EE6133 Experiment-1

Two audio files are available for this experiment:
"music16khz.wav": a piece of music sampled at 16 kHz
"speech8khz.wav": speech sampled at 8kHz

- Downsample each audio signal by 2 *without* any AA filtering. Play back using the new sampling rate.
- Design an equiripple LPF with $\omega_p = 0.45\pi$ and $\omega_s = 0.55\pi$, with appropriate values for δ_p and δ_s. Using this as an AA filter, downsample each audio signal by 2. Play back using the new sampling rate. Compare the two downsampled signals (with and without AA filtering) for each audio input.
- Design an equiripple LPF with $\omega_p = 0.22\pi$ and $\omega_s = 0.28\pi$.
- First upsample each original audio signal by 3. Then apply the LPF with cutoff-$\pi/4$ designed above, and downsample by 4. Play the re-sampled outputs using the new sampling rate. Compare these outputs with the originals.
- First filter each audio original signal with the LPF designed above, and downsample by 4. Then upsample by 3. Finally, interpolate each upsampled signal using an interpolation filter for $L = 3$ with support $[-8, 8]$ designed by any method (including windowing). (If you use "intfilt" of Matlab, set $\alpha = 1$.) Play these outputs and compare them with those of the previous part.