DSP Homework 12

Xing, Yunpeng

November 21, 2022

Contents

Abstract

This week we watched two videos, The Beginner's Guide To Software Defined Radio RTL-SDR and IPv6 Addresses Explained. After watching the video, I feel deeply and think more deeply. In this article, I first obtained the audio of my voice through the matlab program, and then analyzed to get the range of my occurrence. At the end of the article, I obtained the reason why for a grayscale image spectral phase contains more image information than spectral amplitude through consulting the data and analysis.

1 Summary and my thoughts

1.1 The Beginner's Guide To Software Defined Radio RTL-SDR

This video tells us about the relevant knowledge of software defined radio. This video is for beginners, similar to popular science videos, relatively easy to understand. The beginning of the video introduces us to what software radio is. Software radio is a radio communication system that is implemented by software on a personal computer or embedded system. Traditional components are usually implemented on hardware, such as mixers, filters amplifiers, modulators, etc. We can use SDR to make broadcasts of different modes and types that we want, and we can do many cool things. Then the blogger showed us the architecture of SDR. The hardware part mainly needs three parts: SDR Receiver, Antenna(s), Computer Running SDR Software.

The video mostly takes us in on various models and kinds of equipment, giving us more insight into the specifics. However, due to the introduction of too many types, it is a bit complicated for beginners, and it is difficult to remember too much content.

1.2 IPv6 Addresses Explained

This video mainly introduces us to IPV6 addresses. In the beginning, IPV4 appeared first, and compared with IPV6, IPV4 provides fewer addresses. Later, due to the large-scale explosive growth of devices requiring IP addresses, IPV4 was not enough, so IPV6 came into being. Compared with IPV4, IPV6 not only increases the number of address bits, but also makes certain improvements to make it more efficient and usable.

IPV4 is 32 bits long and consists of four octet sections separated by dots, using binary, and each octet can theoretically contain any number between 0 and 255. The length of IPV6 is 128 bits, including 8 parts of 16-bit bytes, separated by colons, using hexadecimal. Next, the video briefly introduces the principle of subnet mask for us. The first part (first 48 bits) of IPV6 is called the global prefix, the next 16 bits are our subnet ID, and the remaining 64 bits are the host or interface ID. At the end of the video, we analyzed the possible reasons for some network errors.

1.3 My thoughts

The first video is mainly about radio transmission, which is related to the course "Communication Electronic Circuits" we offer this semester, so I have a better understanding of filters and amplifiers in analog circuits. It is relatively more convenient to implement radio reception and processing through software, and it is more convenient to realize various radio transmissions at different frequencies. My personal guess is that the software radio is based on digital circuits, and a large amount of data will be involved in the actual processing, which requires high processing capabilities such as computers. Combined with the knowledge about analog chips that we have watched before, the traditional radio transmission based on analog circuits still has great advantages in terms of computing power. In practical applications, we need to choose according to actual needs.

The second video is relatively clear about the structure of IPV6, and the actual implementation details still need to be explored. I was curious about the IP address assigned to me by the network I was connected to, so I looked it up. I found that both IPV4 and IPV6 will be allocated when the dormitory is connected to iLZU, but only IPV4 will be allocated when the teaching building is connected to LZU. However, LZU will also allocate IPV6 before, why not in the near future, you need to ask the network center. In addition, I also checked about why IPV4 and IPV6 will coexist, rather than IPV6 directly replacing IPV4. I understand that most Internet users are still using IPV4 at this stage. If you directly upgrade to IPV6, you need to modify all nodes on the Internet, which is difficult to implement, so use the form of IPV4/IPV6 transition.

Figure 1: LZU IP

Figure 2: iLZU IP

2 Use FFT to determine the frequency range of your voice

2.1 Problem description

Use FFT to determine the frequency range of your voice, i.e., the highest and lowest frequencies your vocal system can produce.

2.2 Get my voice range

This time I use matlab for programming, use the built-in audiorecorder() and recordblocking() to record my own monophonic audio, set the sampling frequency Fs = 10000, set the sampling time to t = 20s, the recording time is slightly longer for subsequent measurement The sound range is processed, and finally the recorded audio is saved as "xypvoice.mav", and the relevant time domain waveform is drawn at the same time. The waveform is shown in Figure 3.

```
Fs=10000;                       %Sampling frequency
t=20;                           %Sampling duration

recObj = audiorecorder(Fs,8,1);
disp('Start.')
recordblocking(recObj,t);
disp('End.');
myvoice = getaudiodata(recObj);

% Plot the time-domain waveform of recording data
t = length(myvoice)/Fs;
sample_points=[1/Fs:1/Fs:t];   %Calculate the abscissa of the sampling point
plot(sample_points,myvoice);
```

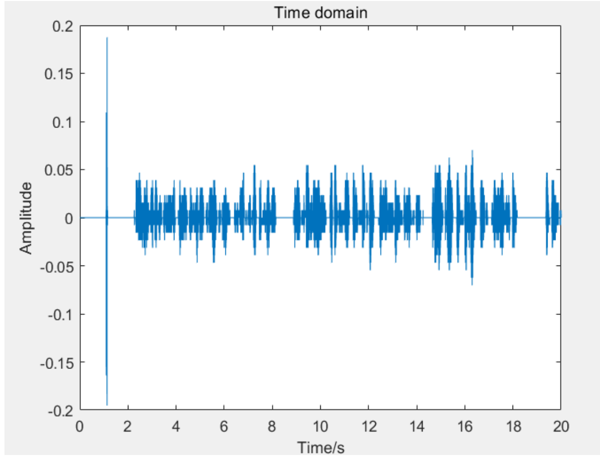


Figure 3: Original Time Domain Waveform

```
title('Time domain')
xlabel('Time/s');
ylabel('Amplitude');

% Store audio
filename = 'xypvoice.wav';
audiowrite(filename,myRecording,Fs);
```

The front is about the audio acquisition part. Next, we need to process the acquired audio, use interception to perform FFT processing on every two sides, and then output the time-domain waveform and frequency-domain waveform to find the lowest frequency and the highest frequency. Considering the impact of environmental noise, the lowest frequency point and the highest frequency point are selected as peak points, and the points with too small amplitude are discarded to eliminate the influence as much as possible. At the same time, since the human frequency range is about 50Hz to 2000Hz, the selection of points should be limited according to the actual situation. For ease of observation, the spectrum is shifted using the fftshift() function.

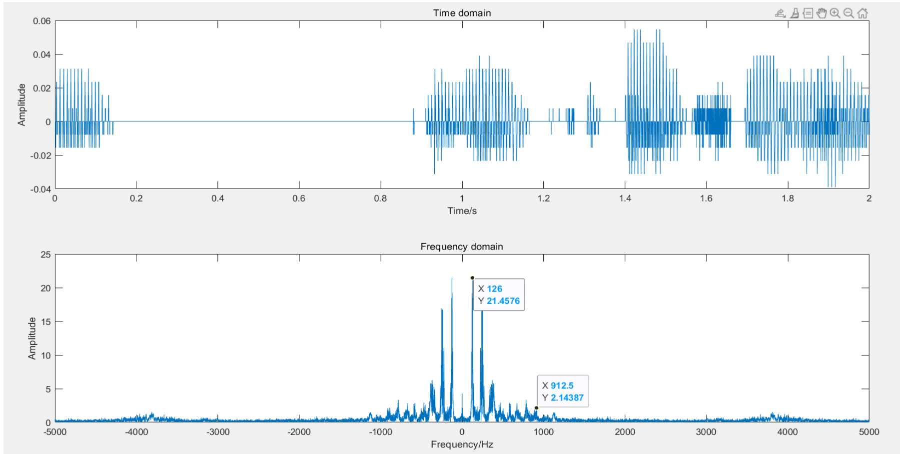
```
%Read raw audio
[origin_signal0,fs] = audioread('xypvoice.wav');
%Intercept 2S audio
start_time = 4;
end_time = 6;
new_signal=origin_signal0((fs*start_time+1):fs*end_time,1);
audiowrite('sound_new.wav',new_signal,fs);

t = length(new_signal)/fs;
sample_points=[1/fs:1/fs:t];
freq_signal=fft(new_signal);
freq_signal=fftshift(freq_signal);

%Plot time-domain waveforms
subplot(211);
plot(sample_points,new_signal);
title('Time domain')
xlabel('Time/s');
ylabel('Amplitude');

%Plot frequency domain waveforms
subplot(212);
plot([-fs:1:fs-1]/2,abs(freq_signal));
title('Frequency domain')
xlabel('Frequency/Hz');
ylabel('Amplitude');
```

By observing the time-domain waveform diagram in Figure 3, we found that there is almost no sound in 0-2S, so we no longer intercept the audio analysis. Table 1 below shows the low frequency f_L corresponding to the audio interception of 2-4S, 4-6S, 6-8S, 8-10S, 10-12S, 12-14S, 14-16S, 16-18S, 18-20S and high frequency f_H. We can find that the low frequency is relatively



Centered spectrum. G

Figure 4: 8-10s($f_H = 912.5Hz$)

Table 1: t-f_L-f_H

t / s	f_L/Hz	f_H/Hz
2-4	131	468
4-6	130	690.4
6-8	129	507.5
8-10	126	912.5
10-12	124	450
12-14	120.5	889
14-16	126	867.5
16-18	127.5	773.5
18-20	128.4	632.4

concentrated, so the average of the low frequency is taken as f_L; the high frequency is relatively scattered, and the average is meaningless, so the maximum value is chosen as f_H. Figures 4 show the time and frequency domains of 8-10s ($f_H = 912.5Hz$) respectively waveform diagram.After calculation, the low-frequency frequency of my voice is f_L=126.9Hz, and the high-frequency frequency is f_H=912.5Hz.

3 Explore the reasons for spectrum phase contains more image information

3.1 Problem description

In class we see that for a grayscale image, its spectrum phase contains more image information than its spectrum amplitude.Explore the reason(s).

3.2 The reasons

In class, we can see that after the phase spectra of the two given images are exchanged, the two images both present the structural outline of the other image, and the local light and shade remain almost unchanged. This shows that in the bispectral reconstruction, the phase spectrum plays a greater role. The specific reasons are as follows:

The phase spectrum includes all edge information and reflects the mutual position information of each frequency component of the signal. The magnitude spectrum of the image represents the brightness information of each pixel in the image, that is, what color the pixel should display, but the generated magnitude spectrum does not know which point each point is in the original image, that is, although the magnitude spectrum stores each pixel The amplitude information of the point, but the position of the original pixel point has been disrupted, so there is no way to reconstruct the original image only by the amplitude spectrum.

Edge information plays an extremely important role in image recognition and reconstruction. For example, when an artist depicts a person, only a few simple lines are needed to outline a person's image. To some extent, we can roughly think that the phase spectrum contains the texture structure information of the image, while the magnitude spectrum contains the contrast information of the image. To some extent, the structure of the image is the result of the combination of various texture information, edge structure information and gray distribution. Among them, we believe that the structure information has a

Many statements need proof.

more important position than the gray distribution, which determines the appearance and structure of the image [1]. Therefore, in bispectral reconstruction, the phase spectrum plays a greater role.

In the images we saw in class, we can also see that the contrast between light and dark is more obvious in the amplitude spectrum, while the phase spectrum is more mottled texture lines. After consulting the data, I also know that if a signal satisfies the minimum phase condition, then the amplitude spectral coefficient and phase spectral coefficient obtained by Fourier transform satisfy the Hilbert transform relationship, so it is possible to accurately restore the original information from the phase spectrum .

References

[1] Lv Yaoxin, Liu Zhiqiang, Zhu Xianghua.Research on Image Feature Extraction Based on Phase Spectrum Analysis Technology[J].Computer Application Research,2005(01):258-260.