DOA Eastimation

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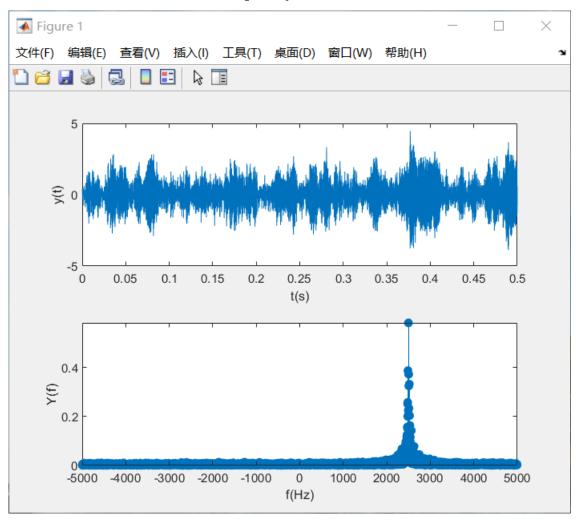
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1 Narrowband DOA Estimation

1.1 implementing procedure

At the beginning of task1 , we load the sound data and set the numbers of sources , sensors , sound velocity and sensor position. then we use matlab to find o ut t he c enter f requency a nd c alculate o ut t he s ignal s ubspace, noise subspace and the covariance matrix to implement MUSIC algorithm.

1.2 the wave of time-domain , magnitude of frequency-domain and the center frequency



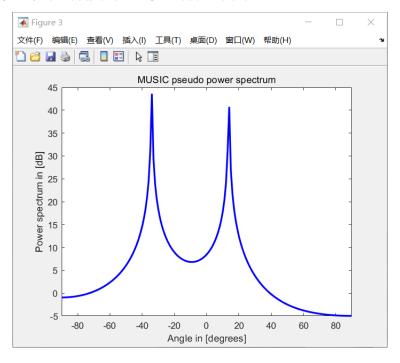
1.3 how to find center frequency

pair Y with frequency , and according to the largest Y to find its index , then the index is its center frequency.

1.4 why narrowband estimation can be implemented

There is only one center frequency in the frequency-domain picture which means there is only one frequency in the data, thus narrowband estimation is feasible.

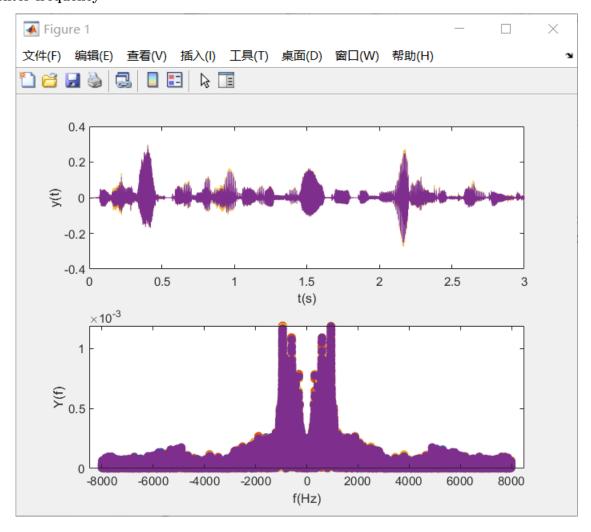
1.5 the result of DOA estimation



2 Wideband DOA Estimation

2.1 part one

the wave of time-domain , magnitude of frequency-domain and the center frequency



why narrowband estimation can't be implemented

The center frequency is not focused on a certain figure , its inaucrate and infeasible to use narrowboard estimation.

a feasible solution

The whole data can be splited into many tiny part thus there is only one center frequency in each part and the source of the sound can be seen as still. In this situation, short time fourier transformation is essential.

our spliting way and specific implementing procedure

The original matrix is 48000*3 thus we set the window as 800 and split the whole 48000*4 matrix into 160*300*4 matrix. For each 1*300*4, the each tiny part can be seen as just exists a center frequency and the source is still, thus narrow board estimation can be implemented. For each splited matrix we can use the formula

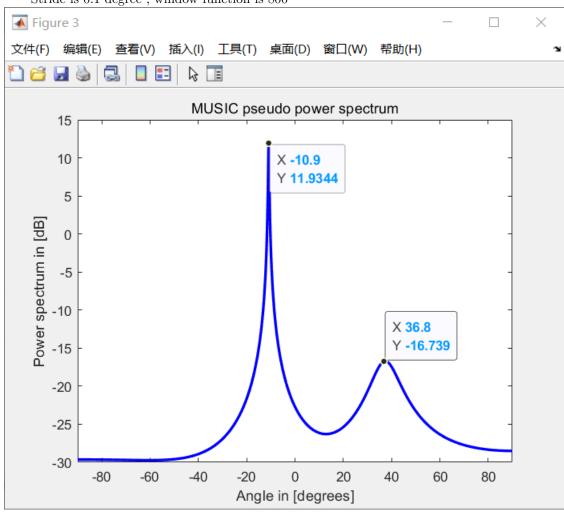
$$a^h(\theta)U_nU_n^ha(\theta) \tag{1}$$

Respectively, and the final result can be calculated by

$$P_{music}(\theta) = \frac{1}{\sum_{n=1}^{300} a^h(\theta) U_n U_n^h a(\theta)}$$
 (2)

the result of wide band estimation

Stride is 0.1 degree , window function is 800

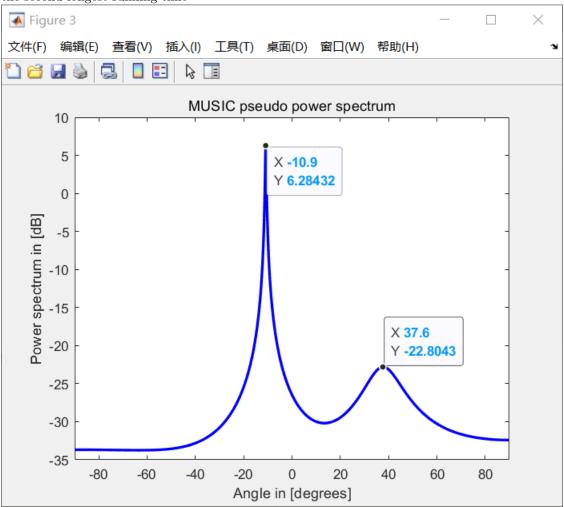


The result is -10.9 degree and 36.8 degree

different results under different width of window function and same stride of 0.1 degree

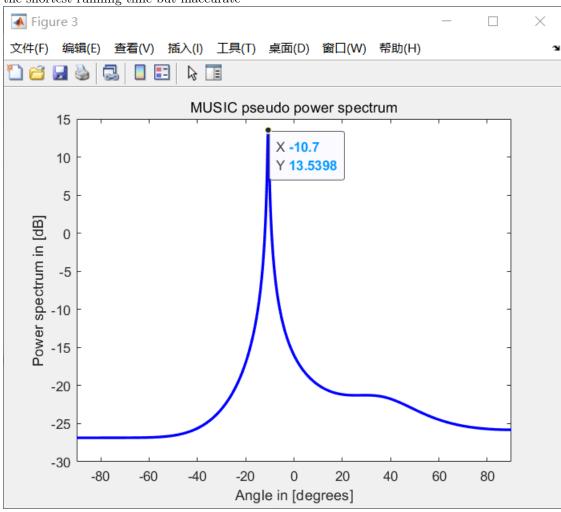
window function = 2000

the second longest running time



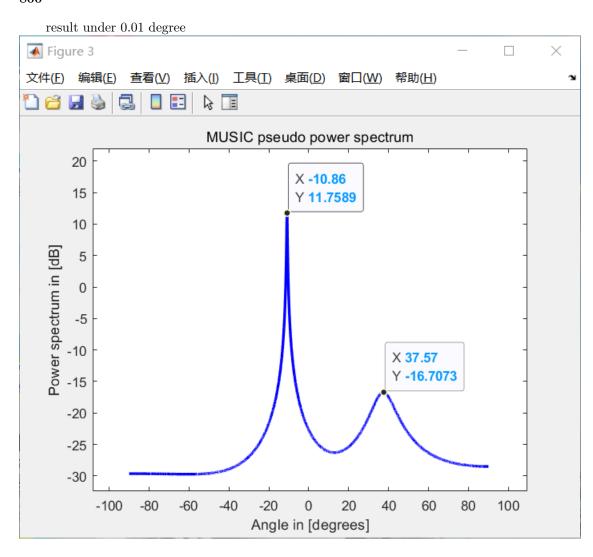
window function = 400

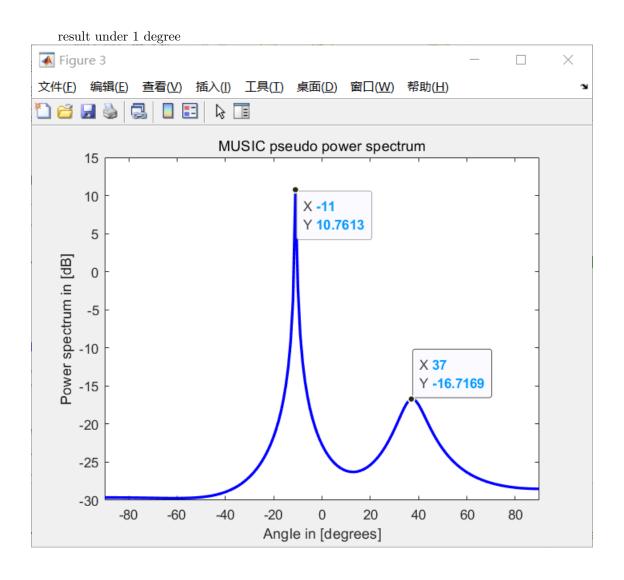
the shortest running time but inaccurate



the accuracy is controlled by the width of window function, and if the length is too small, some data will be lost, while if the length is too big, running time will increase obviously. through the experiment and comparsion, we regard that 800 is a suitable value.

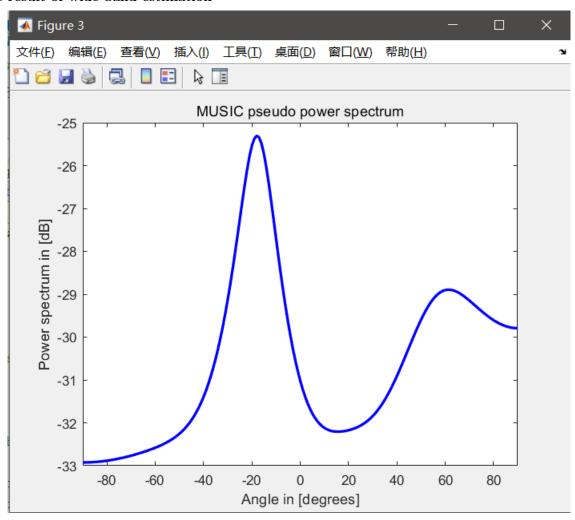
different results under different stride but same window function of 800





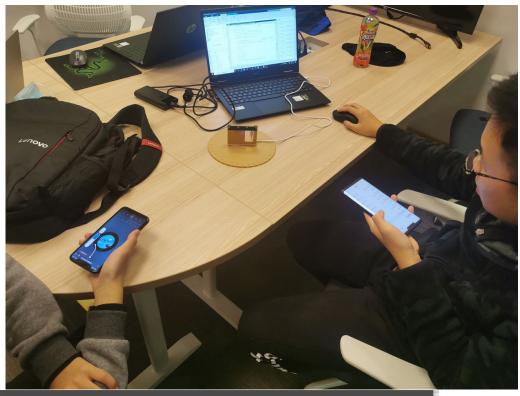
2.2 part two

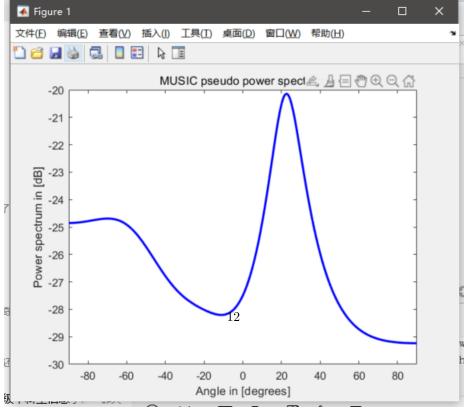
the result of wide band estimation



3 Microphone array experiment

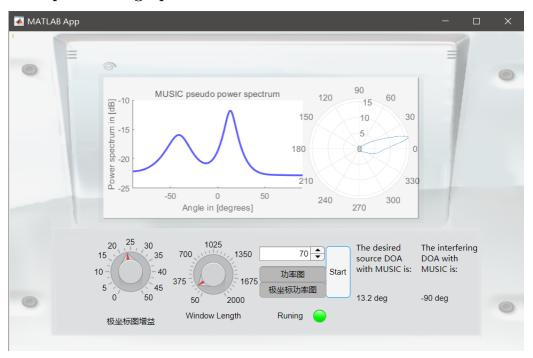
3.1 result of the experiment





4 Bonus

4.1 the picture of graphical user interface



4.2 result

shown in the video