Add to your Time Domain menu framework the following features:

A. Filtering

1) Implement FIR filters, ask the user for the input signal to be filtered, type of filter he wants (low, high, band pass and band stop) and his specifications, according to it choose the appropriate window w (n), the appropriate infinite impulse response h(n), compute N and then compute the needed coefficients, finally convolve the input signal with the computed coefficients, draw the resulted signal and save the coefficients to text file.

Notes

i) the specification will 1) sampling frequency, 2) cut off frequency (in case of low and high filters) f1 and f2 in case of (band pass and band stop filters) 3) stop attenuation δ_s 4) transition band

ii) don't forget to adjust frequencies using half transition band to suit the window method.

iii) frequencies should be normalized by dividing it by sampling frequency after being taken from user.

iv) don't forget that coefficients are symmetric, and N should be odd (type one)

B. Resampling

Ask the user for the input signal, low pass filter specifications, and the values of M & L where M and L are the decimation and interpolation factors respectively. Your application should consider the following cases:

If M =0 & L ≠ 0 then up sample by L factor and then apply low pass filter.

If M ≠ 0 & L = 0 then apply filter first and thereafter down sample by M factor.

If M ≠ 0 & L ≠ 0 this means, we want to change sample rate by fraction. Thus, first up sample by L factor, apply low pass filter and then down sample by M factor.

If M = 0 & L = 0 then return error message.