# Disciplina: Processamento Digital de Sinais

Material \_aula 3
Ambiente Blackboard

# Apresentação

- 1) Implementação de algoritmos: Média Móvel – Matlab/Python, Linguagem C
- 2) Implementação do eco Matlab/Python
- 3) Transformação de variáveis
- 4) Convolução
- 5) Tarefas

#### Sistema Discreto

#### 2.3 Discrete-time systems

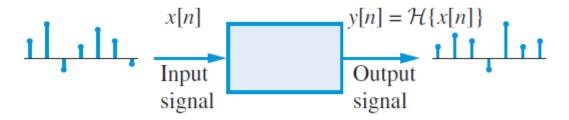


Figure 2.5 Block diagram representation of a discrete-time system.

# Implementação de algoritmos

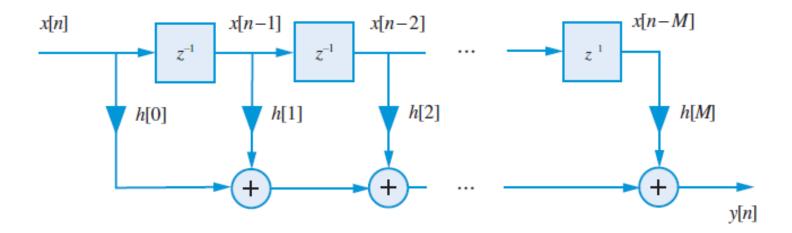
- Média Móvel de k elementos da entrada
- Por exemplo, para k = 4, temos:

$$y[n] = \frac{x[n] + x[n-1] + x[n-2] + x[n-3]}{4}$$

$$y[n] = 1/4 \cdot x[n] + 1/4 \cdot x[n-1] + 1/4 \cdot x[n-2] + 1/4 \cdot x[n-3]$$

# Implementação de algoritmos

Média Móvel de k elementos da entrada



# Implementação Média Móvel

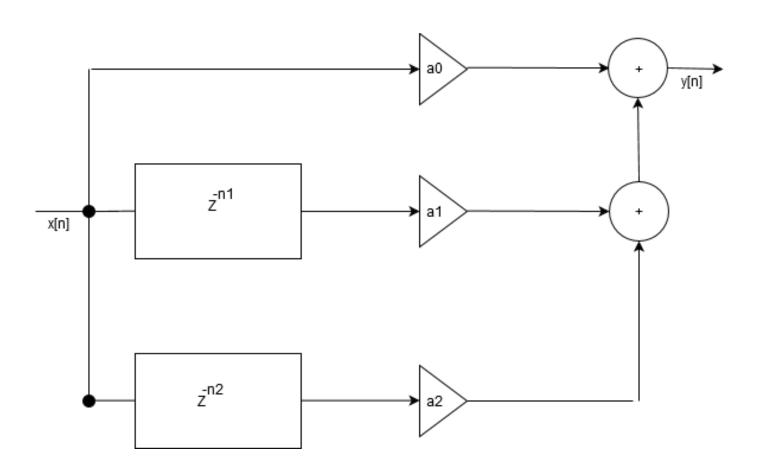
Vetor Arquivo Vetor x[n] Arquivo Coefs de dimensão de Dimensão entrada k.1 saída k,1 2 0 10 8 -2 3 -4 1/k 10 2,5 0 0 10 8 -4 -2 2 1/k 3 2 8 0 0 0 10 -4 8 1/k 5 0 0 0 0 10 2 8 -4 1/k 4 -2 3

### Implementação de algoritmos:

- Média Móvel Matlab / Python
- Média Móvel Linguagem C
- Usar como entrada um sinal de ruído branco gerado no ocenaudio
- Testar com diferentes tamanhos de média e salvar o arquivo de saída, por exemplo, k = 4, k = 8, k = 16, k = 32, etc

## Implementação de algoritmos:

- Média Móvel Matlab / Python
- Usar como entrada o impulso unitário e obtenha a correspondente saída.
- Usar como entrada o degrau unitário e obtenha a correspondente saída.



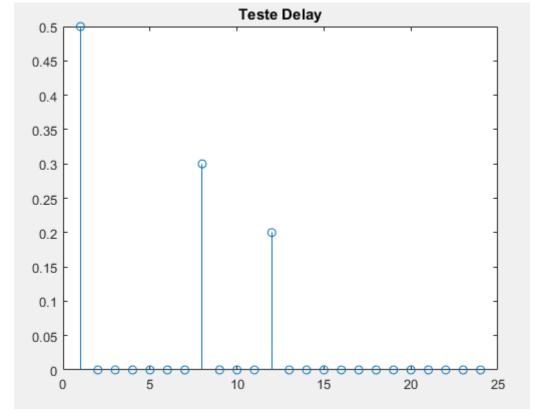
 Implemente um programa em Matlab/Python que execute a equação do diagrama de blocos do eco.

$$y[n] = a0.x[n] + a1.x[n-n1] + a2.x[n-n2]$$

- Utilize Fs= 8k, n1 = correspondendo a um atraso de 10ms e n2 de 15ms.
- Fs = 8k, Ts = 125us.
- Portando: n1 corresponde a 80 amostras de delay e n2 120 amostras
- Use a0 = 0.5; a1 = 0.3 e a2 = 0.2.
- Valide o algoritmo para uma entrada impulso unitário

 Para uma entrada impulso unitário, n1 de 8 amostras de delay e n2 de 12 amostras,

temos:



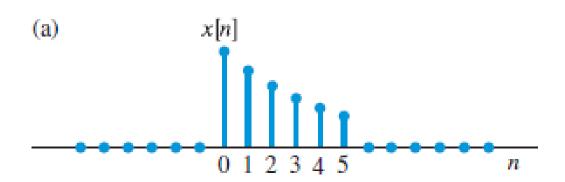
```
% Exemplo delay
clear all; close all; clc;
Fs = 8000;
t1 = 1.0*10^{-3}; t2 = 1.5*10^{-3};
n1 = t1*Fs; n2 = t2*Fs
% Definição dos ganhos
a0 = .5; a1 = .3; a2 = .2;
tama delay = n2;
vetor delay = zeros(tama delay ,1);
% Definindo a entrada
entrada = zeros(2*tama delay,1);
entrada(1,1) = 1; % definindo o impulso unitário
tama loop = length(entrada);
vet saida = zeros(tama loop,1);
for j = 1: tama loop
  input = entrada(j,1);
  vetor delay(1,1) = input;
  y = a0* vetor delay(1,1) + a1 * vetor delay(n1,1) + a2 * vetor delay(n2,1);
  % Desloca o vetor de delay
  for k = 0: tama delay-2
         vetor delay(tama delay-k,1) = vetor delay(tama delay-k-1,1);
  end
 vet saida(j,1) = y;
end
stem(vet saida)
title('Teste Delay');
```

 TAREFA: Modifique o programa implementado para gerar o eco em um arquivo de aúdio (Ocenaudio) com atrasos bem maiores de forma que sejam perceptiveis ao ouvir.

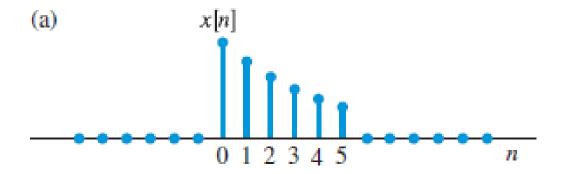
- Transformação na varíavel indepedente n:
- Reversão no tempo
- Deslocamento no tempo

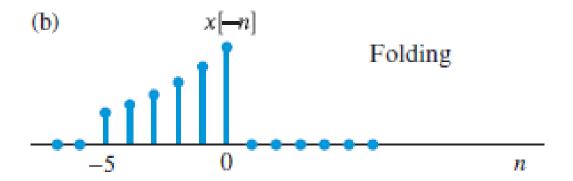
#### Reversão no tempo

Time-reversal or folding, which is an operation defined by y[n] = x[-n], reflects the sequence x[n] about the origin n = 0. Folding a sequence in MATLAB is done using the



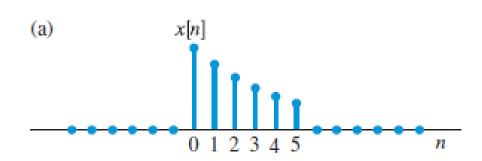
Reversão no tempo



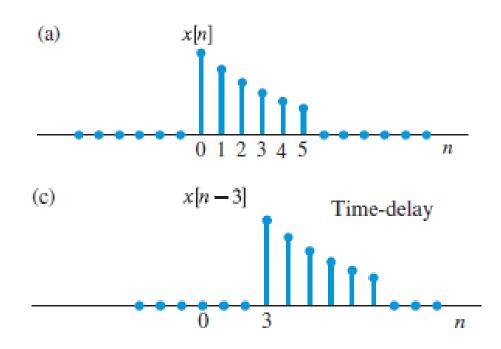


#### Deslocamento no tempo

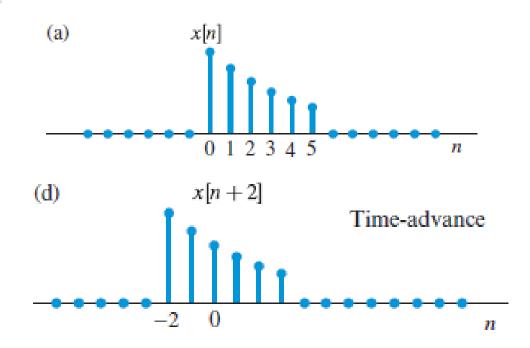
Time-shifting is defined by the formula  $y[n] = x[n - n_0]$ . For  $n = n_0$  we have,  $y[n_0] = x[0]$ ; thus, the sequence x[n] is shifted by  $n_0$  samples so that the sample x[0] is moved to  $n = n_0$ . If  $n_0 > 0$ , the sequence x[n] is shifted to the right; because the sequence "appears later," the shift corresponds to a time-delay. If  $n_0 < 0$ , the sequence is shifted to the left; because the sequence "appears earlier," the shift amounts to a time-advance.

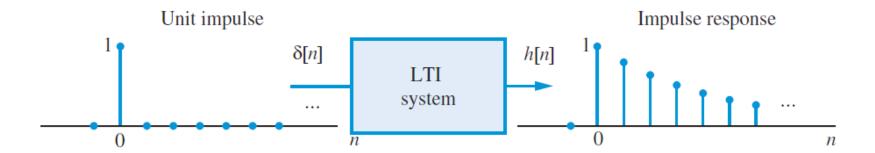


Deslocamento no tempo - Atraso



Deslocamento no tempo - Avanço





Signal decomposition into impulses Let us define the sequence

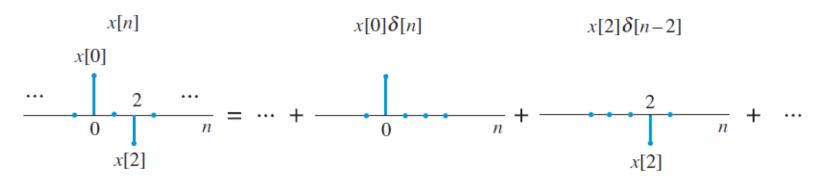
$$x_k[n] = \begin{cases} x[k], & n = k \\ 0, & n \neq k \end{cases}$$
 (2.29)

which consists of the sample x[k] of  $\{x[n]\}$  at n=k and zero elsewhere. The sequence  $x_k[n]$  can be obtained by multiplying the sequence  $\{x[n]\}$  by the sequence

$$\delta[n-k] = \begin{cases} 1, & n=k\\ 0, & n \neq k \end{cases}$$
 (2.30)

Hence, the sequence  $\{x[n]\}$  can be expressed as

$$x[n] = \sum_{k = -\infty}^{\infty} x[k]\delta[n - k]. \quad -\infty < n < \infty$$
 (2.31)



**Figure 2.10** Decomposition of a discrete-time signal into a superposition of scaled and delayed unit sample sequences.

Convolution sum We now illustrate how the properties of linearity and time-invariance restrict the form of discrete-time systems and simplify the understanding and analysis of their operation. More specifically, we show that we can determine the output of any LTI system if we know its impulse response.

We start by recalling that any sequence x[n] can be decomposed into a superposition of scaled and shifted impulses as in (2.31). Consider next a linear (but possibly time-varying) system and denote by  $h_k[n]$  its response to the basic signal  $\delta[n-k]$ . Then, from the superposition property for a linear system (see (2.27) and (2.28)), the response y[n] to the input x[n] is the same linear combination of the basic responses  $h_k[n]$ , that is,

$$y[n] = \sum_{k=-\infty}^{\infty} x[k]h_k[n], \qquad (2.34)$$

which is known as the *superposition summation* formula. Equation (2.34) provides the response of a linear time-varying system in terms of the responses of the system to the impulses  $\delta[n-k]$ .

If we impose the additional constraint that the system is time-invariant, we have

$$\delta[n] \xrightarrow{\mathcal{H}} h[n] \Rightarrow \delta[n-k] \xrightarrow{\mathcal{H}} h_k[n] = h[n-k].$$
 (2.35)

Substitution of (2.35) into (2.34) gives the formula

$$y[n] = \sum_{k=-\infty}^{\infty} x[k]h[n-k]. \quad -\infty < n < \infty$$
 (2.36)

Equation (2.36), which is commonly called the *convolution sum* or simply *convolution* is denoted using the notation y[n] = x[n] \* h[n]. Therefore, the response of a linear time-invariant system to any input signal x[n] can be determined from its impulse response h[n] using the convolution sum (2.36). If we know the impulse response of an LTI system, we can compute its response to any input without using the actual system. Furthermore,

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

The operation described by the convolution sum takes two sequences x[n] and h[n] and generates a new sequence y[n]. We usually say that sequence y[n] is the *convolution* of sequences x[n] and h[n] or that y[n] is obtained by convolving x[n] with h[n]. Convolution

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

$$y[n] = \sum_{k=-\infty}^{\infty} x[k]h[n-k]. \quad -\infty < n < \infty$$

# Convolução - Cálculo

Considere as sequências

$$x[n] = \{ 1 \ 2 \ 3 \ 4 \ 5 \}, \quad h[n] = \{ -1 \ 2 \ 1 \}.$$

x[k]	0	0	0	1	2	3	4	5	
h[k]			-1	2	1				
h[-k]			1	2	-1				

# Convolução - Cálculo

x[k]	0	0	0	1	2	3	4	5	
h[-k]			1	2	-1				
n = -1 h[-k-1]		1	2	-1					y[-1] = -1
n = 0 h[-k-0]			1	2	-1				y[0] = 0
n = 1 h[-k+1]				1	2	-1			y[1] = 2
n = 2 h[-k+2]					1	2	-1		y[2] = 4

## Convolução – Resumo do cálculo

In summary, the computation of convolution of two sequences involves the following steps:

- 1. Change the index of the sequences h[n], x[n] from n to k and plot them as a function of k.
- 2. Flip (or fold) the sequence h[k] about k = 0 to obtain the sequence h[-k].
- 3. Shift the flipped sequence h[-k] by n samples to the right, if n > 0, or to the left, if n < 0.
- 4. Multiply the sequences x[k] and h[n-k] to obtain the sequence  $z_n[k] = x[k]h[n-k]$ .
- 5. Sum all nonzero samples of  $z_n[k]$  to determine the output sample at the given value of the shift n.
- 6. Repeat steps 3-5 for all desired values of n.

# Convolução – Resumo do cálculo

Considere as sequências

$$x[n] = \{ 1 \ 1 \ 1 \ 1 \ 1 \ 1 \}, \quad h[n] = \{ 1 \ 0.5 \ 0.25 \ 0.125 \},$$

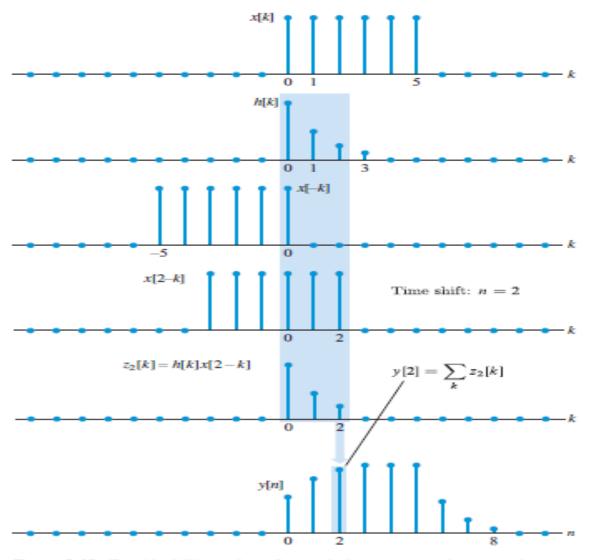
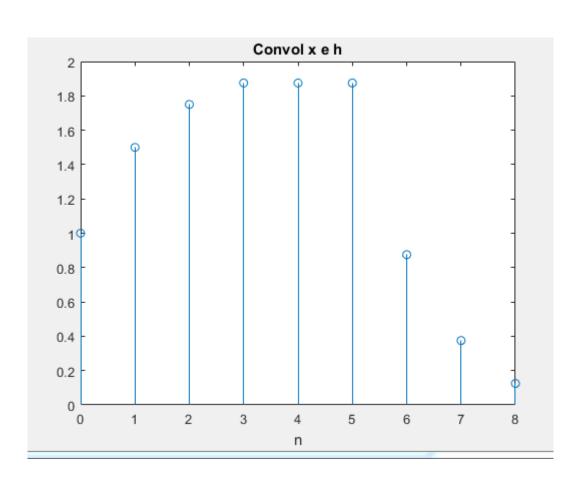


Figure 2.12 Graphical illustration of convolution as a scanning operation.

#### Programa exemplo no Matlab

```
% Exemplo do calculo da convolução
clear all; close all; clc;
x = [1 1 1 1 1 1];
h = [1.5.25.125];
y = conv(x,h)
tama = length(x) + length(h) -1;
n = (0:1:tama-1);
stem(n,y)
title('Convol x e h');
xlabel('n');
```



#### Tarefas:

- Considere o filtro média móvel da aula passada com k = 8, obtenha:
- A saída do filtro para as seguintes entradas: impulso unitário, degrau unitário e o vetor x[n] = [1, 0.5, 0.25, 0.125]

### Próxima aula

- 3) Transformada de Fourier Discreta
- 4) Transformada Z
- 5) Resposta em frequencia

# Disciplina: Processamento Digital de Sinais

Material\_aula 3

**Ambiente Blackboard**