B.M.S. COLLEGE OF ENGINEERING

(Autonomous College Affiliated to Visvesvaraya Technological University, Belgaum)

Bull Temple Road, Basavanagudi, Bangalore-560019

REPORT

ON

"OPEN ENDED EXPERIMENTS"

Submitted in partial fulfilment of the requirements for the partial completion of
COMMUNICATION THEORY 2
(16EC6DCCT2)
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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

OPEN-ENDED EXPERIMENTS

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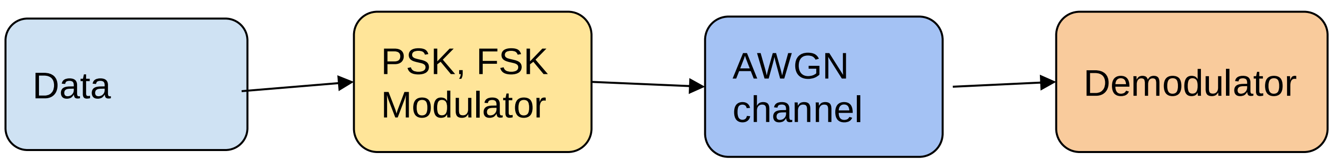
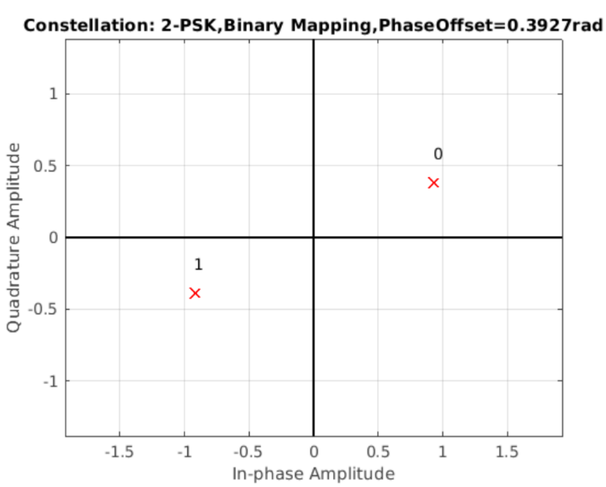
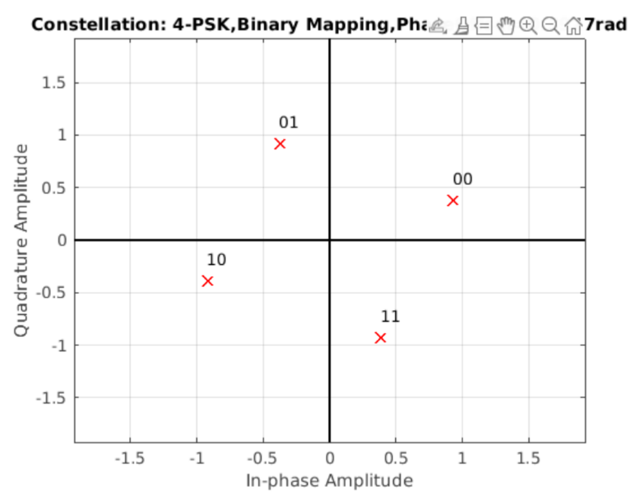
1. **Demonstrate BER performance of a digital communication system employing BPSK, QPSK, FSK modulation scheme you have studied and show the constellation plot. . Assume AWGN channel.**

Block diagram:



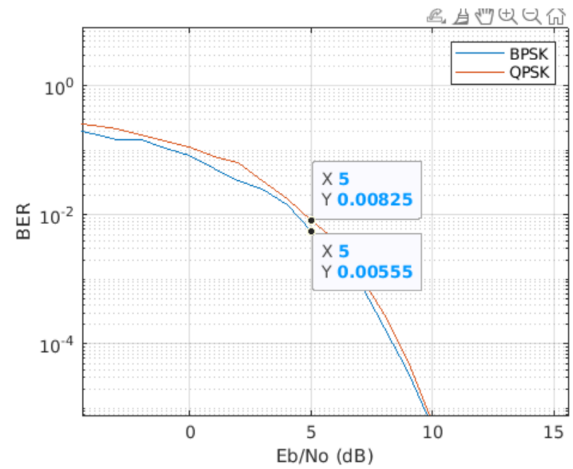




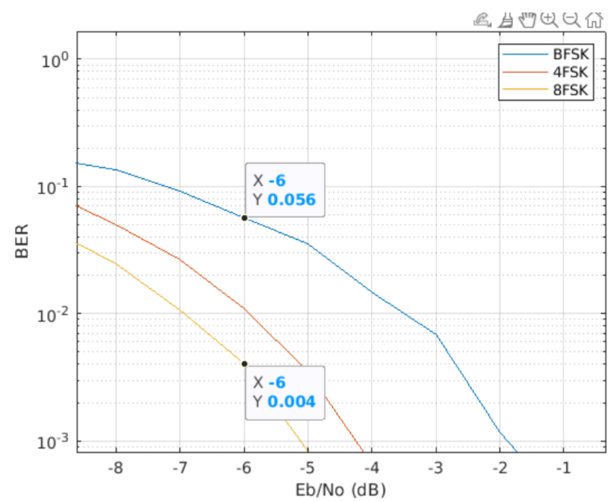
The above figures show constellation plot for 2-PSK and 4-PSK. Binary mapping is used for respective modulations with default phase difference of 0.3927 rad.

Observations:

- It is observed that for the same ratio of bit-energy to noise power spectral density the bit error rate of QPSK is more than the BPSK system.
- We can also say that to achieve a given bit-error rate the Eb/No ratio for QPSK is greater than BPSK system.



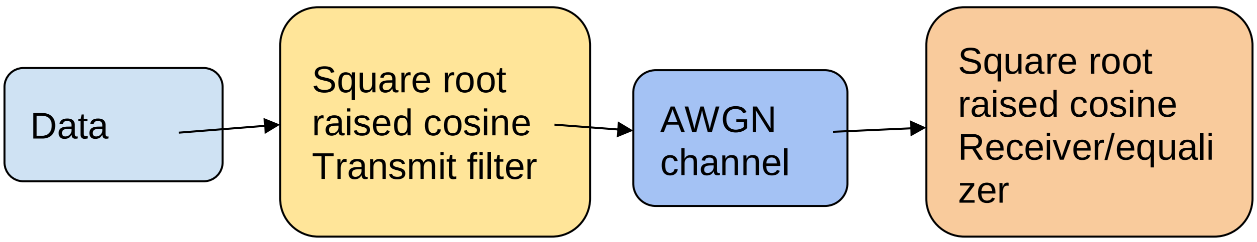
- This figure shows bit-error rate for FSK modulation systems.
- We can observe that as modulation order increases signal to noise ratio needed to achieve the same bit error rate decreases.
- Also all the above systems were simulated for signal to noise ratio varying from -10 dB to 10 dB.



Results:

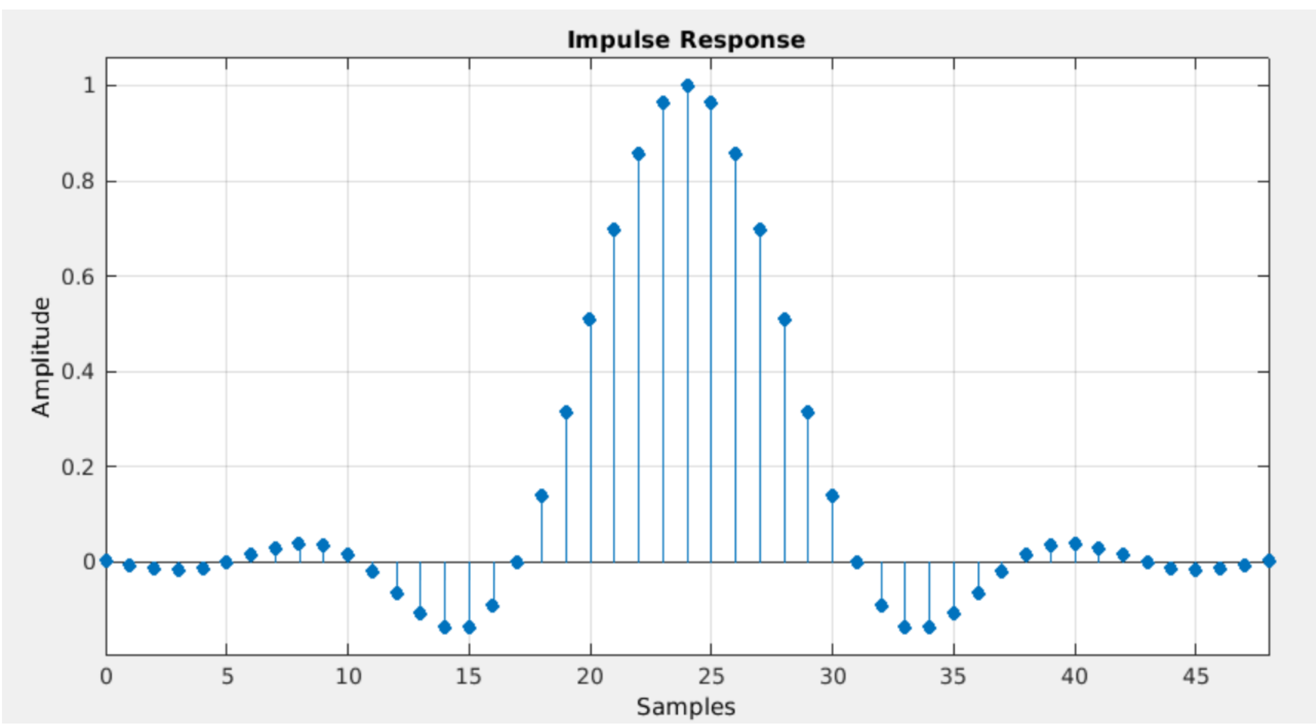
- BER performance of a digital communication system employing BPSK, QPSK, and FSK modulation scheme is studied along with constellation plot.
- As modulation order increases, to achieve a given BER signal to noise ratio decreases for FSK and increases for PSK modulation.

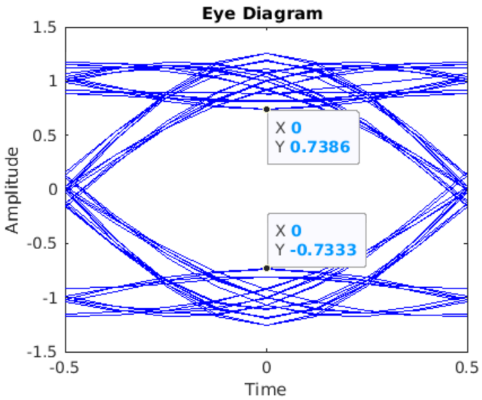
2. **Consider the digital baseband transmission through a bandlimited channel. Design a suitable equalizer to compensate for this channel transfer function in the receiver. Show the eye-pattern prior to and after including this block.**

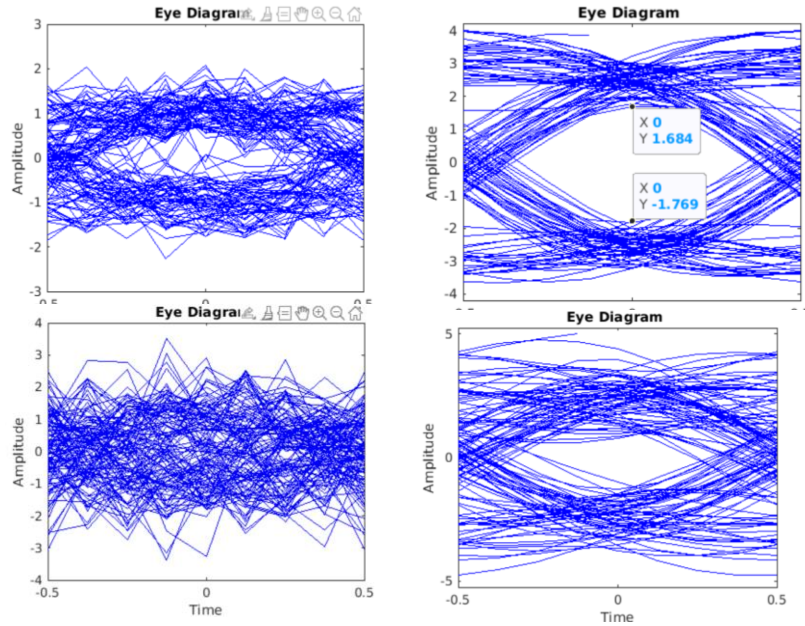


Filter specification for the above experiment:

Transmit and receive filters have roll-off factor of 0.5 with a filter span in duration 6 and upsampling factor of 8. The impulse response is as shown above. The filter used at transmitter and receiver is a square root raised cosine filter.

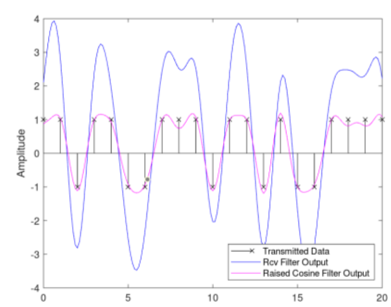






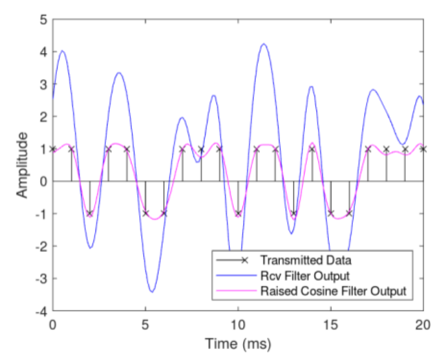
Observations:

- First diagram is the eye-diagram of the data obtained from output of the transmit filter.
- The middle column shows the eye diagram of the data obtained from the channel, and the last column shows the output after applying filter equalizer after the respective channel.
- First row is the output when channel SNR is 5 dB, second row is the output when SNR is -1 dB.



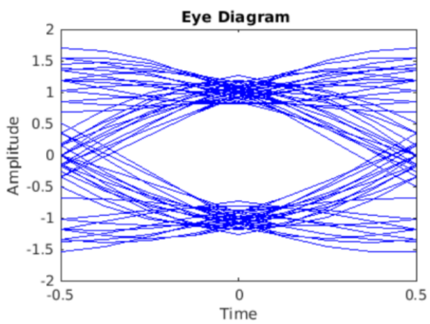
We know that the width of the eye opening defines the time interval over which the received signal can be sampled without ISI.

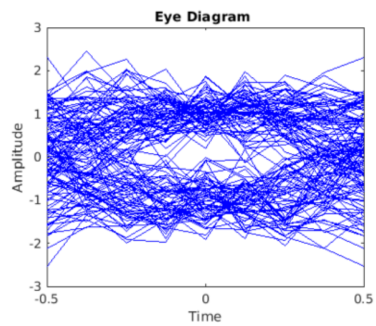
- The width of the eye opening for the channel's output in the first column is very less. However, after applying a receiver filter significant improvement is seen.
- As the SNR decreases the eye opening is significantly less when compared to the 5 dB plots.
- This implies that the probability of error in detection of symbols decreases as receiver-equalizer filter is applied.

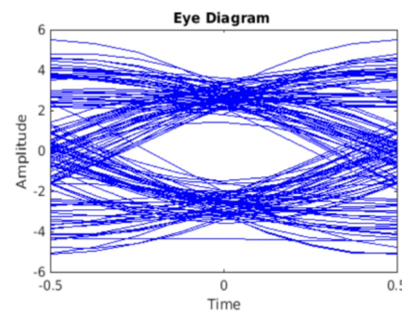


The following figures show the response of the filters when roll of factor is decreased to 0.2. The following observation can be made when compared with the above filter.

- Comparatively the width of the eye opening decreases. This implies that the time interval over which the signal can be sampled also decreases.



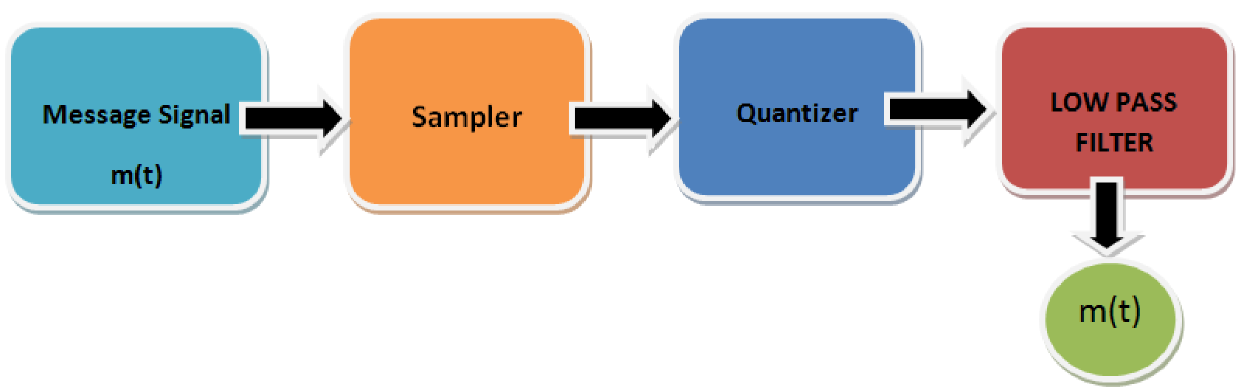




Results: Effects of parameters of the equalizer filter is studied and analysis of eye diagram is done.

3. **Demonstrate sampling, quantization and reconstruction of i) Speech Signal or ii) Audio Signal or iii) Image in MATLAB**

Block Diagram:

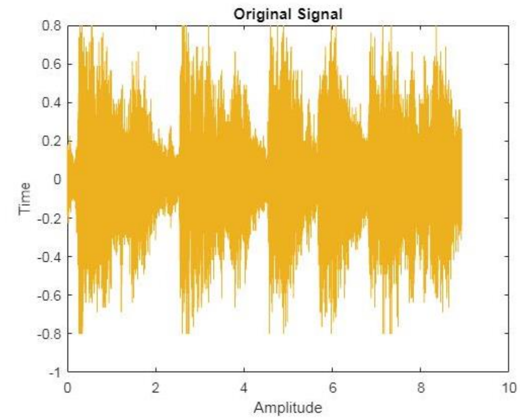


Consider the Original audio signal:

Signal sampled at a rate = 8192

Total number of samples = 73113

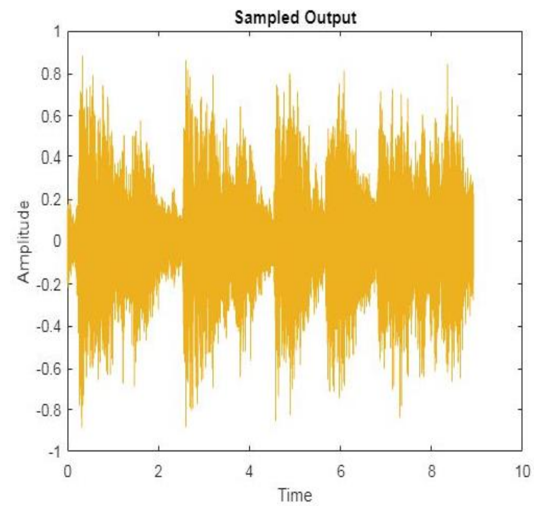
BitsperSample = 16



Sampled Signal:

Input sample frequency rate=4081Hz

Number of quantized levels = 12



Observations:

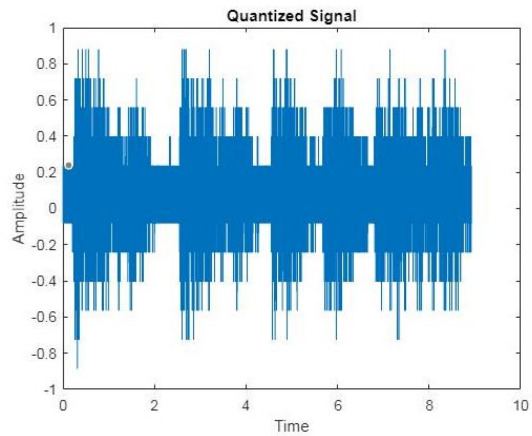
- Sampling rate Fs, is at least twice that of the highest frequency of interest in analog signal. Specifically, for having spectral content extending up to B Hz, we choose **Fs=1/ Ts> 2B** in forming the sequence of samples.
- If a continuous function only contains frequencies within a bandwidth, B Hertz, it is completely determined by its value at a series of points spaced less than (1/2B) seconds apart.
- Sampling process produces frequencies at multiples of the sampling frequency *plus and minus* the sidebands comprising the original signal. Now if the signal frequency is more than half the sampling frequency we will get overlapping called aliasing of signal and sidebands and recovery of the signal by a simple filter is not possible.
- MATLAB function resample is used for sampling. The resample function uses a filter when it performs rate conversion. This filtering is sensitive to large transients in the signal.

- Resample allows us to upsample by an integral factor, p and subsequently decimate by another integral factor, q. In this way you can resample to a rational multiple (p / q) of the original sample rate.

Quantization:

Observations:

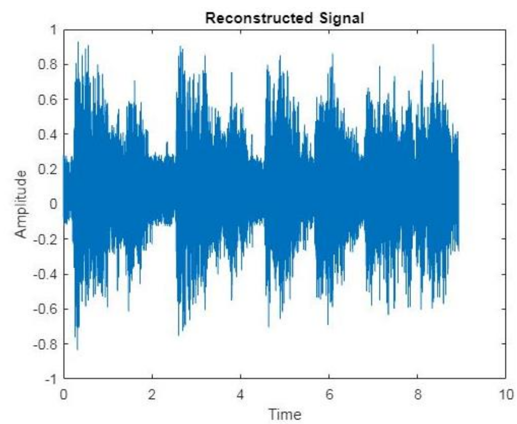
- Quantization is representing the sampled values of the amplitude by a finite set of levels, which means converting a continuous-amplitude sample into a discrete-time signal.
- The quality of a Quantizer output depends upon the number of quantization levels used.
- In music, the signals keep changing continuously, where a regularity is not found in errors. Such errors create a wideband noise.
- To decrease distortion, increase the number of quantized levels by increasing the number of bits.



Reconstructed Signal:

Observations:

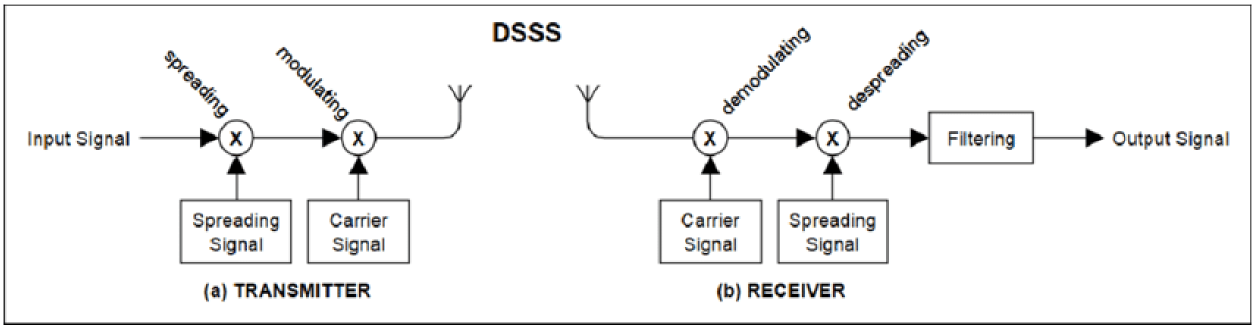
- The LPF filter is used to recover the useful frequency components of the original transmitted signal.
- The LPF makes sure only the information signal within the bandwidth of interest is captured.
- It is used in order to remove the signal noise and improve the ratio SNR.



- Let w_m be the maximum frequency component of the original signal. w_c be the cut-off frequency of Lowpass filter and w_s be the fundamental angular frequency of periodic impulse train. Then for reconstruction of original signal, the necessary condition is: $\mathbf{w_m < w_c < w_s - w_m}$

4. DEMONSTRATING THE USE OF DS SPREAD SPECTRUM FOR CDMA SYSTEM

Block diagram:



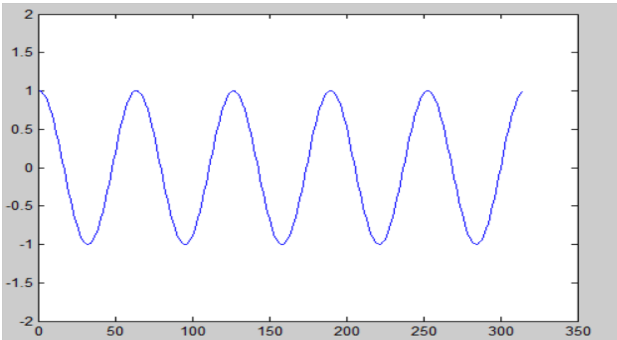
Why dsss?

- It has best discrimination against multipath signals.
- It avoids intentional **interference** such as jamming effectively.
- Most suitable technique for cdma

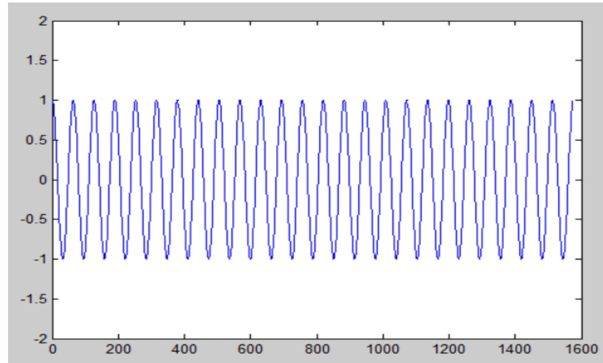
How dsss works:

- The each bit in the data initially is xor-ed with pseudo random sequence for spreading
- Next it is modulated along with the addition of multiple user's data and sent over the channel
- Then at the receiver end , only if the pseudo random sequence is correct for each user , the original data is retrieved

Modulated dsss signal without spreading



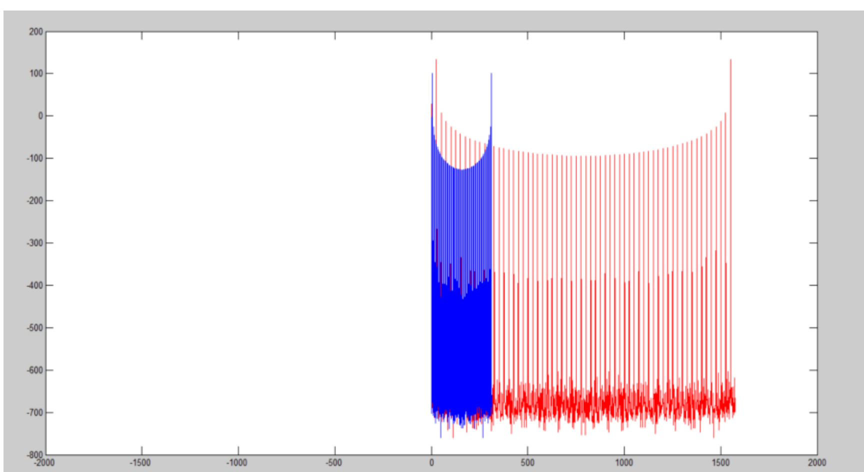
Modulated dsss signal with spreading



Observations:

- We can observe from above figures that by xor ing a pseudo random sequence with the each bit of data, the signal is spreaded
- Also the power of the whole signal will be spreaded across, compared to the original signal
- It is observed that due to spreading the bit rate is increased.

Power density plot comparing both spreaded (red) and non spreaded (blue) one:



Observations:

By observing power density spreading, we can conclude that

- Bands of signals occupy a wide range of frequencies.
- Power density is very low.
- Energy is widespread.

Results:

- Successfully transmitted the data and retrieved the same at receiving by using the dsss technique
- Compared both spreaded and non spreaded data by plotting power density spectrum