATIONS

IC	Integrated Circuit
TDOA	Time Difference Of Arrival
DDOA	Distance Difference Of Arrival
ITD	Inter-aural Time Difference
IPD	Inter-aural Phase Difference
ILD	Inter-aural Level Difference
IID	Inter-aural Intensity Difference
GCC-PHAT	Generalized Cross Correlation PHase Transform
AED	Adaptive Eigenvalue Decomposition
SNR	Signal to Noise Ratio
DAQ	Data AcQuisition
A/D	Analog to Digital



SUMMARY

In this project the goal was to make a bug zapper which can localize a sound source by means of phase difference detection in the measured sound waves at different positions in space and shoot a laser at this source. The sound source localization is done by using five microphones of which one is a reference. A TDOA is detected by cross correlation and peak detection of the measured signals of one of the four microphones with the reference. This TDOA is used to calculate the DDOA and then the sound source localization is calculated by a linear closed loop algorithm. We did not manage to implement a laser into the system but a laser can be implemented with a galvano mirror. The sound source localization works with measuring repeatability of 0.75 cm and localizes the sound source within 2.5 cm error for 11 out of 24 measured locations. This is not accurate enough to realize a bug zapper with minimum measuring repeatability and sound source localization error of 0.3 cm. The difference in measured and desired sound source locations is due to errors in calculating the TDOA's, inaccurate measurements of the positions of the microphones, false assumptions made on the sound source and inaccurate measurements of the temperature.



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INTRODUCTION

Since the dawn of time men has encountered and overcome lots of problems. Today, we rule the world and are largely in control of nature but there still are problems that keep us awake at night. This particular one has cost many of us hours of sleep and almost no one has faced this creature without drawing blood. I am, of course, talking about the mosquito. We have been bothered long enough by these little insects and it is time to stop this menace. We have humbly accepted the task – cast upon us by our supervisor Herman Offerhaus - to free mankind of these awful creatures, with lasers.

The goal of this project is to localize a mosquito using the sound it makes and produce a device which will aim a laser at the mosquito to eliminate it so that it can't harm anyone any longer. The localization of sound is a technique called sound source localization. The human ear is a very good system to localize sound sources. One ear is able to determine in which direction a sound source is located and also give a general idea how far away it is located. One of the characteristics of sound the ear uses is the phase information. The localization of the mosquito will be done by using phase differences of the sound of a mosquito obtained by a microphone array and from that calculate the position of the mosquito.

The methods we applied will be discussed in the theoretical aspects. This will cover the sound source we used and what assumptions were made, what sound source localization is, how it can be done, how we did it and limitations we have to consider when processing the measured signal.

How we implemented these methods in a practical setup will be discussed in the experimental aspects. This will cover the microphones used in the setup, their limitations, the DAQ measuring card and limitations we encountered while digitally processing the signal.

The measurements we performed will be discussed in the results and this will lead to a discussion about what could be improved, a conclusion on our measurements and recommendations on continuing this project.



THEORATICAL ASPECTS

To be able to make a bug zapper which is capable of shooting down mosquitoes we first need to define boundaries by making assumptions and considering the limits of each step in the system. These steps are:

1. The sound source emits a sound wave, which will be measured by our system.
2. The sound source emitting the sound waves needs to be localized. This is split into two parts:
 - Determining phase difference
 - Localization method
3. The sound source needs to be shot down by a laser.

We will also explain two considerations regarding the processing of the measured signals.

SOUND SOURCE

A sound source is a vibrating body which makes sound and is characterized by its frequency spectrum, amplitude spectrum and phase. Next to these characteristics a sound from a sound source also has noise. Noise appears due to irregularities in the vibration or external influences like reverberation. The sound source is assumed to be a stationary point source radiating sound waves spherically outward in all directions and always loud enough to be able to perform a measurement within the system's dimensions.

The bug zapper focuses on insects as sound sources. Insects make sound during flight by flapping their wings. The characteristics of the sound are determined by the flapping speed and the motion of the flapping. The faster and harder an insect flaps its wings, the higher the base frequency and amplitude of the sound will be. Different insects are unique enough in this aspect to be distinguished from one another. Not only are they distinguishable by the base frequency but also by the distribution of higher harmonics. For this study we took the mosquito as the insect of interest. A mosquito is assumed to be a sound source of 1 cm by 0.3 cm by 0.3 cm. The frequency of the sound produced by a mosquito is dependent on the species of mosquito. The sound has a base frequency range between 300-500 Hz [1]. We used a recording of a mosquito recorded by the Acoustical Association of America. The recording is on a female mosquito from the species *Anopheles stephensi*, a malaria carrying species from India [2].



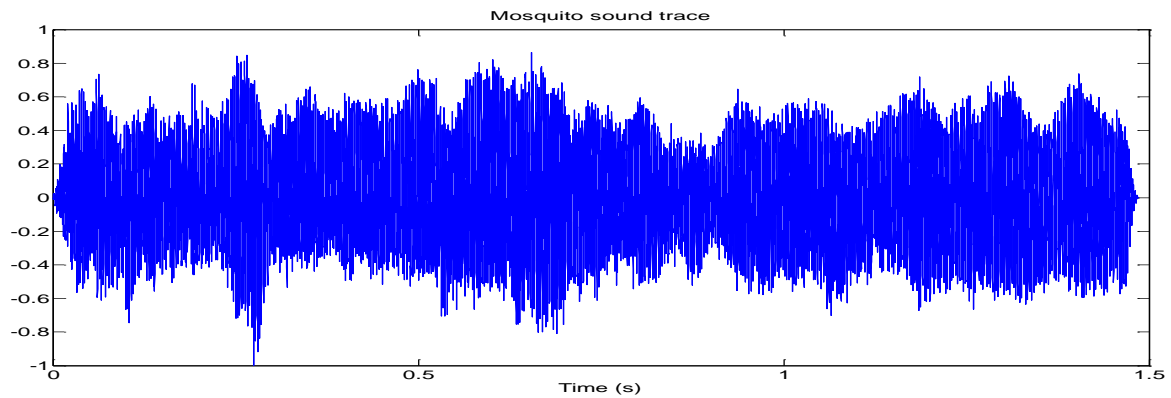


Figure 1: Mosquito sound trace made from the recording by the Acoustical Association of America [3].

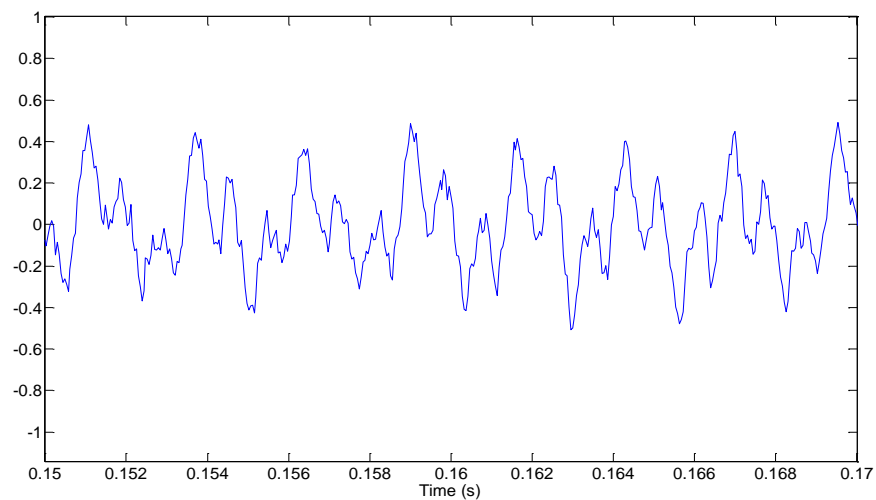


Figure 2: Mosquito sound trace zoom in.

The above figure shows that the mosquito sound has a repeating pattern with three peaks. This means there is at least one higher harmonics in the sound. A Fourier analysis reveals the base frequency and its higher harmonics.

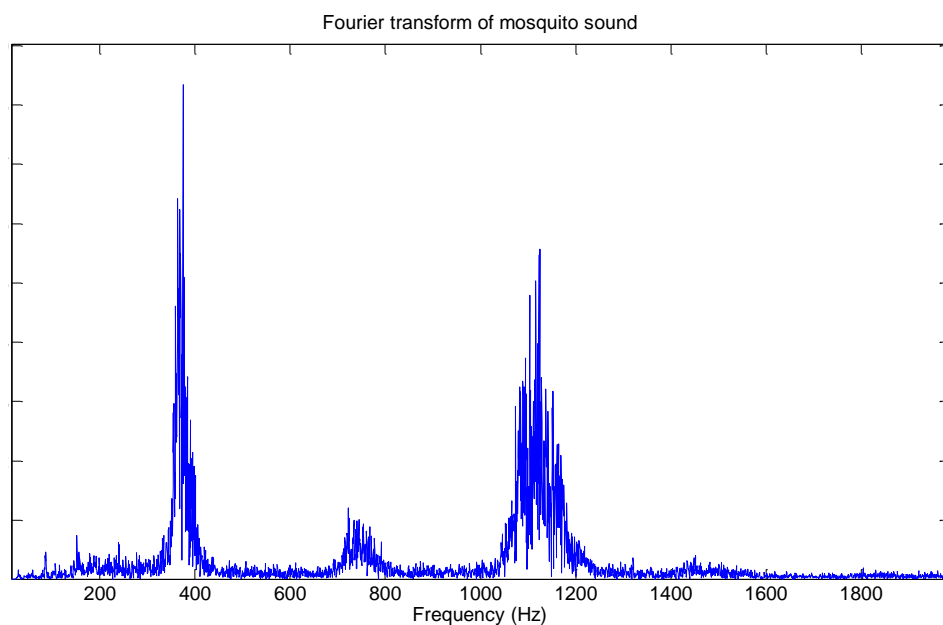
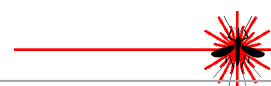


Figure 3: Power Spectral Density plot of the mosquito sound trace.



The base frequency turns out to be around 375 Hz. The first harmonic is around 750 Hz and the second harmonic is around 1125 Hz. Especially the second harmonic is interesting since it has a higher frequency and allows for faster and more accurate computation. The peaks are 100 to 200 Hz wide, indicating that there is noise in the signal and the signal is not perfectly repeated.

SOUND SOURCE LOCALIZATION

Sound source localization is the technique to locate the direction and distance of a sound source. This technique has been extensively studied in the biological sense and for robotic applications. The human ear listens to sound and captures its intensity, spectral and timing information. From this information the brain is able to determine an estimate of the direction and distance of the sound source [4]. This is an immensely complex system which is difficult to reproduce mechanically. Yet there have been numerous attempts to do so, because it is desired to make human-like robots react to their surroundings by listening [5] [6].

In general a sound source has two characteristics which could be used in sound source localization. These are the phase and the energy level of the signal. Based on these two characteristics there are two methods of sound source localization. The first method uses the phase of a signal and is the method of inter-aural time difference (ITD), also known as inter-aural phase difference (IPD) or time difference of arrival (TDOA). The second method uses the energy level of a signal and is the method of inter-aural level difference (ILD) or inter-aural intensity difference (IID).

The ILD method uses the difference in energy level of a signal measured on two or more microphones. In this case the difference in travel time from signal to microphones will result in a difference in the energy level of the measured signals. Sound sources produce sound with a certain energy level. This energy level decays over distance according to the inverse square law. A microphone that is further away from the sound source than another microphone will therefore measure sound with a lower energy level than the measured sound of the microphone closer by the sound source. This difference in energy gives the ILD.

The TDOA method uses the shift in time between two signals measured by two microphones. If the signal has a longer travel time to one microphone than the other there will be difference in the phase of the measured signals. The time shift is directly related to this phase difference. This happens when the microphones have a different distance towards the sound source.

Another method to determine the location of a sound source which is closer to how the human hearing works is by combining TDOA and ILD. The TDOA could be used to find the directionality of the sound source while ILD could provide information on the distance of the sound source.

We used only the method of TDOA because it is possible to completely locate a sound source with the time shift information alone. We think ILD will not be necessary and only unnecessarily complicate our setup if we decide to use it. Considering the size of a mosquito we consider an error in the sound source localization of 0.3 cm to be acceptable.



DETERMINING PHASE DIFFERENCE

When a signal is measured by two microphones at a different distance from the sound source, the signals will be shifted relative to each other. The signal then has a longer travel time to one microphone compared to the other.

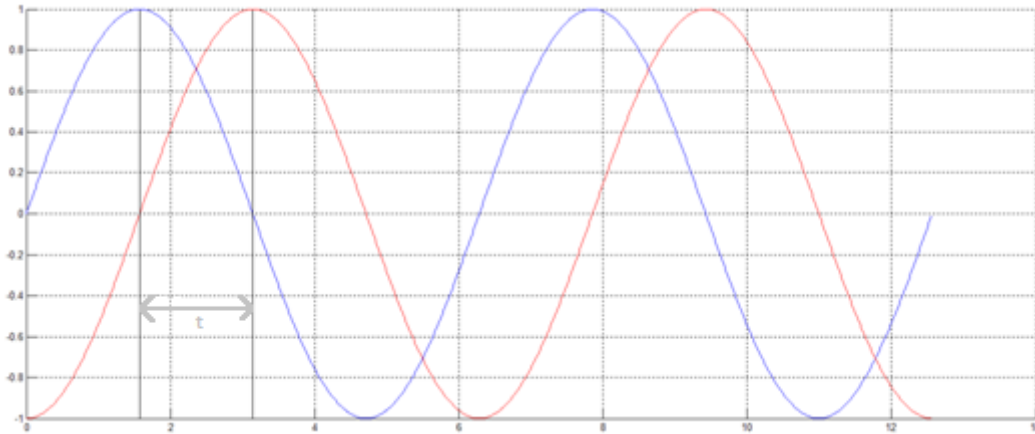


Figure 4: Illustration of measured time delay t due to difference shift between two measured signals. This time delay is the TDOA.

By comparing the signals, the phase difference can be measured. The most commonly used method to extract the phase difference from the measured signals is by using a Generalized Cross Correlation PHase Transform (GCC-PHAT) [7]. One of the signals will be used as a reference signal for all other measured signals. The GCC-PHAT uses a cross correlation to find a time shift between two signals and applies a phase transform to get the phase difference.

Another way to extract phase information is the Adaptive Eigenvalue Decomposition (AED). AED uses eigenvalue decomposition on the covariance matrix of microphone signals. The eigenvector corresponding with the minimum eigenvalue of the covariance matrix contains the impulse responses between source and the microphone signals. All required information to extract a TDOA is present in the eigenvector. The eigenvector turns out to be unique. [8].

A comparison between both methods [9] states that in an environment with considerable noise and a moving sound source the GCC-PHAT method is more accurate than the AED method. From this we conclude the GCC-PHAT is more suitable for the bug zapper. However, we only need the TDOA, so after applying the cross correlation on two signals we measure the time shift by using peak detection. The displacement of the most overlap (highest) peak from zero in the cross correlation is equal to the TDOA.

LOCALIZATION METHOD

Before calculating the sound source location it needs to be clear whether we are in the near-field or far-field regime. In the far-field regime the incoming sound wave front is assumed to be flat, while in the near-field regime a spherical wave front has to be considered. The localization method depends on the regime the sound source is in. A mosquito needs to be in the near-field regime in order to acquire a useable signal in our setup. This means the distance between two microphones is larger than the distance between a microphone and the sound source. With the TDOA known it is possible to calculate the position of the sound



source. The TDOA needs to be converted into the Distance Difference Of Arrival (DDOA). The DDOA is instead of the time difference the sound wave takes in order to travel from the closer microphone to the microphone further away from the sound source, the difference in distance between the closer microphone and the microphone further away from the sound source. To convert the TDOA to DDOA, the TDOA needs to be multiplied by the speed of sound. The speed of sound is dependent on temperature and can be approximated by the formula $v_{air} = 331.3 + 0.606T$ with T the temperature in °C.

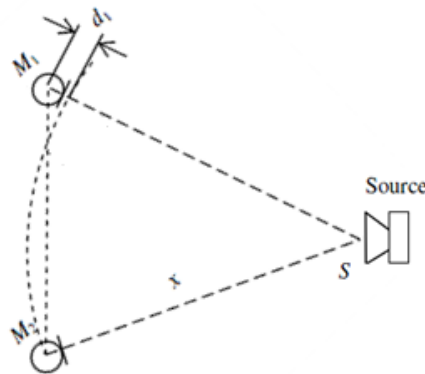


Figure 5: Illustration of the DDOA. Two microphones M1 and M2 located at distances $x+d1$ and x . The distance $d1$ is the DDOA.

To get a grasp on sound source localization we tried to find all possible locations between reference microphone and a second microphone for which the calculated DDOA holds. This means all positions in space for which the distance from one microphone towards this position equals the distance from the other microphone to this position plus the DDOA. All the solutions which obey this statement create a hyperboloid [10]. Repeating this process for at least 3 pairs of microphones gives three hyperboloids which can intersect and determine a unique position. This position then has to be the position of the sound source.

Solving the intersection of three hyperboloids is a complex non-linear problem. We simulated this method numerically to determine the optimal sound source position. However, calculating all the possible solutions requires a lot of calculation time, because for each position within a determined boundary the statement has to be checked for truth. This method is therefore considered unsuitable for the bug zapper, but gives a good general idea of sound source localization.



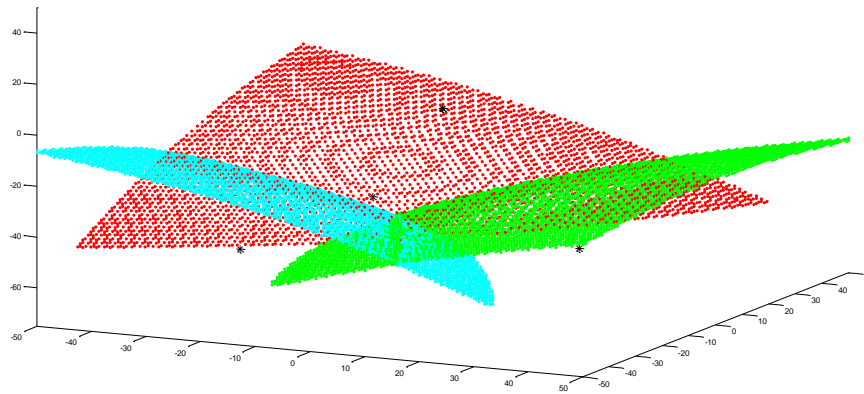


Figure 6: Illustrates the first method to localize a sound source, by means of intersecting hyperboloids.

A suitable algorithm for sound source localization for the bug zapper has to be fast and accurate. We used a linear closed form algorithm proposed by M.D. Gillette and H.F. Silverman [11]. This algorithm calculates the position of the sound source in a near-field regime acquired by the intersecting surfaces of a linear approach of the above method analytically. It needs five microphones instead of four. The advantages are that the calculation is extremely fast since the solution is acquired analytically. M.D. Gillette and H.F. Silverman state that; “In comparison to other methods to localize a sound source this algorithm is shown to have about the same sensitivity. An advantage of this algorithm is that it is easy to program and has a great variety of useable microphone arrays” [12].

LASER

The system is designed to use a laser to shoot down a mosquito after it has been localized. Of course this comes with some safety issues. The laser should only harm the mosquito and not any person walking by unaware of any risks. To be able to eliminate a mosquito by laser the laser needs apply 100 mJ of energy to the mosquito [12]. The best way to apply this is by using a short pulse when the zapper has localized a target. Lasers are usually most harmful to a person when it hits the eye, because the high energy beam will get focused on the retina and may cause permanent damage. This is best prevented by using a laser in the ultraviolet or mid infrared wavelength regions around 100 nm or 10 μm . Ultraviolet and mid infrared wavelengths get absorbed by water better than other wavelength regions. Most of the energy that hits the eye will be absorbed before getting focused on the retina and causes less damage. Especially if the energy is low enough to not cause burning in other areas of the eye.

Another reason the mid infrared wavelengths are desirable is because mosquitoes are insects and insects have a large amount of chitin in their exoskeleton and internal parts of their body. The laser pulse will be absorbed by the mosquito and eliminate it by heating it. A mosquito is considered eliminated when the laser applied enough energy to at least burn its wings. Burning the wings on a mosquito makes it incapable of flight and therefore no longer a threat.

A suitable laser for this task is a 2.9 μm wavelength laser since chitin has a high absorption peak for this infrared wavelength [13]. A 2.9 μm wavelength laser has a Maximum Permissible Exposure of 0.4 J/cm². To effectively eliminate a mosquito you want to hit the



entire surface area of the mosquito. A mosquito has a minimum area of 0.3 cm by 0.3 cm that can be hit, excluding its wings. If we take the laser to be 0.5 cm by 0.5 cm we will surely hit the entire mosquito and also have a chance to burn its wings. If we want to apply 0.1 J in an area of 0.5 cm by 0.5 cm the energy of the laser has to be 0.4 J/cm^2 . The laser has approximately 0.1 ms maximum exposure time with energy of 0.4 J/cm^2 and $2.9 \mu\text{m}$ wavelength. To output 0.1 J of energy in 0.1 ms means a 1000 W laser would be needed. However it is not desired to use a laser over 1 W of power. The maximum allowed exposure time will need to be exceeded and the laser is therefore no longer eye safe.

SIGNAL PROCESSING

The emitted sound measured by the microphones has to be processed by a cross correlation, peak detection and an algorithm before we know the sound source location. There are two more important aspects regarding these processes we have to mention.

After the signal is measured we apply a cross correlation. The cross correlation needs to give an accurate as possible estimation of the phase difference. To improve the accuracy we cut the measured signal and cross correlated this with the reference signal. The cut of the measured signal is done by dividing the signal in four equally large sections and taking the middle two sections to cross correlate. This will prevent the peaks from shifting to the middle of the cross correlation because when the two signals have the same length, they automatically have less overlap when they are not completely on top of each other.

Accurate location estimation is important to hit a mosquito. Considering sound source localization calculation, the distance from sound source to microphone cannot be bigger than the wavelength of the base frequency. If the distance is larger than the wavelength of the base frequency the location of the sound source could be shifted by one wavelength. The cross correlation cannot determine whether the sound wave has traveled more than an entire wavelength. The sound source will be positioned closer than it is in reality.



EXPERIMENTAL ASPECTS

Since it is theoretically possible to locate the position of a sound source we should be able to make the bug zapper work. We built a setup to bring this to the test. Every part of the setup has been thought through thoroughly and has been tested before the whole thing is put together to try to locate a mosquito with high precision.

SOUND SOURCE

For practical reasons, we chose to simulate the mosquito with a small speaker. The main argument for using a speaker is that we can control where and when it makes a sound and at what volume and frequency. To realize a decent volume, we used an old speaker of 5 cm by 5 cm by 1.8 cm. To locate the position of the sound source, it is important that it behaves as a point source. If this isn't the case, DDOA errors are easily generated and these will cause positioning errors.

MICROPHONES

With the emission of sound taken care of, the next step is the reception of the emitted sound. The microphones themselves must be very sensitive to hear the signal so the original sound can be extracted from the ambient noise with the right techniques later in the process. If the quality of the recorded signal is not high enough this introduces bigger errors in determining the TDOA. They must also be unidirectional because if the received signal is distorted dependent on its incoming angle, it will be very difficult to calculate where the signal came from.

As the emitted sound travels through space it picks up noise, reducing its quality. The signal recorded by the microphones can roughly be divided into 3 parts.

- 1) The sound emitted by the speaker. The signal has a base frequency of 375Hz with a strongly present third higher harmonic.
- 2) Random noise is different for every microphone and will not affect the outcome of the calculations because this part reduces when cross correlating.
- 3) Ambient noise consists of signals that are picked up by every microphone but do not originate from the position of the mosquito. For example the sound of a computer fan, air-conditioning, reverberations or the 50Hz (and higher harmonics) of the power grid and reverberations.

This last one is the most unwelcome since the presence of another sound source will alter the outcomes of the cross correlation. To get rid of most of the noise and retrieve the best signal we developed an IC with a bandpass filter and tested it to know its characteristics.



IC REQUIREMENTS

Since the cross correlation should work with the frequencies of the mosquito it is desirable to filter out the frequencies below that of the base frequency of the mosquito. The power grid will account for some disturbances in the signal at 50 Hz and higher harmonics. Besides that there always is noise from the surroundings. This could really be anything¹. To also eliminate the high frequency ambient noise we decided to use a band pass filter. Unfortunately, the ambient noise in band pass frequency range will still be part of the recorded signal.

Besides the noise filtering, amplifying the measured signal makes it have a better SNR ratio when random noise enters after the filtering. The amplification is also necessary to have a good resolution in the minimal Volt range of the DAQ-card ($\pm 0.2V$).

To realize these wishes and minimize the time it takes to execute them we designed an active analog band filter with 100 times amplification.

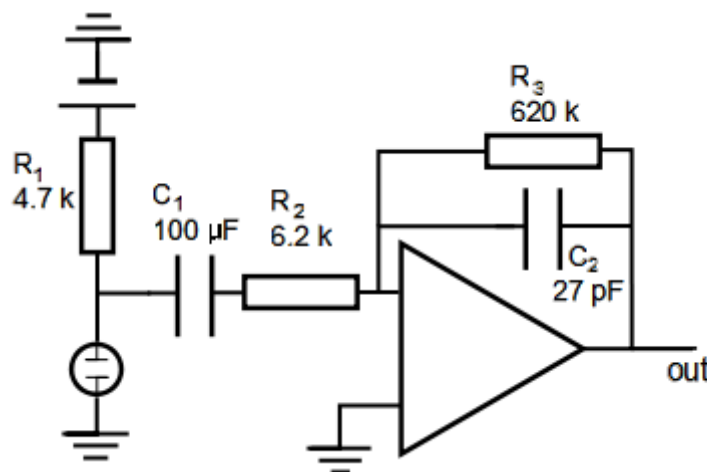


Figure 7: The IC used in every microphone to measure the sound waves.

The base frequency of the mosquito is around 375 Hz. We did not want to filter out the first higher harmonics since they are very characteristic for the mosquito sound and the calculations will provide more precise data with these higher harmonics left in so the stop frequency should be beyond 1400 Hz. An analog filter works by the impedance it has for certain frequencies. A capacitor and a resistor in series provide high impedance for low frequency signals while the same components in parallel configuration provide high impedance for signals with a high frequency. The frequency at which the filter starts/stops working effectively is called the -3dB point. At this point the signal has 50% of its maximum energy. The following formula describes the position of the -3dB point: $\frac{1}{2\pi RC}$. This leaves the following specifications of the IC.

$$R_1 = 4.7 \text{ k}\Omega$$

$$R_3/R_2 = 100$$

$$\frac{1}{2\pi R_2 C_1} \approx 300 \text{ Hz}$$

$$\frac{1}{2\pi R_3 C_2} \approx 6000 \text{ Hz}$$

To obtain the right voltage over the microphone

The maximum amplification the op-amp can handle

-3 dB point of highpass to filter the power grid frequency

-3 dB point of lowpass to filter high frequency noise

¹ For example: the lab technician using the power drill 2 meters from your setup



Solving for these equations gives us the following values for its components.

$$R_1 = 4.7 \text{ k}\Omega$$

$$R_2 = 6.2 \text{ k}\Omega$$

$$R_3 = 620 \text{ k}\Omega$$

$$C_1 = 100 \text{ nF}$$

$$C_2 = 27 \text{ pF}$$

The highpass filter has a -3dB point at approximately 256 HZ and the lowpass filter has a -3dB point at approximately 9.5 kHz.

We chose to work with an OP27 (op-amp) because it has a high SNR and operates at a high frequency which makes it excellent for audio amplification applications. The microphone type is AOM 4540P-R, chosen for its cheap cost.

DATA ACQUISITION CARD

In order to digitally process the signal we need to use an A/D-converter. This device converts an analog signal into a digital signal so they can be read into a computer and analyzed. The device we use to acquire this data is the DAQ-card². It can measure up to 400.000 aggregate samples per second. An aggregate sampling rate means that when data from multiple channels is collected, the first measurement starts at channel 0 and it measures the next channel 2.5 μ s later and so on. The DAQ card is incapable of measuring multiple channels at the same time. The maximum frequency at which it can measure one channel is thus dependable on the number of channels used.

In this setup there will be 5 microphones used so the maximum sampling rate per microphone is 80 kHz. It is important to note that there is a time difference between the n^{th} measurements of each microphone. If there is an error of 2.5 μ s in the TDOA, this accounts for difference 0.8 mm in de DDOA. This should be compensated for.

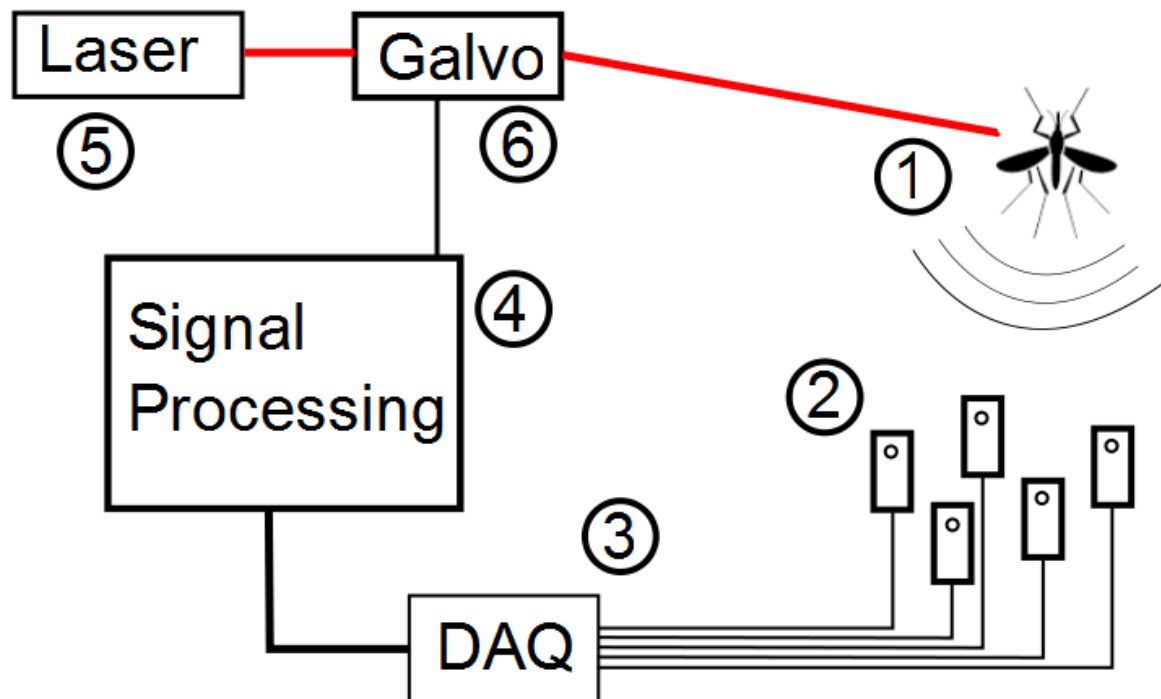
SETUP

The image below shows a schematic overview of the setup as it should be. For our research we have only used elements 1-4³ because we find it important to know for sure that the system works well and is safe before we start to shoot lasers.

As the mosquito (1) makes a sound the microphones (2) measure it and the DAQ-card (3) converts the analog signal to digital. In a computer (4) is determined what the position, speed and direction of the mosquito are. With the current estimated position the angles of

² USB 6212 OEM





- 1) Sound source:
- 2) 5x microphone IC
- 3) DAQ - NI USB-6212
- 4) Live data processing with labview and matlab
- 5) Laser, at least 1W
- 6) 20Kpps laser scanning galvo

Figure 8: Schematic of the setup used to locate and disable a mosquito.

ERROR ANALYSIS

In order for the system to be able to localize a mosquito reliably, the localization will have to be accurate within a distance of 3 mm. To reach such high precision means decreasing errors significantly. Determining the location of the mosquito introduces errors in the analog data acquisition and digital processing.

ANALOG DATA ACQUISITION

Even though the microphones are designed to be similar the electronics could behave slightly different because not all microphones have the same sensitivity and they act as a parasitic capacitance. Not all resistors and capacitors have the exact value we calculated. This creates an offset in the relative phase of the signals due to slightly different phase characteristics.



The signal is not recorded infinitely accurate and while calculating there will undoubtedly be rounding errors. On top of that, the calculations are done with the signals where some errors already have slipped in at the first stage. The calculation for estimating the location of the sound source also requires the exact position of the microphones. Measuring the position of the microphones also introduces errors. The following graph shows a simulation of the error in the calculated position caused by the error in the measured microphone position.

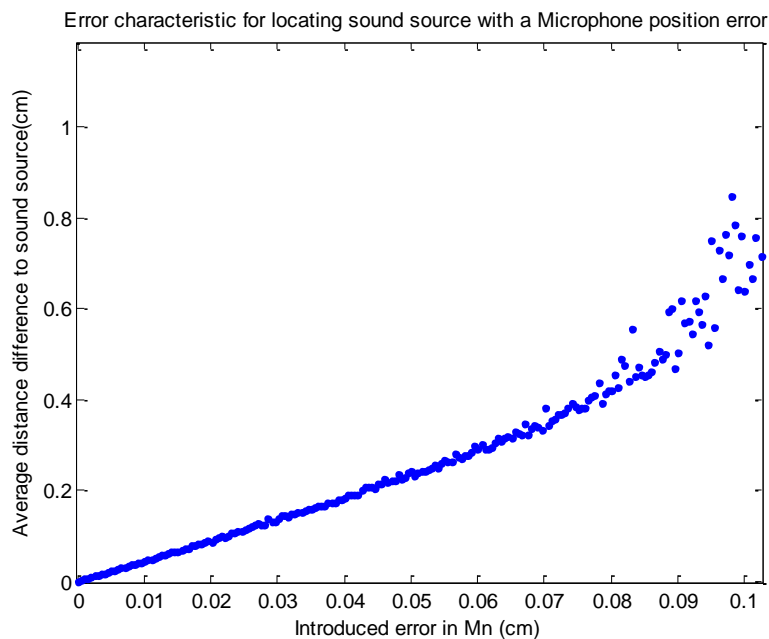


Figure 9: Error characteristics of the sound source localization algorithm.

There will also be an error in the DDOA that is calculated, there can be several reasons for this error to exist: the sound source can be not unidirectional or not a good point source. If there is a wind blowing that distorts the uniform propagation of the sound waves or the speed of sound is not what you expected due to unexpected temperature or humidity changes the DDOA is also affected.⁴ The graph below shows a simulation of the error in the calculated position of the microphone caused by the error in the determined DDOA.

⁴ We actually had some problems with the thermostat so there could be a temperature difference of 5° between morning and evening.



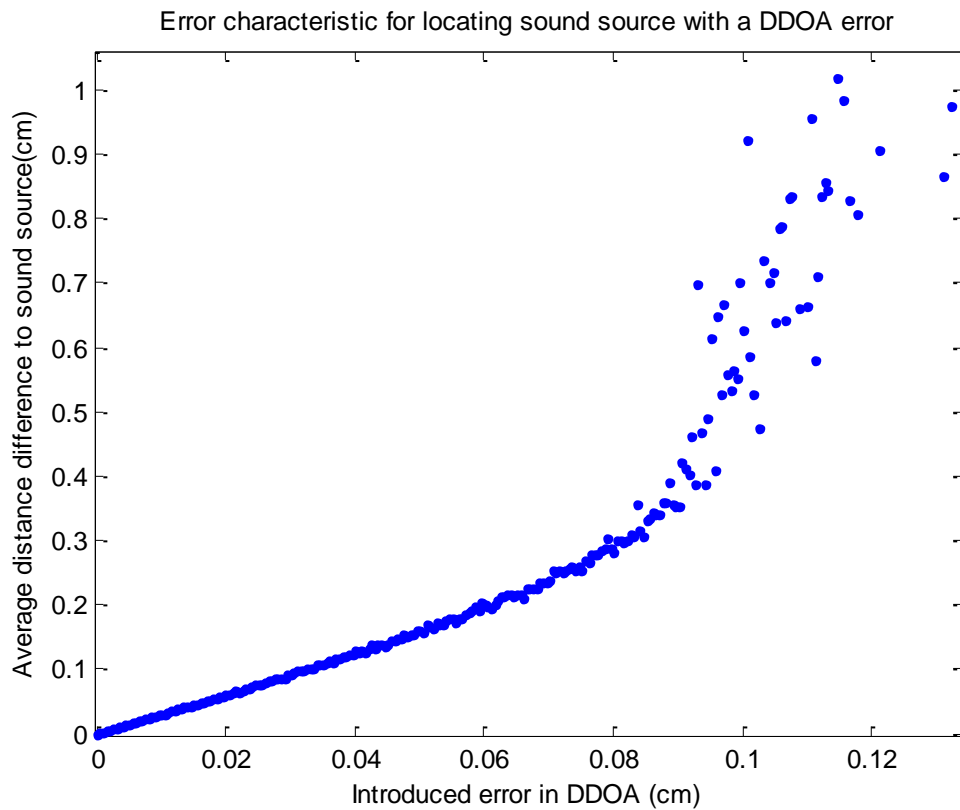


Figure 10: Error characteristic of the error in measured microphone position

ERROR ESTIMATION

There are only a few physical errors but the slightest error in the beginning can have a large effect on the localization in the end. As the figure below shows, the signal collected by the microphone IC is already influenced by ambient noise and imperfections of the IC. With this impure signal the TDOA is calculated. The peak position is determined using a quadratic fit. From the TDOA we multiply with the speed of sound, which depends on temperature. With the DDOA's and positions of the microphones we use the algorithm to localize the sound source.

For the complete system, by this time the mosquito has already moved a bit and with the aid of former measurements its current position has to be estimated. Once this is calculated, a signal is sent to the laser. Aiming the laser with the galvano mirror should be possible with negligible error.



Error chart

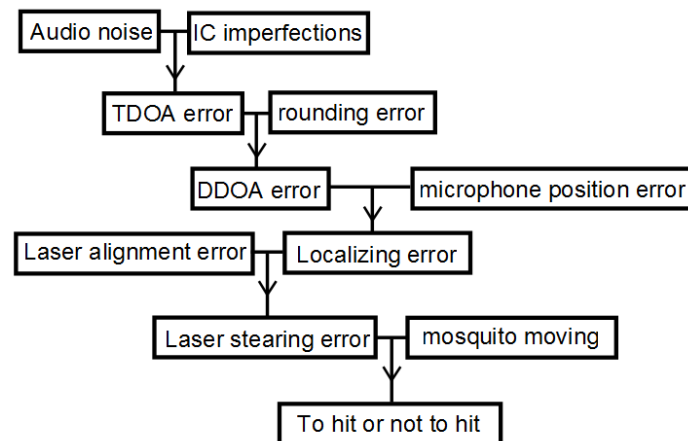


Figure 11: Error characteristics of the sound source localization algorithm.

The influence from the audio noise on the TDOA calculation depends on the SNR ratio. While testing the setup we will make sure to work in a quiet environment and we expect the effects from ambient noise to be negligible. Imperfections in the microphone IC deform the signal and the DAQ card cannot acquire perfectly as well. We expect the error in the TDOA caused by this to be about $0.5 \mu\text{s}$ since the different IC's process their signals slightly different. A quadratic fit is applied to locate the peak position of the cross correlated signal. We expect to find an error of $0.5 \mu\text{s}$ in the TDOA due to the fit. When added to the error from the imperfections in the IC, this $1 \mu\text{s}$ accounts for an error of approximately 0.34 mm in calculating the DDOA.

While calculating the DDOA from the TDOA an error of 0.5% of the DDOA value can be expected due to temperature differences that cause a difference in the speed of sound and therefore an error in calculating the DDOA (0.5% accounts for an unknown shift in temperature of 2°C). DDOA distances can go up to 10 cm so the expected error is 0.5 mm . This adds up to a 0.85 mm error in calculating the DDOA.

The position of the microphones has to be measured since this is used in calculating the sound source position. If measured carefully, this can be determined up to 0.5 mm .

Both these errors determine the error in the positioning algorithm. For the DDOA error of 0.85 mm this means the mosquito position calculation is off by approximately 0.5 cm , as shown in figure 10. The microphone positioning fault of 0.5 mm accounts for an error of 0.25 cm , as shown in figure 9. This adds up to a total localization error of 0.75 cm .



RESULTS

To get a good idea about the performance of the designed system we tested it. All test results of the single parts and the complete system are shown below.

MICROPHONE CHARACTERISTICS

Before we tried sound source localization we measured the microphone characteristics to make sure they have the same response. Some general testing in labview showed us the microphones successfully recorded data. For the measurements we sent signals with different frequencies to the speaker and compared the incoming signal with the emitted signal.

We measured the time delay per frequency in the frequency area we have an interest in. If one microphone would have a different time shift compared to the other microphones for a certain frequency it would cause an incorrect DDOA difference when calculating the DDOA. For a single-frequency signal we could easily alter this systematic error. However, if the time difference error is not the same for all frequencies, the DDOA is likely to be calculated incorrectly and it would take some calculations (and therefore time) to determine the correct DDOA.

The microphones show a frequency-dependable time delay (see figure below) but since this is roughly the same for every IC we expect it will not affect any calculations significantly.

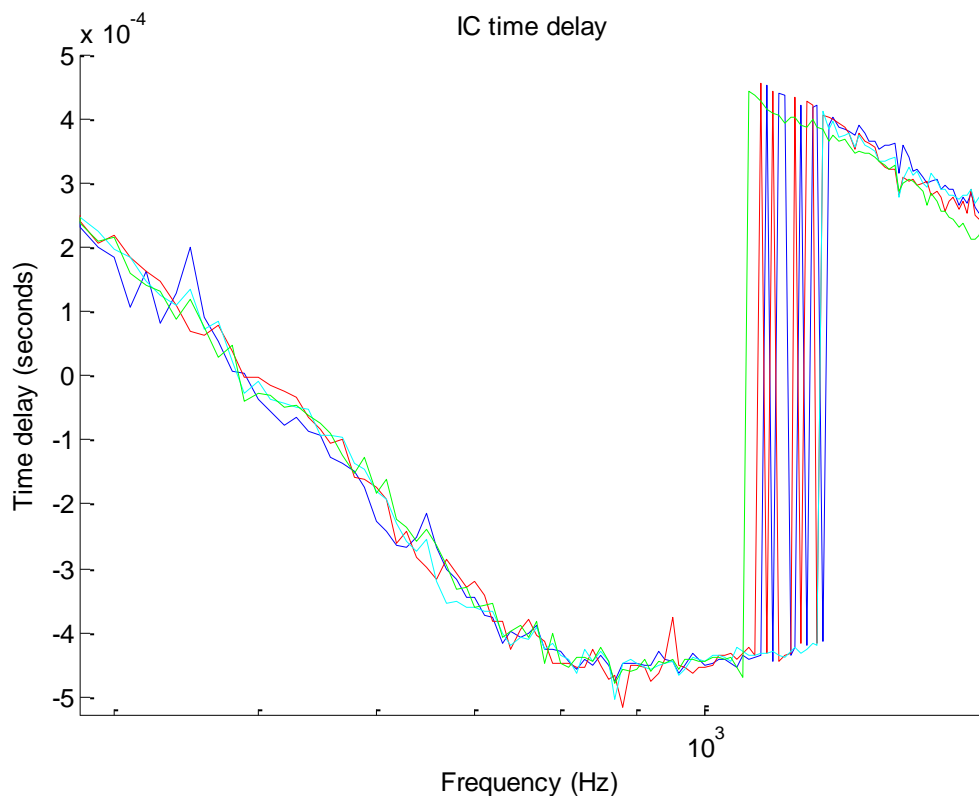


Figure 12: Phase response of 4 microphones.



Besides frequency response the amplification factor for different frequencies is vital to understanding the IC. The graph below shows us that the -3dB point we calculated to be approximately 256 Hz turns out to be around 550 Hz. The speaker possibly emits low frequency sound softer than expected. This weakens the amplitude of the base frequency of the mosquito (375 Hz) quite a lot. This is not a problem since the shorter wavelength of 1125 Hz should provide us accurate information.

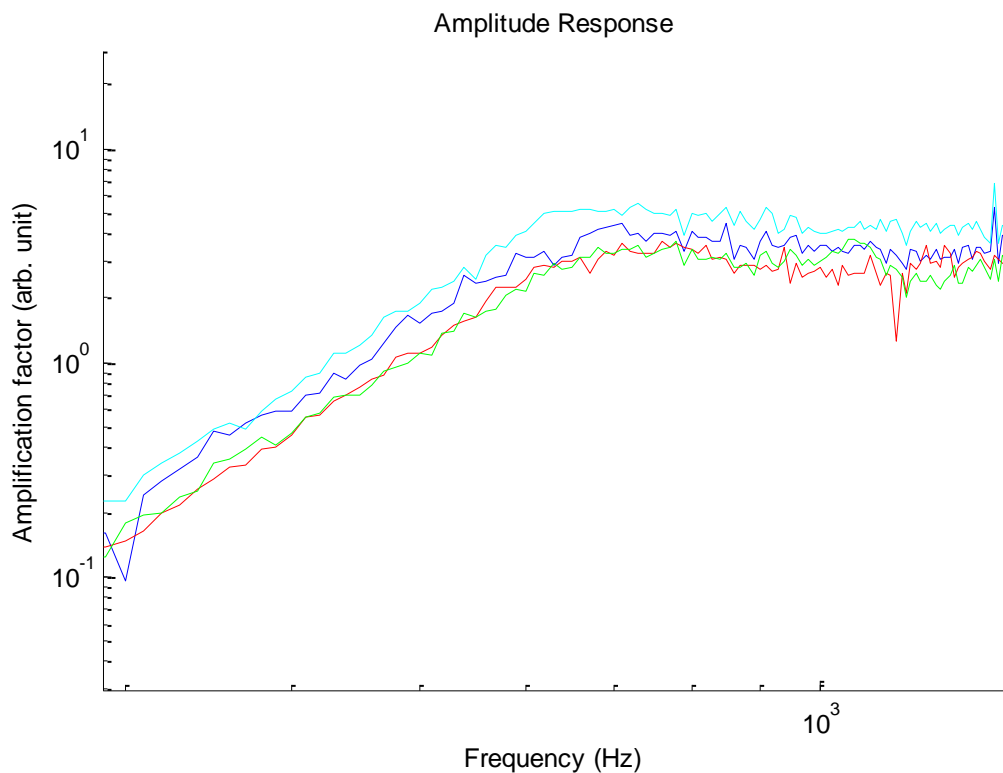


Figure 13: Amplitude response of the 4 microphones.

The difference in amplitude response of microphones relative to each other in the above graph occurs because during the measurements the microphones were not fixed in position. The travel time of the signal was different for each measurement and this resulted in different energy losses per measurement.

DETERMINING SPEAKER UNIDIRECTIONALITY

We assumed the speaker to be a point source emitting sound spherically outward in all directions. The used speaker however has dimensions so we need to test whether this assumption is valid⁵. When the speaker dimension is smaller than the wavelength emitted the speaker is considered subwavelength. Subwavelength speakers emit spherical sound waves better. If the speaker dimensions get closer to the wavelength it emits, the sound wave will be more planar. To see how the wave front of the sound emitted by the speaker is

⁵ We tested this after we tested and improved the system since we only then started doubting the unidirectionality of the speaker. Had we known this, we would have put more effort in finding a sound source



distributed in space, we placed one microphone about 15 cm opposite to the sound source and moved another microphone around to see at what positions the DDOA between the microphones would be equal to zero. The red point represents the microphone opposite of the sound source. The purple points are the expected positions for when the DDOA equals 0 if the source was a perfect point source and the blue points are the positions of the microphone for which we measured the DDOA to be 0. We performed four measurements, each with a different microphone to move around.

Top view of measured wave front of speaker (+) versus expected wave front (x) of four microphones relative to the same reference

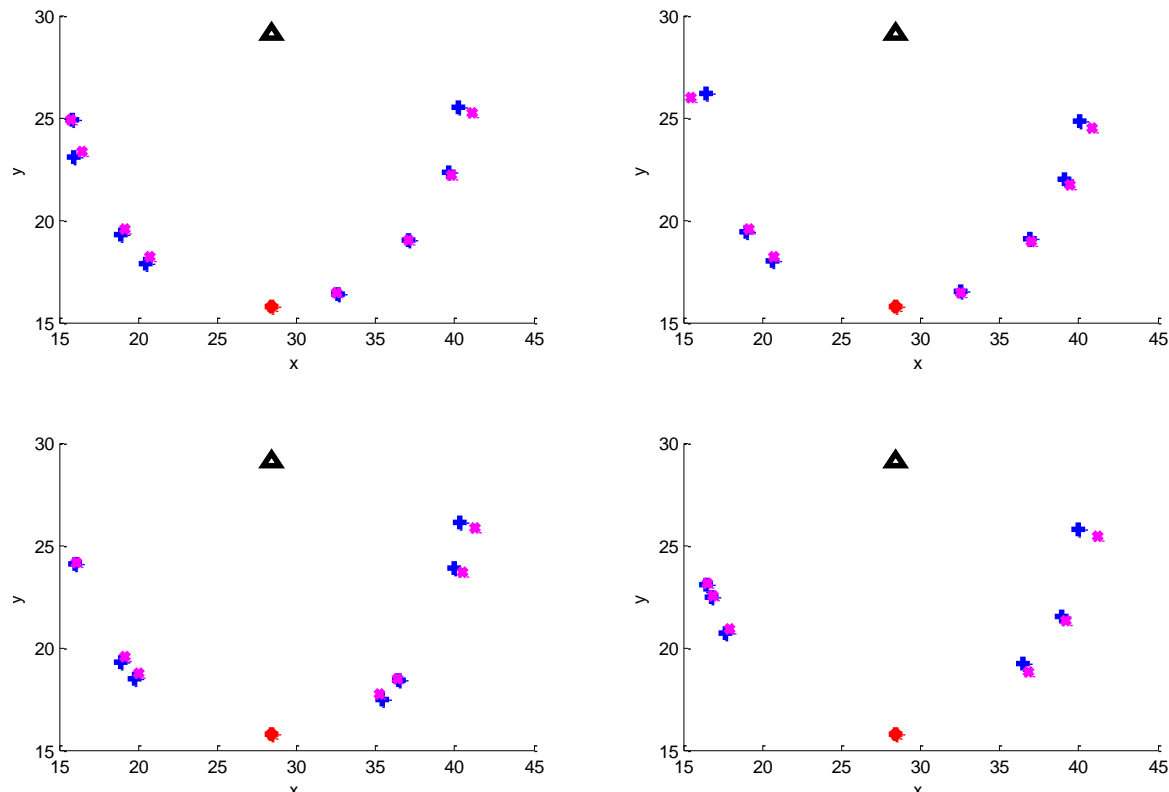


Figure 14: Shows how well the sound source behaves as a point source

The data shows that for larger angles of the speaker, the microphone positions are closer to the speaker than we had expected them to be. We had expected an opposite effect because of the size of the speaker: In the experiment we assume the sound comes from a point source but the speaker is actually 5 cm in diameter. At large angles, the sound could have been emitted from 2.5 cm more towards the microphone than we assumed for the experiment which would make the measured spot of the wave front further away than the expected point indicates.

For both the 1st and 3rd harmonic – the most important frequencies of the sound – the speaker size is subwavelength. For the larger wavelength (375 Hz) the speaker is more subwavelength and thus has less directionality than the smaller wavelength (1175 Hz). This means that at larger angles the lower frequencies are relatively better emitted than the higher frequencies. This causes errors in determining the DDOA.



It's worthy to note that on the right side it is very clear that the measured point is a lot closer than the expected position while this effect does not show clearly on the left side of each graph. This shows the speaker was not a perfect point source and the emitted sound may have been dependent on direction.

SOUND SOURCE LOCALIZATION

With all components added together the position of the sound source can be determined. To determine how well this works, we positioned the sound source at 24 different positions, always aiming in the negative x-direction and measured:

- Position of the sound source
- Position of the speakers
- DDOA's calculated from the cross correlation
- Calculated position from the sound source from the locating algorithm.

The measurements resulted in the following graph. The red dots represent measured locations of blue dots they are connected with. The blue dots represent the actual location of the speaker.

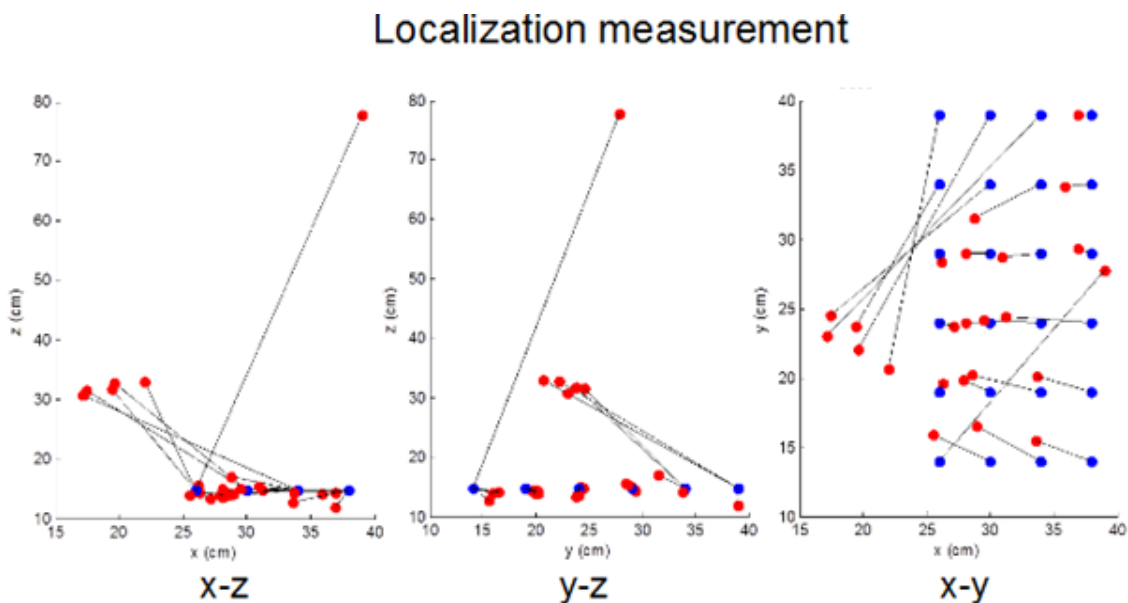


Figure 15: Sound source location measurement results.

These graphs show 6 poorly measured positions. We expect that this happened because the speaker had a large angle to reach the microphones and this accounts for a DDOA error as shown before⁶. The large distance between the speaker and microphone could have caused errors because there could have been an extra wavelength added to the DDOA. With these points excluded this shows a fairly well-defined vector field of errors, best visible in the xy-plot. So whatever is causing these errors, there is very likely some systematic error and not a random one. Several points with the same x-value have the same error in the x-value as

⁶ Determining speaker unidirectionality, page 20,21



shown in the x-y view for the four positions in the lower right corner. The error in the measured position of the microphones could be responsible for causing this field of errors or there could be some relative delay between one and another microphone that we have not accounted for yet.

To find out what was going on, we compared the DDOA's calculated by the cross correlation with the theoretical DDOA's, calculated from knowing the positions of sound source and microphones. For all measurements, this gives the next graph. On the x-axis is the number of measurements and on the y-axis the relative DDOA's: One value determined by cross correlation between two microphone signals (measured) and one value determined by geometrical calculation (expected). The graph shows some constant difference in the measured and calculated DDOA for the red and magenta curves. The green lines show these effects a little bit and the black line looks pretty accurate. The figure clearly shows which 6 measurements gave questionable results.

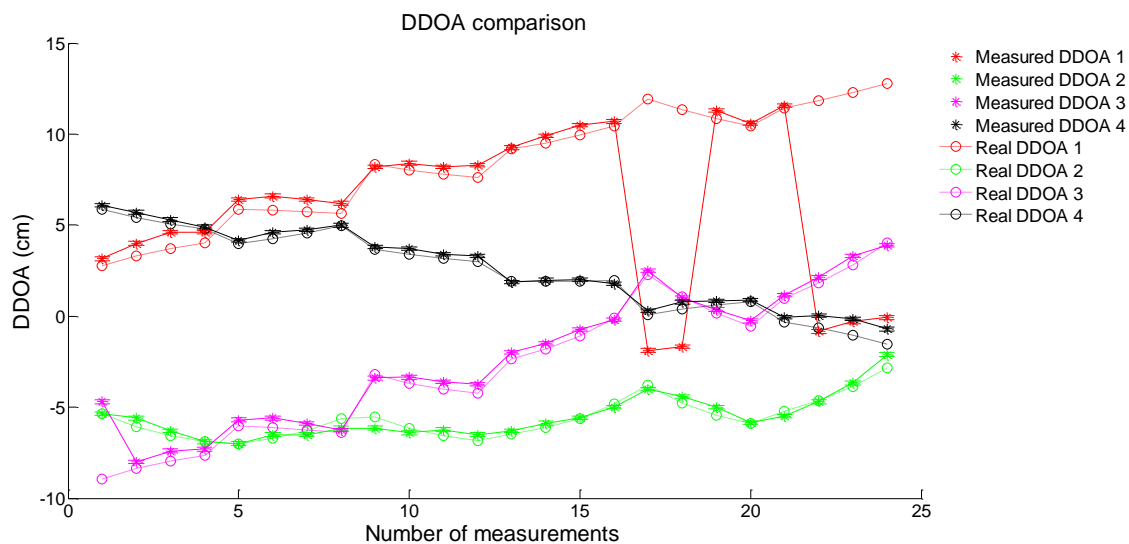


Figure 16: Comparison between real DDOA values and measured DDOA values for 4 microphones relative to a reference.

The errors between real and correctly measured values were likely caused by the systematic errors that still reside in the system. We did the most basic thing to eliminate these errors: After the cross correlation, when the DDOA is calculated we subtract the extra DDOA difference we found for this setup from the calculated DDOA to approach the theoretical value more closely.

After applying this correction we measured a second time to see whether the sound source localization had improved.



2nd localization measurement

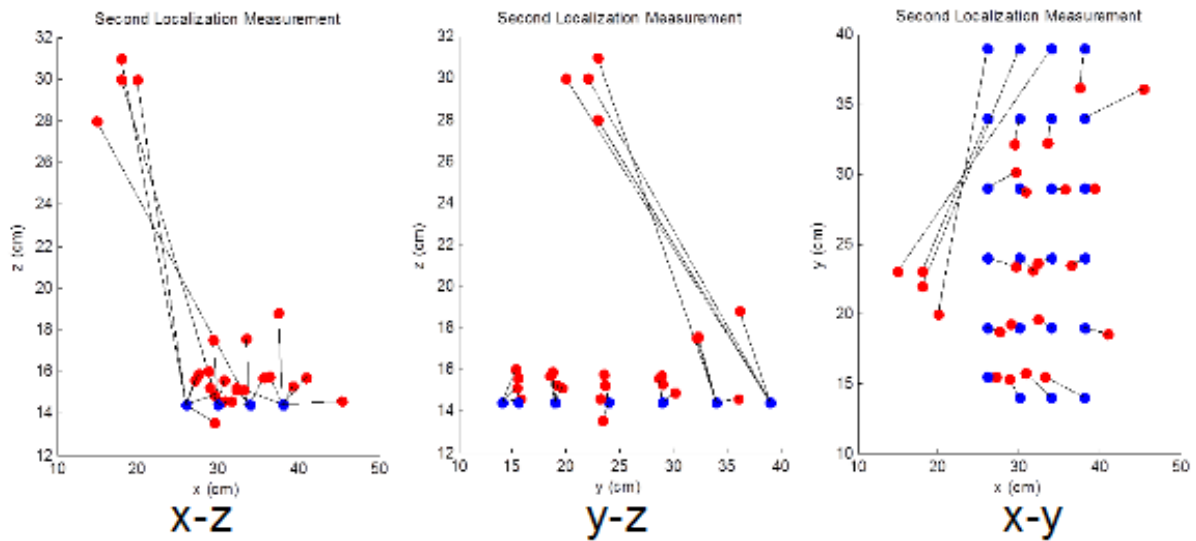


Figure 17: A second series of measurement of sound source locations

A new series of measurements shows we did increase accuracy with this action but not by much. The y-z view looks promising because it shows very clear that almost all position calculations are off in the positive z-direction and the y-coordinate of most measurements is very accurate. A simple adjustment could make all calculated z-positions some 2 cm lower, increasing the accuracy. The x-direction of the measurements is only slightly improved. As can be seen in the x-y view, some area's (line $y=14$) are off in the negative x-direction while elsewhere ($y=29$) the calculated positions are off in the positive x-direction. The histogram below shows the error to the real position and the frequency at which that error occurred. Eleven of the measurements were accurate within 2.5 cm and another 6 were located within 4 cm of the actual position of the sound source.

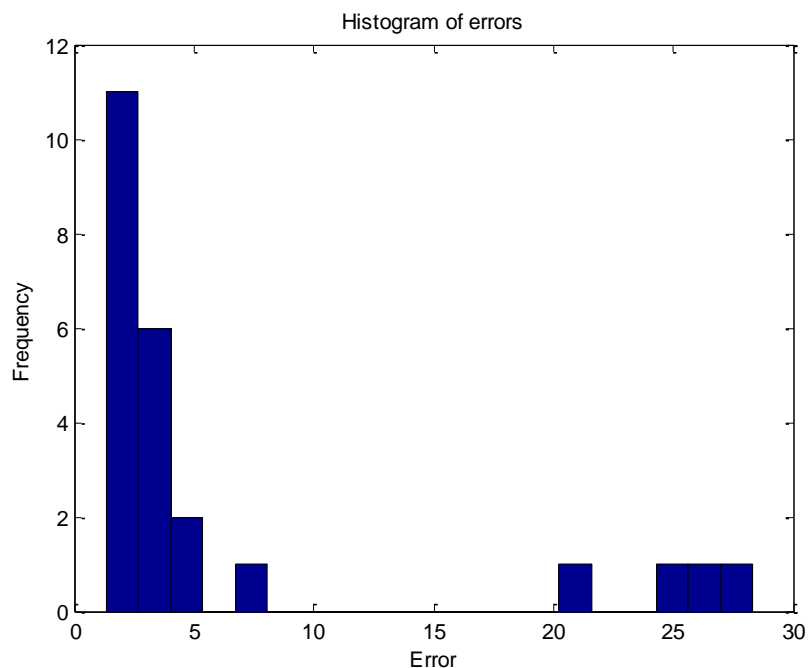


Figure 18: Error characteristics of the sound source localization algorithm.



Measurements showed that the system does have a high repeatability but the accuracy needs to be improved. Succeeding measurements of an unmoving sound source locate the source in the same volume of 0.75 by 0.75 by 0.75 centimeter, as we expected, but this volume is often not so close to the true position. Of all measured positions, 46% lays within 2.5 cm and 71% within 4 cm of their true position. This hints that we have the random errors under control but there are some systematical errors left that need to be dealt with. We expect that these come from the speaker not being a good point source and from errors in the localization of the microphones.

There are also some measurements which are way off from where the speaker was actually located. While measuring we noticed the DDOA of some microphones could differ over 10 cm while moving the speaker only 1 cm. We expect this effect is either caused by the large distance between the sound source and microphone, allowing for more than one wavelength of the 3rd harmonic to fit in this length. This causes the localization to calculate a position an entire wavelength away from the actual position plus the result from the calculated DDOA. This is clearly visible in figure 16 where the measured DDOA for the red line drop approximately 15 cm. Another reason this is so off could be because the directionality of the speaker, which is frequency dependant.

For a certain configuration of the microphones there is a confined volume where the sound source can be detected with high enough accuracy. It is necessary to determine the boundaries of this volume and only try to locate the mosquito there. We have not been able to determine the key parameters that can be used to calibrate the system for each random setup. In the current situation, the system has to be tested and altered for every unique configuration of microphones to make it work.



DISCUSSION

SOUND SOURCE

In trying to make the bug zapper we made a number of assumptions to do sound source localization. In reality not all these assumptions are justified. A mosquito is not a stationary sound source, but actually has to be moving to produce sound by flapping its wings. The bug zapper needs to have a quick reaction time to hit a moving mosquito. A mosquito has a maximum flight speed of 1.4 to 1.8 m/s [13]. The size of a mosquito in width is approximated to be around 3 mm. To hit a mosquito the localization of the mosquito and aiming of the laser needs to be completed within $\frac{3 \cdot 10^{-3}}{1.8} = 1.67 \text{ ms}$. The sound of the mosquito we used has the most dominant frequencies in the base frequency and second harmonic around 375 Hz and 1125 Hz. For a more accurate measurement of the phase difference it is desired to capture at least one wavelength completely. For a frequency of 375 Hz one wave passes by every $375^{-1} = 2.67 \text{ ms}$ which already exceeds the minimum required time for the system to react. Only a part of a wave of the base frequency can be measured which will create inaccuracies in the cross correlation. For a frequency of 1125 Hz one wave passes by every $1125^{-1} = 0.89 \text{ ms}$ which is within the said limit of 1.67 ms and will give a more accurate measurement. This doesn't pose any problem since the used IC in the microphones doesn't weaken this signal.

To improve the accuracy further we did an averaging over the last ten calculated DDOA's. This exceeds the minimum time allowed for calculation by far since every calculation in itself took about 0.25 seconds, due to a large number of samples to cross correlate. A moving mosquito will have left its position during this time. In successive measurements the localization will lag behind the position of the mosquito by 2.5 seconds but still follow its movement. 2.5 Seconds is an unacceptable lag, but if this lag could be reduced it might be possible to extrapolate the measurements to determine the position of the mosquito. A mosquito does not have to move in one direction and could abruptly make a turn. We cannot know when a mosquito makes a turn because its persistence length is unknown. An extrapolation will be ineffective when a mosquito turns during the calculation, because the laser will miss. If both the extrapolation and measurements are tracked they may overlap in case the mosquito holds the same position for a relatively longer time. This could be used as a criterion to determine whether a shot should be fired.

A mosquito will not always be emitting enough sound to get a proper measurement of its sound. The bug zapper needs a way to recognize when a mosquito is within the specified range to be shot down. The specified range should be small enough that all microphones will be able to pick up enough of the sound for a proper measurement. This feature can be implemented along with a filter function which is able to distinguish a mosquito sound from the surrounding noises. Noise in this sense includes other sound sources like different insects. If this works the bug zapper will be made specific. The filter function could always be tuned to different frequencies or wave patterns in the case one would want to eliminate a hornet.



A mosquito does not necessarily behave as a point source. This should be considered when the bug zapper is able to locate life mosquitoes and fire at them. In our setup we used a speaker we assumed to be a point source. In the results we showed this assumption was invalid. Errors occurred due to this false assumption. We compensated for these errors by adding a value to the calculated DDOA's. This is not a proper solution, but did provide a little bit of improvement on the sound source localization.

SOUND SOURCE LOCALIZATION

In sound source localization we discussed the method we used to localize a sound source, a linear closed form algorithm which directly calculates the sound source position. However, there are different methods which could be applied to localize a sound source. These methods have not been tested and could possibly work better. If the system is allowed to lag behind a little the calculation speed of the algorithm becomes less important and this allows slower, more accurate algorithms to be implemented which use better calculations to find the sound source location.

The linear closed form algorithm can only handle one sound source. In the presence of multiple sources the calculation of the TDOA will no longer belong to a single sound source. This will cause the algorithm to calculate an incorrect sound source location. In order for this algorithm to work properly the desired sound will have to be filtered from all ambient noise. It might be best to make a double bandpass filter and let only frequencies around 375 and 1175 Hz pass.

SIGNAL PROCESSING

In figure 1 it is visible the used signal has different amplitudes over the length of the sound. The length of the measured signal is determined by the sample rate and number of samples per calculation. We used a reference signal of 20.000 samples and cross correlated this with a measured signal of 10.000 samples. These samples were gathered at a rate of 80 kHz. Each 0.25 seconds we measured a signal. In this time period the amplitude of the signal could change by a factor of two. This could introduce error in the cross correlation. A period of 0.25 seconds has $\frac{0.25}{.89 \cdot 10^{-3}} \approx 281$ periods at 1125 Hz of the reference sound wave and 140 for the measured sound wave. To detect a phase difference only one period of the measured signal is actually required to be cross correlated over few periods of the reference signal. Doing this will require a lot less calculation time as well as omit large changes in amplitudes which could give errors.

The measured signals had amplitudes of approximately 0.2 V while the DAQ-card is able to handle amplitudes of 5 V. This shows room for improvement on the SNR. However, with the current components in the IC the op-amp was working at a bandwidth of 80 kHz which is the same as the sampling rate. If you would increase the amplification the op-amp will work with a lower bandwidth. This will create a lack of measured values which will have to be accounted for. An improvement in the SNR may seem like an improvement to the accuracy but will make the cross correlation have higher amplitudes and a lower resolution. This may decrease accuracy. There is a trade-off here in which the proper balance has to be found.



We think that the op-amp working at the same frequency as the DAQ-card provides a good balance.

The op-amp has a gain-bandwidth product of 8 MHz but with an amplification of 100 times this bandwidth drops down to 80 kHz. This means the op-amp introduces a lag of 12.5 μ s. The lag introduced by the op-amp should be taken into consideration when the sound source is no longer considered to be stationary. The best solution would be to put another active amplifier behind the first one. It's also possible to use an op-amp with a higher gain-bandwidth product to reach a higher amplification or to use a current feedback op-amp which bandwidth is independent of the gain.

The calculation of the DDOA is dependent on the temperature. In the room we did our measurements there was a thermostat but during measurements the temperature may have changed. Adding a decent thermometer to the system should define temperature within 0.5°C. Experiments did show the DDOA fluctuation is only around 0.8 mm so if we can get rid of the systematic errors the accuracy of the system should increase.

MICROPHONE ARRAY

The microphone array can be changed before a measurement and because their positions are used directly in the sound source localization this directly influences accuracy. We expect that less symmetry is better. To accurately determine differences in x, y and z directions from the linear closed form algorithm there should be as few as possible microphones in the same plane.

The used algorithm also allows more than 5 microphones and more than one 1 reference microphone to be used. It is not guaranteed that the array we used with 5 microphones of which 1 is the reference is optimal. Using seven microphones with one of them as a reference in the middle and two microphones to measure a signal in positive and negative x, y and z directions accuracy might be improved. In sound source localization adding more microphones gives more information by measuring more TDOA's and this in general improves the sound source localization. However, if the DAQ-card is still in use, multiple microphones acquire more of its capacity and the sampling rate per microphone could become too low. If the computation time to compute TDOA can be reduced significantly, one might consider using more than one microphone as a reference or maybe all of them. With 5 microphones 5 sets of 1 reference and 4 microphones can be made. Averaging over 5 position localizations may improve accuracy but will definitely cost some calculation time.



CONCLUSION

We determined that with the setup as described in this paper we can do sound source localization measurements with 0.75 cm repeatability due to errors in calculating the TDOA's. Inaccurate measurements of the positions of the microphones, false assumptions made on the sound source and inaccurate measurements of the temperature make it difficult to accurately localize the source. The sound source localization was accurate within 2.5 cm errors for 11 out of 24 measured positions. To make this a suitable system for a bug zapper the accuracy and repeatability need to be improved to work within 0.3 cm error.

A laser can be implemented in the system which is capable of shooting at a mosquito by steering the laser with a galvano mirror. We expect the errors in steering the laser to be negligible compared to sound source localization errors. In order for the laser to be eye safe it would need to have 1000 W of power.



RECOMMENDATIONS

The bug zapper has come a long way but is still far from perfect. We think this system could be made to work. In order to realize this we recommend looking into the following:

- The system still lacks a laser to shoot at mosquitoes. It will be necessary to explore the options of the galvano mirrors to aim the beam as precise and fast as possible with enough energy to eliminate a mosquito.
- The bug zapper will have to be able to localize a mosquito in a noisy environment. The possibility of filtering the mosquito signal from unwanted signals will need to be studied. With this filtering it is also possible to determine a criterion for the system to activate and shoot. There won't always be a mosquito nearby.
- One could look at the part we compensated for errors by adding a value to the DDOA and try to find solutions for approximating the Distance Difference of Arrival without the errors still in the system. This way the whole system may not need to be calibrated again after moving a microphone.
- The ideal number of microphones and references could be determined. We chose to use five microphone of which one served as a reference, but it is not guaranteed that this is the optimal number. Do consider that too many microphones give a slow data sample acquisition per microphone on a single DAQ card and longer computing time since more data has to be processed but it does increase the accuracy per measurement. It is also possible to use more than one reference microphone which might increase accuracy.
- The positioning of the microphones can be optimized. We positioned the microphones in a way we expected to be good for measuring the TDOAs, but it is again not guaranteed that this is the optimal solution. To find the ideal positioning would require further study on the algorithm or trial and error.



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