

MATLAB 大作业 语音通信系统

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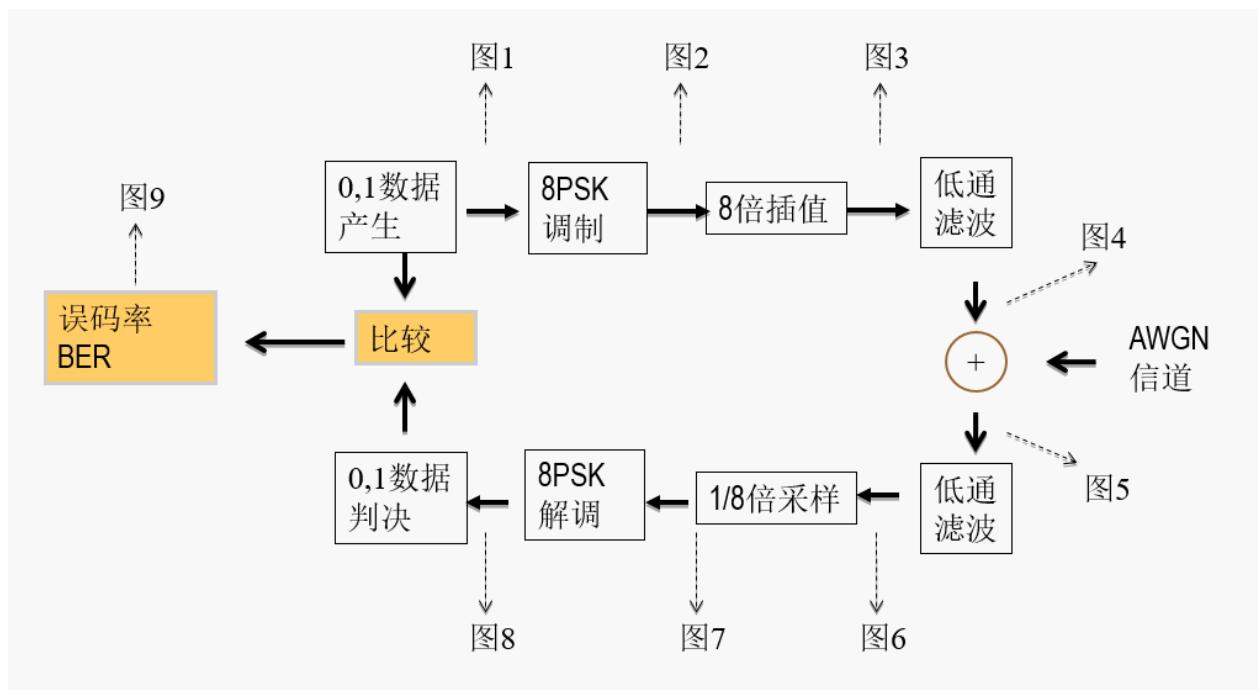
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§1 实验要求

1. 通信系统框图如下：

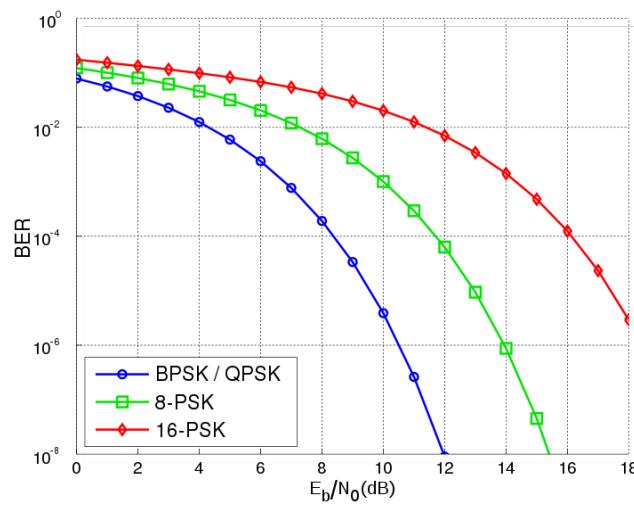


2. 给出原理表达式和原理框图；
3. 给出仿真程序；
4. 给出仿真结果并分析、讨论、比较；
5. 画出理论与仿真误码率曲线；
6. 将自己设计的滤波器同平方根升余弦低通滤波器比较，分析不同。

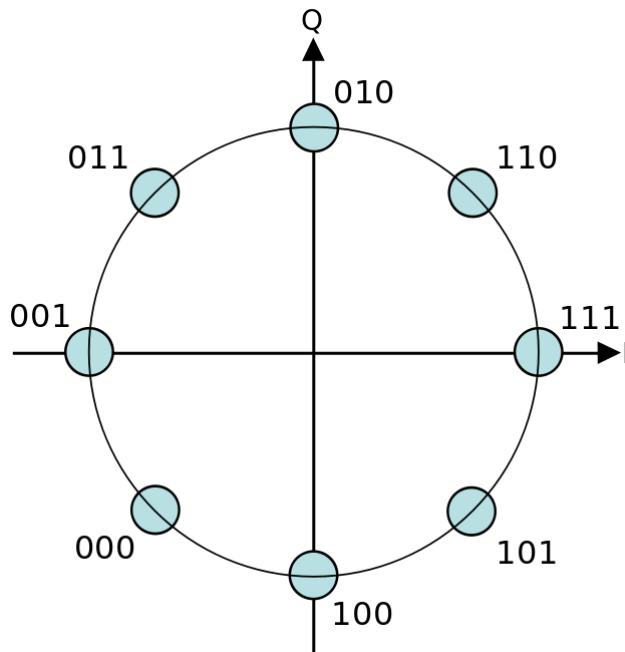
§2 实验原理

2.1 MPSK

1. MPSK - multiple phase shift keying 多进制数字相位调制，又称多相制，是二相制的推广。它是利用载波的多种不同相位状态来表征数字信息的调制方式。多进制数字相位调制也有绝对相位调制（MPSK）和相对相位调制（MDPSK）两种。



2. 8PSK (8 Phase Shift Keying 8 移相键控) 是一种相位调制算法。相位调制 (调相) 是频率调制 (调频) 的一种演变，载波的相位被调整用于把数字信息的比特编码到每一词相位改变 (相移)。因为 8PSK 拥有 8 种状态，所以 8PSK 每个符号 (symbol) 可以编码 3 个比特 (bits)。8PSK 抗链路恶化的能力 (抗噪能力) 不如 QPSK，但提供了更高的数据吞吐容量。



2.2 AWGN

```
awgn(sfilter, EbN0 - 10 * log10(T), 'measured');
```

$$snr(dB) = \frac{E_s}{N_0}(dB) - 10 * \lg(Nsamp) = \frac{E_b}{N_0}(dB) + 10 * \lg(k) - 10 * \lg(Nsamp)$$

Nsamp 为过采样率， k 为每符号比特数。

§3 代码及仿真结果

以自己设计的等波纹滤波器，在 SNR=10 处的结果为例：

3.1 0/1 数据产生

```

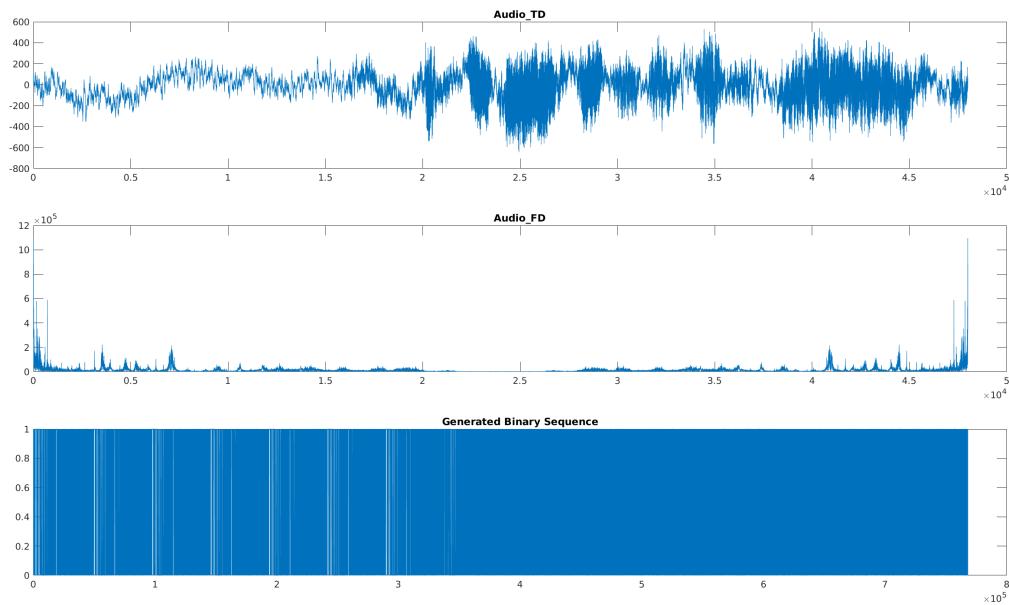
1 % Audio Input
2
3 clear
4 load /home/ykx/Audio_Communication_System/ber_standard.mat
5 M = 8;
6 K = log2(M);
7
8 % sample: datapoints of the audio
9 % fs: frequency of samplerate
10 % cnt_point: number of datapoints
11 % cnt_track: number of tunnels
12 basepath = '/home/ykx/Audio_Communication_System/';
13 wav = [basepath 'music.wav'];
14 [sample, fs] = audioread(wav, 'native');
15 info = audioinfo(wav);
16 [cnt_point, cnt_track] = size(sample);
17
18 % reshape to represent every symbol with 3 bits
19 sample = reshape(sample, [], 1);
20 bits = reshape(double(dectobin(sample, 16)) - 48, 3, []);
21 Ns = length(bits);
22 BER = [];
23 theory = [];

```

```

24
25 % display and save
26 figure(1)
27 subplot(3,1,1)
28 plot(sample)
29 title ('Audio\_TD');
30 subplot(3,1,2)
31 plot(abs(fft (sample)))
32 title ('Audio\_FD');
33 subplot(3,1,3)
34 plot(reshape(bits, 1, []))
35 title ('Generated Binary Sequence');
36 saveas(1, [basepath 'figure/audio.png']);

```

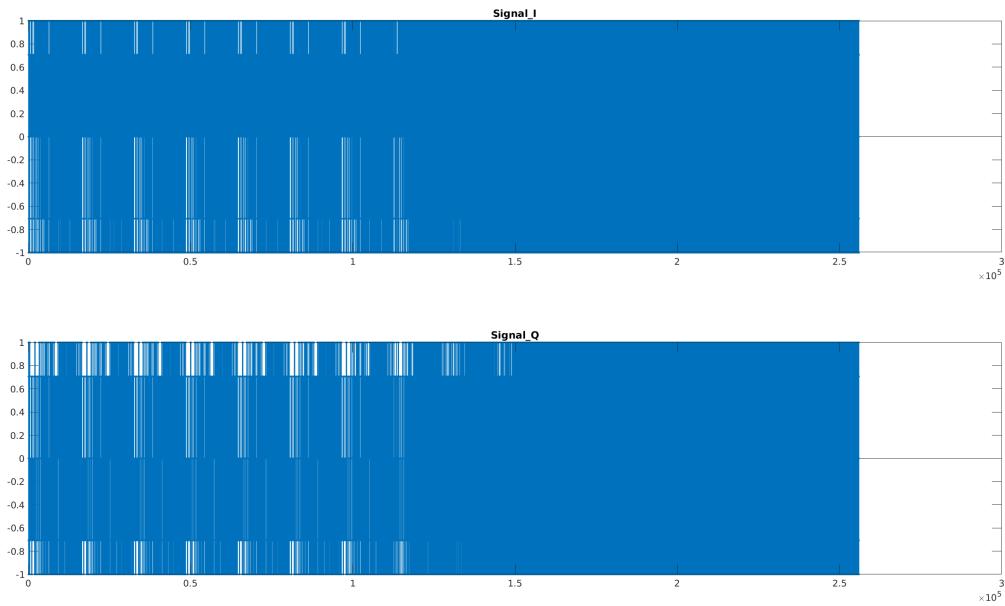


读取`music.wav`文件，受计算能力的限制，音频文件选取采样率为8000Hz，量化位数为16bits，时长为6s，总点数为48000，量化后比特数为768000，以3bits一组，构成 3×256000 数组。

上图显示的依次是音频的时域、频域图像，以及量化成比特序列后的图像。

3.2 8PSK 调制

```
1 % Generate 8PSK Signal
2
3 %8PSK symbols
4 sym_map=[1;(1+1i)/sqrt(2);1i;(-1+1i)/sqrt(2);-1;(-1-1i)/sqrt(2);-1i;(1-1i)/sqrt(2)];
5
6 s_mpsk = [];
7 s_I = [];
8 s_Q = [];
9
10 k = 4 * bits (1, :) + 2 * bits (2, :) + bits (3, :) + 1;
11 s_mpsk = [s_mpsk sym_map(k)];
12 s_I = [s_I real(sym_map(k))]; % the I branch of 8PSK
13 s_Q = [s_Q imag(sym_map(k))]; % the Q branch of 8PSK
14 s = [s_I s_Q];
15
16 % display and save
17 figure(2)
18 subplot(2,1,1)
19 stem(s_I, '.')
20 title ('Signal\_I')
21 subplot(2,1,2)
22 stem(s_Q, '.')
23 title ('Signal\_Q')
24 saveas(2, [basepath 'figure/8psk.png']);
```



3bits 一组做 8PSK 调制，映射成复数信号 s_{mpsk} ，分离成两路实信号 s_I , s_Q 。
上图分别显示了 s_I 和 s_Q 的时域波形。

3.3 8 倍插值

```

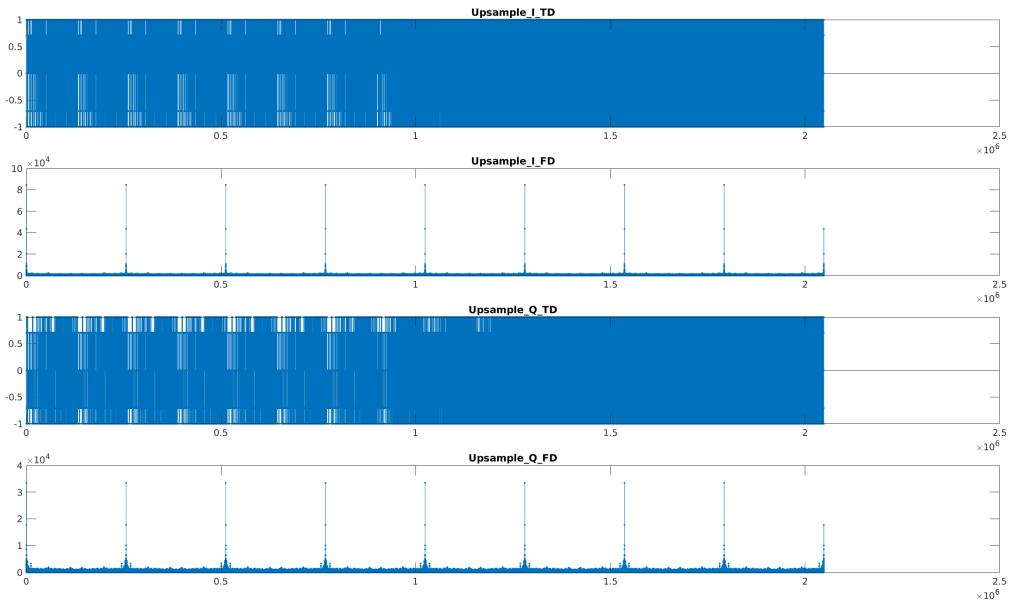
1 % Upsample
2
3 s_upsample = upsample(s, 8);
4
5 % display and save
6 figure(3)
7 subplot(4,1,1)
8 stem(s_upsample(:,1), '.');
9 title ('Upsample\_I\_TD')
10 subplot(4,1,2)
11 stem(abs(fft(s_upsample(:,1))), '.');
12 title ('Upsample\_I\_FD')
13 subplot(4,1,3)
14 stem(s_upsample(:,2), '.');
15 title ('Upsample\_Q\_TD')
16 subplot(4,1,4)

```

```

17 stem(abs(fft(s_upsample(:,2))), ' . ');
18 title ('Upsample\_Q\_FD')
19 saveas(3, [basepath 'figure/upsample.png']);

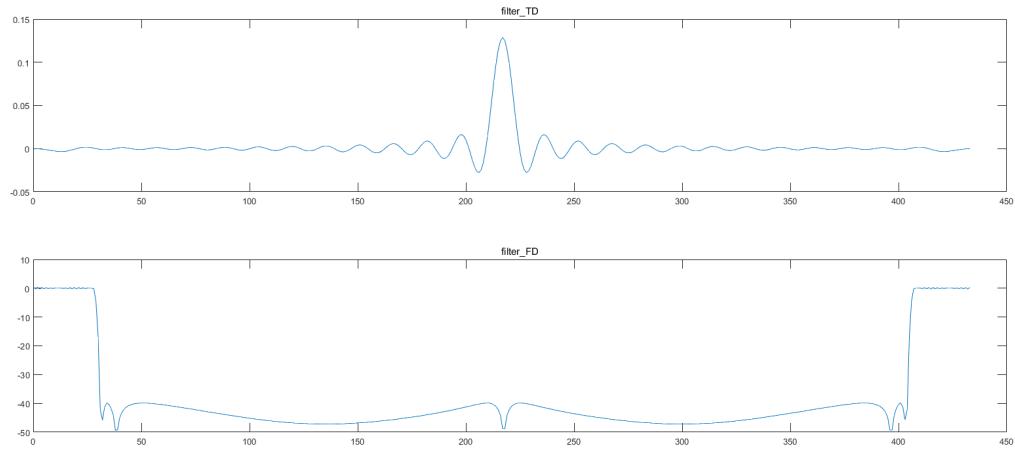
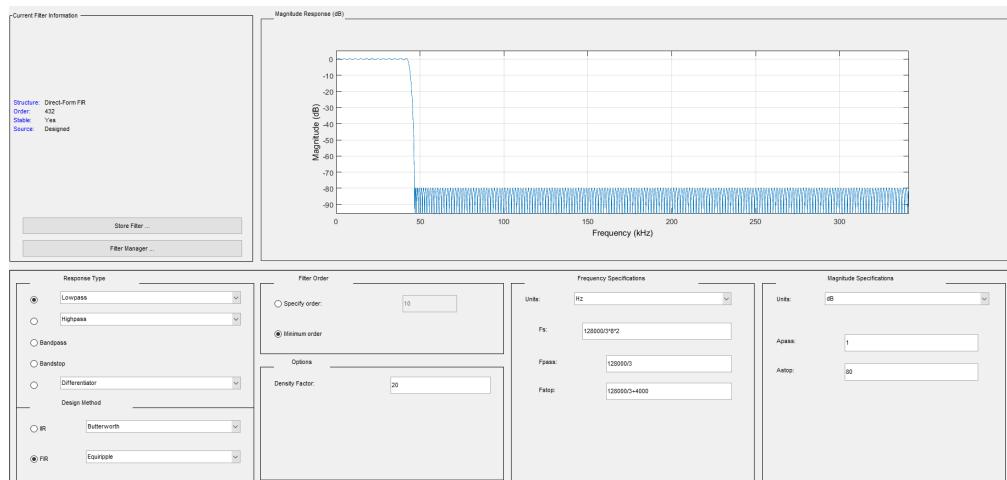
```



分别对两支路做 8 倍上采样得到时域与频域波形如上，可以看到数字频域在 0 到 2π 范围内以 $2\pi/8$ 的间隔重复 8 次。

3.4 成型滤波

首先设计了一个等波纹滤波器，参数设置上， F_s 选取为 2 倍信号频率，即 $2 * 128000 * 8/3 \approx 682666\text{Hz}$ ，通带频率为 $128000/3 \approx 42666\text{Hz}$ ，阻带频率为 $128000/3 + 4000 \approx 46666\text{Hz}$ ，通带和阻带波纹分别为 1dB 和 80dB。



接下来利用该滤波器实现成型滤波：

```

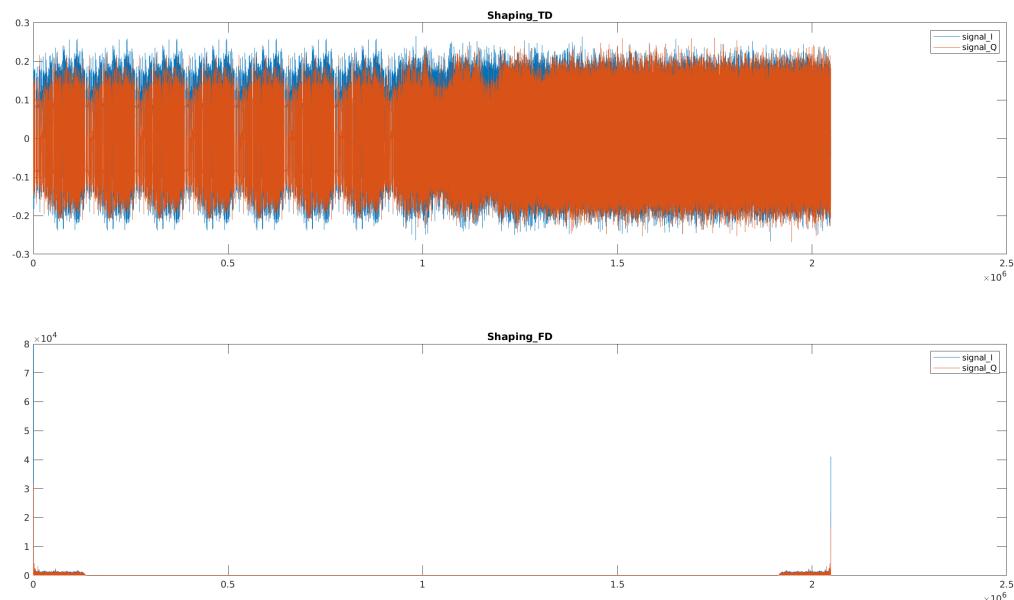
1 % Shaping Filter
2
3 hd = HD;
4 myfilter = hd.Numerator;
5 % add zeros after signals
6 s_transmit = filter (myfilter, 1, [s_upsample; zeros((length(myfilter) - 1) / 2, 2)]);
7 % cut off the prefix zeros
8 s_transmit = s_transmit(((length(myfilter)-1)/2+1): end, 1:2);
9
10 % display and save
11 figure(4)
12 subplot(2,1,1)

```

```

13 plot(s_transmit);
14 legend('signal\_I', 'signal\_Q');
15 title ('Shaping\_TD')
16 subplot(2,1,2)
17 plot(abs(fft (s_transmit)));
18 legend('signal\_I', 'signal\_Q');
19 title ('Shaping\_FD')
20 saveas(4, [basepath 'figure/shaping.png']);
21
22 % combine two branches
23 s_transmit = s_transmit (:,1) + 1i * s_transmit (:,2);

```



由于数字滤波器存在延时效应，所以在滤波前要先在序列末尾补零，滤波后舍去序列中的前导零。

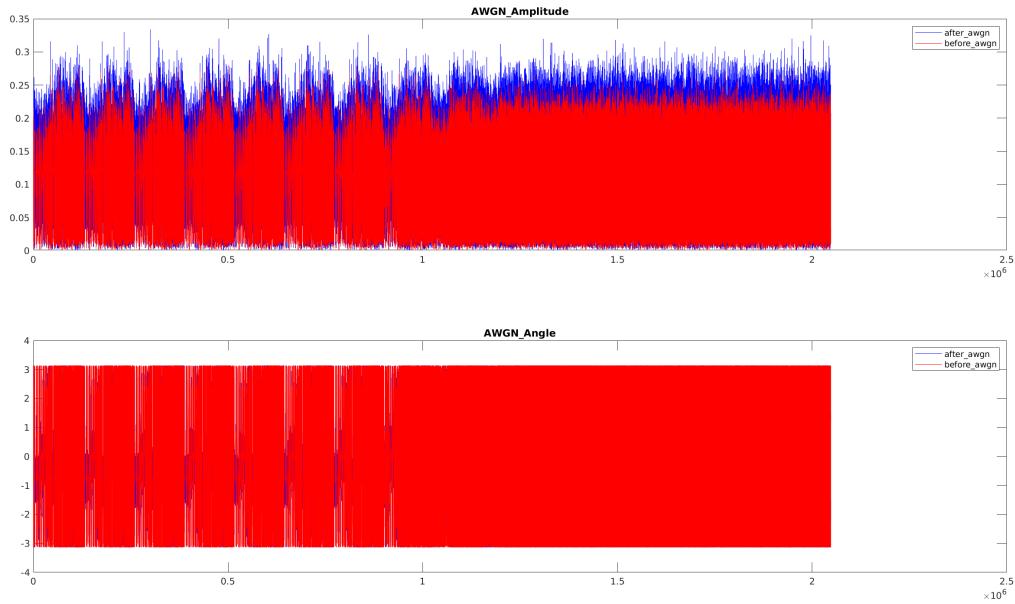
最后将两支路合并成复数信号，准备通过 AWGN 信道。

3.5 AWGN 信道

```
1 % AWGN Channel
```

```
2
```

```
3 Es_range = 10.^([-7] [8:1:22])/10); % Energy per symbol
4 for item = 1:16
5     Es = Es_range(item);
6     Eb = Es/K; % Energy per bit
7     N0 = 2;
8     SNR = 10 * log10((K * Eb/N0) / 8);
9     s_awgn = awgn(s_transmit, SNR, 'measured');
10
11    % display and save
12    figure(5)
13    subplot(2,1,1)
14    plot(abs(s_awgn), 'b');
15    hold on;
16    plot(abs(s_transmit), 'r');
17    legend('after\_awgn', 'before\_awgn');
18    title ('AWGN\_Amplitude');
19    hold off;
20    subplot(2,1,2)
21    plot(angle(s_awgn), 'b');
22    hold on;
23    plot(angle(s_transmit), 'r');
24    legend('after\_awgn', 'before\_awgn');
25    title ('AWGN\_Angle');
26    hold off;
27    saveas(5, fullfile (basepath, 'figure/', string(item), '/awgn.png'));
```



设计一系列的符号功率值，循环以便最终得到 BER-EbN0 图像。从幅值上可以看到通过 AWGN 信道后，有用信号上附加了一定的噪声信号。

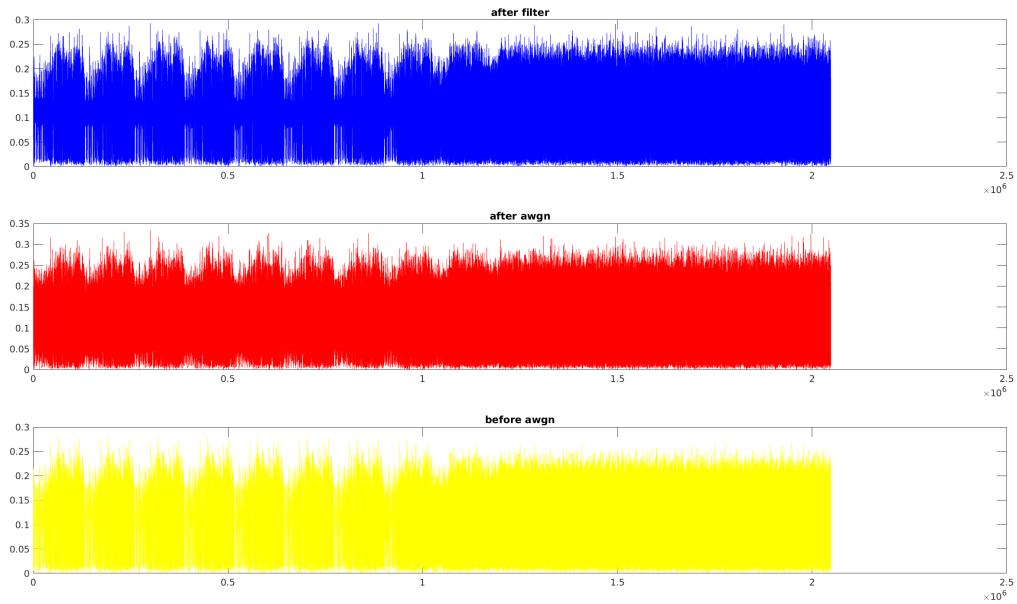
3.6 低通滤波

```

1 % Match Filter
2
3 s_receive = filter (myfilter, 1, [s_awgn; zeros((length(myfilter) - 1) / 2, 1)]);
4 s_receive = s_receive (((length(myfilter)-1)/2+1):end, 1);
5
6 % display and save
7 figure(6)
8 subplot(3,1,1)
9 plot(abs(s_receive), 'b');
10 title ('after filter');
11 subplot(3,1,2)
12 plot(abs(s_awgn), 'r');
13 title ('after awgn');
14 subplot(3,1,3)
15 plot(abs(s_transmit), 'y');
16 title ('before awgn');

```

```
17 saveas(6, fullfile (basepath, 'figure/', string(item), '/matchedfilter.png'));
```



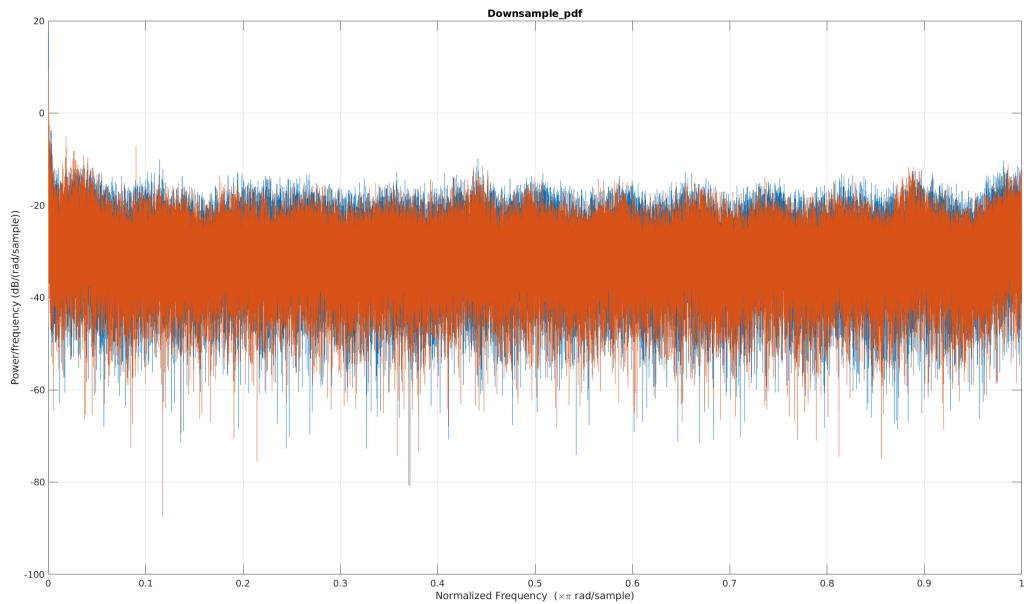
做一次低通滤波，由三条时域图像可以粗略看到，低通滤波后的波形更接近通过 AWGN 信道前的波形，部分噪声被滤除。

3.7 1/8 下采样

```

1 % Downsample
2
3 s_downsample = downsample(s_receive, 8);
4 % divide into two branches
5 s_downsample = [real(s_downsample) imag(s_downsample)];
6
7 %display and save
8 figure(7)
9 periodogram(s_downsample);
10 title ('Downsample\_pdf');
11 saveas(7, fullfile (basepath, 'figure/', string(item), '/downsample.png'));

```



做 $1/8$ 下采样，并将复数信号按照实部和虚部分离成两支路，上图为两支路的功率谱密度函数图像。

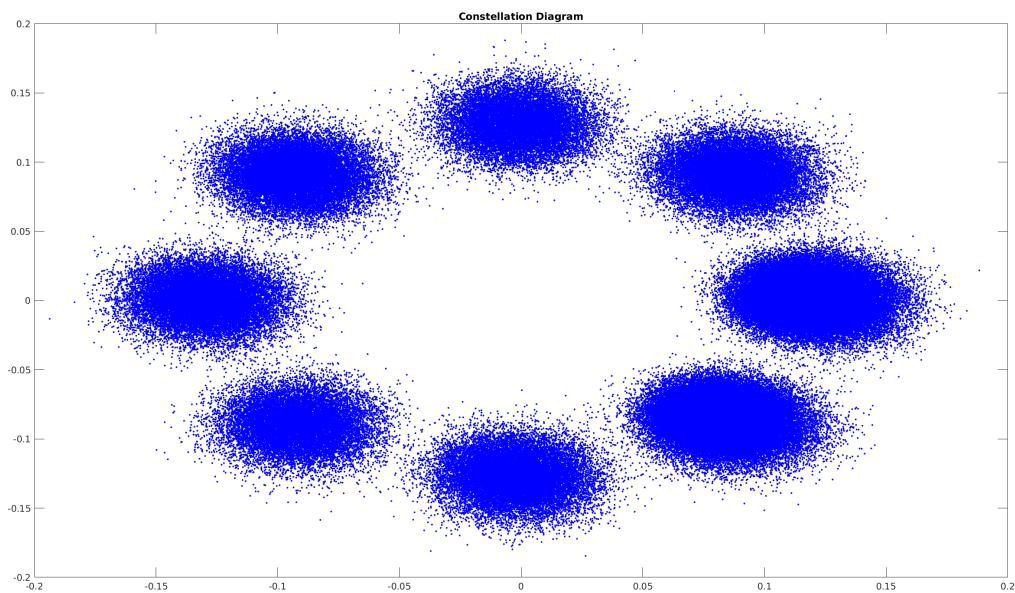
3.8 8PSK 解调及判决

```

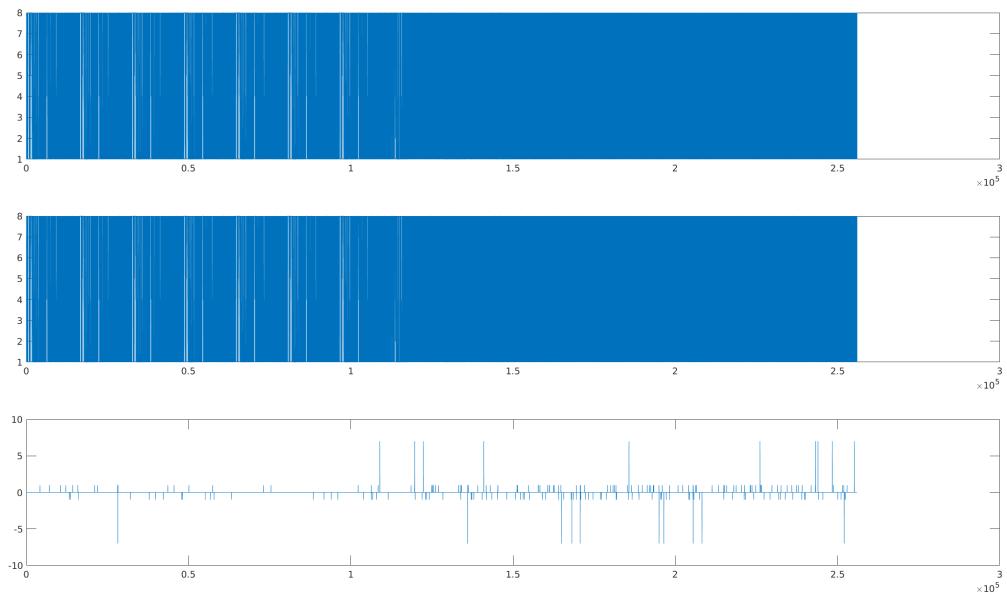
1 % 8PSK Judgement
2
3 s_demodulate = s_downsample;
4
5 % display and save
6 figure(8)
7 plot(s_demodulate(:,1), s_demodulate(:,2), 'b.')
8 title ('Constellation Diagram');
9 saveas(8, fullfile (basepath, 'figure/', string(item), '/constellation.png'));
10
11 % calculate the distance to 8 phase points
12 s_result = s_demodulate(:,1) - 1i * s_demodulate(:,2);
13 s_result = s_result';
14 distance = abs(repmat(s_result, [M, 1]) - repmat(sym_map, [1, Ns]));
15
16 % get the minimum distance and the order

```

```
17 [min_dis, min_pos] = min(uint32(distance .* 10000));  
18  
19 % display and save  
20 figure(9)  
21 subplot(3,1,1)  
22 plot(k)  
23 subplot(3,1,2)  
24 plot(min_pos)  
25 subplot(3,1,3)  
26 plot(k-min_pos)  
27 saveas(9, fullfile (basepath, 'figure/', string(item), '/error.png'));  
28  
29 % demodulate from symbols to bits sequence  
30 min_pos = min_pos - 1;  
31 bits_result = [];  
32 bits_result = [ bits_result sign(bitand(min_pos, 4))] ;  
33 bits_result = [ bits_result ; sign(bitand(min_pos, 2))];  
34 bits_result = [ bits_result ; mod(min_pos, 2)];  
35  
36 % display and save  
37 figure(11)  
38 subplot(2,1,1)  
39 plot(reshape(bits, 1, []), 'b');  
40 title ('transmitted\_bits');  
41 subplot(2,1,2)  
42 plot(reshape(bits_result, 1, []), 'r');  
43 title ('received\_bits');  
44 saveas(11, fullfile (basepath, 'figure/', string(item), '/bits.png'));
```



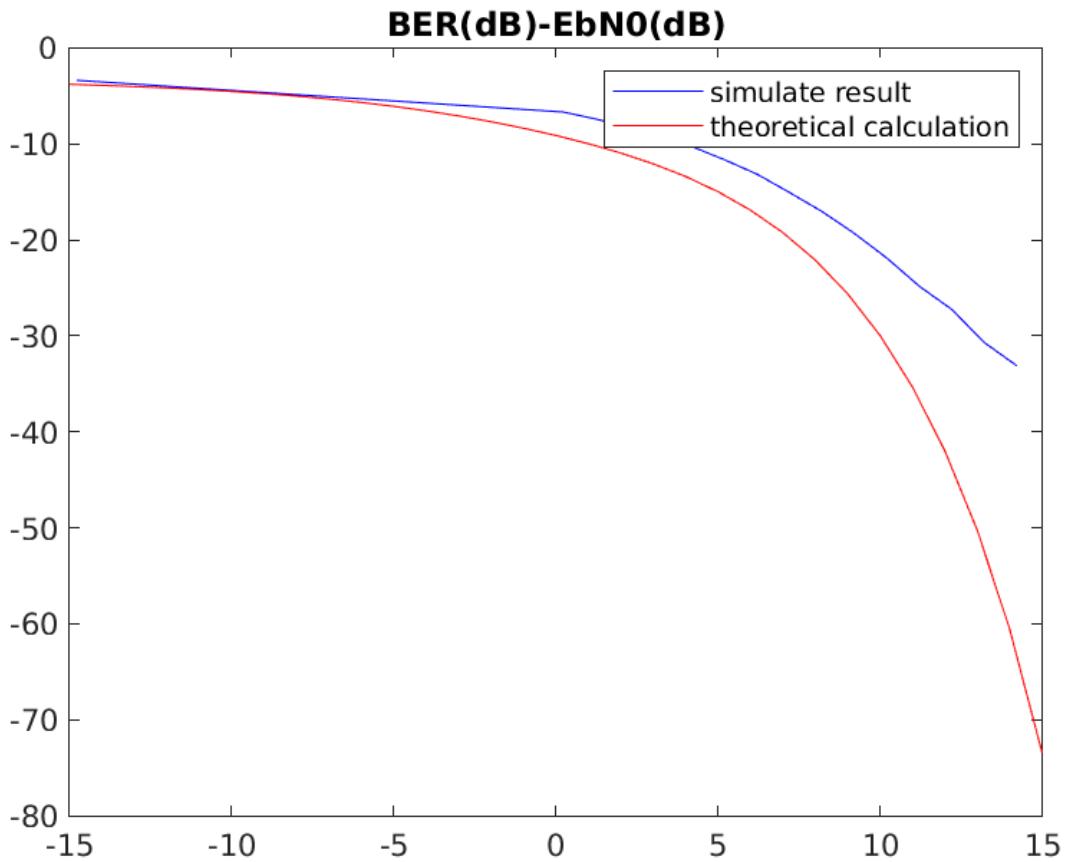
符号判决采用距离比较的方法，将接收到的各个符号同 8PSK 的 8 个相位点求距离，取距离最小的点作为判决结果，得到星座图。由于是 $\text{SNR}=10$ 时的图像，可以看到星座图较为明显地分为了 8 个密集区，对应 8PSK 的 8 个相位点。



同时做出发送比特序列和接收比特序列的图像，由于过于密集不够直观，故做出第三条图像，为二者差值，可以看到存在少量的错判。

3.9 误码率计算

```
1 % Calculate
2
3 bits_error = abs(bits_result - bits);
4 BER = [BER biterr(bits, bits_result)];
5
6 end
7
8 % display and save
9 figure(10)
10 hold off;
11 BE = BER;
12 BER = BER / (3 * length(bits));
13 plot(10 * log10(Es_range/(K * N0)), 10 * log10(BER), 'b');
14 hold on;
15 % plot the theoretical curve
16 plot(bers0.data {1,1}, 10 * log10(bers0.data {1,2}), 'r');
17 title ('BER(dB)-EbN0(dB)')
18 legend('simulate result', 'theoretical calculation');
19 saveas(10, [basepath 'figure/BER-EBN0.png']);
```



最后的误码率曲线采用对数坐标，可以看到当 $EbN0$ 较低时，仿真结果与理论曲线吻合较好，但当 $EbN0$ 大于 10dB 时，两条曲线有数量级的差距，仿真结果明显差于理想曲线。

结合滤波器的时域、频域曲线分析可知，在这种设计下的滤波器存在较强的符号间干扰，随着噪声的逐渐减小，符号间干扰的作用越来越明显，显著影响到判决结果。

§4 比较与分析

以上是对整个流程的简单介绍，下面针对不同信噪比、不同滤波器得到的结果进行比较和分析。

4.1 信噪比的影响

由星座图可以比较明显直观地看到 SNR 对于信道性能的影响。

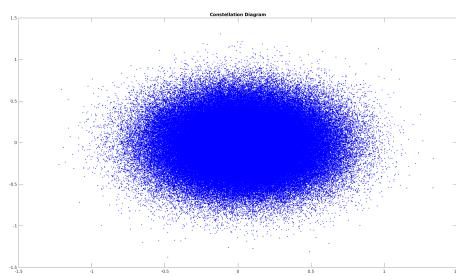


Figure 1: SNR=-19dB

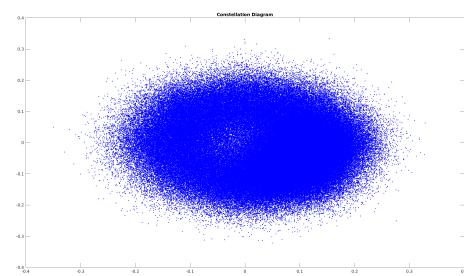


Figure 2: SNR=-4dB

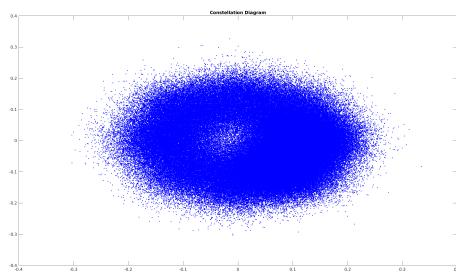


Figure 3: SNR=-3dB

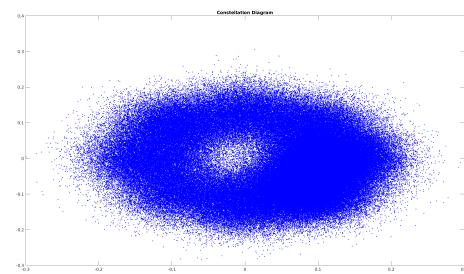


Figure 4: SNR=-2dB

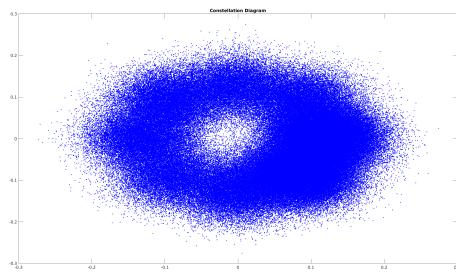


Figure 5: SNR=-1dB

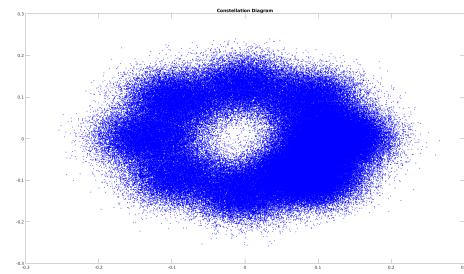


Figure 6: SNR=0dB

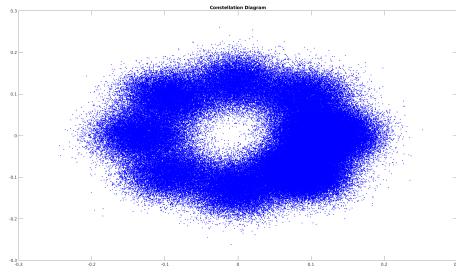


Figure 7: SNR=1dB

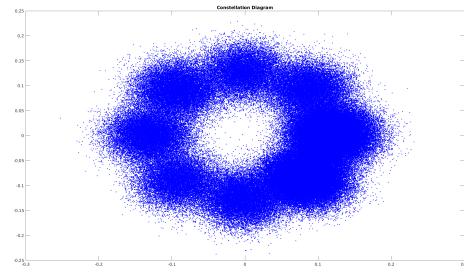


Figure 8: SNR=2dB

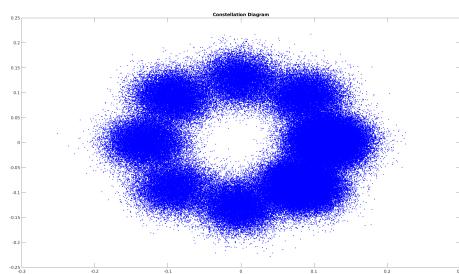


Figure 9: SNR=3dB

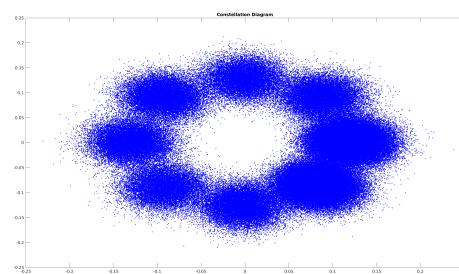


Figure 10: SNR=4dB

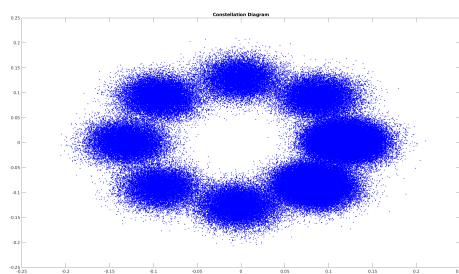


Figure 11: SNR=5dB

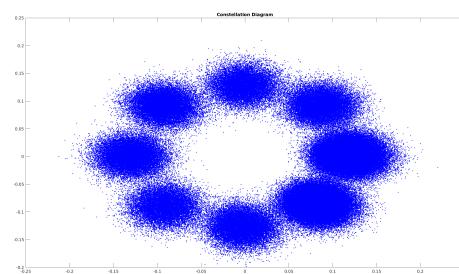


Figure 12: SNR=6dB

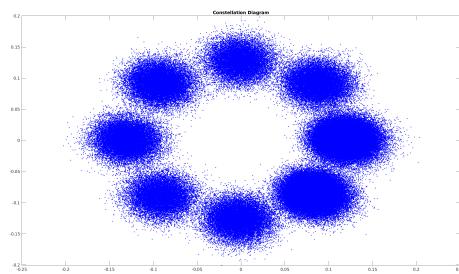


Figure 13: SNR=7dB

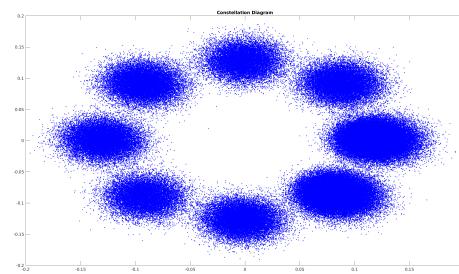


Figure 14: SNR=8dB

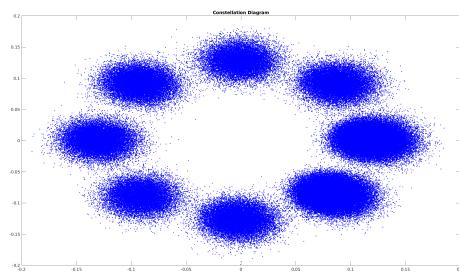


Figure 15: SNR=9dB

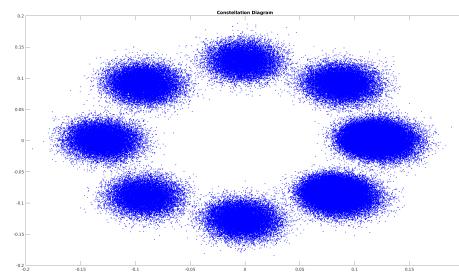
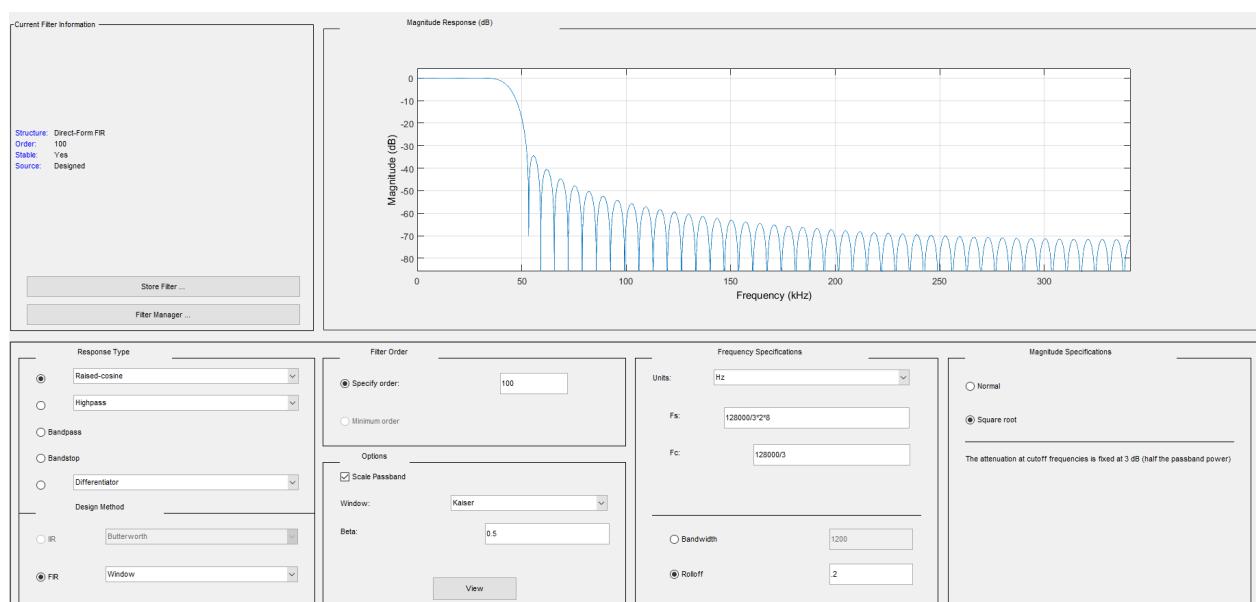


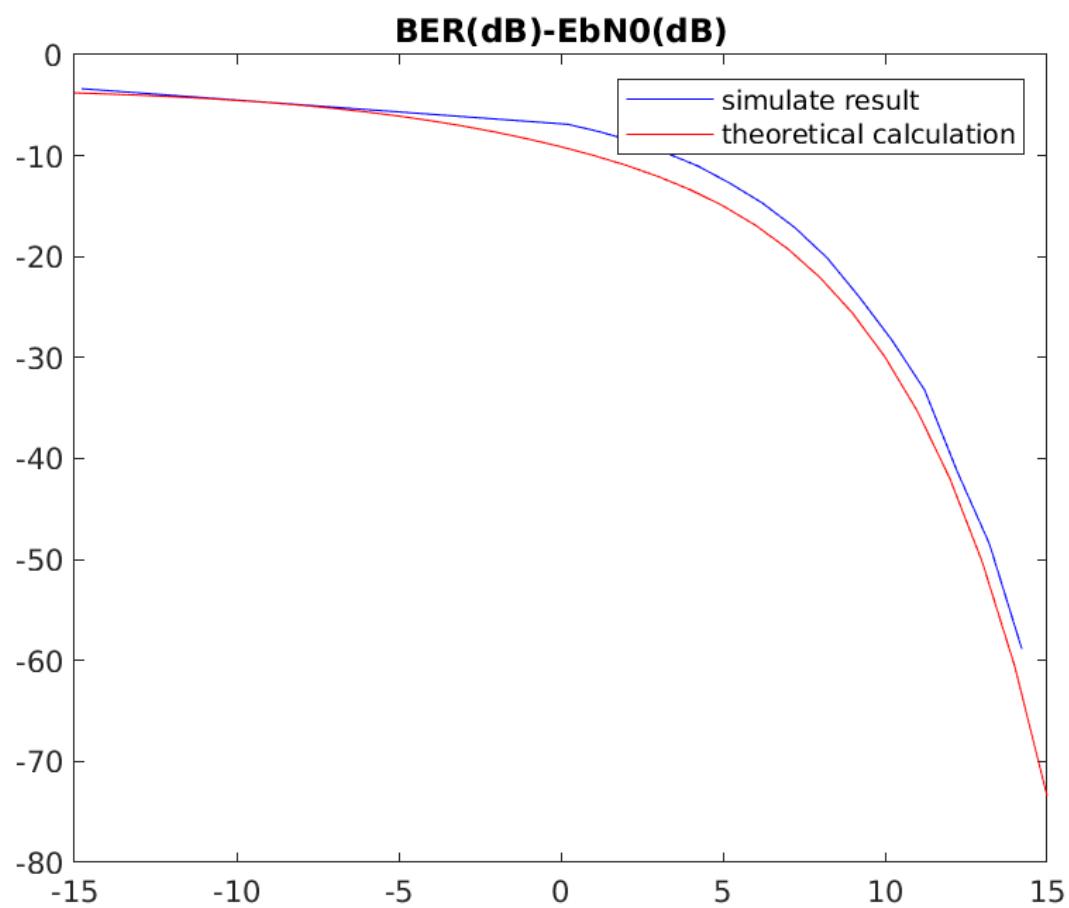
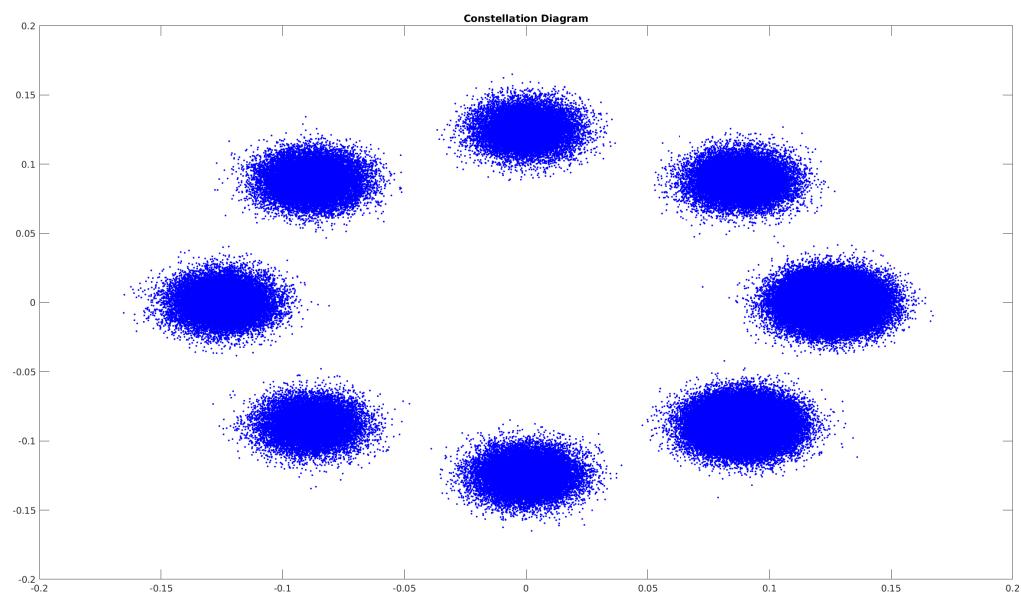
Figure 16: SNR=10dB

可以看到随着 SNR 的提升，星座图 8 个相位点逐渐清晰并彼此分离，表示此时判决的误码率不断下降。

4.2 滤波器的影响

使用等波纹方法设计的低通滤波器在 E_bN_0 较高时性能较差，与理想曲线相距较远，故尝试使用平方根升余弦滤波器，参数设置为凯泽窗，Beta=0.5，阶数为 100 阶， $F_s = 128000/3 * 8 * 2 \approx 682666\text{Hz}$, $F_c = 128000/3 \approx 42666\text{Hz}$ ，参数设计、得到的星座图及最后的误码率曲线如下：





可以看到升余弦滤波器星座图得到的散点更集中于 8 个相位点附近，误码率与理想曲线吻合更好，说明平方根升余弦滤波器在解决符号间干扰的问题上性能突出。

§5 附 0：辅助代码

5.1 dectobin.m

实现十进制整数（包含正负）到二进制的转化。

```
1 function bin = dectobin(dec, bit)
2     bin = dec2bin(abs(dec), bit);
3     for j = 1:length(dec)
4         if dec(j,1) < 0
5             for i = 1:bit
6                 if bin(j, i) == 48
7                     bin(j, i) = 49;
8                 else
9                     bin(j, i) = 48;
10                end
11            end
12            bin(j, bit) = bin(j, bit) + 1;
13            for i = bit:-1:2
14                if bin(j, i) == 50
15                    bin(j, i - 1) = bin(j, i - 1) + 1;
16                    bin(j, i) = 48;
17                end
18            end
19            if bin(j, 1) == 50
20                bin(j, 1) = 48;
21            end
22        end
23    end
24 end
```

5.2 HD.m

自己设计的等波纹滤波器参数内容。

```
1 function Hd = HD
2 %HD Returns a discrete-time filter object.
3
4 % MATLAB Code
5 % Generated by MATLAB(R) 9.4 and DSP System Toolbox 9.6.
6 % Generated on: 20-Jun-2018 00:42:54
7
8 % Equiripple Lowpass filter designed using the FIRPM function.
9
10 % All frequency values are in Hz.
11 Fs = 682666.66667; % Sampling Frequency
12
13 Fpass = 42666.666667; % Passband Frequency
14 Fstop = 46666.666667; % Stopband Frequency
15 Dpass = 0.057501127785; % Passband Ripple
16 Dstop = 0.0001; % Stopband Attenuation
17 dens = 20; % Density Factor
18
19 % Calculate the order from the parameters using FIRPMORD.
20 [N, Fo, Ao, W] = firpmord([Fpass, Fstop]/(Fs/2), [1 0], [Dpass, Dstop]);
21
22 % Calculate the coefficients using the FIRPM function.
23 b = firpm(N, Fo, Ao, W, {dens});
24 Hd = dfilt.dffir(b);
25
26 % [EOF]
```

5.3 SRC.m

自己设计的平方根升余弦滤波器参数内容。

```
1 function Hd = SRC
2 %SRC Returns a discrete-time filter object.
```

```
3
4 % MATLAB Code
5 % Generated by MATLAB(R) 9.4 and DSP System Toolbox 9.6.
6 % Generated on: 19-Jun-2018 23:52:49
7
8 % FIR Window Raised-cosine filter designed using the FIRRCOS function.
9
10 % All frequency values are in Hz.
11 Fs = 682666.66667; % Sampling Frequency
12
13 N = 100; % Order
14 Fc = 42666.666667; % Cutoff Frequency
15 TM = 'Rolloff'; % Transition Mode
16 R = 0.2; % Rolloff
17 DT = 'sqrt'; % Design Type
18 Beta = 0.5; % Window Parameter
19
20 % Create the window vector for the design algorithm.
21 win = kaiser(N+1, Beta);
22
23 % Calculate the coefficients using the FIR1 function.
24 b = firrcos (N, Fc/(Fs/2), R, 2, TM, DT, [], win);
25 Hd = dfilt. dffir (b);
26
27 % [EOF]
```

5.4 ber_standard.mat

8PSK 理想误码率曲线数据内容。