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！

上課方式

(1) 錄影 ，影片將藉由 NTU Cool 下載　http://cool.ntu.edu.tw

(2) 現場 (週三下午 15:30~18:20，明達館205室)

作業和報告繳交方式：

用 NTU Cool 來繳交作業與報告的電子檔　http://cool.ntu.edu.tw

注意，Tutorial 一定要交Word 或 Latex 原始碼

Wiki 要寄編輯條目的連結給老師

上課時間：14 週

2/21,

3/7, 出 HW1

3/13,

3/20, 交 HW1

3/27, 出 HW2

4/3,

4/10, 交 HW2

4/17, 出 HW3

4/24,

5/1, 交 HW3

5/8, 出 HW4

5/15,

5/22, 交 HW4,

5/29, 出 HW5

6/5 之前, Oral Presentation

6/12, 交 HW5 及 term paper

原則上: 3n-1 週出作業, 3n+1 週繳交

- **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，會有額外問答題，每位同學四次，每答對一次加0.8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

方式有五種

(1) 書面報告

10頁以上(不含封面)，中英文皆可，11或12的字體，題目可選擇和信號處理(包括信號、通訊、影像、音訊、生醫訊號、經濟信號處理等等) 有關的任何一個主題。

包括 abstract, conclusion, 及 references，並且要分 sections。 儘量工整

鼓勵多做實驗及模擬，

有做模擬的同學請將程式附上來，會有額外加分。

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)

限十八個名額，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

限十個名額，每個人 15~40分鐘，題目可選擇和課程有關的任何一個主題。口頭報告的同學請在6月7日以前將影片錄好，並且把影片(或連結)寄給老師。有意願的同學，請儘早告知，以先登記的同學為優先。
選擇這個項目的同學，學期成績加2分
口頭報告時，希望同學們至少能參與線上觀看，並將做為第五次作業的其中一題。

(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 5月29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 6月12日

以上(1), (2), (3), (4) 不管選哪個題目，若有做實驗模擬，請附上程式碼，會有額外的加分 (鼓勵不強制)

(5) 程式編寫，協助信號處理程式資料庫的建立

選擇 Page 10 當中其中一組題目，來編寫相關的程式，程式用 Matlab, Python, 或 C 編寫皆可 (共6組題目，6x3 = 18個名額)

選擇這個題目的同學，期末要用 NTUCool 交程式，並另外寫一個 ReadMe 的檔案，說明程式該如何執行，並舉例子顯示程式執行結果。

Tutorial 可供選擇的題目(可以略做修改)

(1) Automatic Music Evaluation

(2) Transformer in Natural Language Processing

(3) Speech Recognition in Multi-Speaker Scenario

(4) Color Coordinate Transform

(5) Advanced Multimedia Security Techniques

(6) Semantic Segmentation

(7) Instance and Panoptic Segmentation

(8) Panorama Image Processing

(9) Reflection Removal in Image

(10) Alternating Direction Method of Multipliers (ADMM) for Optimization

(11) Optimization for L_0 Norm Problems

(12) Beam Forming

Tutorial 可供選擇的題目(可以略做修改)

(13) BM3D Image Denoising Method

(14) Primitive Polynomial

(15) Galois field

(16) Biosignature Identification

(17) Electroretinogram (ERG)

(18) Electrooculogram (EOG)

程式編寫可供選擇的題目

(有意願的同學可選擇其中一組，用 Matlab, Python，或 C++皆可，要加說明檔 ReadMe，不可 copy 網路程式)

(1) (i) Step Invariance IIR Filter Design

(ii) MSE FIR Filter Design with Weights and Transition Bands

(2) (i) Minimax Filter for Types II

(ii) Minimax Filter for Types III

(iii) Minimax Filter for Types IV

(3) (i) Use the DCT to Compute the DFT for Even Inputs

(ii) Use the DCT to Compute the DFT for Odd Inputs

(iii) Discrete-Hartley Transform

(4) (i) Change the Time of a Signal without Varing the Frequency

(ii) Change the Frequency of a Signal without Varing the Time

(5) (i) Sectioned Convolution

(ii) Use the Recursive Method to Implement the Convolution with $a^n u[n]$

(6) (i) Orthogonal Frequency-Division Multiplexing

(ii) Modulation and Demodulation by CDMA Using Walsh Bases

Matlab Program

Download: 請洽台大各系所

參考書目

洪維恩，Matlab 程式設計，旗標，台北市，2013． (合適的入門書)

張智星，Matlab 程式設計入門篇，第四版，碁峰，2016.

預計看書學習所花時間： 3~5 天

Python Program

Download: https://www.python.org/

參考書目

葉難， Python程式設計入門，博碩，2015

黃健庭， Python程式設計：從入門到進階應用 ，全華，2020

The Python Tutorial https://docs.python.org/3/tutorial/index.html

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:

Fourier transform: $X(f) = \int_{-\infty}^{\infty} x(t) e^{-j2\pi f t} dt$

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 W)
- 4. Fast Algorithms
  - Basic (1W)
  - FFT and Convolution (2W)
  - Others (0.5W)
- 5. Orthogonal Transforms
  - Walsh Transform (1W)
  - Number Theory (0.8 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

- filter
  - digital
    - IIR (考慮aliasing)
      - bilinear transform
      - impulse invariant
      - step invariant
    - FIR
      - MSE (mean square error)
      - minimax
      - frequency sampling
  - analog (技術較早開發)

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

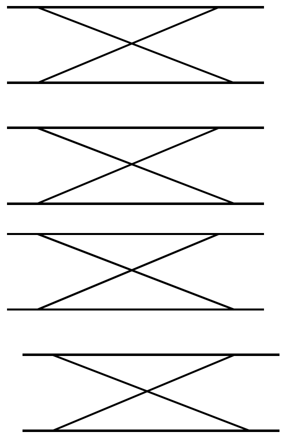
Example: one complex number multiplication

= ? 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？ $DFT\left(x[n]\right)=\sum_{n=0}^{N-1}x\left[n\right]e^{-j\frac{2\pi mn}{N}}$

- Walsh Transform
  (CDMA)

- Number Theoretic Transform

- Orthogonal Frequency-Division Multiplexing (OFDM)

- Code Division Multiple Access (CDMA)

Review 1: Four Types of the Fourier Transform

- **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

(1) Fourier Transform

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

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 frequencies are to satisfy:

$$e^{j2\pi f t}=e^{j2\pi f(t+T)}$$

頻率和 m 之間的關係：$f=\dfrac{m}{T}$

$\dfrac{1}{T}$ 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)

Review 2: Normalized Frequency

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

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

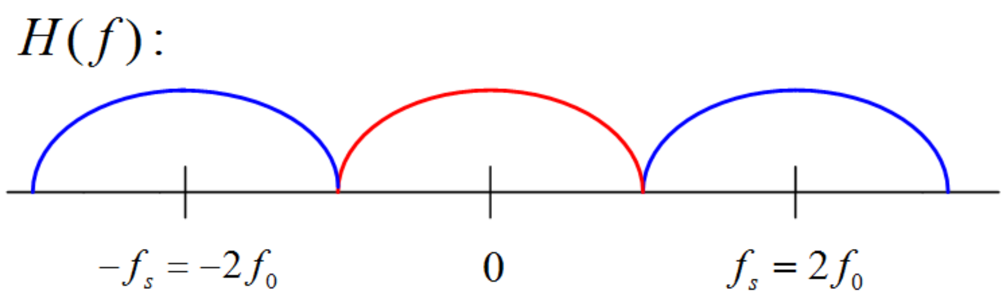
where $f_s = 1/\Delta_t$ (sampling frequency)

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 $0 < f < f_0$)

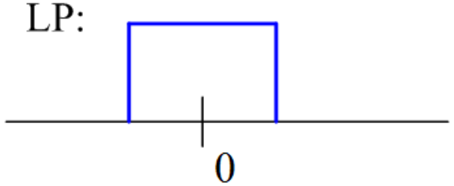
就可以知道全部的 $G(F)$

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

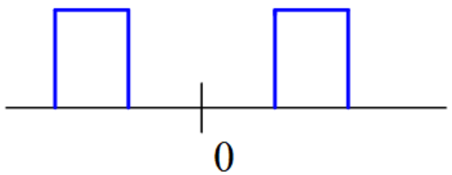
$g[n] = -g[-n]$ (odd) $\longrightarrow$ $G(F) = -G(-F)$

Analog filter: $H(f)$

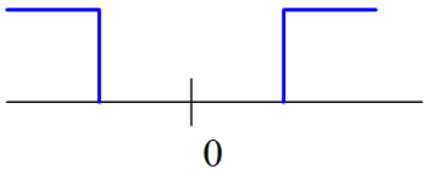
LP:



BP:

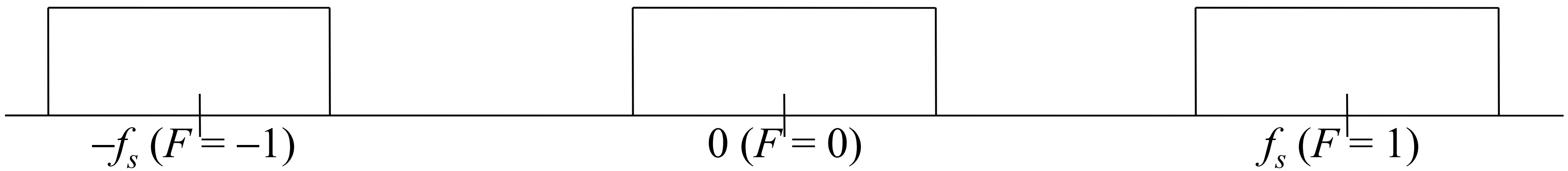


HP:

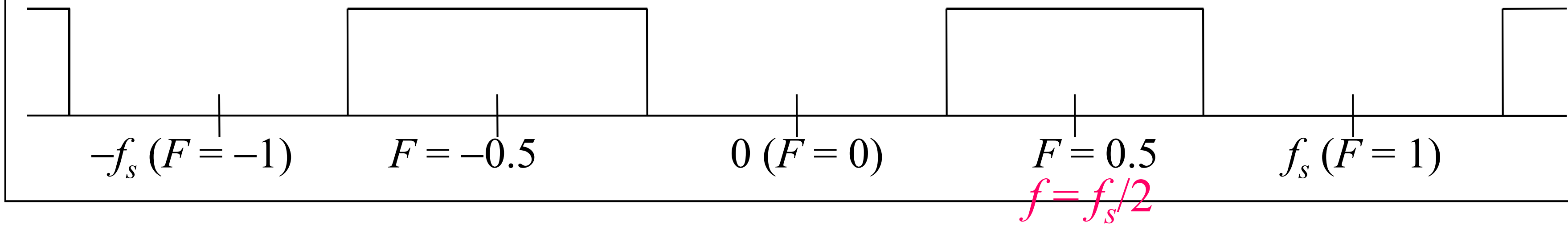


• Discrete time Fourier transform of the lowpass, highpass, and band pass filters

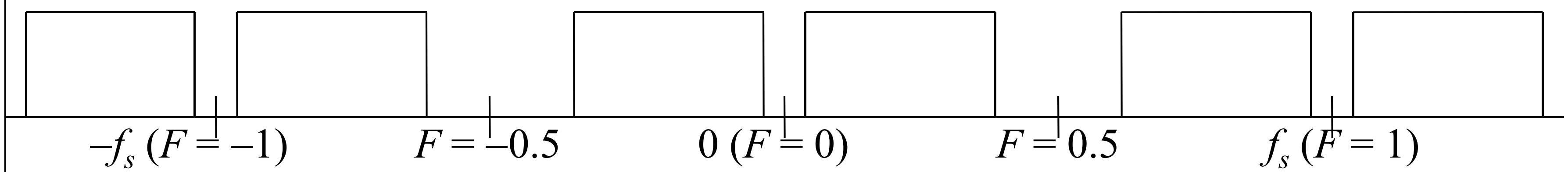
low pass filter（ **pass band 在 f_s 的整數倍附近** ）



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:

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

$s = j2\pi f$

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
- bilinear transform
- impulse invariant
- step invariant

Advantage:

Disadvantage:

An IIR Filter May Not be Hard to Implement

Ex : $h[n] = (0.9)^n$, for $n \geqq 0$, $h[n] = 0$, otherwise

$y[n] = x[n] * h[n]$

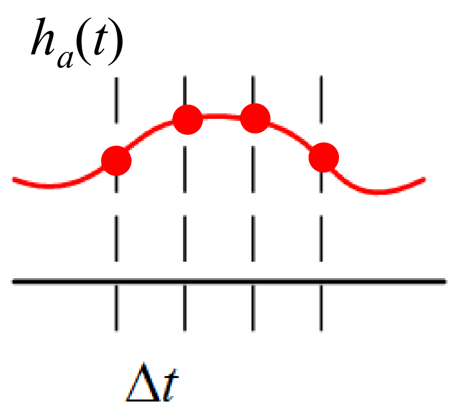
Z transform

Method 1: Impulse Invariance

白話一點，就是直接做 sampling

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

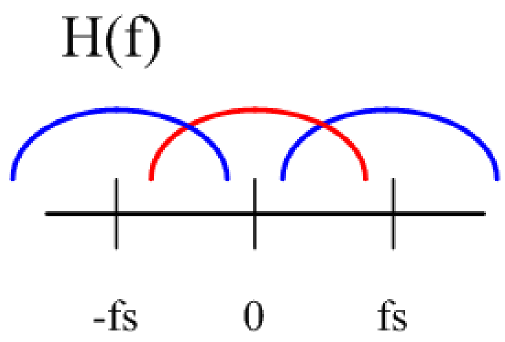
$$\boxed{h[n] = h_a(n\Delta_t)}$$



Advantage : Simple

Disadvantage : (1) infinite

(2)



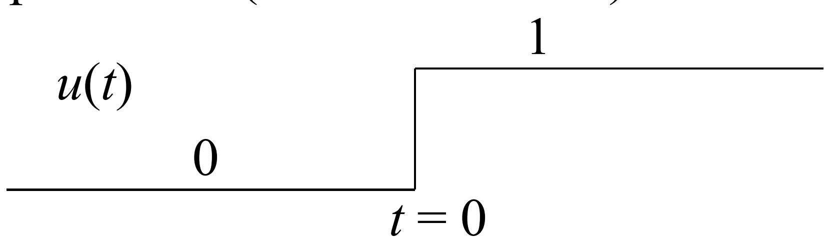
$f_s < 2B$

Method 2: Step Invariance

對 step function 的 response 作 sampling

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

step function (continuous form)



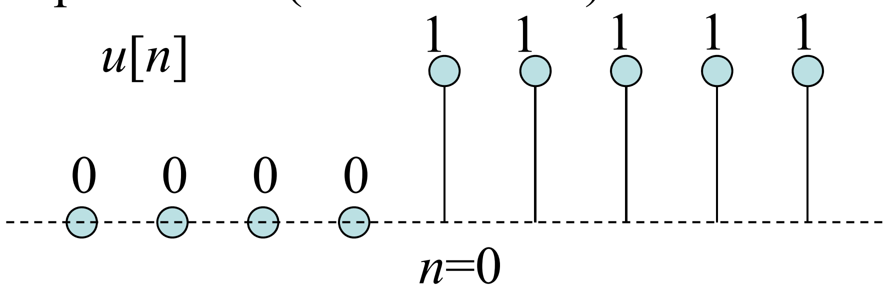
Laplace transform of $u(t)$:

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

Fourier transform of $u(t)$:

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

step function (discrete form)



Z transform of $u[n]$:

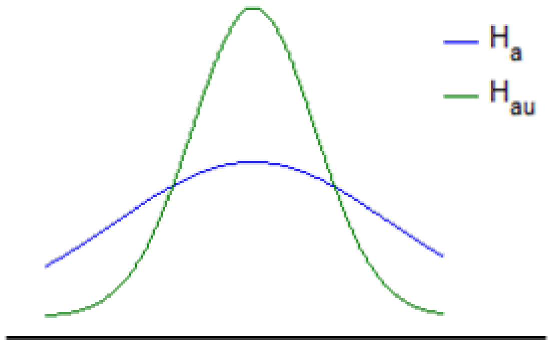
$$\frac{1}{1-z^{-1}}$$

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 $h_u[n] = h[n] * u[n]$

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

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

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 $H(f)$,

$$\boxed{H\left(f_{new}\right) = H_a\left(f_{old}\right)} \qquad f_{old} \in (-\infty, \infty)$$

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

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

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

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

s: index of the Laplace transform

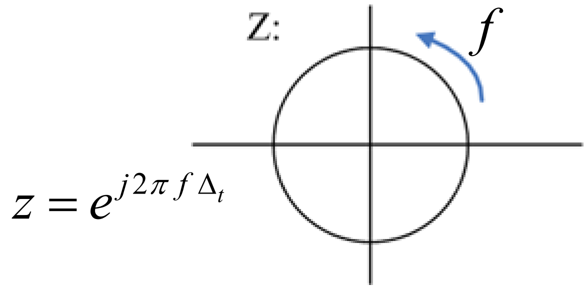
z: index of the Z transform

c: some constant

$$h_a(t) \xrightarrow{Laplace} H_a(s) \rightarrow H(z) = H_a\left(c\frac{1-z^{-1}}{1+z^{-1}}\right) \xrightarrow[Z-transform]{Inverse} h[n]$$

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

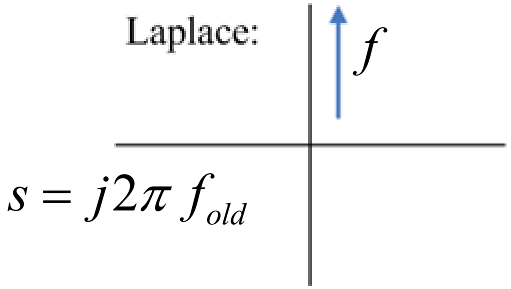
参考page 28、page29



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

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

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



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

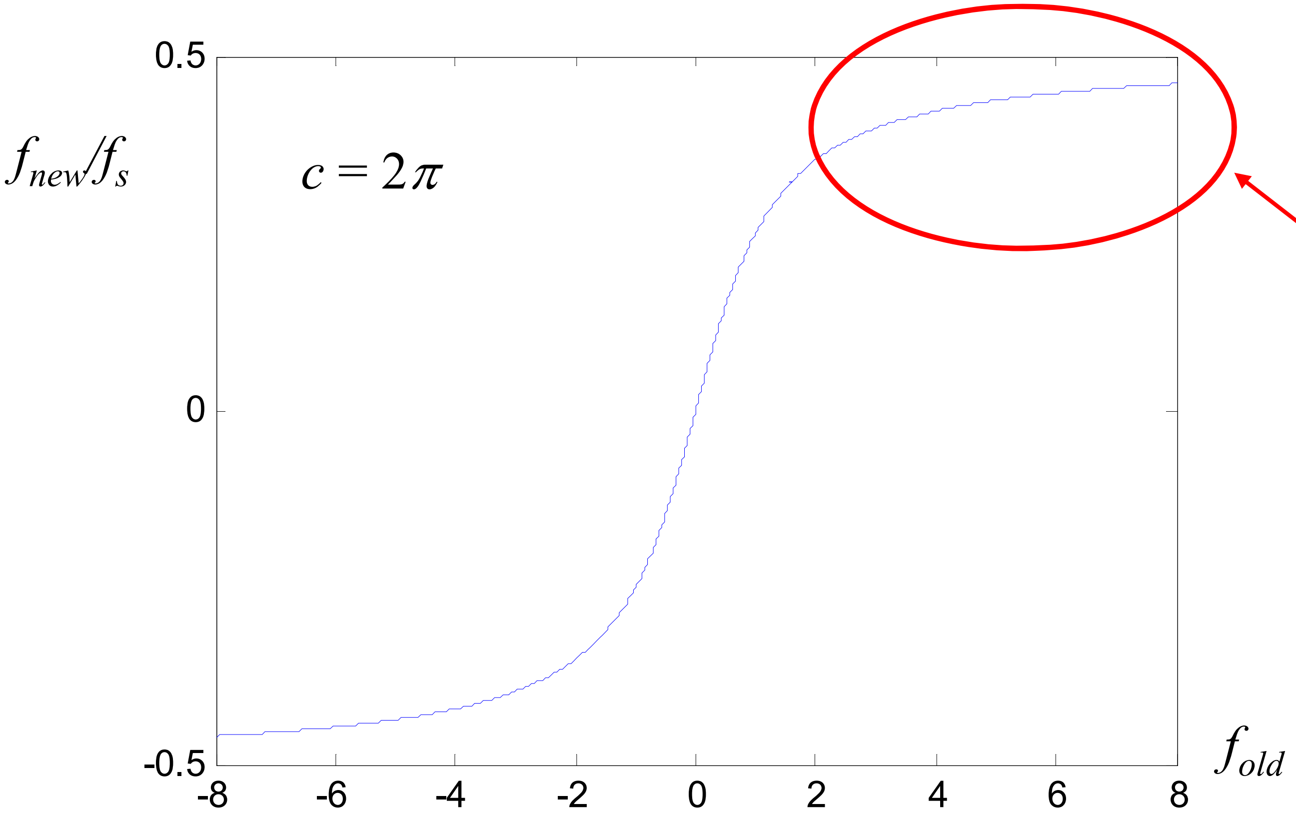
- Suppose that the Laplace transform of the analog filter $h_a(t)$ is $H_{a,L}(s)$

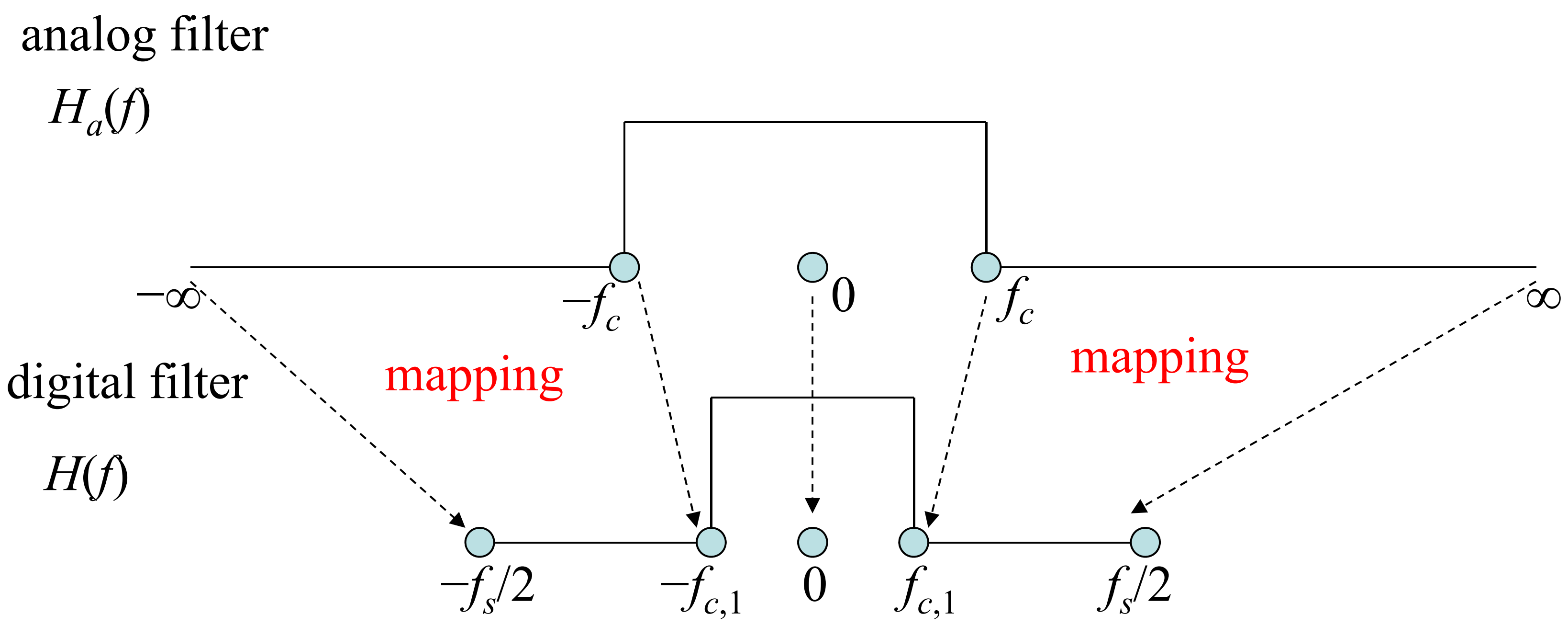
  The Z transform of the digital filter $h[n]$ is $H_z(z)$

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

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

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) Applications: 這個方法要用來處理什麼問題，有什麼應用

(5) Disadvantages: 這方法的缺點是什麼

(5-1) Why? 造成這些缺點的原因是什麼

(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