

濾波器設計與應用

函式定義與傳呼

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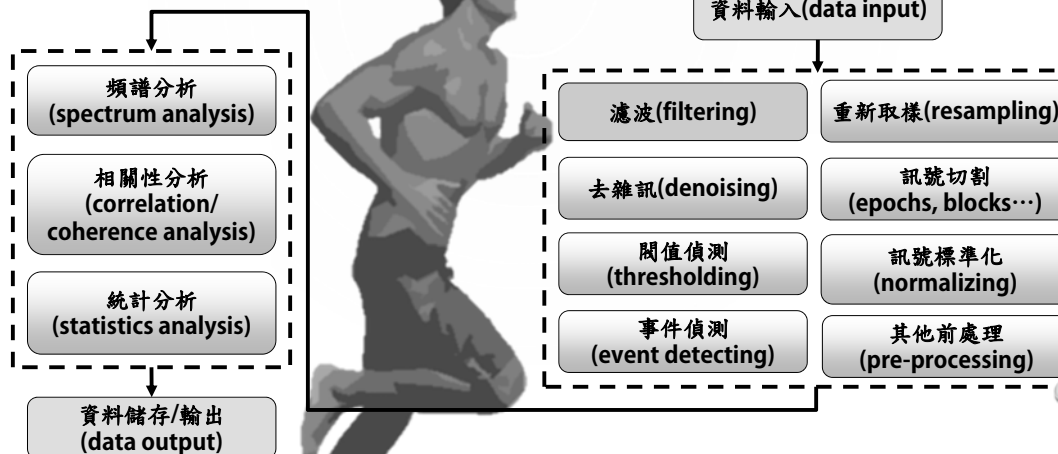
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請先下載本週上課資料

- <http://www.ym.edu.tw/~cflu>
- 點選左欄 [課程資料]
- 下載第8週上課資料 [[demodata_L7.zip](#)]，檔案大小約1MB

訊號分析方法



本週課程內容

- 數位濾波器設計(digital filter design)
- 有限脈衝響應(finite impulse response, FIR)濾波器
- 無限脈衝響應(infinite impulse response, IIR)濾波器
- MATLAB函式定義與傳呼

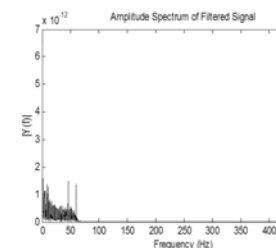
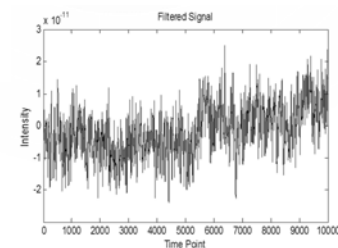
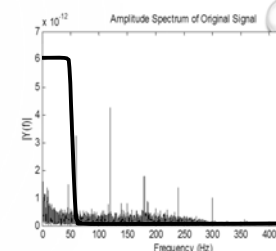
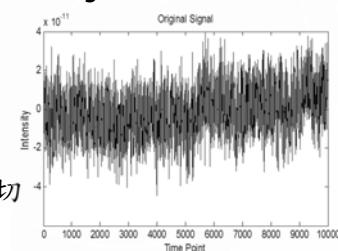
數位濾波器設計原理

濾波的目的

將訊號的頻譜成分作調整

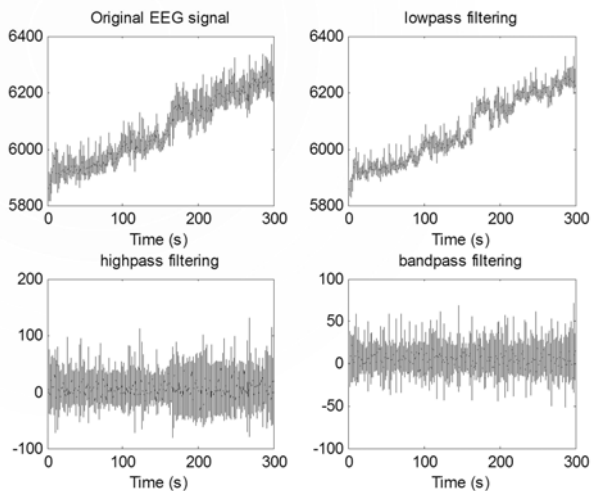
濾波器設計與頻譜分析關係密切

EEG signal



濾波的目的

- 鎖定特定頻帶觀察生理現象
- 移除高頻機器或環境雜訊
- 移除低頻訊號飄移成分



傅立葉轉換的特性

$$\begin{aligned}
 X(f) &= \sum_{n=0}^{N-1} x(n) \cos(2\pi fn / N) - j \sum_{n=0}^{N-1} x(n) \sin(2\pi fn / N) \\
 &= \sum_{n=0}^{N-1} x(n) e^{-j2\pi fn / N}
 \end{aligned}$$

	Time domain	Frequency domain
Convolution	$h(n) * x(n)$	$H(f) \cdot X(f)$
Time shifting	$x(n - k)$	$e^{-j2\pi k / N} X(f)$

Z轉換(Z-TRANSFORM)的特性

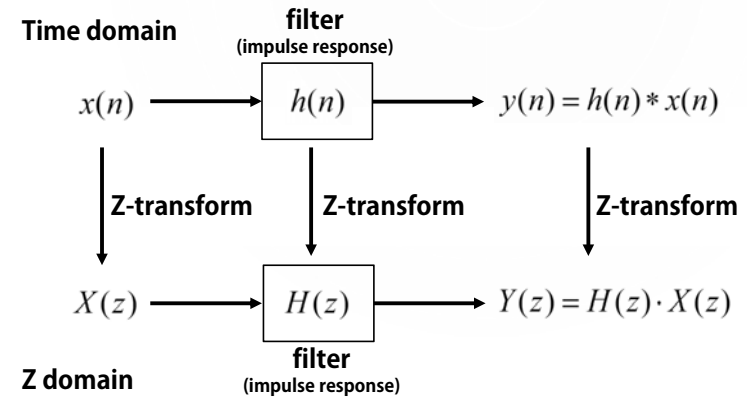
$$X(z) \equiv Z\{x(n)\} = \sum_{n=0}^{N-1} x(n)z^{-n}$$

z is an arbitrary complex value

如將 z 設為 $e^{j2\pi f/N}$ 即為傅立葉轉換!!

	Time domain	Z domain
Convolution	$h(n) * x(n)$	$H(z) \cdot X(z)$
Time shifting	$x(n-k)$	$z^{-k} X(z)$

濾波器的運算原理



轉換函數(DIGITAL TRANSFER FUNCTION)

$$H(z) = \frac{Y(z)}{X(z)} = \frac{\text{numerator}}{\text{denominator}} = \frac{\sum_{k=0}^K b_k z^{-k}}{\sum_{l=0}^L a_l z^{-l}}, \text{ usually } a_0 = 1$$

$$Y(z) = X(z) \cdot H(z) = X(z) \frac{\sum_{k=0}^K b_k z^{-k}}{\sum_{l=0}^L a_l z^{-l}}$$

輸入輸出函式(INPUT-OUTPUT FUNCTION)

$$\text{Z domain: } Y(z) = X(z) \frac{\sum_{k=0}^K b_k z^{-k}}{\sum_{l=0}^L a_l z^{-l}}$$

$$\text{Time domain: } y(n) = \sum_{k=0}^K b_k x(n-k) - \sum_{l=1}^L a_l y(n-l)$$

The order of filter (the number of coefficients)

numerator (moving average)

denominator (Autoregressive)

輸入輸出函式 (INPUT-OUTPUT FUNCTION)

time domain $y(n) = \sum_{k=0}^K b_k x(n-k) - \sum_{l=1}^L a_l y(n-l)$

$\sum_{l=0}^L a_l y(n-l) = \sum_{k=0}^K b_k x(n-k)$, where $a_0 = 1$

$\sum_{l=0}^L a_l Z\{y(n-l)\} = \sum_{k=0}^K b_k Z\{x(n-k)\}$

$\sum_{l=0}^L a_l z^{-l} Z\{y(n)\} = \sum_{k=0}^K b_k z^{-k} Z\{x(n)\}$

linearity of Z-transform

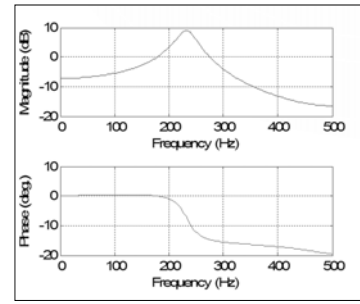
Time shifting of Z-transform

$Z\{y(n)\} = Z\{x(n)\} \frac{\sum_{k=0}^K b_k z^{-k}}{\sum_{l=0}^L a_l z^{-l}}$ or $Y(z) = X(z) \frac{\sum_{k=0}^K b_k z^{-k}}{\sum_{l=0}^L a_l z^{-l}}$ # Z domain

有限脈衝響應濾波器 FINITE IMPULSE RESPONSE, FIR FILTERS

轉換函數(濾波器)的頻譜分析

$H(f) = \frac{Y(f)}{X(f)} = \frac{\sum_{k=0}^K b_k e^{-j2\pi f k / N}}{\sum_{l=0}^L a_l e^{-j2\pi f l / N}} = \frac{fft(b_k)}{fft(a_l)}$



實例：
畫出下列轉換函數的頻譜與相位變化
 $H(z) = \frac{0.2 + 0.5z^{-1}}{1 - 0.2z^{-1} + 0.8z^{-2}}$
請執行 demodata_L7\TransFun_spectrum.m

- 也可直接使用
- [H,faxis] = freqz(b,a,N,'whole',samplerat

有限脈衝濾波器 FIR

轉換函數 $H(z) = \sum_{k=0}^K b_k z^{-k}$

輸入輸出函式

Z domain: $Y(z) = X(z) \sum_{k=0}^K b_k z^{-k}$

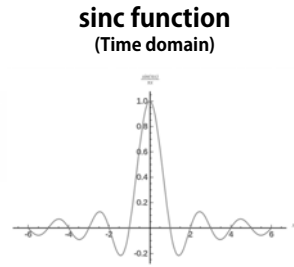
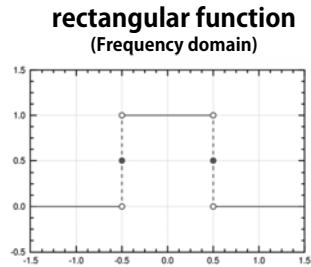
Time domain: $y(n) = \sum_{k=0}^K b_k x(n-k)$

numerator
(moving average)

- [優點]
- 穩定 (stable)
 - 線性相位變化 (linear phase shift)
- [缺點]
- 需要較高階數來滿足濾波條件
 - 計算效率較低

FIR濾波器設計

- 將設定好的頻率響應，反推算出轉換函數的係數**b**
- 係數的計算，可以透過對欲得到的頻率響應做反傅立葉轉換(**inverse Fourier transform**)取得



$$\text{sinc}(x) = \frac{\sin(\pi x)}{\pi x}$$

FIR濾波器設計

F_c is the cutoff frequency, T_s is the sample interval in second, L is the length of filter.

$$b(k) = \frac{\sin[2\pi f_c T_s (k - L/2)]}{\pi(k - L/2)}$$

Lowpass

$$b(k) = \frac{\sin[\pi(k - L/2)]}{\pi(k - L/2)} - \frac{\sin[2\pi f_c T_s (k - L/2)]}{\pi(k - L/2)}$$

Highpass

$$b(k) = \frac{\sin[2\pi f_H T_s (k - L/2)]}{\pi(k - L/2)} - \frac{\sin[2\pi f_L T_s (k - L/2)]}{\pi(k - L/2)}$$

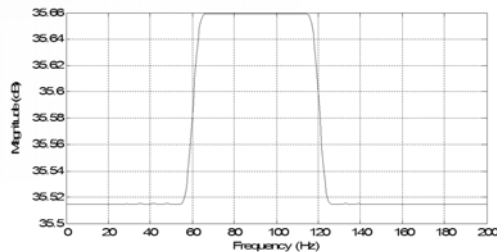
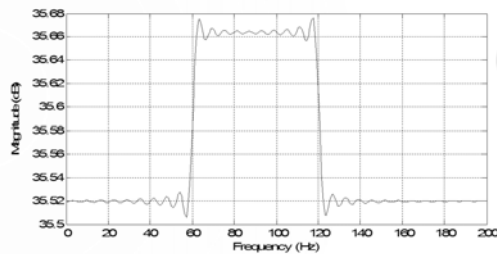
Bandpass

$$b(k) = \frac{\sin[2\pi f_L T_s (k - L/2)]}{\pi(k - L/2)} + \frac{\sin[\pi(k - L/2)]}{\pi(k - L/2)} - \frac{\sin[2\pi f_H T_s (k - L/2)]}{\pi(k - L/2)}$$

Bandstop

FIR濾波器設計

- Window function**對於濾波器的助益
- 請執行 `demodata_L7\FIR_Window.m`
- 比較下列兩者的差異：
 - $H = \text{fft}(b, N);$
 - $HW = \text{fft}(b.*\text{hamming}(L+1)', N);$



MATLAB濾波器設計函式

三階段設計流程

firord
↓
firpm
↓
filtfilt, filter, conv

- 1st-stage:
- 決定濾波器階數(**order**)與截斷頻率(**cutoff frequency**)
- 2nd-stage:
- 產生係數**b**序列
- 3rd-stage:
- 將濾波器套用到訊號上

二階段設計流程

fir1
↓
filtfilt, filter, conv

Data must have length more than 3 times filter order!!

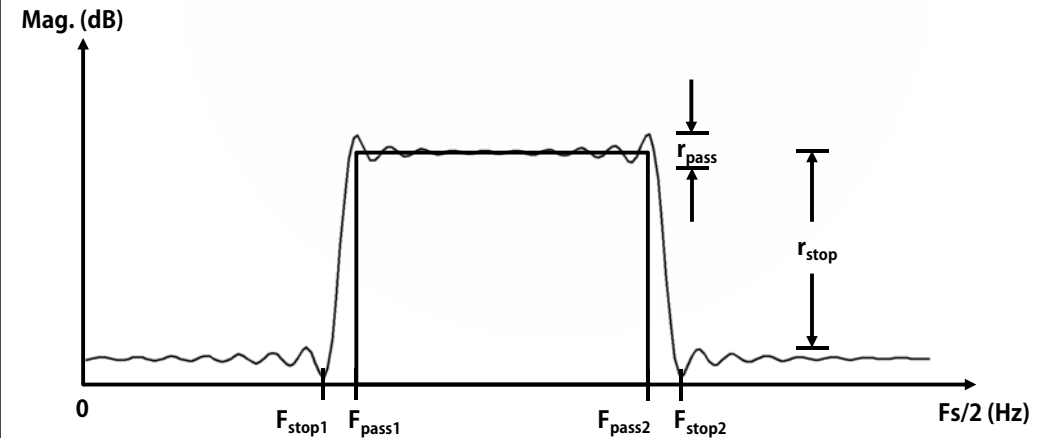
[MATLAB RULE] 建立與傳呼函式(FUNCTION)

- 以 `mean.m` 為例
- 格式為:

```
function y = mean(x,dim)
%MEAN Average or mean value.
% For vectors, MEAN(X) is the mean value of the elements in X.
y = sum(x)/size(x,dim);
```

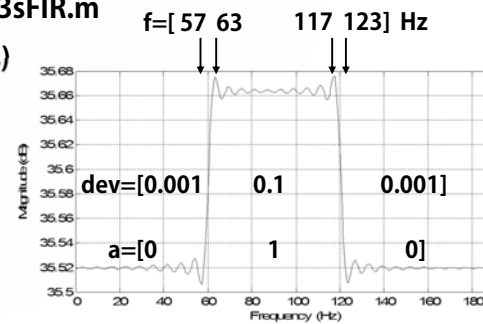
- 檔名需與 **function name** 相同
- 輸出變數一定要存在

濾波器規格(FILTER SPECIFICATION)



MATLAB 三階段FIR濾波處理

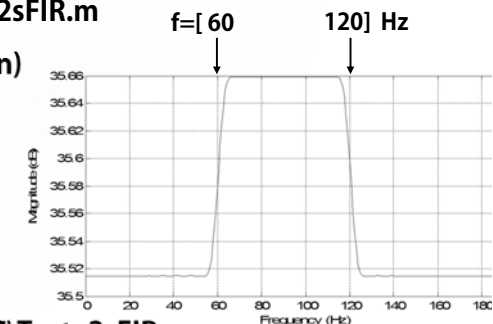
- 三階段FIR濾波處理，`demodata_L7\filter_3sFIR.m`
`[sigfilter,n] = filter_3sFIR(sig,f,a,dev,fs)`



- 欲使用該函式，請開啟並執行 `demodata_L7\Test_3sFIR.m`

MATLAB 二階段FIR濾波處理

- 二階段FIR濾波處理，`demodata_L7\filter_2sFIR.m`
`sigfilter = filter_2sFIR(sig,f,fs,n,type,win)`



- 欲使用該函式，請開啟並執行 `demodata_L7\Test_2sFIR.m`

無限脈衝響應濾波器

INFINITE IMPULSE RESPONSE, IIR FILTERS

無限脈衝濾波器 IIR

• 轉換函數
$$H(z) = \frac{\sum_{k=0}^K b_k z^{-k}}{\sum_{l=0}^L a_l z^{-l}}$$

• 輸入輸出函式

Z domain:
$$Y(z) = X(z) \frac{\sum_{k=0}^K b_k z^{-k}}{\sum_{l=0}^L a_l z^{-l}}$$

Time domain:
$$y(n) = \sum_{k=0}^K b_k x(n-k) - \sum_{l=1}^L a_l y(n-l)$$

numerator
(moving average)

denominator
(Autoregressive)

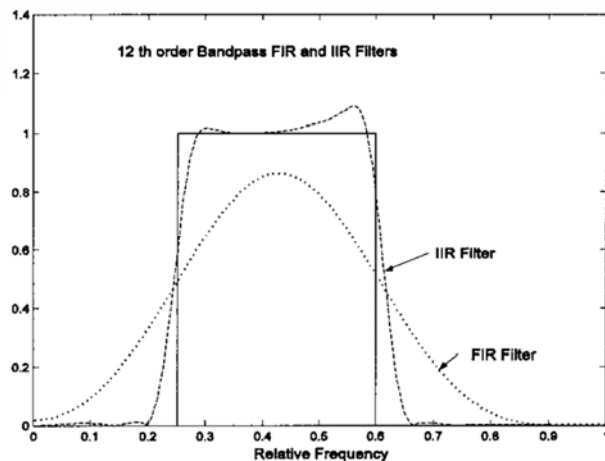
[優點]

- 僅需較低階數來滿足濾波條件
- 計算效率較高，但運算要持續用到整段訊號

[缺點]

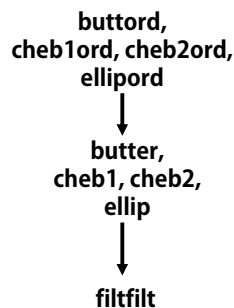
- 可能不穩定(unstable)，分母項為0
- 非線性相位變化(nonlinear phase shift)

同階數 FIR 與 IIR 比較



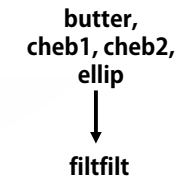
MATLAB 濾波器設計函式

三階段設計流程



- 1st-stage:
- 決定濾波器階數(order)與截斷頻率(cutoff frequency)
- 2nd-stage:
- 產生係數b與a序列
- 3rd-stage:
- 將濾波器套用到訊號上

二階段設計流程

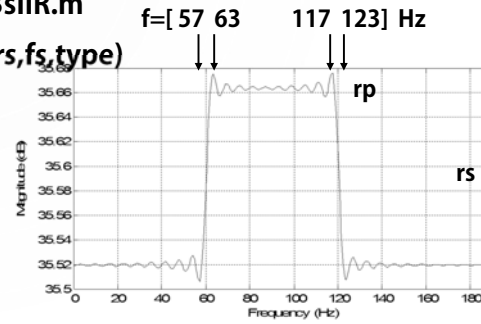


Data must have length more than 3 times filter order!!

MATLAB三階段IIR濾波處理

- 三階段IIR濾波處理，demodata_L7\filter_3sIIR.m

`[sigfilter, n]=filter_3sIIR(sig,wp,ws,rp,rs,fs,type)`

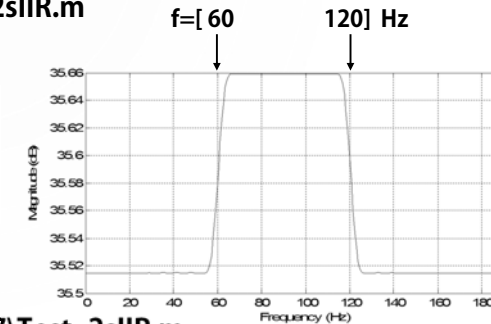


- 欲使用該函式，請開啟並執行demodata_L7\Test_3sIIR.m

MATLAB二階段IIR濾波處理

- 二階段IIR濾波處理，demodata_L7\filter_2sIIR.m

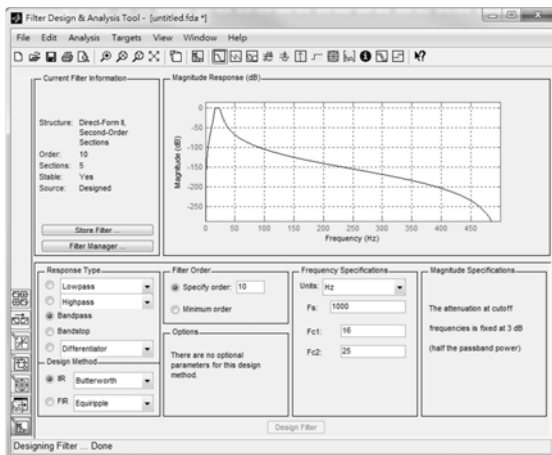
`sigfilter = filter_2sIIR(sig,f,fs,n,type)`



- 欲使用該函式，請開啟並執行demodata_L7\Test_2sIIR.m

MATLAB濾波器設計工具箱

- 在command window下輸入fdatool呼叫濾波器設計工具箱



THE END

<http://www.ym.edu.tw/~cflu>