DSP Homework 04

wang,yaqi

10

September 24,2022

Contents

Abstract

There are four questions in this assignment. In question 1, I explained Dishy, a wireless transmission system with electromagnetic field as the main technology. After learning the video, I had a certain understanding of the technology, and after further searching and thinking, I had a deep learning and understanding of its operation principle. In Problem two, through the review of higher mathematics, I have a deeper understanding of the transformation rules of Fourier transform, and based on this, I proved the problem. In Problem three, it mainly solves two problems: generating corresponding frequency signals and how to test the left and right ears separately, so as to test the maximum and minimum values of sound frequencies that can be heard by the two ears. In question 4, I plan to design a noise decibel detection system based on 51 microcontroller through Proteus, which involves AD transformation and the design of 51 microcontroller and other problems to be solved.

1 Write a summary of this week's video(s) and your further thoughts on the content.

1.1 What do I get?

In this week's video, we mainly discuss Starlink and how it works. Starlink is extremely fast and its data transmission rate is up to hundreds of megabits per second. In order to achieve the purpose of information exchange, the dish antenna will move between different satellites every four seconds. The difference between the satellite receiving antenna of the TV and the Starlink information receiving device DISHY is that the receiving device of the TV can only receive data, but the receiving device of the Starlink can both receive data and send information. In contrast, the receiving devices of Starlink can aggregate more energy over a longer distance, at the cost of greatly reducing the coverage of the devices. The coverage of global signals can only be achieved by increasing the number of devices.

Dishy signal reception and transmission are realized through the transmission of high-frequency signals: during signal transmission, due to the voltage on the high-frequency feeder changes continuously from positive to negative and then to positive in the form of sine wave, the charge on the patch is induced to change continuously from negative to positive and then to negative, resulting in current oscillation. In this way information can be transmitted to the patch by high-frequency currents and sent out in the form of magnetic fields.

The signal reception principle is influenced by the electromagnetic field of the input signal, the electrons in the copper create an oscillating electron flow, and then send the information out in the form of an electric current.

In order to enhance the energy of information transmission, phase control and beamforming are used. The transmission of information is carried out in a constellation diagram consisting of six-bit binary amplitude and phase angle.

1.2 My further thoughts

In the study of video, I learned about the beamforming technology applied by Dishy to ensure the directionality of signal transmission, and further studied and thought about this technology.

Beamforming technology is developed to solve the problem of energy waste during signal transmission in a specific direction. The basic principle is that multi-wave interference technology is used. If the interference between multiple waves is in-phase interference, the interaction between waves will increase the amplitude, which is a constructive interference. However, if the interference between multiple waves is anti-phase interference, the interference between waves will reduce the amplitude, which is a destructive interference. To achieve this effect, the method of antenna array can be adopted,which is in figure 1. In order to verify this method, I used Matlab software to draw the power transmission images of a single antenna and the power transmission images of four antennas respectively for comparison, and obtained the results as shown in figure 2.

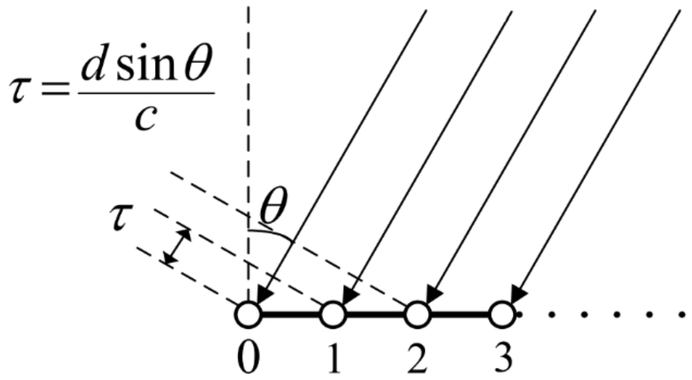

Figure 1: Schematic of the antenna array

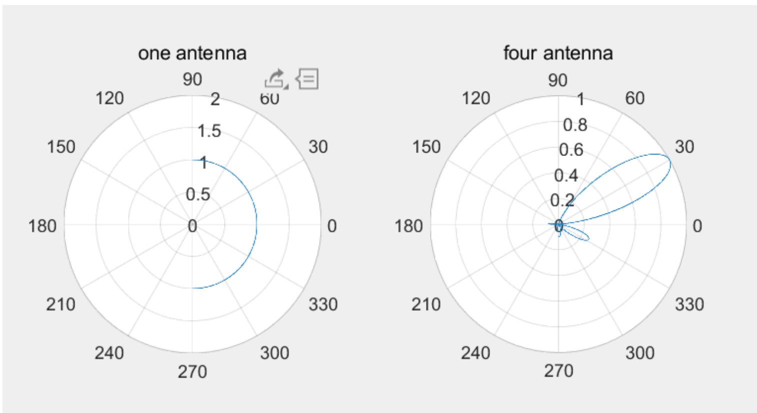

Figure 2: Energy concentration comparison chart

It is obvious from the picture that the energy generated by the superposition of multiple antenna systems is more concentrated in the direction of transmission and has a better effect.

2 Fourier transform proof

2.1 Problem Description

When studying the sampling process, we use the function

$$s(t) = \begin{cases} 1, if \quad t = nT \\ 0, otherwise \end{cases} \tag{1}$$

to express the sampling signal

$$
\begin{aligned}
x_s(t) &= x(t)s(t) \\
&= \begin{cases} x(nT) & ,if \qquad t = nT, \\ 0 & ,otherwise \end{cases}
\end{aligned} \tag{2}
$$

It turns out that such approach is not useful when using the Fourier transform because

$$
\widetilde{s}(f) = \widetilde{x}_s(f) = 0 \tag{3}
$$

Prove (3).

2.2 My answers

The problem is proved in three steps:

1. Consider only one term in the periodic function of s(t),

$$
s_0(t) = \begin{cases} 1,if \qquad t = 0 \\ 0,otherwise \end{cases}
$$

assume that $s_0(t)$ is the gate function which $\tau \to 0$ as shown in the figure 3, and its Fourier transform is $\widetilde{s_0}(f) = E\tau Sa(\frac{wr}{2})$. When $\tau \to 0$, $Sa(w\frac{wr}{2}) = \frac{sin(\frac{wr}{2})}{\frac{wr}{2}} \to 1$,given that E = 1, so $\widetilde{s_0}(f) = 0$ can be obtained.

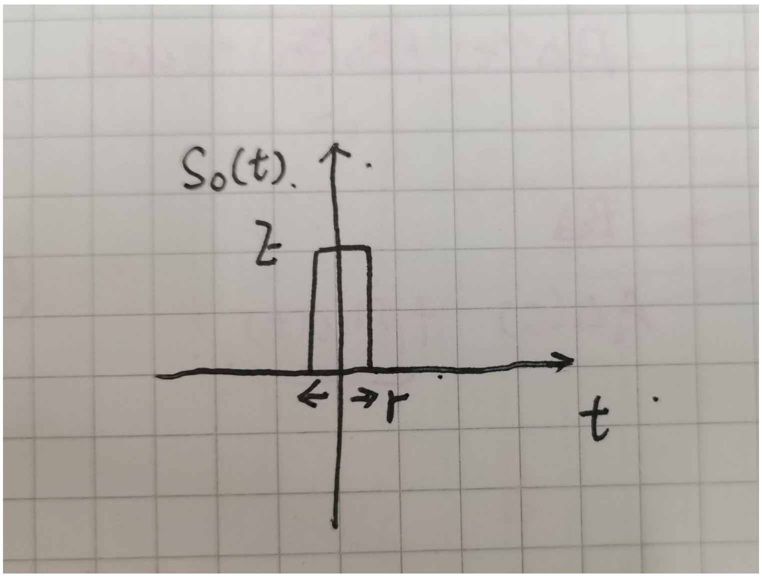

Figure 3: gate function

2. By time-shift characteristic and linear superposition characteristic,$\widetilde{s}(f) = \sum_{n=-\infty}^{\infty} \widetilde{s_0}(f)\delta(f - nf_0)$,which $f_0 = \frac{2\pi}{T}$ .so $\widetilde{s}(f) = 0$

3. According to the convolution characteristics of Fourier transform, $\mathcal{F}(x(t)s(t)) = \int_{-\infty}^{\infty} \widetilde{x}(f)\widetilde{s}(f - \tau)d\tau = 0$

So far, the proof is complete.

The specific process is as follows:

Consider one of the duplicates,take it a gate function like picture 3,which $\tau \to 0$.

$$
s_0(t) = \begin{cases} 1,if \qquad t \in [-\frac{\tau}{2}, \frac{\tau}{2}] \\ 0,otherwise \end{cases}
$$

$$
\mathcal{F}(s_0(t)) = E\tau Sa(\frac{wr}{2})
$$

if $\tau \to 0$,then

$$
\begin{aligned}
\mathcal{F}(s_0(t))_{\tau \to 0} &= 1 \times \tau \times lim_{\tau \to 0}(Sa(\frac{wr}{2})) \\
&= 0
\end{aligned}
$$

By time-shift characteristic and linear superposition characteristic,

$$\begin{aligned} s(t) &= \sum_{n=-\infty}^{\infty} s_0(t-nT) \\ &= \sum_{n=-\infty}^{\infty} s_0(t)e^{j2\pi n f_0}, f_0 = \frac{1}{T} \\ \mathcal{F}(s(t)) &= \sum_{n=-\infty}^{\infty} \widetilde{s_0}(f)\Delta(f-nf_0) \\ &= 0 \end{aligned}$$

According to the convolution characteristics of Fourier transform,

$$\begin{aligned} \mathcal{F}(x_s(t)) &= \mathcal{F}(x(t)s(t)) \\ &= \int_{-\infty}^{\infty} \widetilde{x}(f)\widetilde{s}(f-\tau)d\tau \\ &= 0 \end{aligned}$$

so $\widetilde{s}(f) = \widetilde{x}_s(f) = 0$

3 Design and carry out an experiment to find out the highest and lowest audio frequencies that your left and right ears can hear.

3.1 How do I design it?

The experimental design mainly considers 3 issues:

1. How to generate audio signals at specific frequencies.
2. How to ensure that the left and right ears are tested separately.
3. How to get to f_{max} and f_{min} fastest.

These 3 problems are designed as follows:

First of all, according to specific frequency of audio output problem, because of the need to get the exact numerical synthesis method, therefore, only by machine considering the matlab has generated in the audio signal sound() function, so by way of writing output sine function and sampling frequency is one of the leading f f audio is a controllable frequency signal to input.

Secondly, two methods are considered for the isolation test of the left and right ears. One is the use of soundproof earplugs, but when the amplitude of the signal is high, the earplugs can not achieve good sound insulation effect, so we want to reduce the sound amplitude and add soundproof earmuffs to improve it. Another method, based on the principle that vacuum cannot transmit sound, considered whether it could be blocked by putting a hollow thermos cup over the edge of the outer ear, but found that the existing materials did not match the size of the ear and could not block sound well. Therefore, method 1 is used for sound shielding.

Finally, considering that sound is a continuous signal, the human ear can only hear signals within a certain range of frequencies. If it cannot hear a signal of frequency f_{min}, it will not be able to receive a signal of frequency lower than that. Similarly, if it cannot receive a signal of frequency f_{max}, it will not be able to listen to a signal of frequency higher than that. So can be in the human ear can receive the frequency interval of 20-20000 on the basis of common sense, f_{min} at about 20Hz to find, f_{max} at about 20 000Hz to find. At this point, the design part of the experiment is completed.

3.2 How do I carry it out?

1. Use the device shown in Figure 4 to shield the hearing of the right ear(After wearing earplugs, fasten the earmuffs on your ears and press them tightly.).
2. Use Matlab software to generate a specific frequency of audio signals as shown in Figure 5, $f_{min}$from 10Hz upward gradually search, $f_{max}$from 21000Hz downward gradually search, until the sound can be heard.
3. Record the test results.
4. Mask the hearing of the left ear and measure as described above.

3.3 What's my answers?

The highest frequency the right ear can hear is 21000Hz, and the lowest frequency is 105Hz.
The highest frequency the left ear can hear is 21010Hz, and the lowest frequency is 100Hz.

Figure 4: tools we need

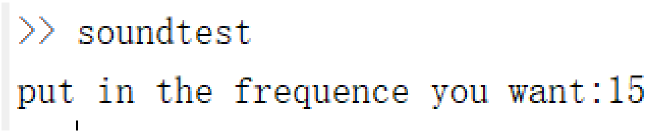

Figure 5: frequency set

4 Write a (improved) proposal for your first project.

4.1 Why I want to do it?

There are a lot of noise around us. Some noise will not have a great impact on us, but some noise has caused great harm to our quality of life and even our health. Although we are trying to avoid noise affecting the lives of others, but in daily life there will always be some have to hold a party at home, it is inevitable not to have an impact on the lives of neighbors and do not know it, inadvertently damage the harmonious relationship between neighbors. In addition, I have a certain degree of learning 51 microcontroller in the last semester, more interested in. Therefore, I want to design a noise detection system through 51 single-chip microcomputer, which can be placed at home to detect the noise. When the device detects that the noise is too large, it will light up the indicator light and alarm through the buzzer to realize the purpose of warning.

4.2 What functions does it have?

System block diagram as shown in figure 6, the first by a small microphone equipment acquisition environment after dealing with the signal into sound db information into the MCU devices, and things through the keyboard input of the block and stored in the Eeprom threshold comparison, determine whether to sound too loud, if judgment to noise LED lights up, the buzzer alarm, The data is displayed on the LCD display screen.

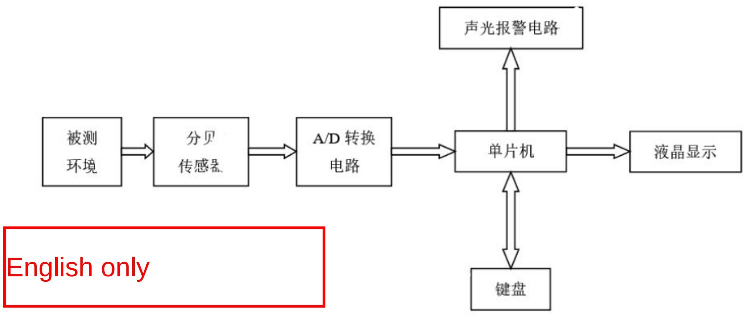

Figure 6: Basic structural block diagram

4.3 How I make it?

The preliminary idea of this design is to take 51 chip as the main part of the design, through the external equipment to collect the sound signal, the sound signal into electrical signal, the use of AD changes the incoming analog signal into digital signal. By writing the code, the 51 microcontroller can read the external input digital signal, and store the threshold information written by the keyboard into the Eeprom, so that when it reaches a certain value, the LED lights up and the buzzer alarms. The contents to be learned include:

1. How can the LCD display the data detected in the MCU?

2. The AD conversion part is completed by PCF8951 inside the MCU, or by external equipment to convert the signal into a digital signal directly into the incoming. The former needs to consider how to convert sound signals into continuous

electrical signals and transmit them into the system, while the latter arises the problem of which chip is used to convert external information into digital signals.

3. How to collect sound signals? Is it necessary to amplify them? If so, what kind of equipment should be used?
4. How to achieve maximum simulation of living environment through Protues.

This is the current thinking of the project, and the subsequent design will be gradually advanced by solving these problems.

Reference

[1]刘光毅,方敏,关皓,李云岗,孙程君. 5G移动通信系统[M].人民邮电出版社:5G丛书, 201611.287.

Appendix A Code Listings

```
%% beamforming
clc;clear;close all;
c=1500;f0=800;
d=c/f0/5;fs=40000;
N=2^10;M=1;
t=(0:N-1)/fs;
xt=cos(2*pi*f0*t);
theta=-90:0.1:90;theta0=30;
for i=1:M
    a(i,:)=exp(1j*2*pi*(i-1)*d*sind(theta));
    w(1,i)=exp(1j*2*pi*(i-1)*d*sind(theta0));
end
BP=conj(w)*a/M;
figure(1)
subplot(121)
polarplot(theta*pi/180,real(BP));title('one antenna');
M = 4;
for i=1:M
    a(i,:)=exp(1j*2*pi*(i-1)*d*sind(theta));
    w(1,i)=exp(1j*2*pi*(i-1)*d*sind(theta0));
end
BP=conj(w)*a/M;
subplot(122)
polarplot(theta*pi/180,real(BP));title('four antenna');

%%sound testing
f = input("put in the frequency you want:");
Fs = 44100; % frequency
T = 4; % time length
n = Fs*T; % sample
y = sin(2*pi*f*T*linspace(0,1,n+1));
sound(y,Fs);
```