V. Homomorphic Signal Processing

◉ 5-A Homomorphism

Homomorphism is a way of "carrying over" operations from one algebra system into another.
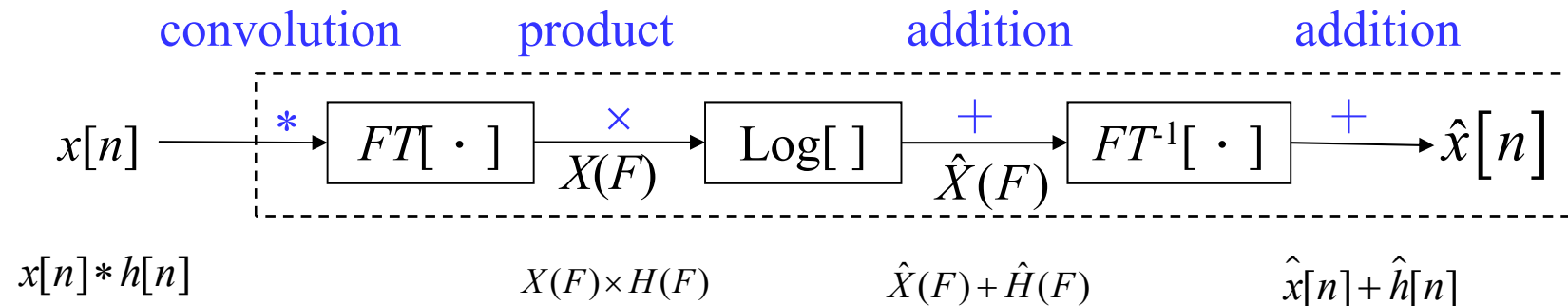
Ex.   convulution $\xrightarrow{\quad Fourier \quad}$ multiplication $\xrightarrow{\quad log \quad}$ addition

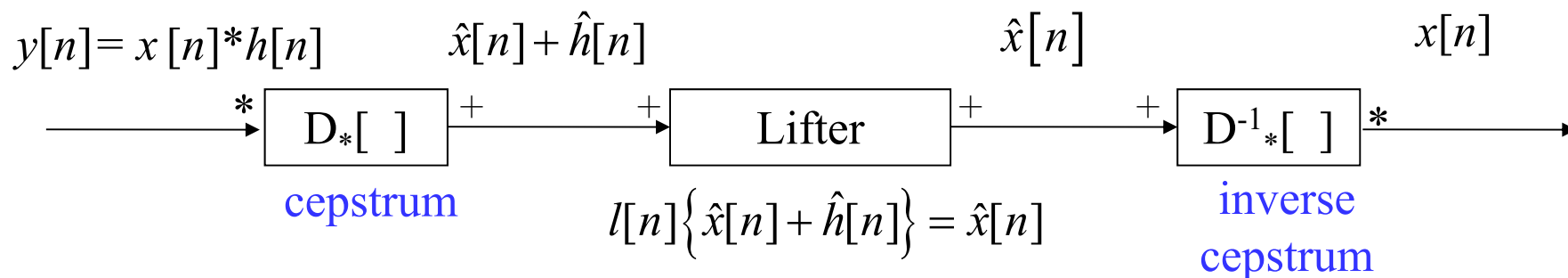把複雜的運算，變成效能相同但較簡單的運算

⊙ 5-B Cepstrum

$$\hat{X}(Z)\Big|_{z=e^{i2\pi F}} = \log X(Z)\Big|_{z=e^{i2\pi F}} = \log \left| X(Z) \right|_{z=e^{i2\pi F}} + j \arg[X(e^{i2\pi F})]$$

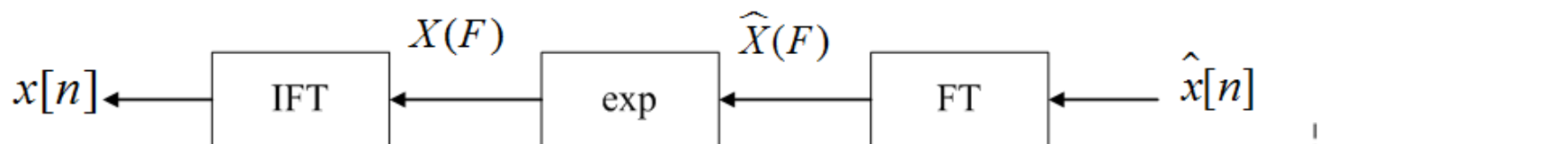For the process of cepstrum (denoted by D∗[·])

convolution	product	addition	addition
$x[n] \xrightarrow{\ *\ }$	$FT[\ \cdot\ ] \xrightarrow{\ \times\ } X(F)$	$\text{Log}[\ ] \xrightarrow{\ +\ } \hat{X}(F)$	$FT^{-1}[\ \cdot\ ] \xrightarrow{\ +\ } \hat{x}[n]$

$x[n]*h[n]$ $X(F)\times H(F)$ $\hat{X}(F)+\hat{H}(F)$ $\hat{x}[n]+\hat{h}[n]$

FT: discrete-time Fourier transform

- 由 $y[n]=x[n]*h[n]$ 重建 $x[n]$

$y[n]=x[n]*h[n]$ $\quad\quad \hat{x}[n]+\hat{h}[n]$ $\quad\quad\quad\quad \hat{x}[n]$ $\quad\quad\quad\quad x[n]$

$*$ → D$_*$[ ] $\xrightarrow{+}$ $\xrightarrow{+}$ Lifter $\xrightarrow{+}$ $\xrightarrow{+}$ D^{-1}$_*$[ ] $\xrightarrow{*}$

cepstrum $\quad\quad$ $l[n]\{\hat{x}[n]+\hat{h}[n]\}=\hat{x}[n]$ $\quad\quad$ inverse cepstrum

For the <u>inverse cepstrum D^{-1}$_*$[·]</u>

$x[n] \leftarrow$ IFT $\xleftarrow{X(F)}$ exp $\xleftarrow{\widehat{X}(F)}$ FT $\xleftarrow{}$ $\hat{x}[n]$

- 有趣的名詞

$\hat{x}[n]$ $\quad$ cepstrum

n $\quad$ quefrency

$l[n]$ $\quad$ lifter

$\hat{x}[n]$ $\quad\quad$ $\hat{h}[n]$ $\quad\quad$ n
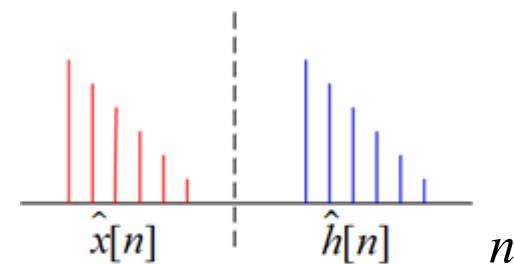
lifter $\quad$ $l[n]$ $\quad\quad$ n

$l[n]\{\hat{x}[n]+\hat{h}[n]\}=\hat{x}[n]$

⊙ 5-C Methods for Computing the Cepstrum

● **Method 1**: Compute the inverse discrete time Fourier transform:

$$\hat{x}[n] = \int_{-1/2}^{1/2} \hat{X}(F) e^{i 2\pi n F} dF \qquad : \text{inverse F.T}$$

$$\text{where} \quad \hat{X}(F) = \log|X(F)| + j \arg[X(F)]$$

ambiguity for phase

Problems: (1)

(2)

Actually, the COMPLEX Cepstrum is REAL for real input

● Method 2 (From Poles and Zeros of the Z Transform)

實際上計算 cepstrum的方法

time delay

$$X(Z) = \frac{A \, Z^r \prod\limits_{k=1}^{m_i}(1-a_k Z^{-1}) \prod\limits_{k=1}^{m_0}(1-b_k Z)}{\prod\limits_{k=1}^{P_i}(1-c_k Z^{-1}) \prod\limits_{k=1}^{P_0}(1-d_k Z)}$$

where

$$\left|a_k\right| , \left|b_k\right| , \left|c_k\right| , \left|d_k\right| \leq 1$$

a_k: <u>zeros inside unit circle</u>

c_k: <u>poles inside unit circle</u>

b_k^{-1}: <u>zeros outside unit circle</u>

d_k^{-1}: <u>poles outside unit circle</u>

$$\therefore \hat{X}(Z) = \log X(Z) = \log A + r \cdot \log Z + \sum_{k=1}^{m_i}\log(1-a_k Z^{-1}) + \sum_{k=1}^{m_0}\log(1-b_k Z)$$

$$-\sum_{k=1}^{P_i}\log(1-c_k Z^{-1}) - \sum_{k=1}^{P_0}\log(1-d_k Z)$$

$$\therefore \hat{X}(Z) = \log X(Z) = \log A + \cancel{r \cdot \log Z} + \sum_{k=1}^{m_i} \log(1 - a_k Z^{-1}) + \sum_{k=1}^{m_0} \log(1 - b_k Z)$$

$$-\sum_{k=1}^{P_i} \log(1 - c_k Z^{-1}) - \sum_{k=1}^{P_0} \log(1 - d_k Z)$$

Z^{-1}
(inverse Z transform)

?

Taylor series

$$f(t) = f(t_0) + \sum_{n=1}^{\infty} \frac{f^{(n)}(t_0)}{n!}(t - t_0)^n$$

$\boxed{\text{Taylor series expansion}}$ $\Big| Z^{-1}$ (Suppose that $r = 0$)

$$\hat{x}[n] = \begin{cases} \log(A) & , n = 0 \\[2mm] -\sum_{k=1}^{m_i} \dfrac{a_k^{\;n}}{n} + \sum_{k=1}^{P_i} \dfrac{c_k^{\;n}}{n} & , n > 0 \\[4mm] \sum_{k=1}^{m_0} \dfrac{b_k^{\;-n}}{n} - \sum_{k=1}^{P_0} \dfrac{d_k^{\;-n}}{n} & , n < 0 \end{cases}$$

Poles & zeros inside unit circle, right-sided sequence

Poles & zeros outside unit circle, left-sided sequence

Note:

(1) $\hat{x}[n]$ always decays with $|n|$.

(2) 在 complex cepstrum domain
<u>Minimum phase</u> 及 maximum phase 之貢獻以 $n = 0$ 為分界切開

(3) For FIR case, there is no c_k and d_k

(4) The complex cepstrum is unique and of infinite duration for both positive & negative n, even though $x[n]$ is causal & of finite durations

$\boxed{\hat{x}[n] \text{ is always IIR}}$

● **Method 3**

$$Z \cdot \hat{X}'(Z) = Z \cdot \frac{X'(Z)}{X(Z)}$$

$$\therefore ZX'(Z) = Z\hat{X}'(Z) \cdot X(Z)$$

$$\downarrow \quad Z^{-1}$$

$$n\, x[n] = \sum_{k=-\infty}^{\infty} k\, \hat{x}[k]\, x[n-k]$$

$$\therefore x[n] = \sum_{k=-\infty}^{\infty} \frac{k}{n} \hat{x}[k]\, x[n-k] \qquad for\ n \neq 0$$

Suppose that $x[n]$ is <u>causal</u> and has <u>minimum phase</u>, i.e. $x[n] = \hat{x}[n] = 0$, $n < 0$

$$x[n] = \sum_{k=-\infty}^{\infty} \frac{k}{n} \hat{x}[k] x[n-k] \qquad \text{for } n \neq 0$$

$$\Rightarrow x[n] = \sum_{k=0}^{n} \frac{k}{n} \hat{x}[k] x[n-k] \qquad \text{for } n > 0 \qquad (\text{causal sequence})$$

$$x[n] = \hat{x}[n] x[0] + \sum_{k=0}^{n-1} \frac{k}{n} \hat{x}[k] x[n-k]$$

For a minimum phase sequence $x[n]$

$$\hat{x}[n] = \begin{cases} 0 & , n < 0 \\ \dfrac{x[n]}{x[0]} - \displaystyle\sum_{k=0}^{n-1} (\frac{k}{n}) \hat{x}[k] \dfrac{x[n-k]}{x[0]} & , n > 0 \\ \log A & , n = 0 \end{cases} \qquad \text{recursive method}$$

Determining $\hat{x}[n]$ from $\hat{x}[0], \hat{x}[1], \cdots, \hat{x}[n-1]$

For <u>anti-causal</u> and <u>maximum phase</u> sequence, $x[n] = \hat{x}[n] = 0,\ n > 0$

$$x[n] = \sum_{k=n}^{0} \frac{k}{n}\hat{x}[k]\,x[n-k] \qquad\qquad ,n < 0$$

$$= \hat{x}[n]\,x[0] + \sum_{k=n+1}^{0} \frac{k}{n}\hat{x}[k]\,x[n-k]$$

For maximum phase sequence,

$$\hat{x}[n] = \begin{cases} 0 & ,n > 0 \\ \log A & ,n = 0 \\ \dfrac{x[n]}{x[0]} - \displaystyle\sum_{k=n+1}^{0} \left(\dfrac{k}{n}\right)\hat{x}[k]\dfrac{x[n-k]}{x[0]} & ,n < 0 \end{cases}$$

⊙ 5-D Properties

P.1) The complex cepstrum decays at least as fast as $\dfrac{1}{n}$

$$\left|\hat{x}[n]\right| < c \left|\frac{\alpha^{n}}{n}\right| \qquad -\infty < n < \infty$$

$$\alpha = \max\left(\left|a_{k}\right|, \left|b_{k}\right|, \left|c_{k}\right|, \left|d_{k}\right|\right)$$

P.2) If $X(Z)$ has no poles and zeros outside the unit circle, i.e. $x[n]$ is minimum phase, then
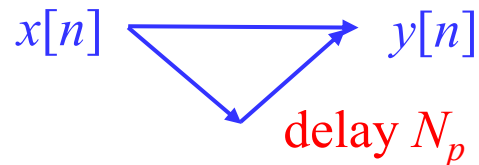
$$\hat{x}[n] = 0 \qquad \qquad \textit{for all } n < 0$$

because of no b_{k}, d_{k}

P.3) If $X(Z)$ has no poles and zeros inside the unit circle, i.e. $x[n]$ is maximum phase, then

$$\hat{x}[n] = 0 \qquad \qquad \textit{for all } n > 0$$

because of no a_{k}, c_{k}

P.4) If $x[n]$ is of finite duration, then

$$\hat{x}[n] \text{ has infinite duration}$$

⊙ 5-E Application of Homomorphic Deconvolution

(1) Equalization for Echo

$$y[n] = x[n] + \alpha\, x[n - N_p]$$

$x[n]$ $y[n]$

delay N_p

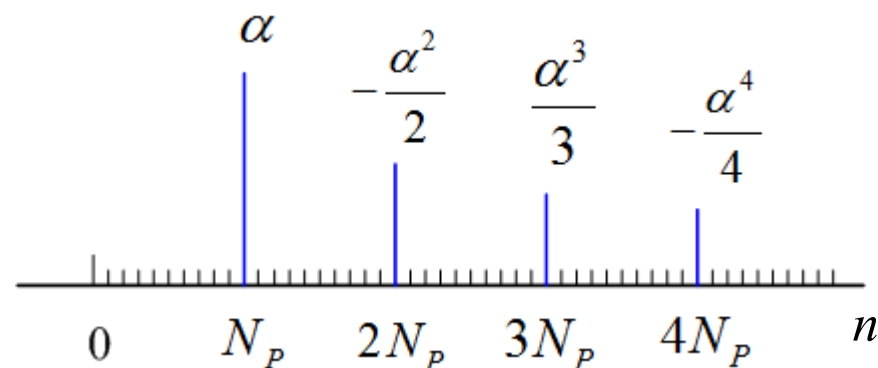Let $p[n]$ be $p[n] = \delta[n] + \alpha\delta[n\text{-}N_p]$

$$y[n] = x[n] + \alpha\, x[n\text{-}N_p] = x[n] * p[n]$$
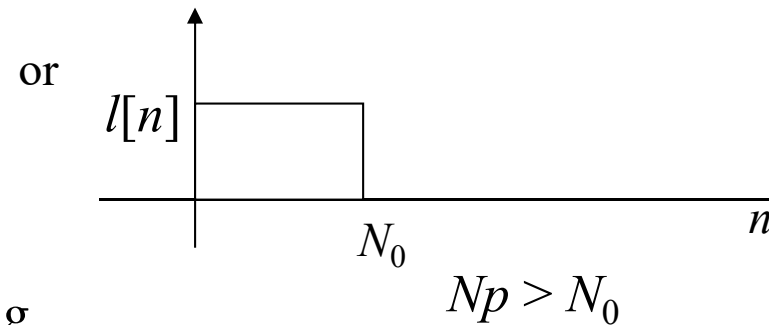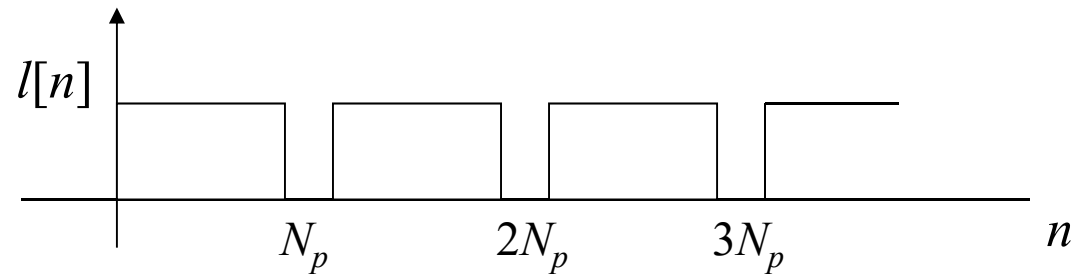
$$P(Z) = 1 + \alpha Z^{-N_p}$$

$$\hat{P}(Z) = \log(1 + \alpha Z^{-N_p}) = \sum_{k=1}^{\infty} (-1)^{k+1} \frac{\alpha^k}{k} Z^{-kN_p}$$

$\downarrow Z^{-1}$

$$\hat{p}[n] = \sum_{k=1}^{\infty} (-1)^{k+1} \frac{\alpha^k}{k} \delta(n - k \cdot N_p)$$

α $-\dfrac{\alpha^2}{2}$ $\dfrac{\alpha^3}{3}$ $-\dfrac{\alpha^4}{4}$

0 N_P $2N_P$ $3N_P$ $4N_P$ n

Filtering out the echo by the following "lifter":

$l[n]$

N_p $2N_p$ $3N_p$ n

or

$l[n]$

N_0 n

Q: For the case where N_p is unknown

$Np > N_0$

(2) Representation of acoustic engineering

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

<u>Synthesiz</u>   <u>music</u>   <u>building effect：e.g. 羅馬大教堂的</u>
<u>ed music</u>   <u>impulse response</u>

(3) Speech analysis

$$s[n] \quad = \quad g[n] \quad * \quad v[n] \quad * \quad p[n]$$

<u>Speech wave</u>   <u>Global wave shape</u>   <u>Vocal tract impulse</u>   <u>Pitch</u>

They can be separated by filtering in the complex cepstrum domain

(4) Seismic Signals

(5) Multiple-path analysis for any wave-propagation problem

用 cepstrum 將 multipath的影響去除

From 王小川，"語音訊號處理"，全華出版，台北，民國94年。
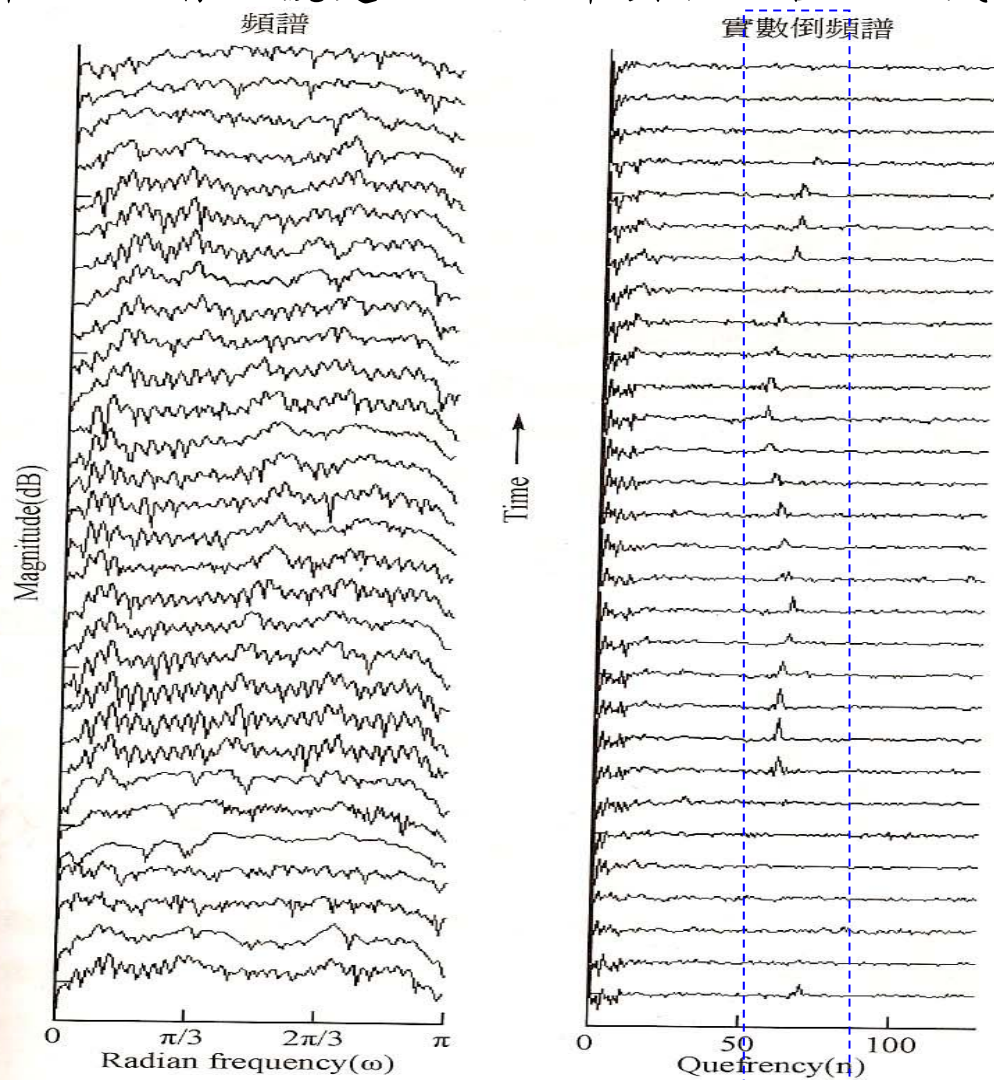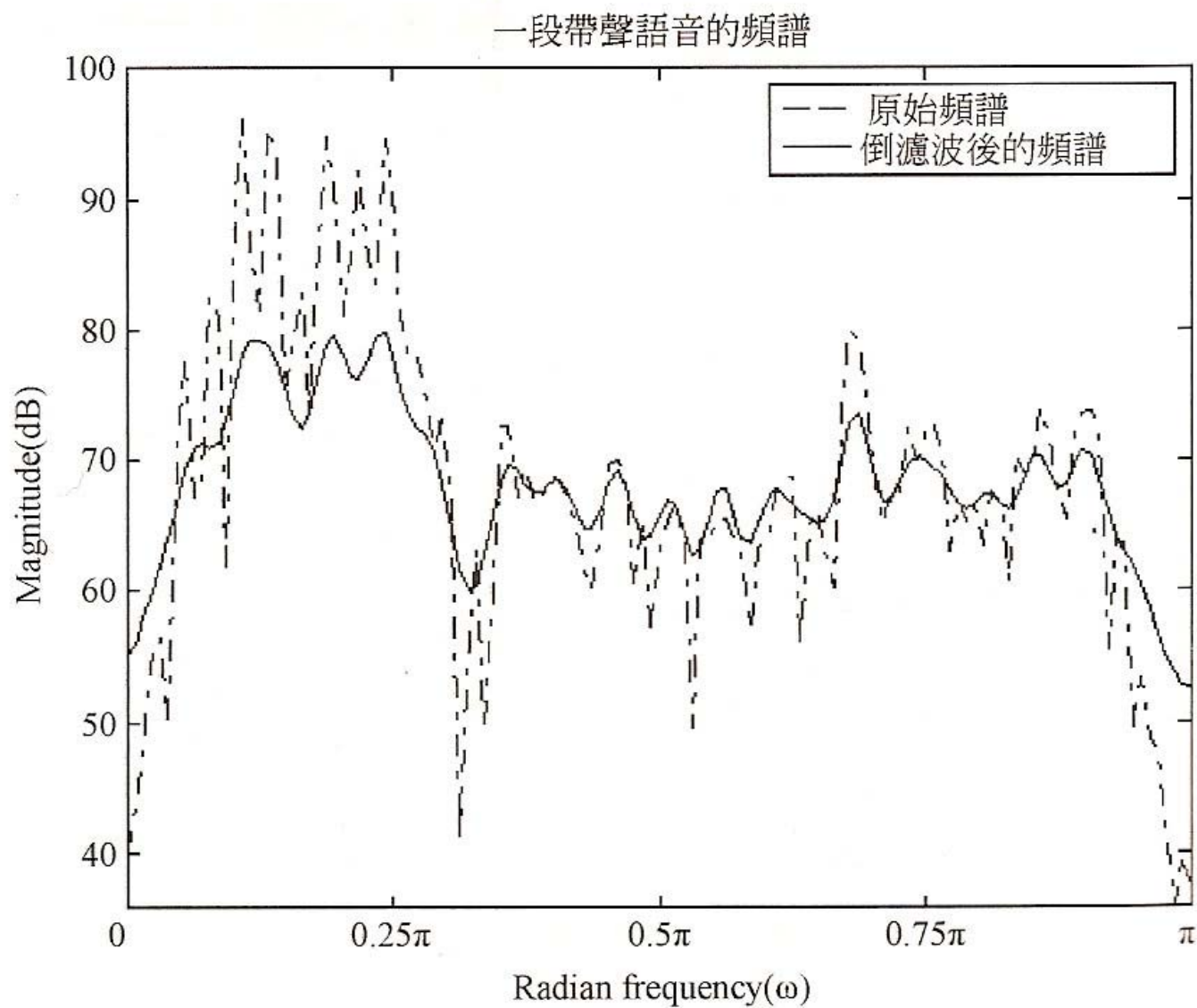

圖 7-5　元音頻譜與倒頻譜

From 王小川，"語音訊號處理"，全華出版，台北，民國94年。

一段帶聲語音的頻譜

圖 7-6　經過倒濾波器作平滑處理的頻譜

⊙ 5-F Problems of Cepstrum

(1) $| \log(X(Z))|$

(2) Phase

(3) Delay Z^{-k}

(4) Only suitable for the multiple-path-like problem

⊙ 5-G Differential Cepstrum

$$\hat{x}_d(n) = Z^{-1}\left[\frac{X'(Z)}{X(Z)}\right] \qquad \text{或} \qquad \hat{x}_d[n] = \int_{-1/2}^{1/2}\frac{X'(F)}{X(F)}e^{i2\pi F}\,dF$$

inverse Z transform

Note: $\dfrac{d}{dZ}\hat{X}(Z) = \dfrac{d}{dZ}\log(X(Z)) = \dfrac{X'(Z)}{X(Z)}$

If $x(n) = x_1(n) * x_2(n)$

$X(Z) = X_1(Z)\cdot X_2(Z)$

$X'(Z) = X_1'(Z)\cdot X_2(Z) + X_1(Z)\cdot X_2'(Z)$

$\dfrac{X'(Z)}{X(Z)} = \dfrac{X_1'(Z)}{X_1(Z)} + \dfrac{X_2'(Z)}{X_2(Z)}$ $\therefore \quad \hat{x}_d(n) = \hat{x}_{1d}(n) + \hat{x}_{2d}(n)$

Advantages: no phase ambiguity

able to deal with the delay problem

- **Properties of Differential Cepstrum**

(1) The differential Cepstrum is shift & scaling invariant

不只適用於 multi-path-like problem

也適用於 pattern recognition

If $y[n] = A\,X[n - r]$

$$\Rightarrow \quad \hat{y}_d(n) = \begin{cases} \hat{x}_d(n) & ,\quad n \neq 1 \\ -r + \hat{x}_d(1) & ,\quad n = 1 \end{cases}$$

(Proof): $Y(z) = Az^{-r}X(z)$

$$Y'(z) = Az^{-r}X'(z) - rAz^{-r-1}X(z)$$

$$\frac{Y'(z)}{Y(z)} = \frac{X'(z)}{X(z)} - rz^{-1}$$

$$y_d(n) = x_d(n) - r\delta(n-1)$$

(2) The complex cepstrum $\hat{C}[n]$ is closely related to its differential cepstrum $\hat{x}_d[n]$ and the signal original sequence $x[n]$

$$\hat{C}(n) = \frac{-\hat{x}_d(n+1)}{n} \qquad\qquad n \neq 0 \qquad\qquad \textit{diff cepstrum}$$

$$\textit{and} \quad -(n-1)\,x(n-1) = \sum_{k=-\infty}^{\infty} \hat{x}_d(n)\,x(n-k) \qquad \textit{recursive formula}$$

Complex cepstrum 做得到的事情, differential cepstrum 也做得到！

(3) If $x[n]$ is minimum phase (no poles & zeros outside the unit circle), then
$\hat{x}_d[n] = 0$ for $n \leq 0$

(4) If $x[n]$ is maximum phase (no poles & zeros inside the unit circle) , then
$\hat{x}_d[n] = 0$ for $n \geq 2$

delay

max phase ↙ min phase

0　1　2
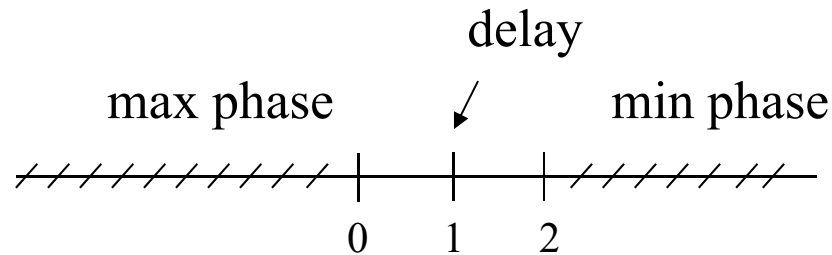
(5) If $x(n)$ is of finite duration, $\hat{x}_d[n]$ has infinite duration

Complex cepstrum decay rate $\propto \dfrac{1}{n}$

Differential Cepstrum decay rate 變慢了, $\because$ $\hat{x}_d(n+1) = n \cdot \hat{c}(n) \propto n \cdot \dfrac{1}{n} = 1$

⊙ 5-H Mel-Frequency Cepstrum (梅爾頻率倒頻譜)

Take log in the frequency mask

gain   mask of Mel-frequency cepstrum

$B_m[k]$

$f_1, f_2, f_3 \ldots\ldots$ 以等比級數增加

$$f_m = \alpha^m f_0$$

frequency
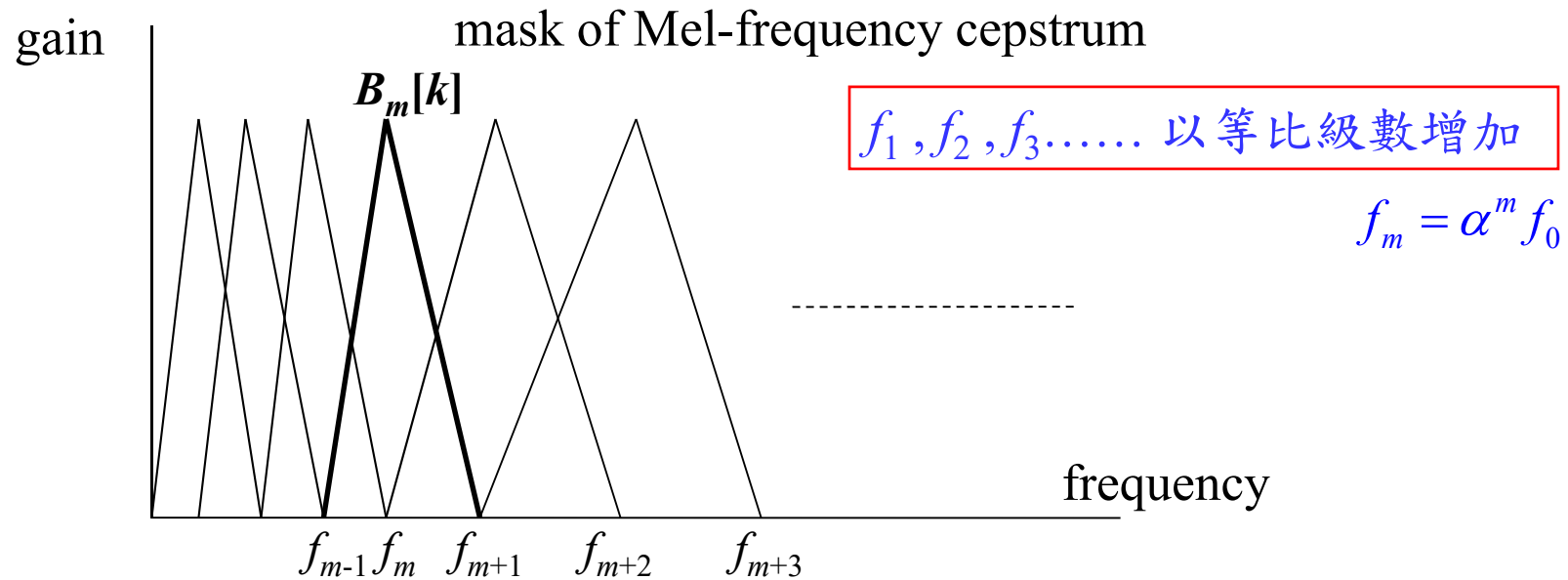
$f_{m-1} f_m \quad f_{m+1} \quad f_{m+2} \quad f_{m+3}$

$B_m[k] = 0$ for $f < f_{m-1}$ and $f > f_{m+1}$

$B_m[k] = (k - f_{m-1})/(f_m - f_{m-1})$ for $f_{m-1} \leq f \leq f_m$

$B_m[k] = (f_{m+1} - k)/(f_{m+1} - f_m)$ for $f_m \leq f \leq f_{m+1}$

$f = k f_s / N$

Process of the Mel-Cepstrum

summation of the effect inside the mth mask

(1) $\quad x[n] \xrightarrow{\;FT\;} X[k]$

(2) $\quad Y[m] = \log\left\{ \sum_{k=f_{m-1}}^{f_{m+1}} \left|X[k]\right|^2 B_m[k] \right\}$

(3) $\quad c_x[n] = \dfrac{1}{M} \sum_{m=1}^{M} Y[m] \cos\left(\dfrac{\pi n(m-1/2)}{M} \right)$

Q: What are the difference between the Mel-cepstrum and the original cepstrum?

Advantages :

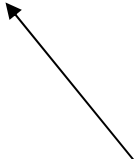Mel-frequency cepstrum 更接近人耳對語音的區別性

用 $c_x[1], c_x[2], c_x[3], \ldots\ldots, c_x[13]$ 即足以描述語音特徵

⊙ 5-I References

- R. B. Randall and J. Hee, "Cepstrum analysis," *Wireless World*, vol. 88, pp. 77-80. Feb. 1982

- 王小川，"語音訊號處理"，全華出版，台北，民國94年。

- A. V. Oppenheim and R. W. Schafer, *Discrete-Time Signal Processing*, London: Prentice-Hall, 3rd ed., 2010.

- S. C. Pei and S. T. Lu, "Design of minimum phase and FIR digital filters by differential cepstrum," *IEEE Trans. Circuits Syst. I*, vol. 33, no. 5, pp. 570-576, May 1986.

- S. Imai, "Cepstrum analysis synthesis on the Mel-frequency scale," *ICASSP*, vol. 8, pp. 93-96, Apr. 1983.

附錄六：聲音檔和影像檔的處理 (by Matlab)

A. 讀取聲音檔

- 電腦中，沒有經過壓縮的聲音檔都是 *.wav 的型態
  有經過壓縮的聲音檔是*.mp3的型態

- 讀取： audioread
  註：2015版本以後的 Matlab，**wavread** 將改為 **audioread**

- 例： [**x**, **fs**] = audioread(C:\WINDOWS\Media\Alarm01.wav');
  可以將 ringin.wav 以數字向量 **x** 來呈現。　**fs**: sampling frequency
  這個例子當中 size(**x**) = 122868　2　　　fs = 22050

- 思考: 所以，取樣間隔多大?

- 這個聲音檔有多少秒？　　　　　　　雙聲道 (**Stereo**，俗稱立體聲)

畫出聲音的波型

time = [0:size(x,1)-1]/fs;　% x 是前頁用 audioread 所讀出的向量

subplot(2,1,1); **plot(time, x(:,1))**; xlim([time(1),time(end)])
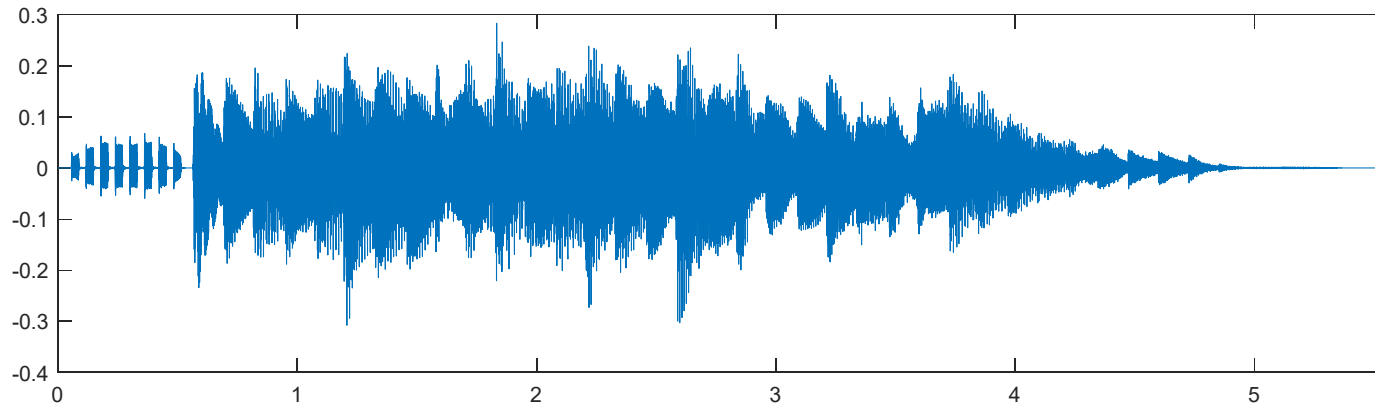
subplot(2,1,2); **plot(time, x(:,2))**; xlim([time(1),time(end)])
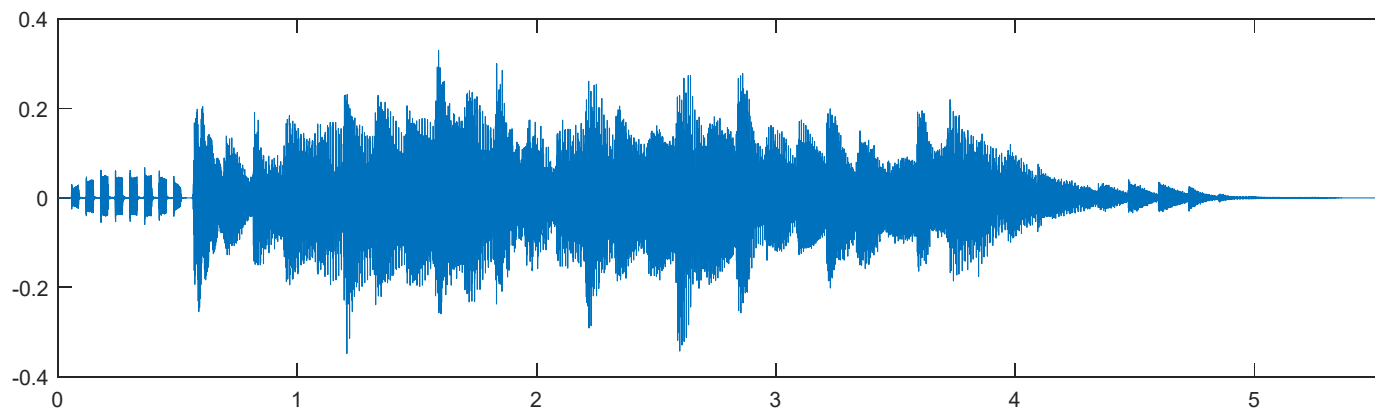
注意：*.wav 檔中所讀取的資料，值都在 −1 和 +1 之間

B. 繪出頻譜 (詳細方法請參考附錄二)

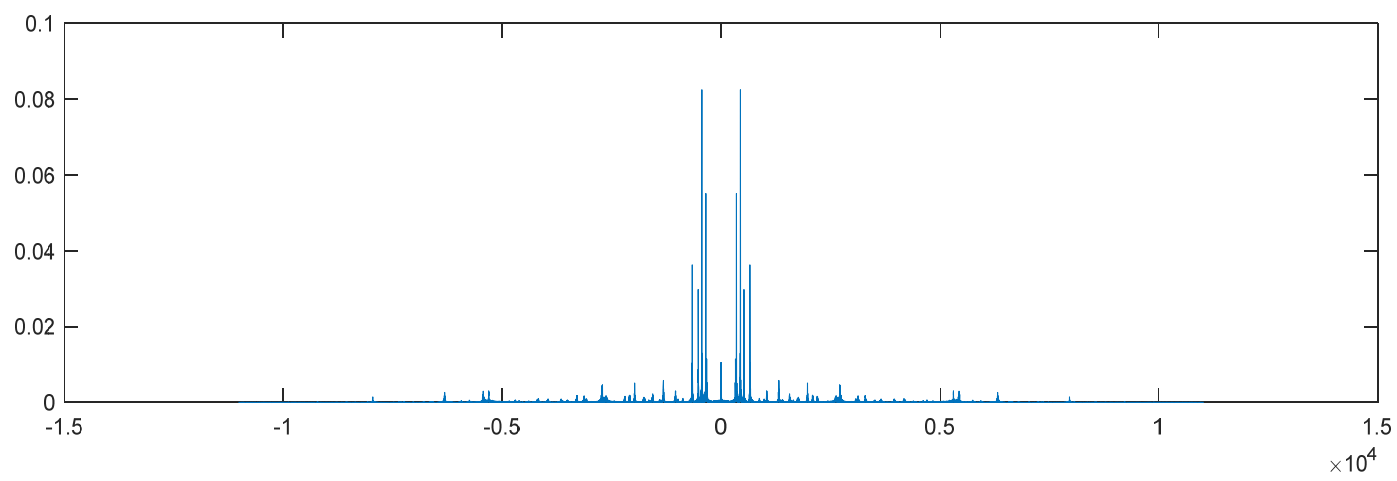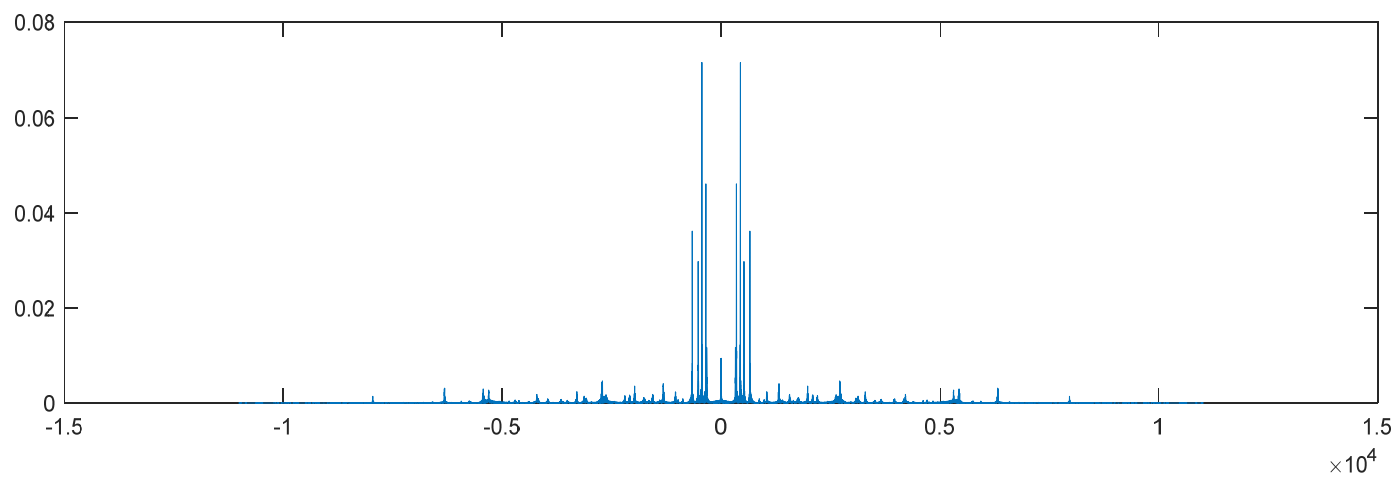**X = fft(x(:,1)); % 只做這一步無法得出正確的頻譜**

```
X=X.';
N=length(X); N1=round(N/2);
dt=1/fs;
X1=[X(N1+1:N),X(1:N1)]*dt;  % shifting for spectrum
f=[[N1:N-1]-N,0:N1-1]/N*fs;   % valid f
plot(f, abs(X1));
```
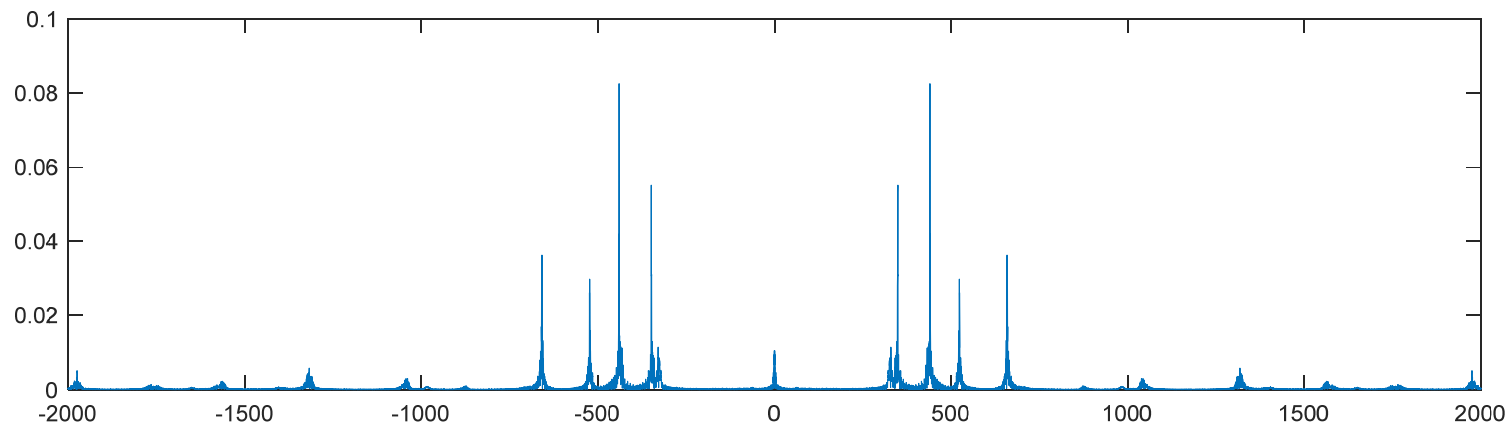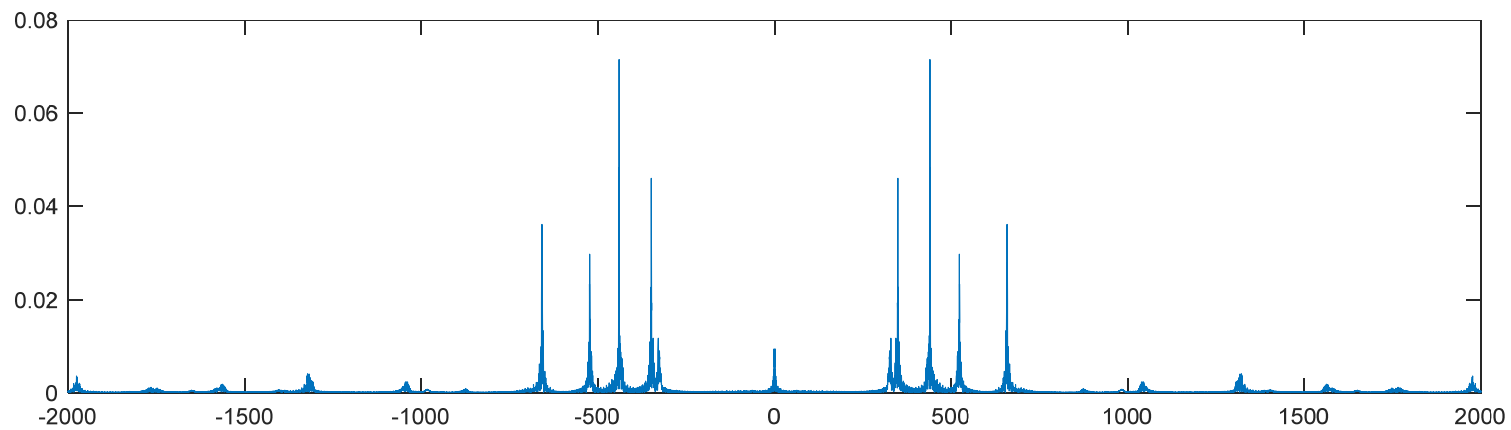
Alarm01.wav 的頻譜

Alarm01.wav 的頻譜

xlim([-2000,2000]) % 只看其中 -2000Hz ~ 2000Hz 的部分

C. 聲音的播放

(1) sound(**x**): 將 **x** 以 8192Hz 的頻率播放

(2) sound(**x, fs**): 將 **x** 以 **fs** Hz 的頻率播放

　Note: (1)~(3) 中 **x** 必需是1 個column (或2個 columns)，且 **x** 的值應該 介 於 −1 和 +1 之間

(3) soundsc(**x, fs**): 自動把 **x** 的值調到 −1 和 +1 之間 再播放

D. 用 Matlab 製作 *.wav 檔： audiowrite

audiowrite(filename, x, fs)

　　　將數據 **x** 變成一個 *.wav 檔，取樣速率為 **fs** Hz

① **x** 必需是1 個column (或2個 columns) ② **x** 值應該 介於 −1 和 +1

E. 用 Matlab 錄音的方法

錄音之前，要先將電腦接上麥克風，且確定電腦有音效卡
(部分的 notebooks 不需裝麥克風即可錄音)

範例程式：

```
Sec = 3;
Fs = 8000;
recorder = audiorecorder(Fs, 16, 1);
recordblocking(recorder, Sec);
audioarray = getaudiodata(recorder);
```

執行以上的程式，即可錄音。

錄音的時間為三秒，sampling frequency 為 8000 Hz

錄音結果為 audioarray，是一個 column vector (如果是雙聲道，則是
兩個 column vectors)

範例程式 (續)：

sound(audioarray, Fs);   % 播放錄音的結果

t = [0:length(audioarray)-1]./Fs;

plot (t, audioarray');   % 將錄音的結果用圖畫出來

xlabel('sec','FontSize',16);

audiowrite('test.wav', audioarray, Fs)   % 將錄音的結果存成 *.wav 檔

指令說明：

recorder = audiorecorder(Fs, nb, nch);　　(提供錄音相關的參數)

　　Fs: sampling frequency,

　　nb: using nb bits to record each data

　　nch: number of channels (1 or 2)

recordblocking(recorder, Sec);　　(錄音的指令)

　　recorder: the parameters obtained by the command "audiorecorder"

　　Sec: the time length for recording

audioarray = getaudiodata(recorder);

　　(將錄音的結果，變成 audioarray 這個 column vector，如果是雙聲道，則 audioarray 是兩個 column vectors)

以上這三個指令，要並用，才可以錄音

F：影像檔的處理

Image 檔讀取:　imread

Image 檔顯示:　imshow,　image,　imagesc

Image 檔製作:　imwrite

　　　　基本概念：灰階影像在 Matlab 當中是一個矩陣

　　　　　　　　彩色影像在 Matlab 當中是三個矩陣，分別代表 Red, Green, Blue

　　　　　　*.bmp:　沒有經過任何壓縮處理的圖檔

　　　　　　*.jpg:　　有經過 JPEG 壓縮的圖檔

Video 檔讀取:　aviread

範例一：(黑白影像)

im=double(imread('C:\Program Files\MATLAB\pic\Pepper.bmp'));
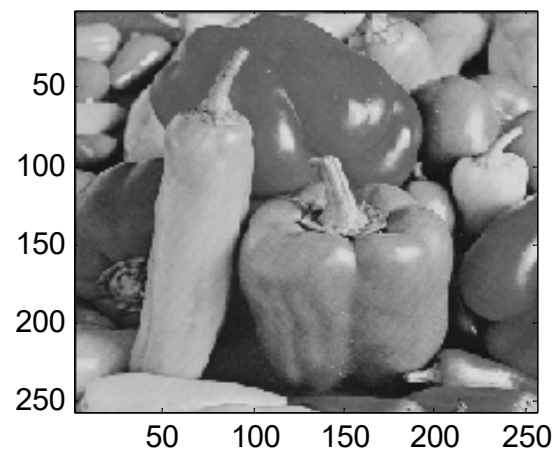
　　　(注意，如果 Pepper.bmp 是個灰階圖，im 將是一個矩陣)

size(im)　　　　(用 size 這個指令來看 im 這個矩陣的大小)

ans =

　256　256

image(im);
colormap(gray(256))

範例二：(彩色影像)

im2=double(imread('C:\Program Files\MATLAB\pic\Pepper512c.bmp'));

size(im2)　　　　　　(注意，由於這個圖檔是個彩色的，所以 im2 將由

ans =　　　　　　　三個矩陣複合而成)

　512　512　③

imshow(im);　　or　　image(im/255);

注意：要對影像做運算時，<u>要先變成 double 的格式</u>

否則電腦會預設影像為 integer 的格式，在做浮點運算時會產生誤差

例如，若要對影像做 2D Discrete Fourier transform

```
im=imread('C:\Program Files\MATLAB\pic\Pepper.bmp');
im=double(im);
Imf=fft2(im);
```

附錄七 聲音檔和影像檔的處理 (by Python)

可以先安裝幾個模組

pip install numpy
pip install scipy
pip install matplotlib # plot
pip install pipwin
pipwin install simpleaudio # vocal files
pipwin install pyaudio

PS: 謝謝2021年擔任助教的蔡昌廷同學協助製作

A. 讀音訊檔

要先import 相關模組： import scipy.io.wavfile as wavfile

讀取音檔：

fs, wave_data = wavfile.read('C:/WINDOWS/Media/Alarm01.wav')

  # fs: sampling frequency

  # If the audio file has one channel, then wave_data is a column vector

  # If the audio file has two channels, then wave_data has two column
    vectors

num_frame = len(wave_data) # 音訊長度:

n_channel = int(wave_data.size/ num_frame) # channels 數量

```
>>> fs
22050
>>> num_frame, n_channel
(1150416, 2)
```
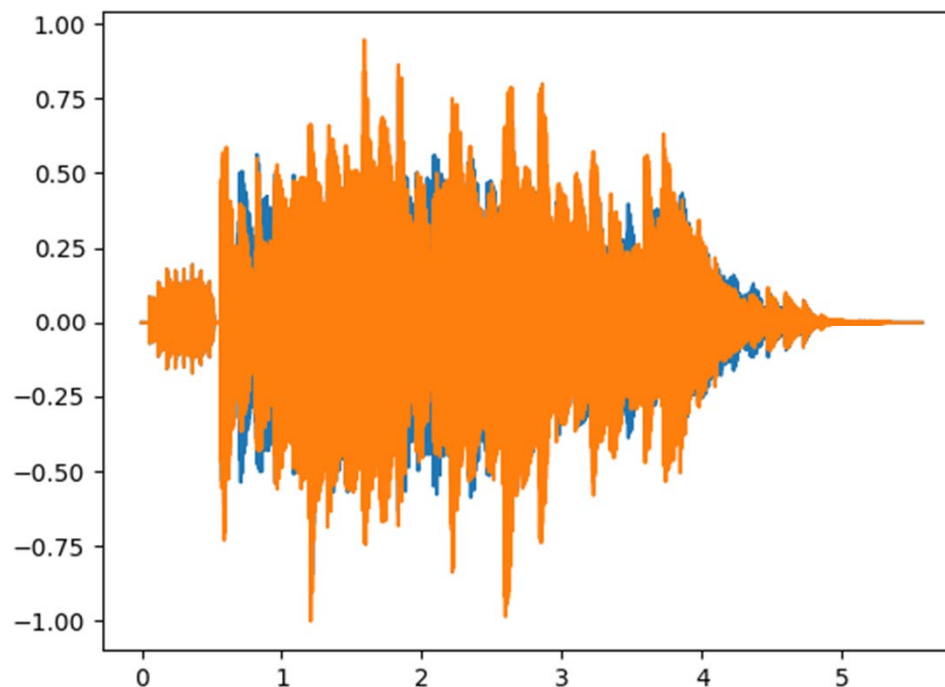
畫出音訊波形圖

要先import 相關模組： import matplotlib.pyplot as plt

- time = np.arange(0, num_frame)*1/fs

- plt.plot(time, wave_data)

- plt.show()
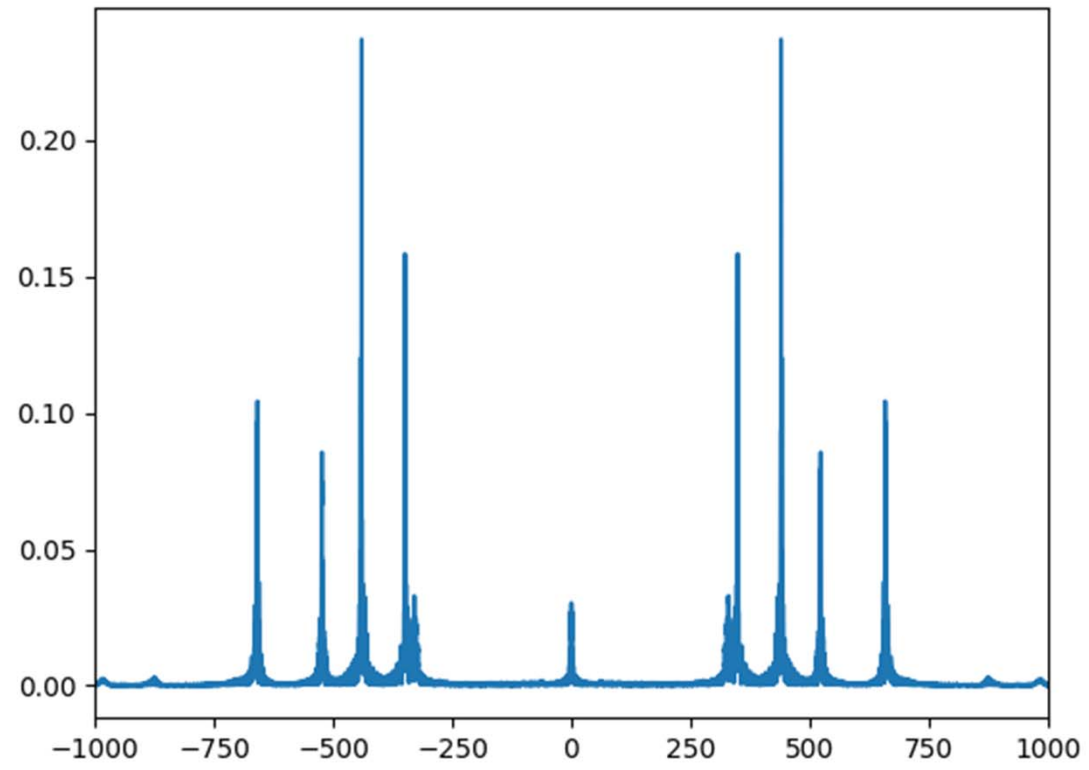
B. 畫出頻譜

要先import 相關模組： from scipy.fftpack import fft

- fft_data = abs(fft(wave_data[:,1]))/fs   # only choose the 1st channel

  # 注意要乘上 1/fs

- n0=int(np.ceil(num_frame/2))

- fft_data1=np.concatenate([fft_data[n0:num_frame],fft_data[0:n0]])

  # 將頻譜後面一半移到前面

- freq=np.concatenate([range(n0-num_frame,0),range(0,n0)])*fs/num_frame

  # 頻率軸跟著調整

- plt.plot(freq,fft_data1)

- plt.xlim(-1000,1000)   # 限制頻率的顯示範圍

- plt.show() # 如後圖

C. 播放聲音

要先import 相關模組： import simpleaudio as sa

- n_bytes =2   # using two bytes to record a data
- wave_data = (2**15-1)* wave_data

  # change the range to $-2^{15} \sim 2^{15}$
- wave_data = wave_data.astype(np.int16)
- play_obj = sa.play_buffer(wave_data, n_channel, n_bytes, fs)
- play_obj.wait_done()

D. 製作音檔

- wavfile.write(file name, fs, data)

  # fs means the sampling frequency

  # data should be a one-column or two column array

Example:

- wavfile.write('Alarm01_test.wav', 22050, wave_data)

E. 錄音

要先import 相關模組： import pyaudio

範例程式

```
import pyaudio
pa=pyaudio.PyAudio()
fs = 44100
chunk = 1024
stream = pa.open(format=pyaudio.paInt16, channels=1,
rate=fs, input=True, frames_per_buffer=chunk)

vocal=[]
count=0
```

```
while count<200: #控制錄音時間
    audio = stream.read(chunk)   #一次性錄音取樣位元組大小
    vocal.append(audio)
    count +=1

save_wave_file('testrecord.wav',vocal)
stream.close()
```

參考
https://codertw.com/%E7%A8%8B%E5%BC%8F%E8%AA%9E%E8%A8%80/491427/

F. 影像檔的處理

可以先安裝幾個模組

　　pip install numpy

　　pip install matplotlib

1. 讀取影像檔

```
import cv2
image = cv2.imread('D:/Pic/peppers.bmp')
```
或
```
import matplotlib.pyplot as plt
image = plt.imread('D:/Pic/peppers.bmp')
```

注意

(1)寫入圖片若為彩色圖片，需要注意 cv2.imread 預設channels 順序為 BGR,

image[:, :, 0] => B, image[:, :, 1] => G, image[:, :, 2] => R

(2) 若使用 plt.imread, 則 3 個 channels 的順序仍為 RGB

image[:, :, 0] => R, image[:, :, 1] => G, image[:, :, 2] => B

(3) 若讀檔讀不出來，有時要將路徑的 \ 改為 /

(4) image.shape 可以看出影像之大小

>>> image.shape

(512, 512, 3)

(5) 可讀 jpg, bmp, png 檔，但不能讀 gif 檔

2. 顯示影像

Case 1: 圖的值格式為 int

以下的指令要配合使用 (彩色，灰階皆可)

    cv2.imshow('test', image) # 以 test 為顯示的圖的名稱

    cv2.waitKey(0)

    cv2.destroyAllWindows()

亦可用以下之指令

    import matplotlib.pyplot as plt

    plt.imshow(image)

    plt.show()

若一開始用 cv.imread 讀圖但要用 plt.imshow來顯示彩色圖，要先將 BGR 的順序轉回 RGB，將第二行改為

    plt.imshow(image[:,:,[2,1,0]])

Case 2: 圖的值格式為 double (非整數)

此時，無論用 cv2.imshow或是 plt.imshow，皆要先除以255，再將圖顯示出來

Example 1:

image = cv2.imread('D:/Pic/peppers.bmp')

Image1 = image*0.5 + 127.5 # lighten the image

cv2.imshow('test', image) # int 不用除255

cv2.waitKey(0)

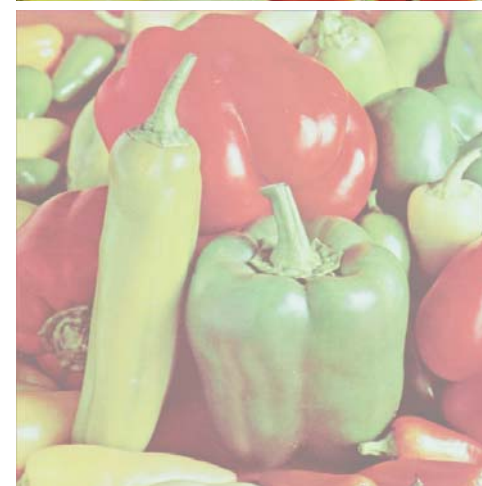cv2.destroyAllWindows()

cv2.imshow('test', image1/255) # 非整數要除255

cv2.waitKey(0)

cv2.destroyAllWindows()

Example 2:

import matplotlib.pyplot as plt

image = plt.imread('D:/Pic/peppers.bmp')

image1 = image*0.5 + 127.5 # lighten the image

plt.imshow(image) # int 不用除255

plt.show()

plt.imshow(image1/255) # 非整數要除255

plt.show()

3. 寫入圖片檔

cv2.imwrite('D:/Pic/jpg', image)

或

plt.imsave('D:/Pic/jpg', image)

注意

(1) 寫入圖片若為彩色圖片，在使用 cv2.imwrite 時需要注意

image[:, :, 0] => B, image[:, :, 1] => G, image[:, :, 2] => R

若用 plt.imsave 則

image[:, :, 0] => R, image[:, :, 1] => G, image[:, :, 2] => B

(2) 若用 cv2.imwrite('D:\Pic\jpg', image) 可能無法存檔，要將 \ 改為 /

(3) 若是使用 plt.imshow 和 plot.show() 來顯示，可以用右下角的 "save the figure" 來存檔