Question (1)

a) In Matlab, read the following sentence and record your own voice for 4 seconds at a rate 40 kHz and generate a "message.wav" file.

"Use a pencil to write the first draft."

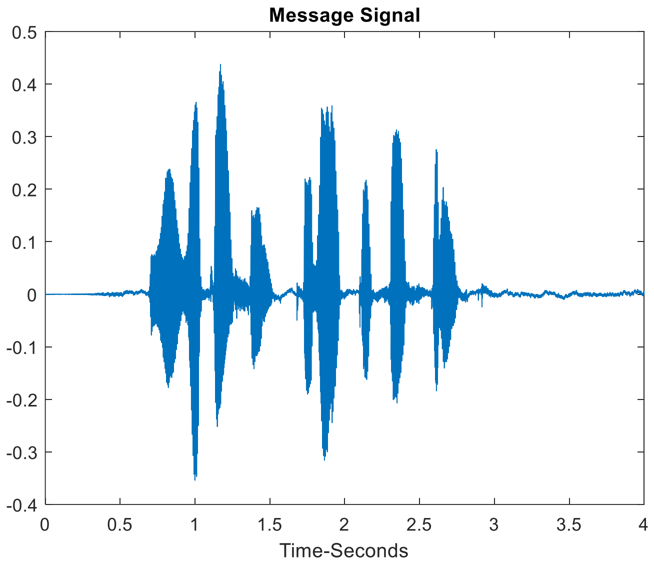

Figure 1 Message Signal

Fig.1 shows the message signal over time.

b) Downsample your message signal at a sampling rate 8 kHz, show the original signal and its discrete time samples for a very short period of time.

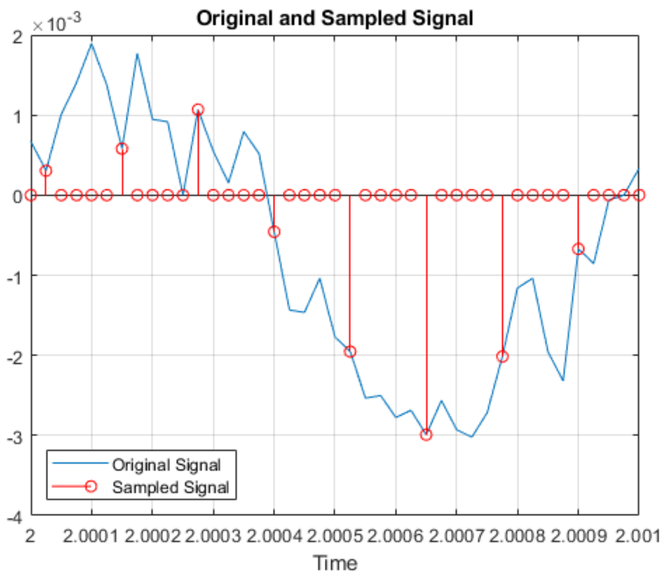

Figure 2 Original and Sampled Signal

Fig.2 shows our original and sampled signal for 8 kHz.

c) Using Spectrum Analyzer, show the two sided spectrum of the original signal and the sampled signal.

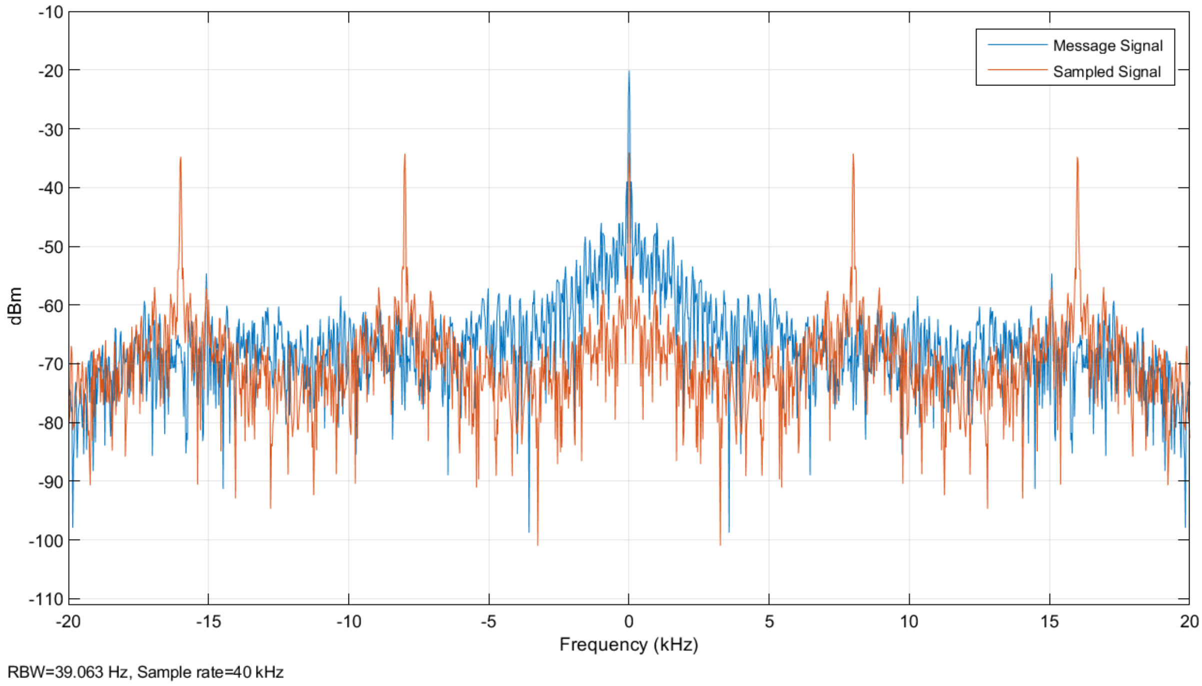

Figure 3 Two-sided Spectrum Analyzer of Message and Sampled Signals

Fig.3 shows the spectrum analysis of our original and sampled signal for 8kHz.

d) Define a low pass filter with suitable cutoff frequencies to recover the original low pass filter from its discrete time samples. Listen the original message signal and recovered message signal at 40 kHz and comment on your results.

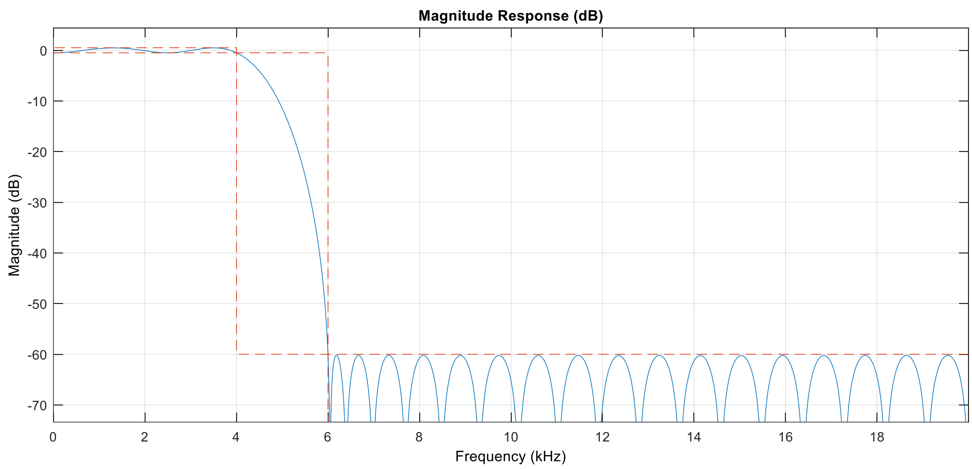

Figure 4 Low- Pass Filter for Sampled Signal

Fig.4 shows the magnitude response of the sampled signal.

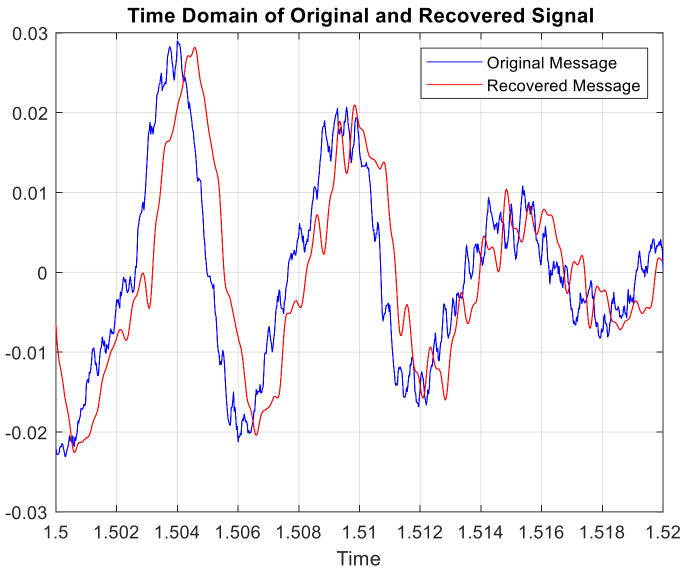

Figure 5 Original and Recovered Signal

Fig.5 shows the original and recovered signal there is a delay in time.

Comment: We chose our sampling frequency(fs) 8 kHz for our message and sampled our message every 5 stops.(N=ts/td=5, Figure 2) We applied a low pass filter in an appropriate frequency interval to recover our message in discrete time sampling(Figure4). As a result, when we listened to our recovery message, I heard a noise in the back according to our original message, and there was a slight phase shift(Figure 5).

e) Repeat the above steps for the new sampling rate 4 kHz. What happens at 4 kHz? Comment on your results.

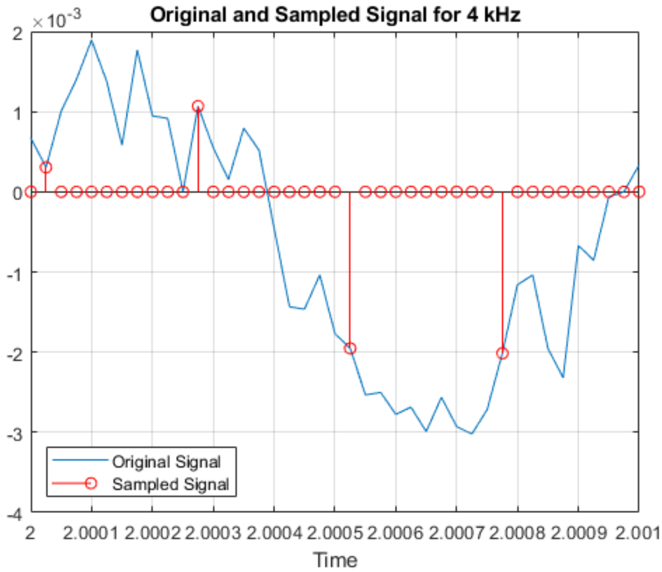

Figure 6 Original and Sampled Signal

Fig.6 shows our original and sampled signal for 4 kHz.

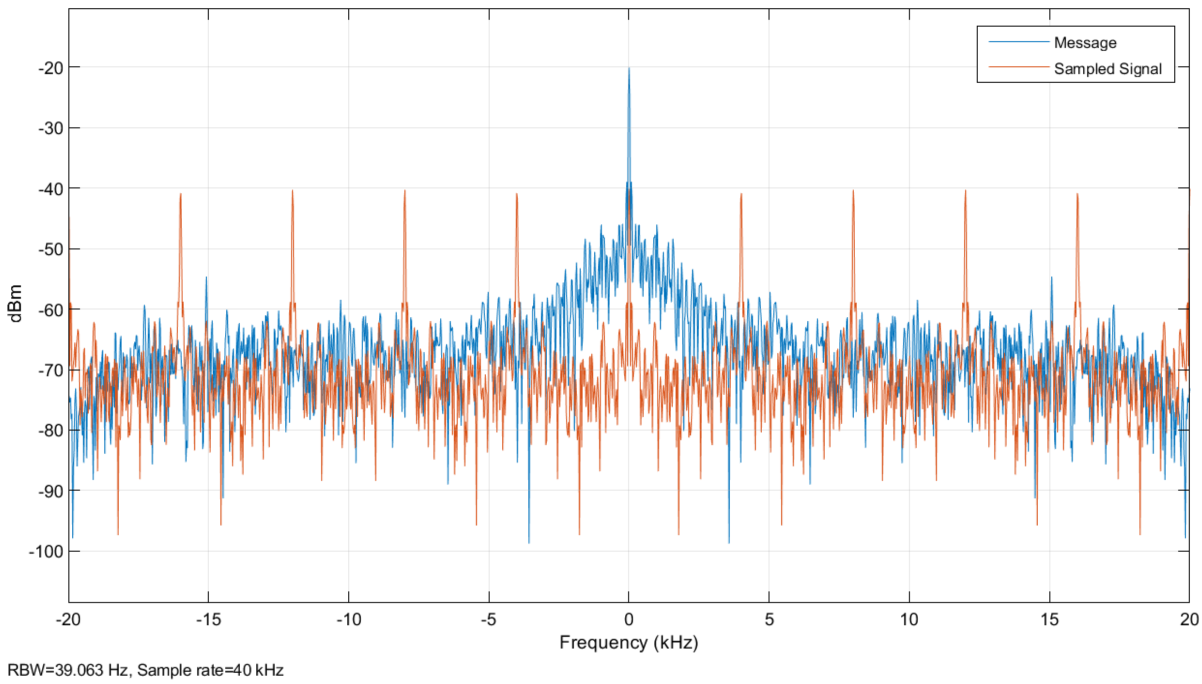

Figure 7 Two-Sided Spectrum Analyzer of Message and Sampled Signals

Fig.7 shows the spectrum analysis of our original and sampled signal for 4kHz.

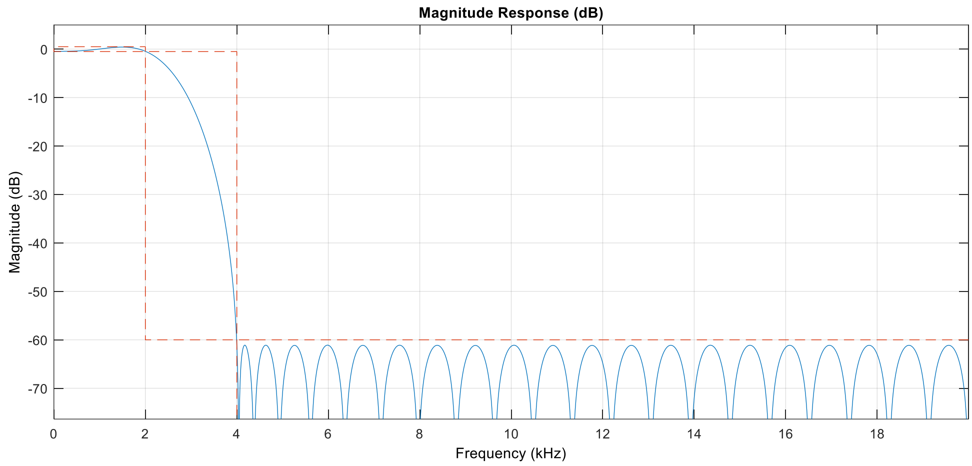

Figure 8 Low- Pass Filter for Sampled Signal

Fig.8 shows the magnitude response of the sampled signal for 4 kHz.

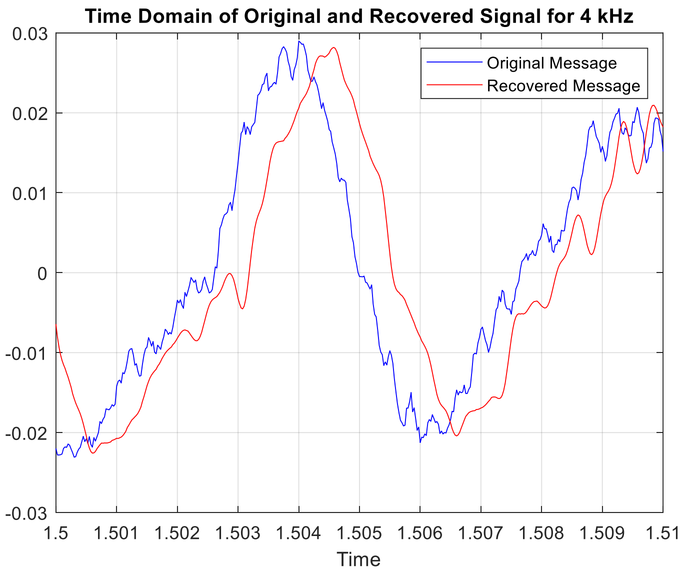

Figure 9 Original and Recovered Signal

Fig.9 shows the original and recovered signal there is a delay in time for 4 kHz.

Comment: We have applied the above operations again by changing our sampling frequency(fs) to 4 kHz. This time we sampled our message every 10 stops(N = ts/td = 10 , Figure 6). When I listened to our recover message, there was more noise compared to the first situation, and the sound quality had also decreased. In other words, we can say that the distortion in the sound increases as our sampling frequency decreases.(Figure 9)

Question(2):

a) In Matlab, read the following sentences and record your own voice for 4 seconds at a rate 40 kHz and generate “message1.wav” and “message2.wav” files.

“The two met while playing on the sand.”

“This is a grand season for hikes on the road.”

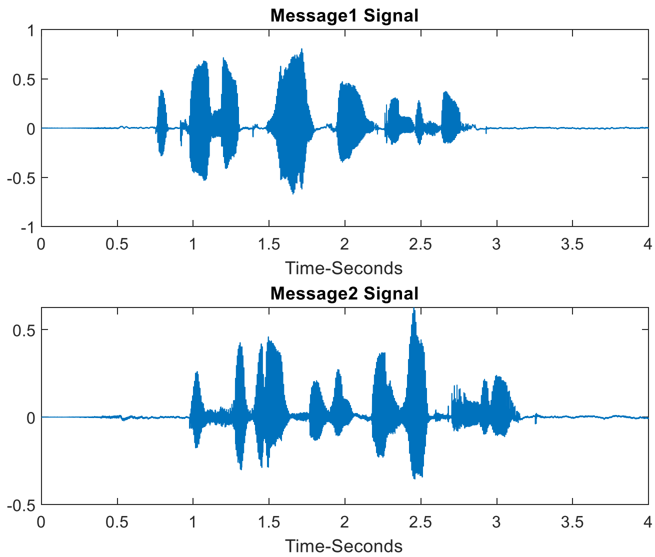

Figure 10 Message1 and Message2 Signals

Fig.10 shows the message signals over time.

b) In Matlab, implement an analog QAM modulator with carrier frequency 8 kHz. Show the two sided QAM spectrum using Spectrum Analyzer.

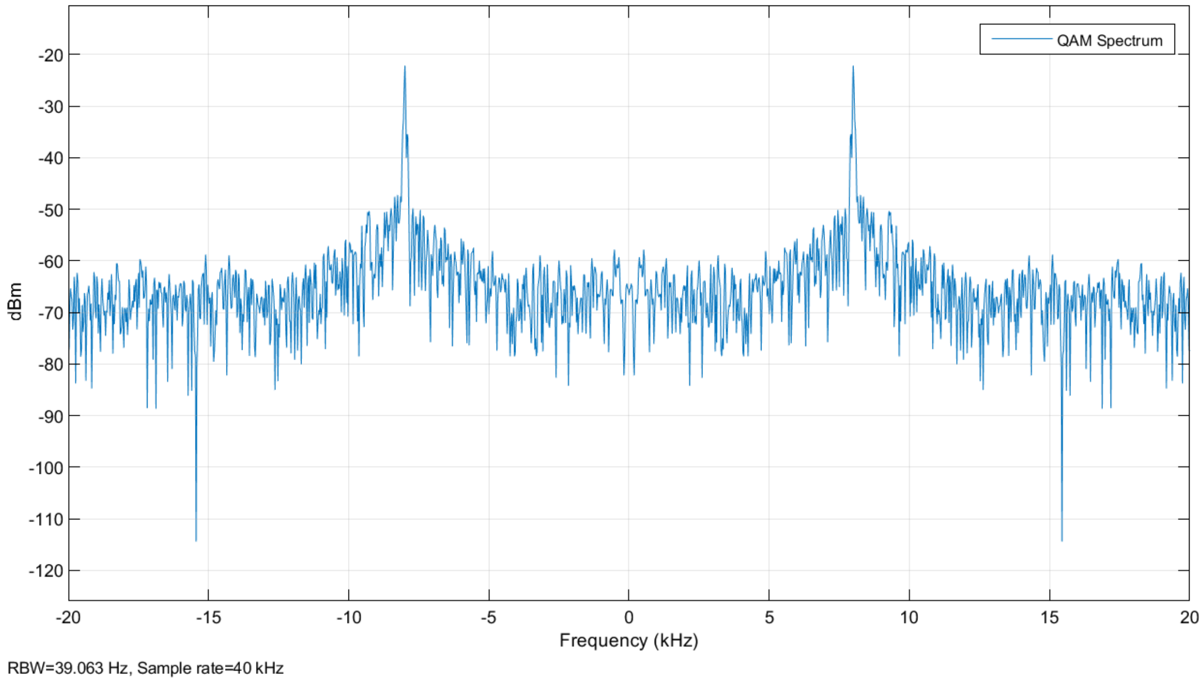

Figure 11 Two-sided Spectrum Analyzer of QAM message1 and message2

Fig.11 shows the spectrum analysis of QAM modulator(message1 and message2) for 8 kHz.

c) Add Gaussian channel noise whose power is 1 micro Watts.

We have set N = e^{-6}.

d) Implement QAM demodulator and recover message 1 and message 2. Listen the original message signals and recovered message signals and comment on your results.

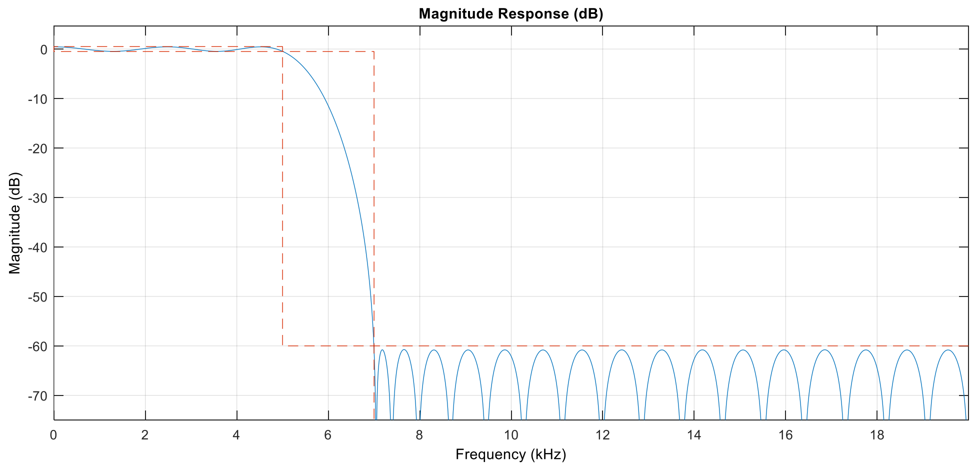

Figure 12 Low-Pass Fİlter of Message1 Signal

Fig.12 shows the magnitude response of the message1 signal.

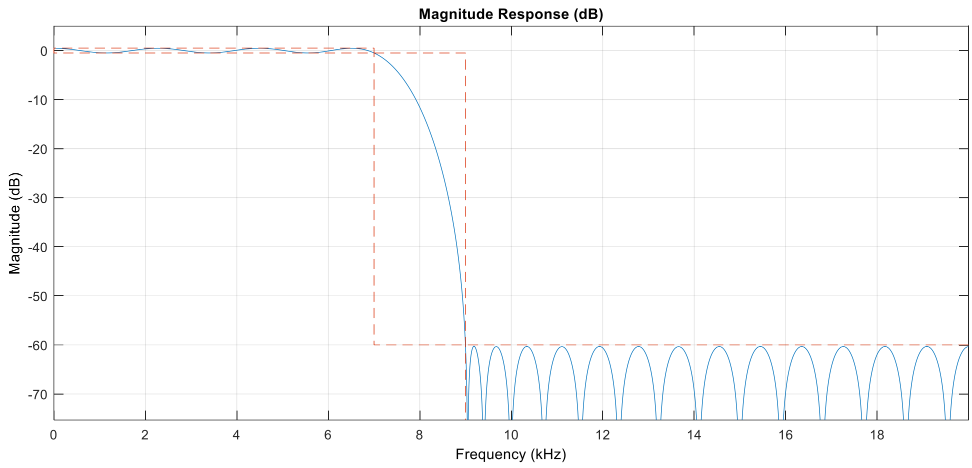

Figure 13 Low-Pass Filter of Message2 Signal

Fig.13 shows the magnitude response of the message2 signal.

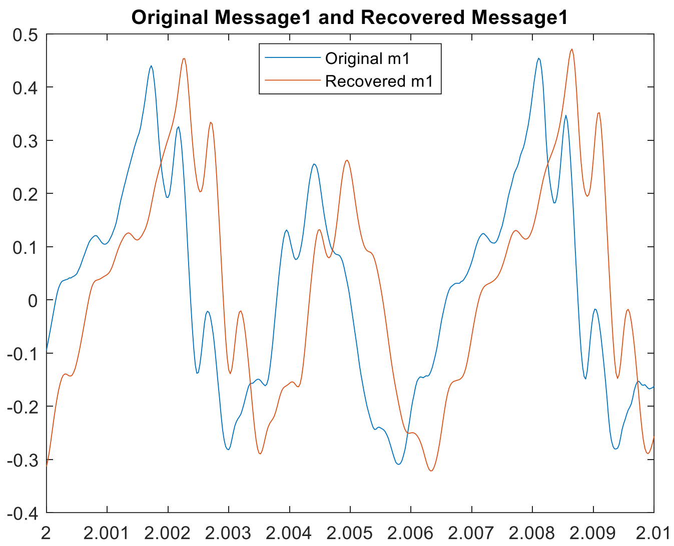

Figure 14 Original and Recovered Signal

Fig.14 shows the original m1 and recovered m1 signal to N = e^{-6}. There is a delay in time.

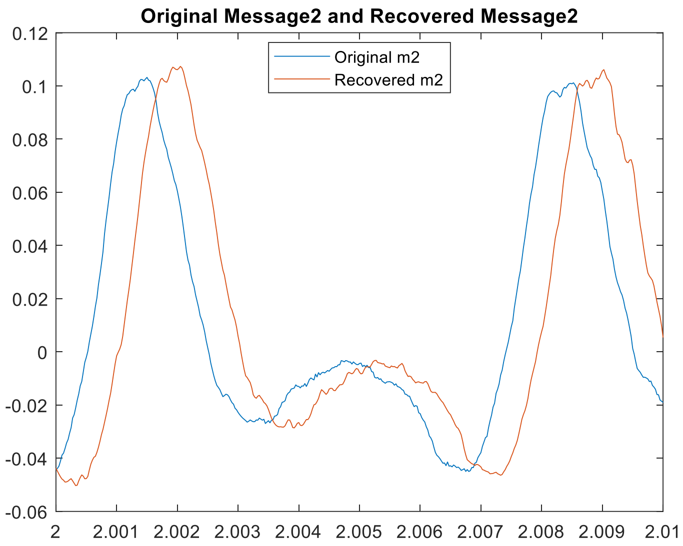

Figure 15 Original and Recovered Signal

Fig.15 shows the original m2 and recovered m2 signal to N = e^{-6}. There is a delay in time.

Comment: We set the carrier frequency to 8 kHz for each of our 2 messages and applied a QAM modulator. We have added noise to our messages in such a way that N = e^{-6}. We have applied QAM demodulator to recover our messages. As a result, when I listened to our original messages and recovered messages, the distortion in the sound was low, but the noise in the back was high(Figure14,Figure15).

e) Repeat the above steps with channel noise whose power is 1 milliWatts. Comment on your results.

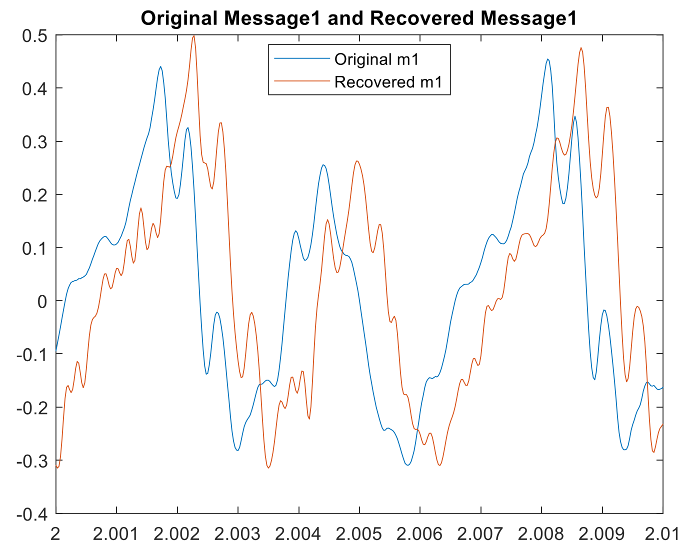

Figure 16 Original and Recovered Signal

Fig.16 shows the original m1 and recovered m1 signal to N = e^{-3}. There is a delay in time.

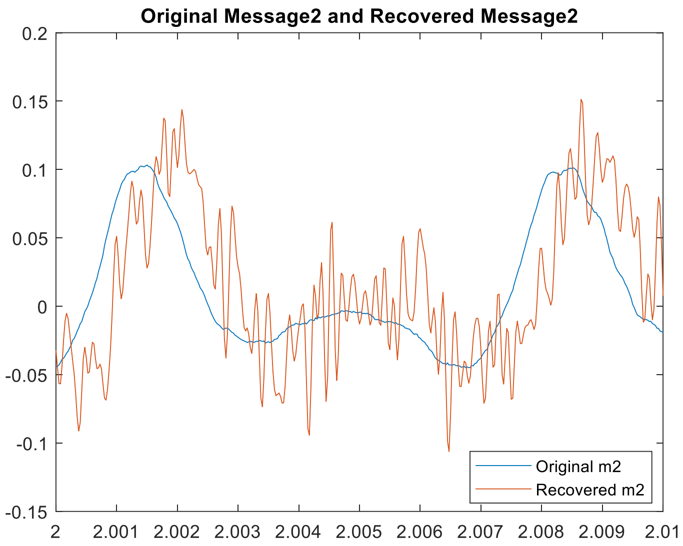

Figure 17 Original and Recovered Signal

Fig.17 shows the original m2 and recovered m2 signal to N = e^{-3}.

Comment: This time we applied the above operations for the noise coefficient N = e^{-3}. When I listened to our messages, there were distortions in the sound, and according to the initial situation, the noise in the back was too much. As a result, as the noise coefficient(N) we add to the sound increases, we cannot get the sound back at the rate we want.

Question(3):

a) In Matlab, read the following sentence and record your own voice for 4 seconds at a rate 40 kHz and generate “message.wav” file,

“We find joy in the simplest things. ”

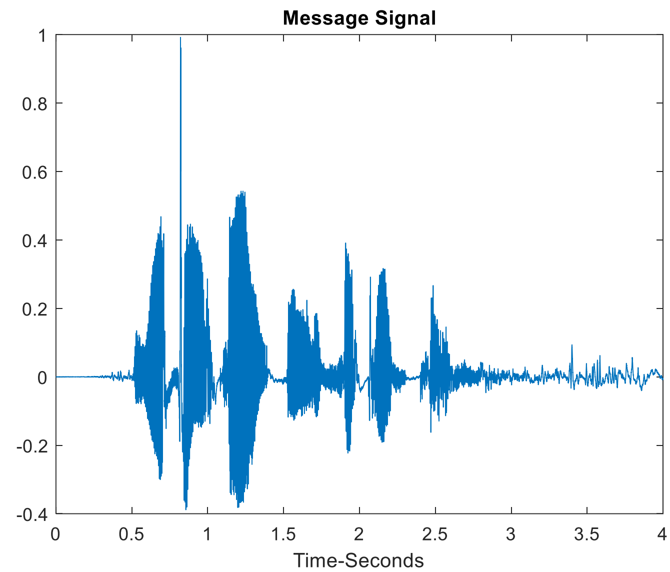

Figure 18 Message Signal

Fig.18 shows the message signals over time.

b) In Matlab, implement an USB modulator with carrier frequency 8 kHz. Show the two sided USB spectrum and DSB spectrum using Spectrum Analyzer.

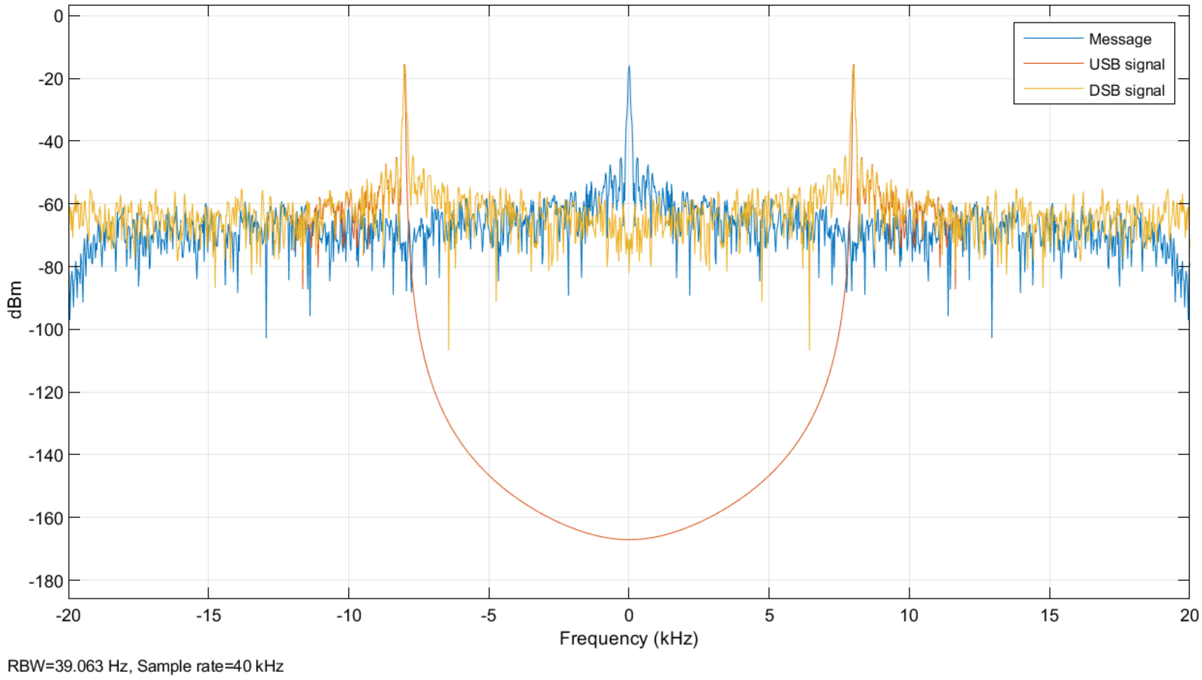

Figure 19 Two-sided Spectrum Analyzer of USB/DSB Modulated Signals

Fig.19 shows the spectrum analysis of message signal(USB/DSB MODULATED) for 8 kHz

c) Add Gaussian channel noise whose power is 1 micro Watts.
We have set N = e^{-6}.

d) Implement USB/LSB demodulators and recover the message. Listen the original message signals and recovered message signals and comment on your results.

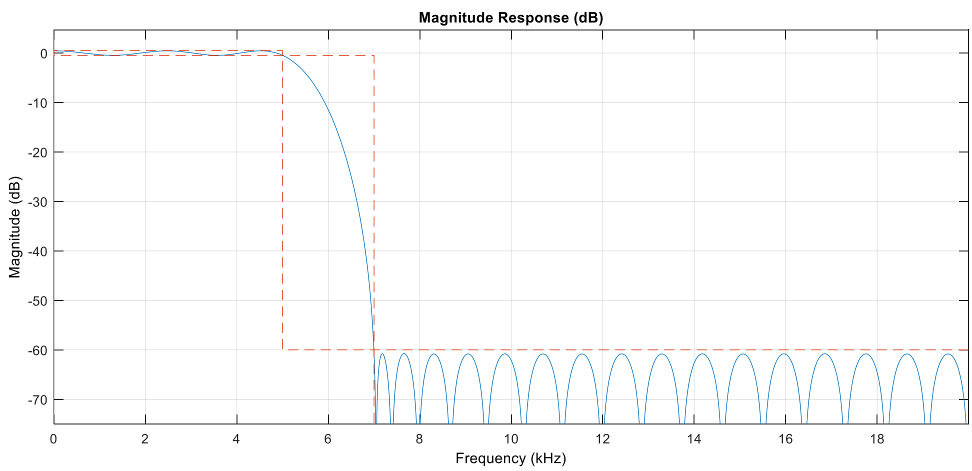

Figure 20 Low-Pass Filter of Message Signal

Fig.20 shows the magnitude response of the message signal to N = e^{-6}.

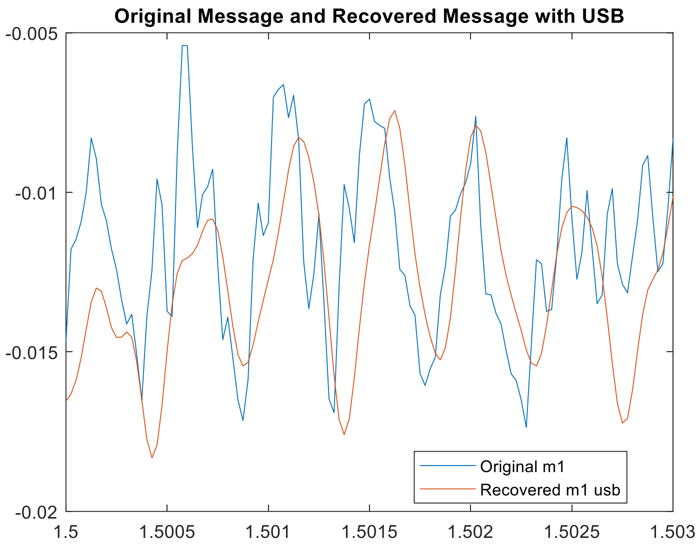

Figure 21 Original and Recovered Signal

Fig.21 shows the original m1 and recovered m1 signal to N = e^{-6} . (usb demodulated)

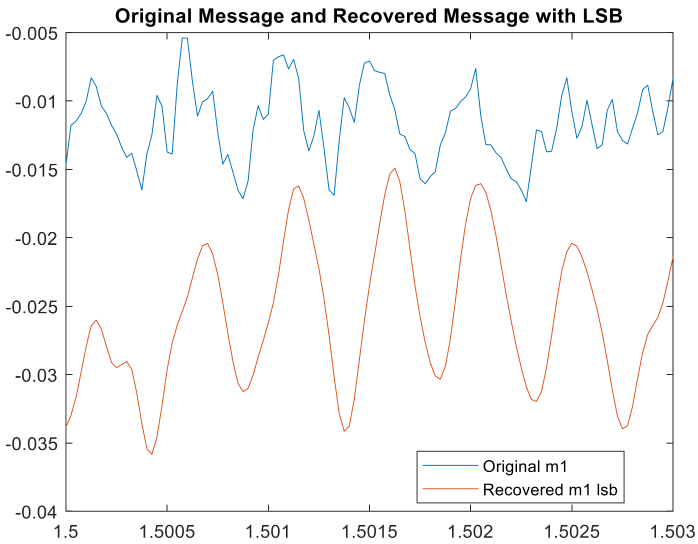

Figure 22 Original and Recovered Signal

Fig.22 shows the original m1 and recovered m1 signal to N = e^{-6} . (lsb demodulated)

Comment: We selected the 8 kHz carrier frequency for our message and showed the USB/DSB modulator spectrum (Figure 19). Then we added noise(N = e^{-6}) to our message and applied the USB/LSB demodulator to recover our message(Figure 21, Figure 22). When I listened to the message that we applied USB demodulator, the power of the sound was almost same and there was noise in the back. When I listened to the message that we applied the LSB demodulator, the power of the sound had increased and there was noise in the back.

e) Repeat the above steps with channel noise whose power is 1 milliWatts. Comment on your results.

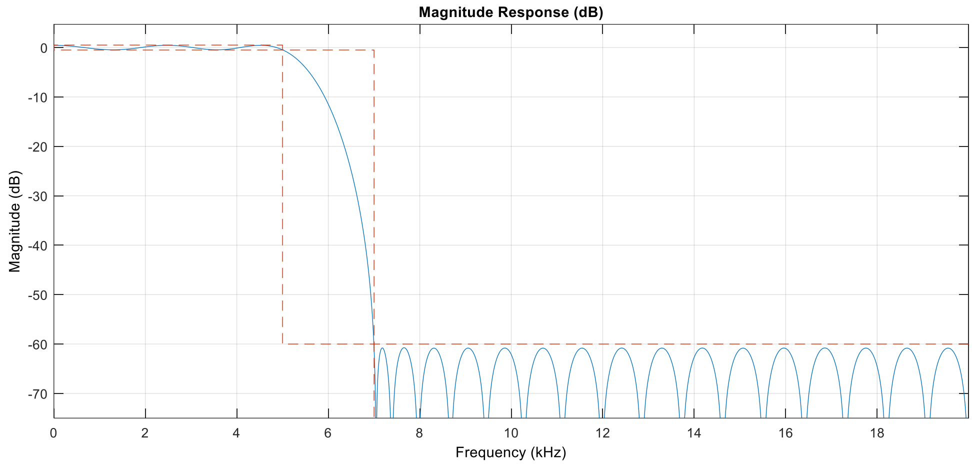

Figure 23 Low-Pass Filter of Message Signal

Fig.23 shows the magnitude response of the message signal to N = e^{-3}.

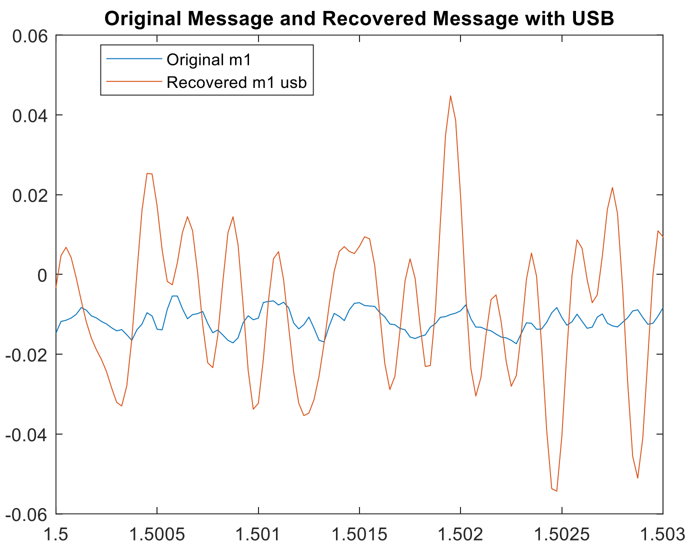

Figure 24 Original and Recovered Signal

Fig.24 shows the original m1 and recovered m1 signal to N = e^{-3} . (usb demodulated)

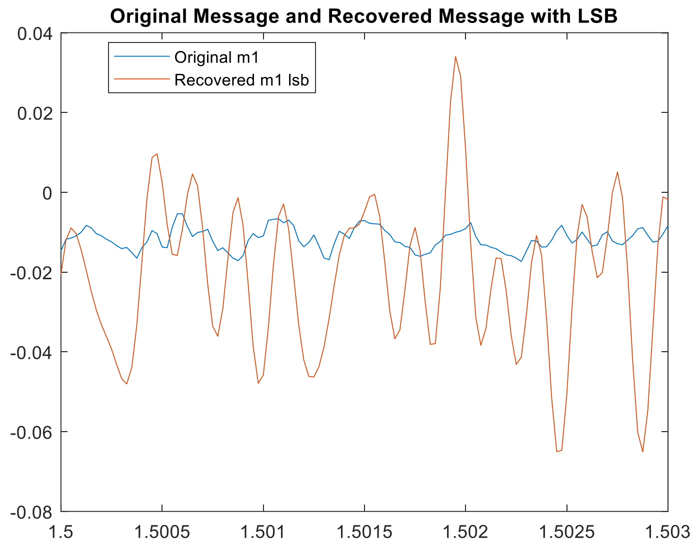

Figure 25 Original and Recovered Signal

Fig.25 shows the original m1 and recovered m1 signal to N = e^{-3} . (lsb demodulated)

Comment: The operations we performed above, this time we set the noise coefficient to N = e^{-3} and applied it. When I listened to our USB demodulator message, the noise in the back had increased. When I listened to our message that we had applied LSB demodulator, the power of the sound had increased and the noise in the back was too much. As a result, as we increase the noise coefficient, we cannot recover the sound at the rate we want, and the LSB demodulator increases the power of the sound(Figure24 , Figure25).