TIMESERIES: THE FINAL CHAPTER?

10.30.2020



HOMEWORK

- * Problem Set 4 was due today!
- * Problem Set 5 will be posted this evening, due in 3 weeks



RECAP

- * power spectrum / psd
- * spectrogram
- * filtering

FILTERING

- * **scipy.signal** is a module in scipy that contains lots of useful functions for filter design
- * scipy.signal.firwin creates "finite impulse response" filters with desired properties

ANALYZING A FILTER

- * scipy.signal.freqz is a great function that tells you what the *frequency* response of your filter looks like
 - * i.e. it tells you what the filter is going to do to your signal

RECALL: FOURIER ANALYSIS

- * fourier transforms have an interesting property related to convolution:
- * given two timeseries, f and g, the fourier transform of their convolution = the element-wise product of their fourier transforms

$$FT(f*g) = F \cdot G$$

* the reverse is also true:

$$F * G = FT(f \cdot g)$$

Time Function Sinc Boxcar $-\frac{\tau}{2}$ 0 $\frac{\tau}{2}$ $G(t) = \begin{cases} 1-|t|/\tau, |t| < \tau \\ 0, |t| > \tau \end{cases}$ Sinc² Triangle 0 $G(t) = e^{-1/2t^2}$ Gaussian Gaussian τ DC Shift Impulse $G(t) = \delta(t)$ = 0, t ≠ 0 0 Single Freq. $G(t) = \cos \omega_0 t$ Sinusoid π/ω 0 π/ω 2π/ω G(t) = comb(t)Comb. Comb. $= \sum_{n=0}^{\infty} \delta(1-n\tau)$ 0 τ 2τ 3τ -2π

Frequency Function

 $S(f) = \tau \operatorname{sinc}(f\tau)$

 $S(f) = \tau \operatorname{sinc}^2(ft)$

 $= (1/\pi f) \sin (\pi f t)$

 $= (1/\pi^2 f^2 \tau) \sin^2 (\pi f t)$

-1/_{\tau} 0 1/_{\tau} 2/_{\tau} 3/_{\tau} 4/_{\tau}

 $S(f) = \frac{1}{2}(\delta(f+f_0) + \delta(f-f_0))$

2π

 4π

 $S(f) = \tau(2\pi)^{1/2} e^{-(\pi f \tau)^2}$

-1/_T 0 1/_T

0

0

 $S(f) = \sum_{-\infty}^{\infty} \delta(f-n/\tau)$

S(f) = 1

 $-f_0$

- * all of the timeseries we work with are discrete or digital, meaning that they are made up of samples separated by some even spacing in time
 - * (note that **sample** is used in a different sense here than in statistics)

- * the number of samples taken per unit time is called the **sampling rate**
 - * e.g. in fMRI our sampling rate is typically 0.5 Hz (1 sample every 2 seconds)
 - * in electrophysiology it could be as high as 25 kHz (25,000 samples per second)

- * the sampling rate limits the frequencies that can be represented in a timeseries
- * the highest frequency that a timeseries can represent is called the **Nyquist** frequency, and it is exactly half the sampling rate

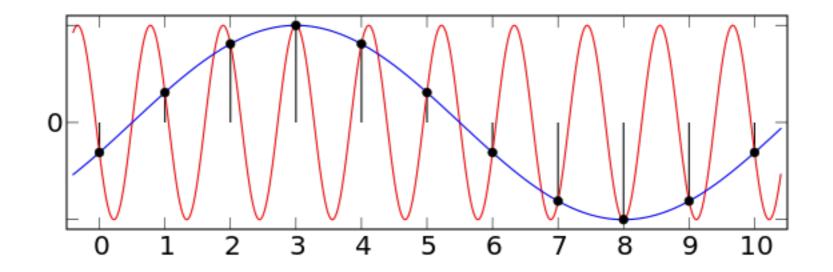


Harry Nyquist

* for example if our fMRI data is sampled at 0.5 Hz, then the Nyquist frequency is 0.25 Hz

* why is this? why can't higher frequency signals be represented?

- * the problem is that any frequency above Nyquist would appear identical to some frequency below Nyquist
- * this is called *aliasing*



SUBSAMPLING

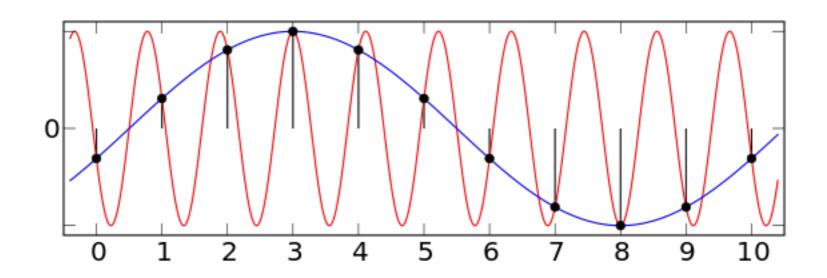
* suppose we have a 20 kHz timeseries and want to downsample it to 2 kHz

SUBSAMPLING

- * one idea: just take every 10th sample!
 - * (this is called **subsampling**)
- * taking every 10th sample is *LITERALLY THE* WORST IDEA
 - * (let's see an example)

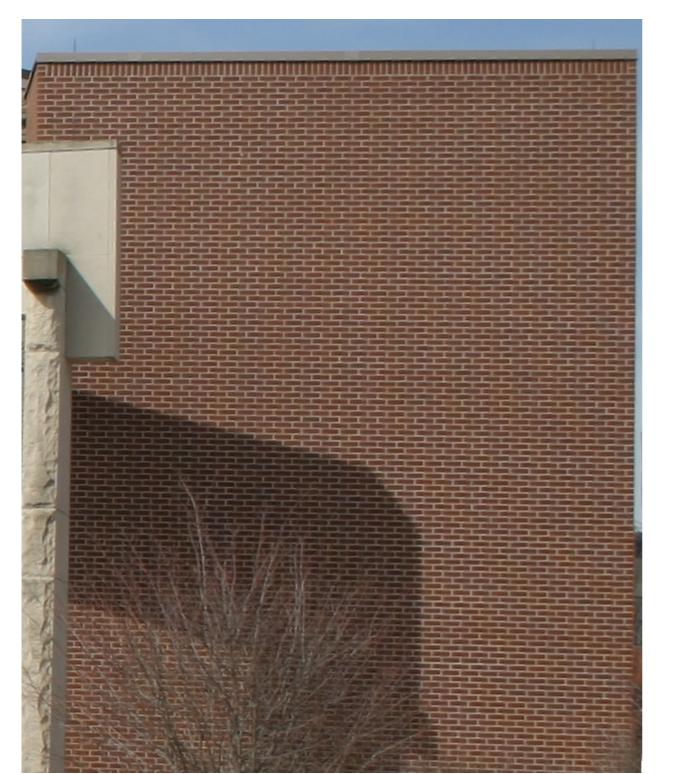
SUBSAMPLING

- * subsampling doesn't remove high frequencies, it turns them into low frequencies
- * this is also aliasing



ALIASING IN IMAGES

Original Image



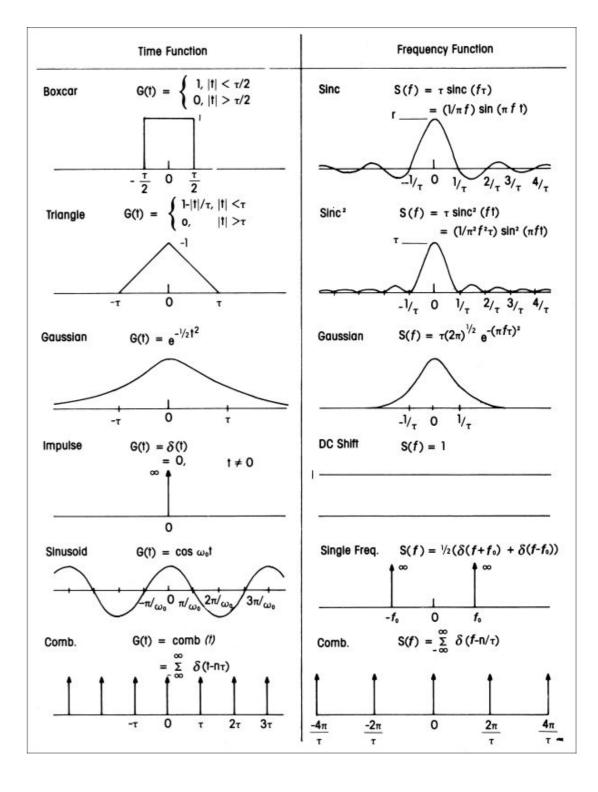
Subsampled



high frequency pattern (bricks) is aliased to low frequency "moiré pattern"

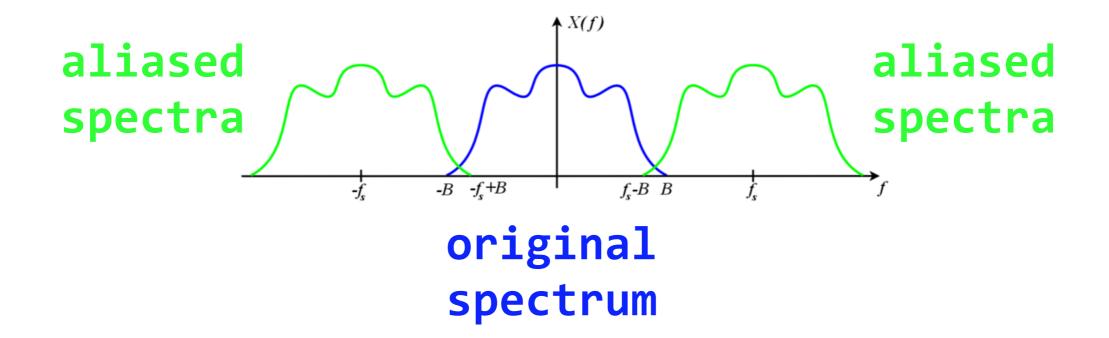
ALIASING

- * sampling is like
 multiplying your
 timeseries by a "comb"
 function
- * ... which is equivalent to convolving the fourier transform of your timeseries by a comb function



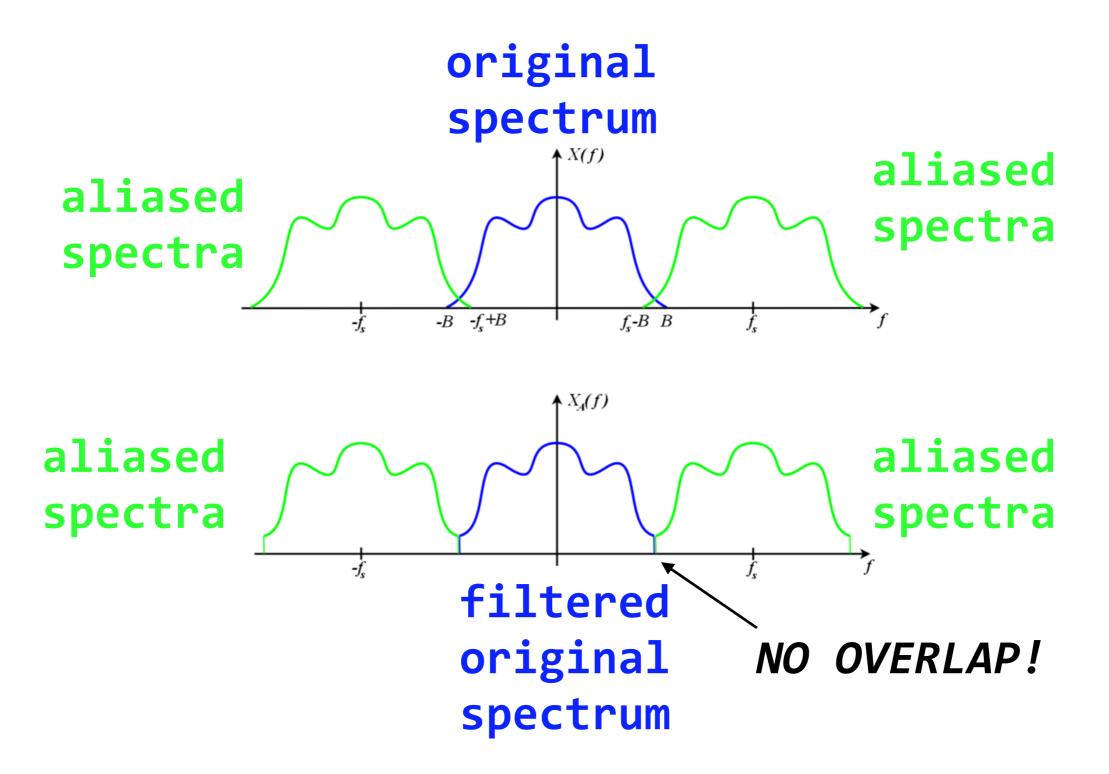
ALIASING

* which means that the fourier transform of the subsampled timeseries can have high frequencies "invading" lower frequencies



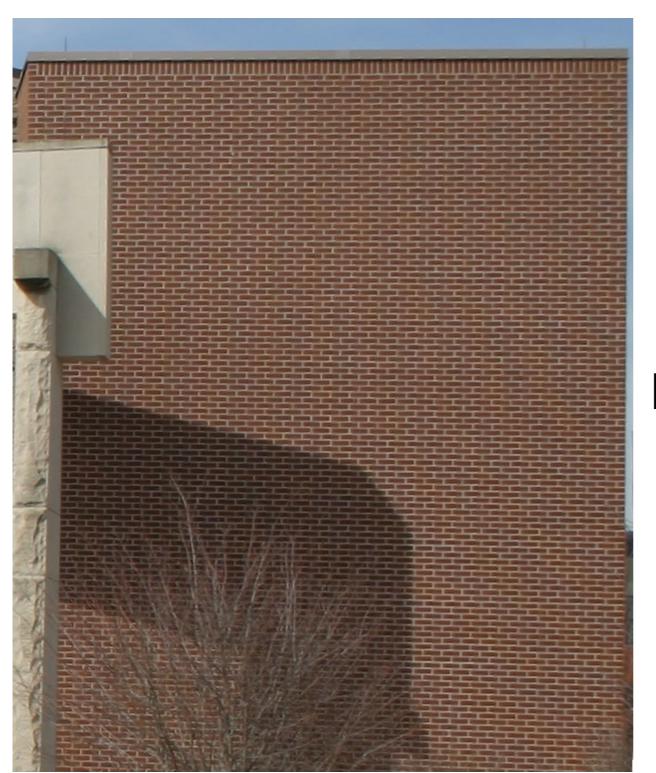
* how do we solve this?

- * we can use an antialiasing filter
- * e.g.: the original signal is sampled at 20 kHz, we want to downsample to 2 kHz
- * the new 2 kHz shouldn't contain any frequencies above Nyquist (1 kHz)
- * so we **low-pass filter** the original signal at 1 kHz, and then subsample



ANTIALIASING IN IMAGES

Original Image



Subsampled



Properly downsampled



- * there are functions in **scipy.signal** for doing good downsampling/resampling
 - * **signal.decimate** is great for downsampling
 - * **signal.resample** can do downsampling or upsampling

END