Intro to DSP

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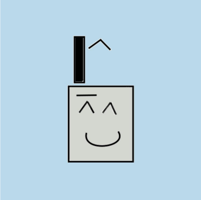
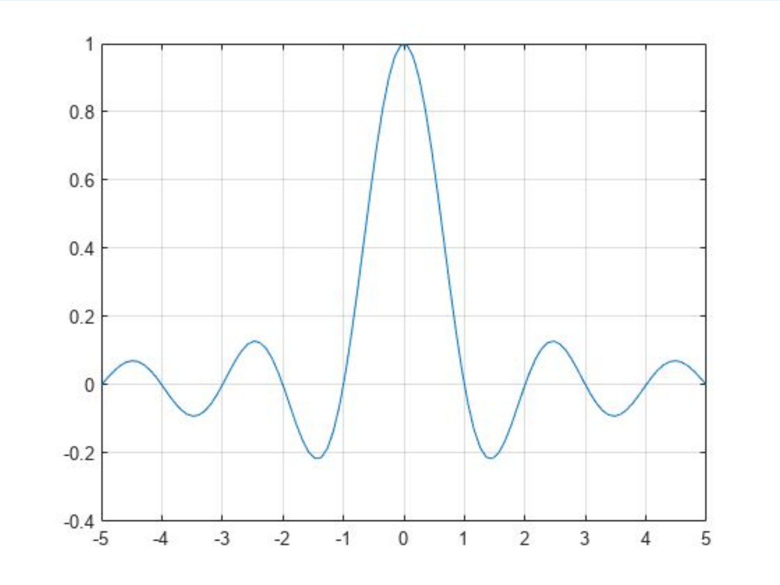
About Me

- BS and MS in Electrical Engineering, University of Rochester
- Ham since 2004, Extra since 2007
  - Callsign AA6XA, formerly KB1KXL
- Ham radio interests include
  - SOTA
  - VHF contesting
  - Homebrewing
  - Microwaves
  - HF CW contesting
- Social Media
  - ham_bitious (youtube.com/@hambitious and Instagram, Mastodon, etc.)
  - AA6XA on Instagram



Outline

- Why Digital Signal Processing?
  - DSP applications
- Signal Classification
  - Analog vs. Digital
  - Continuous Time vs. Discrete Time
- Sampling
- Fast Fourier Transform
- Filtering
- Putting it all together



Why DSP?

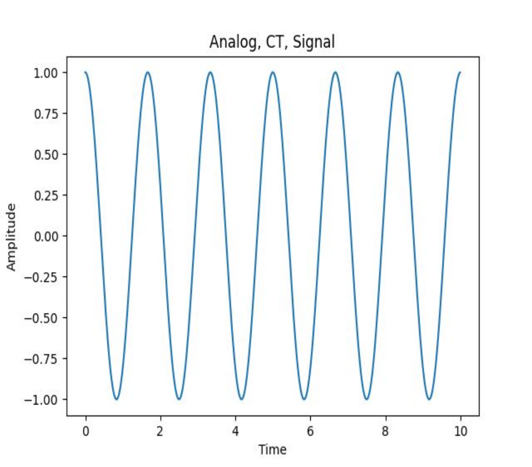
- Analog signal processing is hard
  - Need to worry about component tolerances, etc.
  - Actually have to build a circuit to see how it performs
  - More components needed for sharper filters
  - Redesign required to change parameters (frequency, filter shape, etc)
- Digital Signal Processing solves many of these problems
- DSP has many, many applications
  - Image processing
  - Music and audio processing
  - RADAR, SONAR
  - Telecommunications (Ham radio!)

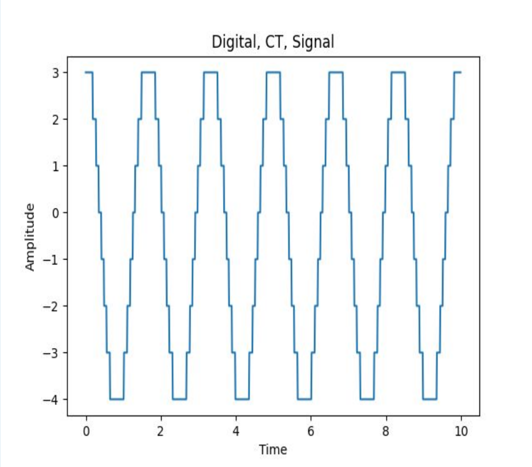
Ham Radio DSP Applications

- Waterfall display
- Filtering (e.g. notch filter)
- Noise reduction - QRM eliminator
- Voice compression for DV modes

Signal Classification: Analog and Digital

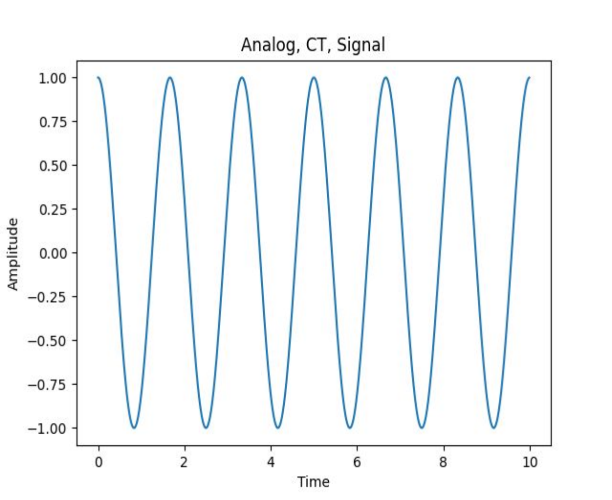
- Analog signals have continuously variable amplitudes
  - Signals found in nature
  - The amplitude range is y(t): [-A, A]
- Digital signals have amplitudes at discrete values
  - Quantized via ADC
  - For a signal with n bits, there are 2^n levels
  - The range is y(t): $[-2^{n-1}, 2^{n-1}-1]$

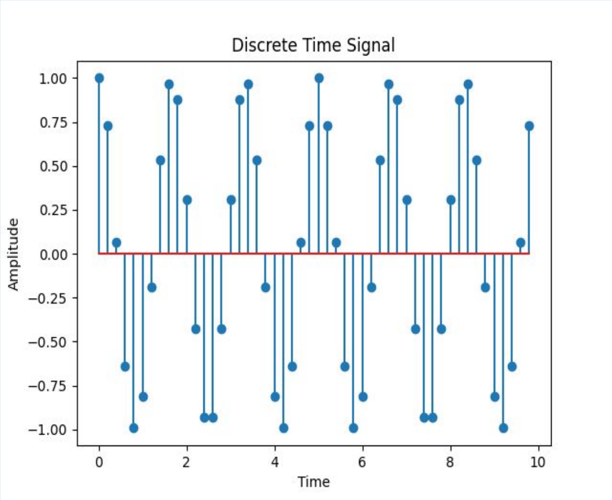




Signal Classification: Continuous Time and Discrete Time

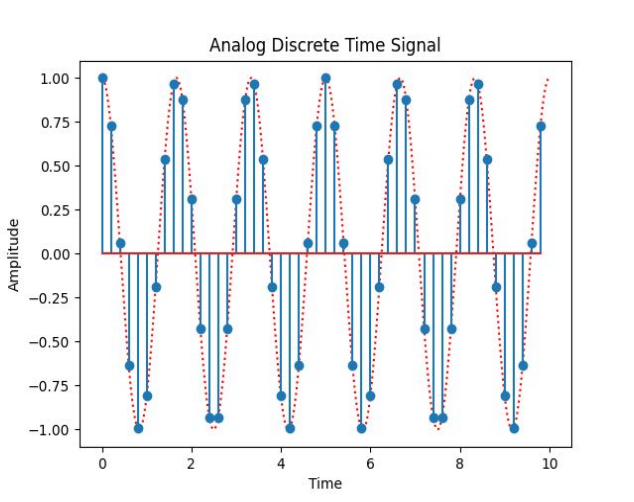
- Continuous Time (CT) signals have a value at every time
  - Generally denoted f(t)
- Discrete Time (DT) signals have values only at given times
  - Sampled at f_s, the sampling frequency
  - Denoted x[n]

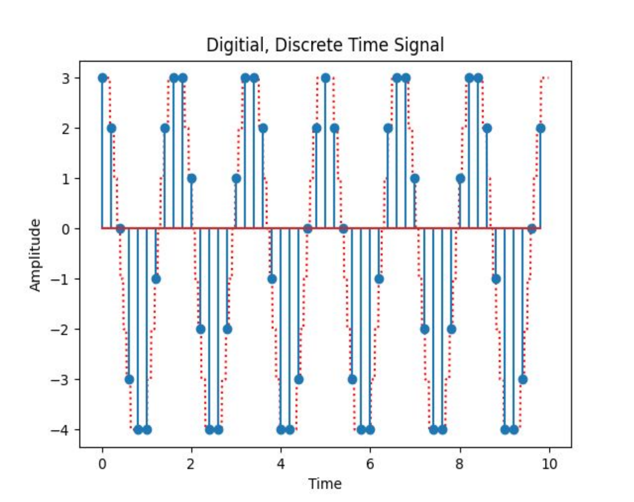




Signal Types Summary

- Signals can be Analog or Digital
- Signals can be Continuous Time (CT) or Discrete Time (DT)
- Any combination is possible, e.g. analog DT
- In the “real world”, signals are analog CT
- Computer processors can only deal with digital DT signals



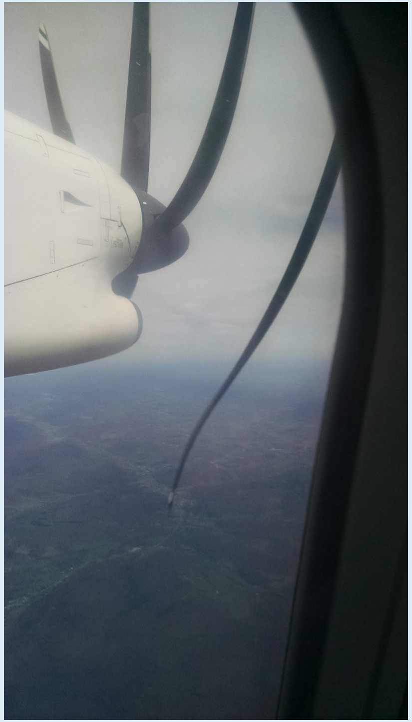


Sampling

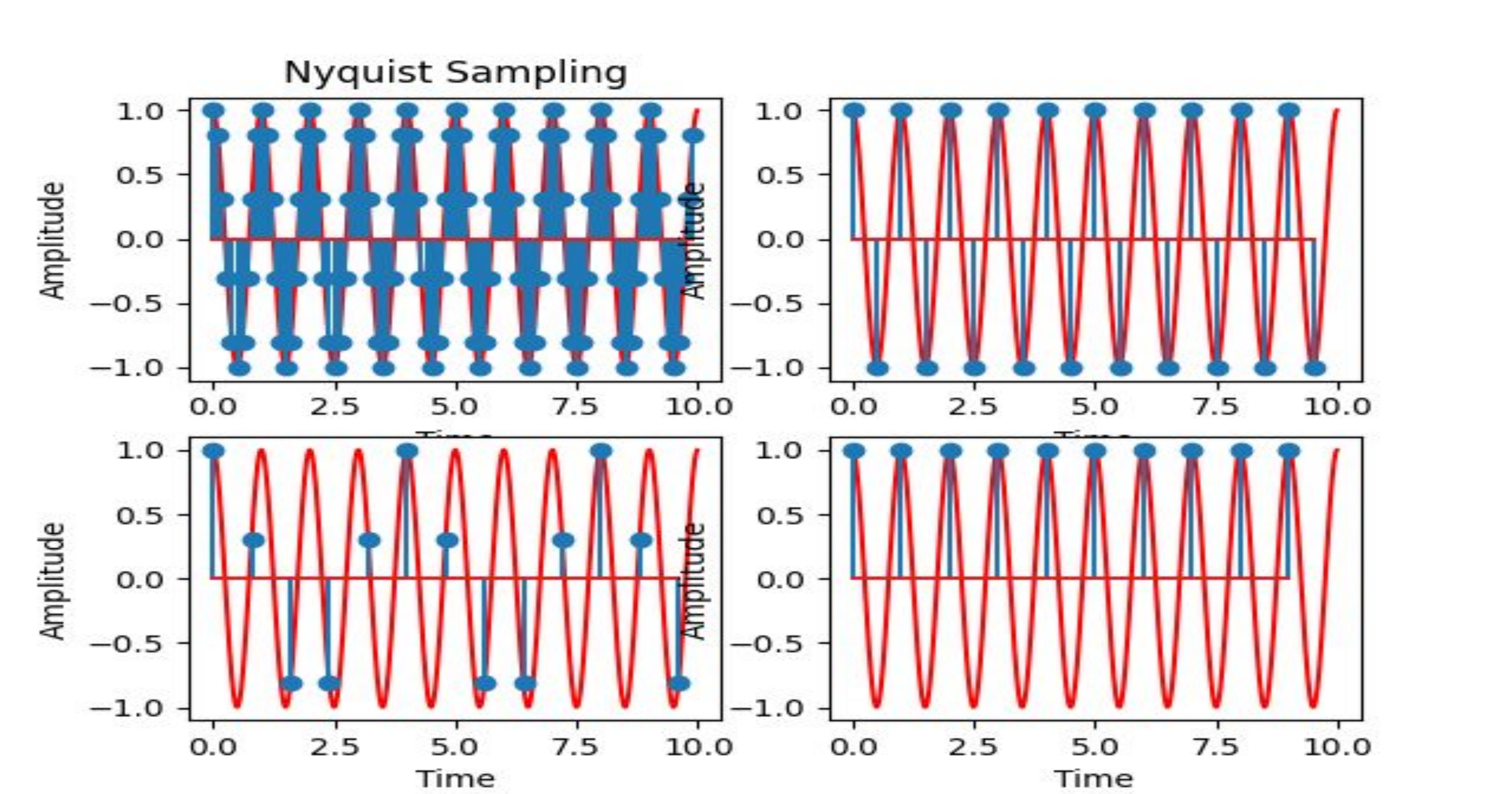
- Sampling is the process of turning the analog CT signal to a digital DT signal
  - Simply, use an ADC (analog-digital converter) and clock
  - At every clock tick, measure the analog signal and digitize it
  - ADCs are a rich subject, a whole presentation could be given on just them
- We want to be able to get our original (analog CT) signal back after sampling
  - If we can get that signal back, then we haven't lost any information

Nyquist Theorem

- How do we ensure we can get our original signal back?
- The Nyquist Theorem tells us how we can do this
  - Nyquist tells us we must use a sampling rate at least twice the highest frequency of the signal of interest
  - For example, if the signal of interest has a frequency of 1kHz, we must sample at at least 2kHz
- Using a lower sampling rate will cause aliasing (wagon wheel effect)



Sampling, cont.



Filtering

- Now that we have our digital signal we want to do something useful, like applying a filter
- Let’s apply a LPF (Low Pass Filter) to a signal
- But our signal is a string of amplitudes over time, how do we know how to apply a LPF, i.e. attenuate only the higher frequency parts?
- We need a way to look at amplitude vs frequency (instead of amplitude vs time)

Enter the Fourier Transform

- The Fourier transform is how we look at a signal in the frequency domain
- The Fourier transform (FT) decomposes a signal into sinusoids
  - This means frequency will be on the x-axis instead of time
  - We are now in the frequency domain
- We can apply the FT to a signal, do some processing, then apply the inverse Fourier transform (IFT) to get back to the time domain

Jean-Baptist Joseph Fourier

Fourier Transform Definition

Fourier Transform

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

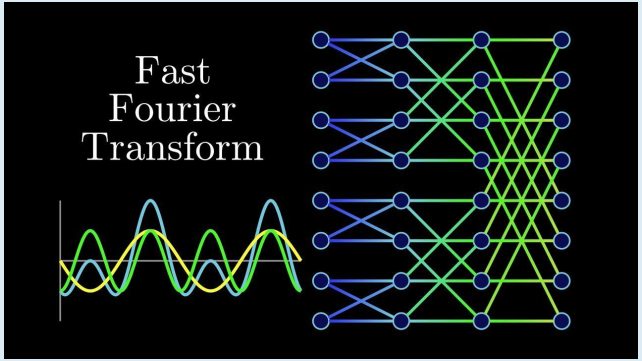
Inverse Fourier Transform

$$f(t) = \int_{-\infty}^{\infty} F(\omega) e^{j\omega t} d\omega$$

Note how similar these two formulas are!

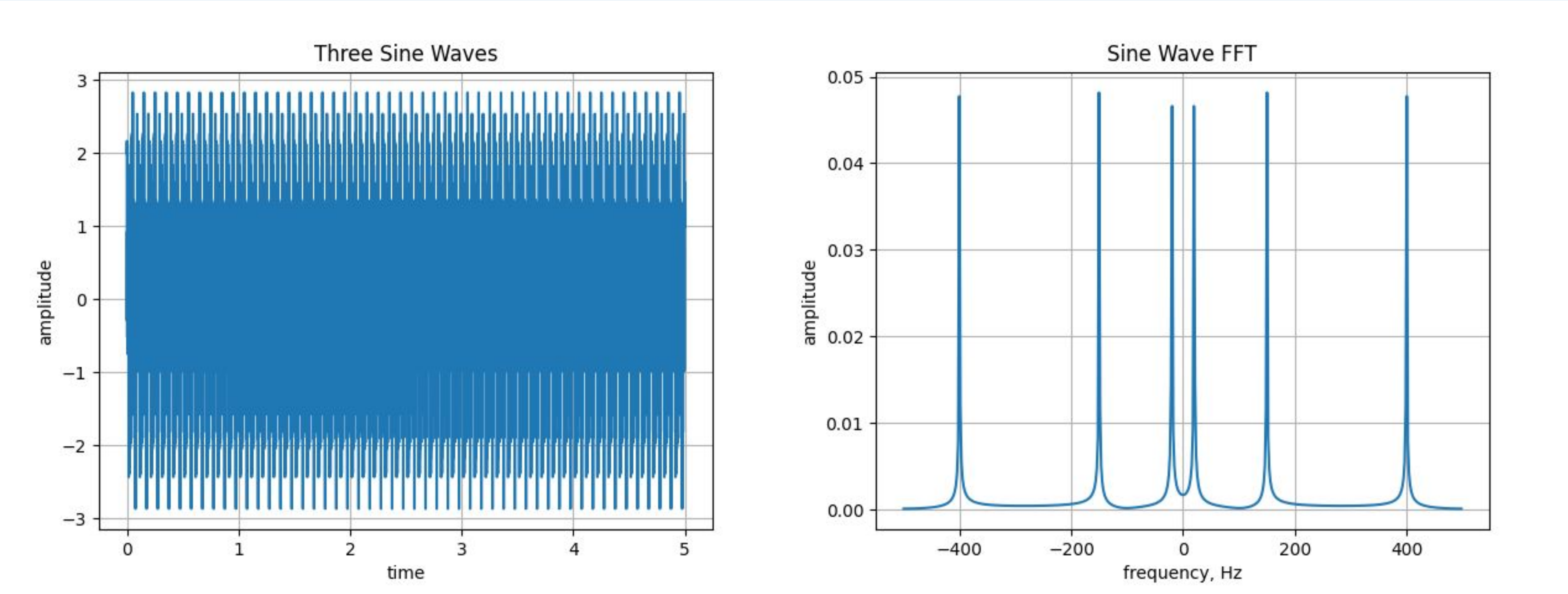
Fast Fourier Transform

- Actually evaluating the Fourier transform can be very hard (i.e. slow)
- The fast Fourier transform (FFT) is a clever algorithm that applies the Fourier transform to a discrete time signal
- It is incredibly efficient - $O(N \log(N))$ instead of $O(N^2)$
- Without the FFT, DSP as we know it would be impossible



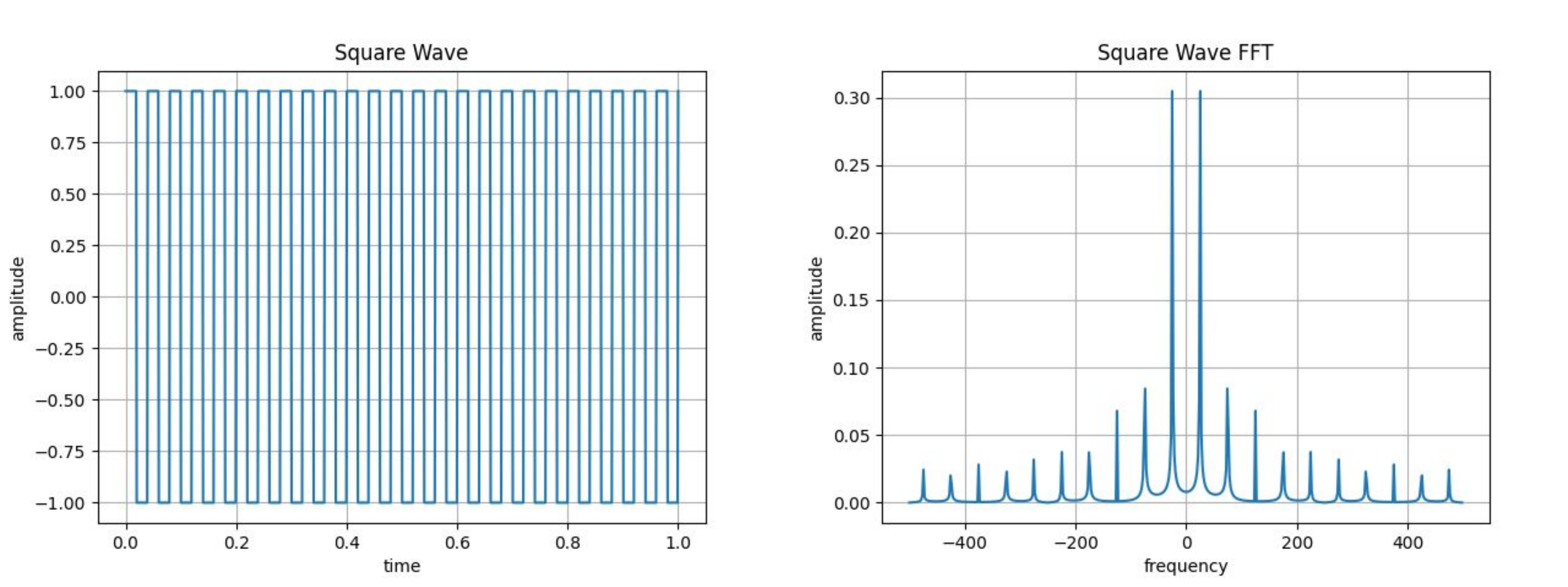
FFT Example

- Sine Waves (20, 150, 400 Hz), Fs = 1kHz



FFT Example

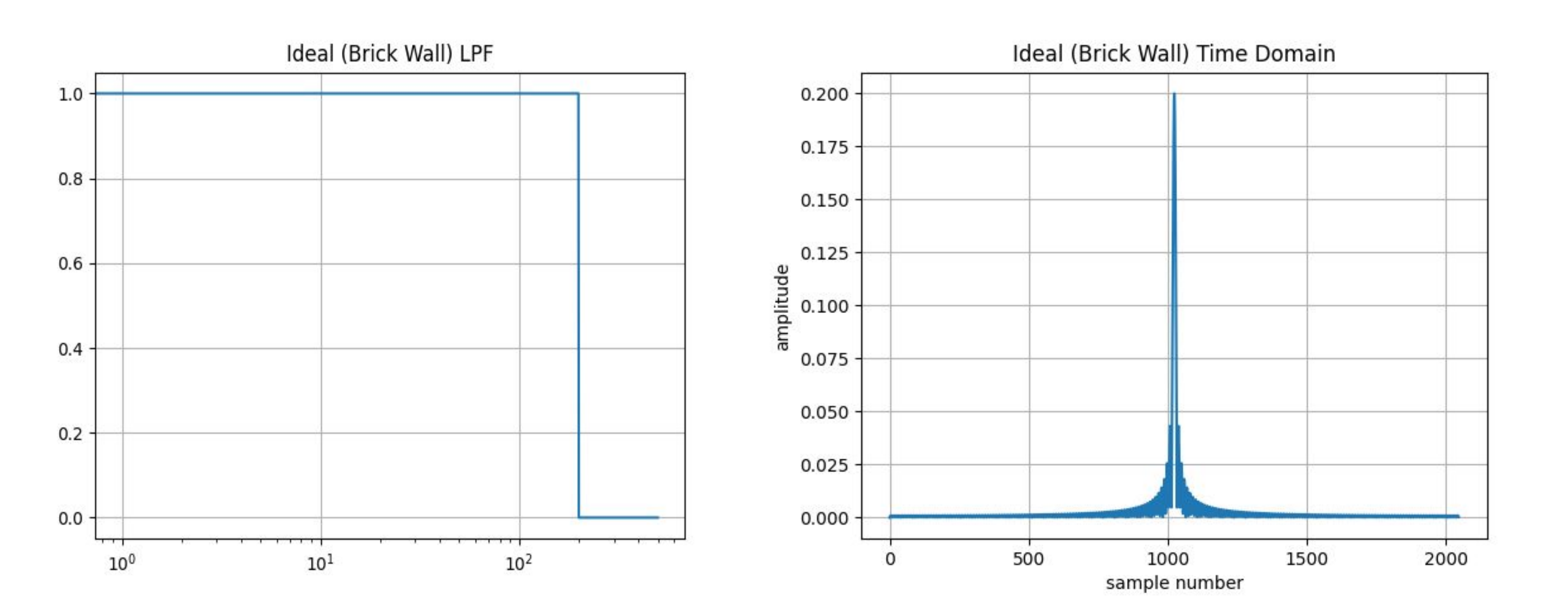
- Square Wave (25Hz), Fs = 1kHz



Ideal Low Pass Filter

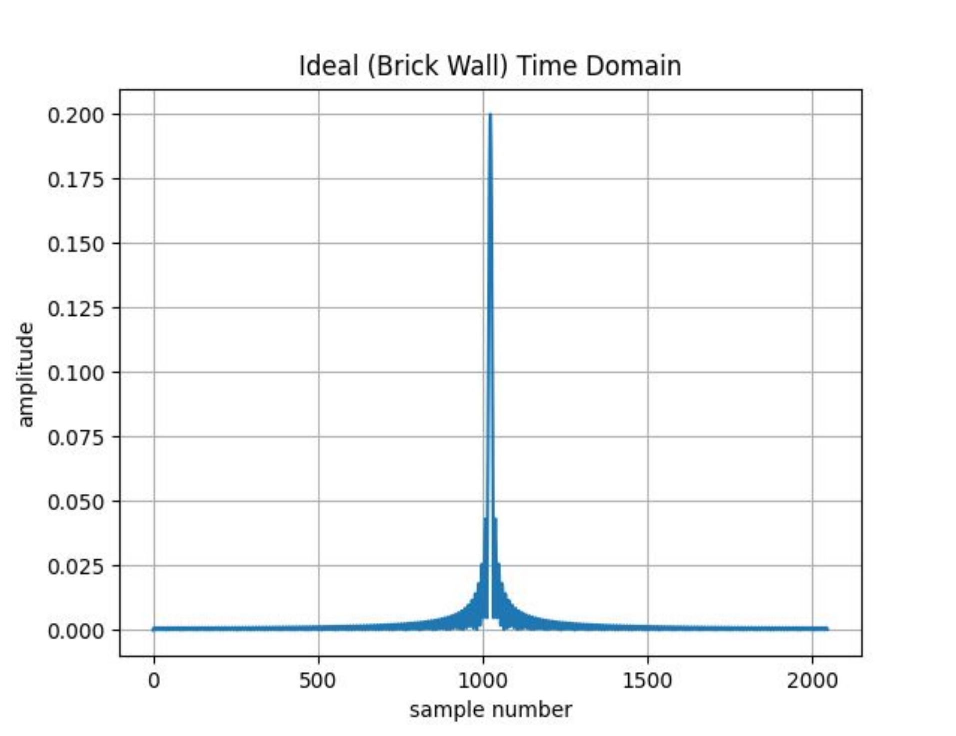
- Now that we can see signals in the frequency domain, it is easier to design a LPF
- Your first thought might be to make an ideal “brick wall” filter: no attenuation up to the cutoff frequency, then no signal above that
- While this looks good in the frequency domain, in the time domain it is not possible - you’d need an infinite number of samples
    - Cutting off some samples will round the corners of your nice brick wall

Ideal LPF

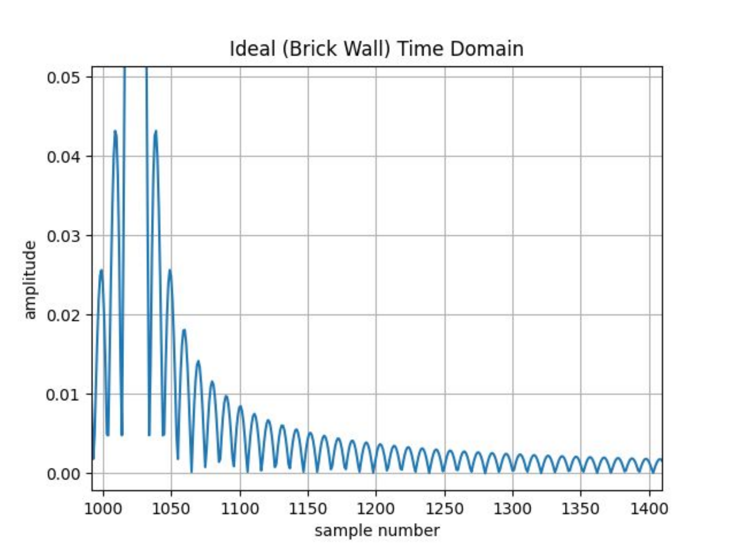


Cutoff frequency 200Hz

Ideal LPF

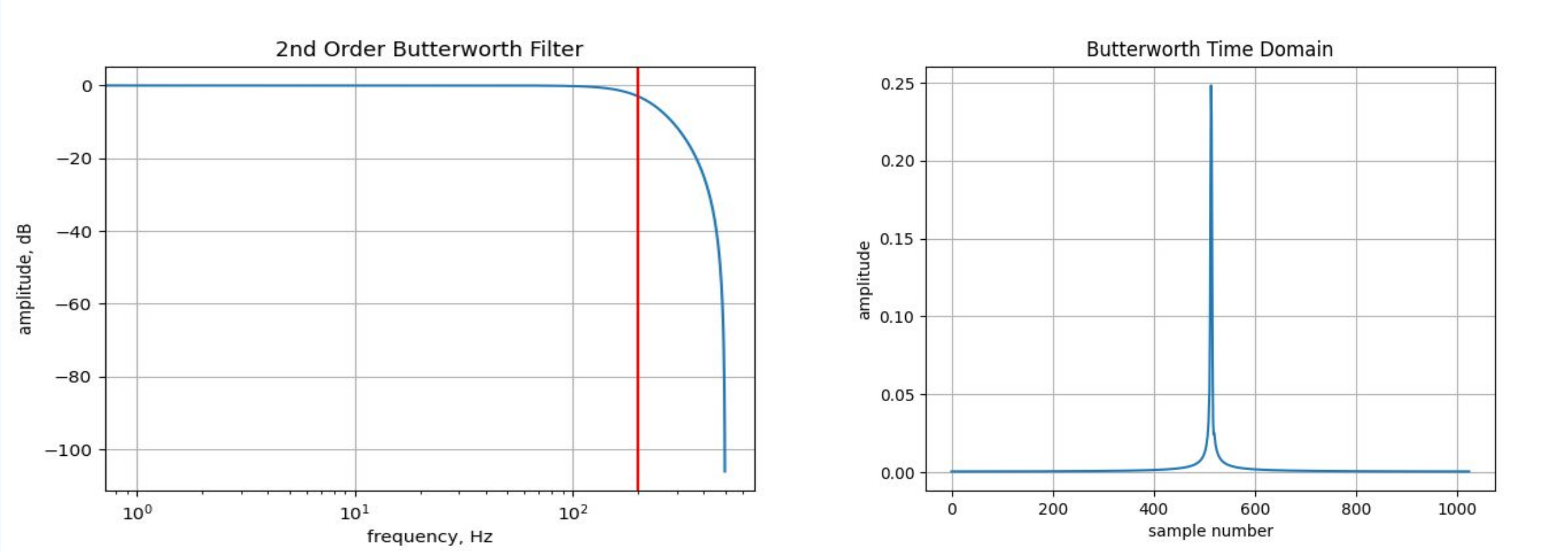


These oscillations are not good!



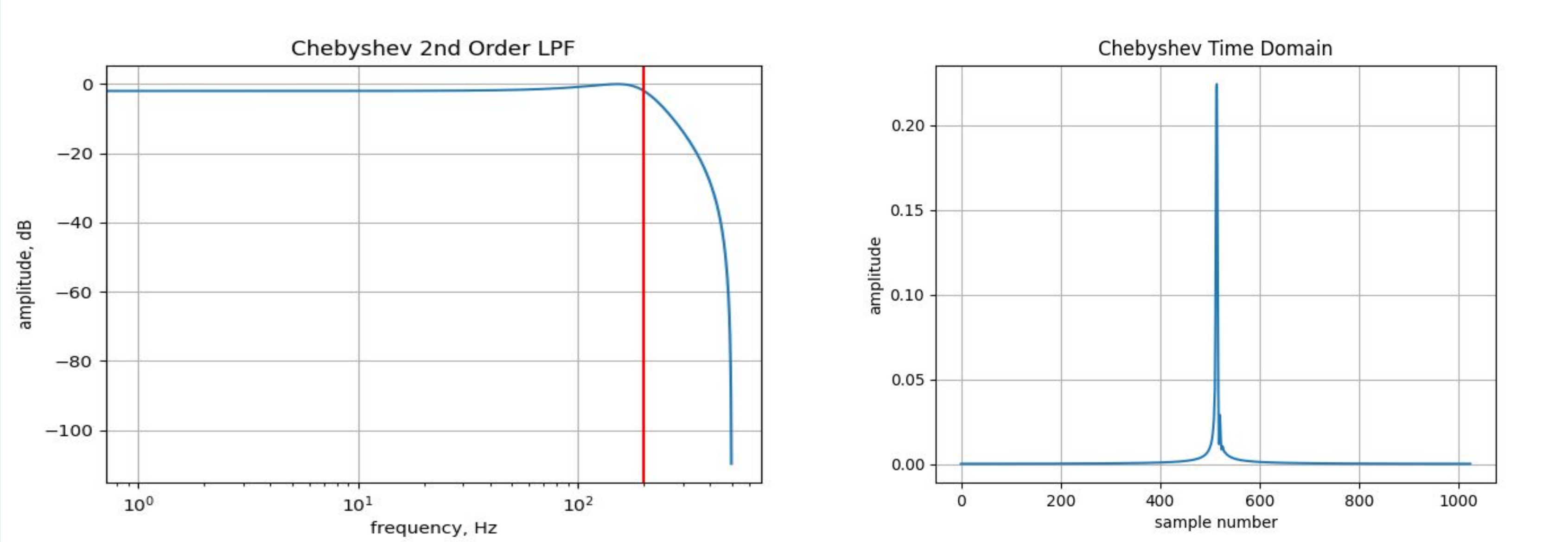
Practical Low Pass Filters

- There are some common filter types
- Butterworth filters have a very flat response in the passband and stopband



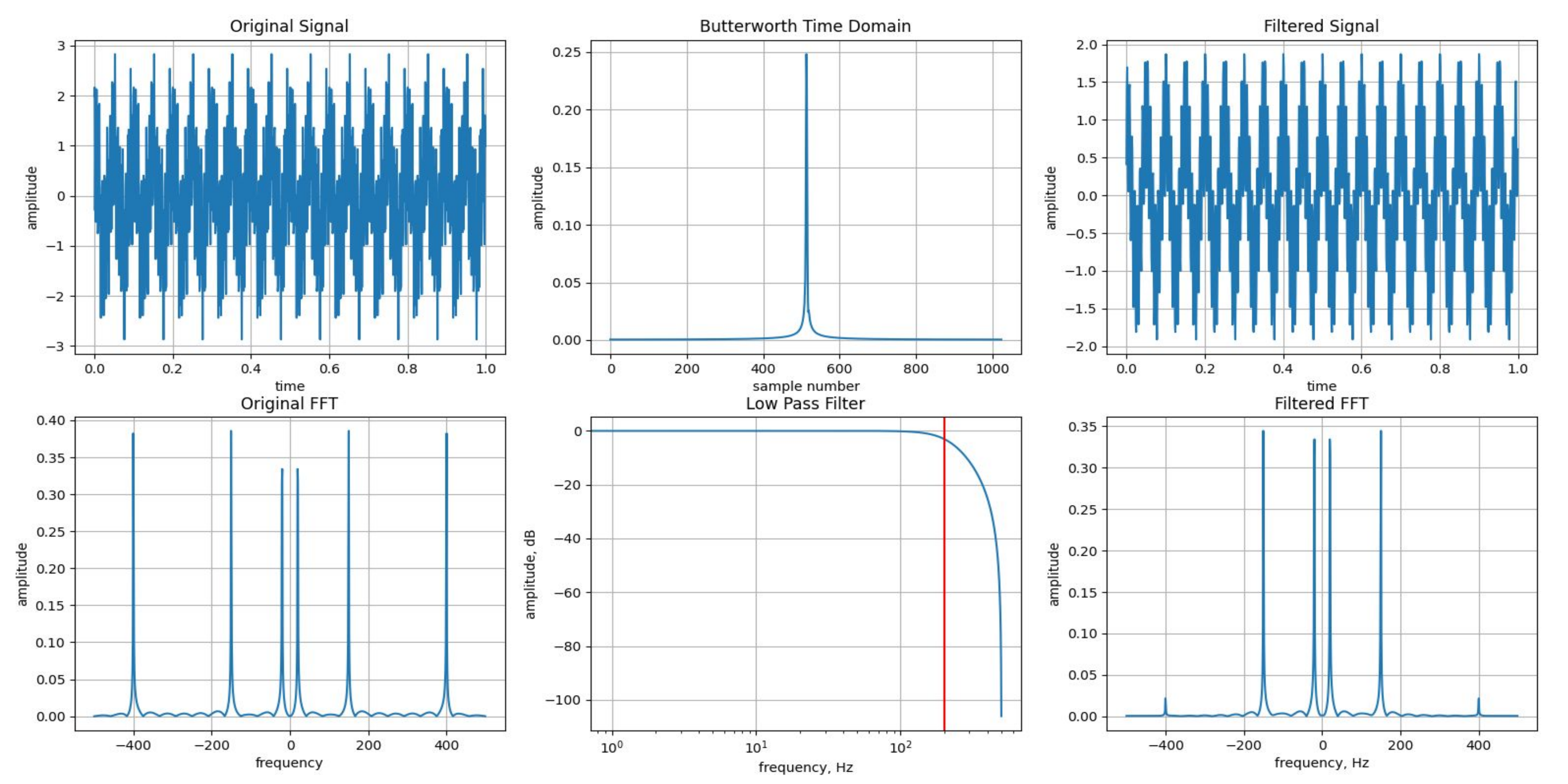
Practical Low Pass Filters

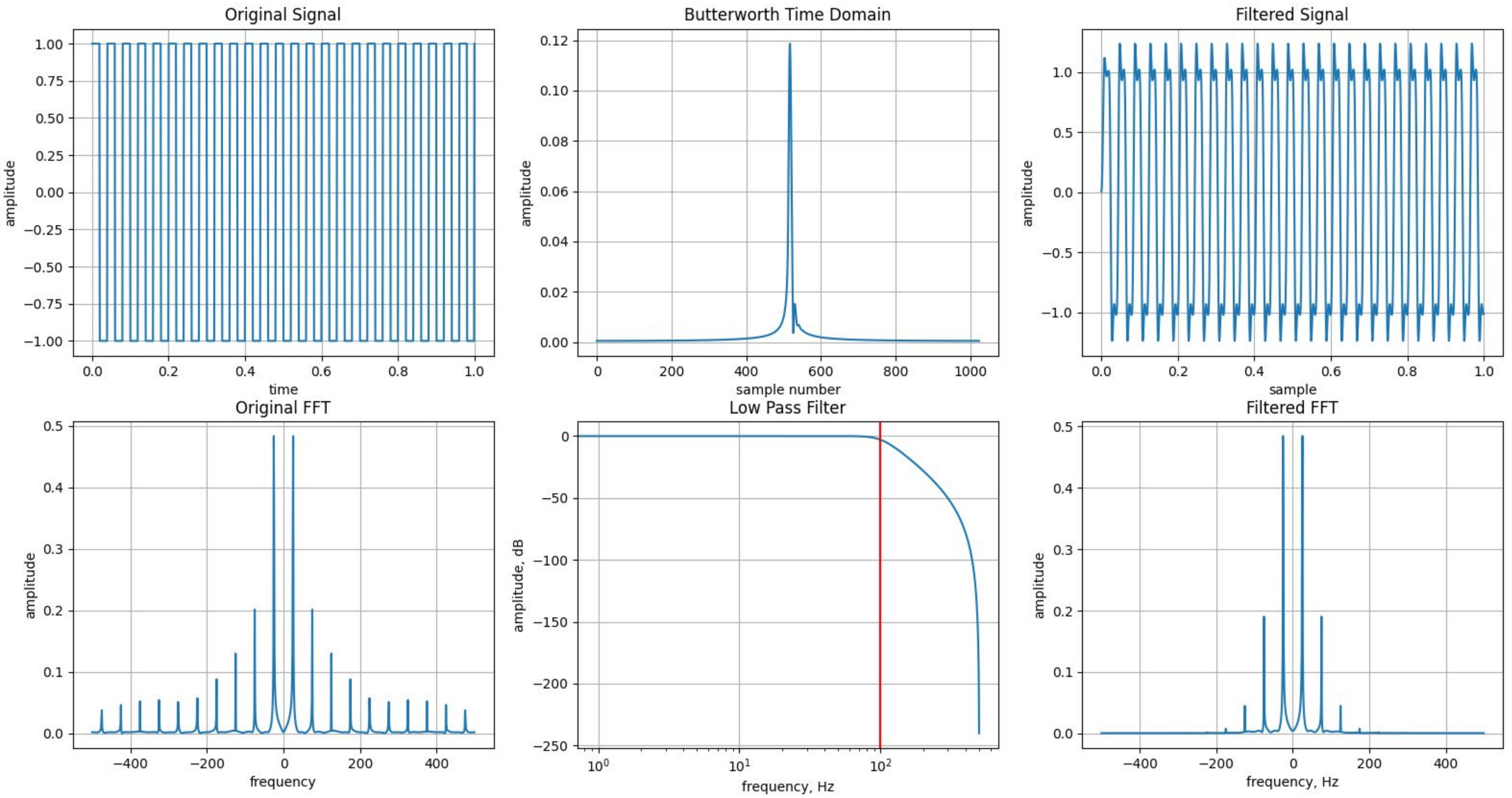
- The Chebyshev filter is also commonly used
- These filters have a much steeper transition, but ripple in either the pass- or stop band



Signal Chain

Original Signal -> FFT -> Apply Filter -> IFFT -> Filtered Signal



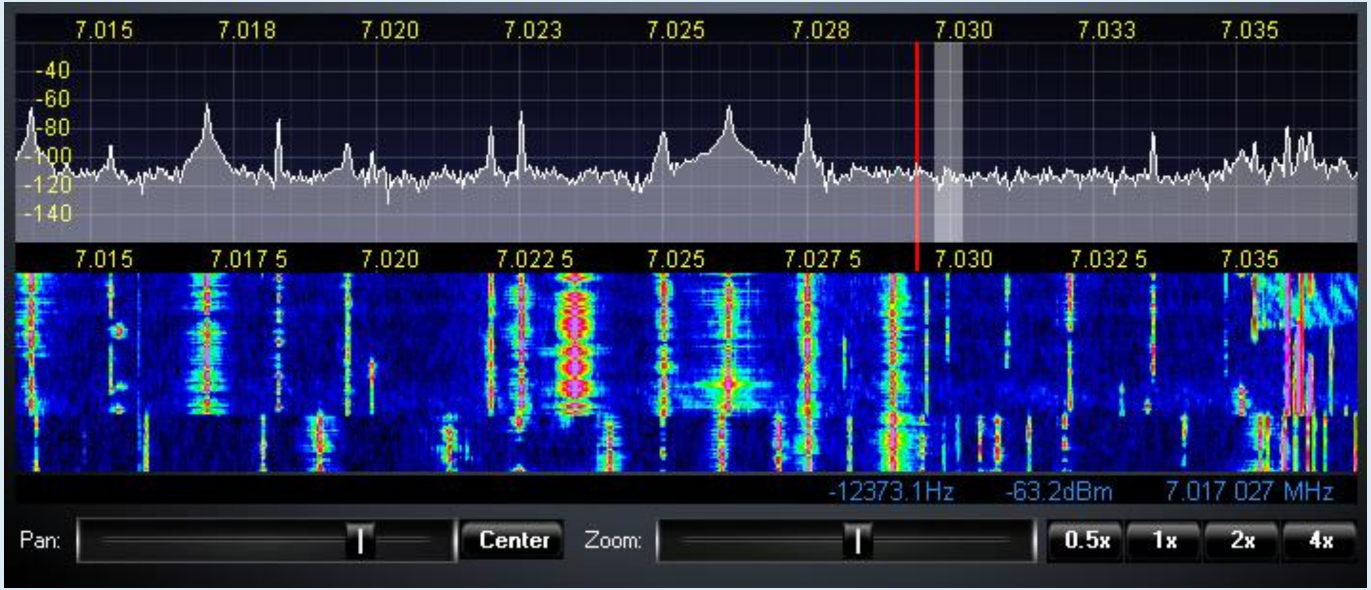

Original Signal
amplitude
time
Butterworth Time Domain
amplitude
sample number
Filtered Signal
amplitude
sample
Original FFT
amplitude
frequency
Low Pass Filter
amplitude, dB
frequency, Hz
Filtered FFT
amplitude
frequency, Hz

Ham Radio DSP Examples

- Notch Filter: Remove very small band from the received signal. With DSP it is easy to change the location and size of the notch
- Pass Band Tuning: Change the width and position of the passband

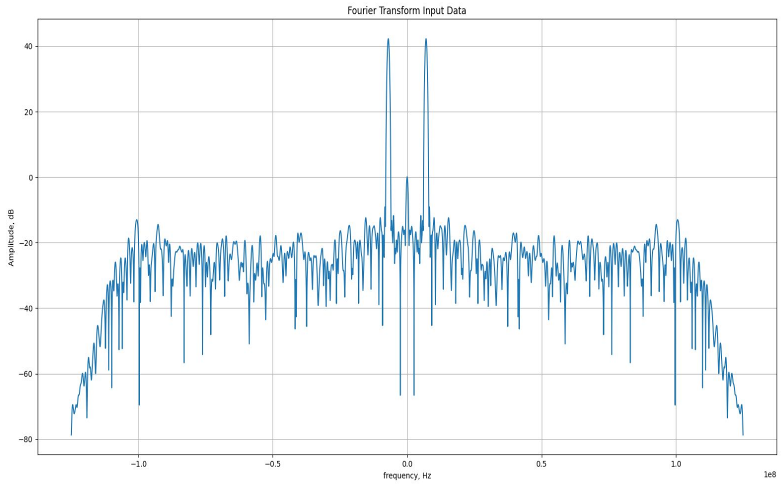
Ham Radio DSP Examples

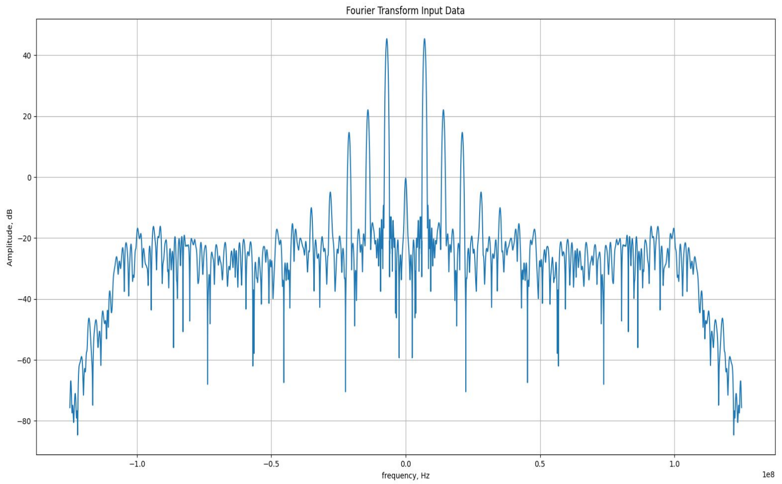
- Waterfall Display: Take repeated snapshots, then take the FFT. Plot these.



- Poor man's spectrum scope - see next slides

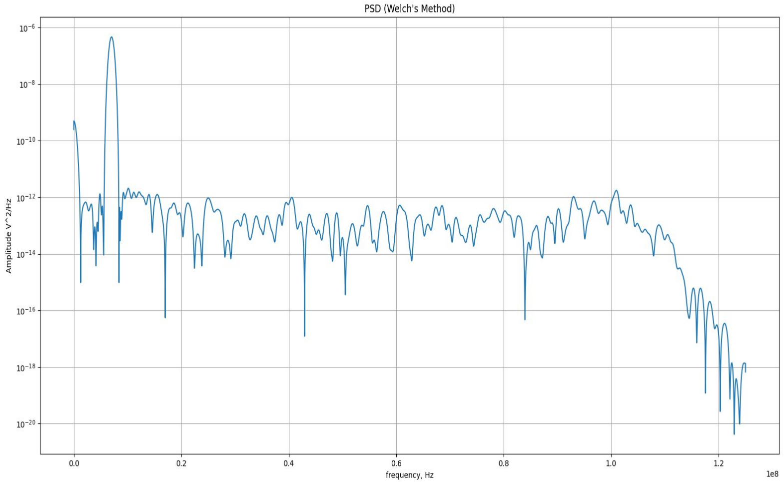
Check Transmitter Spectrum

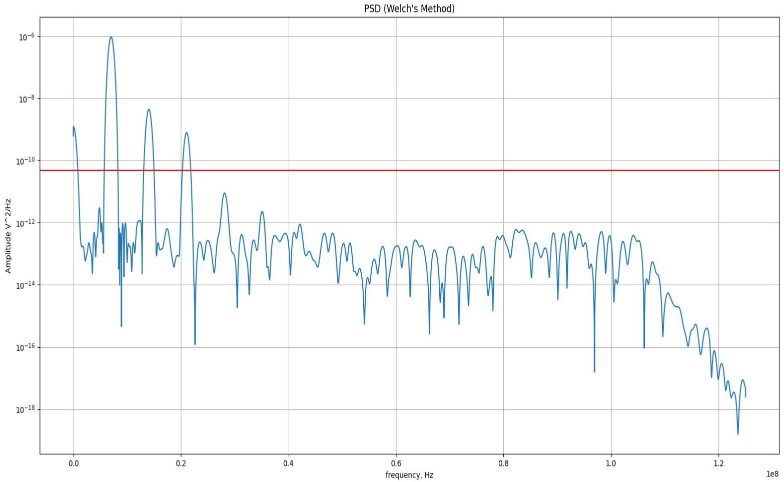




- Spectrum of the KX3 (left) and an eBay Forty-9er (right), both in the 40m band
- Digital oscilloscopes are fairly cheap, and with some DSP knowledge the ‘scope can pull double duty as a spectrum analyzer as well

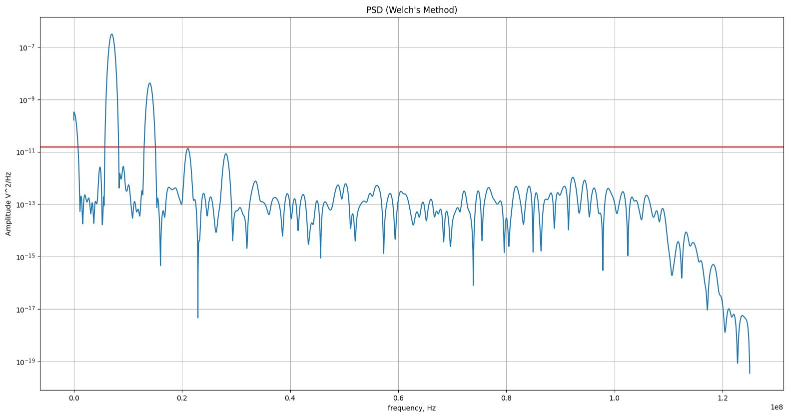
Power Spectral Density

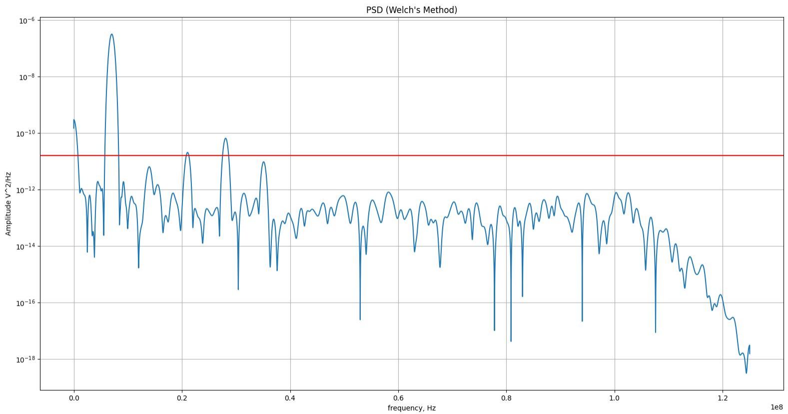




- PSD of the KX3 (left) and eBay Forty-9er (right), both in the 40m band
- Red line is FCC limit for spurious emissions
    - A better output LPF is needed for the Forty-9er

Pixie PSD

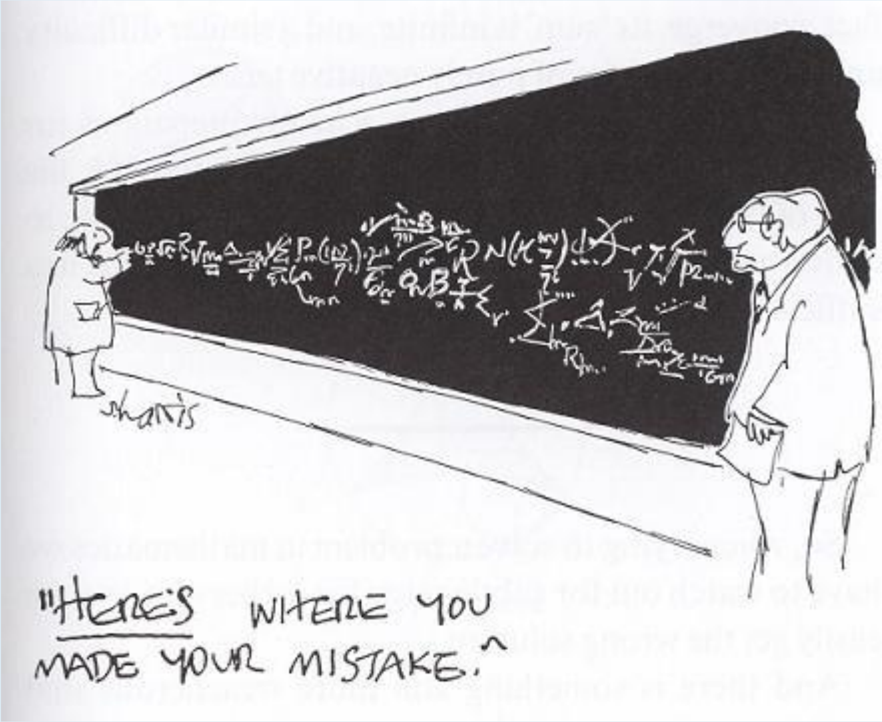




Pixie2 transceiver kit (40m) from eBay. Original LPF on left, 2nd harmonic trap added to spectrum on right.

Surprisingly this spectrum was better than the Forty 9er’s, but still not within FCC limits.

Questions?



More Reading

- http://www.dspguide.com/
- https://www.youtube.com/watch?v=nmgFG7PUHfo
- https://www.youtube.com/watch?v=spUNpyF58BY
- https://www.youtube.com/watch?v=h7apO7q16V0

This presentation and the code used to generate the figures