

## DSP HW 2

Instructor: Chun-Tang Chao 趙春棠

1. Please write a Matlab function  $y = \text{mov\_avg\_filt}(x,n)$  that can filter the signal  $x$  by an n-term moving average filter and  $y$  is the output.

Please verify your program by inputting a random sequence composed of 100 (Uniformly Distributed) random numbers between 2 and 5. Then run your program by setting  $n=3$  and  $n=9$ , respectively, and finally show your opinions about the results.

**Note:** Comparison between the two results is needed. You should write a for-loop to finish the function instead of using functions, e.g. *filter*, in Matlab.

**Hint:** The *rand* function in Matlab can produce uniformly distributed random numbers between 0 and 1.

2. (a) Please design a LP Butterworth filter  $H(s)$  satisfying the specifications  $G_p \geq -3dB$ ,  $G_s \leq -14dB$ ,  $\omega_p = 10^5$  rads/sec,  $\omega_s = 1.5 \times 10^5$  rads/sec

**Hint:** Four steps: (1) Determine order  $n$ , (2) Determine  $\omega_c$ , (3) Determine the normalized  $H(s)$ , (4) Determine the scaled  $H(s)$

- (b) Please finish the design problem in (a) by Matlab.

**Hint:** *buttord*, *butter*, *tf* functions in Matlab.

- (c) Plot the spectrums in (a) and (b) to verify your design.

- (d) Why do most conventional engineers focus on designing LP filters? What should we do if we need a BP filter?

3. Filter Applications (濾波器應用) :

[http://140.114.76.148/jang/books/audioSignalProcessing/10.1-filterApplication.asp?title=11-1%20Filter%20Applications%20\(濾波器應用\)](http://140.114.76.148/jang/books/audioSignalProcessing/10.1-filterApplication.asp?title=11-1%20Filter%20Applications%20(濾波器應用))

Please record an audio file (\*.wav) in your mother tongue and redo the experiments ( 1. LP, 2.HP, 3. One-fold echo, 4. Multiple-fold echo ) in the above web-page.

**Note:** The *filter* instruction in Matlab is used to implement filtering.

※ Reference: <http://140.114.76.148/jang/> Prof. Jyh-Shing Roger Jang (張智星 教授)