Fourier Transform HW

November 26, 2021

#Fourier Transform HW#

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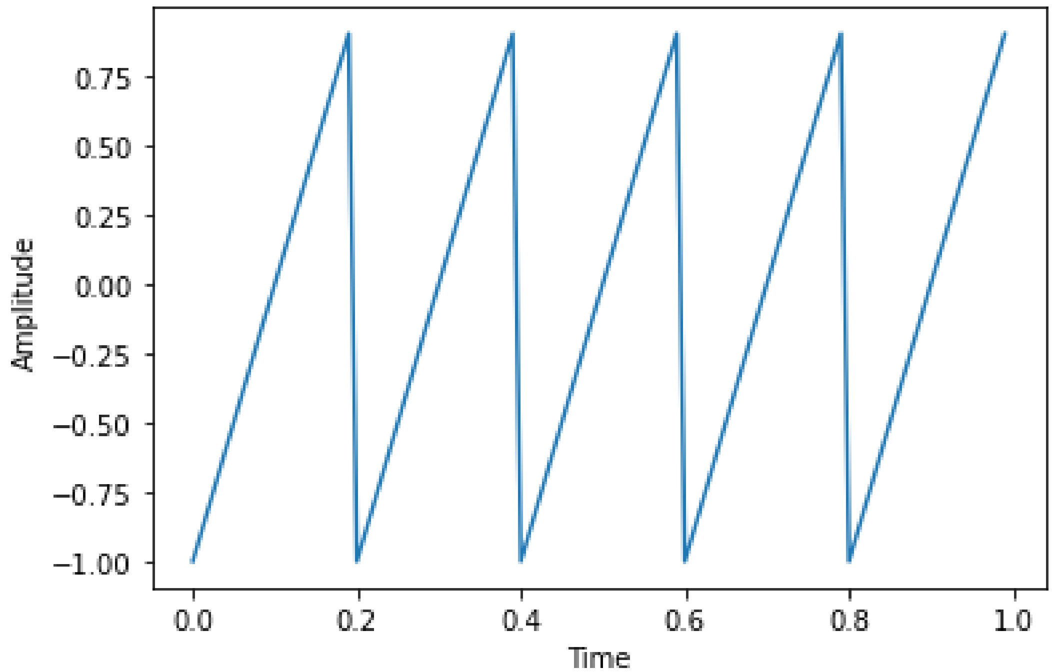
#Question 1#

[3]:
```
import matplotlib.pyplot as plt
from scipy import signal
import numpy as np
from scipy.integrate import simps

Fs = 100
amplitude = 1
duration = 1  # in seconds
N = Fs * duration
t = np.linspace(0, duration, num=N, endpoint=False)

# create the signal
frequency = 5
sig = amplitude * signal.sawtooth(2 * np.pi * frequency * t)
plt.plot(t, sig)
plt.xlabel('Time')
plt.ylabel('Amplitude')
plt.show()
```



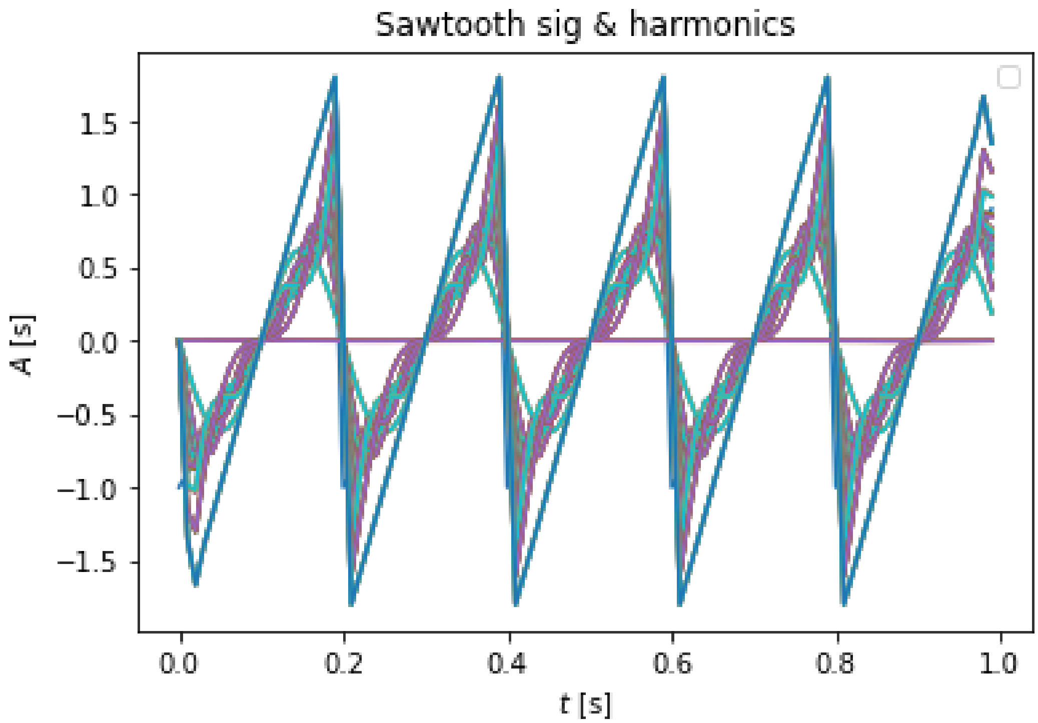
[5]:
```python
def sawtooth_harmonics(number_of_harmonics):
    n = number_of_harmonics
    fig, ax = plt.subplots()
    ax.plot(t, sig)
    sum = 0

    # it's an odd function, fft has only sin
    def b(n):
        return (2.0 / duration) * simps(sig * np.sin(2.0 * np.pi * n * t /
 ↪duration), t)

    for i in range(1, n + 1):
        sum += b(i) * (np.sin(2.0 * np.pi * i * t / duration))
        ax.plot(t, sum)
        ax.set_xlabel('$t$ [s]')
        ax.set_ylabel('$A$ [s]')
    ax.legend()
    ax.set_title("Sawtooth sig & harmonics")
    plt.show()

sawtooth_harmonics(100)
```

```
No handles with labels found to put in legend.
```



After a few trials, we can see that we need around 100 harmonics to represent this wave well.

#Question 2#

[12]:
```
fig, ax = plt.subplots()
Fs = 100
d = 5  # in seconds
N = Fs * d
t = np.linspace(0, d, num=N, endpoint=False)


f1 = 2
amp1 = 2
sig1 = amp1*np.sin(2 * np.pi * f1 * t)

f2 = 10
amp2 = 4
sig2 = amp2*np.sin(2 * np.pi * f2 * t)
ax.plot(t, sig1)
ax.plot(t, sig2)

ax.legend()
plt.xlabel('$t$ [s]')
plt.ylabel('$\sin (t)$')
```

```
ax.set_title("Time domain")
plt.show()
```

```
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```
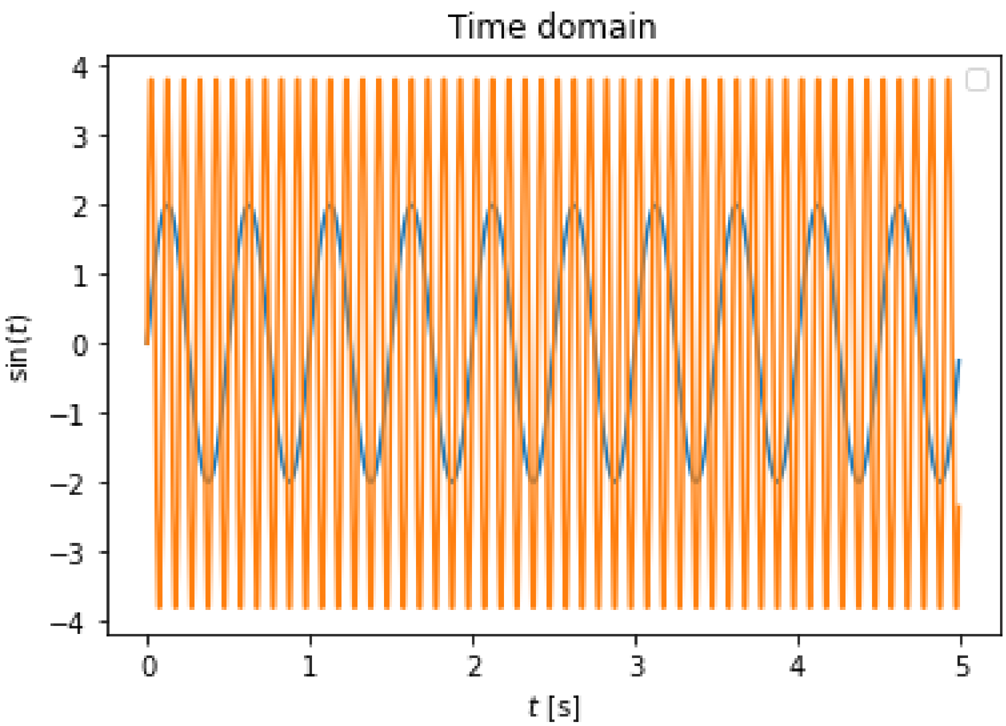
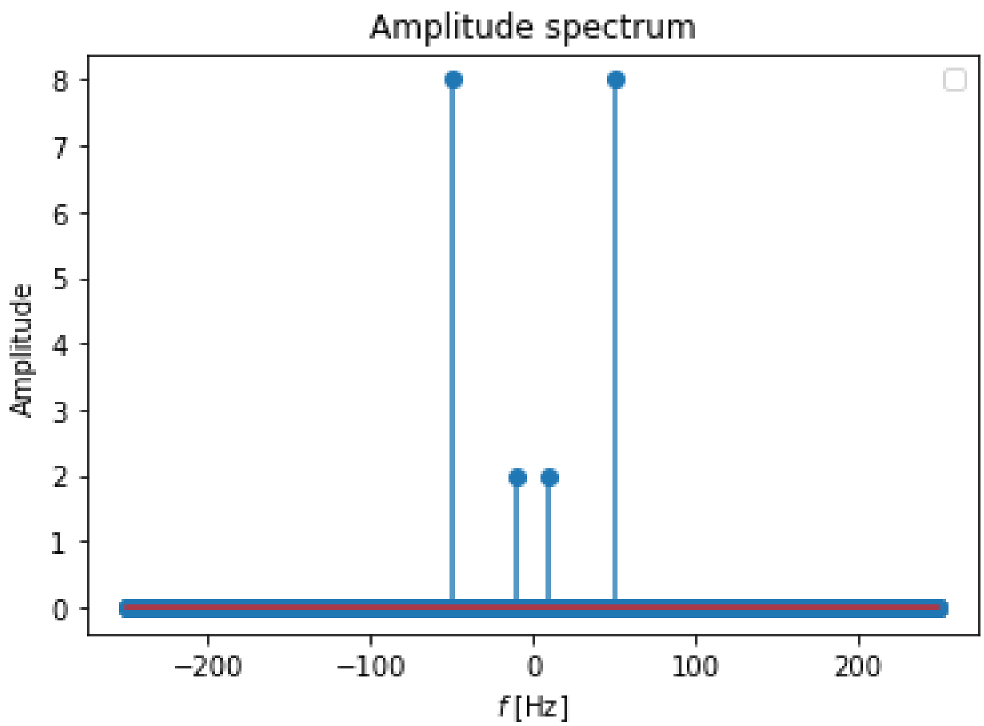




```
[13]: fig2, ax2 = plt.subplots()
fft1 = np.fft.fftshift(np.fft.fft(sig1))
fft2 = np.fft.fftshift(np.fft.fft(sig2))

power1 = np.square(np.abs(fft1)) / N
power2 = np.square(np.abs(fft2)) / N

amp_spec1 = (2 / N) * np.abs(power1)
amp_spec2 = (2 / N) * np.abs(power2)

ax2.stem(np.arange(-N/2, N/2), amp_spec1, use_line_collection=True)
ax2.stem(np.arange(-N/2, N/2),amp_spec2, use_line_collection=True)



ax2.legend()
ax2.set_title("Amplitude spectrum")
ax2.set_xlabel('$f$ [Hz]')
```

```
ax2.set_ylabel('Amplitude')
plt.show()
```

```
No handles with labels found to put in legend.
```



#Question 3#

[24]:
```
from scipy.io import wavfile
Fs, sig = wavfile.read('guitartune.wav')
d = 15  # in seconds
N = Fs * d
t = np.linspace(0, d, num=N, endpoint=False)

fig, ax = plt.subplots()
fft = np.fft.fftshift(np.fft.fft(sig))
power_spectrum = np.square(np.abs(fft))/N
ax.stem(np.arange(-N/2,N/2), power_spectrum, use_line_collection=True)

ax.legend()
ax.set_title("Power spectrum")
ax.set_xlabel('$f$ [Hz]')
ax.set_ylabel('Power')
plt.show()
```

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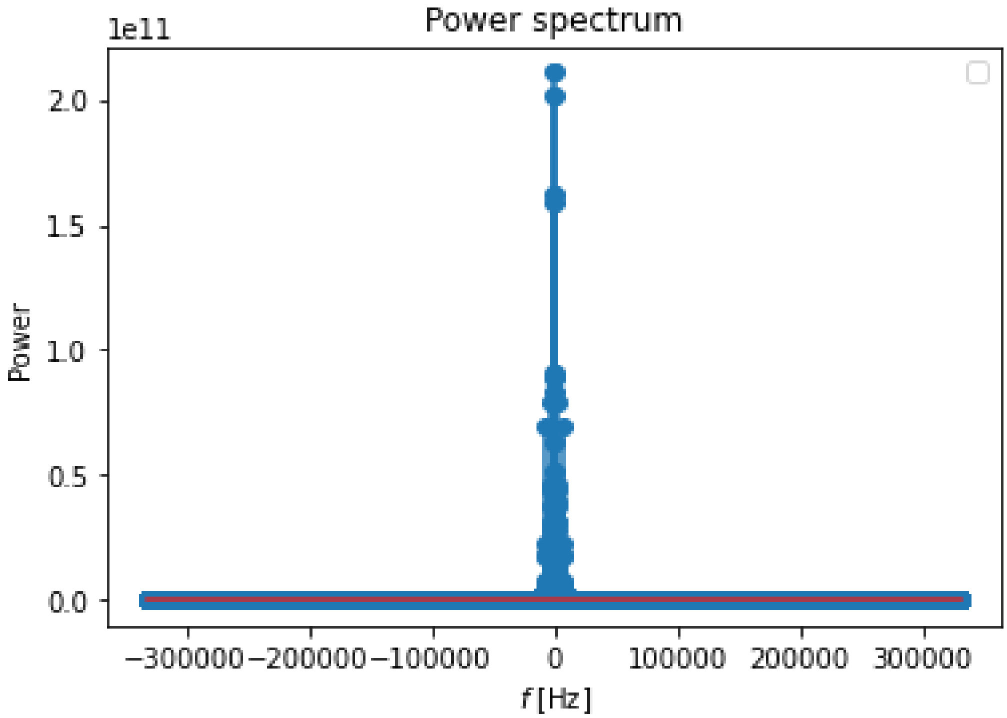

[25]:
```
fig2, ax2 = plt.subplots()
fft2 = fft.copy()
ten_present = int(np.shape(fft2)[0]*0.1)
fft2[(-1)*ten_present:] = 0
fft2[:ten_present] = 0
power_spectrum2 = np.square(np.abs(fft2))/N
ax2.stem(np.arange(-N/2,N/2), power_spectrum2, use_line_collection=True)

ax2.legend()
ax2.set_title("Power spectrum after filtering")
ax2.set_xlabel('$f$ [Hz]')
ax2.set_ylabel('Power')
plt.show()
```

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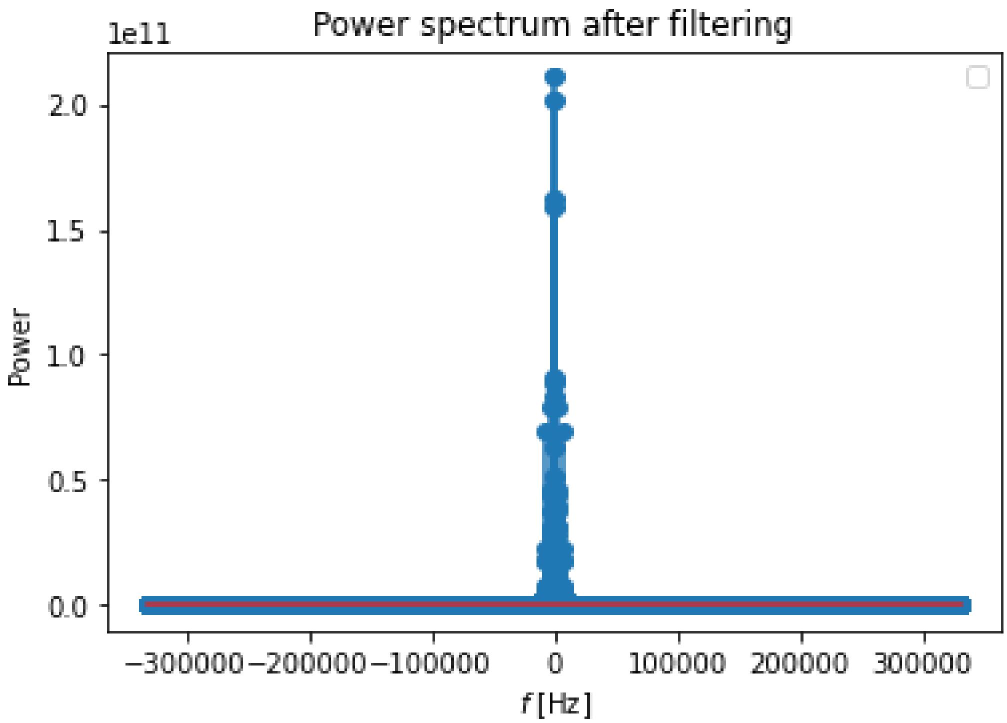

We can see that the Power spectrum hasn't change. Therefore, we didn't need to erase these frequencies. Now, let's apply the inversed Fourier Transform on our file:

[26]:
```
ifft2 = np.fft.ifft(np.fft.ifft(sig))

wavfile.write('ifft guitar.wav', Fs, (ifft2*2**16).astype(np.int16))
```

```
C:\Users\hillu\AppData\Local\Temp/ipykernel_14024/1480286687.py:3:
ComplexWarning: Casting complex values to real discards the imaginary part
  wavfile.write('ifft guitar.wav', Fs, (ifft2*2**16).astype(np.int16))
```

Let's apply some filters: #Butterworth filter#

[27]:
```
b, a = signal.butter(N=5, Wn=200, fs=Fs)
output = signal.filtfilt(b, a, sig)
wavfile.write('Butterworth.wav', Fs, (output*2**16).astype(np.int16))
```

#Bessel filter#

[28]:
```
b2, a2 = signal.bessel(N=5, Wn=0.5, analog=True)
output = signal.filtfilt(b2, a2, sig)
wavfile.write('Bessel.wav', Fs, (output*2**16).astype(np.int16))
```

#Flattop filter#

[29]:
```
window = signal.flattop(20)
b3 = np.ones(window.__len__()) * (1/window.__len__())
output = signal.filtfilt(b3, a2, sig)
wavfile.write('Flattop.wav', Fs, (output*2**16).astype(np.int16))
```

In my opinion, the originals signal sounds the best. But from the results from the filters we applyed, 'Bessel.wav' has less noise, and sounds better.