SI100B SP Project

Requirements:

1. You need to both submit your project report and codes in **Gradescope** and check your program performance with TAs face-to-face.
2. About the report:
   (a) Submit your report in Gradescope. You have to submit only one report for each group.
   (b) Your report should be uploaded in the **PDF** format and the naming format of the file is specified as your_student_id+report.pdf
   (c) No handwritten version is accepted. LaTeX is recommended for preparing the report.
   (d) Prepare your report in English. Report in Chinese is not allowed.
3. About the codes:
   (a) Submit your codes in **Gradescope**.
   (b) Package all your codes into your_student_id+code.zip and upload. In the package, you are recommended to include a file named README.txt to clarify how to run your codes.
   (c) Make sure that your codes can run and are consistent with your report.
4. About face-to-face check: details are described in Module 3. Date is TBD.

Project: Direction of arrival (DoA) estimation

In this project, DOA estimation of sound sources needs to be simulated and experimented. The project is organized with 4 modules: the narrowband and broadband DOA estimation simulation, real-life DOA estimation experiment with a microphone array, and bonus part.

Module 1: Narrow-band DoA estimation

Consider the array processing scenario in a two dimension space which is now depicted in Fig.1. The sensors are located as a $J = 4$ elements uniform linear array (ULA) on the x-axis with a mutual spacing of $dx = 3.4$cm with the origin in the center of the array. The position of the i'th sensor is represented by the row vector $\underline{p}_i = [p_x, p_y]$.

A desired narrowband source signal with center frequency f_c (Hz) is assumed to arrive at the sensor array from the far field under an angle of θ_{s1}. A second interfering source s_2 also has a center frequency of f_c but arrives from another angle θ_{s2}. We know for the desired and interfering sources that $(|\theta_{s1}| < |\theta_{s2}| < \pi/2)$. The medium in which the sound waves are traveling is air which results in a speed of sound of $c = 340$m/s.

The observations are contaminated by spatially and temporally white Gaussian noise. We assume that the noise variances are the same and all signals are uncorrelated with respect to each other. The observations of a simulation of DOA estimation scenario are available in the file *Observations_nb.mat*. The sample rate is $10kHz$, which is also specified in the mat file.

One well known high resolution spectral estimation technique is the MUSIC algorithm. The MUSIC algorithm applies the steering vector for each angle of arrival to a projection matrix that is based on the noise subspace. The resulting pseudo spectrum contains very sharp peaks that represent the angles of arrival.

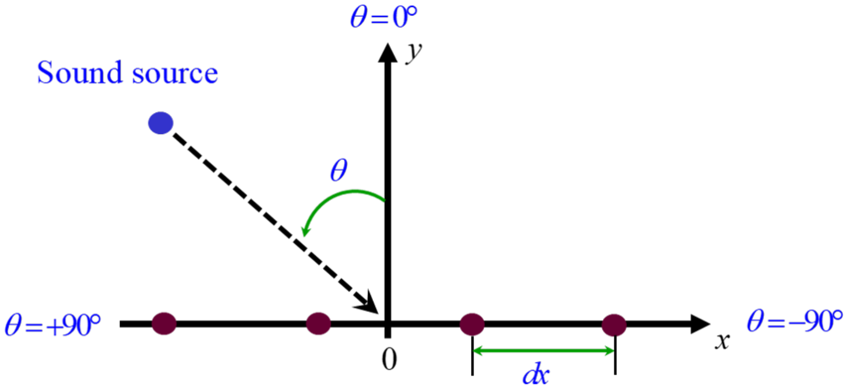

Figure 1: Array configuration

1) Plot the wave of time-domain and magnitude of frequency-domain. Find out the center frequency f_c from the ferquency-domain plot.

2) Complete the code *narrowband_Project.m* and apply the MUSIC algorithm to the observations. In your report, you should give the result of DOA estimation of the two sources and plot the pseudo spatial spectrum for the range of $-\frac{\pi}{2} < \theta < \frac{\pi}{2}$ with a resolution of at least 1°.

Module 2: Broadband DoA estimation

In this module, two broadband male and female speech sources under noisy environment impinge on four elements uniform linear array in different angles. The uniform linear array configuration is the same as module 1 except for the sensor spacing $dx = 2.5$cm. And the observations(simulation) are available in the file *Observation_wb.mat*. The sample rate is $16kHz$, which is also specified in the mat file. The audios recorded by TAs are available in the file folder *Array_output*. While the narrowband formulation of the MUSIC algorithm is straightforward to follow, it does not work well for nonstationary speech signal. The short-time Fourier transform (STFT) is a general way to processing non-stationary speech signals.
For both *Observation_wb.mat* and *Array_output*,

1) Listen to the sounds of the first sensor using *soundsc* command. Plot the wave in time domain with x-axis represented by t (s) and plot the magnitude of the frequency response and represent x-axis in *frequency* (Hz).

2) Perform short-time Fourier analysis on the observations. Divide the observations into different segments, and each segment is regarded as a stationary signal. Then do discrete Fourier analysis for each segment. Treat the same frequency band in all divisions as narrow-band and perform narrow-band MUSIC algorithm in each frequency band. Acquire the overall pseudo spatial spectrum, plot it, and give the DOA estimation result in the report.

Module 3: Microphone array experiment

1) Follow the given document *Audacity_user_manual.docx*, and download the corresponding software.

2) Record two audios in different directions using your microphone array (Fig. 2) and estimate these two directions using your algorithm. You also need to include your experiment result in the report, specifically, comparison of your direction setting of sources with DOA estimation result.

Module 4: Bonus

1) Build a graphical user interface (GUI) to display the direction of the sound sources. Fig. 3 gives an example of the interface.

2) Realize the function of real-time display of sound source direction (within a necessary but small time delay).

References

Direction of Arrival (DoA): https://en.wikipedia.org/wiki/Direction_of_arrival
Sensor array: https://en.wikipedia.org/wiki/Sensor_array
Array processing: https://en.wikipedia.org/wiki/Array_processing
MUSIC algorithm: https://en.wikipedia.org/wiki/MUSIC_(algorithm)
Short-time Fourier transform (STFT): https://en.wikipedia.org/wiki/Short-time_Fourier_transform

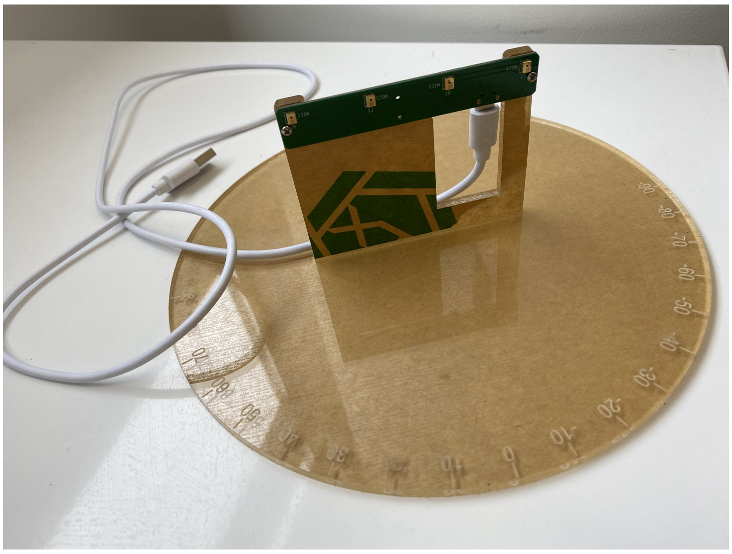

Figure 2: Microphone array.

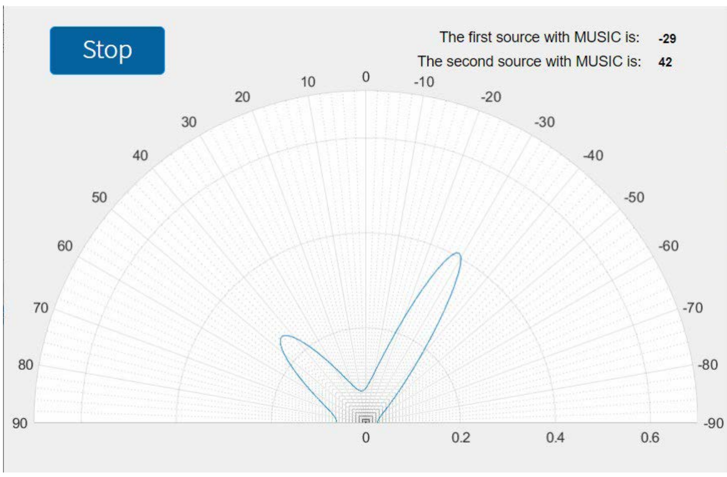

Figure 3: An example of GUI.