



Basic concepts about signals & systems





Outline



- Introduction
- 2. Basic signals and operations with signals
- 3. The Fourier Transform
- 4. Linear Time Invariant Systems
- 5. Filters and resonators
- 6. The source-filter model





Introduction: what's a signal?



- Examples:
 - Speech signal
 - Image or photograph
 - Data about blood pressure, temperature... of a patient

Annual Mean Temperature

- Data about temperature, humidity, atmospheric pressure
- Formally they are represented as a function of one or more variables





Introduction: what's a system?

• System: any process or device that modifies a

signal

– Examples:

Speech production system

Communication system

 Represented by transforms that modify the input signal to produce the output signal

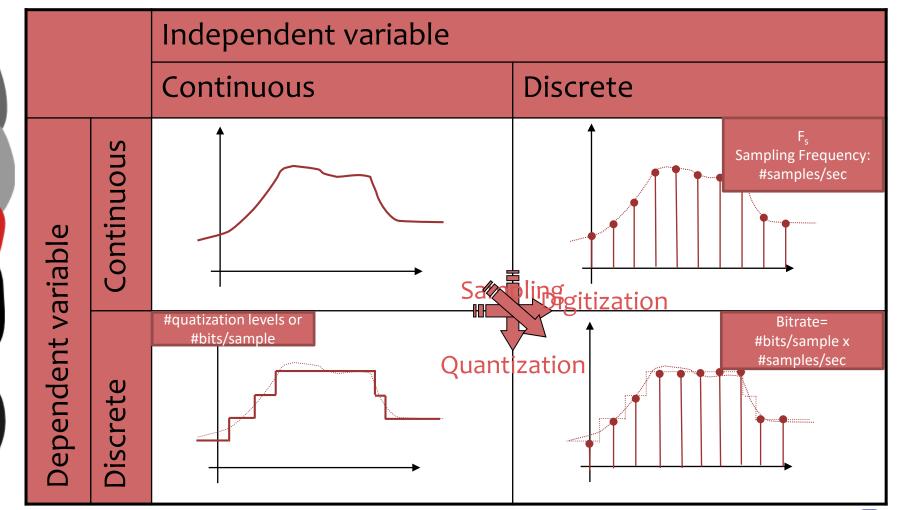
opening

$$x(t)$$
 $T()$ $y(t)=T(x(t))$





Analog vs. digital signals









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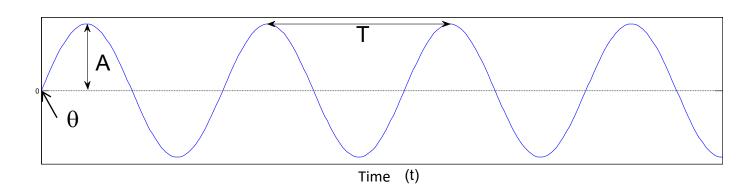


The simplest vibration in nature is the sinusoid

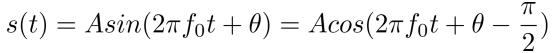


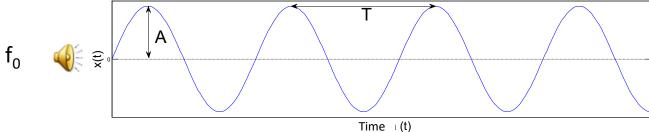
 $s(t) = A\sin(2\pi f_0 t + \theta)$ A: amplitude θ : phase

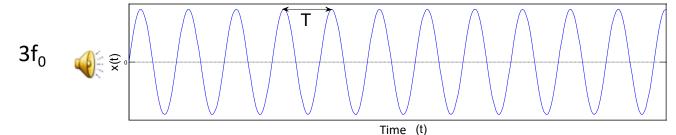
> f₀ (Hz): frequency $T=1/f_0$: period

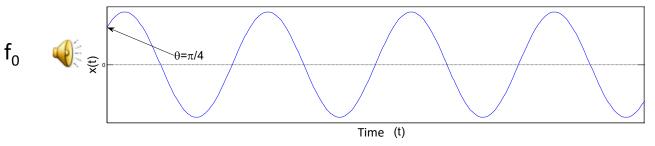










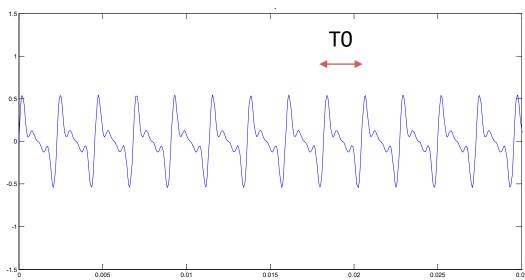






For simplicity, we will ignore the phase of the sinusoides: i.e. we will treat equally sinus (sin) and cosinus (cos)

A periodic signal is a signal that repeats its values in regular intervals or **periods**

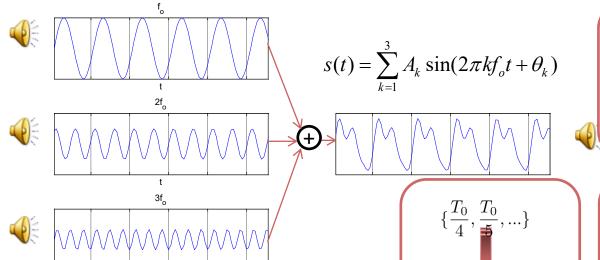


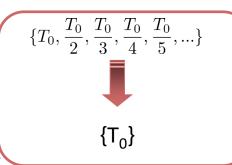
- The minimum repetition interval is the *fundamental period* (measured in seconds s)
- The inverse of the fundamental period is the *fundamental frequency*, it expresses the number of cicles (or periods) by second, and its unit is the Herz (Hz)

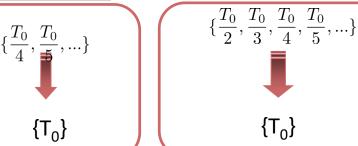


S

If we combine sinusoids with periods multiples one to each other, a periodic signal will result, with a period the **least common multiple** of the individual periods:









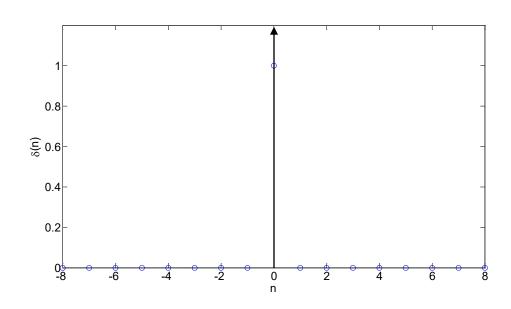


The **least common multiple** of the frequency components is perceived as the 'tone' of the signal (Missing fundamental effect)

Basic signals: The unit impulse signal

- Discrete time unit impulse or delta
 - 0 where n≠0

$$\delta[n] = \begin{cases} 1 & n = 0 \\ 0 & rest \end{cases}$$







Basic signals: The unit impulse signal



- 0 where t≠0
- Area 1

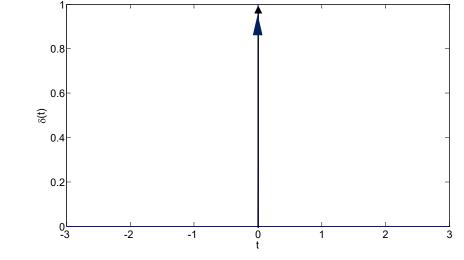
 $\delta(t) = \begin{cases} \infty & t = 0 \\ 0 & rest \end{cases}$

$$\int_{-\infty}^{\infty} \delta(t) dt = 1$$

Possible mathematic definition

$$\delta(t) = \lim_{T \to 0} \frac{1}{T} \prod \left(\frac{t}{T} \right)$$

Represented by an arrow in t=0



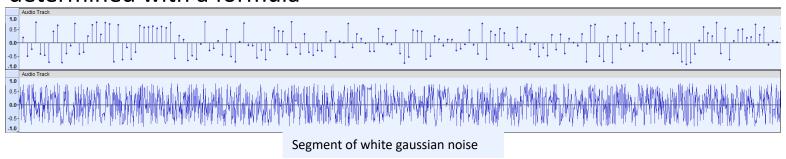






Basic signals: noise

 Noise is a signal with a random nature whose values can not be exactly determined with a formula



- It can be statistically modeled: gaussian noise is the most used model in speech.
- Noises differ also on their frequency components
- In speech modeling:
 - The speech signal for unvoiced sounds is modeled as noise
 - The glottal signal is usually taken as 'White Gaussian Noise' (WGN)



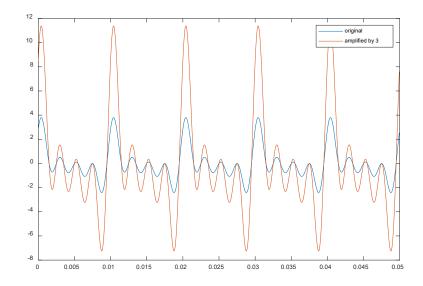








$$x[n] \longrightarrow k \cdot x[n]$$



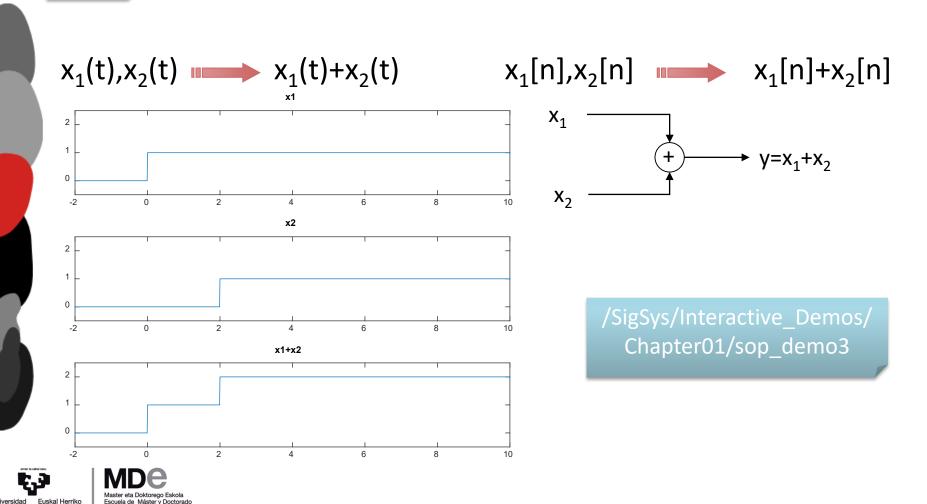


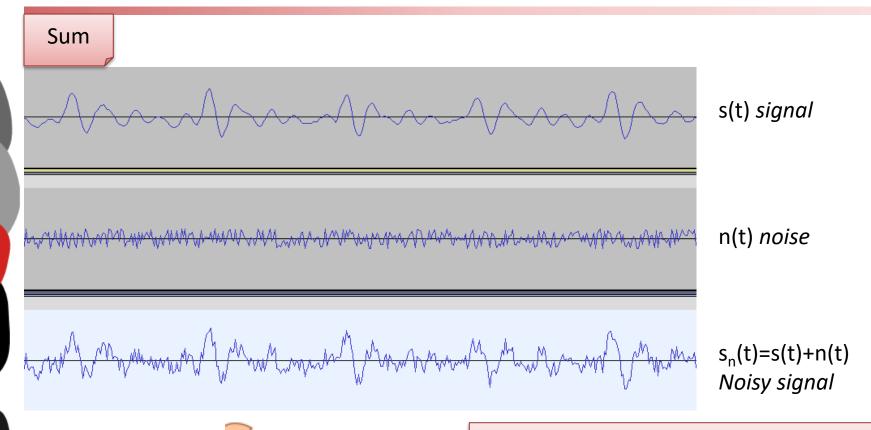






Sum







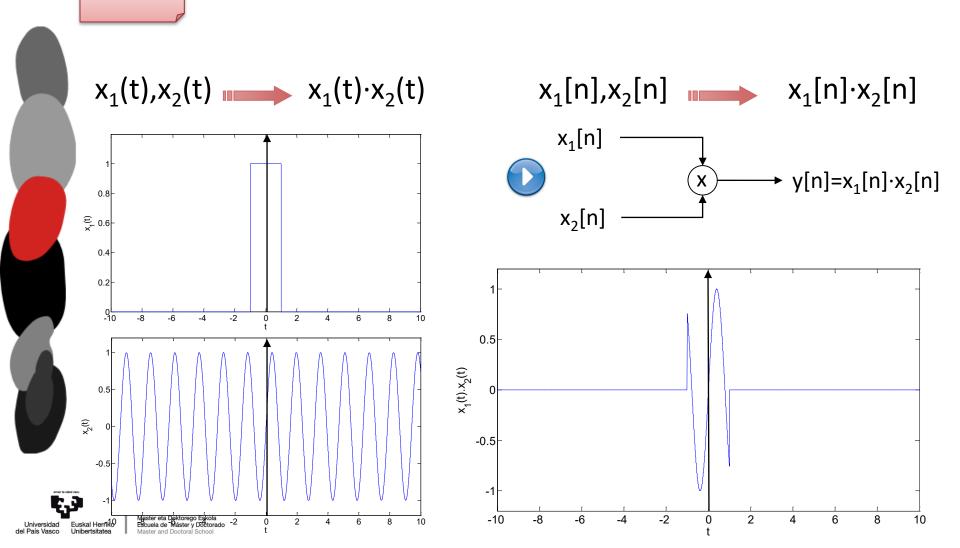


Addition occurs very frequently:

- Noise or other undesired interferences are added to the signals of interest
- Reflections of the main signal are added generating echoes and reverberation
- Signals from stereo or quadraphonic equipment are added before arriving to our ears...

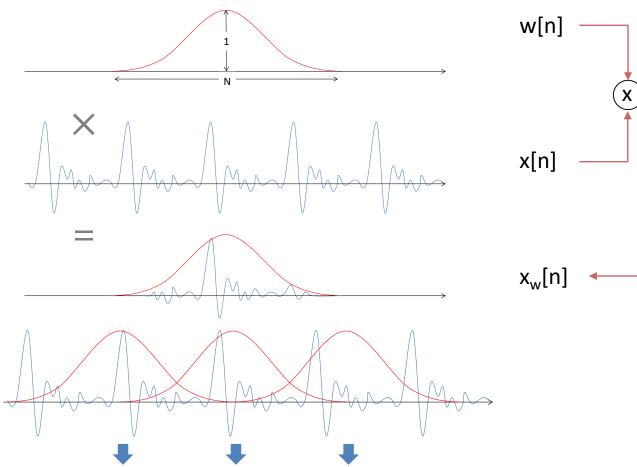


Product



Product

Windowing: the product of a signal and a *window* to select a short segment from the signal



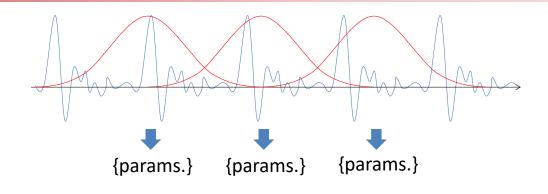
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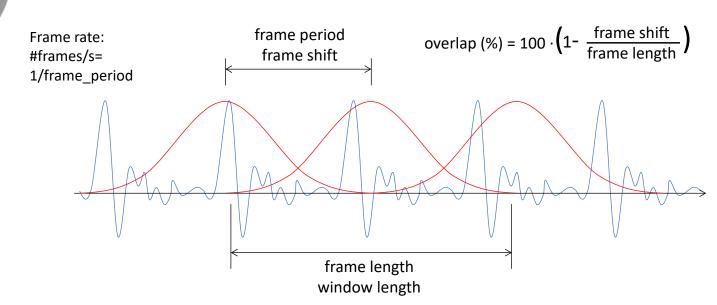




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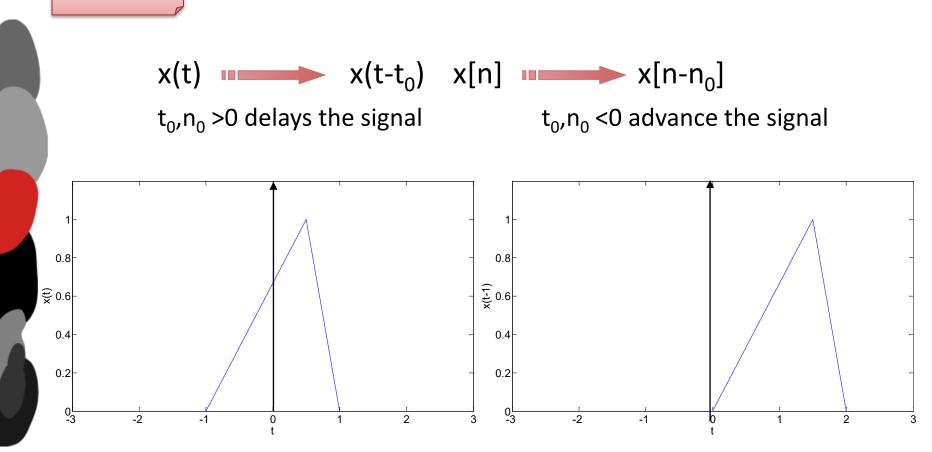






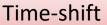


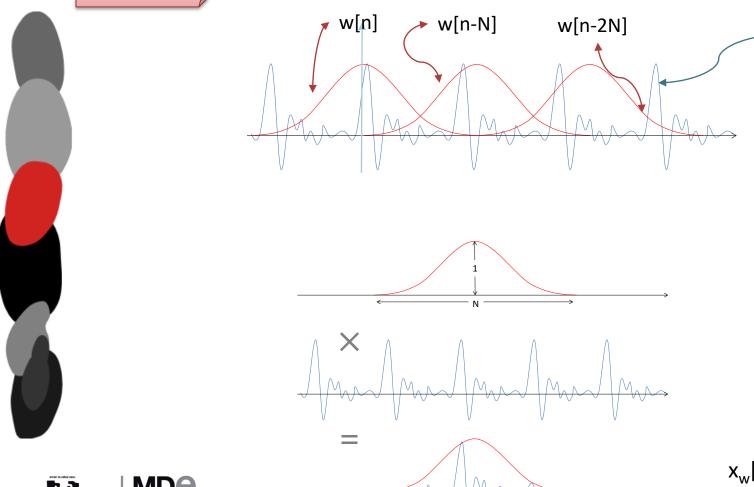
Time-shift

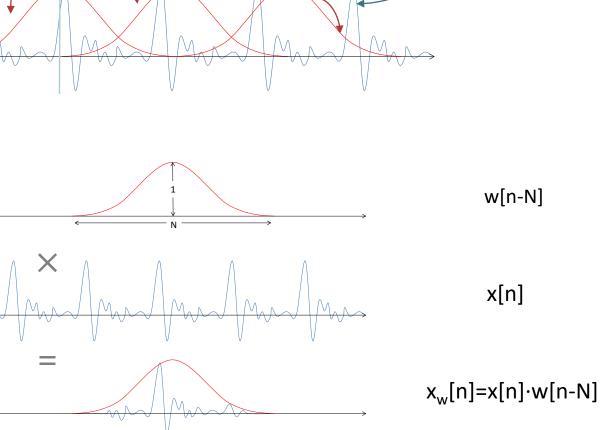












x[n]



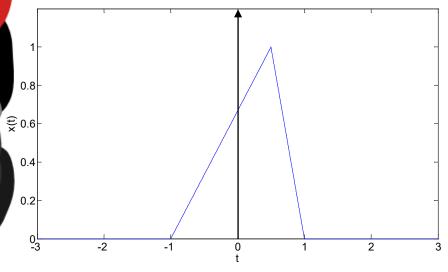


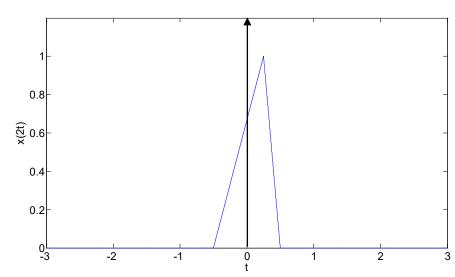


$$x(t) \longrightarrow x(at)$$

$$x[n] \longrightarrow x[an]$$

a>1 compresses the signal (in discrete time samples must be removed)





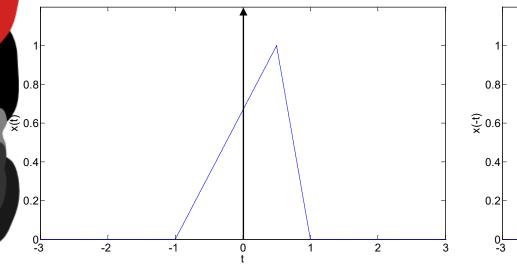


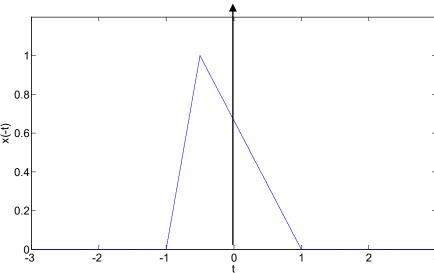




$$x(t) \longrightarrow x(-t)$$

$$x[n] \longrightarrow x[-n]$$



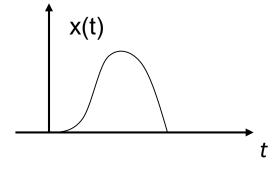






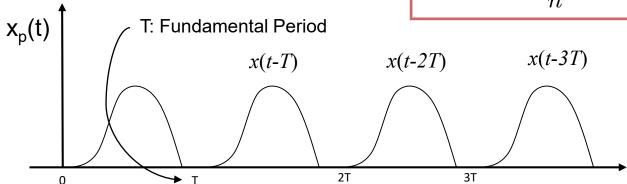
Periodic extension

We can build a periodic signal by repeating a non-periodic signal:



Periodic extension of x(t):

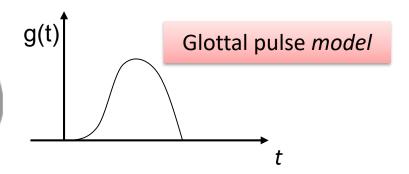
$$x_p(t) = \sum_{n} x(t - nT)$$





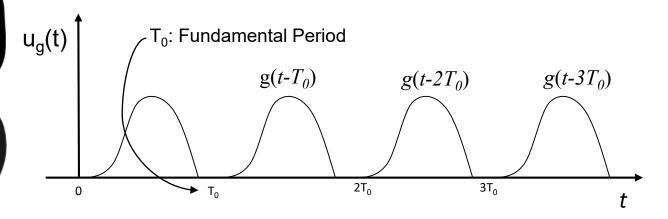


Periodic extension example: glottal signal



Gottal signal:

$$u_G(t) = \sum_{n} g(t - nT_0)$$







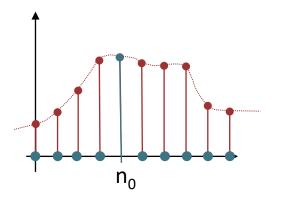




- Properties
 - Sampling

$$x[n]\delta[n]=x[0]\delta[n]$$

 $x[n]\delta[n-n_0]=x[n_0]\delta[n-n_0]$



Expressing any signal as a combination of unit impulses

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

Ch.1 id_demo









Properties

Sampling

$$x(t)\delta(t)=x(0)\delta(t) \qquad x[n]\delta[n]=x[0]\delta[n]$$

$$x(t)\delta(t-t_0)=x(t_0)\delta(t-t_0) \quad x[n]\delta[n-n_0]=x[n_0]\delta[n-n_0]$$

Expressing any signal as a combination of unit impulses



$$x(t) = \int_{-\infty}^{\infty} x(\tau)\delta(t-\tau)d\tau \qquad x[n] = \sum_{k=0}^{\infty} x[k]\delta[n-k]$$

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

Ch.1 id demo





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Why treat signals and systems in frequency?
 Easier to be analyzed

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

Fourier transform of x(t)
Analysis equation

$$x(t) \xrightarrow{F} X(f)$$

Time (s)

Frequency (Hz)

$$x(t) = F^{-1}{X(f)} = \int_{-\infty}^{\infty} X(f)e^{j2\pi ft}df$$

Inverse Fourier transform of X(f)
Synthesis equation

$$X(t)$$
 F^{-1} $X(f)$

Time (s)

Frequency (Hz)









The Fourier Transform of a signal x(t):

- Is a function of *frequency* (Hz): X(f)
- For each value of f, it measures the degree of similarity of x(t) with a pure tone of frequency f
- It can be inverted to recover x(t)
- The module squared $|X(f)|^2$ is a measure of the amount of energy in the signal at each frequency f.
- The module |X(f)| is called the **Spectrum** of x(t).

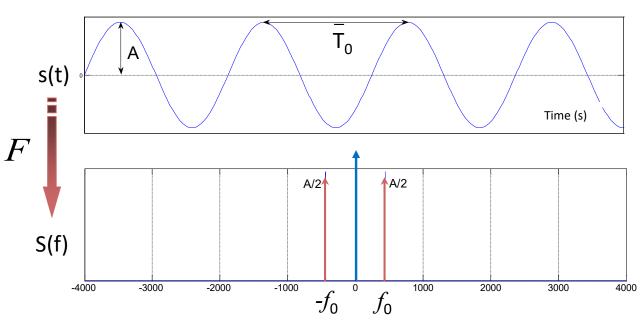






FT of a sinusoidal signal (only module):

$$s(t) = A\cos(2\pi f_0 t)$$



$$|X(f)| = \frac{A}{2} \{\delta(f - f_0) + \delta(f + f_0)\}$$

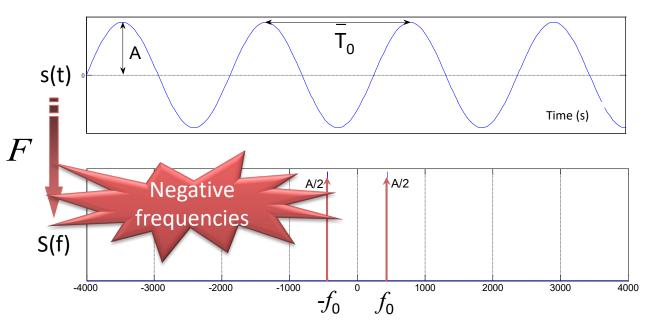






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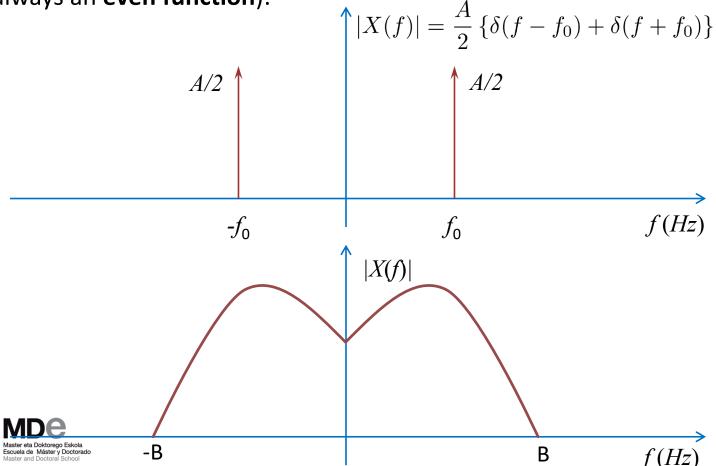
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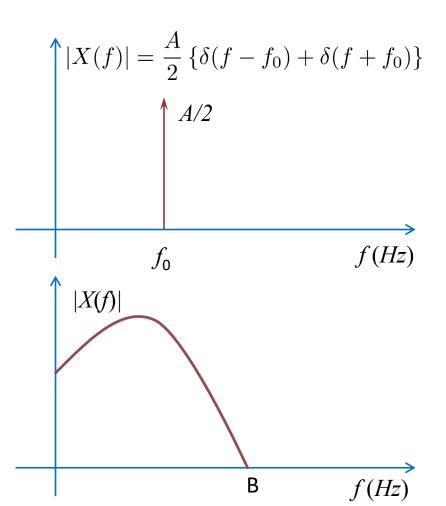


• In the Fourier Transform of a real signal there will always be components at negative and positive frequencies, with **reflected symmetry** (i.e., it is always an **even function**):





And then we usually only draw the positive frequencies:





Fourier transform for continuous times signals: Linearity of the Fourier transform

Linearity

X(t)	X(f)
k x(t)	k X(f)
$x_1(t) + x_2(t)$	$X_1(f)+X_2(f)$
$k_1 x_1(t) + k_2 x_2(t)$	$k_1 X_1(f) + k_2 X_2(f)$

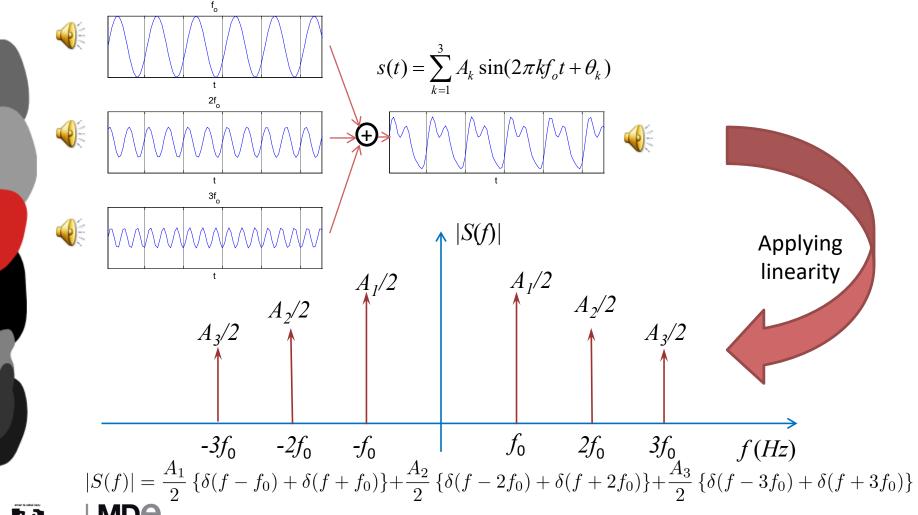
- The Fourier transform of an scaled version of a signal is the scaled version of the FT of the signal.
- The Fourier transform of a sum of two (scaled) signals is the sum of the (scaled) Fourier Transform of each signal.







Fourier transform for continuous times signals: FT of multi-tone signals

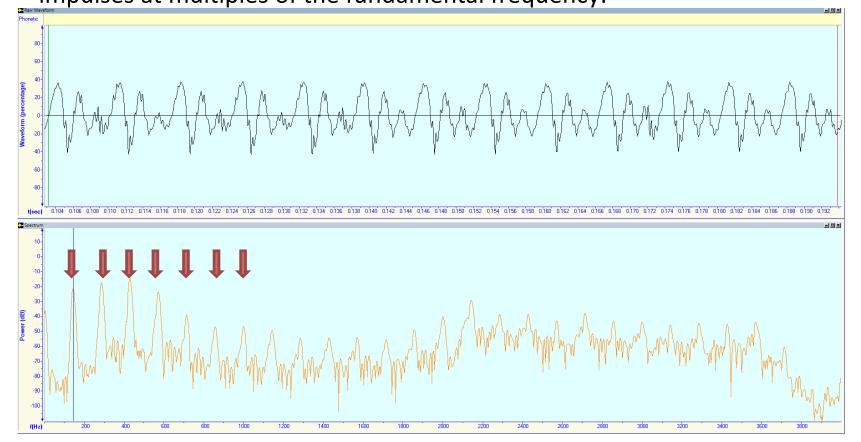






Fourier transform for continuous times signals: FT of periodic signals

The Fourier Transform (*spectrum*) of a periodic signal will always show impulses at multiples of the fundamental frequency:

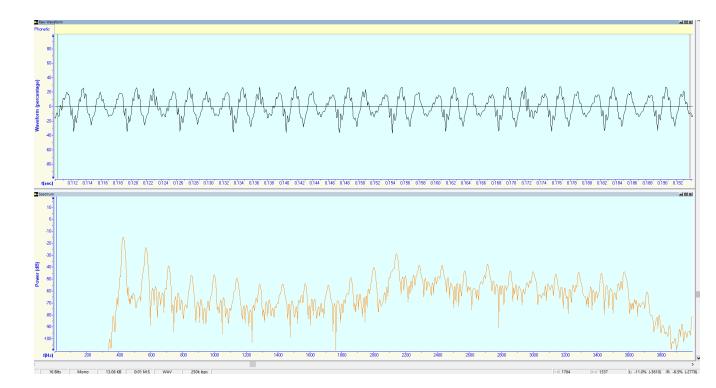






Fourier transform for continuous times signals: FT of periodic signals

- The fundamental F0 is not always present in the periodic speech signal.
- Equally, not all the harmonics are always present in the periodic speech signal.
- The fundamental frequency can be measured in the speech signal as the distance between two consecutive harmonics.



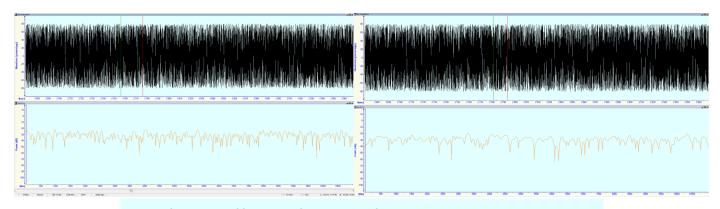






Fourier transform for continuous times signals: FT of noise

- Being noise of random nature, its FT will strongly depend on the selected segment
- It will also look very random



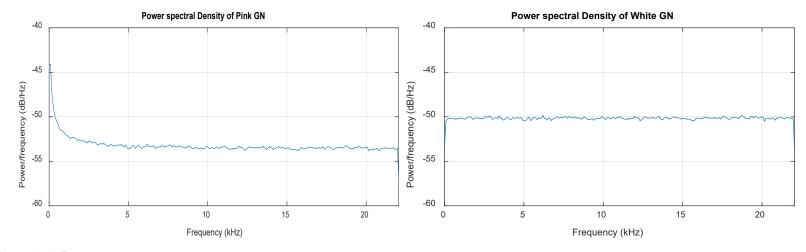






Fourier transform for continuous ting signals: FT of noise & PSD

 Power Specral Density: the spectrums calculated from consecutive frames are averaged to obtain a better estimation of the spectral content of a random signal.











Signals and systems: a Matlab integrated approach
 Oktay Alkin, 2014 CRC Press, ISBN: 978-1-4665-9853-9

http://www.signalsandsystems.org/



