A Microphone-array Acoustic Direction Detector

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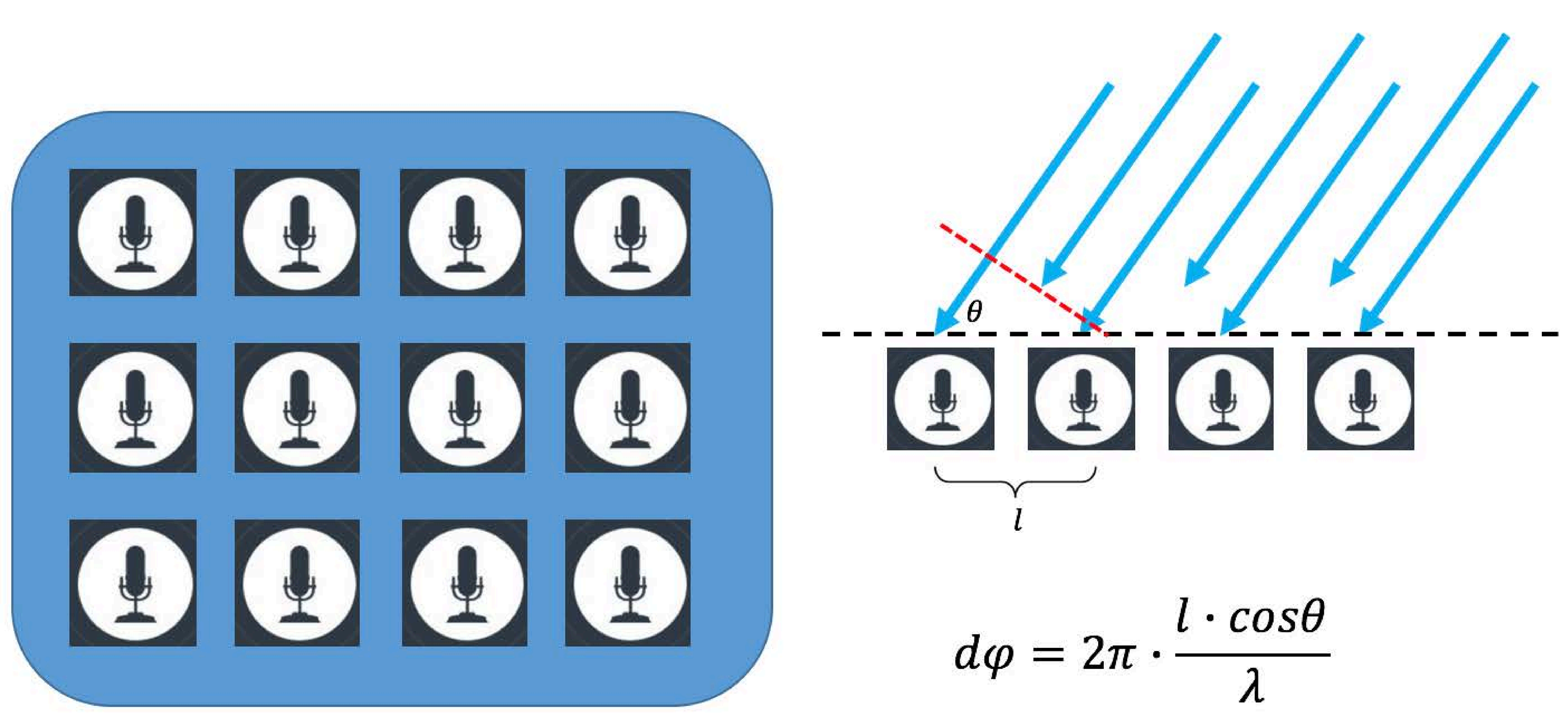
Abstract

In our project, we design an acoustic angle detector. Multiple microphones are incorporated in our device together with a STM32 SCM which is responsible for gathering the data and transmitting it to our computer. By this time, only harmonic acoustic wave is considered and FFT frequency analysis is utilized to derive the phase difference of the received signals by different microphones. Reasonable results are attained though still some systematic errors can not be wiped. Better devices providing higher sampling frequency may provide better results.

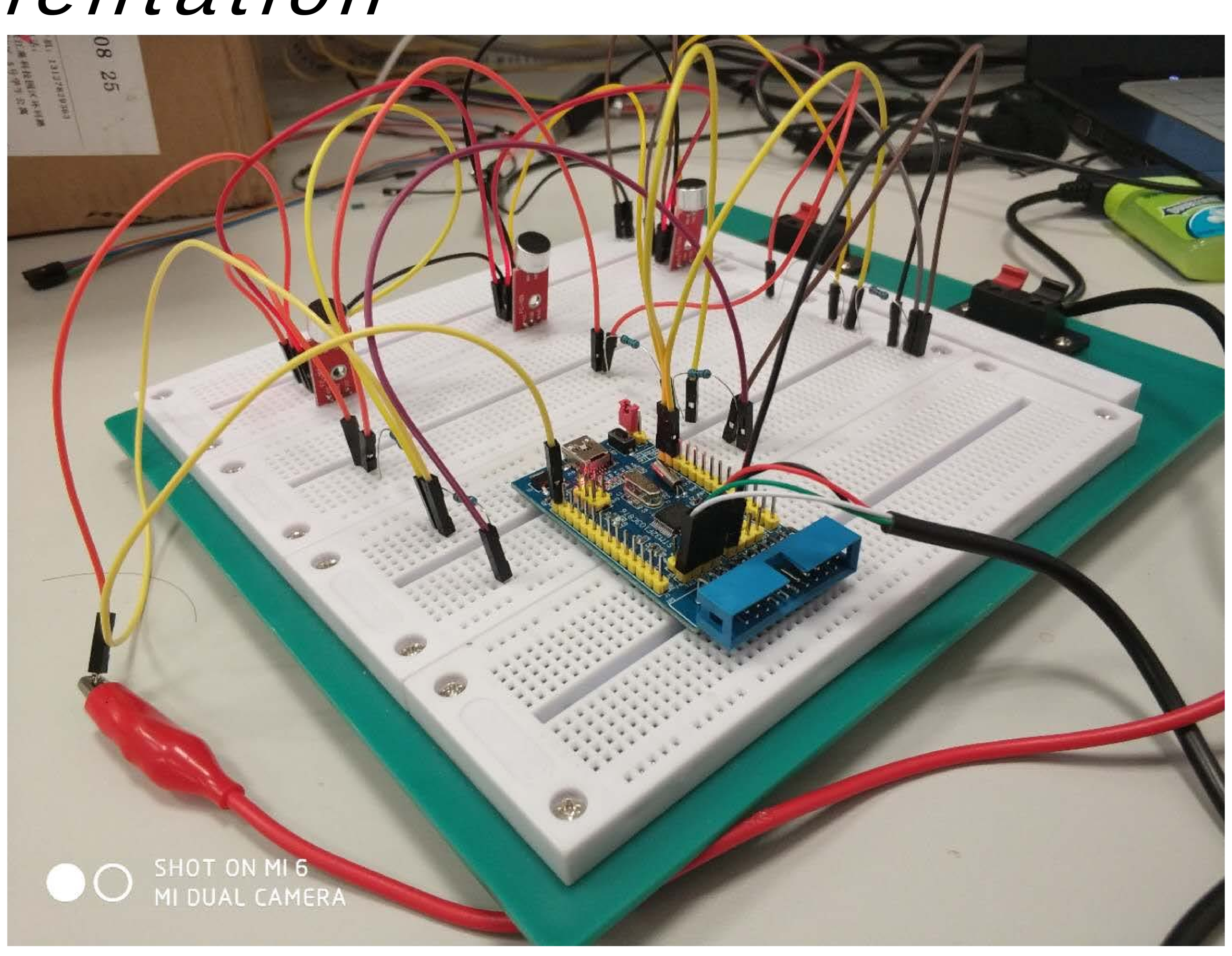
Microphones Array



$$d\varphi = 2\pi \cdot \frac{l \cdot cos\theta}{\lambda}$$

- A 2-D microphones array just as the upper figure shows is planned to be built to detect the all the incident directions in the whole half space the array faces. However, at last we only manage a 1-D microphones queue detecting only one incident angle with respect to the microphones queue dependent on how we put it.
- The idea of estimating the incident angle with the arriving phase difference at different microphones in the array is also shown in the upper figure on the right. Suppose the microphones are lined up with the uniform space being l, then the phase difference of the two neighbor microphones is $2\pi\frac{l \cdot cos\theta}{\lambda}$, with θ being the incident wave and λ the wavelength.

Implementation

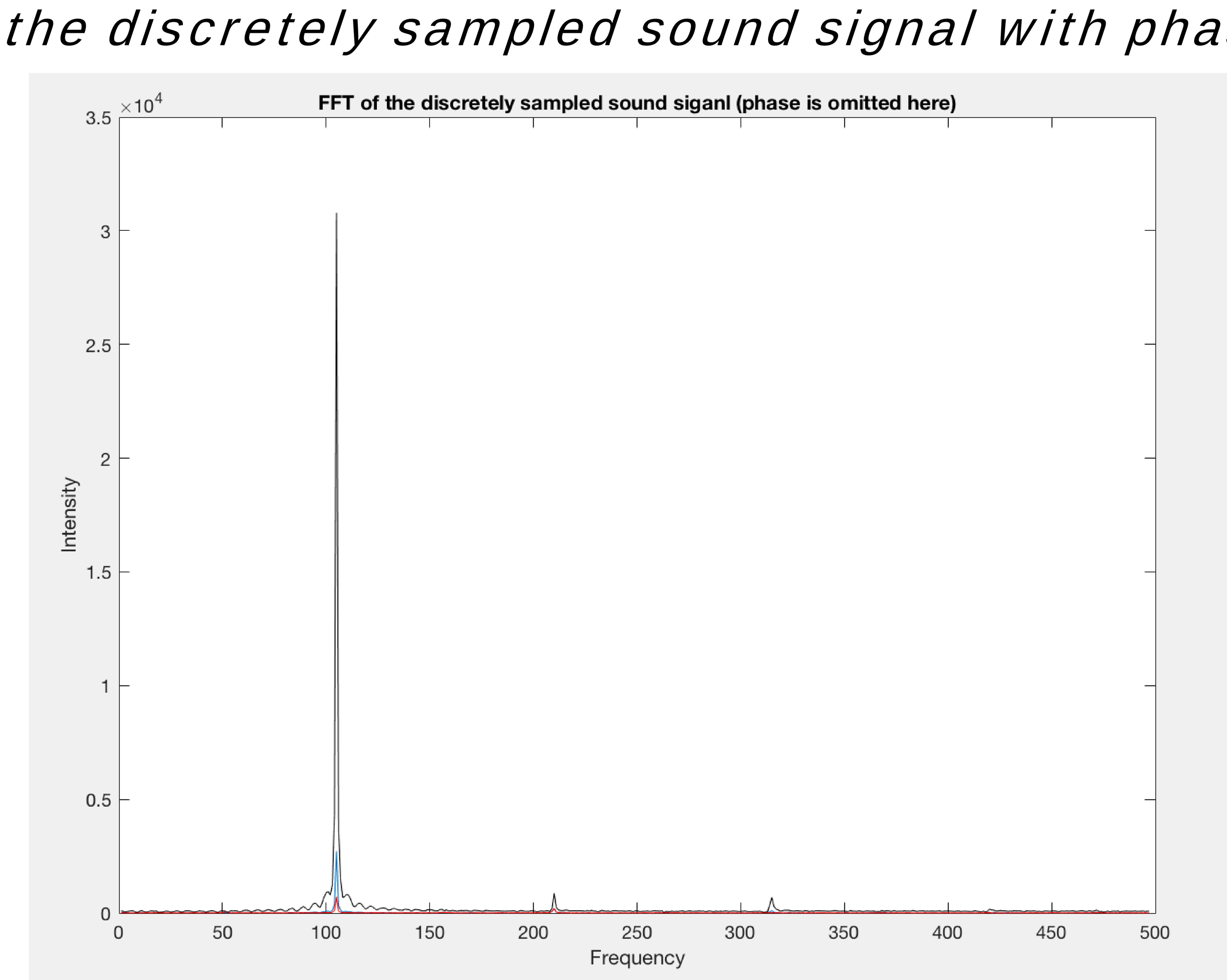


Acoustic Direction Detection and Microphones Array

In certain practical areas, more than the frequency and intensity, we may hope to know the incident direction of the coming acoustic signals. Similar to phased array radar, we try to build a microphones array to manage this direction estimation problem. Different arriving phases of the wave at different microphones can be detected and also the phase difference will vary with different incident direction. That's the basic idea of our device. Further more, FFT is helpful when inferring the phase of the acoustic signal when only discretely sampled signals are given.

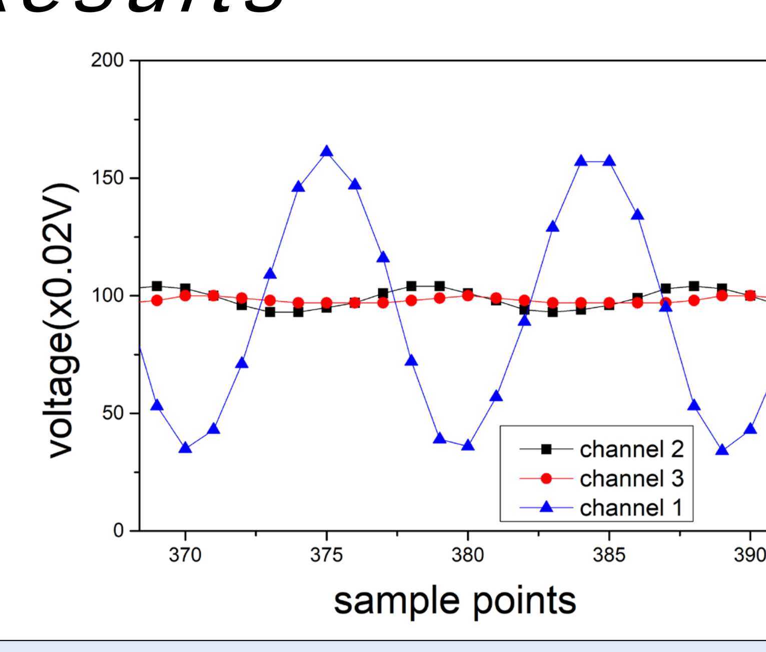
FFT(Fast Fourier Transform)

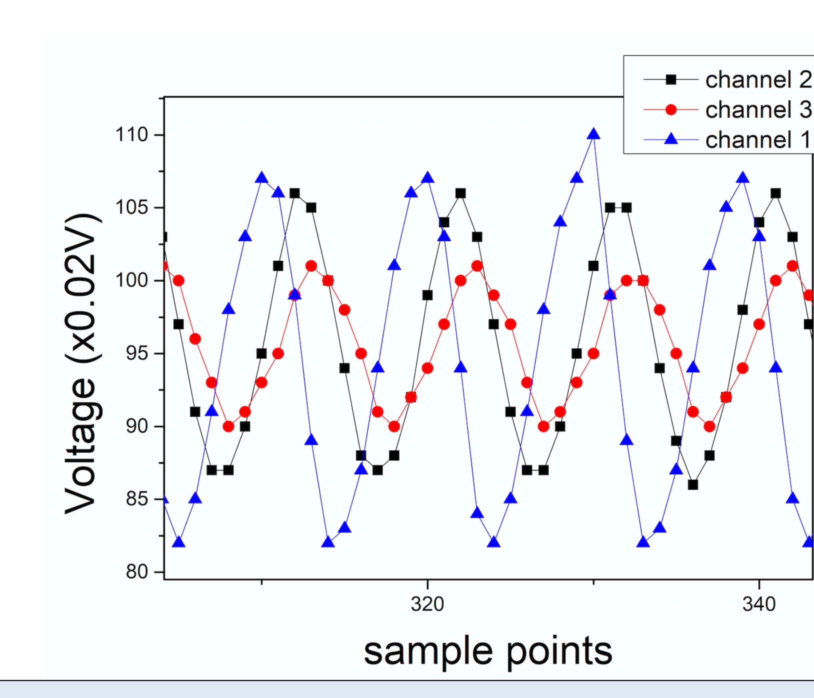
--FFT of the discretely sampled sound signal with phase omitted



- A **fast Fourier transform** (**FFT**) is an algorithm that samples a signal over a period of time (or space) and divides it into its frequency components. Both intensity and phase information of the frequency components is shown.
- In our project, the wave signal is discretely sampled over the time domain and hence FFT is quite fitting. Besides, the phases of the frequency components will be completely preserved without modification which saves us a lot work since the phase information if the sampled data what we actually are interested in.

Results





We evaluate the detecting performance of our device by testing it with the sound source put at different angles. With a 3-microphones queue, the calculated receiving phases' relative error is within 15% and the error of the estimated incident angle is within 0.35 rad.