Homework 2 (Due: 5/4)

(1) Write a Matlab program that uses the frequency sampling method to design a (2k+1)-point discrete Hilbert transform filter (k is an input parameter and can be any integer).

(25 scores)

The transition band can be assigned to reduce the error (unnecessary to optimize). The frequency response (DTFT of $r[n]$, see pages 103 and 104) and the impulse response of the designed filter should be shown. The Matlab code should be emailed to displab531@gmail.com

(2) Suppose that an IIR filter is $H(z) = \dfrac{2z^3 - 4z^2 - z + 2}{2z^2 - 2z + 1}$

(a) Find its cepstrum.

(b) Convert the IIR filter into the minimum phase filter.

(15 scores)

(3) Suppose that $y[n] = \alpha_1 x[n] + \alpha_2 x[n-20] + \alpha_3 x[n-25] + \alpha_4 x[n-30]$

How do we use the cepstrum and the lifter to recover $x[n]$ from $y[n]$?

(10 scores)

(4) Suppose that the impulse response of a filter is

$$h[n] = 0.2^{n} \text{ for } n > 0, \quad h[n] = 0.2^{-n} \text{ for } n < 0, \quad h[0] = 0.5$$

(a) Is the filter a smoother or an edge detector? Why? (5 scores)

(b) Try to implement $y[n] = x[n] * h[n]$ (* means the convolution) with the least number of multiplications. (10 scores)

(5) (a) Why the Notch filter is hard to design? (b) What is the advantage of using the Wiener filter for equalizer? (10 scores)

(6) Why the Mel-frequency cepstrum is more suitable for dealing with the acoustic signal than the original cepstrum? (10 scores)

(7) (a) Which vocal signal sounds louder? (i) $\cos(200\pi t)$, (ii) $\sin(600\pi t)$, (iii) $\cos(1800\pi t)$. (5 scores)

(b) Why the speech and the music signals always have the chord phenomena (10 scores)