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ME465 - Sound and Space

03/06/2020

HW5: Exciting Systems

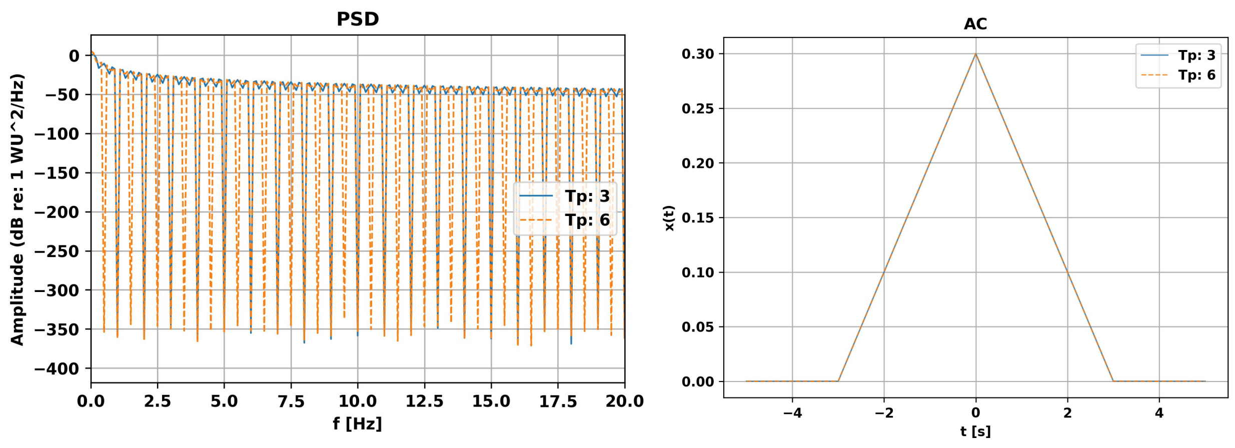
Question 1

The purpose of question one was to create functions for generating time-series impulses, simple sine waves, linear sweeping sine waves, logarithmic sweeping sine waves, and random noise on a subset of some interval where the rest of the values are zero. The appendix section marked ‘Question 1’ includes sample plots for these functions. Notably, to generate random noise, a symmetric, flat frequency response was first created using random phase information before converting it back into the time domain.

Question 2

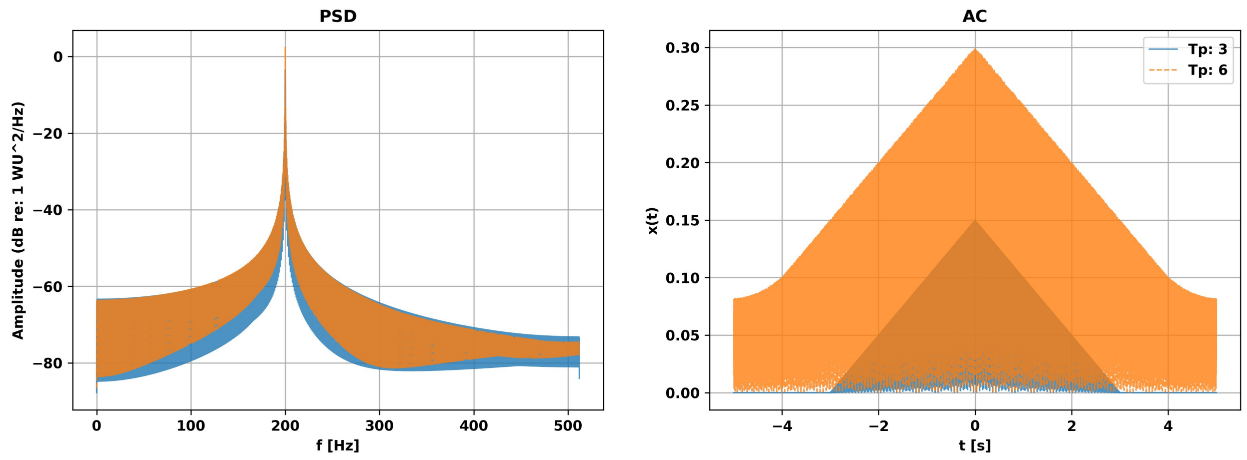
Question 2 asked for a comparison of the power spectral density and auto-correlation for each signal type when the sub-signal length (Tp) was changed. Tp lengths of 3s and 6s were selected for a total signal length of 10s. For the sweeping signals, frequencies from 1 to 200 Hz were implemented. For all signals except impulse and random noise, the power spectrum response was generally greater when the Tp was larger —- generally, longer signals have more energy. The impulse signal’s PSD was studied in detail in Figure 1a (the full spectrum is available in the appendix). Increasing Tp had the effect of increasing the variability in the power spectral density response. Since the impulse response excites a wide range of frequencies, it is useful in theory for quickly learning about how a system’s magnitude responds to all frequencies; increasing the Tp may allow for higher frequency granularity when conducting such studies. For the impulse’s auto-correlation, increasing Tp had no effect on the triangular shaped response. Generally, the relatively high AC that impulses have would not make them a good tool for studying how a system delays a signal, nor would increasing Tp provide more information in such studies.

Figure 1a: Impulse



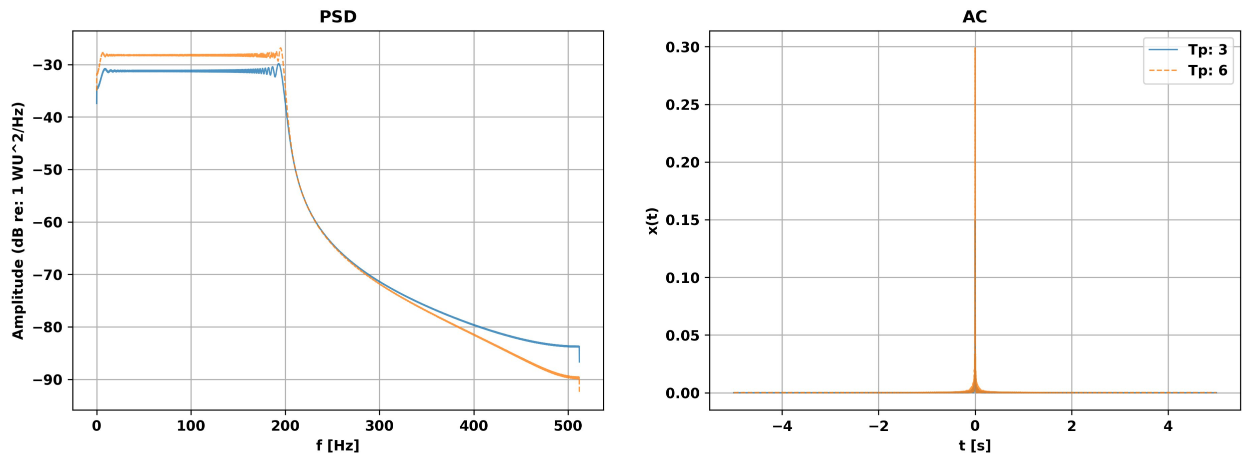
The simple sign wave's PSD had reduced variability when Tp was increased and the peak frequency had more energy. A simple sign wave may be useful to use when there is a specific reason to study a system's behavior at a particular frequency; therefore having prior knowledge about the system may be necessary in order to use this tool effectively. Because of the signal's high auto-correlation, it would not be a good tool for studying the phase delay of a system. The auto-correlation saw increased magnitude when Tp was increased, but this additional information might only make it marginally better at studying the phase delay of a system.

Figure 1b: CW Pulse (f=200 Hz)



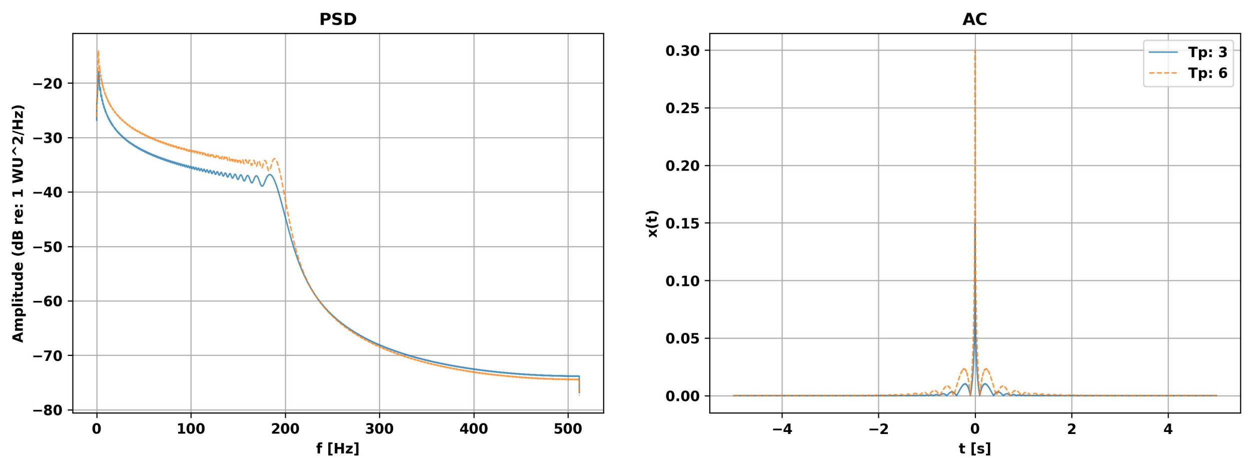
The linear (1c) and logarithmic (2d) sign sweeps had a generally higher PSD and AC when Tp was increased. Notably, the PSD of the linear sweep remained relatively flat in the swept range, while the PSD of logarithmic sweep decreased toward higher frequencies; consequently, it seems like a linear sine sweep would be more useful when studying whether a system amplifies or dampens certain frequencies.

Figure 1c: Linear Sine-Sweep



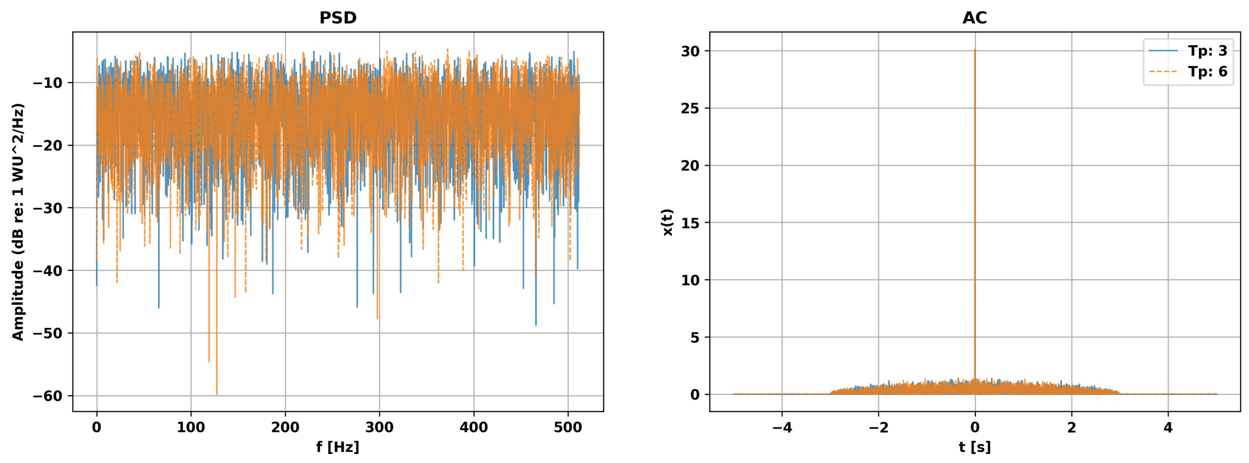
Secondly, the logarithmic sine sweep showed a greater magnitude in its AC across time delays, which would make it slightly harder to study system phase delay. When doing the cross-correlation between an input signal and a measured signal one would ideally like to use an input signal that doesn't have high auto-correlation — the logarithmic sweep isn't as 'bad' for studying system phase as a simple sign wave however.

Figure 1d: Logarithmic Sine-Sweep



Increasing Tp of the white noise signal seems to increase variability of the PSD response, but likely would not provide a benefit when studying the magnitude of a system's response. Conversely however, the AC of the noise is fairly low making it a decent tool for studying how a system transforms a signal's phase (although not as good as the chirps).

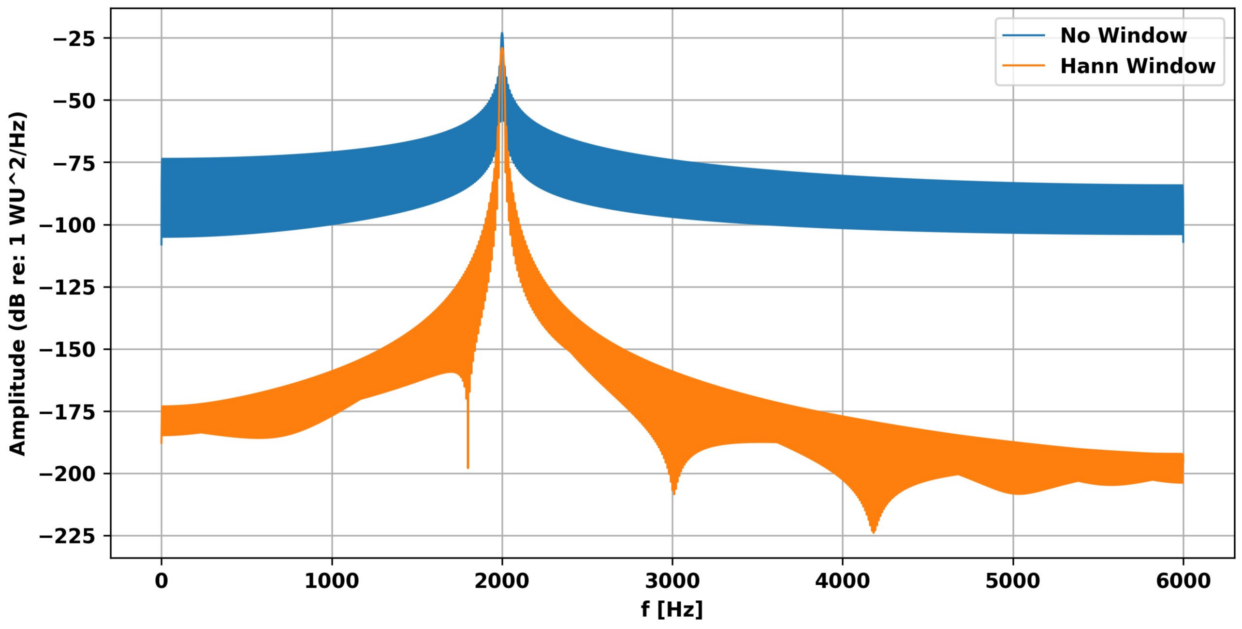
Figure 1e: White Noise



Question 3

The purpose of question 3 was to study the impact of a Hann window on the power spectral density (PSD) of a 2000 Hz sine wave. The taper had the effect of reducing the PSD (shift down) and reducing its variability across most of the spectrum (although increased variability at some frequencies). Most importantly however, the taper reduced variability in the PSD near the signal frequency of 2000 Hz, making it easier to identify the signal frequency.

Figure 2: PSD of Sine Wave (f=2000 Hz) with and without Hann Window



Question 4

The purpose of question 4 was to study how a Hann window affects the PSD response of a linear sine sweep. The window seems to have had two effects - it reduces variability but completely changes the shape of the PSD from a relatively flat response to a curve (Figure 3). Increasing the upper range of the chirp to 8000 Hz increases variability of the PSD with and without the window. In the second scenario, the window resulted in a PSD that did not increase and decrease in the same way it did in the first scenario; instead, it reduced variability particularly in the region above 3500 [Hz] and still transformed a relatively flat low frequency range into a smooth 'increasing' curve.

Figure 3: Linear Sine Sweep with f2 = 5000 Hz

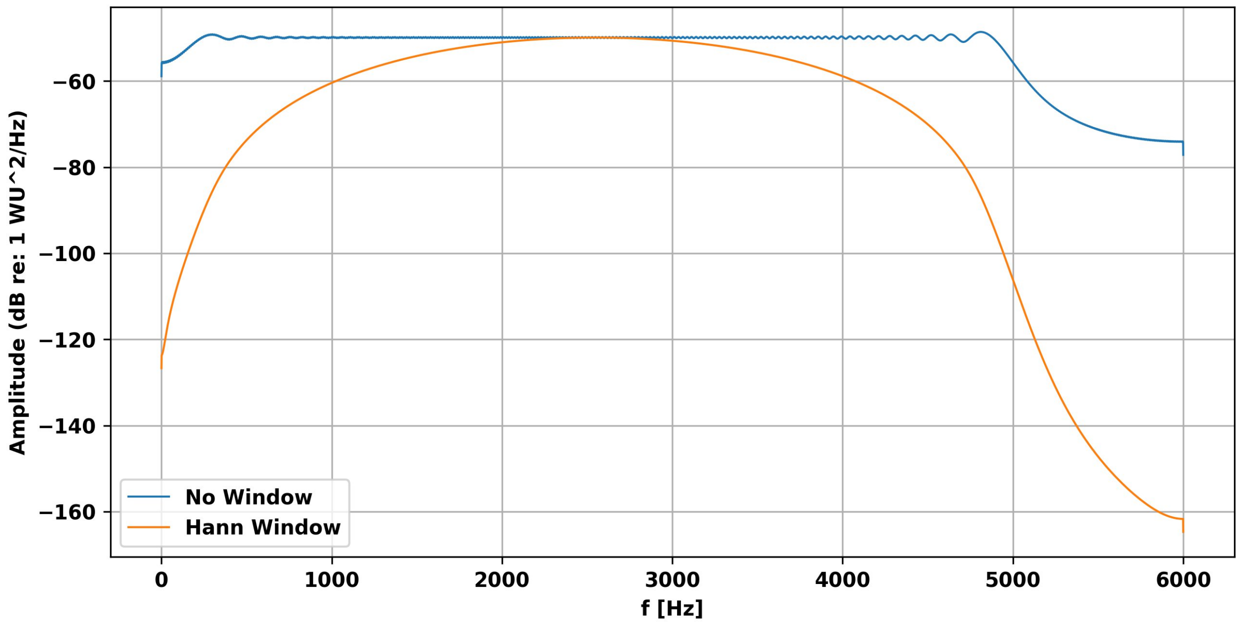
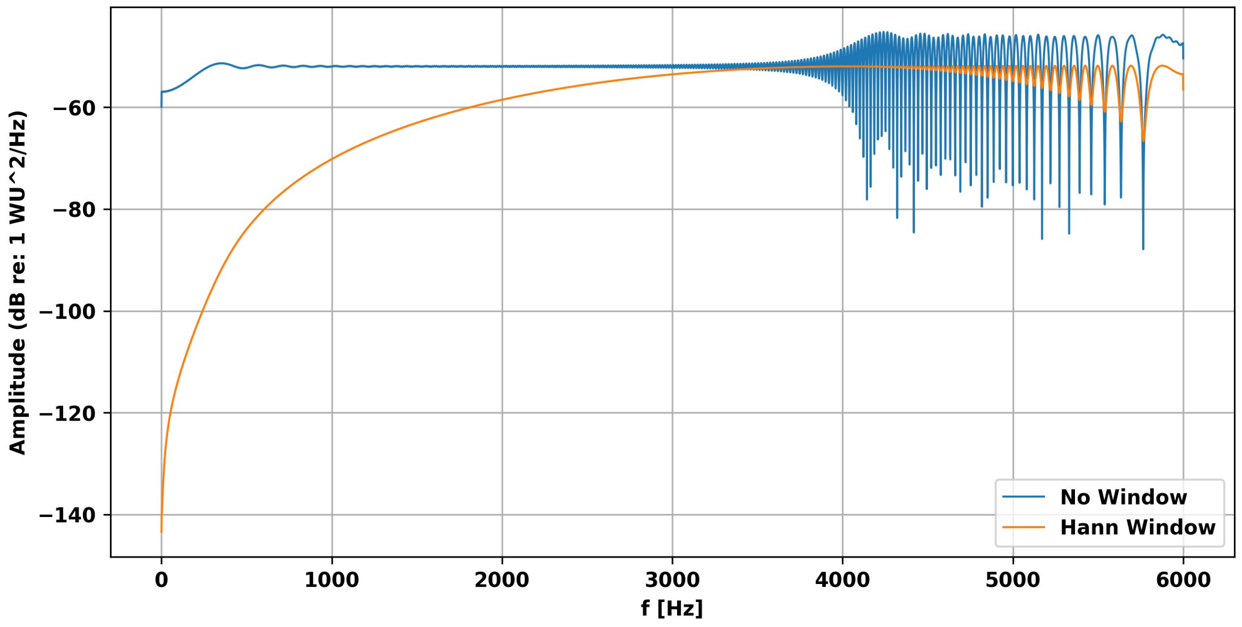


Figure 3: Linear Sine Sweep with f2 = 8000 Hz



Concluding Remarks

The first major take away of the assignment is that one can learn attributes about a system by studying how it responds to a variety of input signals. Some of the input signals are better than others when trying to study particular aspects of a system, depending on whether it is magnitude or phase related information one is after. Generally speaking, a linear sine sweep would be the best to use to learn both phase and magnitude information about the system;

however, the PSD at high frequencies can be highly variable. The second major take away is that the use of a Hanning window can help reduce variability in the PSD of a signal, making it easier to analyze a PSD at certain frequencies.

tzavelis_hw5

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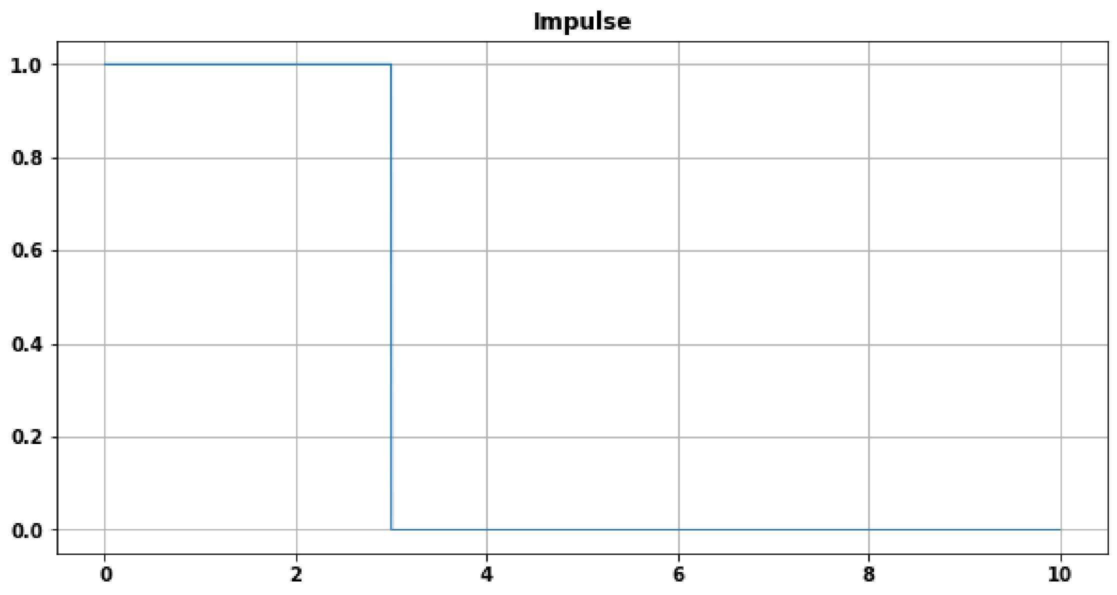
[2]:
```
import numpy as np
import pandas as pd
from scipy import signal
import soundfile as sf
import simpleaudio as sa
import sounddevice as sd
from scipy.io import wavfile
import matplotlib.pyplot as plt
%matplotlib inline
import seaborn as sns

plt.rcParams['font.weight'] = 'bold'
plt.rcParams['axes.labelweight'] = 'bold'
plt.rcParams['lines.linewidth'] = 1
plt.rcParams['axes.titleweight'] = 'bold'

class SignalTB:
    """
        My signal toolbox (SignalTB)!
    """

    def __init__(self, x, fs):
        """
        Arguments:
            x: Time Series
            fs: Sample Frequency
        """
        self.fs = fs; # [hz]
        self.x = x    # time domain series
        self.X = None # frequency domain series
        self.sxx = None
        self.gxx = None
        self.gxx_rms_a = None
        self.gxx_linear_a = None
        self.signals = [self.x] #useful container

        self.N = self.x.shape[0]   # number of samples
        self.L = self.x.index[-1] - self.x.index[0]   # total time of signal [s]
        self.dt = self.L/self.N  # [s]
        self.df = self.fs/self.N

    def get_signals():
        return filter(lambda x: x is not None, [self.X, self.x, self.sxx, self.
↪gxx])

    def my_fft(self):
        """
        Description:
            This method calculates the fft of a time domain signal using␣
↪numpy's fft function and
        adjusting it appropriately to multiplies it by dt.

        Returns:
            Series of frequency domain signal
        """
        freq = np.arange(-np.ceil(self.N/2)+1,
                         np.floor(self.N/2)+1) * self.df
        X = np.fft.fft(a=self.x.values, n=None, axis=-1, norm=None) * self.dt
        X = np.concatenate((X[int(np.floor(self.N/2))+1:],
                            X[0:int(np.floor(self.N/2))+1])) # rearrange the␣
↪frequencies from standard form to sequential. Remember that 1:self.N//2 does␣
↪not grab that second index value
        X = pd.Series(data=X,
                      index=freq,
                      name='X')
        self.X = X
        SignalTB.parseval_thrm(self.x,self.X,self.df,self.dt)  #check Parsevals␣
↪thrm
        self.signals.append(self.X)
        return X

    @staticmethod
    def my_ifft(X, N, df, dt):
        """
        Description:
            This method calculates the ifft of a time domain signal using␣
↪numpy's ifft function and
        adjusting it appropriately to multiplies it by dt.

        Returns:
            Series of frequency domain signal
        """
        t = np.linspace(start=0, stop=N*dt, num=N, endpoint=True)
        X = X.values # these are in sequential, non standard form
        X = np.concatenate((X[int(np.ceil(N/2))-1:],
                            X[0:int(np.ceil(N/2))-1])) #put the fft values in␣
↪standard form so ifft can accept it
        x = np.fft.ifft(a=X, n=None, axis=-1, norm=None) / dt
        SignalTB.parseval_thrm(x,X,df,dt)  #check Parsevals thrm
        return pd.Series(data=x,
                         index=t,
                         name='x2')

    @staticmethod
    def parseval_thrm(x, X, df, dt):
        """
        Description:
             Checks to make sure Parseval's Theorem holds between a time domain␣
↪and FFT holds true

        Arguments:
            x: time domain signal
            X: frequency domain signal
        """
        x = np.absolute(x) #get rid of complex numbers to do calc
        X = np.absolute(X)
        td = round((x**2).sum() * dt, 1)
        fd = round((X**2).sum() * df, 1)
        assert td == fd , "Parseval Theorem not satisfied: {} != {}".
↪format(td,fd)

    def sd(self):
        """
        Descrition:
            Spectral Density
        """
        sxx = np.abs(self.X)**2 / self.L; sxx.name = 'S_xx'; #display('sxx',sxx)
        # mean squared check
        X_ms = round(1/self.L * np.sum(np.abs(self.X)**2)*self.df,1)
        sxx_ms = round(np.sum(sxx)*self.df,1)
        assert X_ms == sxx_ms, 'Mean Squared Value Error: {} != {}'.
↪format(X_ms,sxx_ms)
        self.sxx = sxx
        self.signals.append(self.sxx)

        #gxx
        freq = np.arange(0, np.floor(self.N/2)+1) * self.df;␣
↪#display('freq',freq)
        i_zero = int(np.ceil(self.N/2)-1); #display('i_zero',i_zero)
        X = self.sxx.values[i_zero:] * 2 #grab from the center value all the␣
↪way to the end and double it
        X[0] = X[0]/2
        if self.N%2 == 0: X[-1] = X[-1]/2 #even
        gxx = pd.Series(data=X,
                        index=freq,
                        name='G_xx')

        # mean squared check
        gxx_ms = round(np.sum(gxx) * self.df,1)
        assert sxx_ms == gxx_ms, 'Mean Squared Value Error: {} != {}'.
↪format(sxx_ms,gxx_ms)
        self.gxx = gxx # uts of db
        self.signals.append(self.gxx)
        return self.sxx, self.gxx

    def rms_a(self, n_intervals = 16):
        """
            RMS Averaging for Gxx
        """
        frames=[]
        for i in range(1,n_intervals+1):
            x = self.x.iloc[int((i-1)*self.N/n_intervals):int(i*self.N/
↪n_intervals)]
            m = SignalTB(x=x, fs=self.fs)
            m.my_fft(); #calc fft
            m.sd() #calculate sxx and gxx
            frames.append(m.gxx) #save each gxx for averaging
        assert len(frames) == n_intervals, 'Could not perfectly cut the number␣
↪of samples by the n_interval: {}'.format(n_intervals)
        gxx_rms_a = pd.concat(frames,axis='columns').mean(axis='columns') #␣
↪calculates the mean of at each row (frequency)
        gxx_rms_a.name = 'G_xx_rms_a'
        self.gxx_rms_a = gxx_rms_a
        self.signals.append(gxx_rms_a)
        return gxx_rms_a

    def linear_a(self, n_intervals = 16):
        """
            Linear Averaging for X, then calculation of Gxx
        """
        frames=[]
        for i in range(1,n_intervals+1):
            x = self.x.iloc[int((i-1)*self.N/n_intervals):int(i*self.N/
↪n_intervals)]
            m = SignalTB(x=x, fs=self.fs)
            m.my_fft(); #calc fft
            frames.append(m.X) #save the fft
        assert len(frames) == n_intervals, 'Could not perfectly cut the number␣
↪of samples by the n_interval: {}'.format(n_intervals)
        X_a = pd.concat(frames,axis='columns').mean(axis='columns') #average␣
↪all the X's at each frequency

        #generate a temporary object so that you can perform computations
        m = SignalTB(x=self.x.iloc[int((i-1)*self.N/n_intervals):int(i*self.N/
↪n_intervals)], fs=self.fs) # the time signal passed in doesn't mean␣
↪anything, its just necessary to instatiate object
        m.X = X_a #set the new averaged X_a as the frequency domain signal in␣
↪the temporary object
        m.sd()
        m.gxx.name = 'G_xx_linear_a'
        self.gxx_linear_a = m.gxx
        self.signals.append(m.gxx)
        return m.gxx

    def time_a(self, n_intervals = 16):
        """
            Time Averaging for x, then calculation of Gxx
        """
        frames=[]
        for i in range(1,n_intervals+1):
            x = self.x.iloc[int((i-1)*self.N/n_intervals):int(i*self.N/
↪n_intervals)]
            x = pd.Series(data=x.values,
                          index=self.x.index[0:int(self.N/n_intervals)]) # make␣
↪sure that all the objects have the same time index. This is important for␣
↪taking the average and when we instatiate a new object.
            frames.append(x) #save the fft
        assert len(frames) == n_intervals, 'Could not perfectly cut the number␣
↪of samples by the n_interval: {}'.format(n_intervals)
        x_a = pd.concat(frames,axis='columns').mean(axis='columns');␣
↪#display(x_a);

        m = SignalTB(x=x_a, fs=self.fs) #generate a temporary object
        m.my_fft()
        m.sd()
        m.gxx.name = 'G_xx_time_a'
        self.gxx_time_a = m.gxx
        self.signals.append(m.gxx)
        return m.gxx

    def spectrogram(self, n_intervals = 16, overlap = 0.25):
        """
            Spectrogram!
        """
        p_size = int(np.floor(self.N/n_intervals));#print('psize: {}'.
↪format(p_size));print('n_intervals*p_size: {}'.format(n_intervals*p_size))
        x = self.x.iloc[0:int(n_intervals*p_size)]

        frames=[]
        for i in range(1,n_intervals+1):
            if i == 1:
                f = 0
                l = p_size
            else:
                f = l - int(np.floor(overlap*p_size))
                l = f + p_size
            sig = x.iloc[f:l]
            m = SignalTB(x=sig, fs=self.fs)
            m.my_fft();
            m.sd()
            r = (int((i-1)*self.N/n_intervals) + int(i*self.N/n_intervals))/2
            m.gxx.name = round(r*m.dt,2)  # name the slice at the middle
            frames.append(m.gxx)
        assert len(frames) == n_intervals, 'Could not perfectly cut the number␣
↪of samples by the n_interval: {}'.format(n_intervals)
        gxx_df = pd.concat(frames,axis='columns').sort_index(ascending=False)
        gxx_df.name = 'Gxx_spectro'

        gxx_df.index = gxx_df.index.values.round(decimals=1)
        self.gxx_df = gxx_df
        self.signals.append(self.gxx_df)

        return gxx_df

    def plot_signals(self, xrange=None):
        """
        Description:
            Plots all of the signals in the self.signals container

        Returns:
            Nothing
        """

        for i, sig in enumerate(self.signals):
            if type(sig) != pd.DataFrame:
                if sig.dtype == complex: sig = np.absolute(sig) # ALWAYS the␣
↪magnitude of this in case its a complex number
            fig = plt.figure(figsize=(10,5))
            plt.title(sig.name)
            if sig.name in ['x','x2','time domain signal']:
                plt.ylabel('x(t)'); plt.xlabel('t [s]')
            elif sig.name in␣
↪['X','S_xx','G_xx','G_xx_rms_a','G_xx_linear_a','G_xx_time_a']:
                sig = 10*np.log10(sig); plt.ylabel('X(f)'); plt.xlabel('f␣
↪[Hz]'); #plt.ylim([-30:])
            elif sig.name in ['Gxx_spectro']:
                sns.heatmap(sig, cmap="jet"); plt.ylabel('f [Hz]'); plt.
↪xlabel('t [s]')
                continue
            if xrange != None:
                sig[xrange[0]:xrange[1]].plot();
            else:
                sig.plot();
            plt.grid()

    #Useful functions to generate signals
    @staticmethod
    def impulse(N,L,Tp):
        x = np.zeros(N)
        x[0:int(N*Tp/L)] = 1
        x = pd.Series(data=x,
                      index=np.linspace(start=0, stop=L, num=N, endpoint=True,␣
↪dtype=float),
                      name='x')
        return x

    @staticmethod
    def cw_sin(A,f,N,L,Tp):
        """
        Arguments:
            A: Amplitude
            f: Frequency of signal [hz]
            L: Total length of time [s]
            N: Number of points
            Tp: Pulse length

        Returns:
            Series
        """
        t = np.linspace(start=0, stop=L, num=N, endpoint=True, dtype=float)
        x = np.zeros(N)

        t_s = t[0:int(N*Tp/L)];
        x[0:int(N*Tp/L)] = A*np.sin(2*np.pi*f*t_s)
        return pd.Series(data=x,
                         index=t,
                         name='x')

    @staticmethod
    def lin_sin_s(A,N,L,Tp,f1,f2):
        """
        Arguments:
            A: Amplitude
            f: Frequency of signal [hz]
            L: Total length of time [s]
            N: Number of points
            Tp: Pulse length

        Returns:
            Series
        """
        t = np.linspace(start=0, stop=L, num=N, endpoint=True, dtype=float)
        x = np.zeros(N)

        t_s = t[0:int(N*Tp/L)];
        phi = 2*np.pi*( (f2-f1)/(2*Tp)*t_s**2+f1*t_s)
        x[0:int(N*Tp/L)] = A*np.sin(phi)

        return pd.Series(data=x,
                         index=t,
                         name='x')

    @staticmethod
    def log_sin_s(A,N,L,Tp,f1,f2):
        """
        Arguments:
            A: Amplitude
            L: Total length of time [s]
            N: Number of points
            Tp: Pulse length

        Returns:
            Series
        """
        t = np.linspace(start=0, stop=L, num=N, endpoint=True, dtype=float)
        x = np.zeros(N)

        t_s = t[0:int(N*Tp/L)];
        phi = 2*np.pi*f1*(f2/f1)**(t_s/Tp)*Tp/(np.log(f2/f1))-2*np.pi*f1*Tp/np.
↪log(f2/f1)
        x[0:int(N*Tp/L)] =  A*np.sin(phi)
        x = pd.Series(data=x,
                      index=t,
                      name='x')
        return x

    @staticmethod
    def white_noise(N,L,Tp):
        """
        Arguments:
            L : Total length of time [s]
            N : Number of points

        Returns:
            Series
        """
        fs = N/L
        df = fs/N

        p = np.random.uniform(0,2*np.pi,int(N/2)-1) #random reals
        Z = 1*np.sin(p) + 1*np.cos(p)*1j #random complex numbers with magnitude␣
↪of 1

        Z = np.concatenate((Z, np.array([0]), np.flip(np.conj(Z))))

        if N%2 == 0: Z = np.concatenate((Z, np.array([0])))

        freq = np.arange(-np.ceil(N/2)+1, np.floor(N/2)+1) * df
        X = pd.Series(data=Z,index=freq); #construct the full freq response
        x = SignalTB.my_ifft(X=X, N=N, df=df, dt=dt) # get the full time noise

        y = np.zeros(N,dtype=complex)
        y[0:int(N*Tp/L)] = x.values[0:int(N*Tp/L)] # take only a subset of the␣
↪noisey signal and put it infront of all the zeros

        return pd.Series(data=np.real(y),
                         index=x.index,
                         name='x')

    @staticmethod
    def sin(A,f,L,N):
        """
        Arguments:
            A: Amplitude
            f: Frequency of signal [hz]
            L: Total length of time [s]
            N: Number of points
        Returns:
            Series
        """
        t = np.linspace(start=0, stop=L, num=N, endpoint=True, dtype=float)
        return pd.Series(data=A*np.sin(2*np.pi*f*t),
                         index=t,
                         name='x')

    @staticmethod
    def randn_sig(L,N):
        """
        Arguments:
            L : Total length of time [s]
            N : Number of points

        Returns:
            Series
        """
        return pd.Series(data=np.random.randn(N,),
                         index=np.linspace(start=0, stop=L, num=N,␣
↪endpoint=True),
                         name='x')
    @staticmethod
    def csd(s0,s1):
        """
        Descrition:
            Cross Spectral Density
        """
        #calculate the fft of the objects
        s0.my_fft(); s1.my_fft()

        #sxy
        sxy = np.conj(s0.X)*s1.X / s0.L; sxy.name = 'S_xy';

        #gxy
        freq = np.arange(0, np.floor(s0.N/2)+1) * s0.df; #display('freq',freq)
        i_zero = int(np.ceil(s0.N/2)-1); #display('i_zero',i_zero)

        X = sxy.values[i_zero:] * 2 #grab from the center value all the way to␣
↪the end and double it
        X[0] = X[0]/2
        if s0.N%2 == 0: X[-1] = X[-1]/2 #even
        gxy = pd.Series(data=X,
                        index=freq,
                        name='G_xy')
        return sxy, gxy
    @staticmethod
    def cross_corr(s0,s1):
        """
        Description:
            Cross correlation F^-1(Sxy)
        """
        sxy, gxy = SignalTB.csd(s0,s1)
        X = sxy.values # these are in sequential, non standard form
        X = np.concatenate((X[int(np.ceil(s0.N/2))-1:],
                            X[0:int(np.ceil(s0.N/2))-1])) #put the fft values␣
↪in standard form so ifft can accept it
        x = np.fft.ifft(a=X, n=None, axis=-1, norm=None) / s0.dt
        SignalTB.parseval_thrm(x,X, s0.df, s0.dt)  #check Parsevals thrm
        x = np.concatenate((x[int(np.floor(s0.N/2))+1:],
                            x[0:int(np.floor(s0.N/2))+1])) #put the fft values␣
↪in standard form so ifft can accept it
        t = np.arange(-np.ceil(s0.N/2)+1,np.floor(s0.N/2)+1) * s0.dt

        cross_corr = pd.Series(data=x,
                               index=t,
                               name='Cross Correlation')
        return cross_corr, sxy, gxy
```

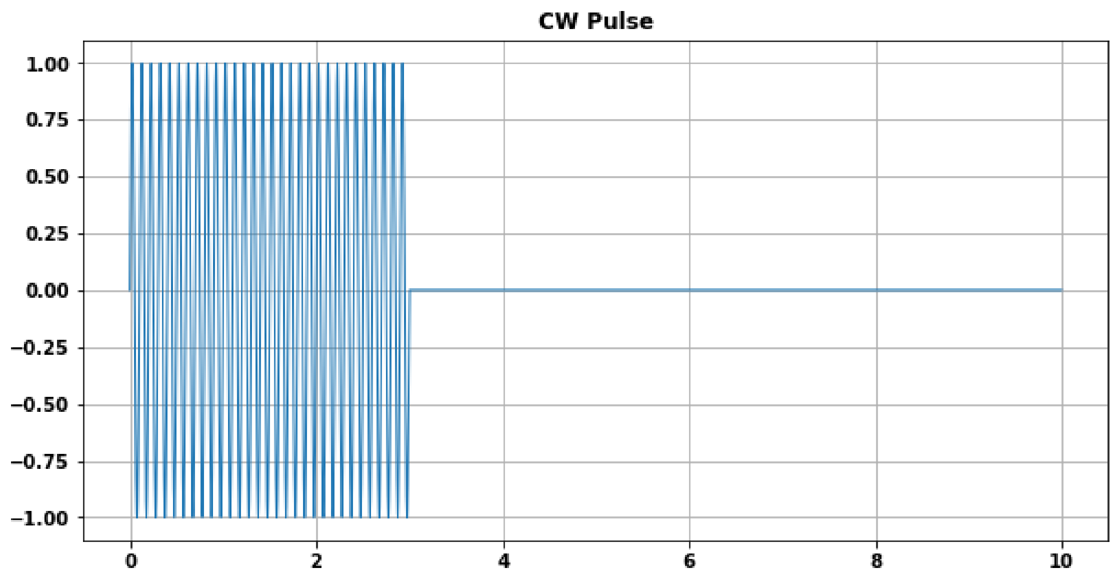
[3]:
```
# QUESTION 1
L = 10 # [s]
fs = 1024 #[hz]
N = int(L/(1/fs));  # generate the number of points based on the sampling␣
↪frequency which is higher than the actual signal frequency
f_sin = 400;
df = fs/N
dt = L/N

Tp = 3
f1 = 1
f2 = 200

x, Z = SignalTB.white_noise(N,L,Tp)
#plt.plot(np.angle(Z))
plt.plot(x)
```

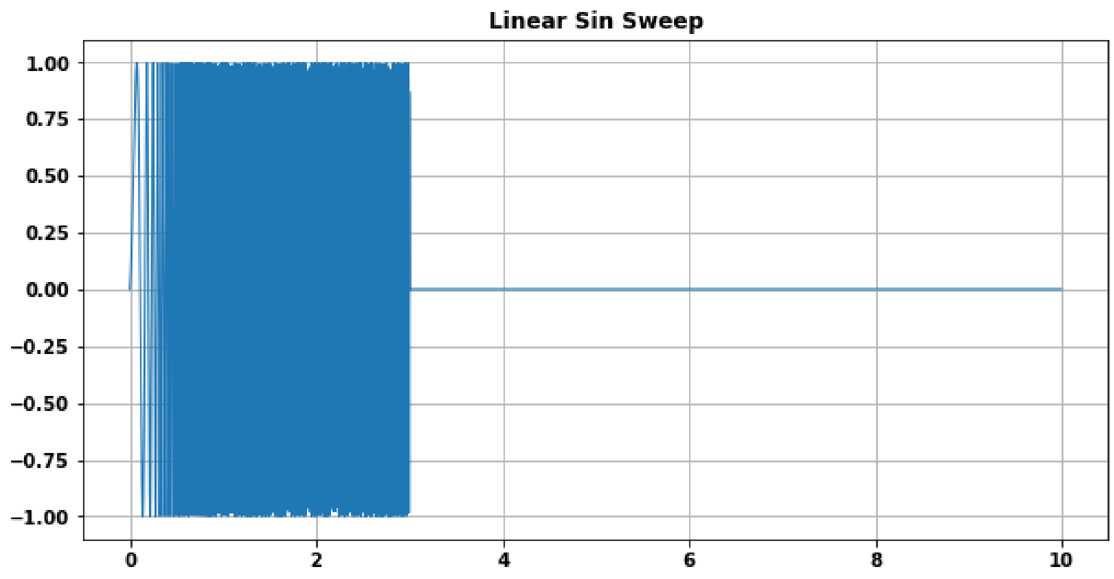
```
      ␣
↪---------------------------------------------------------------------------

       ValueError                                Traceback (most recent call␣
↪last)
```

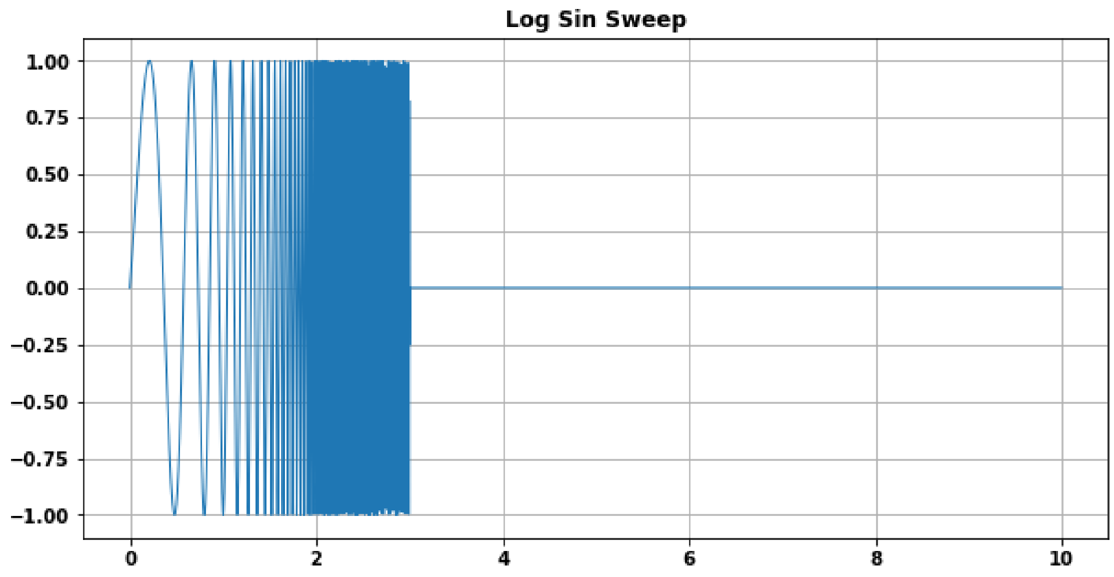
```
    <ipython-input-3-0afc105afb8d> in <module>
     11 f2 = 200
     12
---> 13 x, Z = SignalTB.white_noise(N,L,Tp)
     14 #plt.plot(np.angle(Z))
     15 plt.plot(x)


    ValueError: too many values to unpack (expected 2)
```

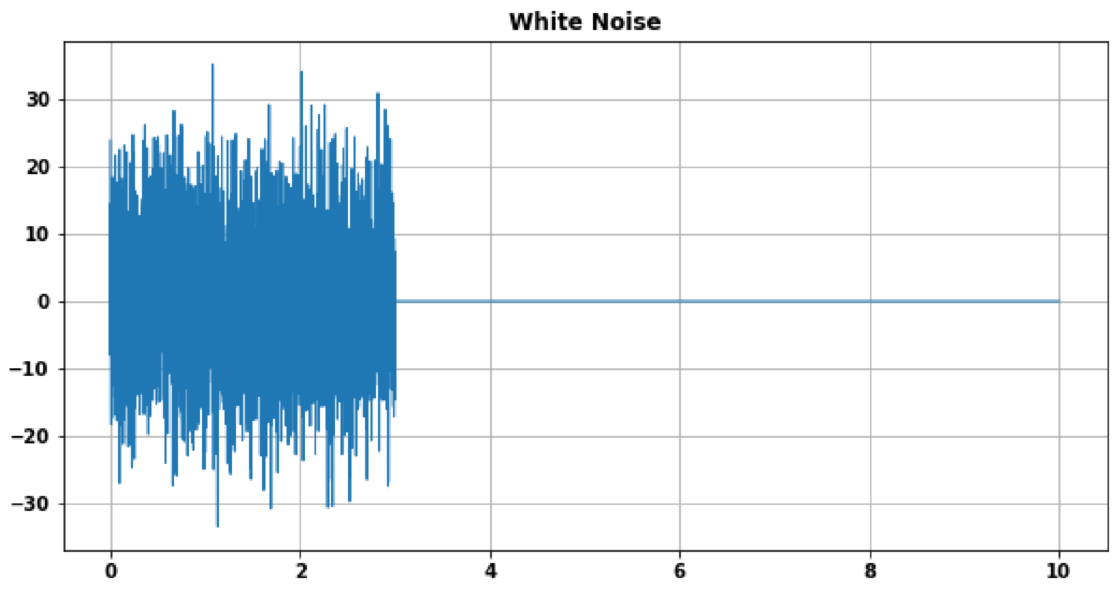
[11]:
```
# QUESTION 1
L = 10 # [s]
fs = 1024 #[hz]
N = int(L/(1/fs));  # generate the number of points based on the sampling␣
 ↪frequency which is higher than the actual signal frequency
f_sin = 400;
df = fs/N
dt = L/N

Tp = 3
f1 = 1
f2 = 200

sigs = [SignalTB.impulse(N=N,L=L,Tp=Tp),
        SignalTB.cw_sin(A=1,f=10, N=N,L=L,Tp=Tp),
        SignalTB.lin_sin_s(A=1,N=N,L=L,Tp=Tp,f1=f1,f2=f2),
        SignalTB.log_sin_s(A=1,N=N,L=L,Tp=Tp,f1=f1,f2=f2),
        SignalTB.white_noise(N=N,L=L,Tp=Tp)]

signals = [SignalTB(x=x, fs=fs) for x in sigs]

titles = ['Impulse','CW Pulse', 'Linear Sin Sweep', 'Log Sin Sweep', 'White␣
 ↪Noise']
#q1
for i, s in enumerate(signals):
    plt.figure(figsize=(10,5))
    plt.title(titles[i])
    s.x.plot(); plt.grid()
    #s.my_fft()
    #plt.plot(np.abs(SignalTB.my_ifft(s.X, N, df, dt)),'--')
```


Impulse
1.0
0.8
0.6
0.4
0.2
0.0
0
2
4
6
8
10


CW Pulse
1.00
0.75
0.50
0.25
0.00
−0.25
−0.50
−0.75
−1.00
0
2
4
6
8
10


Linear Sin Sweep
1.00
0.75
0.50
0.25
0.00
−0.25
−0.50
−0.75
−1.00
0
2
4
6
8
10

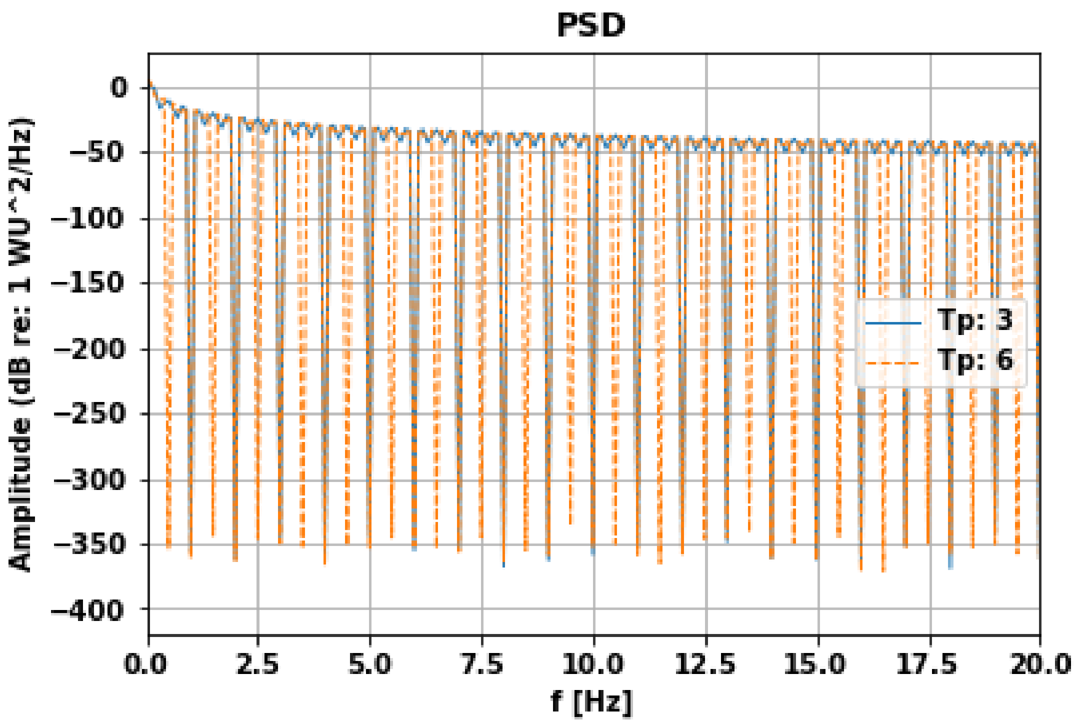

Log Sin Sweep
1.00
0.75
0.50
0.25
0.00
−0.25
−0.50
−0.75
−1.00
0
2
4
6
8
10



[37]:

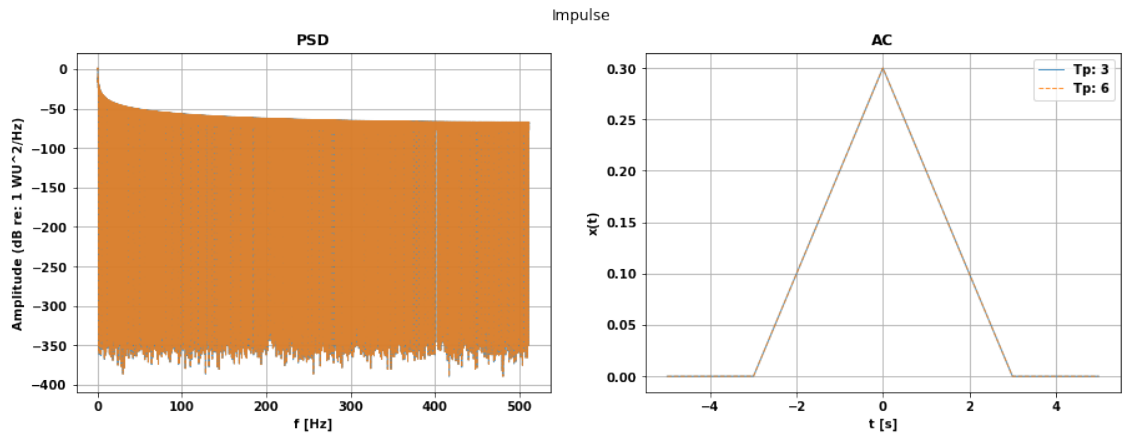
```
# QUESTION 1
L = 10 # [s]
fs = 1024 #[hz]
N = int(L/(1/fs));  # generate the number of points based on the sampling␣
 ↪frequency which is higher than the actual signal frequency
f_sin = 400;
df = fs/N
dt = L/N

Tp = 3
f1 = 1
f2 = 200

x = SignalTB.impulse(L=L,N=N,Tp=Tp)
s = SignalTB(x=x,fs=fs)
s.my_fft() #must calc fft to do sd
#power spectral density
sxx, gxx = s.sd()

fig = plt.figure()
plt.plot(10*np.log10(np.abs(gxx)));

Tp = 6
f1 = 1
f2 = 200

x = SignalTB.impulse(L=L,N=N,Tp=Tp)
```

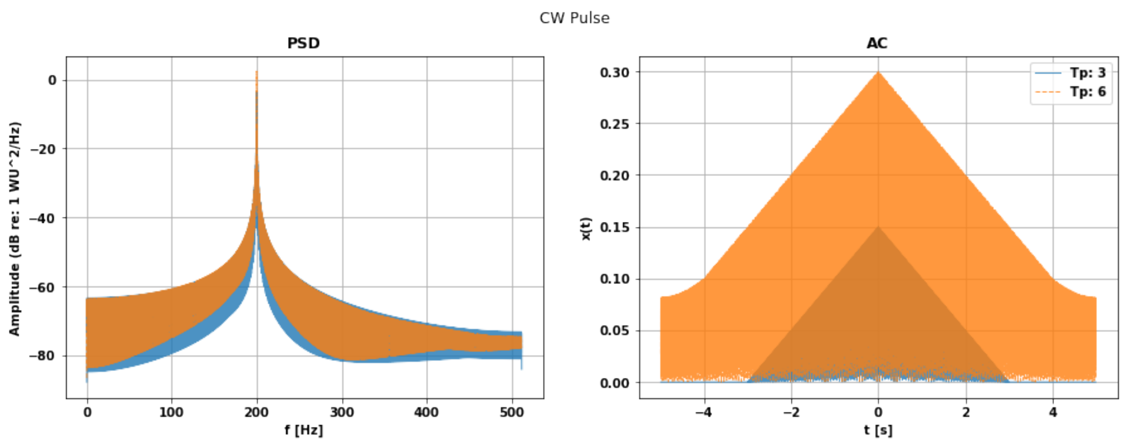
```
s = SignalTB(x=x,fs=fs)
s.my_fft() #must calc fft to do sd
#power spectral density
sxx, gxx = s.sd()

plt.plot(10*np.log10(np.abs(gxx)),'--'); plt.xlim([0,20])
plt.xlim([0,20])
plt.title('PSD'); plt.ylabel('Amplitude (dB re: 1 WU^2/Hz)'); plt.xlabel('f␣
 ↪[Hz]'); plt.grid(True)
plt.legend(['Tp: {}'.format(times[0]),'Tp: {}'.format(times[1])])
fig.savefig('./plots/hw5_'+'zoomed_impulse'+'.png', dpi=300,␣
 ↪bbox_inches='tight')
```

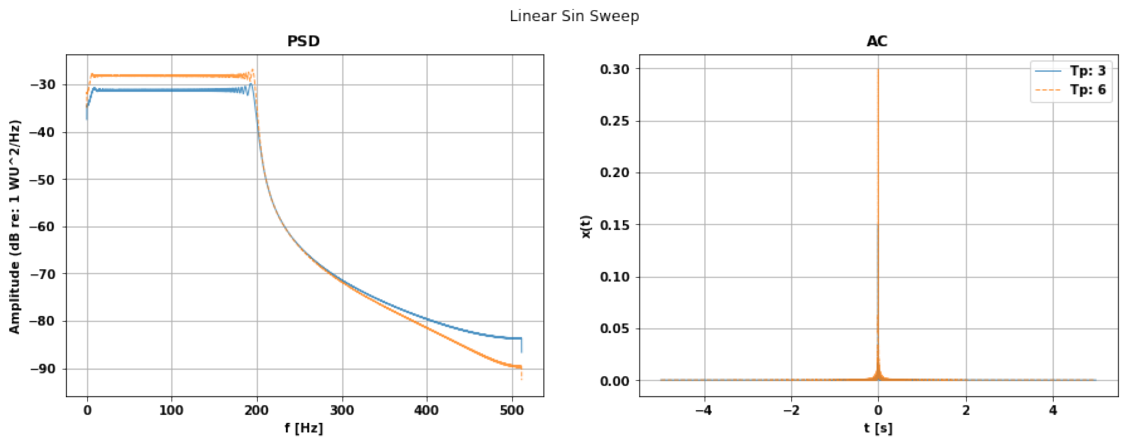


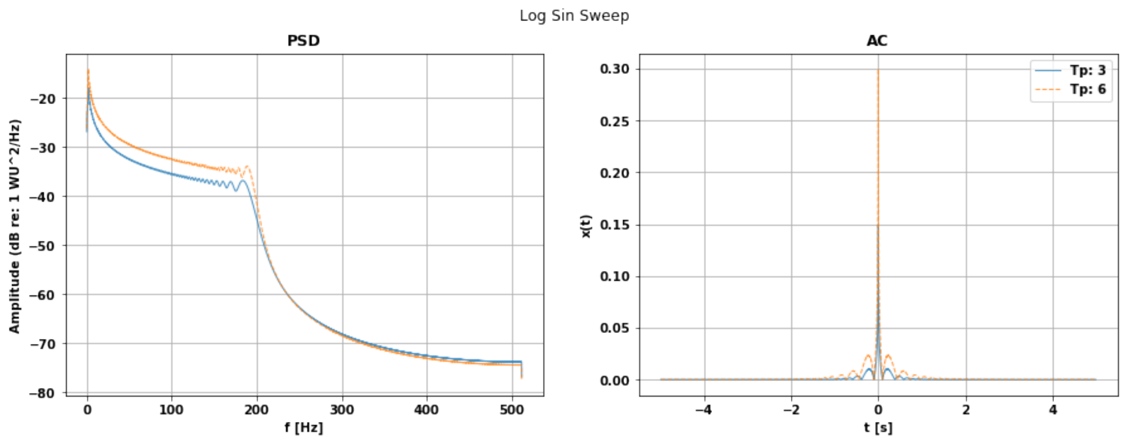
[12]:
```
#QUESTION 2
titles = ['Impulse','CW Pulse', 'Linear Sin Sweep', 'Log Sin Sweep', 'White␣
 ↪Noise']
linestyles = ['-', '--']
for i,t in enumerate(titles):
    fig, axs = plt.subplots(1,2,figsize=(15,5));plt.grid()
    fig.suptitle(t)

    times = [Tp,2*Tp]
    for j, length in enumerate(times):
        sigs = [SignalTB.impulse(L=L,N=N,Tp=Tp),
                SignalTB.cw_sin(A=1,f=f2, N=N,L=L,Tp=length),
                SignalTB.lin_sin_s(A=1,N=N,L=L,Tp=length,f1=f1,f2=f2),
                SignalTB.log_sin_s(A=1,N=N,L=L,Tp=length,f1=f1,f2=f2),
                SignalTB.white_noise(N=N,L=L,Tp=Tp),
               ]
        signals = [SignalTB(x=x, fs=fs) for x in sigs]

        s = signals[i] #select the actual signal we care about
        s.my_fft() #must calc fft to do sd

        #power spectral density
        sxx, gxx = s.sd()
        axs[0].plot(10*np.log10(np.abs(gxx)),alpha=0.8,linestyle=linestyles[j]);
        axs[0].set_title('PSD'); axs[0].set_ylabel('Amplitude (dB re: 1 WU^2/
↪Hz)'); axs[0].set_xlabel('f [Hz]'); axs[0].grid(True)

        #autocorrelation
        cross_corr, sxy, gxy = SignalTB.cross_corr(s,s)
        axs[1].plot(np.abs(cross_corr),alpha=0.8,linestyle=linestyles[j]);
        axs[1].set_title('AC'); axs[1].set_ylabel('x(t)'); axs[1].set_xlabel('t␣
↪[s]'); axs[1].grid(True);
    plt.legend(['Tp: {}'.format(times[0]),'Tp: {}'.format(times[1])])
    fig.savefig('./plots/hw5_'+t+'.png', dpi=300, bbox_inches='tight');
```

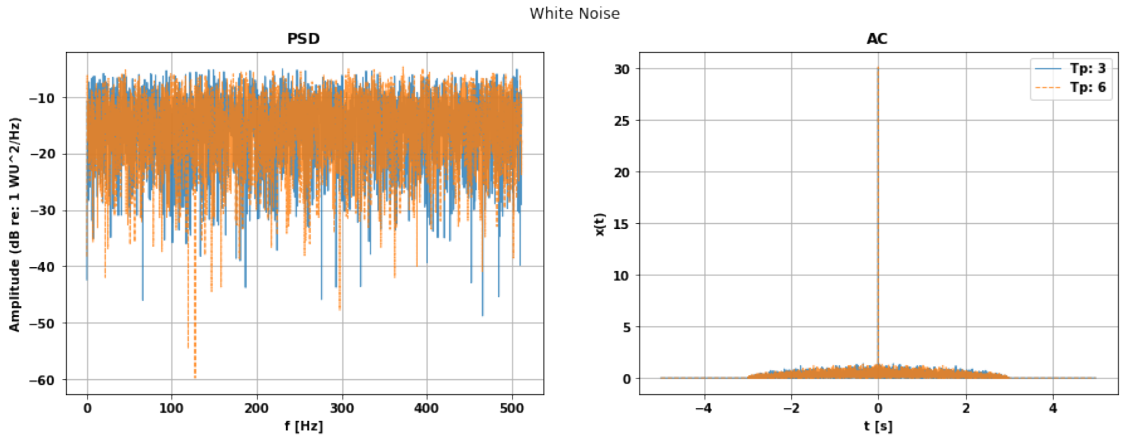
```
/home/m4rz910/anaconda3/lib/python3.7/site-packages/pandas/core/series.py:679:
RuntimeWarning: divide by zero encountered in log10
  result = getattr(ufunc, method)(*inputs, **kwargs)
```



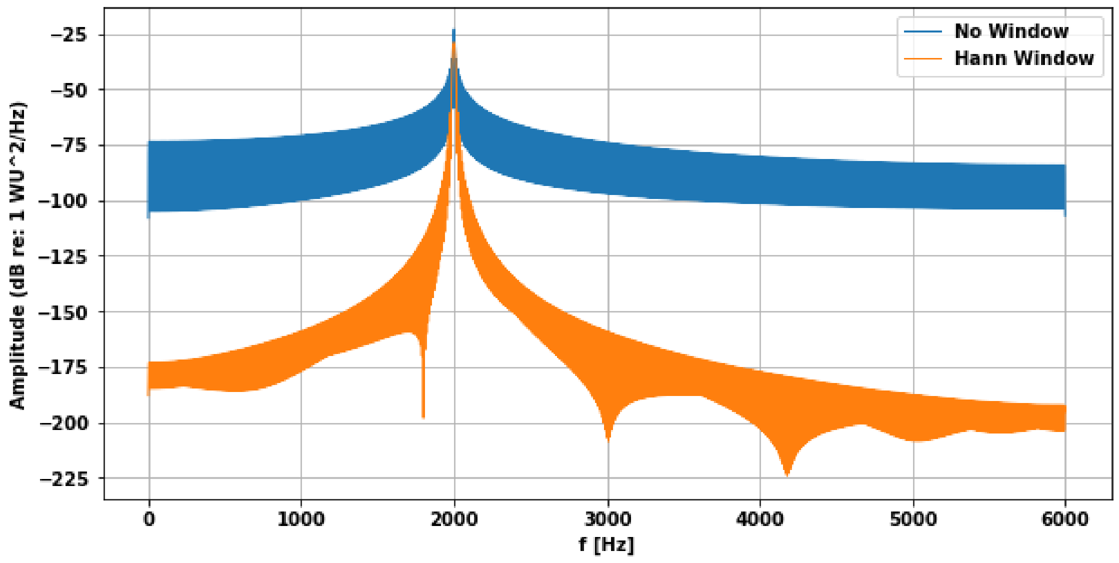

CW Pulse
PSD
Amplitude (dB re: 1 WU^2/Hz)
f [Hz]
AC
x(t)
t [s]
Tp: 3
Tp: 6


Linear Sin Sweep
PSD
Amplitude (dB re: 1 WU^2/Hz)
f [Hz]
AC
x(t)
t [s]
Tp: 3
Tp: 6


Log Sin Sweep
PSD
Amplitude (dB re: 1 WU^2/Hz)
f [Hz]
AC
x(t)
t [s]
Tp: 3
Tp: 6

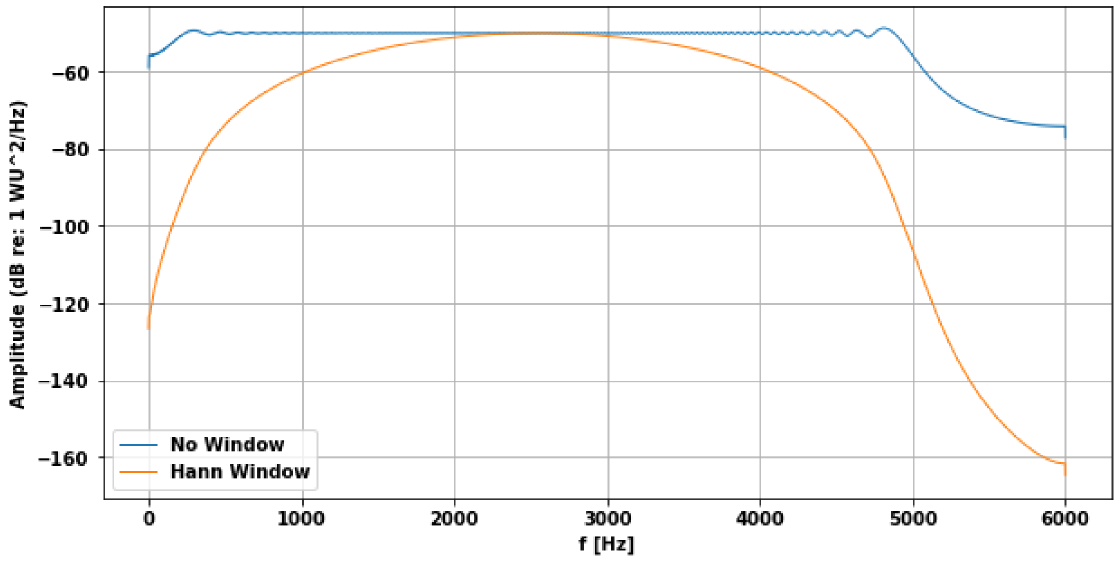


[40]:
```
# 3. TAPER AND WINDOW
L = 1
Tp=0.1
fs=12000
f=2000
N = int(L/(1/fs));

fig = plt.figure(figsize=(10,5));plt.grid()
#without  window
x = SignalTB.cw_sin(A=1,f=f,N=N,L=L,Tp=Tp)
s0 = SignalTB(x,fs)
s0.my_fft();
s0.sd();
#plt.plot(s0.x); #plt.xlim([0,.01])
plt.plot(10*np.log10(np.abs(s0.gxx)))

#with window
w = np.zeros(N)
w[0:int(N*Tp/L)] = np.hanning(int(N*Tp/L))

x = SignalTB.cw_sin(A=1,f=f,N=N,L=L,Tp=Tp)*w
s1 = SignalTB(x,fs)
s1.my_fft();
s1.sd();
plt.plot(10*np.log10(np.abs(s1.gxx)))
#plt.plot(s1.x);
#plt.xlim([0,5000])

plt.ylabel('Amplitude (dB re: 1 WU^2/Hz)'); plt.xlabel('f [Hz]');
plt.legend(['No Window', 'Hann Window']);
fig.savefig('./plots/hw5_'+'q3taper'+'.png', dpi=300, bbox_inches='tight');
```



[42]:
```
# 4. Sine Sweep
# 3. TAPER AND WINDOW
L = 1
Tp=0.1
fs=12000
f1=100
f2=5000
N = int(L/(1/fs));

fig = plt.figure(figsize=(10,5));plt.grid()
#without  window
x=SignalTB.lin_sin_s(A=1,N=N,L=L,Tp=Tp,f1=f1,f2=f2)
#x=SignalTB.log_sin_s(A=1,N=N,L=L,Tp=Tp,f1=f1,f2=f2)
#x = SignalTB.cw_sin(A=1,f=f,N=N,L=L,Tp=Tp)
s0 = SignalTB(x,fs)
s0.my_fft();
s0.sd();
#plt.plot(s0.x); #plt.xlim([0,.01])
plt.plot(10*np.log10(np.abs(s0.gxx)))

#with window
w = np.zeros(N)
w[0:int(N*Tp/L)] = np.hanning(int(N*Tp/L))

x=SignalTB.lin_sin_s(A=1,N=N,L=L,Tp=Tp,f1=f1,f2=f2)*w
#x=SignalTB.log_sin_s(A=1,N=N,L=L,Tp=Tp,f1=f1,f2=f2)*w
#x = SignalTB.cw_sin(A=1,f=f,N=N,L=L,Tp=Tp)*w
s1 = SignalTB(x,fs)
```

```
s1.my_fft();
s1.sd();
plt.plot(10*np.log10(np.abs(s1.gxx)))
#plt.plot(s1.x);
#plt.xlim([0,5000])

plt.ylabel('Amplitude (dB re: 1 WU^2/Hz)'); plt.xlabel('f [Hz]');
plt.legend(['No Window', 'Hann Window']);
fig.savefig('./plots/hw5_'+'q3taper'+'.png', dpi=300, bbox_inches='tight');
```


Amplitude (dB re: 1 WU^2/Hz)
f [Hz]
No Window
Hann Window