(a) Use the MATLAB function *remez.m* to design a lowpass filter, i.e.,

$$b_l[n] = remez(102, [0\ .5\ .6\ 1], [1\ 1\ 0\ 0]).$$

Plot the log-magnitude of its Fourier transform over the interval $[0, \pi]$, using a 1024-point FFT. In doing this exercise, use MATLAB function *fft.m*.

(b) Normalize $b_l[n]$ by the sum of its values and then design a highpass filter as

$$b_h[n] = \delta[n - 52] - b_l[n].$$

Plot the log-magnitude of the Fourier transform of $b_h[n]$ over the interval $[0, \pi]$, using a 1024-point FFT. In doing this exercise, use MATLAB function *fft.m* and the script *high_low.m* provided in the directory Chap_exercises/chapter2.

(c) Using the MATLAB function *conv.m*, filter the speech waveforms *speech2_10k* and *speech3_10k* with the lowpass and highpass filters you designed in parts (a) and (b).

(d) It is sometimes said that speech possesses redundant information in frequency, i.e., the same information may appear in different frequency bands. Using the MATLAB function *sound.m*, listen to the lowpass- and highpass-filtered speech from part (c) and describe its intelligibility. Comment on the notion of spectral redundancy.

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