SEISMOLOGY I

Laurea Magistralis in Physics of the Earth and of the Environment

Linear systems

Fabio ROMANELLI

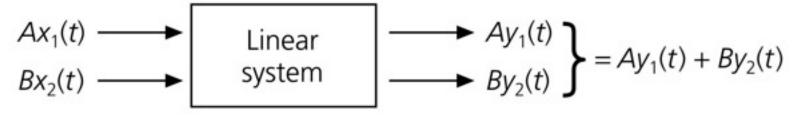
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Figure 6.3-1: Definition of a linear system.



$$x(t) = \int x(\tau)\delta(\tau - t)d\tau$$

$$\int x(\tau)h(\tau - t)d\tau$$

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

(remember GF definition)









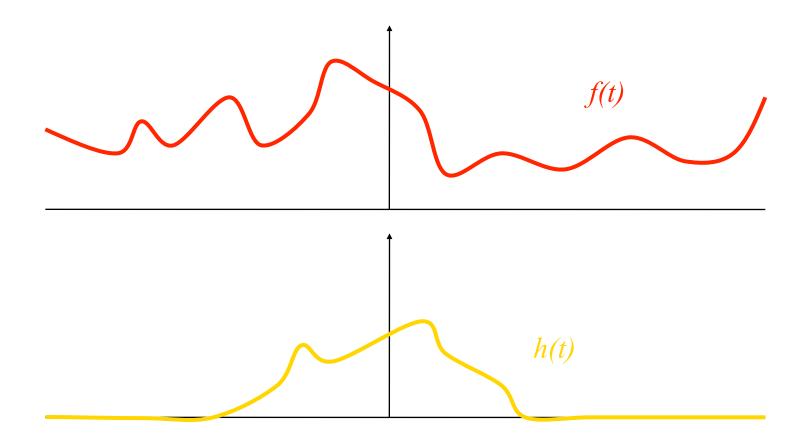
$f(t) * h(t) = \int_{-\infty}^{\infty} f(\tau)h(t - \tau)d\tau$







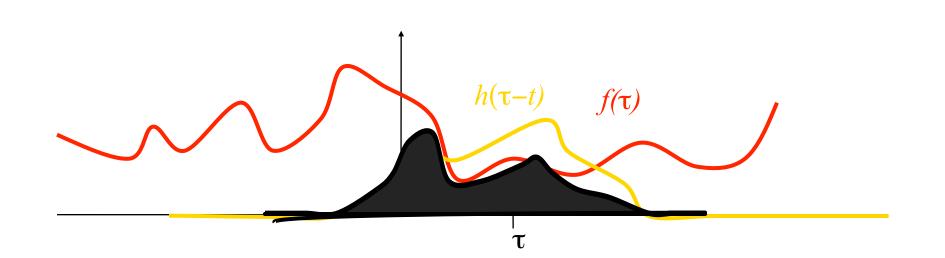










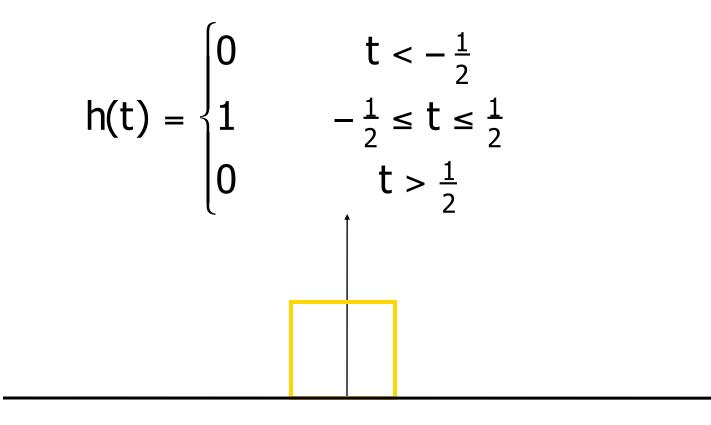








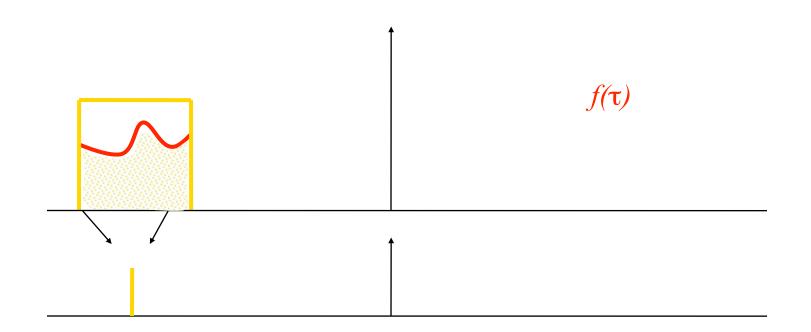
Consider the function (box filter):







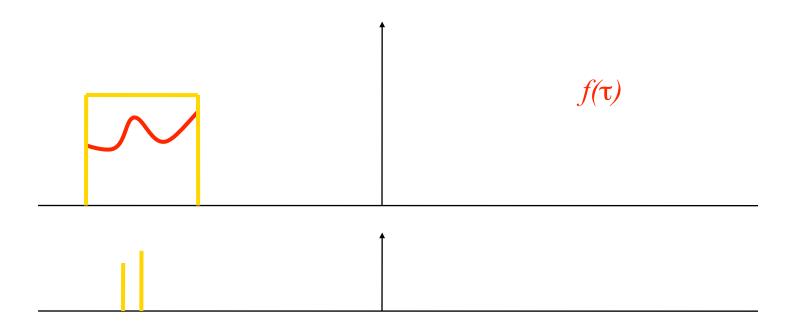








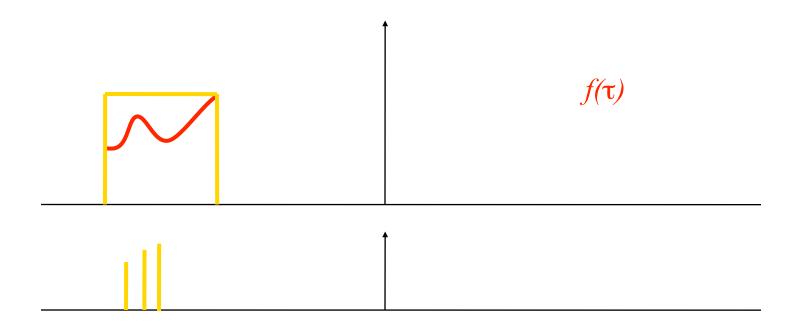








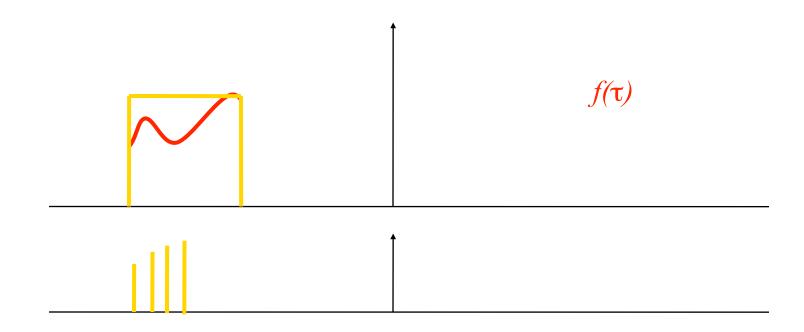








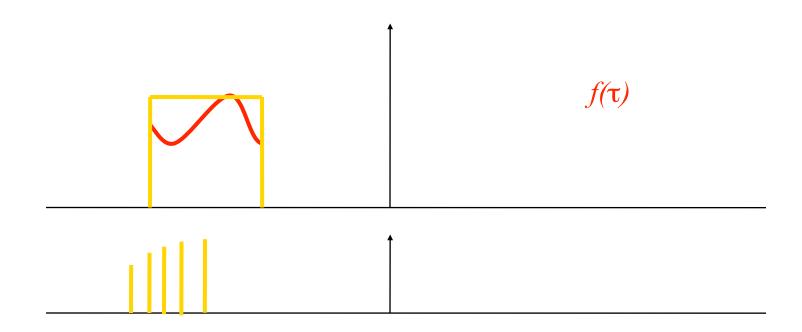








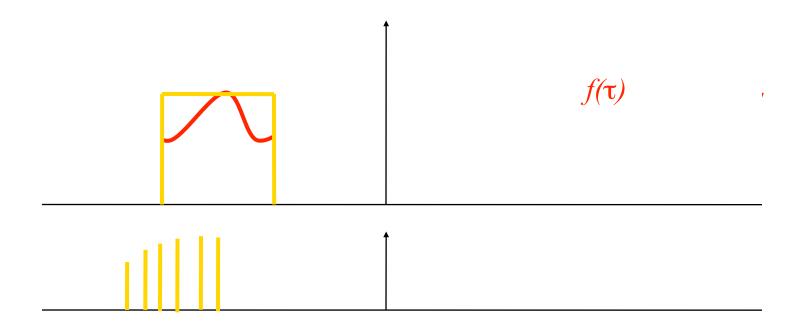








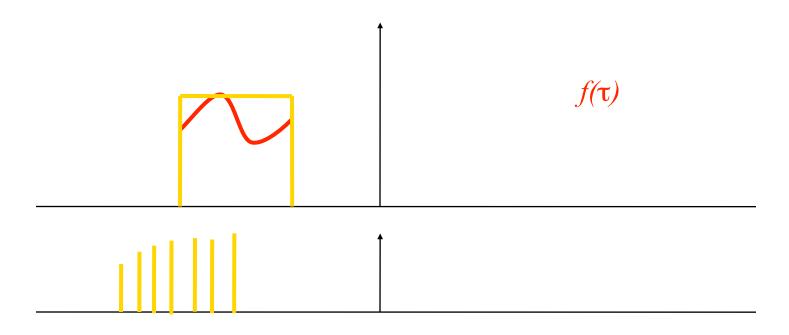








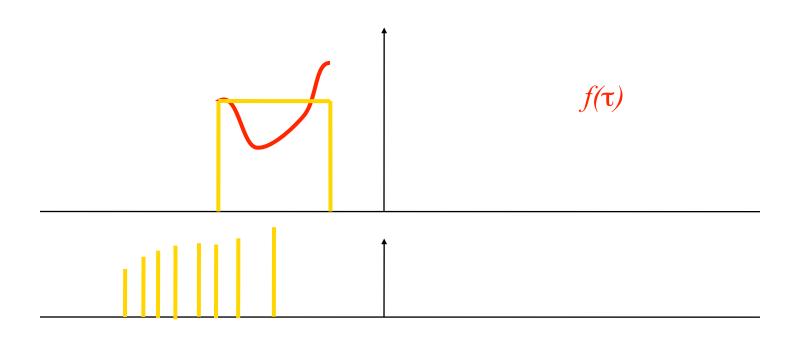








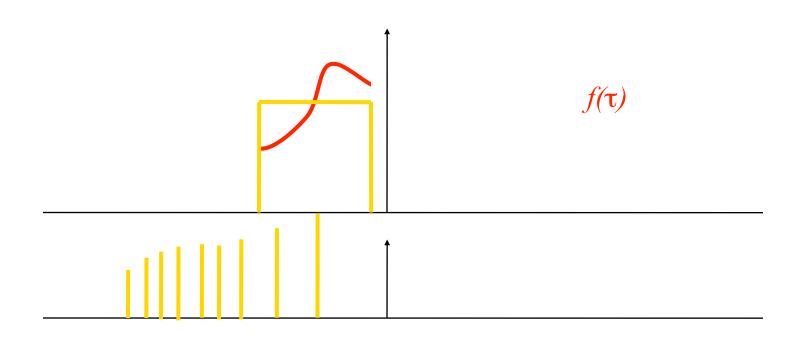








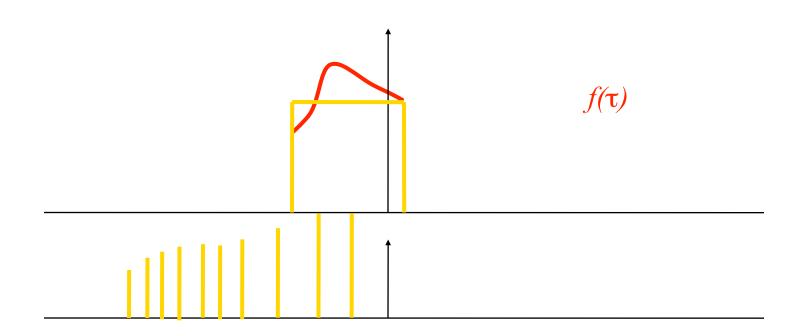








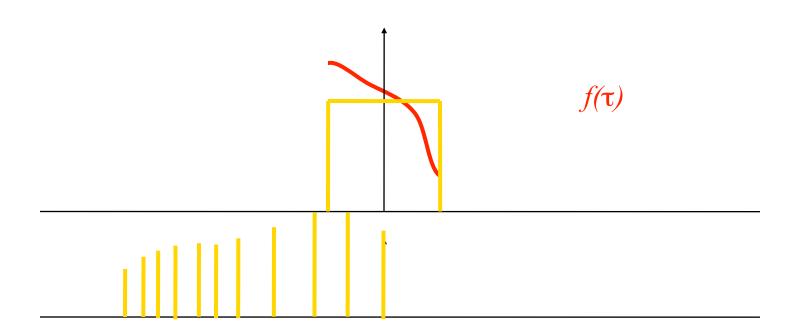








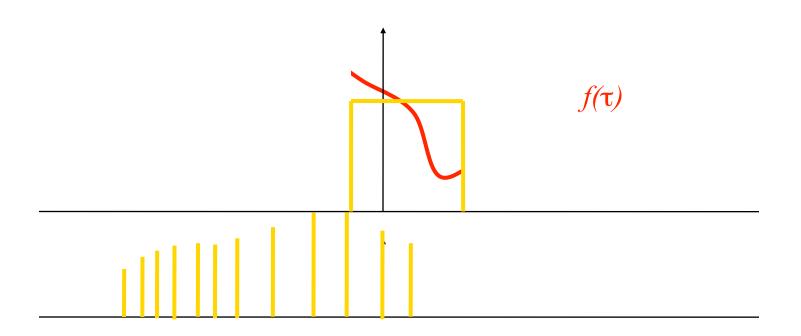








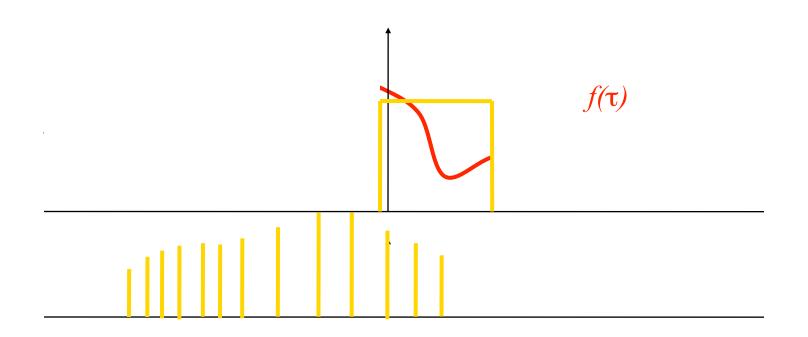








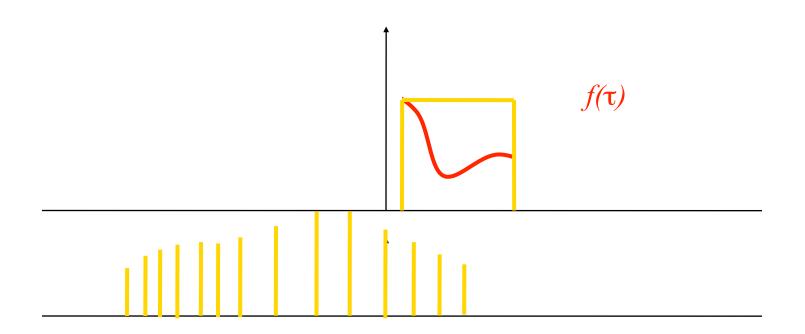








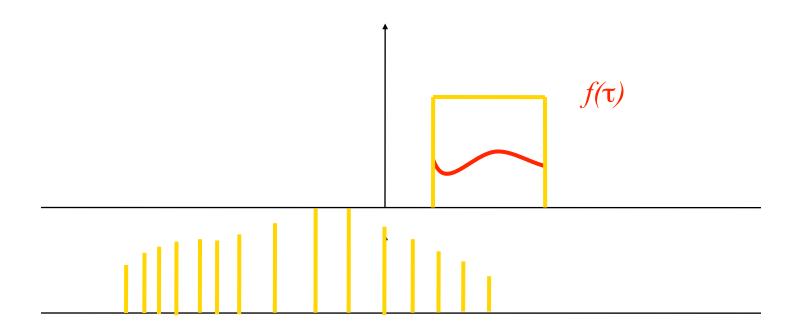








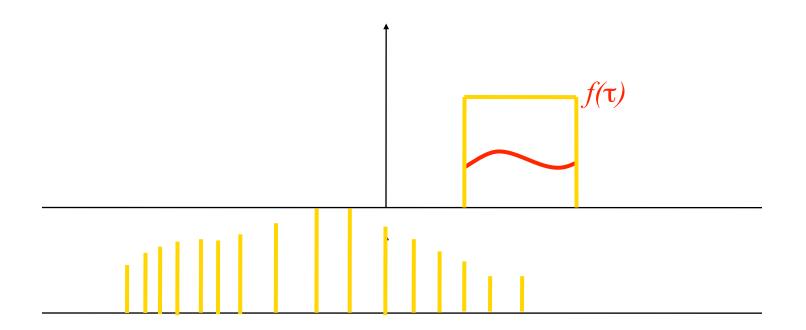








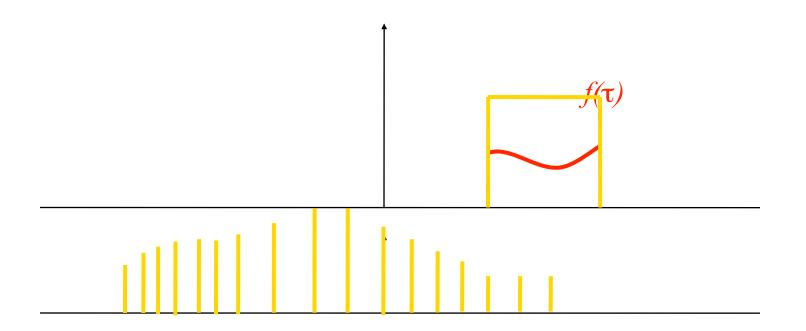








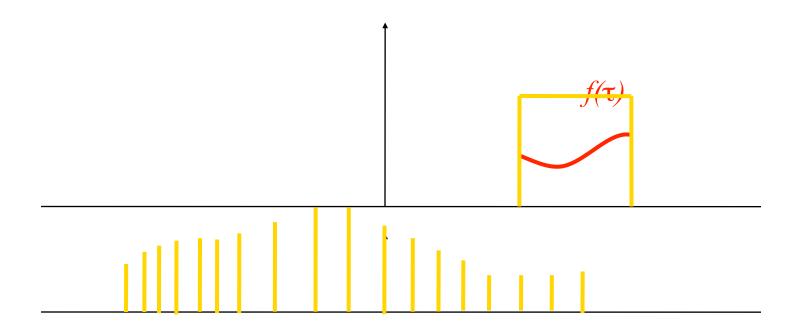








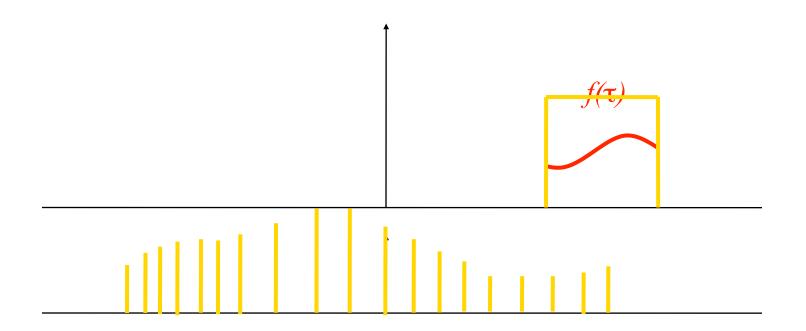








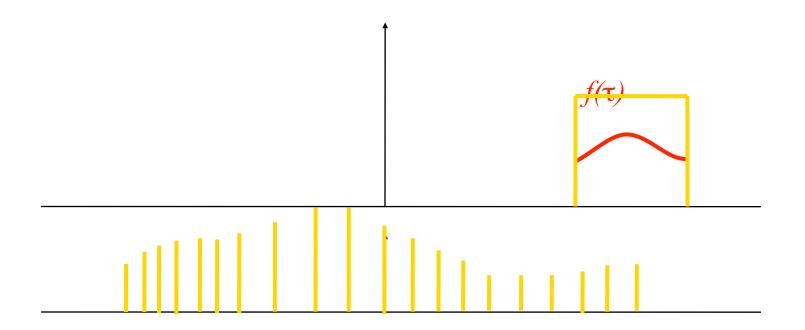








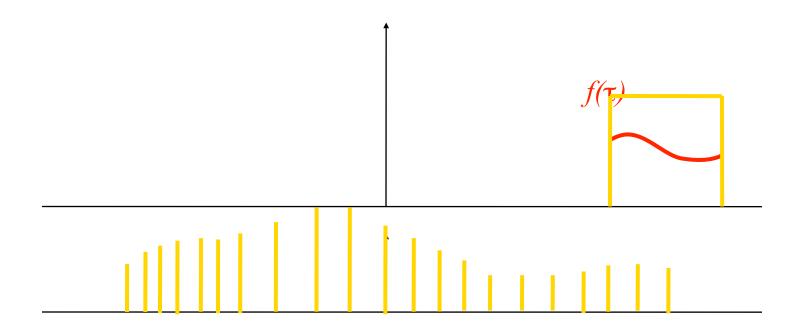








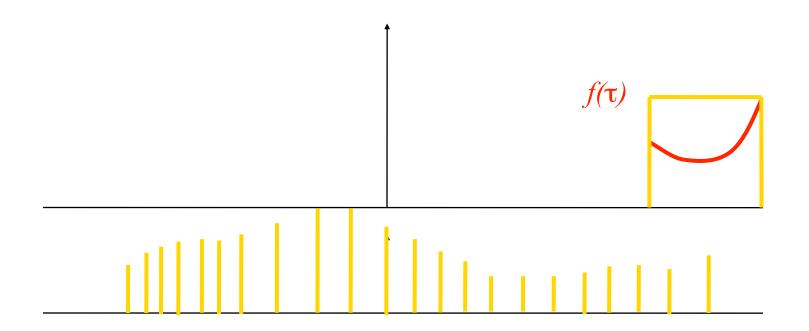










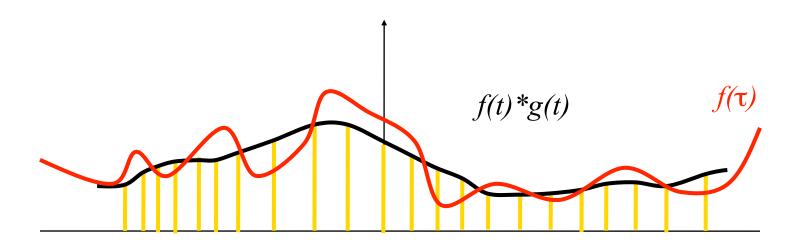






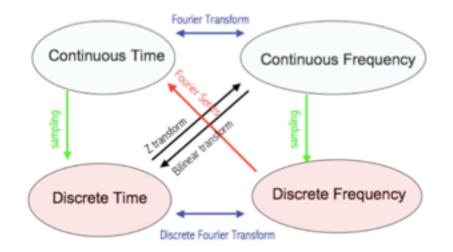


This particular convolution smooths out some of the high frequencies in f(t).

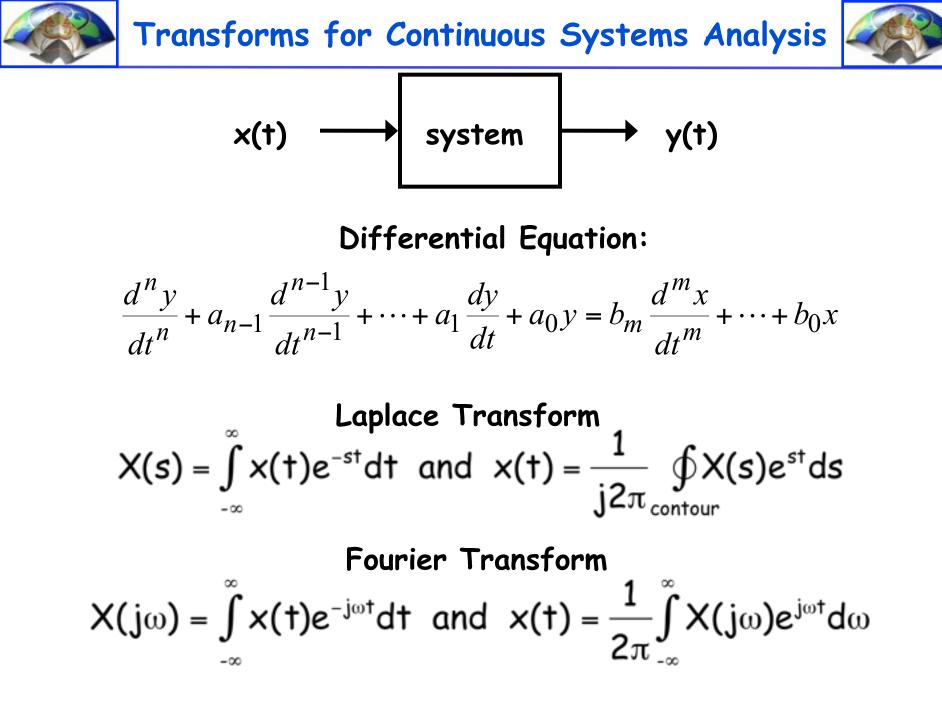








Signal type	Continuous time	Discrete time	Transform Domain
Finite duration	Laplace	Z	Continuous complex frequency (s-plane)
Finite duration	Fourier	Discrete-time Fourier (DTFT)	Continuous real frequency
Periodic	Fourier Series	Discrete Fourier Series (DFS)	Discrete real frequency







$$\frac{d^{n}y}{dt^{n}} + a_{n-1}\frac{d^{n-1}y}{dt^{n-1}} + \dots + a_{1}\frac{dy}{dt} + a_{0}y = b_{m}\frac{d^{m}x}{dt^{m}} + \dots + b_{0}x$$
Transfer Function

$$s^{n}Y(s) + a_{n-1}s^{n-1}Y(s) + \dots + a_{1}s^{1}Y(s) + a_{0}s^{0}Y(s) = b_{m}s^{m}X(s) + b_{m-1}s^{m-1}X(s) + \dots + b_{0}X(s)$$

$$H(s) = \frac{Y(s)}{X(s)} = \frac{s^{m} + b_{m-1}s^{m-1} + \dots + b_{1}s + b_{0}}{s^{n} + a_{n-1}s^{n-1} + \dots + a_{1}s + a_{0}} = \frac{(s - z_{1})(s - z_{2})\cdots(s - z_{m})}{(s - p_{1})(s - p_{2})\cdots(s - p_{n})}$$
Frequency Response

$$(j\omega)^{n}Y(j\omega) + a_{n-1}(j\omega)^{n-1}Y(j\omega) + \dots + a_{0}Y(j\omega) = b_{m}(j\omega)^{m}X(j\omega) + \dots + b_{0}X(j\omega)$$

$$H(j\omega) = \frac{Y(j\omega)}{X(j\omega)} = \frac{b_{m}(j\omega)^{m} + b_{m-1}(j\omega)^{m-1} + \dots + a_{1}(j\omega) + b_{0}}{(j\omega)^{n} + a_{n-1}(j\omega)^{n-1} + \dots + a_{1}(j\omega) + a_{0}}$$

The values of where the numerators is zero are referred to as **zeros**, as the response is zero at this frequency, regardless of the amplitude of the input signal. Conversely, frequencies for which the denominator is zero are called **poles**, as the response becomes very large at these frequencies.





Difference Equation:

 $y[n] + a_1y[n-1] + ... + a_ky[n-k] = b_0m[n] + b_1m[n-1] + ... + b_mm[n-l]$

$$X(z) = \sum_{n=-\infty}^{\infty} x[n]z^{-n} \text{ and } x[n] = \frac{1}{j2\pi} \oint X(z)z^{n-1}dz$$

Discrete Time Fourier Transform $X(e^{j\omega}) = \sum_{n=1}^{\infty} x[n]e^{-j\omega n}$ and $x[n] = \int X(e^{j\omega})e^{j\omega n}d\omega$ $n = -\infty$ 2π





$y[n] + a_1y[n-1] + ... + a_ky[n-k] = b_0m[n] + b_1m[n-1] + ... + b_mm[n-l]$

Transfer Function - z transforms

$$z^{n}Y(z) + a_{n-1}z^{n-1}Y(z) + \dots + a_{1}zY(z) + a_{0}Y(z) = b_{m}z^{m}X(z) + b_{m-1}z^{m-1}X(z) + \dots + b_{0}X(z)$$
$$H(z) = \frac{Y(z)}{X(z)} = \frac{b_{m}z^{m} + b_{m-1}z^{m-1} + \dots + b_{1}z + b_{0}}{z^{n} + a_{n-1}z^{n-1} + \dots + a_{1}z + a_{0}} = \frac{(z - z_{1})(z - z_{2})\cdots(z - z_{m})}{(z - p_{1})(z - p_{2})\cdots(z - p_{n})}$$

Frequency Response

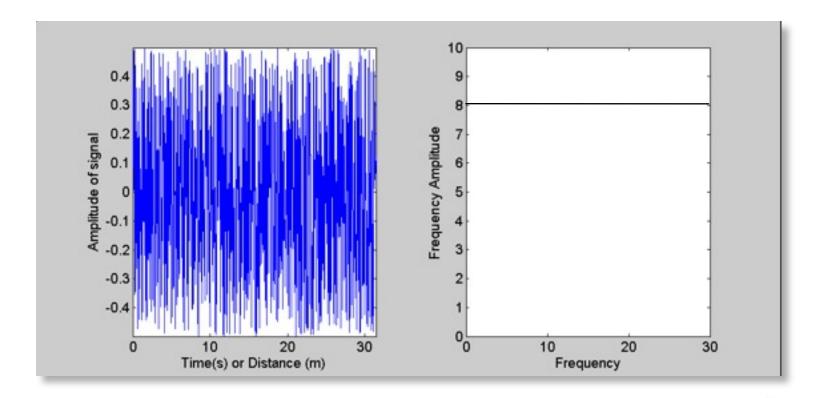
 $e^{j\omega n}Y(e^{j\omega}) + a_{n-1}e^{j\omega(n-1)}Y(e^{j\omega}) + \ldots + a_0Y(e^{j\omega}) = b_m e^{j\omega m}X(e^{j\omega}) + \ldots + b_0X(e^{j\omega})$

$$H(e^{j\omega}) = \frac{Y(e^{j\omega})}{X(e^{j\omega})} = \frac{b_m e^{j\omega m} + b_{m-1} e^{j\omega(m-1)} + \dots + b_1 e^{j\omega} + b_0}{e^{j\omega n} + a_{n-1} e^{j\omega(n-1)} + \dots + a_1 e^{j\omega} + a_0}$$

The values of where the numerators is zero are referred to as **zeros**, as the response is zero at this frequency, regardless of the amplitude of the input signal. Conversely, frequencies for which the denominator is zero are called **poles**, as the response becomes very large at these frequencies.



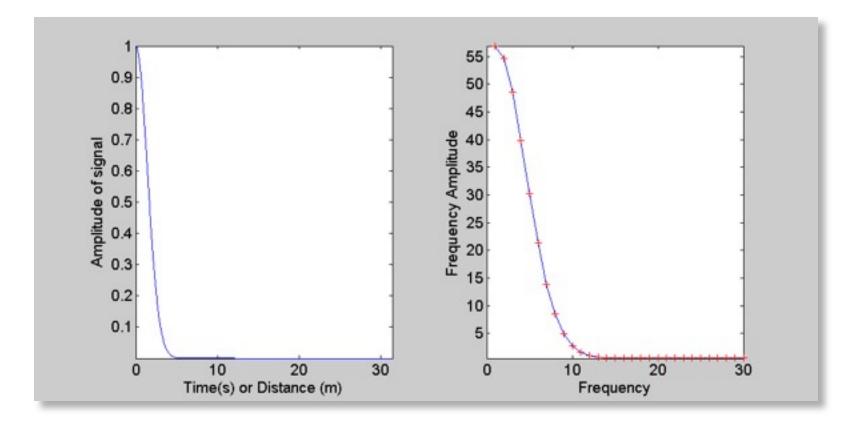
random signals



Random signals may contain all frequencies. A spectrum with constant contribution of all frequencies is called a white spectrum



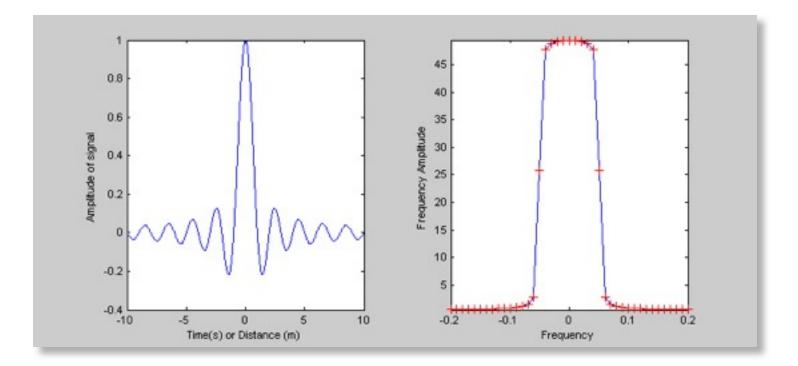
Gaussian signals



The spectrum of a Gaussian function will itself be a Gaussian function. How does the spectrum change, if I make the Gaussian narrower and narrower?



Transient waveform

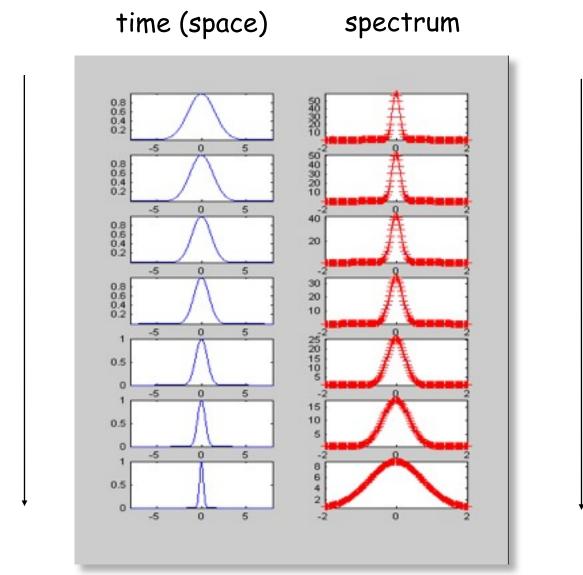


A transient wave form is a wave form limited in time (or space) in comparison with a harmonic wave form that is infinite





Widening frequency band



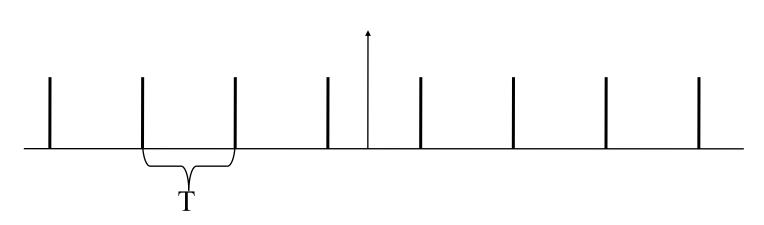
Narrowing physical signal





• A Sampling Function or Impulse Train is defined by: $S_{T}(t) = \sum_{k=-\infty}^{\infty} \delta(t - kT)$

where T is the sample spacing.







- The Fourier Transform of the Sampling Function is itself a sampling function.
- The sample spacing is the inverse.

$S_{T}(t) \Leftrightarrow S_{\frac{1}{T}}(\omega)$





The convolution theorem states that convolution in the spatial domain is equivalent to multiplication in the frequency domain, and viceversa.

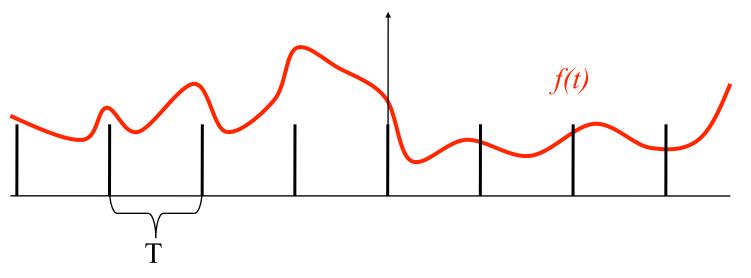
$f(t) * g(t) \Leftrightarrow F(\omega)G(\omega)$ $f(t)g(t) \Leftrightarrow F(\omega) * G(\omega)$





This powerful theorem can illustrate the problems with our point sampling and provide guidance on avoiding aliasing.

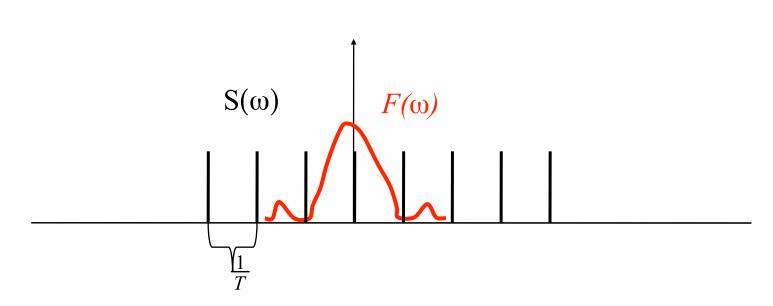








What does this look like in the Fourier domain?

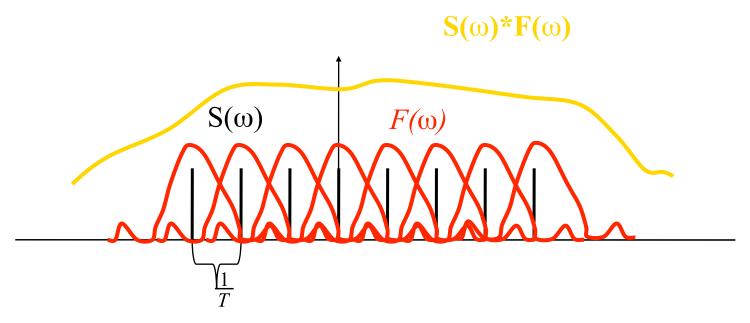


Seismology I - LS&FT





In Fourier domain we would convolve









- What this says, is that any frequencies greater than a certain amount will appear intermixed with other frequencies.
- In particular, the higher frequencies for the copy at 1/T intermix with the low frequencies centered at the origin.





- Note, that the sampling process introduces frequencies out to infinity.
- We have also lost the function f(t), and now have only the discrete samples.
- This brings us to our next powerful theory.





The Shannon Sampling Theorem

- A band-limited signal f(t), with a cutoff frequency of λ , that is sampled with a sampling spacing of T
 - may be perfectly reconstructed from the discrete values f[nT] by convolution with the sinc(t) function, provided:

$$\lambda < \frac{1}{2T}$$





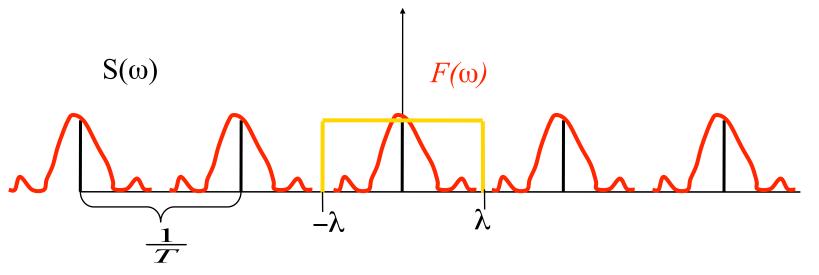
Why is this?

- The Nyquist limit will ensure that the copies of F (ω) do not overlap in the frequency domain.
- \bigcirc I can completely reconstruct or determine f(t) from F(ω) using the Inverse Fourier Transform.





- In order to do this, I need to remove all of the shifted copies of $F(\omega)$ first.
- This is done by simply multiplying $F(\omega)$ by a box function of width 2λ .

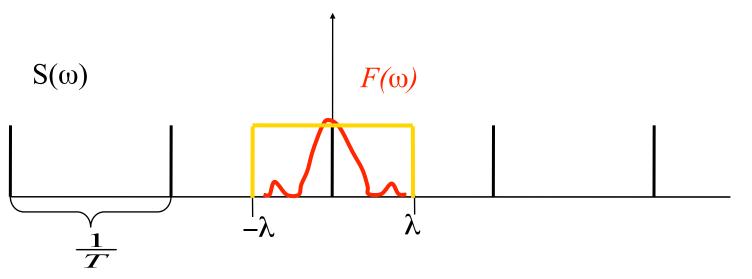


Seismology I - LS&FT





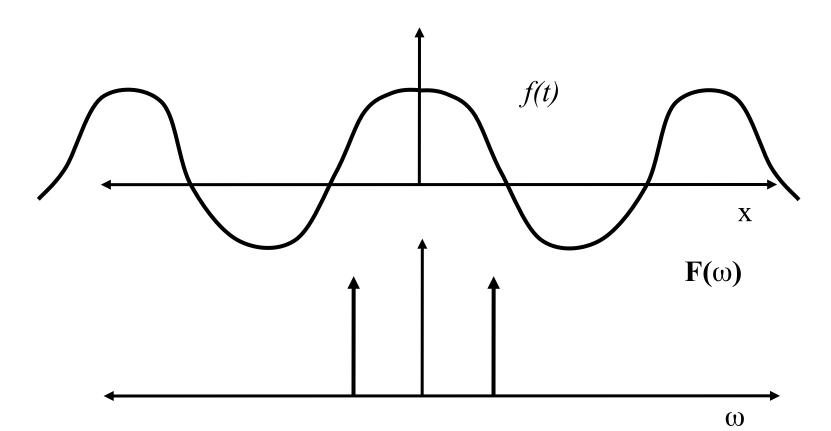
- In order to do this, I need to remove all of the shifted copies of $F(\omega)$ first.
- This is done by simply multiplying $F(\omega)$ by a box function of width 2λ .







Consider the function $f(t) = cos(2\pi t)$.







So, given f[nT] and an assumption that f(t) does not have frequencies greater than 1/2T, we can write the formula:

 $f[nT] = f(t) S_{T}(t) \Leftrightarrow F(\omega)^{*} S_{T}(\omega)$ $F(\omega) = (F(\omega)^{*} S_{T}(\omega)) Box_{1/2T}(\omega)$

therefore,

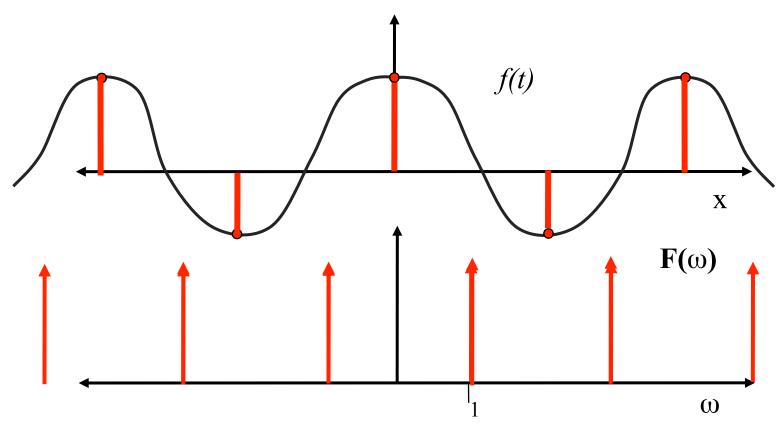
f(t) = f[nT] * sinc(t)







Now sample it at T=1/2



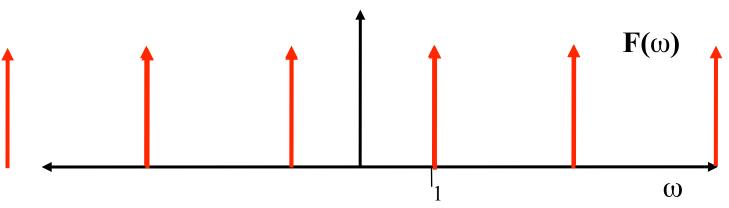




Problem:

The amplitude is now wrong or undefined.

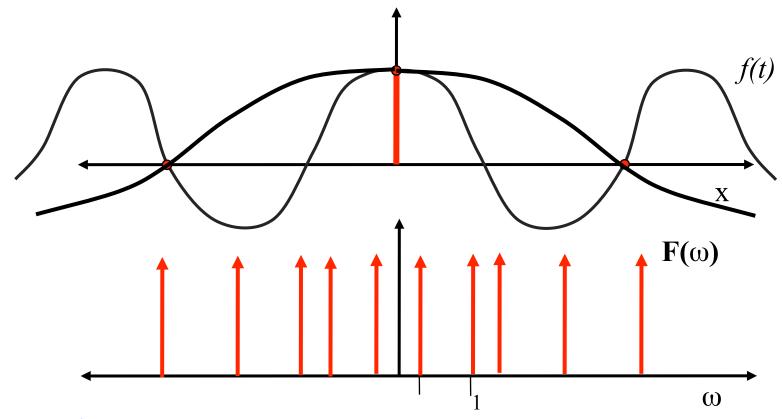
Note however, that there is one and only one cosine with a frequency less than or equal to 1 that goes through the sample pts.







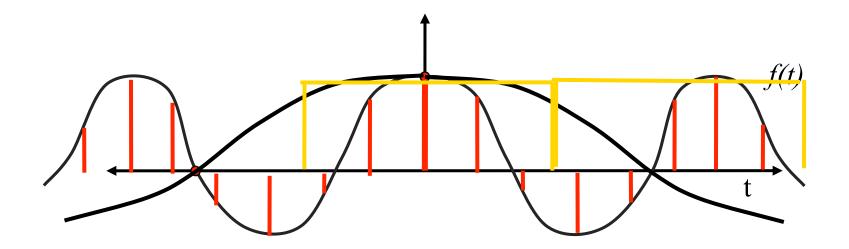
What if we sample at T=2/3?







Supersampling increases the sampling rate, and then integrates or convolves with a box filter, which is finally followed by the output sampling function.







The problem:

- The signal is not band-limited.
- Uniform sampling can pick-up higher frequency patterns and represent them as low-frequency patterns.

