

## **Section 2: Frequency Domain Analysis**

### **Contents**

**2.1 Fourier Series**

**2.2 Fourier Transforms**

**2.3 Convolution**

**2.4 The Sampling Theorem**

**2.5 The Analytic Signal**

**2.6 Applications of the Analytic Signal**

**(i) Phasors**

**(ii) Single Sideband Signals**

## **2. Frequency Domain Analysis**

### **2.1 Fourier Series**

A signal  $x(t)$  which has period  $T$  can be expressed as:

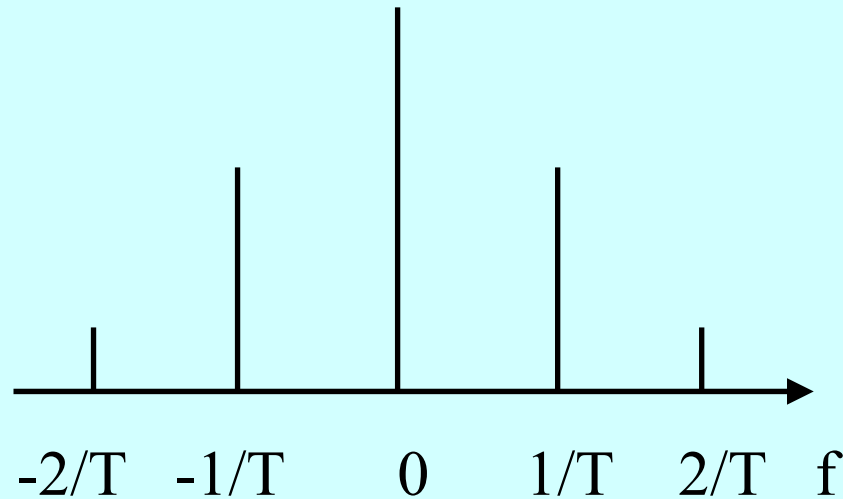
$$x(t) = \sum_{n=-\infty}^{\infty} X_n e^{jn2\pi t/T}$$
$$X_n = \frac{1}{T} \int_{(T)} x(t) e^{-jn2\pi t/T} dt$$

## Notes:

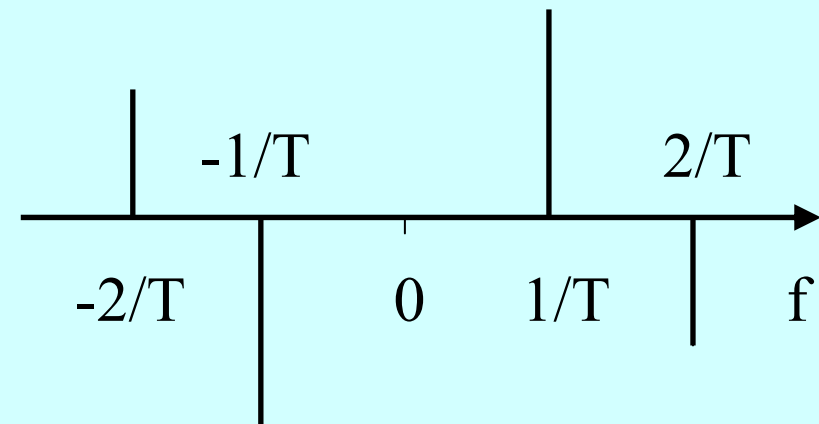
- (T) means integration over any time interval of length T.
- We will use  $f$  in Hz as the frequency variable. The symbol  $\omega$  will only be used to represent  $2\pi f$ .
- Frequency  $f$  is a property of the complex exponential  $e^{j2\pi ft}$  and may be positive or negative. Real signals contain both positive and negative frequencies. eg.  $\cos(2\pi ft) = 0.5e^{j2\pi ft} + 0.5e^{-j2\pi ft}$ .

- The number  $X_n$  is a complex number (dimensions V) and represents a component of frequency  $n/T$  Hz, and for a real signal  $X_{-n} = X_n^*$ . This is called the *Hermitian* property.
- The *fundamental frequency* is  $f_0 = 1/T$  corresponding to  $n = 1$ . All other frequency components are multiples of this frequency.
- The *amplitude spectrum*  $|X_n|$  of a real signal is an even function of frequency, whereas the *phase spectrum*  $\arg X_n$  is an odd function.

## Amplitude Spectrum



## Phase Spectrum

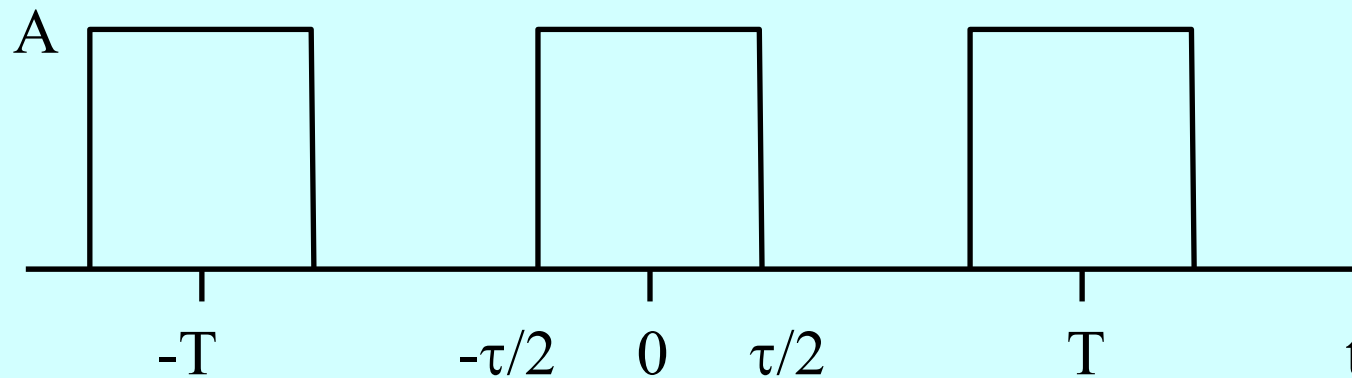


- The average **power** is obtained by averaging over one period.

$$\begin{aligned} P &= \frac{1}{T} \int_{(T)} |x(t)|^2 dt = \frac{1}{T} \int_{(T)} x(t) x^*(t) dt \\ &= \frac{1}{T} \int_{(T)} x(t) \left[ \sum_{n=-\infty}^{\infty} X_n^* e^{-jn2\pi t/T} \right] dt \\ &= \sum_{n=-\infty}^{\infty} X_n X_n^* = \sum_{n=-\infty}^{\infty} |X_n|^2 \end{aligned}$$

This is ***Parseval's Theorem***. Note that P is actually the mean square value.

**Example:** Rectangular pulse train.

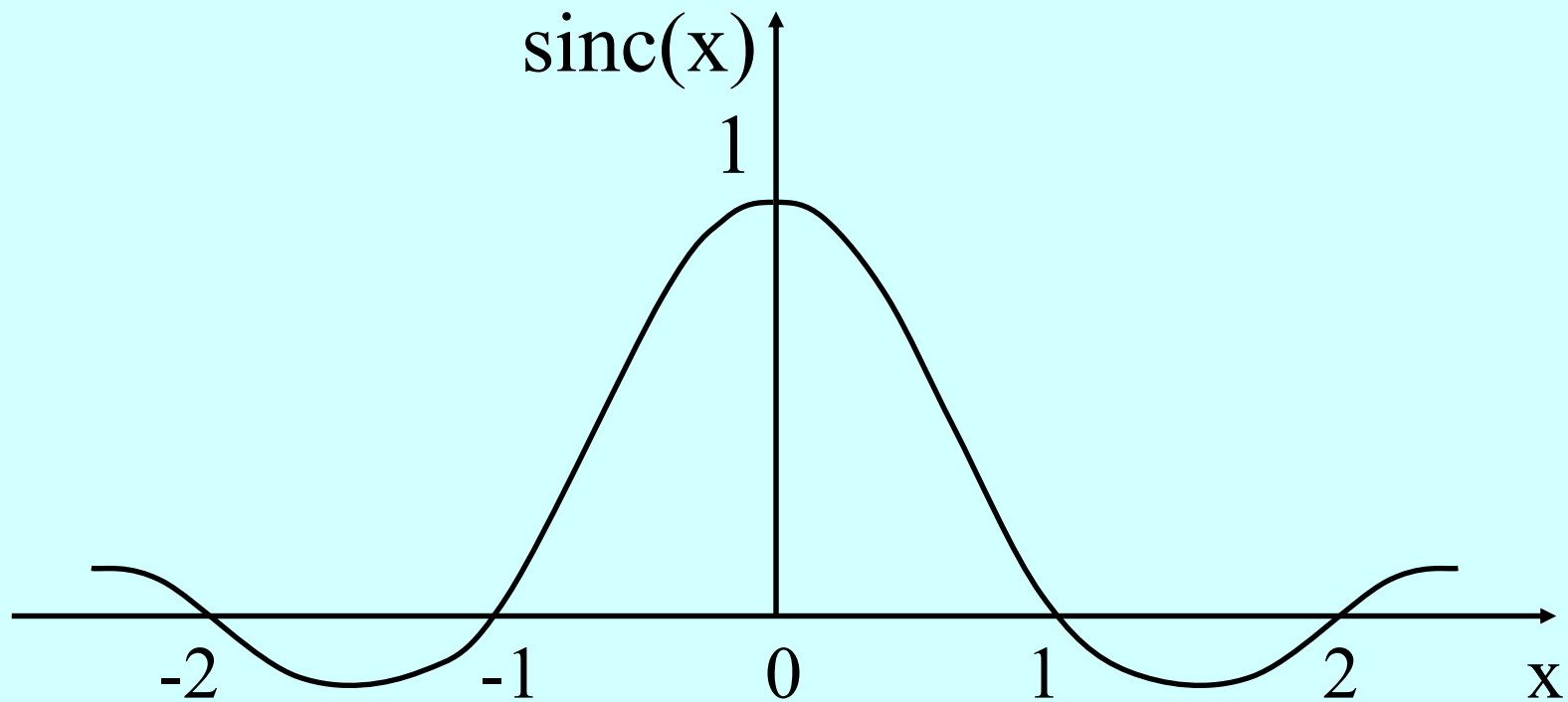


$$x(t) = \text{rep}_T \{A \text{rect}(t / \tau)\}$$

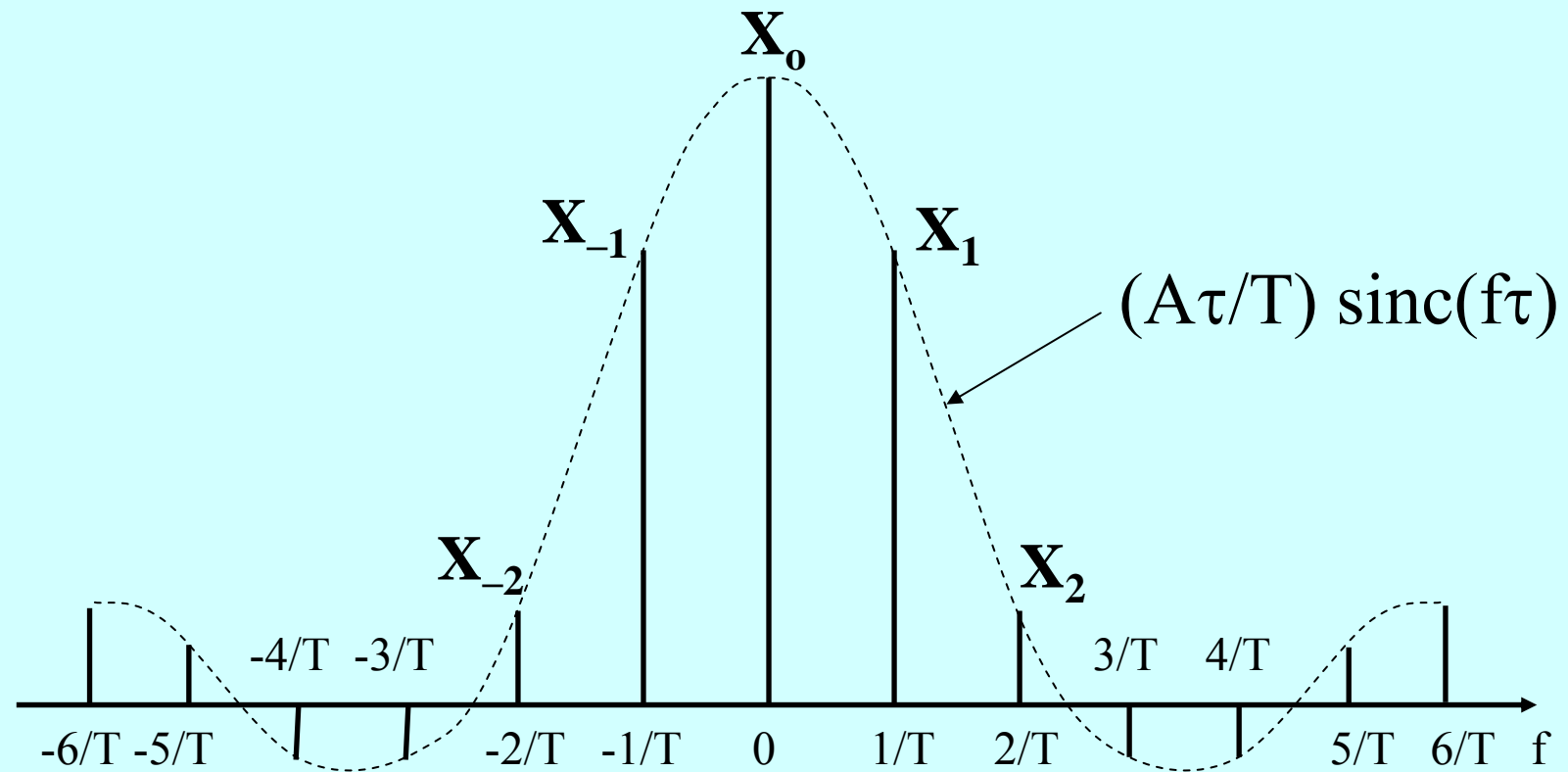
$$\begin{aligned} X_n &= \frac{1}{T} \int_{-\tau/2}^{\tau/2} A e^{-jn2\pi t/T} dt \\ &= \frac{A}{j2\pi n} \left( e^{jn\pi\tau/T} - e^{-jn\pi\tau/T} \right) \\ &= \frac{A\tau}{T} \frac{\sin(n\pi\tau/T)}{n\pi\tau/T} \\ &= \frac{A\tau}{T} \text{sinc}(n\tau/T) \end{aligned}$$



where  $\text{sinc}(x) = \sin(\pi x)/(\pi x)$ .



The Fourier series spectrum is:



## 2.2 Fourier Transforms

The *Fourier transform* is an extension of Fourier series to non-periodic signals.

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

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

We will use the notation  $x(t) \leftrightarrow X(f)$ . If  $x(t)$  is in volts, the dimensions of  $X(f)$  are volt-sec or V/Hz.

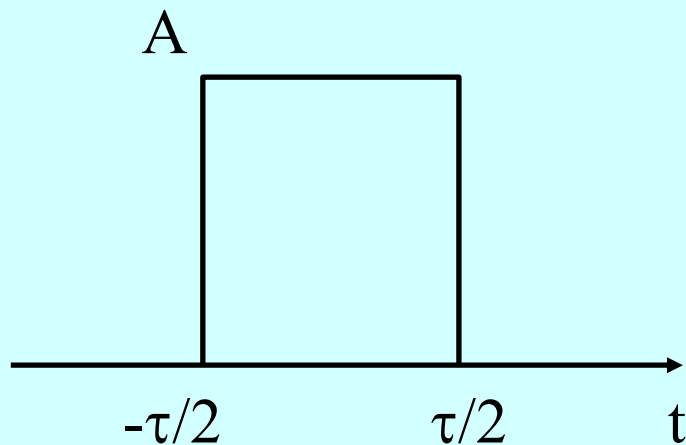
Fourier transforms exist for signals of *finite energy*.

$$\begin{aligned} E &= \int_{-\infty}^{\infty} |x(t)|^2 dt = \int_{-\infty}^{\infty} x(t) x^*(t) dt \\ &= \int_{-\infty}^{\infty} x(t) \left\{ \int_{-\infty}^{\infty} X^*(f) e^{-j2\pi f t} df \right\} dt \\ &= \int_{-\infty}^{\infty} X(f) X^*(f) df = \int_{-\infty}^{\infty} |X(f)|^2 df \end{aligned}$$

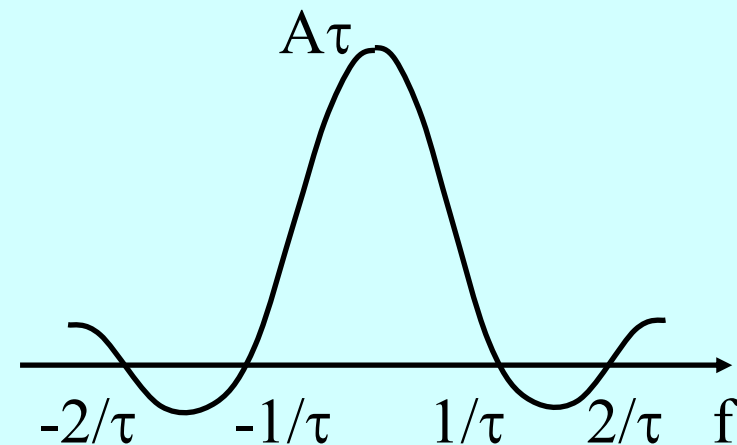
This is *Rayleigh's Energy Theorem*.

Note that  $G_{xx}(f) = |X(f)|^2$  is also called the *energy spectral density* of the finite energy signal.

**Example:** Rectangular pulse

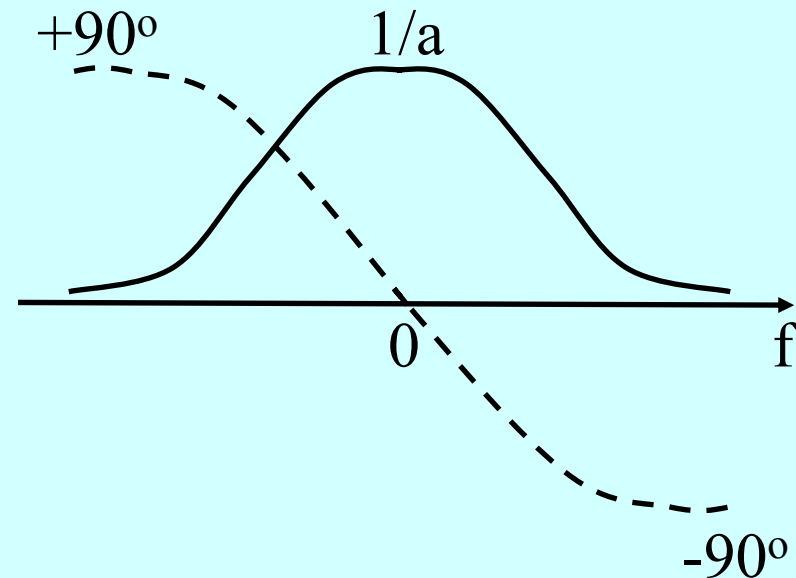
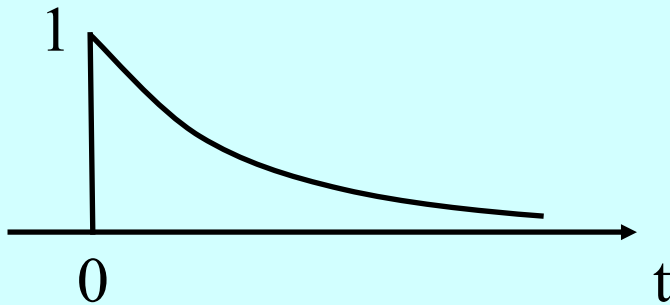


$$x(t) = A \operatorname{rect}(t / \tau)$$



$$X(f) = A\tau \operatorname{sinc}(f\tau)$$

## Example: Exponential pulse



$$x(t) = u(t)e^{-at}$$

$$X(f) = \frac{1}{a + j2\pi f}$$

Make sure that you know how to deal with:

- Time scaling
- Frequency scaling
- Time shifts
- Frequency shifts
- Time inversion
- Differentiation
- Integration
- Multiplication by  $t$
- See the Fourier transform sheet provided

For periodic signals we have:

$$x(t) = \sum_{k=-\infty}^{\infty} g(t - kT) = \text{rep}_T \{g(t)\} = \sum_{n=-\infty}^{\infty} X_n e^{jn2\pi t/T}$$

$$X(f) = \sum_{n=-\infty}^{\infty} X_n \delta(f - n/T)$$

$$X_n = \frac{1}{T} \int_{(T)} x(t) e^{-j2\pi n t/T} dt = \frac{1}{T} G(n/T)$$

$$G(f) = \int_{-\infty}^{\infty} g(t) e^{-j2\pi f t} dt$$

**Exercise**: Prove that  $X_n = (1/T) G(n/T)$ .



## 2.3 Convolution

If we have  $V(f) = X(f)Y(f)$ , what is  $v(t)$ ?

$$\begin{aligned} v(t) &= \int_{-\infty}^{\infty} X(f) Y(f) e^{j2\pi ft} df \\ &= \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} x(\lambda) e^{-j2\pi f\lambda} d\lambda Y(f) e^{j2\pi ft} df \\ &= \int_{-\infty}^{\infty} x(\lambda) y(t - \lambda) d\lambda = x(t) \otimes y(t) \end{aligned}$$

This is called the *convolution* of  $x(t)$  and  $y(t)$ .  
We can also define convolution in the frequency domain.

$$x(t) y(t) \leftrightarrow X(f) \otimes Y(f) = \int_{-\infty}^{\infty} X(\lambda) Y(f - \lambda) d\lambda$$

$$X(f) Y(f) \leftrightarrow x(t) \otimes y(t) = \int_{-\infty}^{\infty} x(\lambda) y(f - \lambda) d\lambda$$

**Exercise**: Prove  $\int_{-\infty}^{\infty} x(\lambda) x^*(\lambda - t) d\lambda \leftrightarrow |X(f)|^2 = G_{xx}(f)$

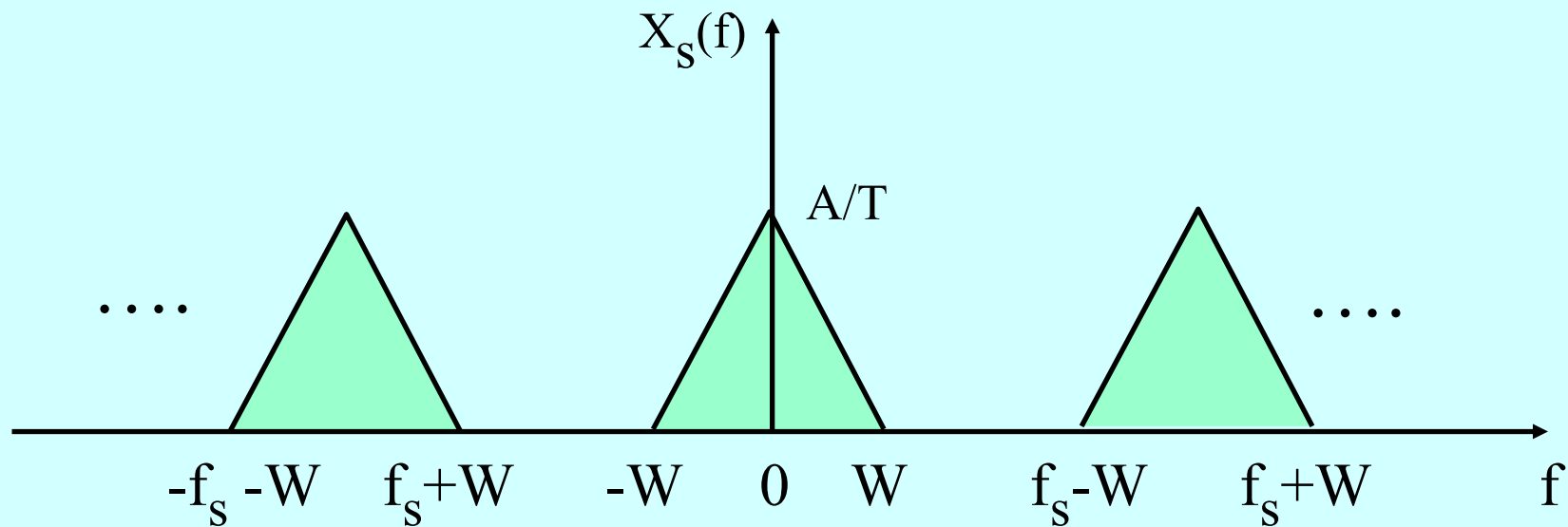
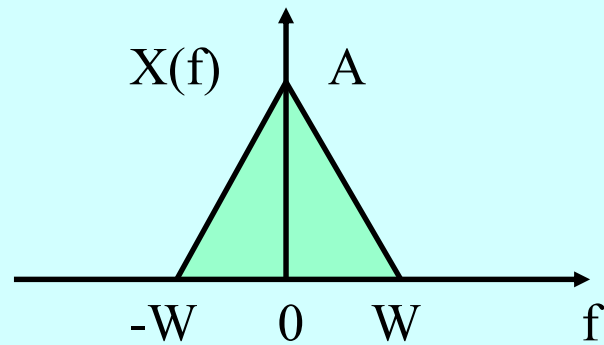
## **2.4 The Sampling Theorem**

If a signal is sampled at intervals of  $T = 1/f_s$  we have:

$$x_s(t) = \sum_{n=-\infty}^{\infty} x(nT) \delta(t - nT) = x(t) \text{comb}_T(t)$$

$$X_s(f) = \frac{1}{T} \sum_{k=-\infty}^{\infty} X(f - k/T) = \frac{1}{T} \text{rep}_{1/T} \{X(f)\}$$

We can recover  $x(t)$  from  $x_s(t)$  by low pass filtering if the bandwidth of  $x(t)$  is less than  $1/2T$  (ie. less than one half the sampling rate).

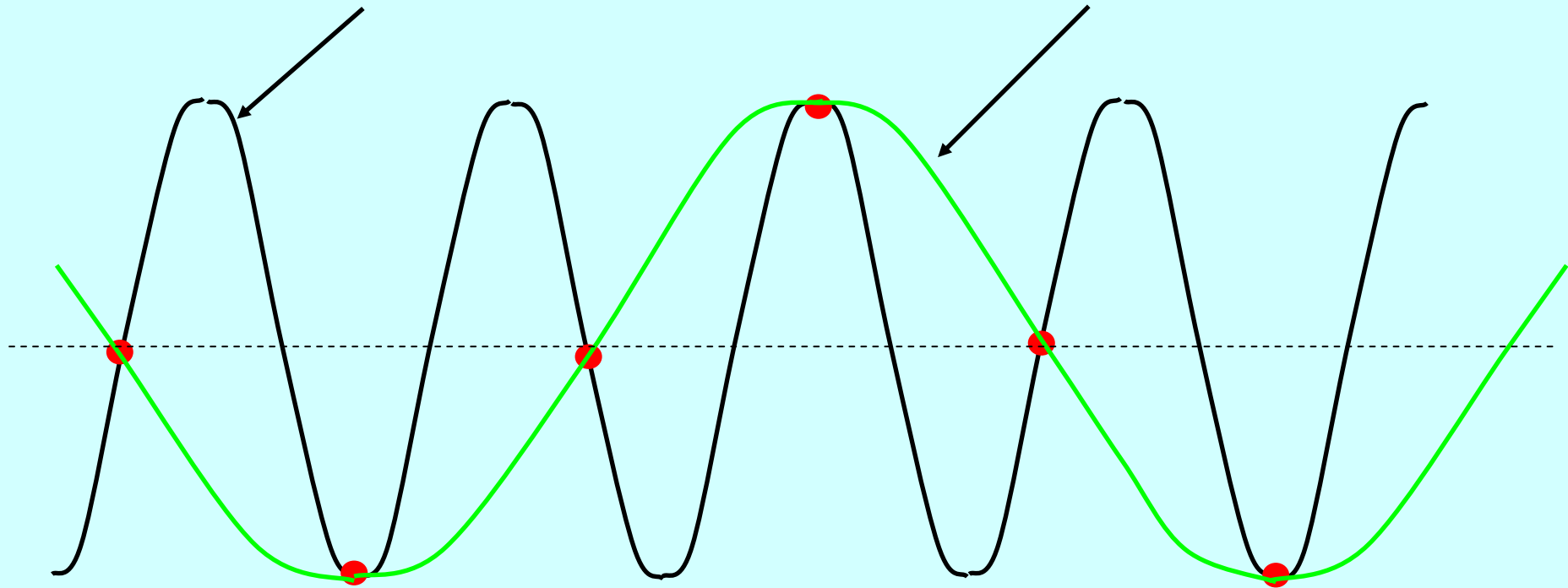


If we sample at lower than the required rate of  $f_s = 2W$ , we get *aliasing*. This is where signals at frequencies greater than the *Nyquist frequency*  $f_s/2$ , reappear at frequencies mirror imaged about  $f_s/2$ . For instance if the sampling rate is 10 kHz, a 6.5 kHz signal will appear as a 3.5 kHz signal when we try to recover the signal.

To avoid this, frequencies higher than  $f_s/2$  must be removed by an anti-aliasing filter before sampling.

Frequency 7500 Hz

Frequency 2500 Hz



Sampling frequency 10000 Hz

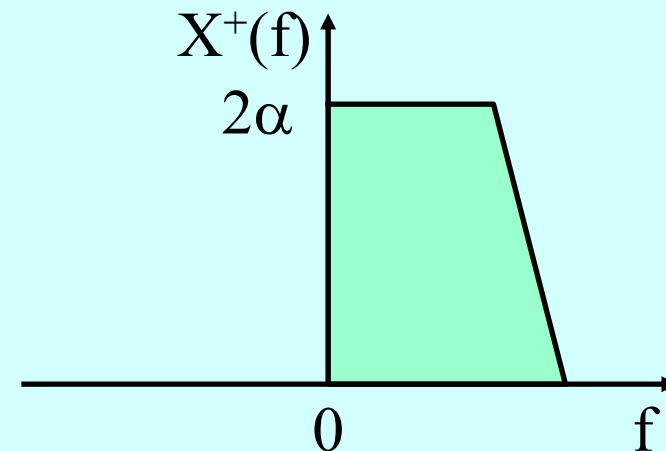
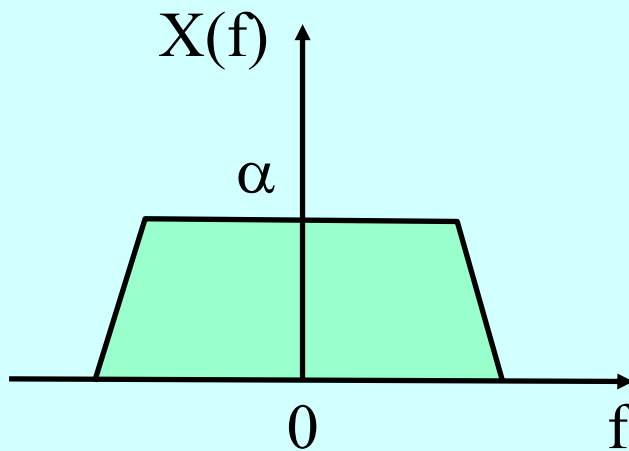
## 2.5 The Analytic Signal

With a real signal  $x(t)$  we have  $X(-f) = X^*(f)$ , so the negative frequency part is redundant. As for sinewaves, it is convenient to deal with signals which contain only positive frequencies.

$$\begin{array}{ccc} A \cos(2\pi f_0 t + \theta) & = & \operatorname{Re}\{A e^{j(2\pi f_0 t + \theta)}\} \\ \text{(Real signal)} & & \text{(Analytic signal)} \end{array}$$

The analytic signal is also called the ***pre-envelope***.

To obtain the analytic signal, we simply discard the negative frequency part and double the positive frequency part. For a real signal  $x(t)$ , we designate the analytic signal as  $x^+(t)$ .



$$X^+(f) = 2u(f)X(f)$$



## Example:

$$x(t) = \frac{a}{a^2 + t^2}$$

$$X(f) = \pi e^{-2\pi|f|a}$$

$$X^+(f) = 2\pi u(f) e^{-2\pi f a}$$

$$x^+(t) = \frac{1}{a - jt} = \left\{ \frac{a}{a^2 + t^2} \right\} + j \left\{ \frac{t}{a^2 + t^2} \right\}$$

$$X^+(f) = 2\pi u(f) e^{-2\pi f a}$$

$$u(t) e^{-at} \leftrightarrow \frac{1}{a + j2\pi f}$$

$$u(-f) e^{af} \leftrightarrow \frac{1}{a + j2\pi t} \quad \text{using} \quad X(t) \leftrightarrow x(-f)$$

$$u(f) e^{-af} \leftrightarrow \frac{1}{a - j2\pi t} \quad \text{using} \quad x(-t) \leftrightarrow X(-f)$$

$$2\pi u(f) e^{-2\pi f a} \leftrightarrow \frac{2\pi}{2\pi a - j2\pi t} = \frac{1}{a - jt} = x^+(t)$$

The real part of the analytic signal is the original signal, the imaginary part is called the *Hilbert transform* of  $x(t)$  and is denoted  $\hat{x}(t)$  .

$$Ae^{j(2\pi f_o t + \theta)} = A \cos(2\pi f_o t + \theta) + jA \sin(2\pi f_o t + \theta)$$

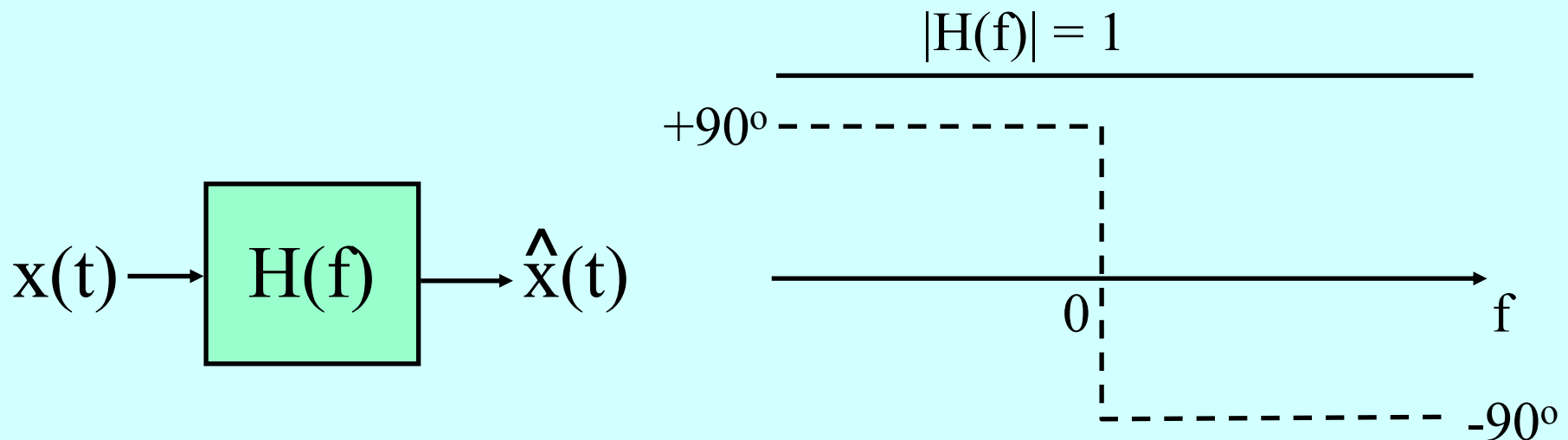
$$x(t) = A \cos(2\pi f_o t + \theta) = \frac{1}{2} Ae^{j(2\pi f_o t + \theta)} + \frac{1}{2} Ae^{-j(2\pi f_o t + \theta)}$$

$$\hat{x}(t) = A \sin(2\pi f_o t + \theta) = -\frac{j}{2} Ae^{j(2\pi f_o t + \theta)} + \frac{j}{2} Ae^{-j(2\pi f_o t + \theta)}$$

$$\hat{X}(f) = -j \operatorname{sgn}(f) X(f)$$

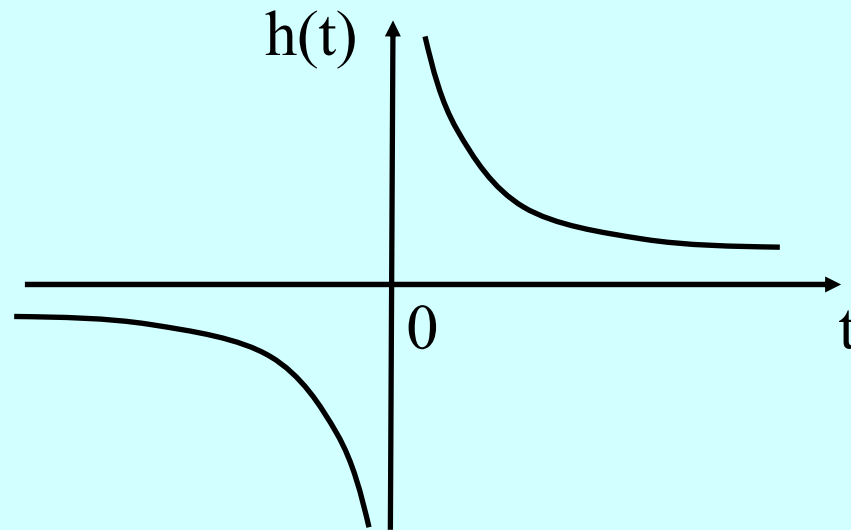
We recognise this as a filtering operation with a filter:

$$H(f) = -j \operatorname{sgn}(f).$$



The impulse response of the Hilbert transformer is:

$$h(t) = \frac{1}{\pi t} \quad \Rightarrow \quad \hat{x}(t) = \frac{1}{\pi} \int_{-\infty}^{\infty} \frac{x(\lambda)}{t - \lambda} d\lambda$$



The Hilbert transformer is *non-causal*.  
*Bandlimiting* removes the infinity at  $t = 0$ .

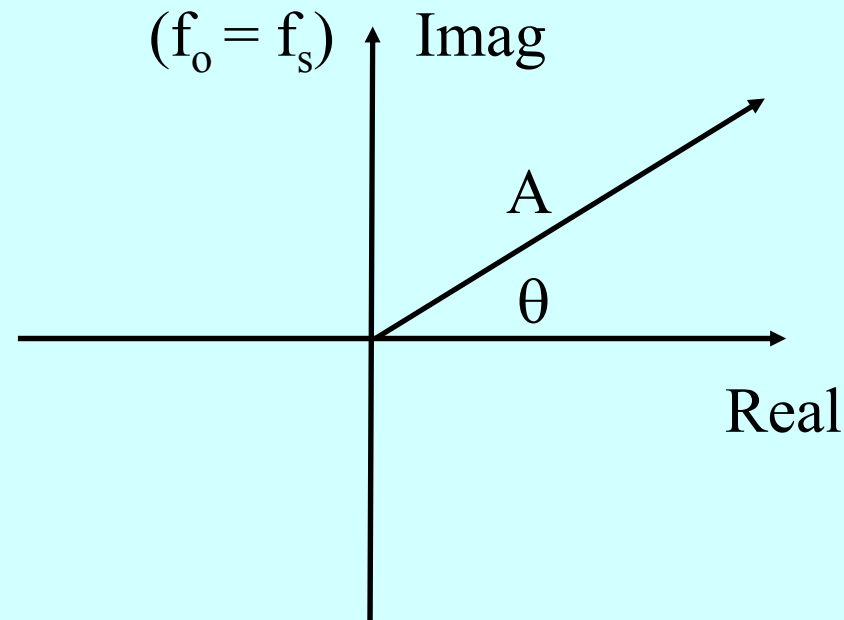
## 2.6 Applications of the Analytic Signal

### (i) Phasors

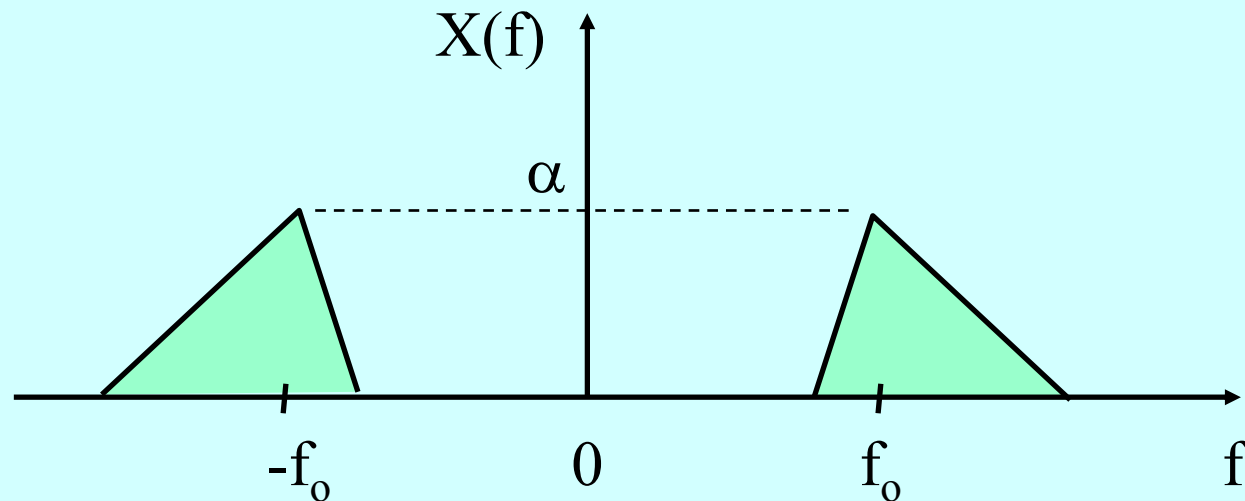
With a sinusoid  $x(t) = A \cos(2\pi f_0 t + \theta)$  the analytic signal is  $x^+(t) = A e^{j(2\pi f_0 t + \theta)}$ . If we factor out the  $e^{j2\pi f_0 t}$  term, the remaining factor  $Ae^{j\theta}$  is the *phasor* representing  $x(t)$ .

An important aspect of the phasor is the reference frequency  $f_0$  (specified separately).

The reference frequency  $f_o$  is arbitrary, but for a sinewave  $A \cos(2\pi f_s t + \theta)$  we usually choose it as the frequency of the sinewave, since then the phasor is a constant.



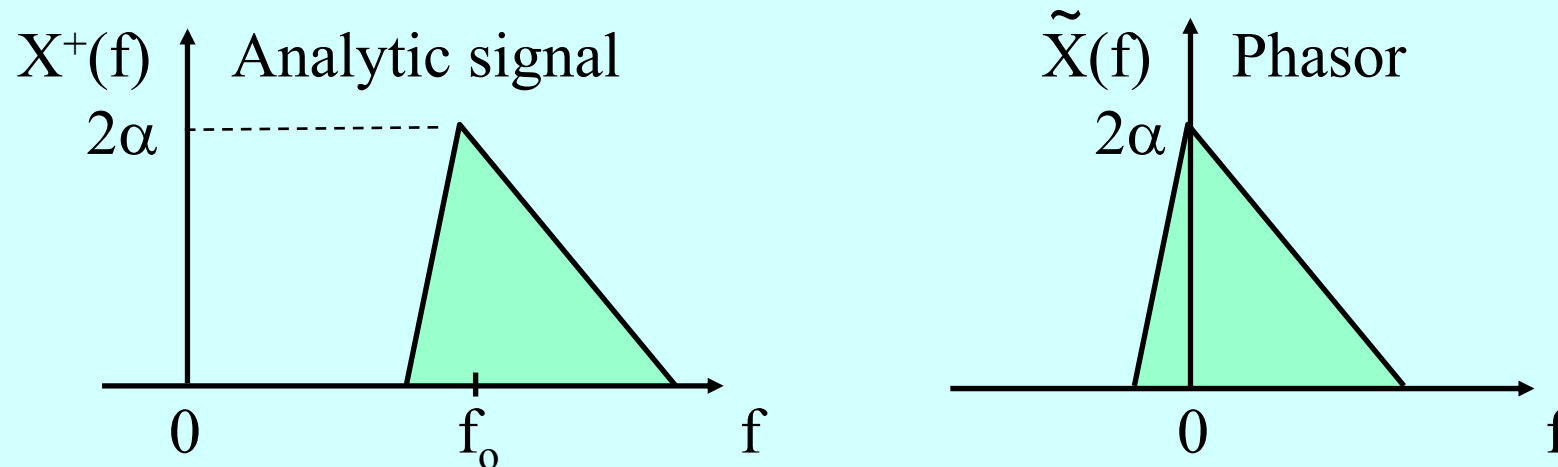
Phasors are of most use when we consider *narrowband* signals. These are signals which have their frequency components concentrated near some frequency  $f_o$ .





The phasor is denoted  $\tilde{x}(t)$  and is simply a frequency down-shifted version of the analytic signal. It contains all the information about  $x(t)$  **except** the carrier frequency  $f_o$ .

$$\tilde{x}(t) = x_c(t) + jx_s(t) = x^+(t)e^{-j2\pi f_o t}$$



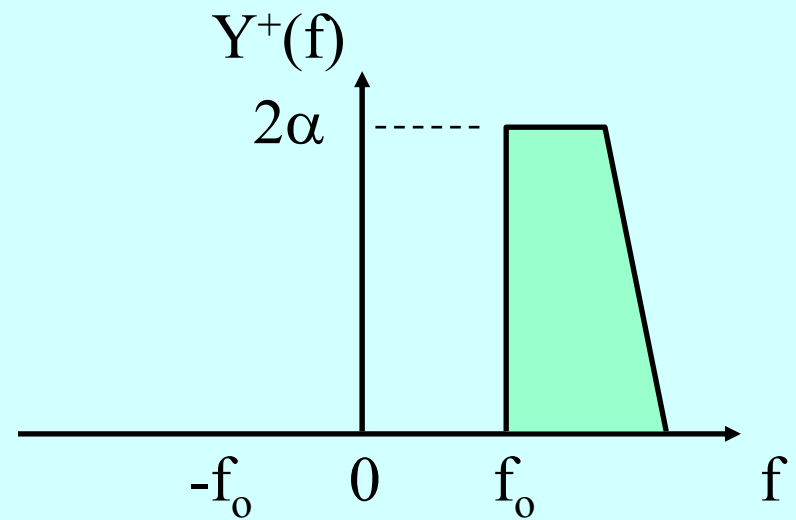
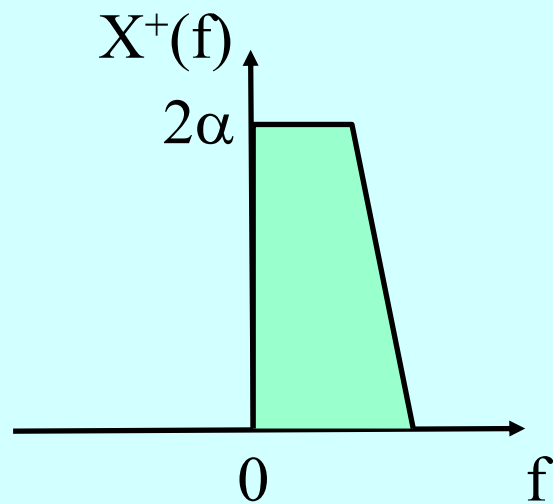
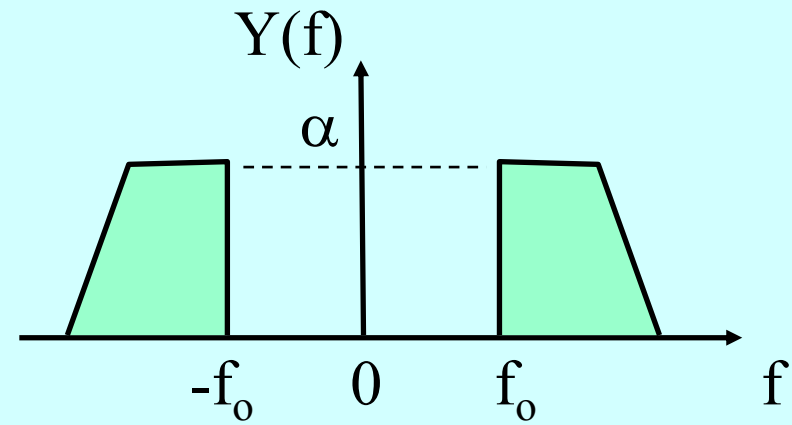
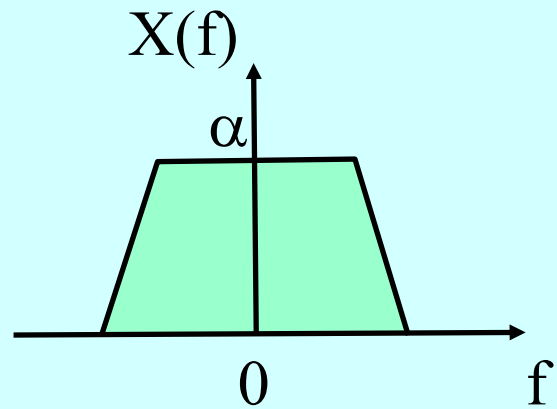
$$\begin{aligned}\tilde{x}(t) &= x^+(t)e^{-j2\pi f_o t} \\ &= x_c(t) + jx_s(t) = r(t)e^{j\theta(t)} \\ x(t) &= \text{Re} \{ x^+(t) \} = \text{Re} \{ \tilde{x}(t)e^{j2\pi f_o t} \} \\ &= x_c(t)\cos(2\pi f_o t) - x_s(t)\sin(2\pi f_o t) \\ &= r(t)\cos\{2\pi f_o t + \theta(t)\}\end{aligned}$$

- **Envelope**  $r(t) = |\tilde{x}(t)|$
- **Relative phase**  $\theta(t) = \arg \tilde{x}(t)$
- **Frequency deviation**  $= d\theta(t)/dt$  ; rad/sec

## **(ii) Single Sideband Signals**

A single sideband (SSB) signal is one in which the positive frequency components of a baseband signal  $x(t)$  are translated up by  $f_o$  and the negative frequency components down by  $f_o$ .

To determine an expression describing how the SSB signal  $y(t)$  is related to the baseband signal  $x(t)$ , we first consider the relation between the respective analytic signals.



$$Y^+(f) = X^+(f - f_o)$$

$$y^+(t) = x^+(t) e^{j2\pi f_o t}$$

$$y(t) + j\hat{y}(t) = \{x(t) + j\hat{x}(t)\} e^{j2\pi f_o t}$$

$$y(t) = x(t) \cos(2\pi f_o t) - \hat{x}(t) \sin(2\pi f_o t)$$

**Exercise:** For  $x(t) = 4 \text{ sinc}(2t)$  and  $f_o = 10 \text{ Hz}$ , calculate the spectrum of  $\hat{x}(t)$  and hence that of  $y(t)$  from the expression above. Sketch the spectra obtained in each case.

**Exercises**: You are expected to attempt the following exercises in Proakis & Salehi. Completion of these exercises is part of the course. Solutions will be available later.

**2.23**

**2.24**

**2.25**

**2.28**

**2.49**

**2.55**

**2.58 ( $f = 0$  should be  $f > 0$ )**