Advanced Digital Signal Processing

高等數位訊號處理

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課程網頁：http://djj.ee.ntu.edu.tw/ADSP.htm

歡迎大家來修課，也歡迎有問題時隨時聯絡！

- **評分方式：**

Basic: 15 scores

原則上每位同學都可以拿到 12 分以上，
另外，上課回答問題，每回答一次加1分

Homework: 60 scores (5 times, 每 3 週一次)

請自己寫，和同學內容極高度相同 ，將扣 70% 的分數
就算寫錯但好好寫也會給 40~95% 的分數，
遲交分數打 8 折，不交不給分。 不知道如何寫，可用 E-mail 和我聯絡，或於上課時發問
禁止 Ctrl-C Ctrl-V 的情形。

Term paper 25 scores

Term paper 25 scores

方式有四種

(1) 書面報告

10頁以上(不含封面)，中英文皆可，11或12的字體，題目可選擇和課程有關的任何一個主題。

格式和一般寫期刊論文或碩博士論文相同，包括 abstract, conclusion, 及 references，並且要分 sections，必要時有subsections。 References 的寫法, 可參照一般 IEEE 的論文的寫法
鼓勵多做實驗及模擬，有創新更好。
嚴禁 Ctrl-C Ctrl-V 的情形，否則扣 70% 的分數

(2) Tutorial (對既有領域做淺顯易懂的整理)

限十七個名額，和書面報告格式相同，但頁數限制為18頁以上(若為加強前人的 tutorial，則頁數為 (2/3)N + 13 以上，N 為前人 tutorial 之頁數)，題目由老師指定，以**清楚且有系統**的介紹一個主題的基本概念和應用為要求，為上課內容的進一步探討和補充，**交 Word 檔**。

選擇這個項目的同學，學期成績加 4分

(3) 口頭報告

限四個名額，每個人 40分鐘，題目可選擇和課程有關的任何一個主題。口頭報告將於 5月1日(第 8 週)進行。有意願的同學，請儘早告知，以先登記的同學為優先。

口頭報告時，鼓勵大家提問（包括口頭報告的同學，也可針對其他同學的報告內容提問）。曾經提問的同學，加分同上課回答問題。

選擇這個項目的同學，學期成績加 2分

(4) 編輯 Wikipedia

中文或英文網頁皆可，至少 2 個條目，但不可同一個條目翻成中文和英文。總計 80行以上。限和課程相關者，自由發揮，越有條理、有系統的越好

選擇編輯 Wikipedia 的同學，請於 6月26日前，向我登記並告知我要編緝的條目(2 個以上)，若有和其他同學選擇相同條目的情形，則較晚向我登記的同學將更換要編緝的條目

書面報告和編輯 Wikipedia，期限是 7月2日

Tutorial 可供選擇的題目(共 17 個，可以略做修改)

(1) Guided Filter

(2) Image Registration

(3) Recent Development of Equalizer

(4) Echo Cancellation

(5) Rain Removal for Images

(6) Distance Estimation for Sound Source

(7) Signal Processing for Big Data

(8) Fuzzy Logic in Signal Processing

(9) Multiple Signal Classification (MUSIC) Algorithm

(10) Image Stitching

Tutorial 可供選擇的題目(可以略做修改)

(11) Photoacoustic Image Processing

(12) Bioacoustics

(13) Learning Based Image Compression Techniques

(14) Shape Adaptive Image Compression

(15) H.265 Video Compression Architecture

(16) Anomaly Detection Using Neural Networks

(17) Learning Based Prediction Techniques

上課時間：15 週

3/6,

3/13,

3/20, 出 HW1

3/27,

4/10, 交 HW1

4/17, 出 HW2

4/24,

5/1, Oral Presentation, 交 HW2

4/3 放假

5/8, 出 HW3

5/15,

5/22, 交 HW3

5/29, 出 HW4

6/5,

6/12, 交 HW4,

6/19, 出 HW5

7/2, 交 HW5 及 term paper

原則上: 3*n* 週出作業, 3*n*+2 週繳交

Matlab

Download: 請洽台大各系所

參考書目

洪維恩，Matlab 7 程式設計，旗標，台北市，2010. .　(合適的入門書)

張智星，Matlab 程式設計入門篇，第三版，基峰，2011.

蒙以正，數位信號處理：應用 Matlab，旗標，台北市，2007.

繆紹綱譯，數位影像處理：運用-Matlab，東華，2005.

預計看書學習所花時間： 3~5 天

研究所和大學以前追求知識的方法有什麼不同？

研究所：觀念的學習

大學：

Question:

Why should we use the Fourier transform?

Is the Fourier transform the best choice in any condition?

I. Introduction

Outline

- Introduction (1.3W)
- 1. Filter Design
  - FIR (2W)
  - IIR, Others (1.7W)
- 2. Homomorphic Signal Processing (1W)
- 3. Applications
  - Acoustics (1W)
  - Compression, Image (1.3 W)
- 4. Fast Algorithms
  - Basic (1W)
  - FFT and Convolution (2W)
  - Others (0.3W)
- 5. Orthogonal Transforms
  - Walsh Transform (1W)
  - Number Theory (0.7 W)
  - OFDM, CDMA (0.7 W)

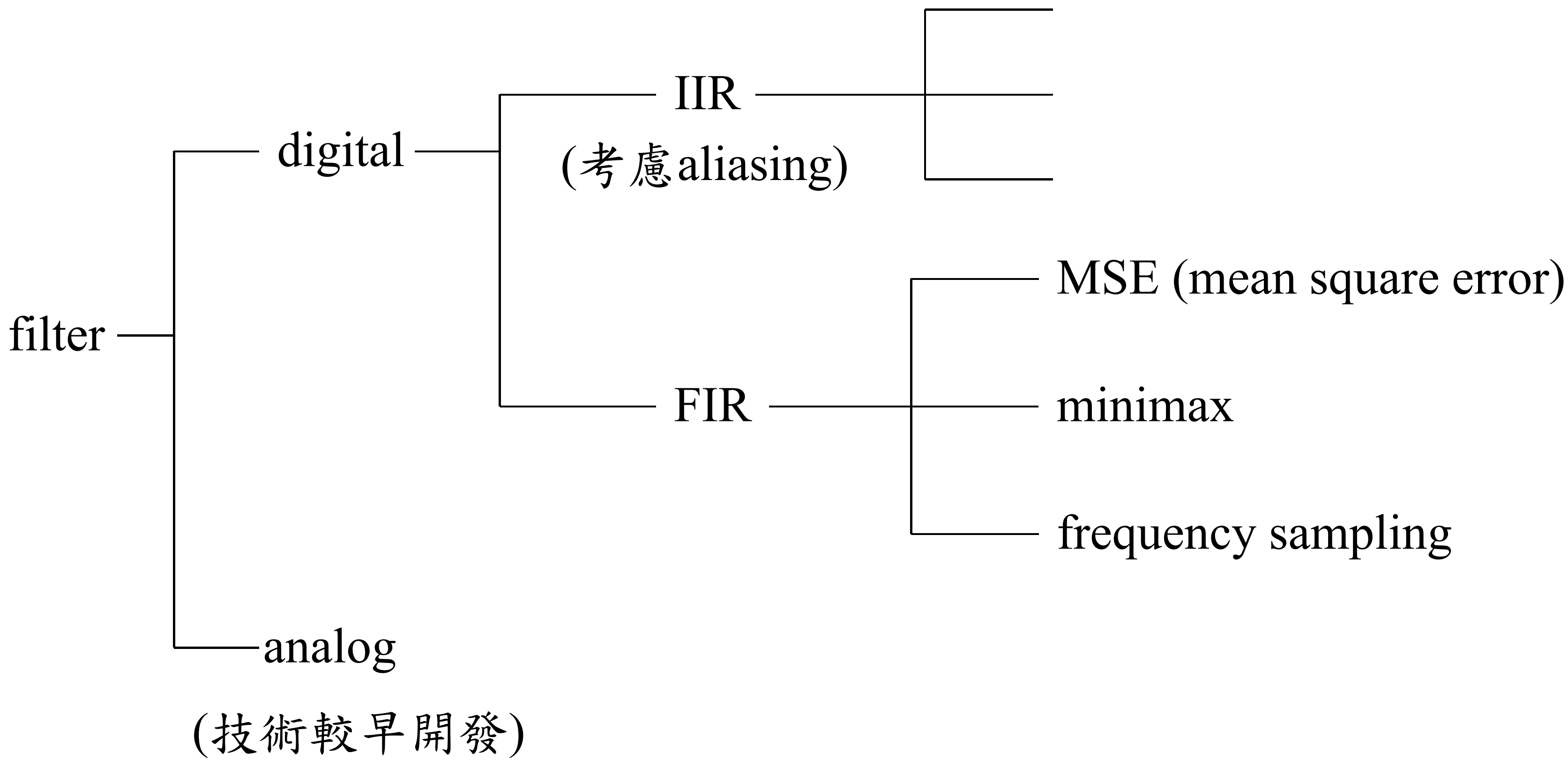
目標：

(1) 對 Digital Signal Processing 作更有系統且深入的了解

(2) 學習 Digital Signal Processing 幾個重要子領域的基礎知識

Part 1: Filter

- Filter 的分類



IIR filter 的優點：(1) easy to design

(2) (sometimes) easy to implement

缺點：

FIR filter 的優點：

缺點： An FIR filter is impossible to have the ideal frequency response of

Part 2: Homomorphic Signal Processing

- 概念：把 convolution 變成 addition

Part 3: Applications of DSP

filter design, data compression (image, video, text), acoustics (speech, music), image analysis (structural similarity, sharpness), 3D accelerometer

- **Part 4: Fast Algorithms**
- Basic Implementation Techniques

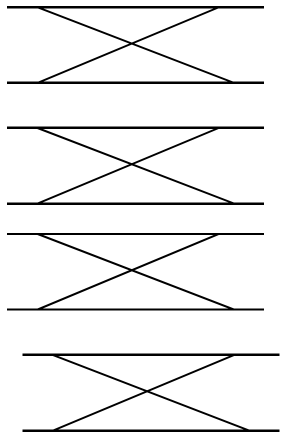
Example: one complex number multiplication

= ? Real number multiplication.

Trade-off: “Multiplication” takes longer than “addition”

• FFT and Convolution

Due to the Cooley-Tukey algorithm (butterflies),
the complexity of the FFT is:



The complexity of the convolution is: 3個 DFTs, $O(N\log_2 N)$

• Part 5: Orthogonal Transforms

DFT 的兩個主要用途:

Question: DFT 的缺點是什麼？　$DFT\left(x[n]\right)=\sum_{n=0}^{N-1} x[n] e^{-j\frac{2\pi mn}{N}}$

- Walsh Transform
  (CDMA)

- Number Theoretic Transform

- Orthogonal Frequency-Division Multiplexing (OFDM)

- Code Division Multiple Access (CDMA)

Review 1: Four Types of the Fourier Transform

(1) Fourier Transform

$$X(f)=\int_{-\infty}^{\infty}x(t)e^{-j2\pi ft}dt \quad , \qquad x(t)=\int_{-\infty}^{\infty}X(f)e^{j2\pi ft}df$$

Alternative definitions

$$X(\omega)=\int_{-\infty}^{\infty}x(t)e^{-j\omega t}dt \quad , \qquad x(t)=\frac{1}{2\pi}\int_{-\infty}^{\infty}X(\omega)e^{j\omega t}d\omega$$

(2) Fourier series (suitable for period function)

$$X[m]=\int_{0}^{T}x(t)e^{-j\frac{2\pi m}{T}t}dt \qquad x(t)=T^{-1}\sum_{m=-\infty}^{\infty}X[m]e^{j\frac{2\pi m}{T}t}$$

T: 週期 $x(t)=x(t+T)$

possible periods:

possible frequencies:

頻率和 m 之間的關係： $f=\frac{m}{T}$ $\qquad \frac{1}{T}$ 整數倍

(3) Discrete-time Fourier transform (DSP 常用)

$$X(f)=\sum_{n=-\infty}^{\infty}x[n]e^{-j2\pi f n\Delta_t}\quad,\quad x[n]=\Delta_t\int_0^{1/\Delta_t}X(f)e^{j2\pi f n\Delta_t}df$$

$$t=n\Delta_t$$

Δ_t : sampling interval

$$X(\omega)=\sum_{n=-\infty}^{\infty}x[n]e^{-j\omega n\Delta_t}\qquad x[n]=\frac{\Delta_t}{2\pi}\int_0^{2\pi/\Delta_t}X(\omega)e^{j\omega n\Delta_t}d\omega$$

(4) Discrete Fourier transform (DFT) (DSP 常用)

$$X[m]=\sum_{n=0}^{N-1}x[n]e^{-j\frac{2\pi mn}{N}}\quad,\quad x[n]=\frac{1}{N}\sum_{m=0}^{N-1}X[m]e^{j\frac{2\pi mn}{N}}$$

頻率和 m 之間的關係：$f=\dfrac{m}{N\Delta_t}=\dfrac{m}{N}f_s$

where $f_s = 1/\Delta_t$ (sampling frequency)

- **四種 Fourier transforms 的比較**

	time domain	frequency domain
(1) Fourier transform	continuous, aperiodic	continuous, aperiodic
(2) Fourier series	continuous, periodic (or continuous, only the value in a finite duration is known)	discrete, aperiodic
(3) discrete-time Fourier transform	discrete , aperiodic	continuous, periodic
(4) discrete Fourier transform	discrete, periodic (or discrete, only the value in a finite duration is known)	discrete, periodic

Review 2: Normalized Frequency

(1) Definition of **normalized frequency *F***:

$$F = \frac{f}{f_s} = f\,\Delta_t = \frac{\omega \Delta_t}{2\pi}$$

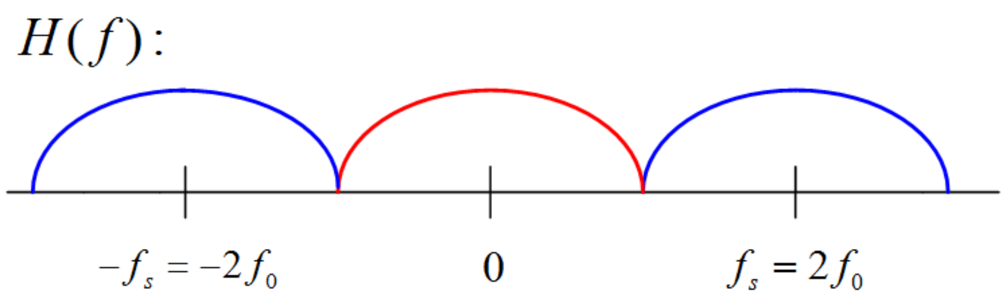
where $f_s = 1/\Delta_t$ (sampling frequency)

Δ_t : sampling interval

(2) folding frequency f_0

$$f_0 = \frac{f_s}{2}$$

若以 normalized frequency 來表示，
folding frequency = 1/2



For the discrete time Fourier transform

(1) $G(f) = G(f + f_s)$ $\longrightarrow$ i.e., $G(F) = G(F + 1)$.

(2) If $g[n]$ is real $\longrightarrow$ $G(F) = G^*(-F)$ (* means conjugation)

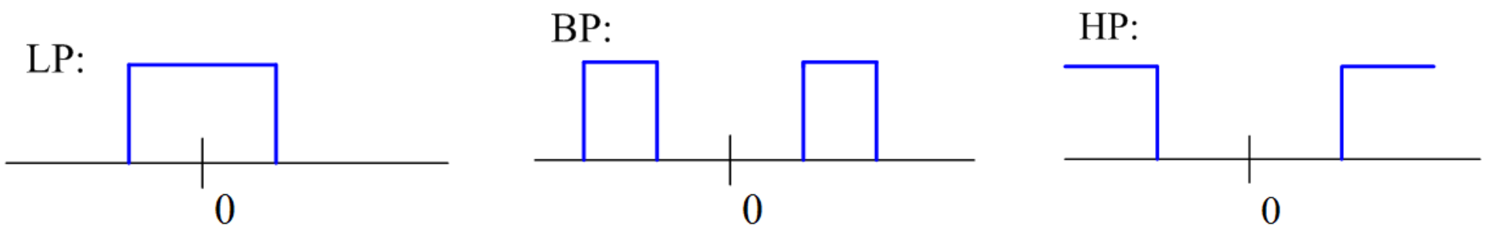
只需知道 $G(F)$ for $0 \le F \le ½$ (即 $0 < f < f_0$)

就可以知道全部的 $G(F)$

(3) If $g[n] = g[-n]$ (even) $\longrightarrow$ $G(F) = G(-F)$,

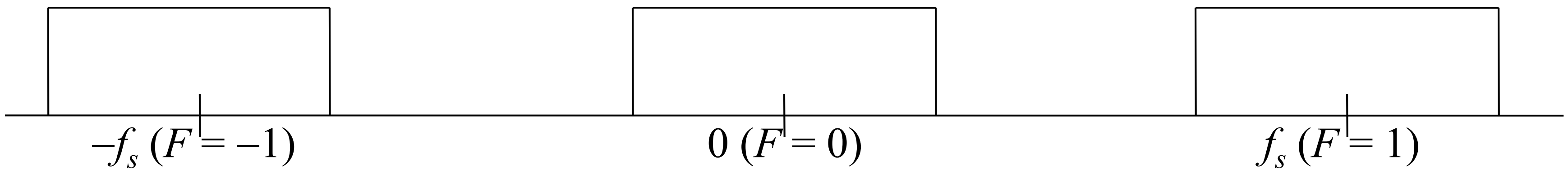
$g[n] = -g[-n]$ (odd) $\longrightarrow$ $G(F) = -G(-F)$

Analog filter: $H(f)$

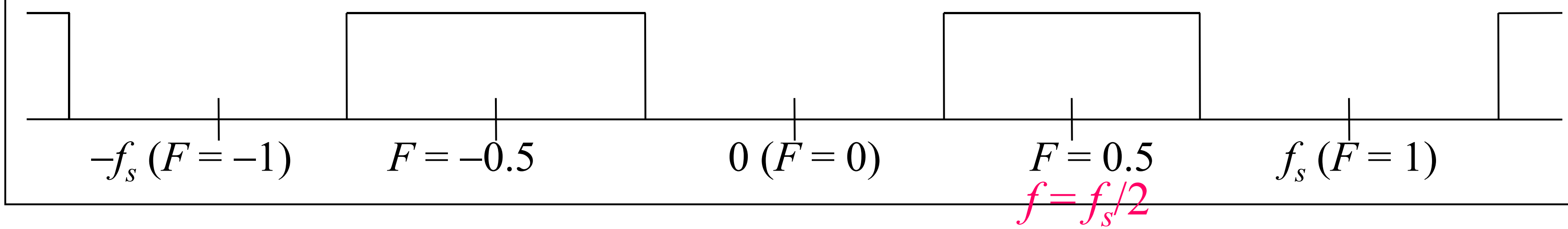


- Discrete time Fourier transform of the lowpass, highpass, and band pass filters

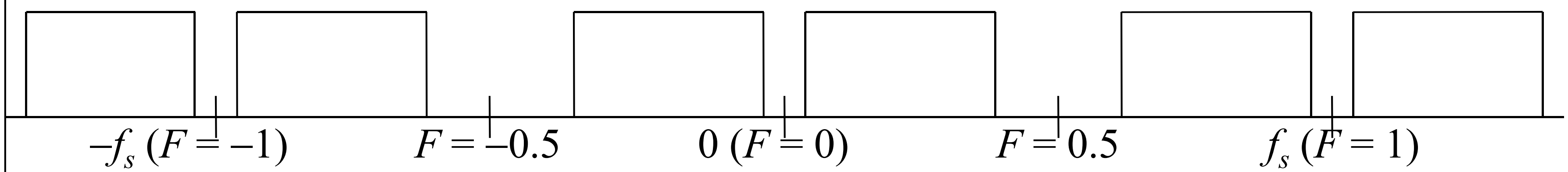
low pass filter（ **pass band 在 fs 的整數倍附近** ）



high pass filter



band pass filter



Review 3: Z Transform and Laplace Transform

- ***Z*-Transform**

suitable for discrete signals

$$G(z) = \sum_{n=-\infty}^{\infty} g[n] z^{-n}$$

Compared with the discrete time Fourier transform:

$$G(f) = \sum_{n=-\infty}^{\infty} g[n] e^{-j2\pi f n \Delta_t} \qquad z = e^{j2\pi f \Delta_t}$$

• Laplace Transform

suitable for continuous signals

One-sided form $G(s) = \int_{0}^{\infty} g(t) e^{-st} dt$

Two-sided form $G(s) = \int_{-\infty}^{\infty} g(t) e^{-st} dt$

Compared with the Fourier transform:

$$G(f) = \int_{-\infty}^{\infty} g(t) e^{-j2\pi f t} dt \qquad s = j2\pi f$$

Review 4: IIR Filter Design

Two types of digital filter:

(1) IIR filter (infinite impulse response filter)

(2) FIR filer (finite impulse response filer)

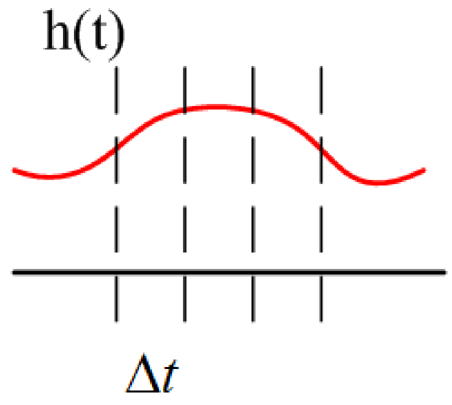
There are 3 popular methods to design the IIR filter.

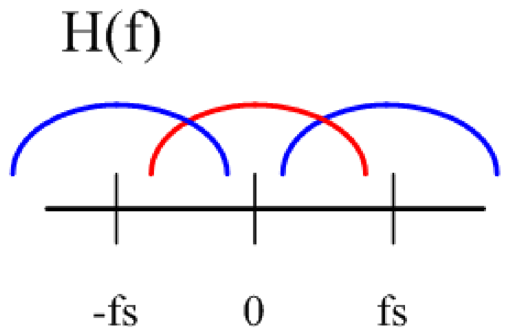
Method 1: Impulse Invariance

白話一點，就是直接做 sampling

analog filter $h_a(t)$ digital filter $h[n]$

$$h[n]=h_a(n\Delta_t)$$





Advantage : Simple

Disadvantage : (1) infinite

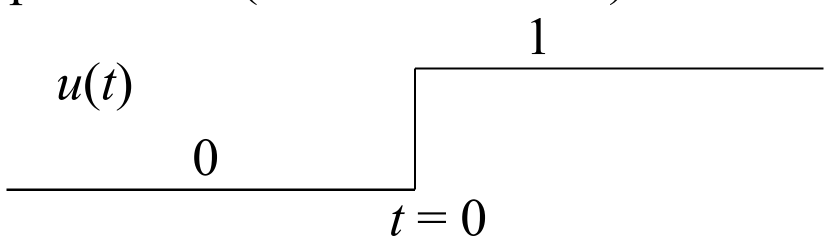
(2)

Method 2: Step Invariance

對 step function 的 response 作 sampling

analog filter $h_a(t)$ digital filter $h[n]$

step function (continuous form)



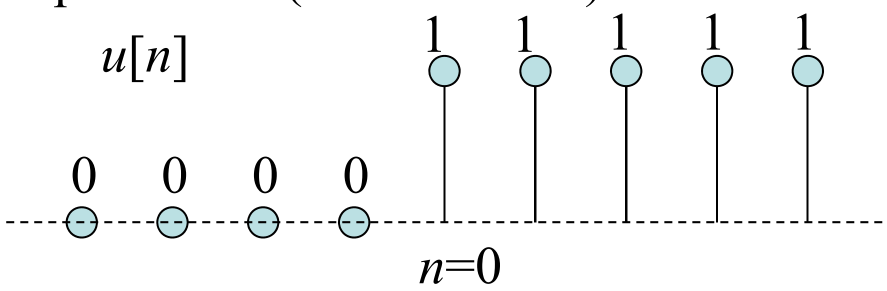
Laplace transform of $u(t)$:

$$\frac{1}{s}$$

Fourier transform of $u(t)$:

$$\frac{1}{j2\pi f}$$

step function (discrete form)



Z transform of $u[n]$:

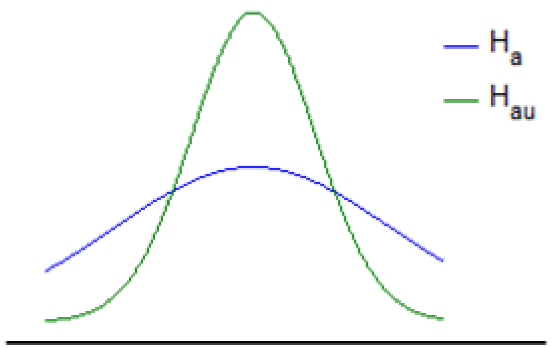
$$\frac{1}{1-z^{-1}}$$

Step 1 Calculate the convolution of $h_a(t)$ and $u(t)$

$$h_{a,u}(t) = h_a(t) * u(t) = \int_{-\infty}^{\infty} h_a(\tau) u(t-\tau) d\tau = \int_{-\infty}^{t} h_a(\tau) d\tau$$

$$H_{a,u}(f) = \frac{H_a(f)}{j2\pi f}$$

(其實就是對 $h_a(t)$ 做積分)



Step 2 Perform sampling for $h_{a,u}(t)$

$$h_u[n] = h_{a,u}(n\Delta_t)$$

Step 3 Calculate $h[n]$ from $h[n] = h_u[n] - h_u[n-1]$

Note: Since $h_u[n] = h[n] * u[n]$ $\quad H_u(z) = \frac{1}{1-z^{-1}} H(z)$

$$H(z) = (1 - z^{-1}) H_u(z)$$

so $h[n] = h_u[n] - h_u[n-1]$

Advantage of the step invariance method:

＊主要 Advantage:

Disadvantage of the step invariance method:

較為間接，設計上稍微複雜

Method 3: Bilinear Transform

Suppose that we have known an analog filter $h_a(t)$ whose frequency response is $H_a(f)$.

To design the digital filter $h[n]$ with the frequency response $H(f)$,

$$H\left(f_{new}\right) = H_a\left(f_{old}\right)$$

$f_{old} \in (-\infty, \infty)$

$f_{new} \in (-f_s/2,\ f_s/2)$

$f_s = 1/\Delta_t$ (sampling frequency)

- The relation between f_{new} and f_{old} is determined by the mapping function

$$s = c\frac{1-z^{-1}}{1+z^{-1}}$$

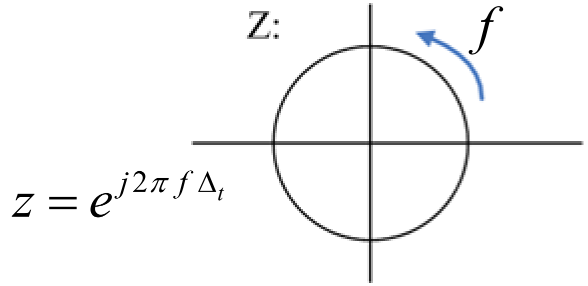
s: index of the Laplace transform

z: index of the Z transform

c: some constant

$s = c\dfrac{1-z^{-1}}{1+z^{-1}}$ $s = j2\pi f_{old}$ $z = e^{j2\pi f_{new}\Delta_t}$ 代入

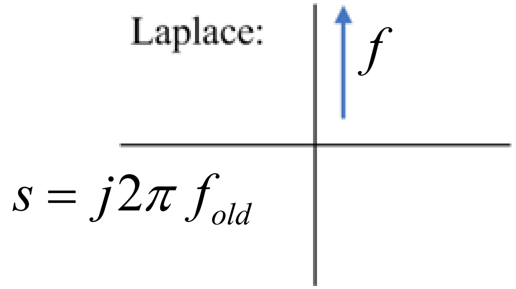
參考page 25、page26



$$j2\pi f_{old} = c\frac{1-e^{-j2\pi f_{new}\Delta_t}}{1+e^{-j2\pi f_{new}\Delta_t}} = c\frac{e^{j\pi f_{new}\Delta_t} - e^{-j\pi f_{new}\Delta_t}}{e^{j\pi f_{new}\Delta_t} + e^{-j\pi f_{new}\Delta_t}}$$

$$= c\frac{j\sin\left(\pi f_{new}\Delta_t\right)}{\cos\left(\pi f_{new}\Delta_t\right)}$$

$$2\pi f_{old} = c\tan\left(\pi f_{new}\Delta_t\right)$$



$$f_{new} = \frac{1}{\pi\Delta_t}\text{atan}\left(\frac{2\pi}{c}f_{old}\right) = \frac{f_s}{\pi}\text{atan}\left(\frac{2\pi}{c}f_{old}\right)$$

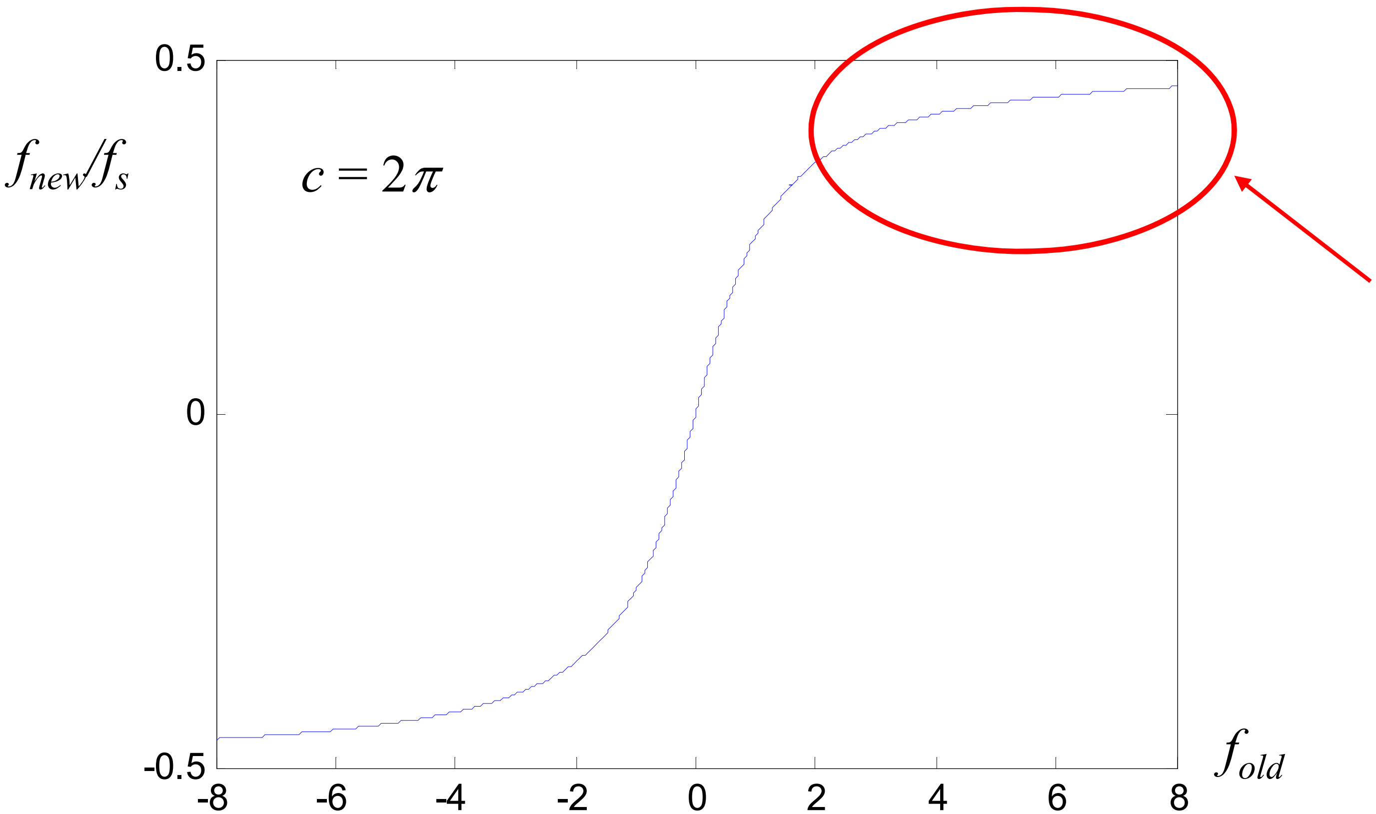
- Suppose that the Laplace transform of the analog filter $h_a(t)$ is $H_{a,L}(s)$

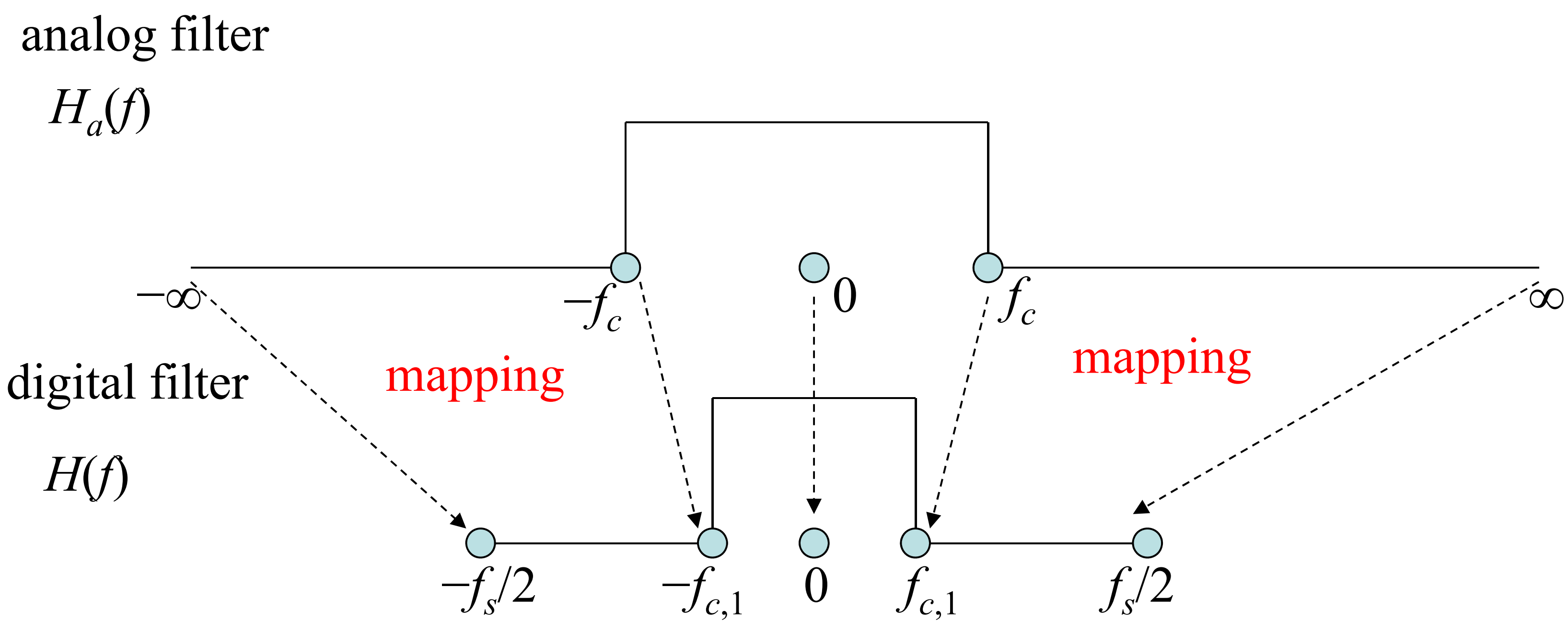
  The Z transform of the digital filter $h[n]$ is $H_z(z)$

$$H_z\left(z\right) = H_{a,L}\left(c\frac{1-z^{-1}}{1+z^{-1}}\right)$$

$$f_{new} = \frac{f_s}{\pi}\,\text{atan}\left(\frac{2\pi}{c} f_{old}\right)$$

f_{old}	$-\infty$	0	∞	1
f_{new}				





$$f_{c,1} = \frac{f_s}{\pi} \operatorname{atan}\left(\frac{2\pi}{c} f_c \right)$$

Advantage of the bilinear transform

Disadvantage of the bilinear transform

附錄一： 學習 DSP 知識把握的要點

(1) Concepts: 這個方法的核心概念、基本精神是什麼

(2) Comparison: 這方法和其他方法之間，有什麼相同的地方？
有什麼相異的地方

(3) Advantages: 這方法的優點是什麼

(3-1) Why? 造成這些優點的原因是什麼

(4) Disadvantages: 這方法的缺點是什麼

(4-1) Why? 造成這些缺點的原因是什麼

(5) Applications: 這個方法要用來處理什麼問題，有什麼應用

(6) Innovations: 這方法有什麼可以改進的地方
或是可以推廣到什麼地方