

Audio Filter Assignment

EE23BTECH11025 - ANANTHA KRISHNAN*

I. DIGITAL FILTER

- I.1 Download the sound file used for this code from the below link

<https://github.com/Gandubs/Signals-and-Systems/blob/master/audio%20filter/codes/Ananth-singing.wav>

- I.2 Listed below is the python code for removal of out of band noise.

```
import soundfile as sf
import numpy as np
from scipy import signal
#read .wav file
input_signal,fs = sf.read('Ananth-singing.
    wav')

#sampling frequency of Input signal
sampl_freq=fs

#order of the filter
order=4

#cutoff frequency
cutoff_freq=1000.0

#digital frequency
Wn=2*cutoff_freq/sampl_freq

# b and a are numerator and denominator
    polynomials respectively
b, a = signal.butter(order, Wn, 'low')
#print(fs)

#filter the input signal with butterworth filter
output_signal = signal.filtfilt(b, a,
    input_signal,padlen=1)
#output_signal = signal.lfilter(b, a,
    input_signal)

#write the output signal into .wav file
sf.write('Sound_With_ReducedNoise.wav',
    output_signal, fs)
```

- I.3 The audio file is now analyzed using a spectrogram from the website <https://academo.org/demos/spectrum-analyzer>.

Dark lines in a spectrogram often indicate areas of low intensity and bright lines represent areas of high intensity in the signal.

In the filtered image, there is a higher concentration of bright lines within the cutoff frequency($f_c = 1kHz$)

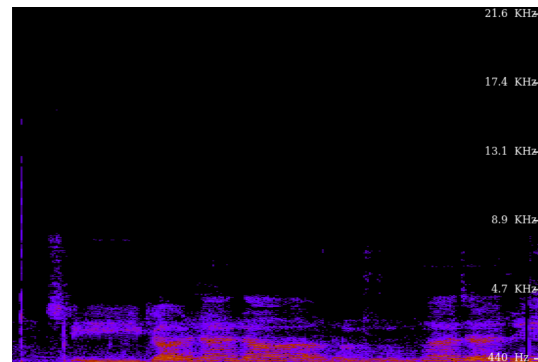


Fig. 1. Spectrogram of the audio file pre Filtering

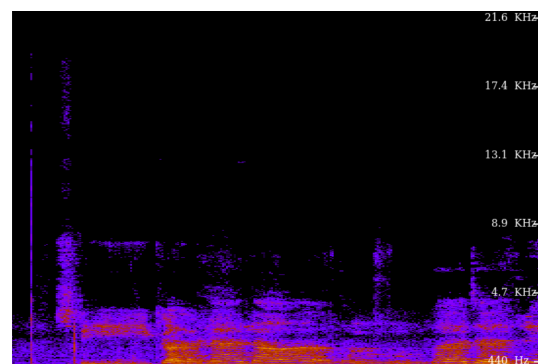


Fig. 2. Spectrogram of the audio file post Filtering

II. DIFFERENCE EQUATION

- II.1 Let

$$x(n) = \left\{ \underset{\uparrow}{1}, 2, 3, 4, 2, 1 \right\} \quad (1)$$

Sketch $x(n)$.

II.2 Let

$$y(n) + \frac{1}{2}y(n-1) = x(n) + x(n-2),$$

$$y(n) = 0, n < 0 \quad (2)$$

Sketch $y(n)$.

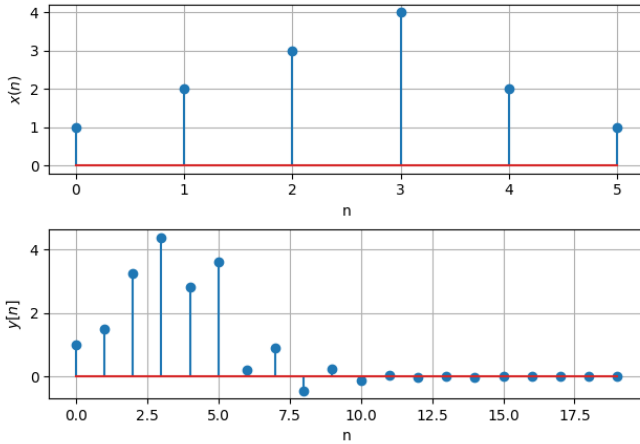
Solve

Solution: The below C code computes the values of $y(n)$ till $n = 20$

<https://github.com/Gandubs/Signals-and-Systems/blob/master/audio%20filter/codes/2.2.c>

The following code plots $x[n]$ and $y[n]$

<https://github.com/Gandubs/Signals-and-Systems/blob/master/audio%20filter/codes/2.2.py>



Solution: From (3)

$$a^n u(n) \xleftrightarrow{Z} \sum_{n=0}^{\infty} (az^{-1})^n \quad (19)$$

$$= \frac{1}{1 - az^{-1}} \quad \forall |z| > |a| \quad (20)$$

III.5 Let

$$H(e^{j\omega}) = H(z = e^{j\omega}). \quad (21)$$

Plot $|H(e^{j\omega})|$. Comment. $H(e^{j\omega})$ is known as the *Discrete Time Fourier Transform (DTFT)* of $x(n)$.

Solution: The below code plots $|H(e^{j\omega})|$.

<https://github.com/Gandubs/Signals-and-Systems/blob/master/audio%20filter/codes/3.5.py>

Substituting $z = e^{j\omega}$ in (11), we get

$$|H(e^{j\omega})| = \left| \frac{1 + e^{-2j\omega}}{1 + \frac{1}{2}e^{-j\omega}} \right| \quad (22)$$

$$= \sqrt{\frac{(1 + \cos(2\omega))^2 + (\sin(2\omega))^2}{\left(1 + \frac{1}{2}\cos(\omega)\right)^2 + \left(\frac{1}{2}\sin(\omega)\right)^2}} \quad (23)$$

$$= \frac{4|\cos(\omega)|}{\sqrt{5 + 4\cos(\omega)}} \quad (24)$$

$$\left| H(e^{j(\omega+2\pi)}) \right| = \frac{4|\cos(\omega + 2\pi)|}{\sqrt{5 + 4\cos(\omega + 2\pi)}} \quad (25)$$

$$= \frac{4|\cos(\omega)|}{\sqrt{5 + 4\cos(\omega)}} \quad (26)$$

$$= |H(e^{j\omega})| \quad (27)$$

Its fundamental period is 2π , which verifies that the DTFT of a signal is always periodic.

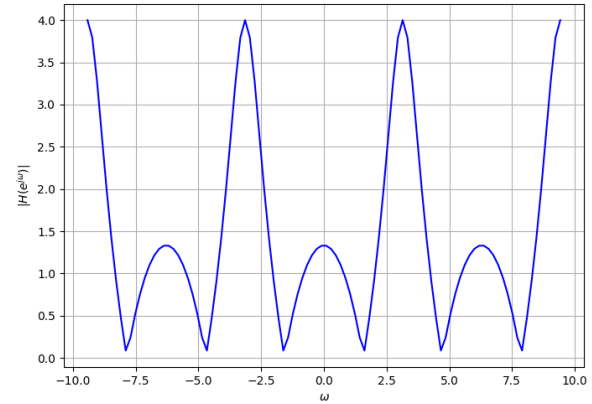


Fig. 4. $|H(e^{j\omega})|$

IV. IMPULSE RESPONSE

IV.1 Find an expression for $h(n)$ using $H(z)$, given that

$$h(n) \xleftrightarrow{Z} H(z) \quad (28)$$

and there is a one to one relationship between $h(n)$ and $H(z)$. $h(n)$ is known as the *impulse response* of the system defined by (2).

Solution: From (11),

$$H(z) = \frac{1}{1 + \frac{1}{2}z^{-1}} + \frac{z^{-2}}{1 + \frac{1}{2}z^{-1}}, \quad |z| > \frac{1}{2} \quad (29)$$

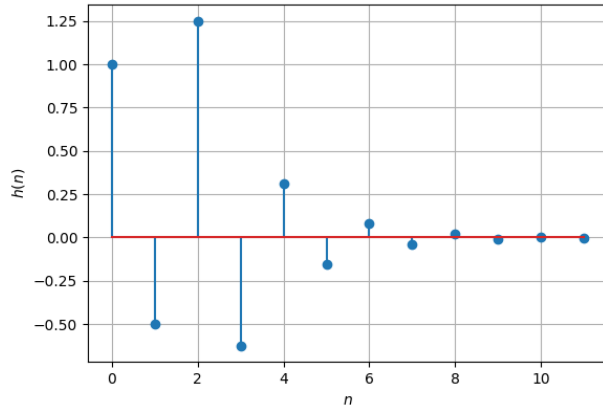
$$\Rightarrow h(n) = \left(-\frac{1}{2}\right)^n (u(n)) + \left(-\frac{1}{2}\right)^{n-2} (u(n-2)) \quad (30)$$

using (18) and (8).

IV.2 Sketch $h(n)$. Is it bounded? Convergent?

Solution: The following code plots $h(n)$

<https://github.com/Gandubs/Signals-and-Systems/blob/master/audio%20filter/codes/4.2.py>

Fig. 5. $h(n)$

IV.3 The system with $h(n)$ is defined to be stable if

$$\sum_{n=-\infty}^{\infty} h(n) < \infty \quad (31)$$

Is the system defined by (2) stable for the impulse response in (28)?

Solution: For a stable system, the R.O.C of $H(z)$ should contain the boundary of unit circle $|z| = 1$. From (29)

$$|z| = 1 \subset |z| > \frac{1}{2} \quad (32)$$

Therefore it converges and hence stable.

IV.4 Compute and sketch $h(n)$ using

$$h(n) + \frac{1}{2}h(n-1) = \delta(n) + \delta(n-2), \quad (33)$$

This is the definition of $h(n)$.

Solution:

Below is the plot of $h(n)$ using the difference equations. 5.

<https://github.com/Gandubs/Signals-and-Systems/blob/master/audio%20filter/codes/4.4.py>

Fig. 6. $h(n)$ from definition is same as Fig. 5

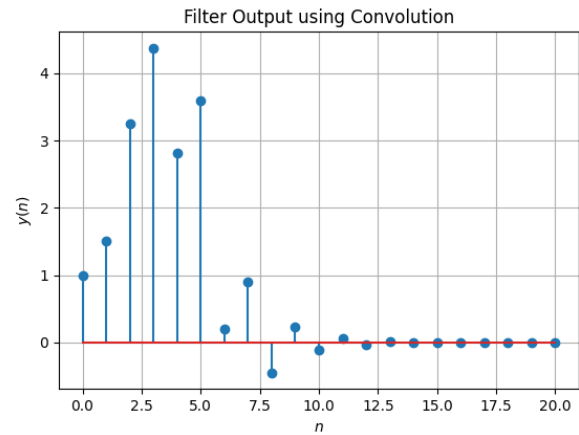
IV.5 Compute

$$y(n) = x(n) * h(n) = \sum_{n=-\infty}^{\infty} x(k)h(n-k) \quad (34)$$

Comment. The operation in (34) is known as *convolution*.

Solution: The following code plots Fig. 7 using convolution. 3.

<https://github.com/Gandubs/Signals-and-Systems/blob/master/audio%20filter/codes/4.5.py>

Fig. 7. $y(n)$ from the definition of convolution

IV.6 Show that

$$y(n) = \sum_{n=-\infty}^{\infty} x(n-k)h(k) \quad (35)$$

Solution: In (34), substituting $k \rightarrow n-k$, we get

$$y(n) = \sum_{n-k=-\infty}^{\infty} x(n-k)h(k) \quad (36)$$

$$= \sum_{k=-\infty}^{\infty} x(n-k)h(k) \quad (37)$$

V. DFT AND FFT

V.1 Compute

$$X(k) \triangleq \sum_{n=0}^{N-1} x(n)e^{-j2\pi kn/N}, \quad k = 0, 1, \dots, N-1 \quad (38)$$

and $H(k)$ using $h(n)$.

V.2 Compute

$$Y(k) = X(k)H(k) \quad (39)$$

V.3 Compute

$$y(n) = \frac{1}{N} \sum_{k=0}^{N-1} Y(k) \cdot e^{j2\pi kn/N}, \quad n = 0, 1, \dots, N-1 \quad (40)$$

Solution: The above three questions are solved using the following code :

https://github.com/Gandubs/Signals-and-Systems/blob/master/audio%20filter/codes/5_sol.py

V.4 Repeat the previous exercise by computing $X(k)$, $H(k)$ and $y(n)$ through FFT and IFFT.

Solution: The solution of this question can be found in the code below.

<https://github.com/Gandubs/Signals-and-Systems/blob/master/audio%20filter/codes/5.4.py>

This code verifies the result by plotting the obtained result with the result obtained previously.

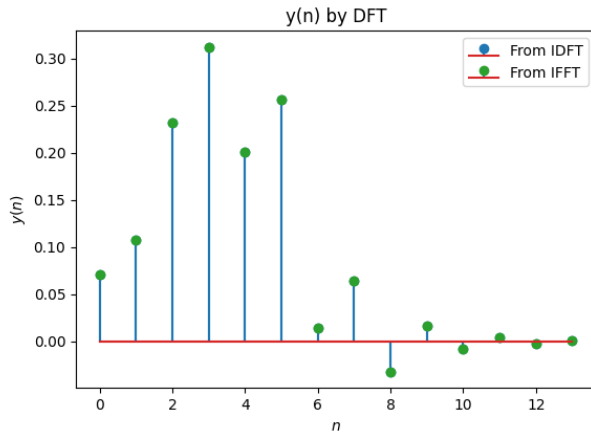


Fig. 8. $y(n)$ obtained from IDFT and IFFT is plotted and verified

V.5 Wherever possible, express all the above equations as matrix equations.

Solution: The DFT matrix is defined as :

$$\mathbf{W} = \begin{pmatrix} \omega^0 & \omega^0 & \dots & \omega^0 \\ \omega^0 & \omega^1 & \dots & \omega^{N-1} \\ \vdots & \vdots & \ddots & \vdots \\ \omega^0 & \omega^{N-1} & \dots & \omega^{(N-1)(N-1)} \end{pmatrix} \quad (41)$$

where $\omega = e^{-\frac{j2\pi}{N}}$. Now any DFT equation can be written as

$$\mathbf{X} = \mathbf{W}\mathbf{x} \quad (42)$$

where

$$\mathbf{x} = \begin{pmatrix} x(0) \\ x(1) \\ \vdots \\ x(n-1) \end{pmatrix} \quad (43)$$

$$\mathbf{X} = \begin{pmatrix} X(0) \\ X(1) \\ \vdots \\ X(n-1) \end{pmatrix} \quad (44)$$

Thus we can rewrite (39) as:

$$\mathbf{Y} = \mathbf{X} \odot \mathbf{H} = (\mathbf{W}\mathbf{x}) \odot (\mathbf{W}\mathbf{h}) \quad (45)$$

where the \odot represents the Hadamard product which performs element-wise multiplication.

The below code computes $y(n)$ by DFT Matrix and then plots it.

<https://github.com/Gandubs/Signals-and-Systems/blob/master/audio%20filter/codes/5.5.py>

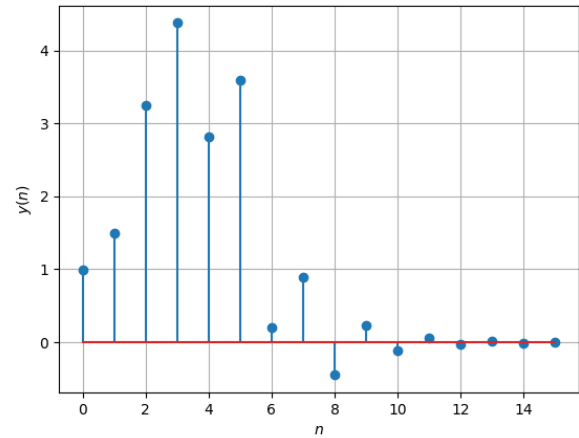


Fig. 9. $y(n)$ obtained from DFT through matrix computations

VI. EXERCISES

Answer the following questions by looking at the python code in Problem I.2.

VI.1 The command

```
output_signal = signal.lfilter(b, a,
                                input_signal)
```

in Problem I.2 is executed through the following difference equation

$$\sum_{m=0}^M a(m)y(n-m) = \sum_{k=0}^N b(k)x(n-k) \quad (46)$$

where the input signal is $x(n)$ and the output signal is $y(n)$ with initial values all 0. Replace **signal.lfilter** with your own routine and verify.

Solution: The below code gives the output of an Audio Filter without using the built in function `signal.lfilter`(Slightly modified audio file here).

<https://github.com/Gandubs/Signals-and-Systems/blob/master/audio%20filter/codes/6.1.py>

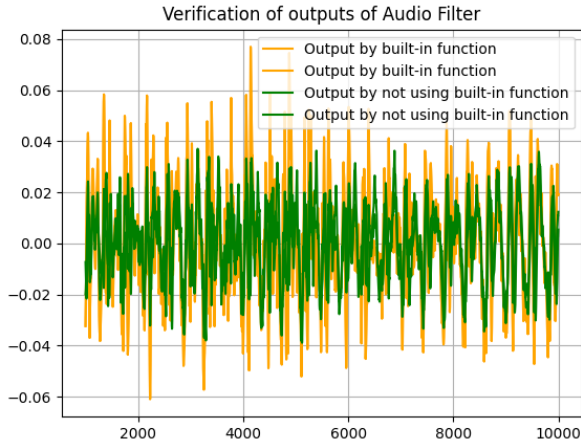


Fig. 10. Both the outputs using and without using function overlap

VI.2 Repeat all the exercises in the previous sections for the above a and b .

Solution: The code in I.2 generates the values of a and b which can be used to generate a difference equation.

And,

$$M = 5 \quad (47)$$

$$N = 5 \quad (48)$$

From 46

$$a(0)y(n) + a(1)y(n-1) + a(2)y(n-2) + a(3)y(n-3) + a(4)y(n-4) = b(0)x(n) + b(1)x(n-1) + b(2)x(n-2) + b(3)x(n-3) + b(4)x(n-4) \quad (49)$$

$$y(n-3) + a(4)y(n-4) = b(0)x(n) + b(1)x(n-1) + b(2)x(n-2) + b(3)x(n-3) + b(4)x(n-4)$$

Difference Equation is given by :

$$\begin{aligned} & y(n) - (3.658)y(n-1) + (5.0314)y(n-2) \\ & - (3.083)y(n-3) + (0.710)y(n-4) \\ & = (1.555 \times 10^{-5})x(n) + (6.220 \times 10^{-5})x(n-1) \\ & + (9.331 \times 10^{-5})x(n-2) + (6.220 \times 10^{-5})x(n-3) \\ & + (1.555 \times 10^{-5})x(n-4) \end{aligned} \quad (50)$$

From (46)

$$H(z) = \frac{b_0 + b_1z^{-1} + b_2z^{-2} + \dots + b_Mz^{-M}}{a_0 + a_1z^{-1} + a_2z^{-2} + \dots + a_Nz^{-N}} \quad (51)$$

$$H(z) = \frac{\sum_{k=0}^N b(k)z^{-k}}{\sum_{k=0}^M a(k)z^{-k}} \quad (52)$$

Partial fraction on (52) can be generalised as:

$$H(z) = \sum_i \frac{r(i)}{1 - p(i)z^{-1}} + \sum_j k(j)z^{-j} \quad (53)$$

Now,

$$a^n u(n) \xleftrightarrow{z} \frac{1}{1 - az^{-1}} \quad (54)$$

$$\delta(n - k) \xleftrightarrow{z} z^{-k} \quad (55)$$

Taking inverse z transform of (53) by using (54) and (55)

$$h(n) = \sum_i r(i)[p(i)]^n u(n) + \sum_j k(j)\delta(n - j) \quad (56)$$

The below code computes the values of $r(i)$, $p(i)$, $k(i)$ and plots $h(n)$

<https://github.com/Gandubs/Signals-and-Systems/blob/master/audio%20filter/codes/6.2.py>

$r(i)$	$p(i)$	$k(i)$
$0.06029142 - 0.14682007j$	$0.88475217 + 0.0445749j$	$2.19006287e - 05$
$0.06029142 + 0.14682007j$	$0.88475217 - 0.0445749j$	–
$-0.06029459 + 0.02518904j$	$0.94427798 + 0.11485352j$	–
$-0.06029459 - 0.02518904j$	$0.94427798 - 0.11485352j$	–

TABLE I
VALUES OF $r(i)$, $p(i)$, $k(i)$

Stability of $h(n)$:

According to (31)

$$H(z) = \sum_{n=0}^{\infty} h(n)z^{-n} \quad (57)$$

$$H(1) = \sum_{n=0}^{\infty} h(n) = \frac{\sum_{k=0}^N b(k)}{\sum_{k=0}^M a(k)} < \infty \quad (58)$$

As both $a(k)$ and $b(k)$ are finite length sequences they converge.

The below code plots Filter frequency response

https://github.com/Gandubs/Signals-and-Systems/blob/master/audio%20filter/codes/6_filter_response.py

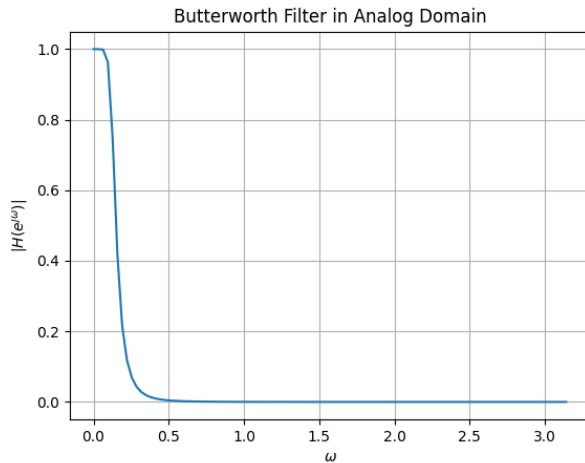


Fig. 11. Frequency Response of Audio Filter

The below code plots the Butterworth Filter in analog domain by using bilinear transform.

$$z = \frac{1 + sT/2}{1 - sT/2} \quad (59)$$

https://github.com/Gandubs/Signals-and-Systems/blob/master/audio%20filter/codes/analog_filt.py

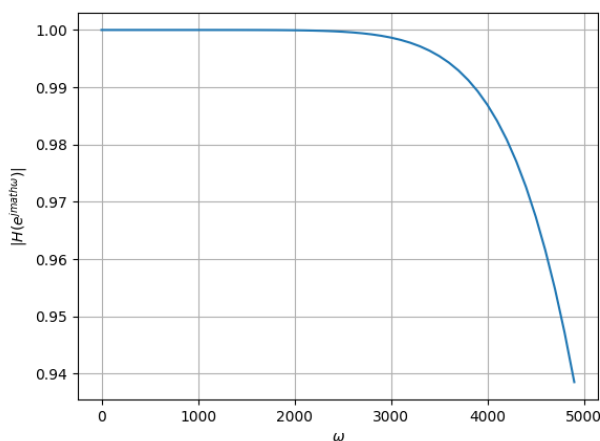


Fig. 12. Butterworth Filter Frequency response in analog domain

The below code plots the Pole-Zero Plot of the frequency response.

https://github.com/Gandubs/Signals-and-Systems/blob/master/audio%20filter/codes/6.2_pole-zero.py

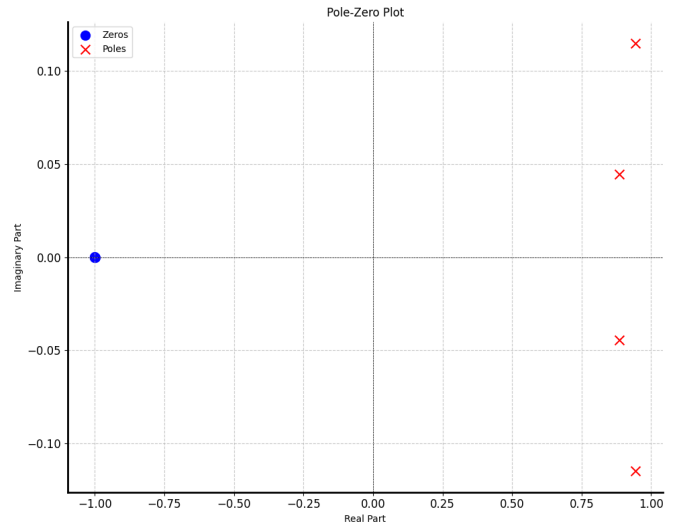


Fig. 13. There are complex poles. So $h(n)$ should be damped sinusoid.

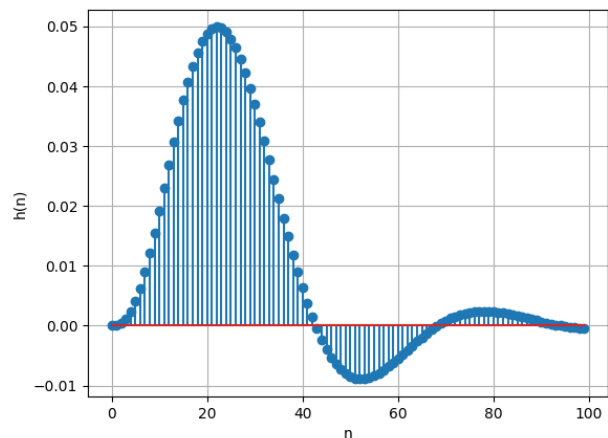


Fig. 14. $h(n)$ of Audio Filter. It is a damped sinusoid.

VI.3 What is the sampling frequency of the input signal?

Solution: The Sampling Frequency is 48KHz

VI.4 What is type, order and cutoff-frequency of the above butterworth filter

Solution: The given butterworth filter is low-pass with order=4 and cutoff-frequency=1kHz.

VI.5 Modify the code with different input parameters and get the best possible output.

Solution: A better filtering was found on setting the order of the filter to be 5.