

CT PROJECT REPORT

NAME : ARYAMAN MAHAJAN, 2022102034
: YASH NITIN DUSANE, 2022102078

System Design Approach (4-Ary ASK)

The audio file is taken as input by the '**audioread**' function. In case of stereo audio, i.e., 2 channel input, we just take the first channel for processing. The reason for adopting this approach rather than conversion to mono audio is that in case of both channels being out of phase, the conversion to mono by taking mean results on cancellation of the audio and results in just silence. Thus, we may consider only the first channel for processing further.

The conversion of the wav data to bits is carried by the inbuilt **dec2bin** function, as per the link provided in the project document's sheet. The similar inbuilt process using **bin2dec** function was also carried out for the reverse quantization as well.

For given scheme of 4-ary ASK, the bitstream generated was read and each group of 2 bits were consecutively assigned a number based on the pattern:

0. 4-ary amplitude-shift keying (ASK)

Here you will map 2 consecutive bits to an amplitude level, as follows,

$$00 \rightarrow 0, 01 \rightarrow 1, 10 \rightarrow 2, 11 \rightarrow 3$$

For line coding on these symbols, the symbol train was convolved with a raised cosine pulse. Upon testing for the entire system's performance, we realised that on decreasing the roll off factor, the audio heard at the output was slightly better in quality. But on taking it very close to zero, the rc pulse start to resemble a sinc wave, which again might not give the best results. Hence, the value for the roll off factor was chosen to be of around **rof = 0.25**. Before convolution, the datastream was upsampled by a factor of **oversampling factor = m = 4**. The oversampling serves to reduce the error in sampling at the output and provides better resolution.

For convolution with the rectangular pulse, a similar oversampling of the original data is carried out by a factor of **$m=4$** .

This process of convolution generates the baseband communication sequence. For transmission we need to multiply it to higher frequency for practical implementation in terms of antenna size, power consumption etc. Thus, as per the instruction, we modulate it with a ***cosine carrier of 1M Hz***, using ***DSB-SC modulation***.

The reason for employing DSB-SC as opposed to SSB modulation is that SSB requires Hilbert transform's implementation which is complex in actual hardware. Also, since the carrier generated from Hilbert transform requires a different demodulation, which is also complex in hardware, we used DSB-SC which requires more bandwidth but is easier to implement.

This modulated signal is transmitted through the channel to the receiver. The two different channels, i.e, AWGN Channel with memory and AWGN channel without memory were implemented independently and plots were generated for both.

For AWGN Channel without memory, at different values of sigma (standard deviation), the audio heard at the final output was different. On increasing the sigma, the quality of audio heard at the output starts to deteriorate. This is because on increasing the variance (square of sigma), the probability of the noise taking values farther from zero mean, that is larger value, starts to increase. Hence, for better perception of the received audio, we set ***sigma = 0.001***. For clarity in plots regarding effect of noise, we use larger values, such as ***sigma = 1 or sigma = 0.1***.

For AWGN Channel with memory, the transfer function is modelled using an array with zeros between the end point values. The first value in the array is a , and the last value is $1-a$. The number of zeros appended between the endpoints is b times T_b , as specified in the project doc. It is observed that on increasing the value of a , the effect of the memory dies out as the delayed impulse is of magnitude $1-a$. Hence, on small values of a , there is significant effect of memory, however for larger values, there is very little effect. Thus, similar as above, for clarity in plots regarding memory effect, we use smaller values of a , around ***a=0.4***. But for improved quality at the output, we set ***a = 0.9*** to minimise the effect of memory.

At the demodulation side, we again multiply the noisy signal by the cosine carrier of the same frequency, thus implementing coherent demodulation. Further we pass this multiplied carrier into a lowpass filter with cutoffs determined according to the carrier

and sampling frequency. The cutoffs thus are **$f_c = 48000$, and $f_s = 48000 * 8$** The output of the lowpass is the noisy baseband transmission.

This is sampled at the output and the sampled values are sent to the line decoder. The decoder uses MAP detection to round the values to the amplitude level most probably transmitted.

This output is further sent to the decoder and then converted back to bits. The D2A converter converts the bitstream back to audio samples.

Using '**audiowrite**' we write this audio data to a new file.

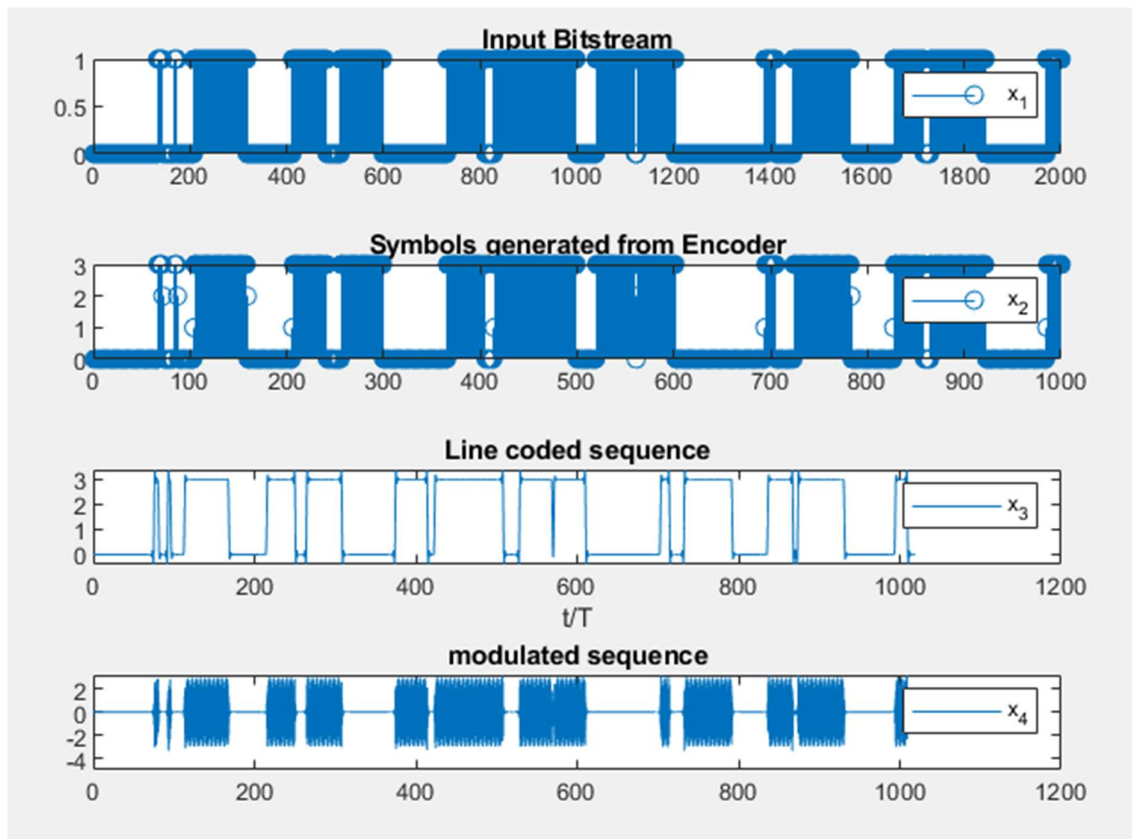
Rectangular vs Raised Cosine Comparison :

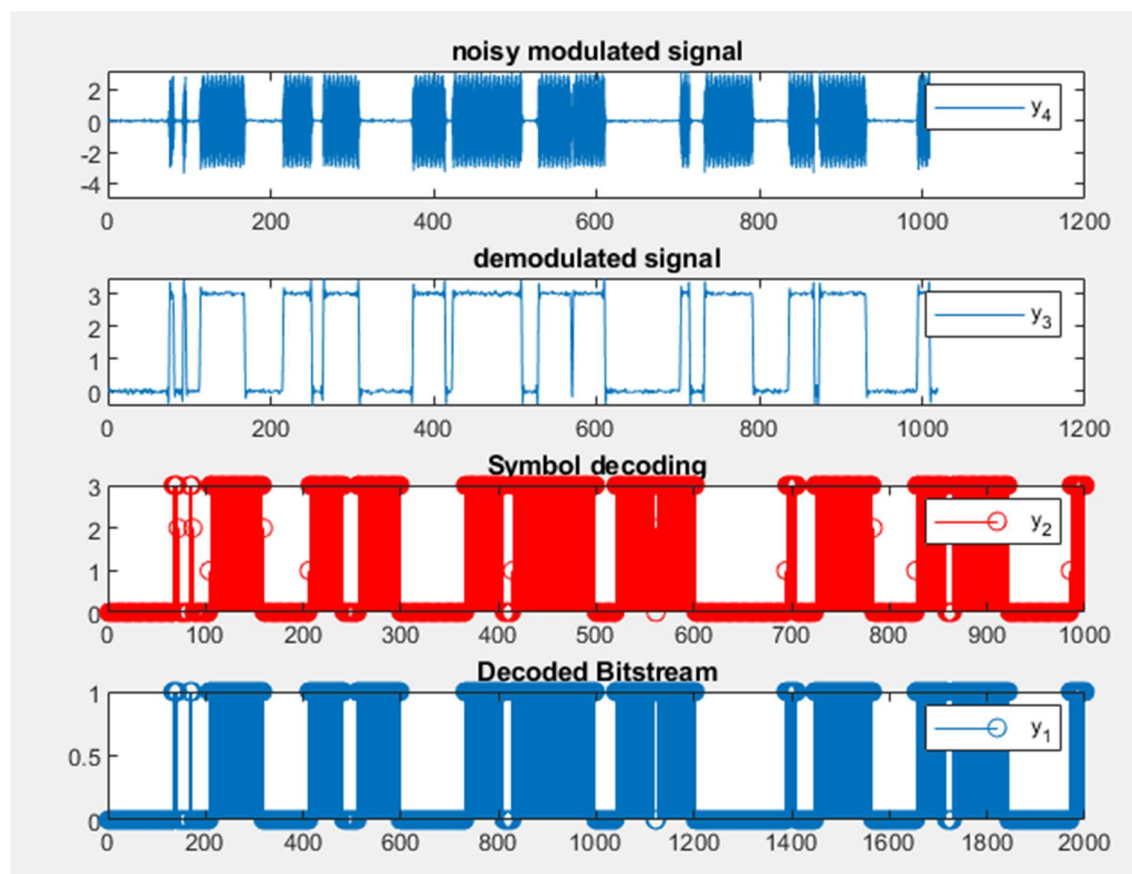
Since the raised cosine pulse is spectrally shaped to avoid ISI, it theoretically as well as practically leads to better performance in terms of BER, perceived audio etc.

The pulse is shaped as per Nyquist's second criterion. The more the roll off factor increases, the more it expands in frequency and analogously compresses in time. For roll off factor =0, the pulse is just a sinc wave.

OUTPUT PLOTS

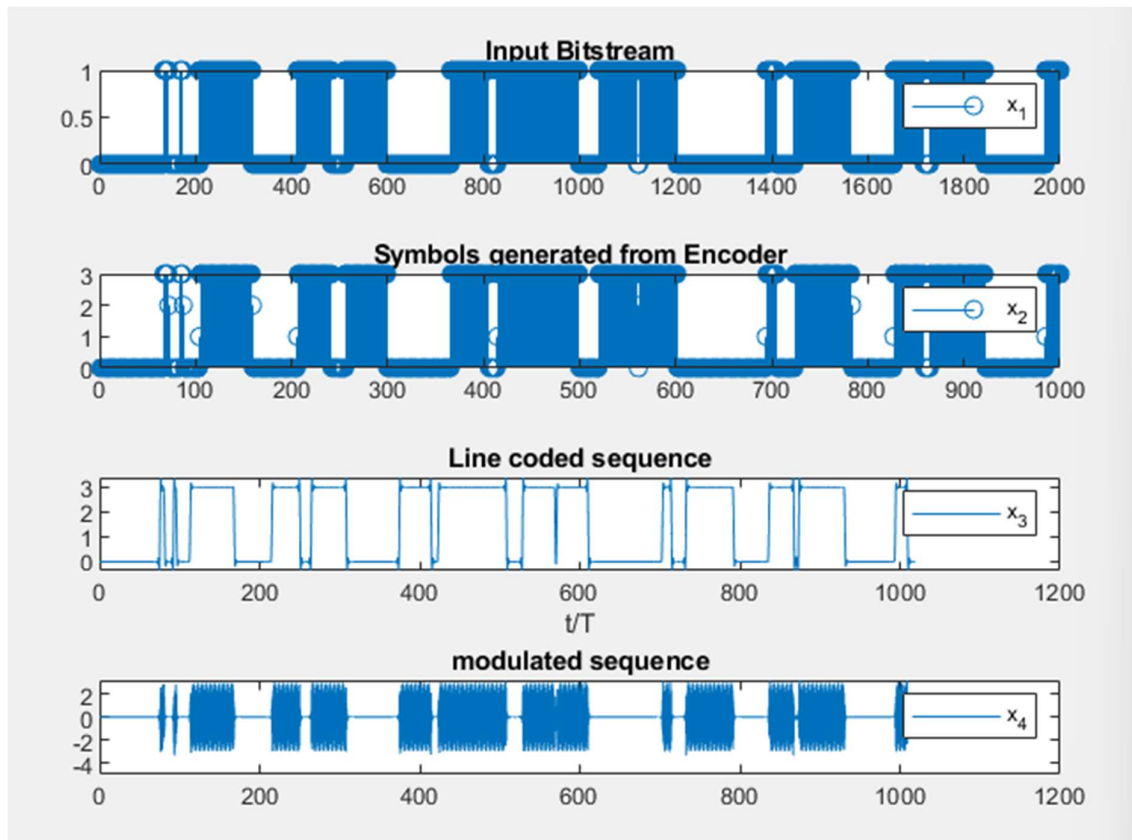
- $\sigma = 0.05$

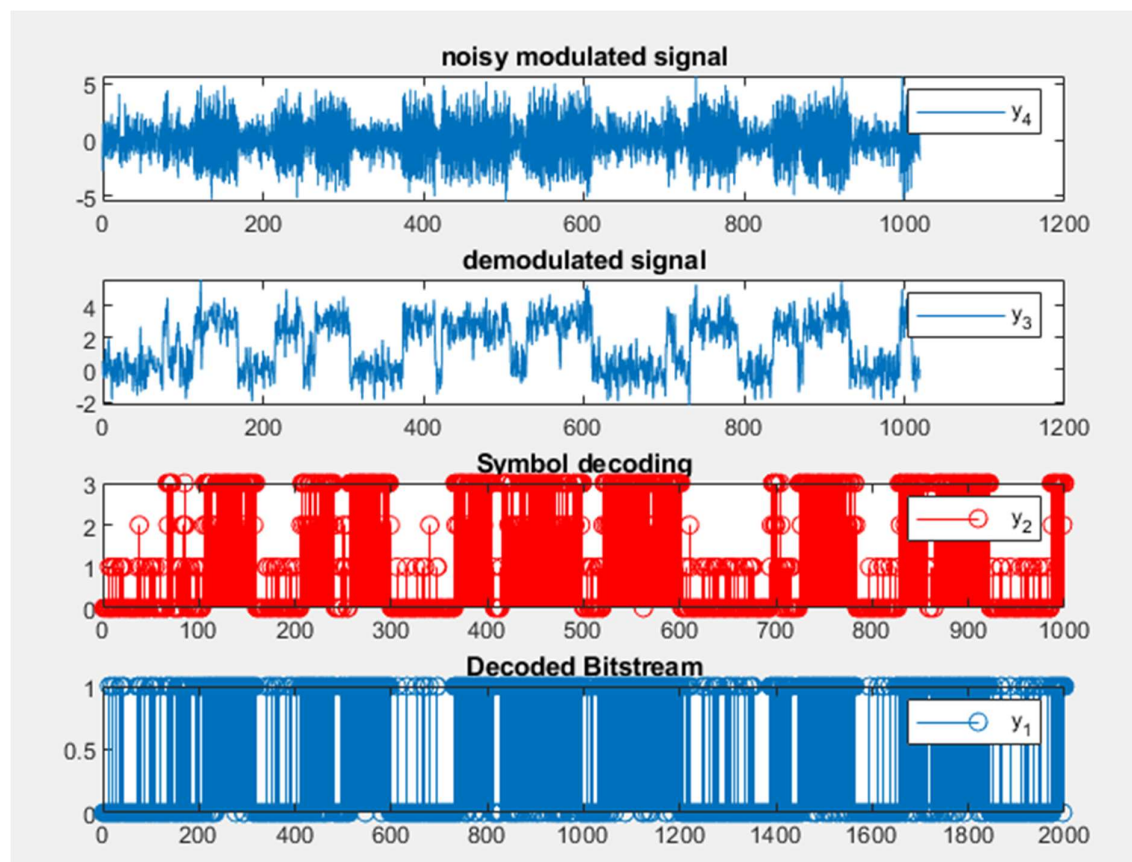




Raised Cosine Pulse Memoryless channel

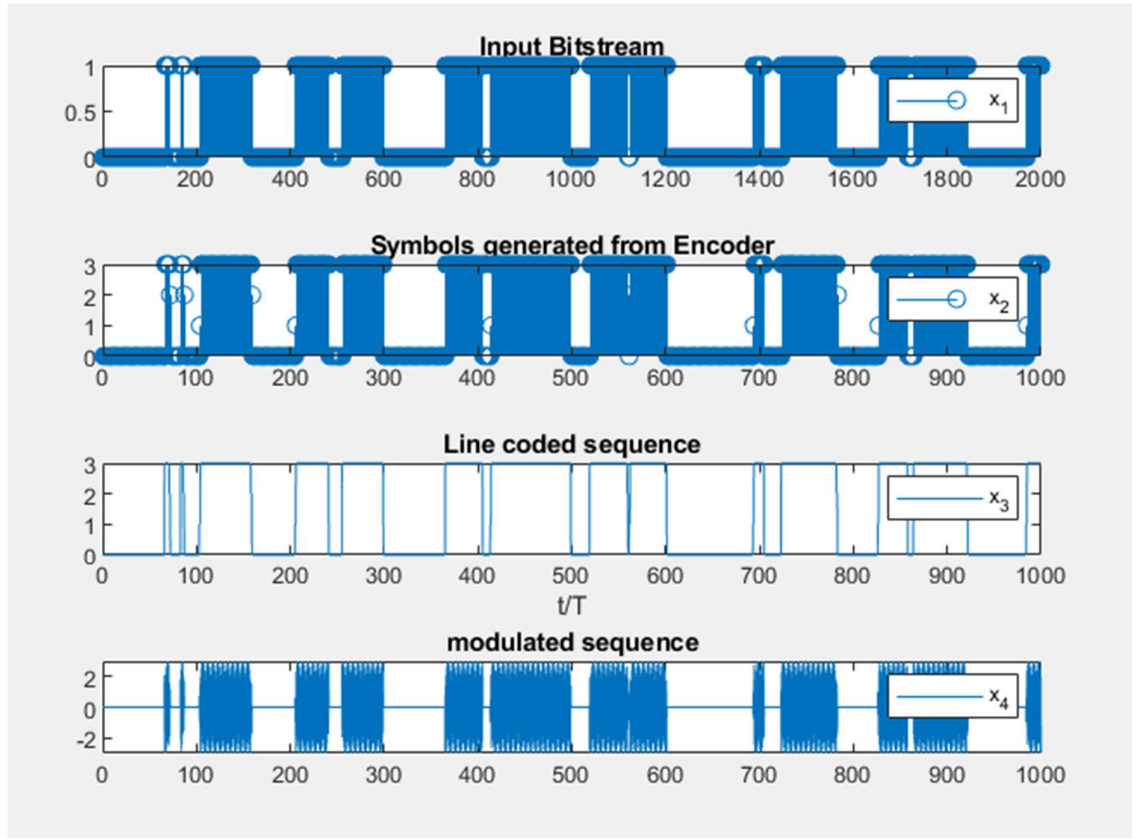
- $\sigma = 1$

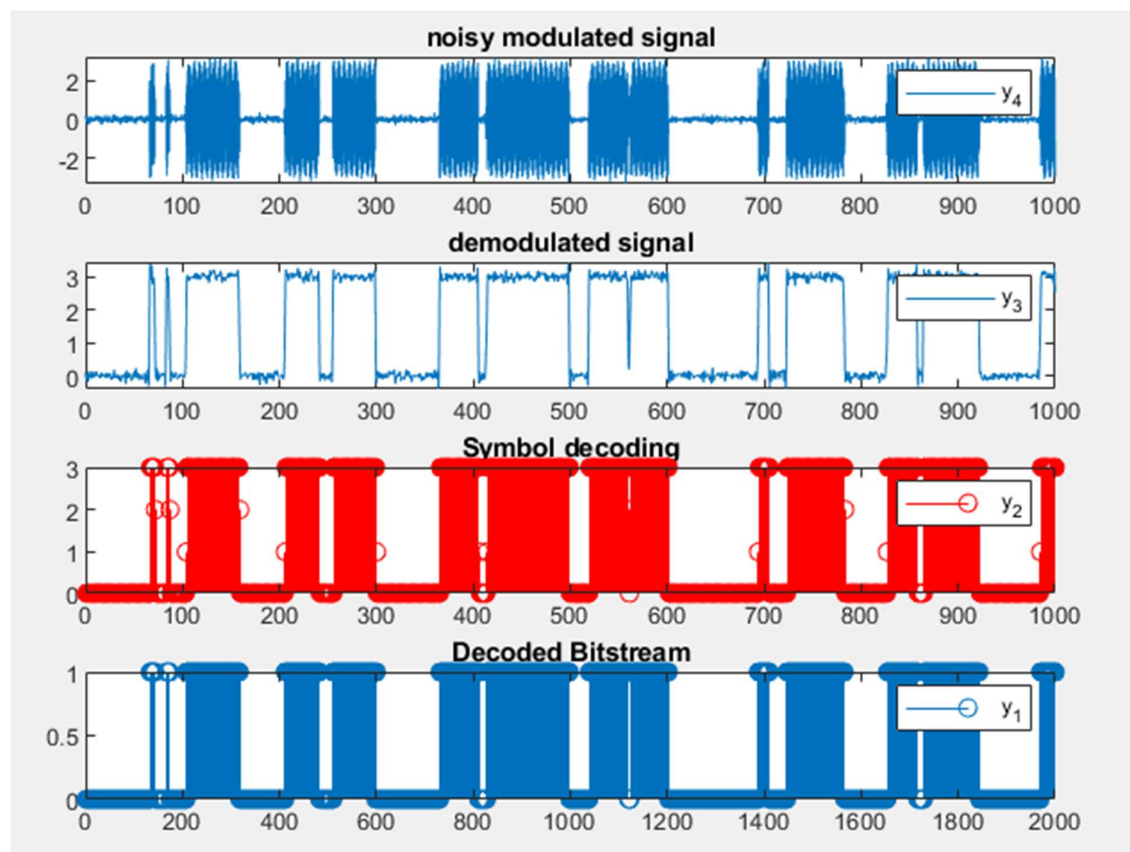




Rectangular Pulse Memoryless channel

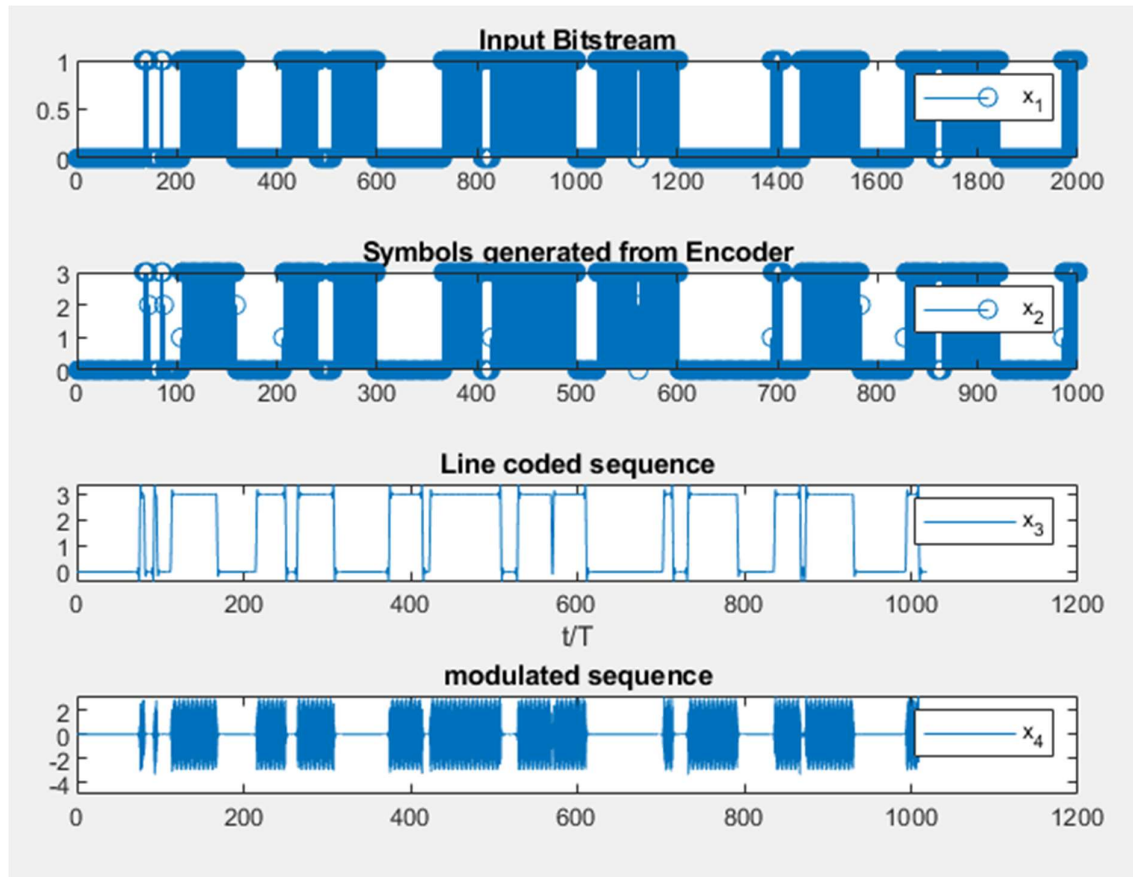
- $\sigma = 0.01$

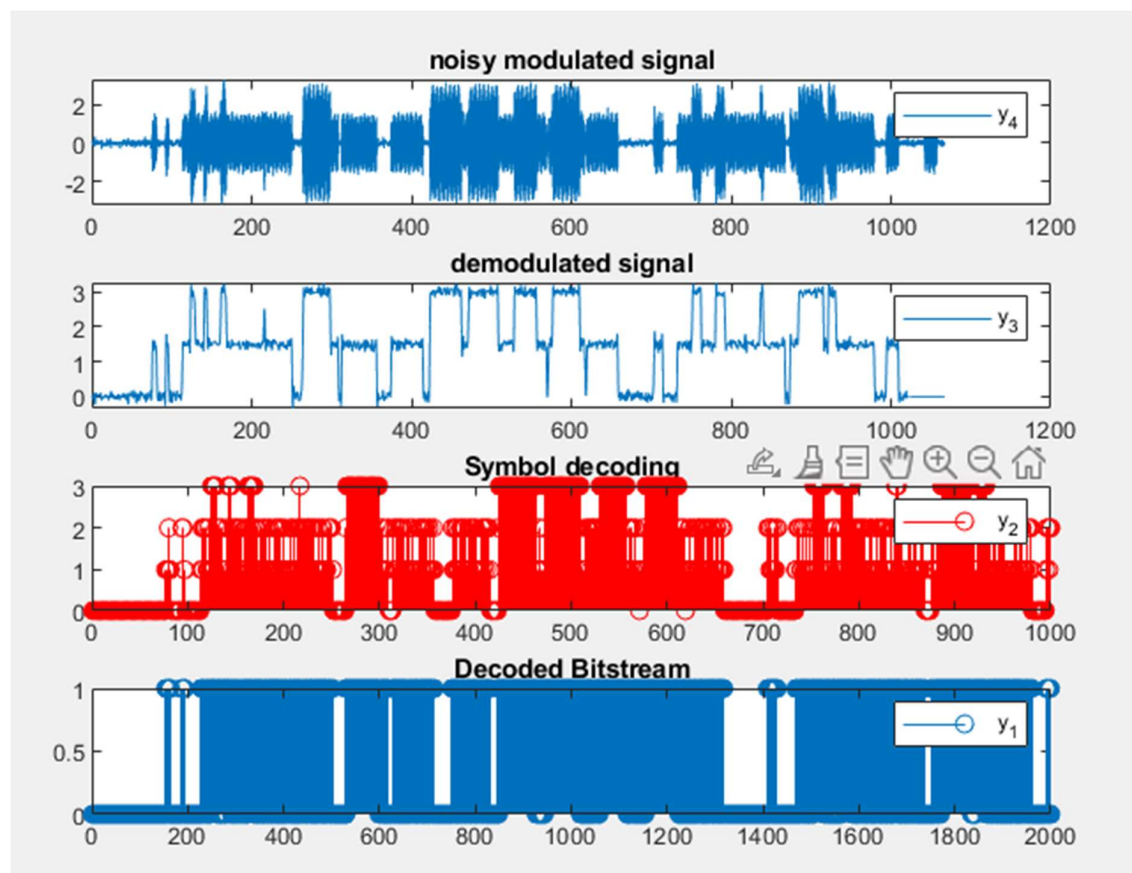




Raised Cosine Pulse with memory channel

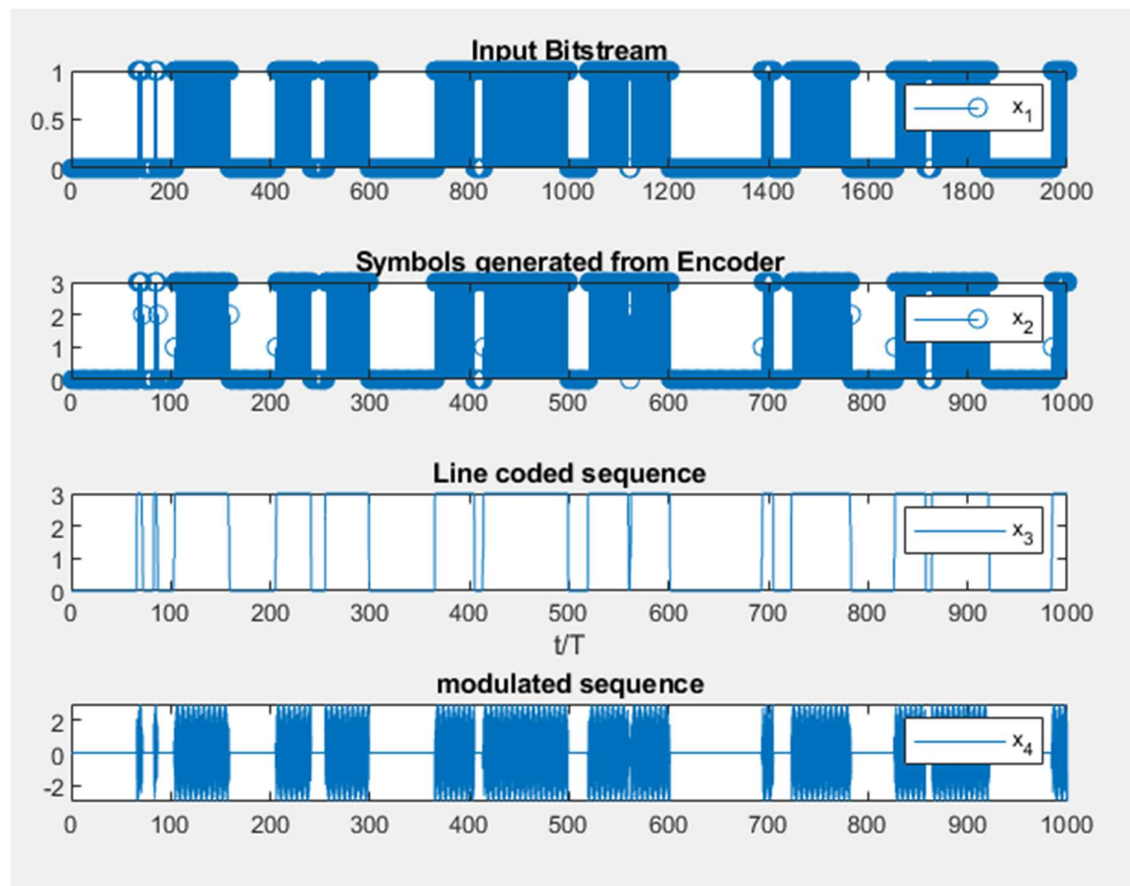
- $\sigma = 0.01$, $a = 0.5$, $b = 4$

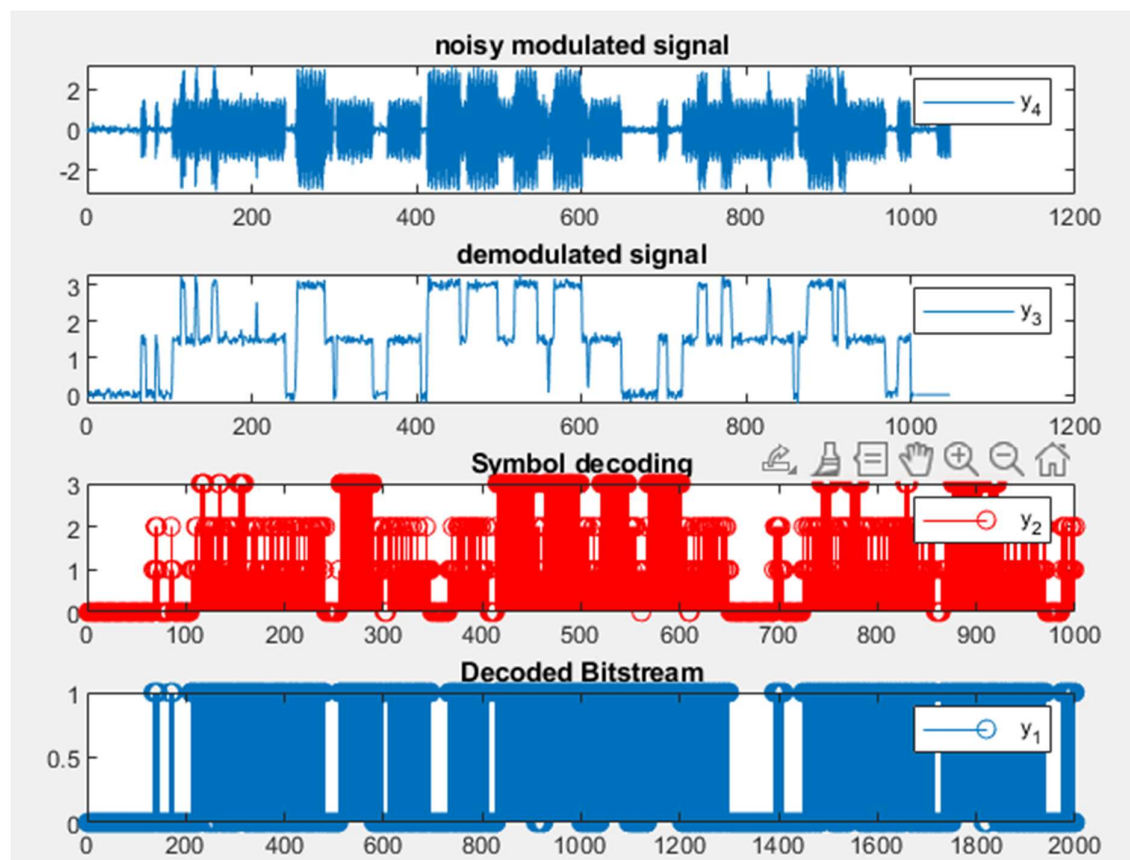




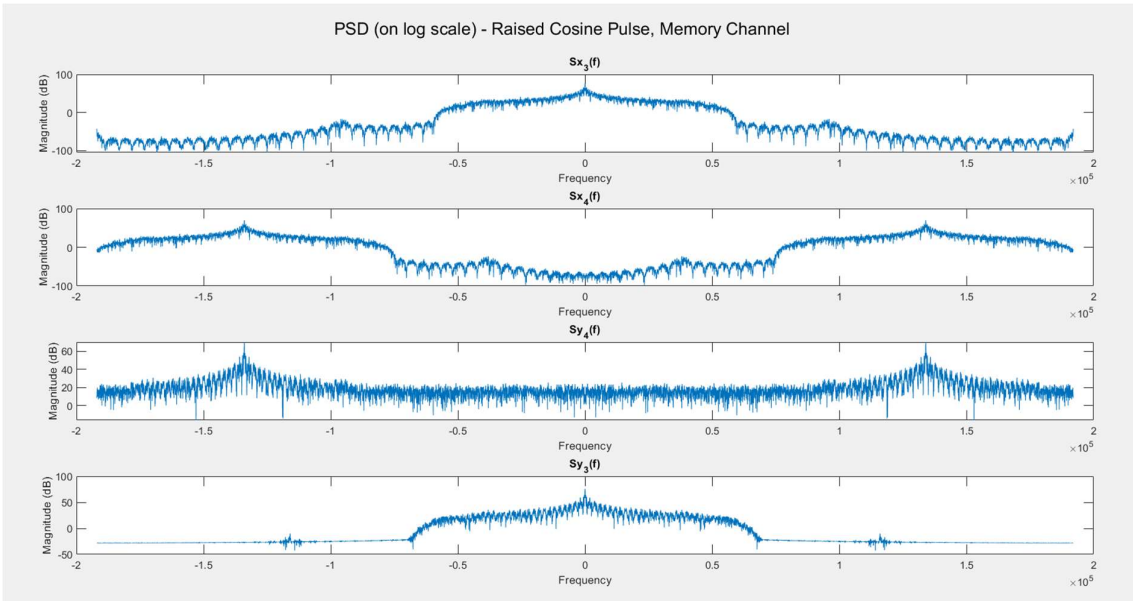
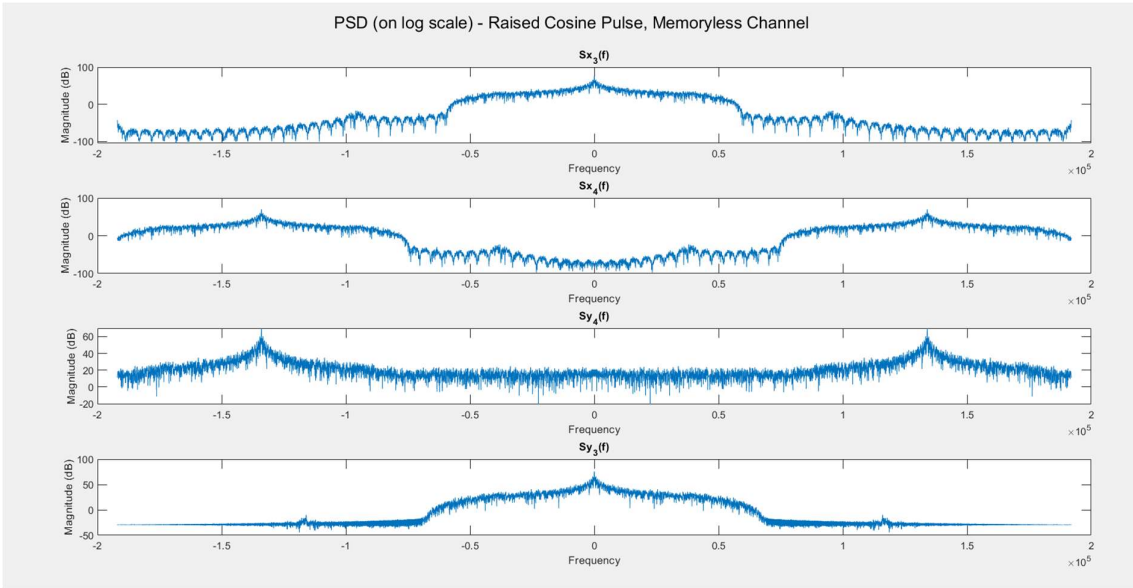
Rectangular Pulse with memory channel

- $\sigma = 0.01$, $a = 0.5$, $b = 4$

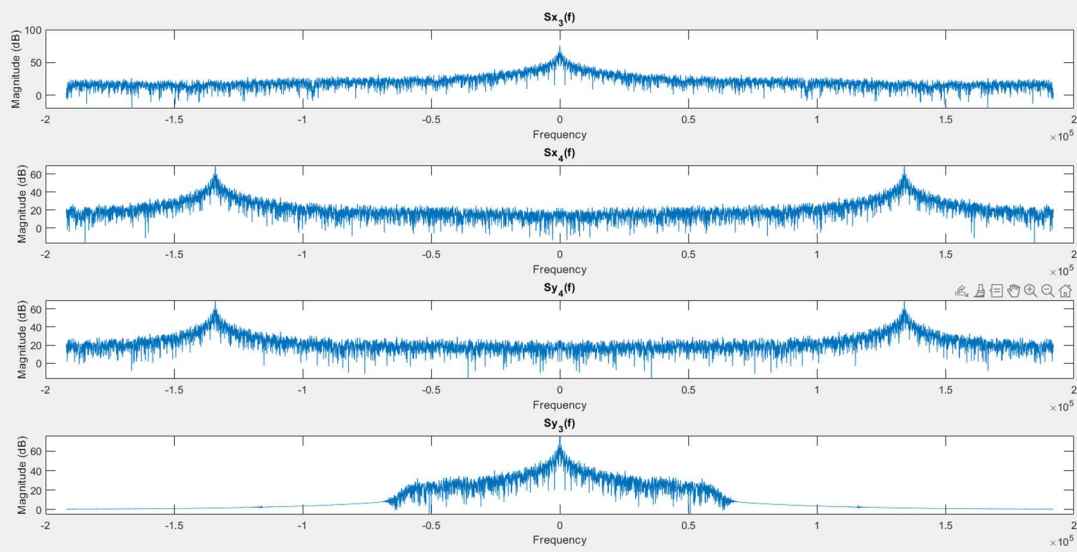




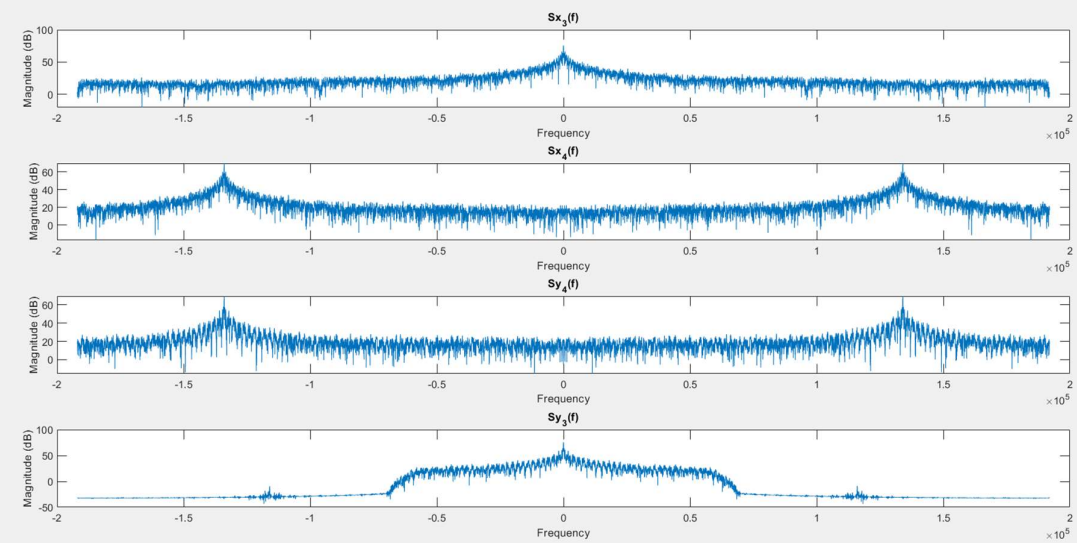
PSD's



PSD (on log scale) - Rectangular Pulse, Memoryless Channel



PSD (on log scale) - Rectangular Pulse, Memory Channel



Power Spectral Density is defined as the power contained per unit Hz of a signal's frequency spectrum. It can be calculated using the fourier transform of the autocorrelation of a time domain signal.

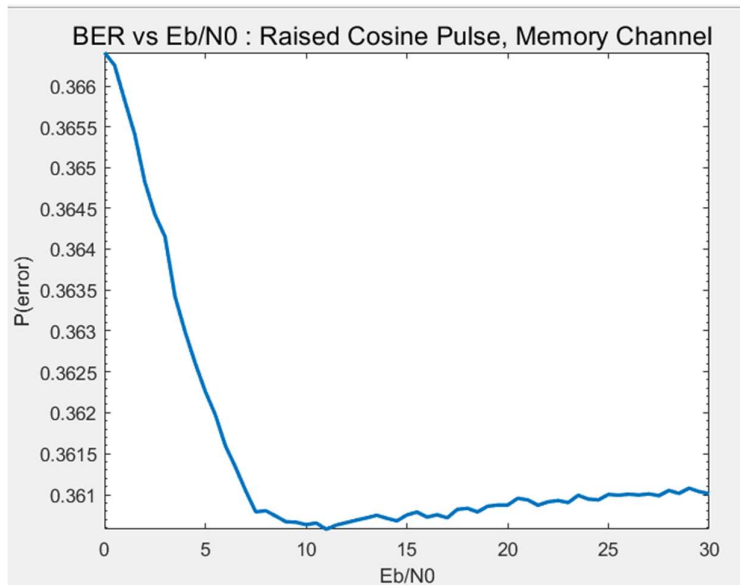
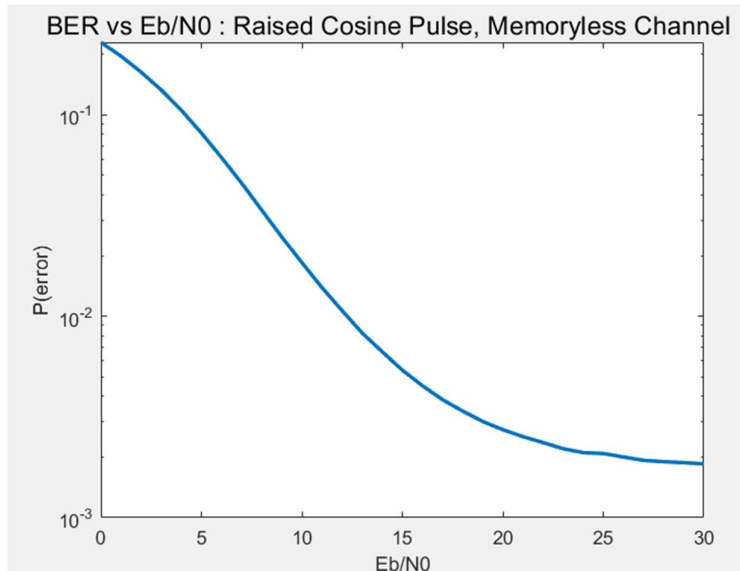
Thus, to calculate the psd's, we obtain the autocorrelation using 'xcorr' function/. Then we take the fourier transform of the autocorrelation using 'fft' function.

The following observations are made on the psd values :

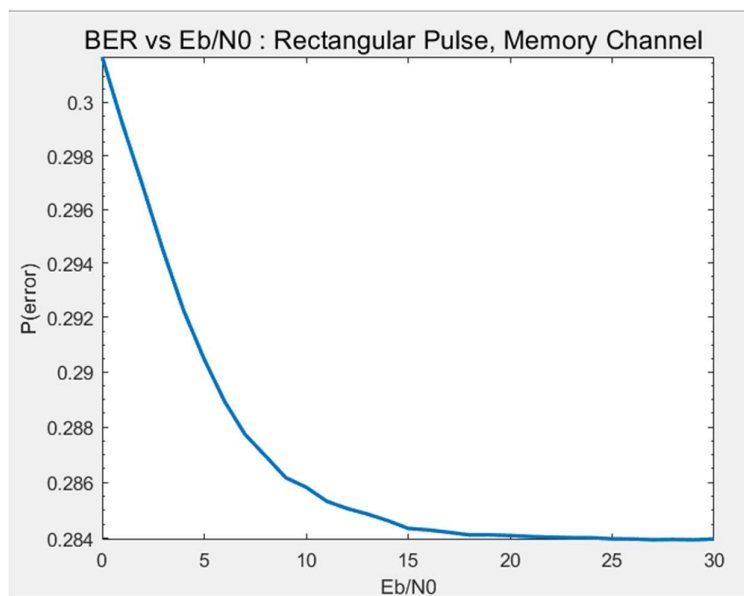
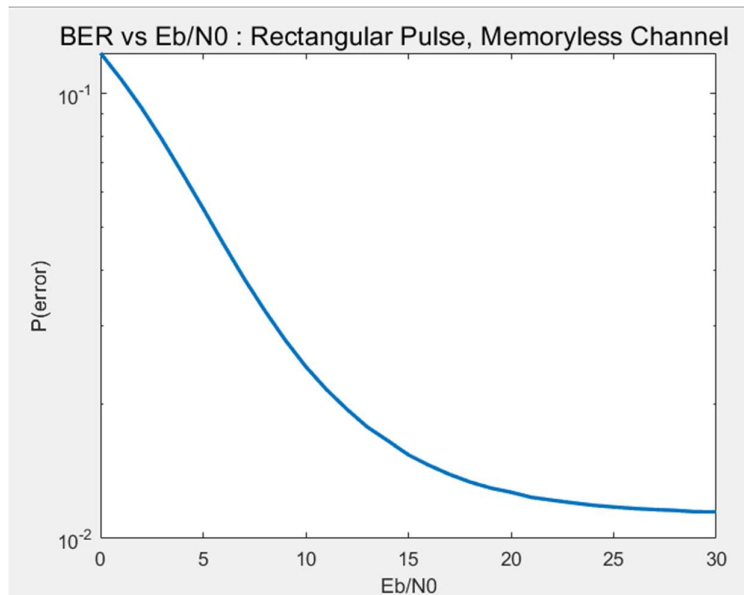
- Since AWGN has a constant PSD, the output PSD of the channel is largely of the same shape as the channel input, but records different values on the y axis due to the addition of the constant noise psd.
- The channel output when passed through demodulator observes a sharp drop in the psd. This drop is beyond the cutoff values of the lowpass filter used in the demodulator. Hence, since the filter allows only low frequencies to pass, the power of the higher frequencies is attenuated as observed in the psd plots.
- The modulated signal's psd observes two peaks at $+f_c$ and $-f_c$, in accordance with the carrier frequencies.
- Since the baseband signal does not have any major high frequency components, the output psd decreases as the frequency increases.
- The output of the memory channel has more disturbances than the channels without memory, representing the effect of memory on the output psd.

Plot: Bit Error Rate vs Eb/No ratio

Adding noise according to the ratio of E_b/N_0 ratio, showcases the effect of noise on Probability of Error. So as E_b/N_0 increases, BER decreases, for all the plots obtained



It can be clearly observed that in case of memory AWGN channel, the error increases significantly as compared to memoryless AWGN channel.



The similar observation could be seen here for rectangular pulse for memoryless and memory AWGN channel.

Now comparing raised cosine, and rectangular pulse, it can be inferred that for low E_b/N_0 , rectangular pulse gives less BER, while for higher ratio, the raised cosine pulse gives effectively less BER. Thus, raised cosine pulse is effective for low power transmission.

Comparing it with the ideal BER, which can be shown for Gray Code gives

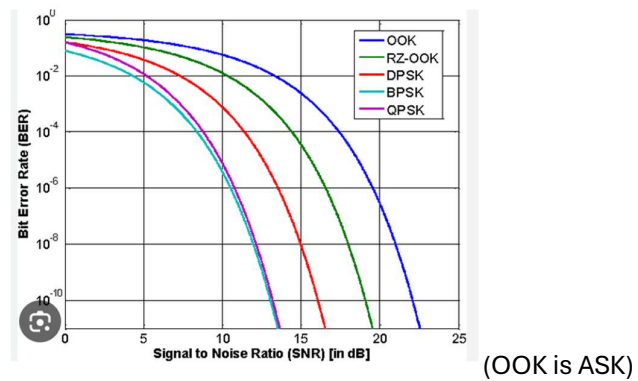
$$Q(\sqrt{4E_b/5N_0}).$$

As it is also not gray Code, and also we implemented memory channel, BER plot would be comparatively worse than this one.

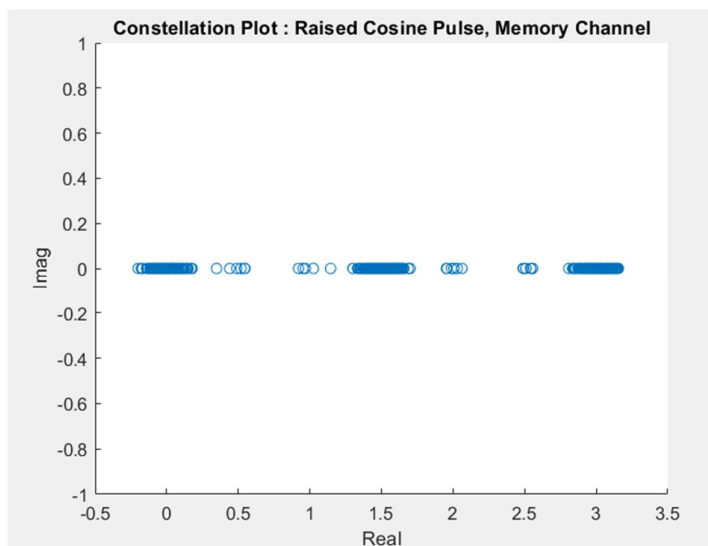
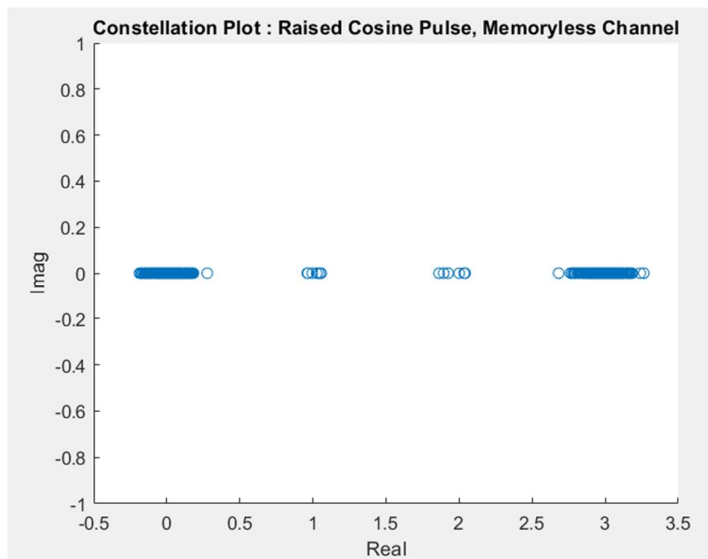
Upon listening the audio files and also the plots, it can be said that the effective scheme is :

1. Raised Cosine + Memoryless AWGN channel
2. Raised Cosine + Memory AWGN channel
3. Rectangular + Memoryless AWGN channel
4. Rectangular + Memoryless AWGN channel

So comparing with other schemes, ASK is equivalent to 16QAM, but worse than other schemes such as FSK, BPSK, QPSK.



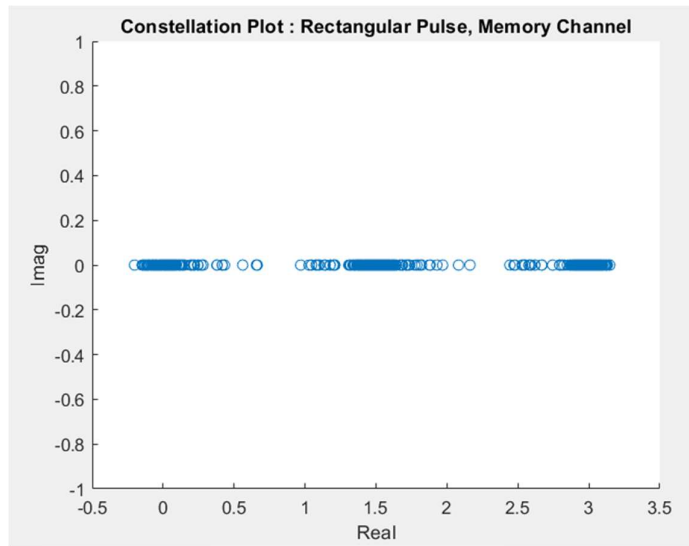
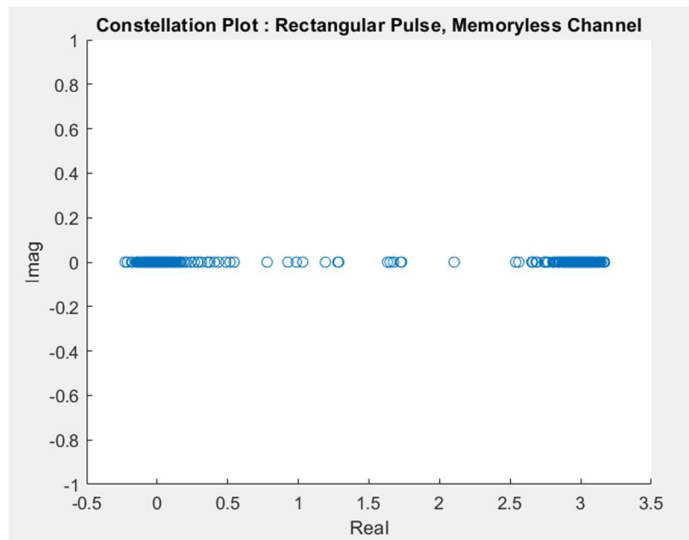
Constellations Plots:



(Did for less no. of points to showcase difference)

As per discussion of BER for different cases, we could see the corresponding scatter of points, and the effect of noise on it.

For Raised Cosine, the memory AWGN channel has more noise, and thus more scattering in the constellation.



Similar observation is seen for Rectangular Pulse.

Further comparing Raised Cosine to rectangular, Raised Cosine shows less deviation of points (less BER).

Thus, the discussion of order of BER as discussed in BER plot is also showcased by these constellations.

