1 Bandlimited channels and intersymbol interference

In the following tasks you can assume that timing recovery and frame synchronization has been perfectly achieved at the receiver. Assume in the following tasks that signals are sampled at a frequency f_s.

Baseband channel modelling

1. A baseband channel can be modelled as an LTI low-pass filter with a cutoff frequency of f_c. The channel can be modelled as introducing attenuation and ISI. Your task is to write a Matlab function which can model a baseband channel.
   a) Write a Matlab *channel* function that takes as input a sampled signal $x[n]$ and produces an output $y[n]$.
   b) Modify your channel function to low-pass filter the input $x[n]$; the low pass filter should have a gain g in the passband as input, and a passband cutoff frequency of f_c.
2. Plot the frequency and phase response of the channel that you have modelled for your choice of channel parameters.

Intersymbol interference - baseband channels

1. We will study and visualize the effect of intersymbol interference (ISI) using the baseband channel model developed above.
2. Consider the baseband channel modelled as a low pass filter. We will investigate what happens to a baseband waveform as it passes through a low pass or baseband channel.
3. We will use a sampling frequency of 100 Hz for this task.
4. Use a low pass filter with a passband edge of 10 Hz with a passband gain of 1 to model the channel. Obtain and plot the magnitude spectrum of the channel.
5. Generate a baseband BPSK signal $b(t)$ corresponding to a random sequence of 100 bits for $T_b = 0.1s$.
6. Plot the eye diagram of this baseband BPSK signal. You should observe that this is the eye diagram of a signal without ISI.
7. Pass the baseband BPSK signal through the channel model and plot the eye diagram of the channel output. Comment on what you have observed.
8. Plot the eye diagrams for $T_b = 0.05$ and $T_b = 0.2$. Comment on your observations.
9. Suppose the channel output is passed through a matched filter for the case of $T_b = 0.1$. Plot the eye diagram of the matched-filtered received signal. What differences do you observe?

Pulse shaping

1. We will use a sampling frequency of 100 Hz for this task.
2. Generate a baseband BPSK signal $b_r(t)$ corresponding to a random sequence of 1000 bits for $T_b = 0.1s$. Note that this baseband BPSK signal should be generated using the rectangular pulse shape.

3. Generate a baseband BPSK signal $b_s(t)$ corresponding to the same random sequence of bits used above, but using the sinc pulse shape.

$$AT_b \frac{sin\left(\pi \frac{t}{T_b}\right)}{t}$$

Note that the sinc pulse shape needs to be truncated to duration of $5T_b$ so that the truncated pulse shape is symmetric.

4. Generate a baseband BPSK signal $b_c(t)$ corresponding to the same random sequence of bits used above, but using the raised cosine pulse shape.

$$AT_b \frac{sin\left(\pi \frac{t}{T_b}\right)}{t} \frac{cos\left(\pi\alpha \frac{t}{T_b}\right)}{\left(1 - \left(2\alpha \frac{t}{Tb}\right)^2\right)}.$$

Note that the raised cosine pulse shape needs to be truncated to duration of $5T_b$ so that the truncated pulse shape is symmetric.

5. Plot the power spectral densities of the signals $b_r(t), b_s(t)$, and $b_c(t)$. Comment on the differences in the three spectra.

6. Plot the eye diagrams of $b_r(t), b_s(t)$, and $b_c(t)$. Comment on the differences and main features of each eye diagram.

7. Plot the spectra and eye diagrams of $b_c(t)$ for different values of α. What do you observe?

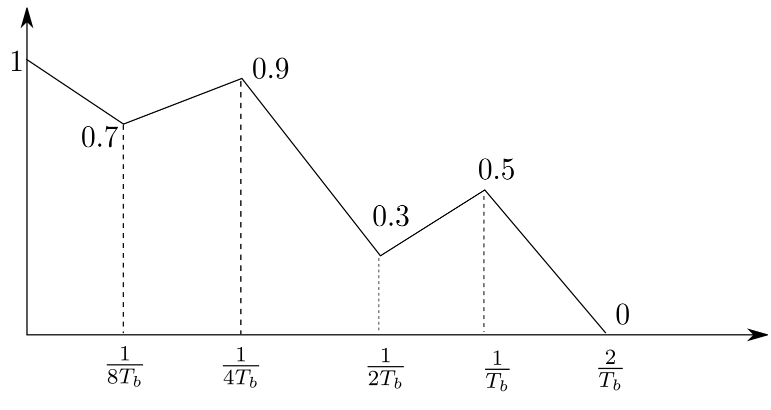
8. For each of the signals $b_r(t), b_s(t), b_c(t)$, and $b_{rc}(t)$ obtain the channel output when the respective baseband signals are sent through a baseband channel. Use a low pass filter with a passband edge of 10 Hz with a passband gain of 1 to model the channel.

9. Obtain the PSD and eye diagram of the channel output in each of the cases. Compare the bandwidth and eye width properties for each of the pulse shapes for $T_b = 0.1, 0.05$ and 0.2.

Channel inversion

1. In this task, you will consider a baseband channel which has a non-flat response in the passband.

2. First, you have to modify the channel model that you have written to model a baseband channel which has the frequency response shown in Figure 2 (approximately and with linear phase response).



3. We will use a sampling frequency of 100 Hz for this task.

4. Generate a baseband BPSK signal using rectangular pulse corresponding to a random sequence of 100 bits with $T_b = 0.1$.

5. Obtain the eye diagram of the channel output when the above baseband signal is passed through the channel.

6. In order to undo the effect of the channel, whose response $H(f)$ is known to us, we can use a equalizer which undoes the effect of the channel by channel inversion, i.e., the equalizer is a receive/transmit filter which has a response $G(f)$ such that $H(f)G(f) = 1$ over the effective bandwidth of the signal. Design a equalizer using channel inversion.

7. Modify your receiver to process the received signal out of the channel using the equalizer.

8. Visualize the output from the equalizer using an eye diagram. What do you observe?

Linear equalizer - Tapped delay line

1. Assume that the baseband communication channel has a response $H(f)$ as in the above tasks. Let $h(t)$ be the corresponding time domain impulse response.

2. We note that the effect of ISI just needs to reduced at the sampling instants.

3. We assume that the output of this communication channel is fed into an equalizer with impulse response $q(t)$, so that the effective pulse shape from the input to the channel to the output is $g(t) = p(t) \star h(t) \star q(t)$ (here $\star$ denotes convolution and $p(t)$ is the pulse shape used).

4. Assume that in order to reduce the effect of ISI at sampling instants we require that

$$\begin{aligned} g(0) &= 1, \\ g(nT_b) &= 0, \text{ for } n = \pm 1, \pm 2, \pm N; \end{aligned}$$

   where T_b is the sampling period (or bit duration).

5. Design the equalizer impulse response $q(t)$ assuming that $q(t)$ is a tapped delay line filter (with delays of magnitude T_b). Note that the design should therefore specify the values of the tap weights of the filter. Do this design for $N = 2, 5, 10, 50$.

6. Visualize the output of the equalizer using an eye diagram for different values of N.