Fourier Transforms! PHYS 250 (Autumn 2018) – Lecture 13

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Outline

- Reminders
 - Reminders from Lecture 12
- Follow-up with the Square Wave
 - Square Wave
- 3 Extension to the Fourier Transform
 - Euler's Formula
 - Fourier Transforms

Reminders from last time

We discussed both Poisson's equation and Fourier Series in the last lecture.

PDEs and Fourier Series

- PDEs: Poisson's Equation
 - We discussed and then wrote a function for computing the solution to Poisson's equation
 - Specifically, we elaborated on how we could structure the (relatively simple) function in order to take various constraints (i.e. sources) for the solution into account more easily
- Fourier Series:
 - Started discussing the basics of Fourier Series
 - Evaluated, computationally, the coefficients of a simple series for both a square and sawtooth wave

Today we will go much more in depth with Fourier Transforms and Analysis! This will be a mixture of Python Notebooks and Lecture Slides

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Square Wave

First, I want to do a little follow-up with the Fourier Series for the **square** wave function that we discussed last time.

Recall that we determined the coefficients of a sine and cosine expansion to be:

$$a_n = \frac{2}{n\pi} \sin\left(n\omega_0 \frac{\pi}{2}\right) \tag{1}$$

which yields a discrete function:

$$f(x) = \frac{a_0}{2} + \sum_{n=1}^{\infty} a_n \cos(nx)$$
 (2)

Now open up the Fourier-Transforms-Analysis.ipynb jupyter notebook so that we can look at this in more detail!

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Euler's Formula

We can generalize this by making use of Euler's Formula.

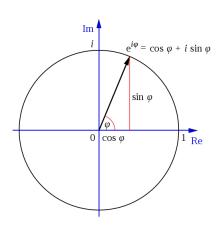
Euler's formula states that for any real number ϕ :

$$e^{i\phi} = \cos(\phi) + i\sin(\phi)$$
 (3)

When $\phi = \pi$, Euler's formula evaluates to

$$e^{i\pi} + 1 = 0, (4)$$

which is known as Euler's identity.



The implication is that it is possible to recover the amplitude of each wave in a Fourier series using an integral, which has many useful properties (in particular, that it's then continuous).

Fourier Transforms (I)

I will use the following definitions for the Fourier transform $\hat{f}(\xi)$ of a function f(x), where x typically represents either a **spatial or time domain**, and ξ typically represents a corresponding inverse notion of **spatial or time frequency**.

$$\hat{f}(\xi) = \int_{-\infty}^{\infty} f(x)e^{-2\pi i\xi x} dx \tag{5}$$

The **inverse transform** is then obtained via

$$f(x) = \int_{-\infty}^{\infty} \hat{f}(\xi)e^{2\pi i\xi x}d\xi \tag{6}$$

In the case of spatial coordinates, x denotes length and ξ denotes inverse wavelength: $\xi = \frac{1}{\lambda}$. In the time domain, x denotes time and ξ denotes frequency. In the case that x = t is in seconds, but ξ is **angular** frequency ω then a factor of 2π appears to get the normalization correct.

Fourier Transforms (II)

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$$\hat{f}(\omega) = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{\infty} f(t)e^{-i\omega t}dt$$

$$f(t) = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{\infty} \hat{f}(\omega)e^{i\omega t}d\omega$$
(8)

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Since
$$\omega = 2\pi \xi = \frac{2\pi}{\lambda}$$
.

The $\frac{1}{\sqrt{2\pi}}$ factor in both these integrals is a common normalization in

quantum mechanics but maybe not in engineering where only a single $\frac{1}{2\pi}$ factor is often used.

If $\hat{f}(\omega)$ or f(t) are known analytically or numerically, the Fourier transform integrals can be evaluated using the integration techniques studied earlier.

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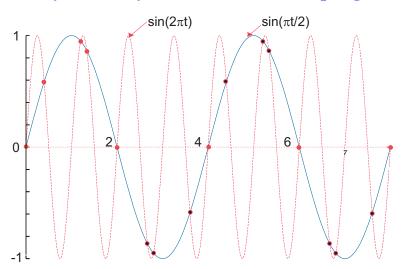
In this way the DFT can be thought of as a technique for interpolating, compressing, and extrapolating data.

Discussion

Do you see any issues with this "sampling"?

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The DFT algorithm results from evaluating the integral not from 1 to +1 but rather from time 0 to time T over which the signal is measured, and from approximating the integration of the integral by computing a discrete sum:

$$\hat{f}(\omega_n) = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{\infty} f(t)e^{-i\omega_n t} dt$$
 (9)

$$\simeq \frac{1}{\sqrt{2\pi}} \int_0^T f(t)e^{-i\omega_n t} dt \tag{10}$$

$$\simeq \frac{1}{\sqrt{2\pi}} \sum_{k=1}^{N} h f(t_k) e^{-i\omega_n t_k}$$
 $(h \equiv \text{stepsize})$ (11)

$$\simeq \frac{h}{\sqrt{2\pi}} \sum_{k=1}^{N} f_k e^{-2\pi i k n/N} \tag{12}$$

$$\hat{f}_n \equiv \frac{\hat{f}(\omega_n)}{h} = \frac{1}{\sqrt{2\pi}} \sum_{k=1}^{N} f_k e^{-2\pi i k n/N}$$
(13)

We then need the inverse as well, which we can obtain with $d\omega \to 2\pi/Nh$ we invert the \hat{f}_n

$$f_k = \frac{1}{\sqrt{2\pi}} \sum_{n=1}^{N} \frac{2\pi}{Nh} \hat{f}_n e^{-i\omega_n t}$$
 (14)

Once we know the N values of the transform \hat{f}_n , we can use this expression to evaluate f(t) for any time t. The frequencies ωn are determined by the number of samples taken and by the total sampling time T = Nh as

$$\omega_n = n \frac{2\pi}{Nh} \tag{15}$$

Clearly, the larger we make the time T=Nh over which we sample the function, the smaller will be the frequency steps or resolution. Accordingly, if you want a smooth frequency spectrum, you need to have a small frequency step $2\pi/T$.

Lastly, we can simplify this expression to yield a clear computational approach:

$$f_k = \frac{\sqrt{2\pi}}{N} \sum_{n=1}^{N} Z^{-nk} \hat{f}_n \qquad (Z = e^{-2\pi i/N})$$
 (16)

$$\hat{f}_n = \frac{1}{\sqrt{2\pi}} \sum_{k=1}^{N} Z^{nk} f_k \qquad (n = 0, 1, \dots, N)$$
 (17)

With this formulation, the computer needs to compute only powers of Z.