7.1 Down and Up Sampling LabWork

In this lab, we will learn how a signal can be compressed by the down-sampling process and how it can be reconstructed by the up-sampling process. You need to first load '**LabSample.mat**' to get the original signal **x** and the sampling frequency $\mathrm{F_s}$. After that, you should follow the below procedures:

(a) Find the sampling period $\mathrm{T_s}$ and the time duration of given signal **x**.

(b) Construct the time vector **t** with respect to $\mathrm{T_s}$ and the time duration which is found in (a).

(c) Choose the down(up)-sampling period D as a 2,4, and 8, respectively.

(d) Generate 3 different impulse-trains according to D given in (c).

(e) Obtain 3 different down-sampled signals by using ***dot-product*** between **x** and 3 different impulse-trains in (d).

(f) Calculate the Fourier Transforms of these 4 signals including **x** and plot their frequency spectrums in a single figure by using the subplot(22.) command.

(g) We will now reconstruct our signals. In this procedure, we should interpolate our down-sampled signals for getting closer to the original one. To this end, the interpolation can be done **by simply designing the low-pass filter** considering the frequency spectrum of each down-sampled signal as in (f). (Remember the previous lab!) Set the number of filter coefficient n as 200 and obtain these coefficients for each low-pass filters by using *firpm()* command! (**Note that: you should decide the frequencies and amplitudes of each down-sampled signal!**)

(h) After finding the filter coefficients, you can now use the '**convolution**' operation over the filter coefficients and the down-sampled signals for reconstruction.

(i) You may notice that each reconstructed signal has more components than the original one due to the convolution process. So, you need to remove the unnecessary parts of each signal.

(j) Then you can measure the similarities between each up-sampled signal and the original signal **x** by Mean Squared Error formula which is given below,

$$\mathbf{MSE} = \int (x(t) - x_r(t))^2 dt \approx \frac{1}{N}\sum_{n=1}^{N}(x[n] - x_r[n])^2 = \frac{1}{N}||\mathbf{x} - \mathbf{x_r}||^2$$

where N is the number of samples and $\mathbf{x_r}$ is the reconstructed signal.

(k) Or else, you can just add a new command that is $\mathbf{sound(x_r,F_s)}$ to see the difference.

(l) Make comments about which D is the best? Why?