EEE 4107 Signals and Communication I.

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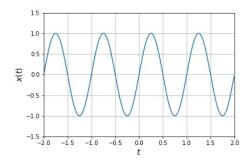
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Today's Lecture

1. Fourier Transform

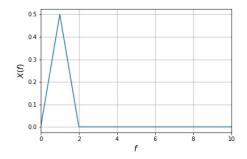
Introduction

- The Fourier transform provides the link between the time domain and frequency domain
- Consider the following sinusoid. What is its frequency?



Introduction

- The Fourier transform provides the link between the time domain and frequency domain
- The same signal can be viewed in the frequency domain



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Fourier Transform

► For a nonperiodic signal x(t), its Fourier transform is defined as

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

where the variable f denotes frequency.

• The time domain signal can be recovered from X(f) as

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

x(t) and X(f) constitute a Fourier Transform pair

Fourier Transform

We can use the notation

$$X(f) = \mathcal{F}[x(t)]$$

and

$$x(t) = \mathcal{F}^{-1}[X(f)]$$

Fourier Transform

- The Fourier transform allows us to express nonperiodic signals as sums of complex exponentials
- Recall

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

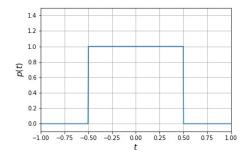
We have

$$\lim_{\Delta f \to 0} \sum_{n = -\infty}^{\infty} X(n\Delta f) e^{j2\pi n\Delta ft} \Delta f = \int_{-\infty}^{\infty} X(f) e^{j2\pi ft} df$$

► Thus we can view the signal x(t) as consisting of complex exponentials at frequency n∆f with each component weighted by X(n∆f)∆f

• Consider the pulse p(t) given by

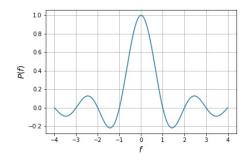
$$p(t) = \begin{cases} 1 & -\frac{1}{2} \le t \le \frac{1}{2} \\ 0 & \text{otherwise} \end{cases}$$



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- We see that the pulse consists of regions with no variation and two transitions at t = −1/2 and t = 1/2
- The regions with no variation correspond to low frequency content f = 0 while the sharp transitions correspond to high frequency content
- We have

$$P(f) = \int_{-\infty}^{\infty} p(t)e^{-j2\pi ft} dt = \int_{-1/2}^{1/2} e^{-j2\pi ft} dt$$
$$= \frac{e^{j\pi f} - e^{-j\pi f}}{j2\pi f} = \frac{\sin(\pi f)}{\pi f}$$
$$= \operatorname{sinc}(f)$$



- Note that sinc(0) = 1
- We see that most of the energy of the signal is located around f = 0 as expected

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- Consider the Dirac delta function $\delta(t)$.
- Recall the sifting property.

$$\int_{-\infty}^{\infty} x(t) \delta(t-t_0) dt = x(t_0)$$

Let's compute its Fourier transform

$$\mathcal{F}[\delta(t)] = \int_{-\infty}^{\infty} \delta(t) e^{-j2\pi f t} dt = e^{-j2\pi f t} \Big|_{t=0} = 1$$

 The Dirac delta function has equal energy over the entire frequency spectrum

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Linearity

$$\mathcal{F}[a_1x_1(t) + a_2x_2(t)] = a_1X_1(f) + a_2X_2(f)$$

Time Shifting

$$\mathcal{F}[x(t-t_0)] = X(f)e^{-j2\pi f t_0}$$

$$\mathcal{F}[x(t-t_0)] = \int_{-\infty}^{\infty} x(t-t_0) e^{-j2\pi f t} dt$$

$$= \int_{-\infty}^{\infty} x(t') e^{-j2\pi f(t'+t_0)} dt' \quad t-t_0 = t'$$

$$= e^{-j2\pi f t_0} \int_{-\infty}^{\infty} x(t') e^{-j2\pi f t'} dt'$$

$$= e^{-j2\pi f t_0} X(f)$$

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Frequency Shifting

$$\mathcal{F}[x(t)e^{j2\pi f_c t}] = X(f - f_c)$$

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

Duality. If

$$\mathcal{F}[x(t)] = X(f)$$

Then

$$\mathcal{F}[X(t)] = x(-f)$$

- Example: Since $\mathcal{F}[\delta(t)] = 1$, then $\mathcal{F}[1] = \delta(-f) = \delta(f)$
- Thus all the energy of a DC signal is located at f = 0

• Determine the Fourier transform of $e^{j2\pi f_c t}$

► Recall $\mathcal{F}[x(t)e^{j2\pi f_c t}] = X(f - f_c)$ In this case x(t) = 1 and $\mathcal{F}[x(t)] = \delta(f)$. Thus $\mathcal{F}[e^{j2\pi f_c t}] = \delta(f - f_c)$

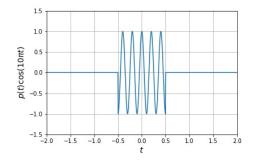
Similarly by exploiting linearity we can show that

$$\mathcal{F}[\cos(2\pi f_c t)] = \frac{1}{2} \Big(\delta(f - f_c) + \delta(f + f_c) \Big)$$

• Recall the pulse p(t) given by

$$p(t) = \begin{cases} 1 & -\frac{1}{2} \le t \le \frac{1}{2} \\ 0 & \text{otherwise} \end{cases}$$

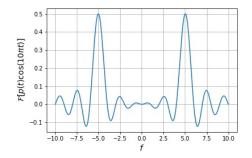
• Determine the Fourier transform of $p(t) \cos(10\pi t)$



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► We have
$$P(f) = \operatorname{sinc}(f)$$
. Thus
 $\mathcal{F}[p(t)\cos(2\pi f_c t)] = \frac{1}{2} \Big(P(f - f_c) + P(f + f_c) \Big)$

with $f_c = 5$



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Convolution in time.

$$\mathcal{F}\Big[\int_{-\infty}^{\infty}h(\tau)x(t-\tau)d\tau\Big]=H(f)X(f)$$

Recall that the output y(t) of an LTI system in response to x(t) is given by the convolution integral

$$y(t) = \int_{-\infty}^{\infty} h(\tau) x(t-\tau) d\tau$$

- We have Y(f) = H(f)X(f). H(f) is also known as the transfer function of the system as it relates the Fourier transform of the output to that of the input.
- $H(f) = \mathcal{F}[h(t)]$

- Consider an LTI system with impulse response $h(t) = e^{-t}u(t)$
- Show that

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

• In general, H(f) is complex and we can write

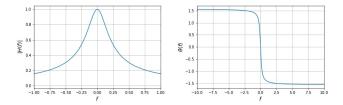
$$H(f) = |H(f)|e^{j\theta(f)}$$

where |H(f)| is known as the magnitude spectrum and $\theta(f)$ is known as the phase spectrum.

Show that

$$|H(f)| = rac{1}{\sqrt{1+4\pi^2 f^2}}$$

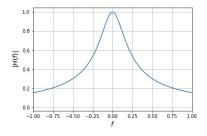
 $heta(f) = - an^{-1}(2\pi f)$



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Filter

- The linear time invariant system in example 5 is a low pass filter
- Filters are frequency selective systems
- Frequency components of the input signal that are present in the output lie within the passband of the filter.
- Frequency components that are present in the input but absent from the output lie in the stopband of the filter.



Filters

- The bandwidth of a filter is a number used to measure the extent of significant frequency content (in the positive frequency range) that is allowed to pass through the filter.
- An ideal lowpass filter has a frequency response given by

$$H_{LP}(f) = \left\{egin{array}{cc} 1 & |f| \leq W \ 0 & ext{otherwise} \end{array}
ight.$$

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The bandwidth of this filter is W.

An ideal bandpass filter has a frequency response given by

$$H_{BP}(f) = \begin{cases} 1 & W_1 \le |f| \le W_2 \\ 0 & \text{otherwise} \end{cases}$$

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The bandwidth of this filter is $W_2 - W_1$.

> An ideal highpass filter has a frequency response given by

$$H_{HP}(f) = \left\{ egin{array}{cc} 1 & |f| \geq W \ 0 & ext{otherwise} \end{array}
ight.$$

Filters

- ► For a non-ideal low pass filter with maximum magnitude response at f = 0, the 3dB bandwidth is the frequency at which the magnitude response is $|H(0)|/\sqrt{2}$.
- At this frequency the power transfer is half that at the origin.
- For the low pass filter with $h(t) = e^{-t}u(t)$ show that

$$f_{3dB} = \frac{1}{2\pi}$$

