ELEC9721: Digital Signal Processing Theory and Applications

Lab 5 Multirate

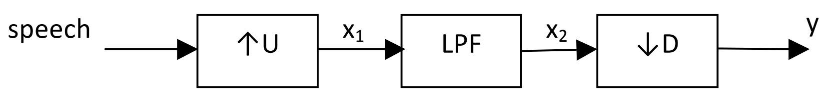
You may not use Matlab's in-built functions downsample, upsample, resample for this lab.

Question 1: show your preparation *2 marks*

Question 2:

The speech signal sampled at 16kHz using in this question is obtained using the following command in Matlab: *speech=audioread('speech.wav')*. ("wavread" if you have an older version of Matlab)

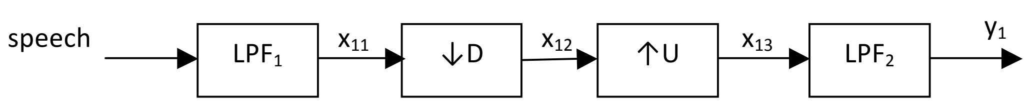
a. This speech signal is passed through a system below to perform resampling. The cut-off frequency of the low pass filter LPF is fc. It is required that the sampling rate of the signal y is 24 kHz.



i. Determine U and D and fc.
ii. Implement the upsampling, low pass filtering and downsampling in Matlab. You can use the *firpm* algorithm to design the LPF; the order of the filter should be N=80.
iii. Also show what happens (play the sound) if the filter is not implemented.
iv. Plot the spectrum of the *speech, x_1, x_2 and y* when the x-axis is in Hz and between zero to half of the sampling rate of the corresponding signal.

4 marks

b. The above speech can be resampled at 24 kHz using another scheme as below:



v. Determine the cut-off frequency (in Hz) of the low pass filters LPF_1 and LPF_2
vi. Implement the resampling process with the new scheme in Matlab
vii. Again, play the sound when the filters are not implemented
viii. Compare the resampled speech signals using two different schemes by observing their spectra and listening them. Which scheme is better and why?

4 marks