

Temari

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Professorat i evaluació

Professor: **Albert Oliveras** D5213 Dill: 09:30-12:00 Dij: 16:00-19:00

1r control:	7/10 Abril
2n control:	19/22 Maig
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Nota:	max{E.F, 0.6E.F.+0.2P1+0.2P2}

Introducció

Objectius

“Estudi de les tècniques i algoritmes de processat digital del senyal, enfocat a aplicacions en comunicacions i tractament d'imatge i veu”

Aplicacions

- Anàlisi espectral
- Caracterització paramètrica del senyal
- Filtrat adaptatiu
- Codificació i compressió de la imatge

Notació matricial

$x(n)$: seqüència

$$\underline{x} : \text{vector} = \begin{vmatrix} x(0) \\ \vdots \\ x(N-1) \end{vmatrix} \quad h^H \underline{x} = \sum_{n=0}^{N-1} h^*(n)x(n)$$

$$\underline{x}^T = \begin{vmatrix} x(0) & \cdots & x(N-1) \end{vmatrix} \quad \| \underline{x} \|^2 = \sum_{n=0}^{N-1} |x(n)|^2 = \underline{x}^H \underline{x}$$

$\underline{\underline{x}}$: matriu

desigualtat de Schwartz

$$|h^H \underline{x}|^2 \leq \|h\|^2 \|\underline{x}\|^2 \quad \text{sii } h(n) = \mathbf{a}x(n) \Rightarrow \text{igualtat}$$

Coherència

$$|\mathbf{g}|^2 = \frac{|h^H \underline{x}|^2}{\|h\|^2 \|\underline{x}\|^2} \leq 1 \quad \text{sii } h(n) = \mathbf{a}x(n) \Rightarrow \mathbf{g} = 1$$

Run-time vector

$$\underline{X}_n^T = [x(n), x(n-1), \dots, x(n-Q+1)]$$

Processat del senyal

Autocorrelació

Distància entre senyals

$$D_{xy} = \sum_{n=0}^{N-1} |x(n) - y(n)|^2 = (\underline{x} - \underline{y})^H (\underline{x} - \underline{y}) = \|\underline{x}\|^2 + \|\underline{y}\|^2 - \underline{x}^H \underline{y} - \underline{y}^H \underline{x}$$

Autocorrelació de x(n)

$$r(m) = \frac{1}{M} \sum_n x^*(n-m)x(n) = \frac{1}{M} \sum_n x^*(n)x(n+m)$$

Matriu de correlació a partir de \underline{X}_n

$$\underline{R} = \frac{1}{M} \sum_n \underline{X}_n \underline{X}_n^* \equiv \begin{bmatrix} r(0) & r(1) & \cdots & r(Q-1) \\ r(-1) & r(0) & \cdots & r(Q-2) \\ \vdots & \vdots & \ddots & \vdots \\ r(-Q+1) & r(-Q+2) & \cdots & r(0) \end{bmatrix}$$

Filtrat

$$y(n) = \sum_{q=0}^{Q-1} h^*(q)x(n-q) = h^H \cdot \underline{x}_n = \begin{vmatrix} h(0)^* & \cdots & h^*(Q-1) \end{vmatrix} \begin{vmatrix} x(0) \\ \vdots \\ x(Q-1) \end{vmatrix}$$

$$|y(n)|^2 = \underline{h}^H \underline{X}_n \underline{X}_n^H \underline{h}$$

$$r_y(0) = \frac{1}{M} \sum_n |y(n)|^2 = \underline{h}^H \underline{R} \underline{h}$$

$$r_y(0) \leq 0$$

Transitori

$x(n)$, N mostres $h(n)$, Q mostres (filtre FIR)

$$\underline{y}_T(n) = \left[\underbrace{y(0), y(1), \dots, y(Q-1)}_{\text{pre-transitori } Q \text{ mostres}}, \underbrace{y(Q), \dots, y(N-1)}_{\text{règim transitori } N-Q \text{ mostres}}, \underbrace{y(N), \dots, y(N+Q-1)}_{\text{post-transitori } Q \text{ mostres}} \right]$$

$$E_y = \underline{y}^H \underline{y} = \underline{y}^T \underline{y}^*$$

$$P_y = \frac{1}{M} \underline{y}^H \underline{y} = \underline{h}^H \frac{1}{M} \underline{X} \underline{X}^H \underline{h}$$

$$\underline{X} = \begin{bmatrix} x(0) & x(1) & \cdots & x(Q-1) & \cdots & x(N-1) & 0 & \cdots & 0 \\ 0 & x(0) & \cdots & x(Q-2) & \cdots & x(N-2) & x(N-1) & \cdots & 0 \\ \vdots & \vdots & \ddots & \vdots & \ddots & \vdots & \vdots & \ddots & 0 \\ 0 & 0 & \cdots & x(0) & \cdots & x(N-Q) & x(N-Q+1) & \cdots & x(N-1) \end{bmatrix}$$

$$\underline{y} = \underline{h}^H \underline{X}$$

Sinus en soroll i el lema de la inversa

$$s(n) = A_0 e^{j(2p f_0 n + q)} = A e^{j2p f_0 n}$$

$$\underline{S}_n = A e^{j2p f_0 n} \begin{bmatrix} 1 \\ e^{-j2p f_0} \\ \vdots \\ e^{-j2p f_0 (Q-1)} \end{bmatrix} = A e^{j2p f_0 n} \underline{S}_0$$

$$\underline{R}_0 = \frac{1}{M} \sum_n \underline{S}_n \underline{S}_n^H = |A|^2 \underline{S}_0 \underline{S}_0^H$$

$$y(n) = s(n) + w(n)$$

$$\underline{R} = |A|^2 \underline{S}_0 \underline{S}_0^H + \underline{s}^2 \underline{I}$$

Si existeixen $M < Q$ sinus ortogonals, existiran M termes iguals.

$$\underline{R} = \sum_{m=1}^M |A_m|^2 \underline{S}_m \underline{S}_m^H + \underline{s}^2 \underline{I} = \underline{R}_S + \underline{s}^2 \underline{I}$$

Lema de la inversa

$$\text{si } \underline{R} = \underline{G} \underline{G}^H + \underline{R}_a \Rightarrow$$

$$\underline{R}^{-1} = \underline{R}_a^{-1} - \underline{R}_a^{-1} \underline{G} (\underline{I} + \underline{G}^H \underline{R}_a^{-1} \underline{G})^{-1} \underline{G}^H \underline{R}_a^{-1}$$

Processat del senyal

Minimització amb restriccions

Disseny de filtres o sistemes pot formular-se sempre com a minimització d'una funció objectius, subjecta a restriccions. Per processos gaussians i sistemes lineals, aquesta funció és l'error quadràtic mig (MSE).

$$MSE = \underline{x} = \underline{a}^H \underline{C} \underline{a}$$

restriccions: $\underline{\Phi} \underline{a} = \underline{f}$

$$MSE_{min} = \underline{f}^H (\underline{\Phi} \underline{C}^{-1} \underline{\Phi}^H)^{-1} \underline{f} \text{ amb } \underline{a} = \underline{C}^{-1} \underline{\Phi}^H (\underline{\Phi} \underline{C}^{-1} \underline{\Phi}^H)^{-1} \underline{f}$$

Autovalors i autovectors

$$\underline{R} \underline{e}_q = \underline{I}_q \underline{e}_q$$

\underline{e}_q : autovector

\underline{I}_q : autovalor

Per ser \underline{R} hermítica ($\dim Q \times Q$) existeixen Q autovectors diferents que permeten fer la descomposició següent:

$$\underline{R} = \sum_{q=1}^Q \underline{I}_q \underline{e}_q \underline{e}_q^H = [\underline{e}_1 \dots \underline{e}_Q] \begin{bmatrix} \underline{I}_1 & \dots & 0 \\ \vdots & \ddots & \vdots \\ 0 & \dots & \underline{I}_q \end{bmatrix} \begin{bmatrix} \underline{e}_1^H \\ \vdots \\ \underline{e}_Q^H \end{bmatrix} = \underline{E} \underline{D} \underline{E}^H$$

$$\underline{I} = \underline{E}^H \underline{E} = \underline{E} \underline{E}^H$$

Propietats

- Si la matriu és definida positiva, tots els seus autovalors ho són.
- Si la matriu R és de rang $M < Q$, el número d'autovalors zero és $Q - M$
- El determinant de la matriu R és el producte dels seus autovalors, i la seva traça és la suma dels autovalors.
- $\ln\{\underline{R}\} = \underline{E} \ln\{\underline{D}\} \underline{E}^H$
- $e^{\{\underline{R}\}} = \underline{E} e^{\{\underline{D}\}} \underline{E}^H$
- $\underline{R}^m = \underline{E} \underline{D}^m \underline{E}^H$
- $\underline{R}^{-1} = \underline{E} \underline{D}^{-1} \underline{E}^H$

Valors singulars

$$\dim\{\underline{R}\} = Q \times Q \quad \underline{R} = \underline{X} \underline{X}^H$$

$$\dim\{\underline{X}\} = Q \times N \quad \underline{X} = \underline{Q} \underline{X} \underline{N}$$

$$\underline{X} = \sum_{q=1}^Q \underline{f}_q \underline{u}_q \underline{v}_q^H = [\underline{u}_1 \dots \underline{u}_Q] \begin{bmatrix} \underline{f}_1 & \dots & 0 \\ \vdots & \ddots & \vdots \\ 0 & \dots & \underline{f}_q \\ 0 & \dots & 0 \\ \vdots & \ddots & \vdots \\ 0 & \dots & 0 \end{bmatrix} \begin{bmatrix} \underline{v}_1^H \\ \vdots \\ \underline{v}_N^H \end{bmatrix} = \underline{U} \underline{\Phi} \underline{V}^H$$

$$\underline{I} = \underline{Q}^H \underline{Q} = \underline{V}^H \underline{V}$$

$$(\underline{X} \underline{X}^H) \underline{u}_q = \underline{f}_q^2 \underline{u}_q, q = 1..Q \quad (\underline{X}^H \underline{X}) \underline{v}_q = \begin{cases} \underline{f}_q^2 \underline{v}_q & q = 1..Q \\ 0 & q = Q+1..N \end{cases}$$

$$\underline{R} = \underline{U} \underline{\Phi}^2 \underline{U}^H = \underline{U} \underline{D} \underline{U}^H$$

Sistemas adaptativos

Introducción

Los diseños óptimos dan lugar a degradación de las prestaciones cuando las condiciones del escenario cambian.

De este modo, cuando un sistema de tratamiento de señal es capaz de adaptarse automáticamente al entorno de señal, se dice que es **adaptativo**.

Amplificador: $d(n) = g(n)x(n)$

En el sistema de apredizaje, se dispone de la señal de referencia $r(n)$, o simplemente de su potencia P_r , basta pues calcular la potencia de la señal de salida según:

$$P_d(n) = \mathbf{b}P_d(n-1) + (1-\mathbf{b})d^2(n)$$

$$\mathbf{e}(n) = P_r - P_d(n)$$

$$g(n+1) = g(n) + \mathbf{m}\mathbf{e}(n)$$

MSE y métodos del gradiente

$d(n)$: señal de referencia

\underline{a}_n : respuesta impulsional el filtro FIR

$$MSE = \mathbf{x}(\underline{a}_n) = E\left\{\left|d(n) - \underline{a}_n^H \underline{X}_n\right|^2\right\}$$

\underline{X}_n : vector de datos

$$\mathbf{x}(\underline{a}_n) = P_d + \underline{a}_n^H \underline{\underline{R}} \underline{a}_n - \underline{a}_n^H \underline{P} - \underline{P}^H \underline{a}_n$$

$$\frac{\partial \mathbf{x}(\underline{a}_n)}{\partial \underline{a}_n^H} = \nabla \mathbf{x} = \underline{\underline{R}} \underline{a}_{opt} - \underline{P} = 0 \Rightarrow \underline{a}_{opt} = \underline{\underline{R}}^{-1} \underline{P}$$

$$\mathbf{x}(\underline{a}_n) = \mathbf{x}_{min} + (\underline{a}_n - \underline{a}_{opt})^H \underline{\underline{R}} (\underline{a}_n - \underline{a}_{opt})$$

$$\mathbf{x}_{min} = P_d - \underline{P}^H \underline{\underline{R}} \underline{P}$$

$$\underline{a}_{n+1} = \underline{a}_n - \mathbf{m} \nabla \mathbf{x}(\underline{a}_n) \quad \mathbf{x}(\underline{a}_n) = \mathbf{x}_{min} + \underline{a}_n^H \underline{\underline{R}} \underline{a}_n$$

$$\nabla \mathbf{x}(\underline{a}_n) = \underline{\underline{R}} \underline{a}_n$$

$$\mathbf{x} = \mathbf{x}_{min} + \underline{\tilde{a}}_n^H (\underline{\underline{E}} \underline{\underline{\Delta}} \underline{\underline{E}}^H) \underline{\tilde{a}}_n = \mathbf{x}_{min} + \underline{z}_n^H \underline{\underline{\Delta}} \underline{\underline{z}}_n^H = \mathbf{x}_{min} + \sum_{i=1}^Q I_i |z_n(i)|^2$$

$$\underline{z}_n = \underline{\underline{E}}^H (\underline{\tilde{a}}_n - \underline{a}_{opt})$$

Diseño y convergencia de algoritmos del gradiente

$$\underline{a}_{n+1} = \underline{a}_n - \mathbf{m} \nabla \mathbf{x}_n = \underline{a}_n - \mathbf{m} (\underline{\underline{R}} \underline{a}_n - \underline{P})$$

$$\mathbf{m} < \frac{2}{Traza\{\underline{\underline{R}}\}} < \frac{2}{I_{max}} \quad P_x(n+1) = \mathbf{b}P_x(n) + (1-\mathbf{b})\underline{X}_n^H \underline{X}_n$$

$$\mathbf{m} = \frac{2\mathbf{a}}{P_x(n)} \quad \frac{1}{1-\mathbf{b}} : \text{número de muestras de la memoria}$$

Usualmente se establece un umbral mínimo P_{xo} .

$$N_c = \frac{2,30}{-\ln(1 - \mathbf{m} I_{min})} = \frac{2,30}{-\ln(1 - 2\mathbf{a} \frac{I_{min}}{I_{max}})} \approx \frac{2,30}{2\mathbf{a}} \frac{I_{max}}{I_{min}}$$

$$TN_c = \frac{2,30}{2\mathbf{a}} Q \frac{P_s}{P_w} = \frac{2,30}{2\mathbf{a}} QSNR$$

El algoritmo LMS (Least Mean Square algorithm)

$$y(n) = \underline{a}_n^H \underline{X}_n$$

$$\mathbf{e}(n) = d^*(n) - y^*(n) \quad \mathbf{m} < \frac{2}{\underline{X}_n^H \underline{X}_n}$$

$$\underline{a}_{n+1} = \underline{a}_n + \mathbf{m} \underline{X}_n \mathbf{e}^*(n)$$

$$E\{\underline{a}_n\} = \underline{a}_{opt}$$

$\underline{\tilde{a}}_n = \underline{a}_n - \underline{a}_{opt}$: error de coeficientes

$$\underline{a}_{n+1} = \underline{\tilde{a}}_n - \mathbf{m} \mathbf{e}^*(n) \underline{X}_n$$

$$d(n) = w(n) + \underline{a}_{opt}^H \underline{X}_n$$

$$\mathbf{e}(n) = w(n) - \underline{\tilde{a}}_{opt}^H \underline{X}_n$$

Processat del senyal

$$\underline{\Sigma}_a = E\{\tilde{a}_{opt} \tilde{a}_{opt}^H\} \equiv \frac{\mathbf{m}}{2} \mathbf{x}_{min} \underline{I} : \text{covarianza de los coeficientes}$$

$$\mathbf{x}(n) = \mathbf{x}_{min} + \tilde{a}_n^H \underline{R} \tilde{a}_n$$

$$E\{\tilde{a}_n^H \underline{R} \tilde{a}_n\} = \frac{\mathbf{m}}{2} \mathbf{x}_{min} \text{Trazza}\{\underline{R}\}$$

$$\aleph \equiv \frac{E\{\mathbf{x}(n)\} - \mathbf{x}_{min}}{\mathbf{x}_{min}} 100\% : \text{exceso de error}$$

$$\aleph = \frac{\mathbf{m}}{2} \text{Trazza}\{\underline{R}\}$$

$$\mathbf{m} = \frac{2\mathbf{a}}{P_x(n)}$$

\mathbf{a} : ruido de desajuste (i.e. $\mathbf{a} = 0.1 \Rightarrow \aleph = 10\%$)

Conclusiones:

- Probar convergencia al óptimo
- Analizar la velocidad de convergencia
- En la fase de seguimiento, se ha de determinar el ruido de desajuste o lo que es lo mismo, su velocidad de seguimiento o capacidad de reacción a cambios.
- A altas velocidades de convergencia (o aprendizaje rápido) implican desajustes elevados.
- Hay un compromiso entre desajuste y velocidad de convergencia.

DSD y métodos de búsqueda aleatoria

DSD (Differential Steepest Descent), calcula el gradiente independientemente para cada coeficiente del filtro.

$$\nabla_n(q) = \frac{\mathbf{x}_n(q, \mathbf{d}) + \mathbf{x}_n(q - \mathbf{d})}{2\mathbf{d}}$$

$$\underline{a}_{n+1} = \underline{a}_n - \frac{\mathbf{m}}{2} \nabla_n$$

$$\mathbf{x}_n(q, -\mathbf{d}) = \mathbf{x}_n - \mathbf{x}_n^T \mathbf{d} + \mathbf{x}_n^T \frac{\mathbf{d}^2}{2}$$

$$\mathbf{x}_n(q, +\mathbf{d}) = \mathbf{x}_n + \mathbf{x}_n^T \mathbf{d} + \mathbf{x}_n^T \frac{\mathbf{d}^2}{2}$$

$$\mathbf{g} = \frac{\mathbf{x}_n^T \mathbf{d}^2}{\mathbf{x}_{min}} = \frac{\mathbf{d}^T \mathbf{r}_{xx}(0)}{\mathbf{x}_{min}}$$

$$\mathbf{g} = \frac{\mathbf{d}^T \text{Trazza}\{\underline{R}\}}{Q \mathbf{x}_{min}}$$

$$\mathbf{x}_n(q, \mathbf{d}) = \frac{1}{M} \sum_n^{n+M-1} |e(n)|^2$$

Ventajas

- Desajuste disminuye con la raíz cuadrada del número de términos empleados en la estimación del MSE.
- No es necesario el vector de datos

Desventaja:

- Se requieren $2MQ$ vectores por cada iteración.

$$\aleph_{total} = \frac{3}{4} \frac{m Q \mathbf{x}_{min}}{M \mathbf{d}^2} + \frac{\mathbf{d}^T \text{Trazza}\{\underline{R}\}}{Q \mathbf{x}_{min}}$$

$$\mathbf{m} = \frac{2\mathbf{a}}{\text{Trazza}\{\underline{R}\}}$$

$$\mathbf{d}_{opt}^2 = \sqrt{\frac{3}{2} \frac{Q^2 \mathbf{a} \mathbf{x}_{min}^2}{M \text{Trazza}^2\{\underline{R}\}}}$$

$$\aleph_{total \text{ optimo}} = \sqrt{\frac{6\mathbf{a}}{M}}$$

Algoritmo LRS (Linear Random Search)

Para cada instancia n ,

- Generar un vector aleatorio con distribución uniforme en sus componentes \underline{d}_n .
- Actualizar los pesos como: $\underline{a}_{n+1} = \underline{a}_n + \frac{\mathbf{m}}{2} \underline{d}_n$
- Calcular el nuevo error usando M vectores de datos \mathbf{x}_{n+1}
- $\underline{a}_{n+1} = \begin{cases} \underline{a}_n & \mathbf{x}_{n+1} > \mathbf{x}_n \\ \underline{a}_{n+1} & \text{otherwise} \end{cases}$
- $n++$;

$$\aleph = \frac{m Q s_d^2}{2M}$$

Algoritmo GRS (Guided Random Search)

Igual que LRS pero se diferencia en que cuando se encuentra una perturbación que hace disminuir el error, se avanza en esa dirección incrementando la \underline{m} hasta que el error empeora y entonces $\underline{m} = \underline{m}_{\text{básico}}$ y se busca un vector de perturbaciones aleatorio ortogonal al presente.

Algoritmo RLS (Recursive Least Squares)

$$MSE(n) = \underline{b}MSE(n-1) + (1 - \underline{b}) |\underline{e}(n)|^2$$

$$\underline{\underline{R}}_{n+1} = \underline{b}\underline{\underline{R}}_n + (1 - \underline{b})\underline{X}_n\underline{X}_n^H$$

$$\underline{P}_{n+1} = \underline{b}\underline{P} + (1 - \underline{b})\underline{X}_n d^*(n)$$

$$\underline{a}_{n+1} = \underline{\underline{R}}_{n+1}^{-1} \underline{P}_{n+1}$$

$$M = \frac{1}{1 - \underline{b}} : \text{número de datos de la señal de entrada}$$

instante n: datos $\underline{a}_n, d(n), \underline{X}_n, \underline{\underline{R}}_n^{-1}$

- $y(n) = \underline{a}_n^H \underline{X}_n$
- $\underline{e}(n) = d(n) - y(n)$
- $\underline{f} = \underline{a} \left(\underline{X}_n^H \underline{\underline{R}}_n^{-1} \underline{X}_n \right) \underline{a} = \frac{1 - \underline{b}}{\underline{b}}$
- $\underline{K}_n = \frac{\underline{a} \underline{\underline{R}}_n^{-1} \underline{X}_n}{1 + \underline{f}} : \text{vectr ganancia}$
- $\underline{a}_{n+1} = \underline{a}_n + \underline{K}_n \underline{e}^*(n)$
- $\underline{\underline{R}}_n^{-1} = \frac{1}{\underline{b}} \underline{\underline{R}}_n^{-1} - \underline{K}_n \underline{K}_n^H \frac{1 + \underline{f}}{1 - \underline{b}}$
- $n++;$

Introduction

Discrete-Time Sinusoidal

$$x(n) = a \cos(\omega n + \phi), -\infty < n < +\infty$$

$$\omega = 2\pi f_0$$

$$x(n) \text{ periodic} \Leftrightarrow x(n) = x(N+n) \Leftrightarrow f \in Q$$

$$2\pi f_0 n = 2\pi k p$$

$$f_0 = \frac{k}{N}$$

$$\cos(\omega n + \phi) = \cos((2\pi p k + \omega)n + \phi) = \cos(\omega_k n + \phi), \omega_k = \omega + 2\pi p k$$

$$\text{maximum oscillation} \Rightarrow \omega = \pm p, f = \pm \frac{1}{2} \Rightarrow x(n) = \{+1, -1, +1, -1, +1, \dots\}$$

Sampling of analog signals

$$x(n) = x_a(nT) \quad f = \frac{F}{F_m} : \text{normalized frequency}$$

$$t = nT = \frac{n}{F_m} \quad \omega = \Omega T$$

Continuous-time signals

$$\Omega_{\text{radians/sec}} = 2\pi F_{\text{Hz}}$$

$$-\infty < \Omega < +\infty$$

$$-\infty < F < +\infty$$

Discrete-time signals

$$\omega_{\text{radians/sample}} = 2\pi f_{\text{cycles/sample}}$$

$$-p \leq \omega \leq +p$$

$$-\frac{1}{2} \leq f \leq +\frac{1}{2}$$

Sampling Theorem

If the highest frequency contained in an analog signal $x_a(t)$ is $F_{max} = B$ and the signal is sampled at a rate $F_s > 2F_{max} \equiv 2B$, then $x_a(t)$ can be exactly recovered from its sample values.

Quantization

$$x_q(n) = Q\{x(n)\} \quad |e_q(n)| \leq \frac{\Delta}{2}$$

$$e_q(n) = x_q(n) - x(n) \quad \Delta = \frac{x_{max} - x_{min}}{L-1}$$

Quantization of sinusoidal signals

$$\left. \begin{aligned} P_q &= \frac{\Delta^2}{12} \\ \Delta &= \frac{2A}{2^b} \end{aligned} \right\} P_q = \frac{A^2 / 3}{2^{2b}} \Rightarrow SNQR = \frac{3}{2} 2^{2b}, SNQR_{db} = 1.76 + 6.02b$$

Discrete-Time Signals and Systems

Elementary signals

$$d(n) \equiv \begin{cases} 1 & n = 0 \\ 0 & n \neq 0 \end{cases} \quad \text{unit sample sequence, unit impulse}$$

$$u(n) \equiv \begin{cases} 1 & n \geq 0 \\ 0 & n < 0 \end{cases} \quad \text{unit step signal}$$

$$u_r(n) \equiv \begin{cases} n & n \geq 0 \\ 0 & n < 0 \end{cases} \quad \text{unit ramp signal}$$

$$x(n) = a^n = (re^{j\phi})^n = r^n (\cos \phi n + j \sin \phi n) \quad \text{exponential signal}$$

Classification of signals

- $E \equiv \sum_{n=-\infty}^{+\infty} |x(n)|^2 \quad 0 < E < \infty \Rightarrow x(n)$ is an energy signal
- $P \equiv \lim_{N \rightarrow +\infty} \frac{1}{2N+1} \sum_{n=-N}^{+N} |x(n)|^2 \quad 0 < P < \infty \Rightarrow x(n)$ is a power signal
- periodic signal \Rightarrow power signal
- $x(n) = x(-n) \Rightarrow$ even signal $x_e(n) = \frac{1}{2}(x(n) + x(-n))$
- $x(n) = -x(-n) \Rightarrow$ odd signal $x_o(n) = \frac{1}{2}(x(n) - x(-n))$

Discrete-time systems $x(n) \xrightarrow{f} y(n)$

- static: $y(n) = f\{x(n)\}$ memoryless
- dynamic: $y(n) = f\{x(n), x(n-1), \dots\}$ with memory

Processat del senyal

- time variant: $x(n) \xrightarrow{\Gamma} y(n)$
 $x(n-k) \xrightarrow{\Gamma} y(n-k)$
- linear: $\Gamma\{a_1x_1(n) + a_2x_2(n)\} = a_1\Gamma\{x_1(n)\} + a_2\Gamma\{x_2(n)\}$
- causal: output only depends on past and present.
- Stable: every bounded input produces a bounded output.

Analysys of Discrete-time systems Linear Time-Invariant

$$y(n) = -\sum_{k=1}^N a_k y(n-k) + \sum_{k=0}^M b_k x(n-k)$$

$$y(n) = x(n) * h(n) = \sum_{k=-\infty}^{+\infty} h(k)x(n-k) = \sum_{k=-\infty}^{+\infty} x(k)h(n-k)$$

- Commutative law: $x(n) * h(n) = h(n) * x(n)$
- Associative law: $[x(n) * h_1(n)] * h_2(n) = x(n) * [h_1(n) * h_2(n)]$
- Distributive law: $x(n) * [h_1(n) + h_2(n)] = x(n) * h_1(n) + x(n) * h_2(n)$
- Causal: $h(n) = 0 \forall n < 0$
- Stable: $\sum_{k=-\infty}^{+\infty} |h(n)| < \infty$
- FIR: $h(n) = 0 \forall n < 0 \text{ and } n \geq M$

Correlation of signals

$$y_{received}(n) = \mathbf{a}x(n-D) + w(n)$$

$$r_{xy}(m) = \sum_{n=-\infty}^{+\infty} x(n)y(n-m) = \sum_{n=-\infty}^{+\infty} x(n+m)y(n) = r_{yx}(-m)$$

$$\sum_{n=-\infty}^{+\infty} |\mathbf{a}x(n) + \mathbf{b}y(n-m)|^2 = a^2 r_{xx}(0) + b^2 r_{yy}(0) + 2ab r_{xy}(l)$$

$$|r_{xy}(m)| \leq \sqrt{E_x E_y} \quad |r_{xx}(m)| \leq E_x$$

$$r_{xx}(m) = \frac{r_{xx}(m)}{r_{xx}(0)} : \text{normalized autocorrelation}$$

$$r_{xy}(m) = \frac{r_{xy}(m)}{\sqrt{r_{xx}(0)r_{yy}(0)}} : \text{normalized crosscorrelation}$$

Correlation of periodic signals

$$r_{xy}(m) = \lim_{M \rightarrow \infty} \frac{1}{2M+1} \sum_{n=-M}^M x(n)y(n-m) = \frac{1}{N} \sum_n x(n)y(n-m)$$

$$y(n) = x(n) + w(n) \Rightarrow r_{yy}(m) = r_{xx}(m) + r_{xw}(m) + r_{wx}(m) + r_{ww}(m)$$

Convolution realisation of correlation

$$r_{xy}(m) = x(m) * y(-m)$$

$$r_{xx}(m) = x(m) * x(-m)$$

Input-Output correlation sequences

$$y(n) = x(n) * y(n)$$

$$r_{yx}(m) = r_{xx}(m) * h(l)$$

$$r_{xy}(m) = r_{xx}(m) * h(-l)$$

$$r_{yy}(m) = r_{xx}(m) * r_{hh}(m) = r_{xx}(m) * h(m) * h(-m)$$

Frequency Analysis of signals and systems

Continous-Time signals

Jean Baptiste Joseph Fourier (1768-1830)

Periodic-Signals

	$x(t) = \sum_{k=-\infty}^{+\infty} c_k e^{j2\pi k F_0 t}$
Synthesis equations	$x(t) = c_0 + 2 \sum_{k=1}^{+\infty} c_k \cos(2\pi k F_0 t + \angle c_k)$
	$x(t) = a_0 + \sum_{k=1}^{+\infty} (a_k \cos(2\pi k F_0 t) + b_k \sin(2\pi k F_0 t))$
	$c_k = \frac{1}{T_p} \int_{-T_p/2}^{T_p/2} x(t) e^{-j2\pi k F_0 t} dt$
Analysis equations	$a_0 = c_0$ $a_k = 2 c_k \cos \angle c_k$ $b_k = 2 c_k \sin \angle c_k$

Power Density Spectrum of Periodic Signals

$$P_x = \sum_{k=-\infty}^{+\infty} |c_k|^2 = a_0^2 + \frac{1}{2} \sum_{k=1}^{+\infty} (a_k^2 + b_k^2) \quad G_{xx}(f) = \sum_{k=-\infty}^{+\infty} |c_k|^2 \delta(f - F_0 k)$$

Aperiodic-signals

	$x(t) = \int_{-\infty}^{+\infty} X(f) e^{j2\pi f t} df$
Analysis equations	$X(t) = \int_{-\infty}^{+\infty} x(t) e^{-j2\pi f t} dt$

Power Density Spectrum of aperiodic Signals

$$S_{xx}(f) = |X(f)|^2 \quad E_{xx} = \int_{-\infty}^{+\infty} S_{xx}(f) df = \int_{-\infty}^{+\infty} |x(t)|^2 dt$$

Relationship between periodic and aperiodic transform

$$c_k = \frac{1}{T_p} X(kF_0) = \frac{1}{T_p} X\left(\frac{k}{T_p}\right)$$

Discrete-Time signals

Periodic-Signals

Synthesis equations	$x(n) = \sum_{k=0}^{N-1} c_k e^{j\frac{2\pi}{N} kn}$
Analysis equations	$c_k = \frac{1}{N} \sum_{n=0}^{N-1} x(n) e^{-j\frac{2\pi}{N} kn} \quad c_k = c_{k+N}$
	$P_x = \frac{1}{N} \sum_{n=0}^{N-1} x(n) ^2 = \sum_{n=0}^{N-1} c_k ^2 \quad E_x = \sum_{n=0}^{N-1} x(n) ^2 = N \sum_{n=0}^{N-1} c_k ^2$

Aperiodic-Signals

Synthesis equations	$x(n) = \frac{1}{2\pi} \int_{-\pi}^{\pi} X(\omega) e^{j\omega n} d\omega$
Analysis equations	$X(\omega) = \sum_{n=-\infty}^{+\infty} x(n) e^{-j\omega n}$
	$E_x = \sum_{k=-\infty}^{+\infty} x(n) ^2 = \frac{1}{2\pi} \int_{-\pi}^{\pi} S_{xx}(\omega) d\omega \quad S_{xx}(\omega) = X(\omega) ^2 \Rightarrow \begin{cases} S_{xx}(\omega) = S_{xx}(-\omega) \\ \angle X(\omega) = -\angle X(-\omega) \end{cases}$

Relationship with Z-transform:

$$X(\omega) = Z\{x(n)\} \mid z = e^{j\omega}$$

Basic rectangular pulse transform:

$$F\{u(n) - u(n-L)\} = A e^{-j(\omega/2)(L-1)} \frac{\sin(\omega L/2)}{\sin(\omega/2)}$$

Properties of the Fourier Transform for Discrete-Time signals

Processat del senyal

Jesús Sanz Marcos Introducció

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$$a_1 x_1(n) + a_2 x_2(n) \xrightarrow{F} a_1 X_1(\mathbf{w}) + a_2 X_2(\mathbf{w})$$

$$x(n-k) \xrightarrow{F} e^{-j\mathbf{w}k} X(\mathbf{w})$$

$$x(-n) \xrightarrow{F} X(-\mathbf{w})$$

$$x^*(n) \xrightarrow{F} X^*(\mathbf{w}^*)$$

$$nx(n) \xrightarrow{F} j \frac{dX(\mathbf{w})}{d\mathbf{w}}$$

$$x_1(n) * x_2(n) \xrightarrow{F} X_1(\mathbf{w}) X_2(\mathbf{w})$$

$$x_1(n) x_2(n) \xrightarrow{F} \frac{1}{2\mathbf{p}} \int_{-\mathbf{p}}^{+\mathbf{p}} X_1(\mathbf{I}) X_2(\mathbf{w} - \mathbf{I}) d\mathbf{I}$$

$$r_{x_1 x_2}(m) = x_1(m) * x_2(-m) \xrightarrow{F} X_1(\mathbf{w}) X_2(-\mathbf{w})$$

$$r_{xx}(m) \xrightarrow{F} S_{xx}(\mathbf{w})$$

$$e^{j\mathbf{w}_0 n} x(n) \xrightarrow{F} X(\mathbf{w} - \mathbf{w}_0)$$

$$\cos(\mathbf{w}_0 n) x(n) \xrightarrow{F} \frac{1}{2} (X(\mathbf{w} - \mathbf{w}_0) + X(\mathbf{w} + \mathbf{w}_0))$$

$$\sin(\mathbf{w}_0 n) x(n) \xrightarrow{F} \frac{1}{2j} (X(\mathbf{w} - \mathbf{w}_0) - X(\mathbf{w} + \mathbf{w}_0))$$

Parseval's relation: $\sum_{n=-\infty}^{+\infty} x_1(n) x_2^*(n) = \frac{1}{2\mathbf{p}} \int_{-\mathbf{p}}^{+\mathbf{p}} X_1(\mathbf{w}) X_2^*(\mathbf{w}) d\mathbf{w}$

Some useful Fourier Transform Pairs

$$\mathbf{d}(n) \xrightarrow{F} 1$$

$$x(n) = \begin{cases} 1 & |n| \leq L \\ 0 & |n| > L \end{cases} \xrightarrow{F} \frac{\sin\left(L + \frac{1}{2}\right)\mathbf{w}}{\sin\frac{\mathbf{w}}{2}}$$

$$x(n) = \begin{cases} \frac{\mathbf{w}_c}{\mathbf{p}} & n=0 \\ \frac{\sin\mathbf{w}_c n}{pn} & n \neq 0 \end{cases} \xrightarrow{F} \prod \left(\frac{\mathbf{w}}{2\mathbf{w}_c} \right)$$

Frequency-Domain LTI Systems

$$H(\mathbf{w}) = H(z) \Big|_{z=e^{j\mathbf{w}}} = \sum_{k=-\infty}^{+\infty} h(n) e^{-j\mathbf{w}n}$$

$$Y(\mathbf{w}) = X(\mathbf{w}) H(\mathbf{w})$$

$$|Y(\mathbf{w})| = |X(\mathbf{w})| |H(\mathbf{w})|$$

$$\angle Y(\mathbf{w}) = \angle X(\mathbf{w}) + \angle H(\mathbf{w})$$

$$S_{yy}(\mathbf{w}) = S_{xx}(\mathbf{w}) |H(\mathbf{w})|^2$$

$$|H(\mathbf{w})|^2 = H(\mathbf{w}) \overline{H}(\mathbf{w})$$

$$r_{yy}(m) = r_{xx}(m) * r_{hh}(m) \quad S_{yy}(\mathbf{w}) = S_{xx}(\mathbf{w}) |H(\mathbf{w})|^2$$

$$r_{yx}(m) = r_{xx}(m) * h(m) \quad S_{yx}(\mathbf{w}) = S_{xx}(\mathbf{w}) H(\mathbf{w})$$

Random Input signals

$$y(n) = x(n) * h(n)$$

$$m_y \equiv E\{y(n)\} = m_x \sum_{k=-\infty}^{+\infty} h(k) = m_x H(0)$$

$$\mathbf{g}_{yy}(m) = E\{y^*(n) y(n+m)\} = \sum_{k=-\infty}^{+\infty} \sum_{j=-\infty}^{+\infty} h(k) h(j) \mathbf{g}_{xx}(k-j+m)$$

$$\mathbf{g}_{xx}(m) = \mathbf{s}_x^2 \mathbf{d}(m) \Rightarrow \mathbf{g}_{yy}(m) = \mathbf{s}_x^2 \sum_{k=-\infty}^{+\infty} h(k) h(k+m)$$

(AWGN)

$$\mathbf{g}_{yy}(0) = P_y = \mathbf{s}_x^2 \sum_{k=-\infty}^{+\infty} |h(k)|^2 = \mathbf{s}_x^2 \int_{-1/2}^{1/2} |H(f)|^2 df$$

$$\Gamma_{yy}(\mathbf{w}) = \sum_{m=-\infty}^{+\infty} \mathbf{g}_{yy}(m) e^{-j\mathbf{w}m} = \Gamma_{xx}(\mathbf{w}) |H(\mathbf{w})|^2$$

$$\Gamma_{yx}(\mathbf{w}) = \Gamma_{xx}(\mathbf{w}) H(\mathbf{w})$$

Frequency-Selective Filters

Processat del senyal

$$h_{lp}(n) = \frac{\sin \mathbf{w}_c \mathbf{p} n}{\mathbf{p} n} \quad H_{hp}(\mathbf{w}) = H_{lp}(\mathbf{w} - \mathbf{p}) \quad h_{hp}(\mathbf{w}) = (-1)^n h_{lp}(\mathbf{w}) = (e^{j\mathbf{p}})^n h_{lp}(\mathbf{w})$$

Digital resonators: $p_{1,2} = re^{\pm j\mathbf{w}_0}$, r near ≤ 1

Notch filters: $z_{1,2} = e^{\pm j\mathbf{w}_o}$

Comb filters: nulls occur periodically across the frequency band.

All-Pass filters: $|H(\mathbf{w})| = 1 \forall \mathbf{w}$

$$y(n) = x(n) * h(n) \stackrel{\text{casual}}{=} \sum_{k=0}^n h(k)x(n-k)$$

$$h(0) = \frac{y(0)}{x(0)}$$

$$h(n) = \frac{y(n) - \sum_{k=0}^{n-1} h(k)x(n-k)}{x(0)}, n > 0$$

$$S_{yx}(\mathbf{w}) = H(\mathbf{w})S_{xx}(\mathbf{w})$$

$$H(\mathbf{w}) = \frac{S_{yy}(\mathbf{w})}{S_{xx}(\mathbf{w})} = \frac{S_{yx}(\mathbf{w})}{|X(\mathbf{w})|^2}$$

Inverse Systems

$$y(n) = x(n) * h(n) \quad h_l(n) = 0 \forall n < 0 \Rightarrow h_l(0) = \frac{1}{h(0)}$$

$$\mathbf{d}(n) = h(n) * h_{INV}(n) \quad h_{INV}(n) = -\sum_{k=1}^n \frac{h(k)h_l(n-k)}{h(0)} \quad n \geq 1$$

- $|z_k| < 1 \forall k \Rightarrow \angle H(\mathbf{p}) = \angle H(0)$ minimum phase system
- $|z_k| > 1 \forall k \Rightarrow \angle H(\mathbf{p}) - \angle H(0) = M\mathbf{p}$ maximum phase system
- otherwise: mixed-phase system

$H(z)$ minimum phase $\Rightarrow H^{-1}(z)$ stable

$H(z)$ stable $\Rightarrow H^{-1}(z)$ minimum phase

$H(z)$ non minimum phase $= H_{min}(z)H_{all-pass}(z)$

$$H(z) = \frac{A(z)}{B(z)} \Rightarrow \begin{cases} H_{min}(z) = \frac{B_1(z)B_2(z^{-1})}{A(z)}, B(z) = B_1(z)B_2(z) \\ H_{all-pass}(z) = \frac{B_2(z)}{B_2(z^{-1})} \end{cases}$$

Deconvolution

(system identification)

Processat del senyal

Z-Transform

$$X(z) = \sum_{n=-\infty}^{+\infty} x(n)z^{-n} \quad \text{Region of Convergence (ROC)}$$

Finite-Duration Signals

Causal: $z \in C - \{0\}$

Anticausal: $z \in C - \{-\infty\}$

Two-sided: $z \in C - \{0, \infty\}$

Infinite-Duration Signals

Causal: $|z| > r_2$

Anticausal: $|z| < r_1$

Two-sided: $r_2 < |z| < r_1$

Properties

$$a_1 x_1(n) + a_2 x_2(n) \xleftarrow{Z} a_1 X_1(z) + a_2 X_2(z)$$

$$x(n-k) \xleftarrow{Z} z^{-k} X(z)$$

$$a^n x(n) \xleftarrow{Z} X(a^{-1}z)$$

$$x(-n) \xleftarrow{Z} X(z^{-1})$$

$$x^*(n) \xleftarrow{Z} X^*(z^*)$$

$$\operatorname{Re}\{x(n)\} \xleftarrow{Z} \frac{1}{2}(X(z) + X^*(z^*))$$

$$\operatorname{Im}\{x(n)\} \xleftarrow{Z} \frac{1}{2}(X(z) - X^*(z^*))$$

$$nx(n) \xleftarrow{Z} -z \frac{dX(z)}{dz}$$

$$\text{Parseval's relation: } \sum_{n=-\infty}^{+\infty} x_1(n)x_2^*(n) = \frac{1}{2\pi j} \oint_C X_1(v)X_2^*(\frac{z}{v})v^{-1}dv$$

Basic Z-Transform Pairs

$$d(n) \xleftarrow{Z} 1$$

$$u(n) \xleftarrow{Z} \frac{1}{1-z^{-1}}$$

$$a^n u(n) \xleftarrow{Z} \frac{1}{1-az^{-1}}$$

$$na^n u(n) = \frac{az^{-1}}{(1-az^{-1})^2}$$

Rational Z-Transforms

$$X(z) = \frac{N(z)}{D(z)} = \frac{\sum_{k=0}^M b_k z^{-k}}{\sum_{k=0}^N a_k z^{-k}} = \frac{b_0}{a_0} z^{-M+N} \frac{(z-z_1)(z-z_2)\dots(z-z_M)}{(z-p_1)(z-p_2)\dots(z-p_N)} = Gz^{N-M} \frac{\sum_{k=0}^M (z-z_k)}{\sum_{k=0}^N (z-p_k)}$$

$$y(n) = x(n) * h(n) \xleftarrow{Z} Y(z) = X(z)H(z)$$

$$H(z) = \frac{Y(z)}{X(z)} = \sum_{n=-\infty}^{+\infty} h(n)z^{-n}$$

$$y(n) = -\sum_{k=1}^N a_k y(n-k) + \sum_{k=0}^M b_k x(n-k) \Rightarrow H(z) = \frac{\sum_{k=0}^M b_k z^{-k}}{1 + \sum_{k=1}^N a_k z^{-k}}$$

Analysis of LTI systems in the Z-domain

Any $H(z)$

causality: $\Leftrightarrow ROC | z | > z_1$

stability: $\Leftrightarrow ROC \in |z| = 1$

Rational $H(z)$

causality $\Leftrightarrow p_k$ most far away pole $\begin{cases} ROC | z | > |p_k| \\ \text{order}\{\text{num}\} \leq \text{order}\{\text{denom}\} \end{cases}$

stability: $\Leftrightarrow |p_k| < 1 \forall k$

Unilateral Z-Transform

$$X(z) = \sum_{n=0}^{+\infty} x(n)z^{-n}$$