DSP Homework 04

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September 26, 2022

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Abstract

In this article I write a summary about video we watched in class, it is about how starlink works, it classifies the difference between TV satellite and starlink satellite at beginning. Next it introduce the physical structure of inside dishy, and then it shows how a single antenna works. Then it introduces how a single antenna combined with others to form beam send to out space. Last it introduced the 64QAM antennas use to communicate with the satellite. To find the max and min frequency our ear can hear, my first idea is using signal generator to drive a loudspeaker, we change the signal frequency and judge if we can listen to it, but due to I and my laboratory teacher don't have same leisure time, I need to find another way. So I write a program to let our computer generate different frequency signal and use earphone to listen to it, how to find accurate frequency we can hear please go to **chapter 3.2**. In the third part, I proved that the Fourier Transform of discrete signal is zero from its definition and Lebesgue's integral theorem two ways. In the last I write a propose about my first project, which is a face detect and warn system.

1 How does starlink work?

1.1 Summary of the Video

This video introduce how starlink works, at the beginning of the video, it classifies the difference between TV satellite and starlink satellite, namely TV satellites can only receive signals from space while the starlink satellites can both receive and transmit signals. To help how the starlink works, it introduce the physical structure of inside dishy, and then it shows how a single antenna works. A single antenna in dishy is composed of 6 layers, but to make things simple, the video simplify it to two parts: feed line and antenna patch. For transmit signals, a high frequency signal is applied to the feed line, at the tip of line will accumulate positive or negative charges, this charges repel charges with same property. So that at the antenna patch, it induces charges of the opposite polarity. When polarity of signal applied to the feed line changes, the distribution of positive and negative charges on patch changes too. So that the electric field on the patch is changing, then changed electric fields produce changed magnetic fields, so that electromagnetic wave comes. As for receive signals, the chip connected to the antenna will change its mode, and the changed electric filed from the satellite will influence the electrons on the patch and then generate oscillating stream of electrons, which will send to the chip for procession. Then it introduces how a single antenna combined with others to form beam send to out space. The electromagnetic field produced by 1280 single antenna, will happen constructive and destructive interference. After add or subtract of each other, the electromagnetic field will be more focused. What interesting is that the power of antenna enlarge not 1280 times but 3500 times. To point at satellite moves 27000 km/h, it use phased array beam steering not a motor to achieve it. Different phase difference between different antenna will produce different direction of maximum radiation. With continuously change the phase difference we can scan different direction. The PCB with many chips inside the dishy also compute the position of satellite and how many degrees the phase difference should be to make sure dishy use its main lobe point at the satellite. To communicate with satellite, it use amplitude and phase modulation. 6 bits are arranged to different amplitude and phase, there are 64 kinds situation. It arranges these 64 kinds situation in a graph. We can draw a line from zero points to one point, the distance represents amplitude of modulated signal and the angel with positive horizontal axis represents phase of modulated signal, this technology is called 64 QAM.

1.2 Worries about radiation

After watching how starlink works, we know that it use electromagnetic to send information between antenna and satellite, but the electromagnetic field produced by antenna has side lobe and main lobe, it use main lobe to send information while the side lobe while produce radiation to the environment and human. Just a starlink needs 42,000 satellites, what about the number of our TV satellites, phone satellites, military satellites, commercial satellites$\cdots$ There are so many satellites on our heads, they send electromagnetic wave every seconds, what's more our cell phone signal base station also send electromagnetic wave every time. So man-made electromagnetic wave is everywhere, it do has some radiation to our body. Recent years many people suffer from all kind of cancel, but why this cancel is not exist in hundreds of years ago. The development of technology do bring convenient to our life, but it also bring harm to us. How can we do to avoid these radiation, I don' know also. Maybe we could invent a new clothing materials which can absorb these radiation and is convenient to wear. Good thought

2 A way to test the highest and lowest audio frequency our left and right ear can hear

2.1 Ideas to solve this problem

I think the key part to solve this problem is to produce an adjustable-frequency signal each time, we know that our signal generator can produce adjustable-frequency signal, such as sine wave, triangular wave and sawtooth wave. So the first way I had thought is that we use a signal generator to produce adjustable-frequency signal, and power it up(if it is unable to drive a loudspeaker), and then use it to drive a loudspeaker. Then we can rotate the frequency button as the Figure 1 shows to change the signal frequency. If we can hear the sound from the loudspeaker it means we can hear this frequency. As for testing left ear we can use some cotton or earplug to plug our right ear and plug our left ear while testing for right ear.

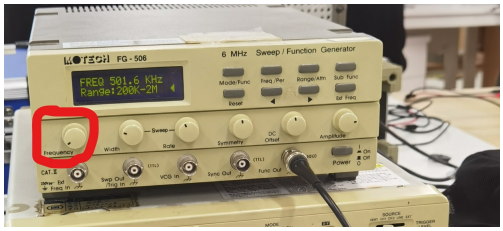


Figure 1: Signal generator

But what a pity is that both I and my laboratory teacher is busy this week, we don't have a same leisure time. To solve this problem before the deadline, I come up another idea. We usually use our computer to watch movies or listen to music, what makes me exciting is that our computer can play sound signal, so I write a program to let our computer play a certain frequency signal we want and using a circle to locate the frequency we can hear.

On the second way, I use numpy and sounddevice library to generate and play a certain frequency signal. I first use *numpy* to generate a 2 seconds sine wave, and use *sounddevice.paly()* to play it out. But to write the code I need to know a rough range of frequency our ear can hear. According to Wikipedia, our ear can hear about signal at 20Hz-20KHz. In my code, I set the frequency at 30000Hz at beginning when test for the max frequency and 10Hz at beginning for testing the min frequency, and we add or subtract the frequency each time until we can hear it.

But how to locate the accurate frequency is also a little difficult. When frequency of signal is closer to the boundary between we can hear and we can't hear, we need to make the change of frequency smaller and smaller. So I divide process of testing max frequency to 4 stage and process of min frequency to 2 stage. As the Figure 2 shows. On stage 0 we subtract f 2000Hz per time, on stage 1 we add f 100Hz per time, on stage we subtract f 10Hz per time and on stage 3 we add f 1Hz per time until we can't hear. On stage 4 we add f 5Hz per time, while on stage 5 we subtract f 1 Hz per time until we can't hear. Then the result of max or min frequency should be f - 1. When we test the left ear we just wear left earphone and only wear the right ear phone when testing the right ear.

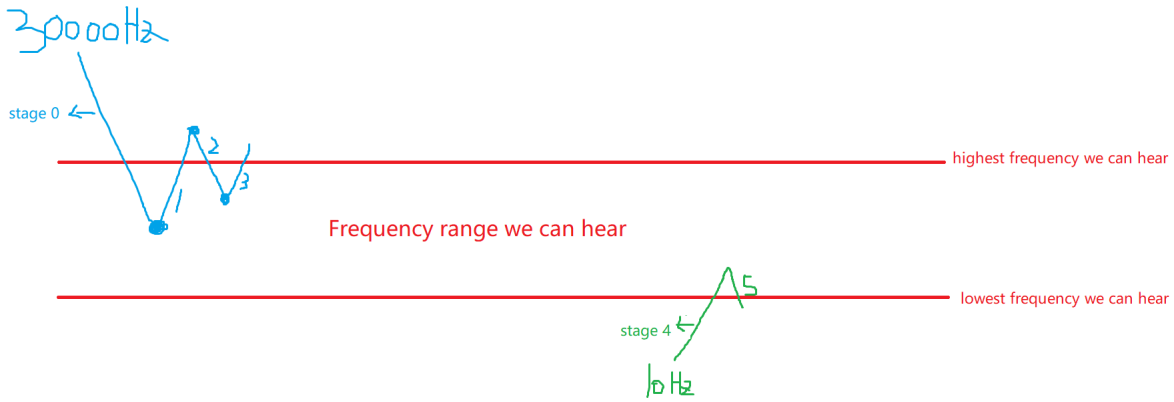


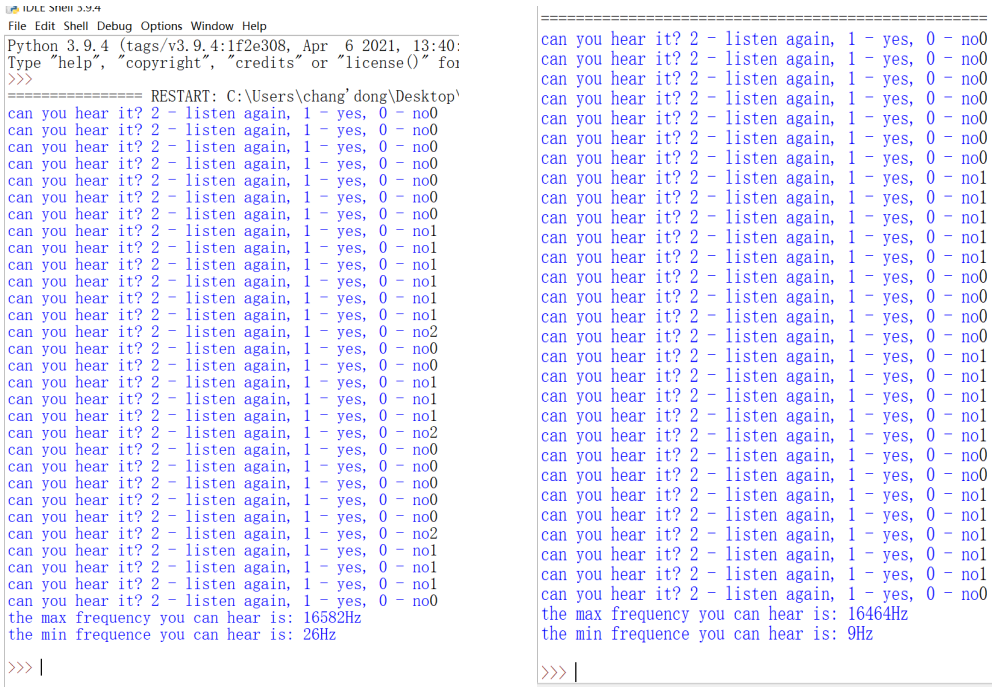
Figure 2: To locate the frequency

2.2 The highest and lowest frequency my left and right ear can hear

I run the program and test my left and right ear singly, the result shows as the Figure 3.

2.3 Analyze of result

From the result, I know that the range of frequency my two ears can hear is different, my right ear seems can hear lower frequency than my left ear. I do not do the test for several times and get average value, so there may be some coincidence, also the environment have noise may influence our test. What's more, when I listen to too much signal, my ear feels tired, so this may lead to a little error. All in all, the result is only for reference, it is not so accurate.



(a) left ear (b) right ear

Figure 3: Highest and lowest frequency my two ear can hear

3 Prove $\tilde{s}(f) = 0$

3.1 Prove from the definition of integral

We know that $x_s(t)$ may not be 0 only at t = nT, where n = $\cdots -1, 0, 1, \cdots$, and n is a integer, so

$$\begin{aligned}\tilde{s}(f) &= \int_{-\infty}^{\infty} x_s(t) e^{-j2\pi ft} dt \\ &= \sum_{n=-\infty}^{\infty} \int_{nT_-}^{nT_+} x(nT) e^{-j2\pi fnT} dt\end{aligned} \tag{1}$$

The integral is the area that our function covered with the axis of variable we integrate, so

$$\tilde{s}(f) = \sum_{n=-\infty}^{\infty} x(nT) e^{-j2\pi fnT} \Delta nT \tag{2}$$

While the value of all existed signal in our nature is finite, so $x(nT)e^{-j2\pi fnT}\Delta nT = 0$.
Thus $\sum_{n=-\infty}^{\infty} x(nT)e^{-j2\pi fnT}\Delta nT = 0$, so $\tilde{s}(f) = 0$.

3.2 Using Lebesgue's integral theorem

We let $E = nT, n = \cdots -1, 0, 1, 2, 3 \cdots$, so the measure of set E, $m(E) = 0$.

OK, but more basic proof is

$$\begin{aligned}\tilde{s}(f) &= \int_{-\infty}^{\infty} x_s(t) e^{-j2\pi ft} dt \\ &= \int_{E} x_s(t) e^{-j2\pi ft} dt + \int_{R-E} x_s(t) e^{-j2\pi ft} dt \\ \because x_s(t) &= 0, when \quad t \neq nT \\ \therefore \int_{R-E} x_s(t) e^{-j2\pi ft} dt &= 0 \\ \therefore \tilde{s}(f) &= \int_{E} x_s(t) e^{-j2\pi ft} dt\end{aligned} \tag{3}$$

because all signal that exist in our world have its boundary, so we let M be the least upper bound of x(t) and N be the highest lower bound of x(t), then $N <= x_s(t) <= M$, so:

$$\begin{aligned} &\int_E Ne^{-j2\pi ft}dt <= \tilde{s}(f) <= \int_E Me^{-j2\pi ft}dt = M * m(E) \\ &\int_E Ne^{-j2\pi ft}dt = N * m(E) = 0 \\ &\int_E Me^{-j2\pi ft}dt = M * m(E) = 0 \end{aligned} \tag{4}$$

so $\tilde{s}(f) = 0$.

4 Proposal of my first project

Good choice. Try out some face detection softwar

4.1 Why Is It?

In this summer holiday, I watched a lot of action movies, of which some of them are about the spy, such as *Mission: Impossible* series, *Bourne Ultimatum* series and *007* series. In this movie, the backstage command center can directly identify a specific person through the monitor on street. I think it is so cool, also it has a strong application value. But due to lack of all people's face picture, I can't do a system to detect all the people in the front of the camera, so I want to do a small face identification project about people we put them in our system.

4.2 What is it

In my project, I want to achieve that identify each people in the front of our camera dynamically among people in my detect system. If there is a people we don't know who he is, then our system will send a email to manager's email(in this project, send the email to my qq email). I have two program in my program, one is named *image_collect.py*, we can use this program to collect people's face picture and save them to our folder *data*. We first input the name of people we will collect and then it will run, after collect his or her information we push key 'n' if we need to collect other people's picture and we push key 'q' if we finish collecting pictures of people. The other program is named *face_detect.py*, in this program we can use camera on our computer to identify people and show his name on screen. Like the Figure 4 shows. The most interesting part is that in this program if a person we don't have their information in our data folder stay in front of our computer camera for too long, we will send a email which consisted of 3 pictures about this people to our manager(in this program, send it to my qq email). We can understand these two program like this: program 1 for manager to collect people's picture and after we have people's picture we can run program2 to detect people in front of our computer.

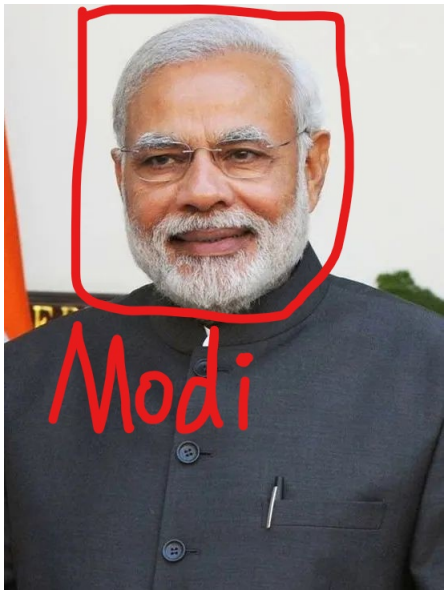


Figure 4: Effect of project

4.3 How to do it

For collecting people's picture, we can use *cv2.VideoCapture* to open our camera, and use the intern classifier of opencv to detect people' face, and after we detect it, we use*cv2.imwrite()* to save it. Too achieve colletc different people's face continuously,

we need to use a circle.

For face detect of program 2, we can first definite a function named *read_image* to read the picture we have saved, and return people's names, picture and corresponding train data we collected. Then we need to definite a function named *face_rec* to detect people's face and judge if this people is in our database, we put his name on screen, else we count time, if he stay too long, we send a email to my qq-email. In *face_rec*, we need to use *.train()* to generate train data, next we use our camera on laptop to see people in front of our computer.Then we find this people's facial feature to match in our train model to judge who he is. To achieve send email, we need too use smtplib library in python, I once meet it, but not so familiar with it so I think I still need to write a function named *send_email* to achieve sending email. Our qq email can open SMTP service so it maybe convenient for us to achieve send email.

This is a roughly ideas about my first project, to achieve I think it is much complex, and some details only can be find when we start do it.

5 Conclusions

(1) The Fourier Transform of discrete signal is zero.

(2) The frequency range our left can hear is different with our right ear's.

Appendix A Code listings

```
import numpy as np
import sounddevice as sd
import time
fs = 70000    #sampling rate more large frequency of sound signal

f = 30000     #it's said that the highest frequency our ear can hear is about 20000Hz so 30000 is enough
length = 2    #sound lasting time

df = 2000     #change value for f each time
f_max = 0     #highest frequency we can hear
f_min = 0     #lowest frequency we can hear

stage = 0
'''
stage reprensents the stage our test is on
on stage 0, 1, 2, 3 we are testing the highest frequency we can hear
    on stage 0 we subtract f 2000Hz per time
    on satge 1 we add f 100Hz per time
    on stage 2 we subtract f 10 Hz per time
    on stage 3 we add f 1Hz per time
on stage 4, 5 we are testing the lowest frequency we can hear
    on stage 4 we add f 5Hz per time
    on satge 5 we subtract f 1Hz per time
'''

while True:   # a circle to find the max and min frequency we can hear
    myarray = np.arange(fs * length) #produce nums in 0 to length * fs - 1
    myarray = np.sin(2 * np.pi * f / fs * myarray)#produce sound signal
    sd.play(myarray, fs) #play the sound
    time.sleep(2)    #wait for 2s
    a = eval(input('can you hear it? 2 - listen again, 1 - yes, 0 - no'))
    if a == 2:       #listen again
        continue
    elif a == 1:     #operation in judge part is corresponding with the stage
        if stage == 0:
            stage = 1
            df = 100
            f = f + df
        elif stage == 1:
            f = f + df
        elif stage == 2:
            df = 1
            f = f + df
```

```
            stage = 3
        elif stage == 3:
            f = f + df
        elif stage == 4:
            stage = 5
            df = 1
            f = f - df
        elif stage == 5:
            f = f - df

    else:
        if stage == 0:
            f = f - df
        elif stage == 1:
            df = 10
            f = f - df
            stage = 2
        elif stage == 2:
            f = f - df
        elif stage == 3:
            f_max = f - 1
            stage = 4
            f = 10
            df = 5
        elif stage == 4:
            f = f + df
        elif stage == 5:
            f_min = f - 1
            break

print(f'the max frequency you can hear is: {f_max}Hz\nthe min frequence you can hear is: {f_min}Hz\n') #print result
```