EE5400 Project: Designing communication signals for underwater acoustic communications

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Introduction

There are great advances in research and practical applications of underwater acoustic communications over the past two decades. One important area that underwater acoustic communications once used for is military. Once admitted lumbering single-digit to 100 b/s links, commercially available systems running in small-form-factor submersible buoys can achieve data rates in excess of 10 kb/s [1, 2]. Recent years underwater acoustic communication systems are paid great attention in several of applications such as commercial fishing and oil exploration. By remotely controlling equipment people try to probe, sense from a surface vehicle or station. For more precisely environmental sensing, wildlife monitoring and tracking and exploration of the potential of marine shorelines of the world, the high-rate digital acoustic communications is becoming an indispensable and important method. These applications range from command and control links to submarines and autonomous underwater vehicles in research, military, and search-and-rescue contexts, to remote operation and control of sensing equipment in deep sea fishing, off-shore oil exploration, and environmental monitoring[3].

As we know, information is sent in the form of electromagnetic waves in radio frequency (RF) communication. One or more sinusoidal components that have been modulated in amplitude and phase will compose the information bearing signal. This process will lead to single carrier or a multicarrier carrier. However, electromagnetic waves cannot propagate in the ocean for a long distance. The seawater conductivity results in rapid attenuation of the electromagnetic signal, especially at higher frequencies. An acoustic wave that used to carry information in underwater communication channel will propagate in a long period for a long distance, but the underwater channel offers several unique challenges for the design of high data rate digital communication systems.

The Major challenges in the design of underwater acoustic networks includes the available bandwidth is severely limited, the comparatively low velocity of acoustic propagation (compared with RF propagation) [1, 2]. While frequency-dependent propagation losses yield relatively small available signal bandwidth, and extremely variable, the underwater channel is severely impaired, high bit error rates and temporary losses of connectivity (shadow zones) can be experienced, due to the extreme characteristics of the underwater channel, battery power is limited and usually batteries cannot be recharged, also because solar energy cannot be exploited, underwater sensors are prone to failures because of fouling and corrosion[4]. Although there are many difficulties to implement underwater acoustic networks, by design suitable community system, researchers can successful collect kinds of data for further research. Our project is to design a communication signals for underwater acoustic communications.

Before to understand the system, it is better to clarify some basic principle. In general baseband signals have a spectrum that starts close to DC. Most analog information sources that are considered in communications are baseband such as audio, video, signals from sensors, images, etc. Also, any signal that is a sum of pulses of different shapes is a baseband signal. Yet another definition is that a baseband signal has not been modulated, or its spectrum has not been shifted, for the purpose of matching it to a particular communication channel. Passband signals were originally baseband, but were modulated, or frequency-shifted (or up-converted) for some purpose, usually to make them fit in a passband channel [5]. In this report, the main system contains original signal data, transducer, channel underwater and receiver. The up-conversion process involves changing the amplitude, frequency and/or phase of a high frequency carrier signal proportionally to the information-bearing baseband signal. The receiver uses the amplitude, frequency and/or phase of the carrier to reconstruct the baseband signal.

The figure1 shows the system for underwater acoustic communications. The input signal or original signal is b_k and after coding, it can be transport as C_n. The bit to symbol mapping can be easily realized in matlab since they already have great function. So we get our BPSK signal S_n. The signal is a complex number, using -1 to present 0 and using +1 to present 1. Pulse shaping filter is implement to generate the BPSK symbol. It is a square-root raised cosine filter. Then do carrier modulation for the baseband BPSK signals. After that we design a linear chirp and prepend it to the BPSK signal. This is the process to design a passband signal. Since S_n, pulse shaping filter and carrier signal have different sampling frequency, we use up-sample method to make fs being same. According to the given information, the frequency band is 16kHz to 21kHz. So we take 2.5kHz as the f_{symbol} ($T_{sym} = 1/ f_{sym}$). Assume sampling frequency of S_n is f_s, then the sampling frequency for the carrier signal should be $F_s = sps*L/ T_{sym}$ where sps is samples that one symbol contains. L is the ratio of fs and Fs. By define sps and L. we can gain F_s. The carrier frequency is given as 18.5kHz, it satisfy sampling theorem to avoid aliasing.

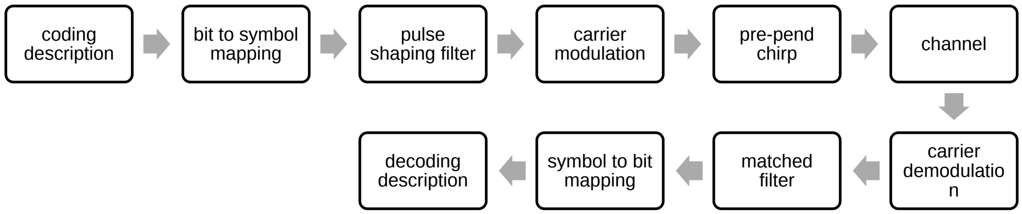

figure1. block chart of project

The process of modifying passband signal and down-converting to base signals is opposite procedures to generating passband signal. The passband signal with chirp pass through channel and then be demodulated. The carrier demodulation contains two parts, one is multiply an exponential and another is convolute with a low-pass filter. This baseband signal y_n then be down-sampled to keep same sample frequency with a matched filter. The carrier signal frequency is still 18.5 kHz, but the sampling frequency is multiples of 10240Hz, and the bandwidth change to 5120 Hz.

Implementations

Files: BPSK.mat

Functions:

Rcosdesign - Raised cosine FIR pulse-shaping filter design

Upsample - Upsample discrete-time models

conv2 - 2-D convolution

resample - Resample time-domain data by decimation or interpolation

chirp - Swept-frequency cosine

spectrogram - Spectrogram using short-time Fourier transform

fir1 - Window-based FIR filter design

freqz - Frequency response of digital filter

filter - 1-D digital filter

downsample - Decrease sampling rate by integer factor

audiowrite - Write audio file

audioread - Read audio file

Filter design: (1) rcosdesign(rolloff,span,sps); pulse shaping filter

where rolloff is 0.25, spam is number of symbols spanned by the filter, we use 8. Sps is number of samples per symbol, we use 10 here. Therefore, the transmitted waveform is correctly sampled at the receiver, the original symbol values can be recovered completely. Although there some noise.

(2) fir1(150,0.0833,'low',chebwin(151,80)); low pass filter with cutoff 2.5KHz. This low-pass filter only let low frequency signal pass. It has a good performance.

Results

Part A.1

Symbol Sampling frequency of 2.5KHz was selected for the BPSK signal. Code used to load and sample data:

```
load('BPSK.mat');
xn=BPSK(1,:);
Tsym=1/2.5e3;%sampling rate of BPSK Symbols in baseband
```

Part A.2

Square root cosine filter is designed to span eight symbols with each symbol constructed with ten samples. Matlab function rcosdesign is used to construct filter.

```
span=8;%number of symbols spanned by the filter
sps=10;%number of samples per symbol
rolloff=0.25;
b=rcosdesign(rolloff,span,sps);
```

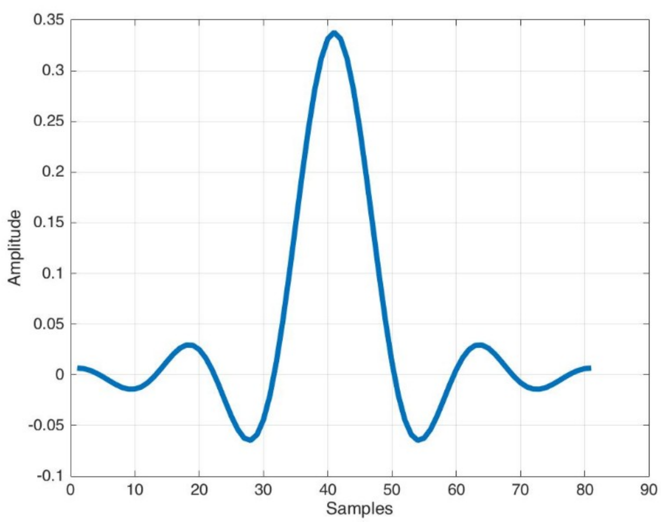

Figure 1Square root Cosine pulse shaping filter

BPSK data was upsampled (by 'sps' samples per symbol assumed in the design of filter) to match sampling rate of square root filter with the code given and convolved with the coefficients of the filter.
xn1=upsample(xn,sps);%sampling rate of filter is sps times sampling rate of BPSK data set
xn2=conv2(xn1,b);

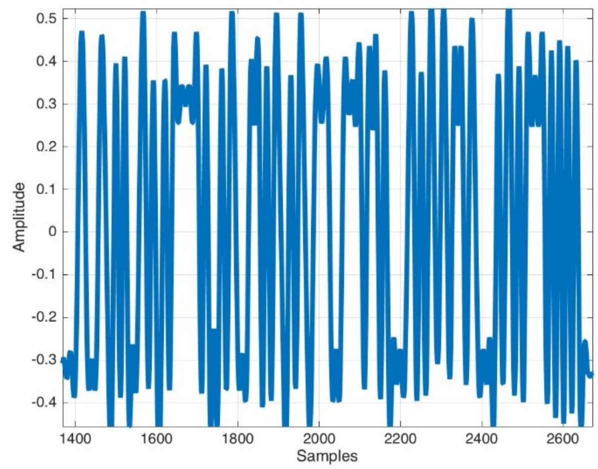

Figure 2 BPSK symbols generated by convolving with Filter

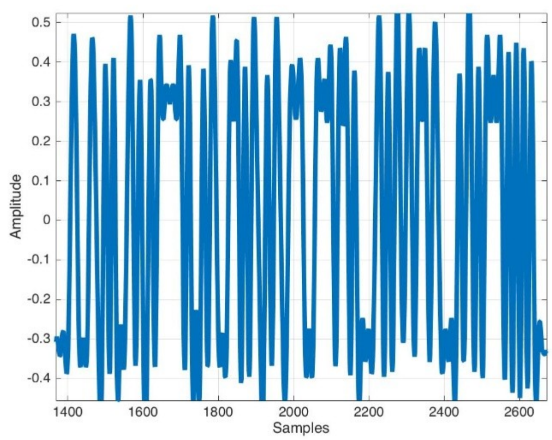

Figure 3 Zoom in picture of baseband signals

Part A.3

Baseband signal was modulated to passband using carrier frequency of 18.5KHz (center frequency of passband). Sampling frequency of carrier was chosen to be atleast greater than twice the highest frequency component present in the passband, 21KHz.
Baseband signal was upsampled to match sampling frequency of carrier signal. Upsampled baseband signal was multiplied by the carrier signal, to generate the passband signal. Length of carrier signal is matched to the length of baseband signal.
fc=18.5e3;%frequency of carrier, converting to passband
L=12;%number of times the sampling rate is a multiple of the sampling rate of filter

```
fs=sps*L/Tsym;
xn3=resample(xn2,72,6); %upsampled baseband signal
n=1:length(xn3);%modulation by carrier
cn=cos(2*pi*fc*n/fs);
xn4=xn3.*cn;
```

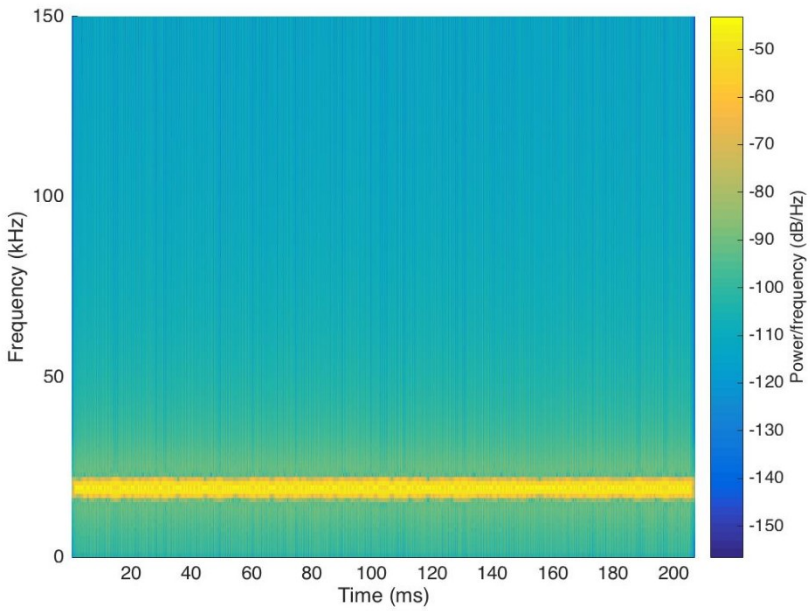

Figure 4 Spectrogram of Signal Modulated to Passband

Part A.4

Linear chirp is designed as per specifications of frequency sweep between 16KHz and 21KHz in 10 milliseconds.

```
t = 0: 1/fs:10e-3;%sweeps passband in 10 milliseconds match the sampling rate of chirp
y = chirp(t,16e3,10e-3,21e3);%sweeps between frequency ranges 16k and 21k
spectrogram(y,256,250,256,100e3,'yaxis');
wn=[y,xn4];%prepend chirp to passband signal
```

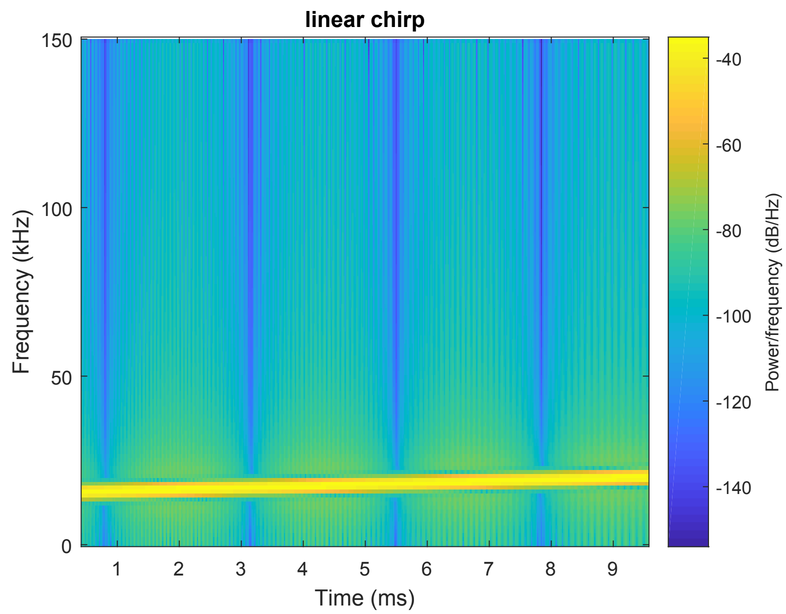

Figure 5 Chirp signal sweeping between 16KHz and 21KHz

Part A.5

Audiowrite function is used to create the .wav file. Sampling frequency used is the same as that used to sample carrier signal.
audiowrite('BPSK.wav',wn,fs);%match sampling rate

Part A.6

Audioread function is used to read back the .wav file.
[xn5, fs]=audioread('BPSK.wav');%read the audio file
spectrogram(xn5,256,250,256,fs,'yaxis');

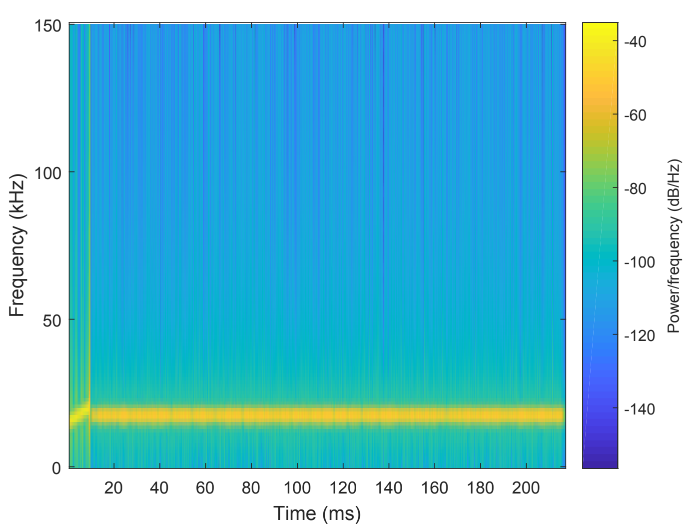

Figure 6 Spectrogram of generated audio signal

Part B.1

Sampling frequency is set to be a multiple of 10000Hz instead of 10240Hz to be a multiple of sampling frequency of carrier used in part A. Carrier frequency is same as that used for carrier signal in Part A. Bandwidth is 2.5Khz since baseband signal should have bandwidth half that of the signal in passband.

```
n=60;%sampling frequency should be matched to that of carrier in part A

Fsb=n*10000;%sampling frequency greater than twice highest frequency component and an integer multiple of carrier sampling frequency.

Fc=18.5e3;%when demodulation is performed signal is transformed into baseband, carrier frequency needs to matched

Fbw=2.5e3;%bandwidth of signal in baseband
```

Part B.2

Signal demodulation is performed by multiplying by complex exponential signal and filtering the high frequency content. Chebyshev low pass filter of order 150 is used.

```
%Demodulate the signal

xn4b=upsample(xn4,2);%sampling frequency should be matched between carrier and passband signal

nb=1:length(xn4b);

cnb=exp(i*2*pi*Fc*nb./Fsb);%defining carrier signal as complex signal

xn6=xn4b.*cnb;%multiplying by carrier signal both should have same sampling freq

spectrogram(xn6,512,500,512,Fsb,'yaxis');


%design of lowpass filter

b1 = fir1(150,0.0833,'low',chebwin(151,80));%low pass filter with cutoff 2.5KHz

[h,w]=freqz(b1,1);

plot(w/pi,abs(h));
```

```
legend('firpm Design');
xlabel 'Radian Frequency (\omega/\pi)', ylabel 'Magnitude';
ylim([0 50e3]);

%creating the complex baseband signal
xn7=filter(b1,1,xn6);
spectrogram(xn7,512,500,512,Fsb,'yaxis');
```

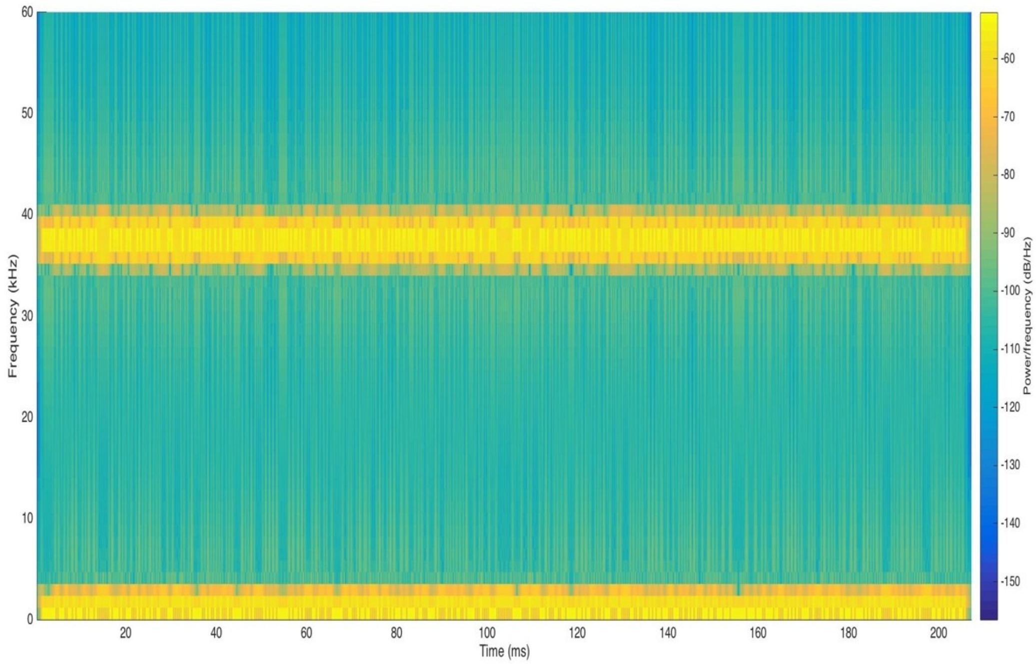

FIGURE 7 PASSBAND SIGNAL MULTIPLIED BY CARRIER

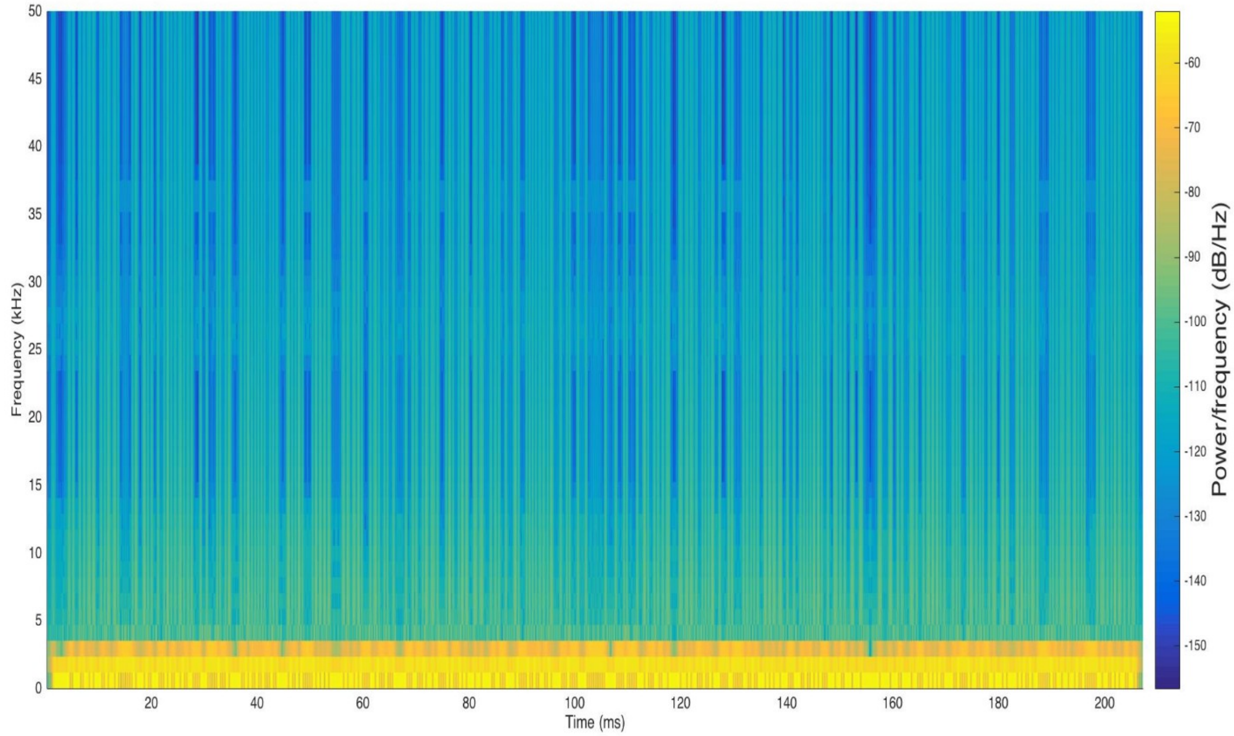

FIGURE 8 SPECTROGRAM OF COMPLEX DEMODULATED SIGNAL IN BASEBAND

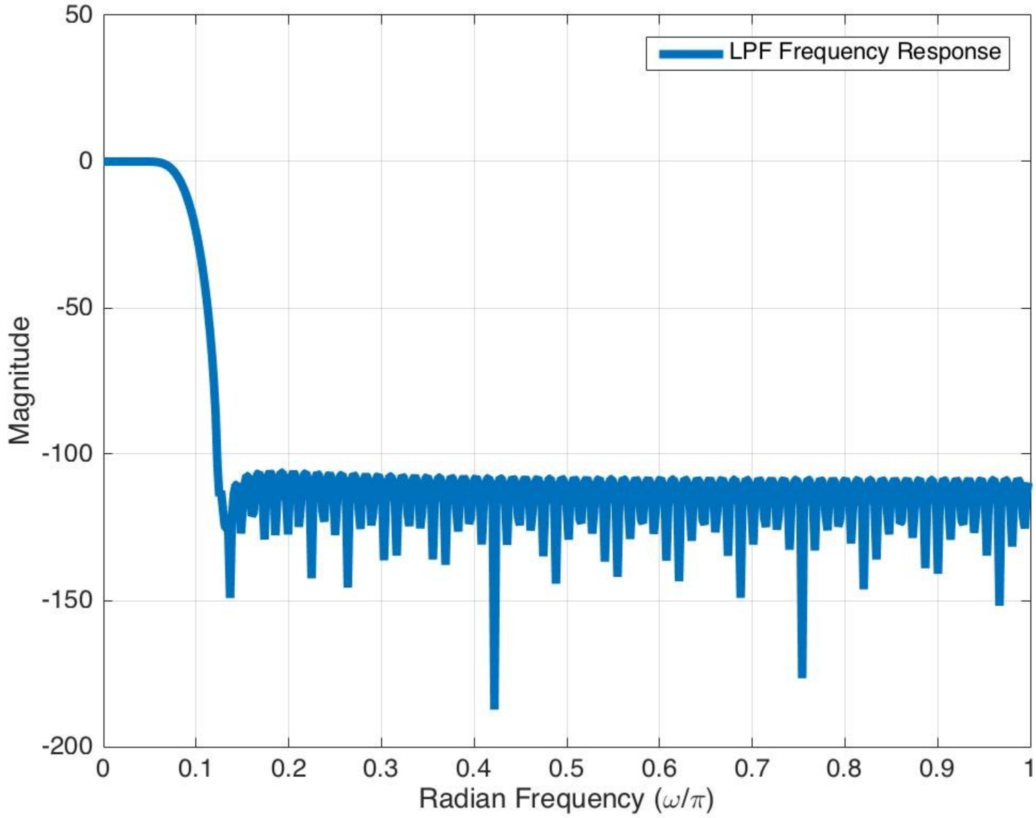

FIGURE 9 FREQUENCY RESPONSE OF LPF

Part B.3

Signal is downsampled using ‘downsample’ function in MATLAB.

```
%down-sampling the demodulated complex signal to 10000Hz
xn8=downsample(xn7,n);%downsampling to 10000Hz
fsb=10000;
%creating audio file for real part
audiowrite('BPSK1.wav',real(xn8),fsb);%match sampling rate
[xn9, fsb]=audioread('BPSK1.wav');%read the audio file
spectrogram(xn10,256,250,256,fsb,'yaxis');
```

```
%creating audio file for imaginary part
audiowrite('BPSK2.wav',imag(xn8),fsb);%match sampling rate
[xn10, fsb]=audioread('BPSK2.wav');%read the audio file
spectrogram(xn10,256,250,256,fsb,'yaxis');
```

Conclusion

After two weeks to do the project of designing communication signal, we have leant plenty of knowledge about how to design matched filter, how and when to use up-sample, down-sample and how to modulate and demodulate signal. Using pulse shaping filter we can generate BPSK symbols. To modulate, the baseband signal was multiply by a carrier signal. In order to increase the sampling rate, we use up-sample. The meaning of adding chirp is to make the signal become distinctive. To demodulate the signal, we multiple with an exponential and it was then be filtered by a low-pass filter. The reason of that using low-pass filter before the down-sample is that this will not lead to aliasing. Last, using down-sample to decrease the sample rate. One good acoustic communication system should have the ability to satisfy the limit bandwidth and variable in order to fit different work environment.

We work two weeks for the project to really understand the principle of every step. Both of us are working on codes and we discuss with each other. Xiaotong Tang worked on introduction, implement and conclusion. After getting the results, Santhosh Ranga Chavalla worked to analysis these results and comparing them.

Our final results is not good enough. In part A, the spectrogram shows the distribution of passband signal after modulate with a chirp signal along time. However, in part B, maybe there are some problem with the demodulation or somewhere the sampling frequency are not same. By changing parameter of different functions, we may get a better signals. There are some parameters still need to be determined, such as the function of resample, the p and q. This details will affect the final results.

References

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