Audio filtering

EE23BTECH11010 - Venkatesh Bandawar*

Parameter	Description
x(n)	Input Audio Signal
y(n)	Output Audio Signal
H(z)	Discrete Time Fourier Transform of x(n)
h(n)	Impulse Response

TABLE 1: Input Table

Frequency Response of Input Audio signal $x(n)$, given as :

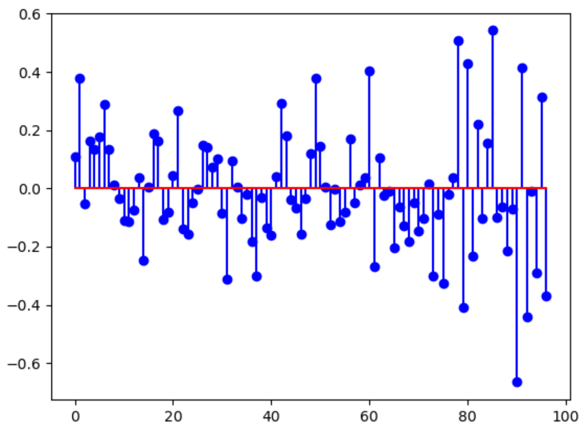

Fig. 1: Graph of x(n)

The Output response can be obtained from Difference equation, given as :

$$\sum_{m=0}^{M} a(m)\, y(n-m) = \sum_{k=0}^{N} b(k)\, x(n-k) \tag{1}$$

The values of a and b are obtained in 'filter.py' file.

$$x[n] \overset{\mathcal{F}}{\longleftrightarrow} X(\omega) \tag{2}$$

$$y[n] \overset{\mathcal{F}}{\longleftrightarrow} Y(\omega) \tag{3}$$

$$h[n] \overset{\mathcal{F}}{\longleftrightarrow} H(\omega) \tag{4}$$

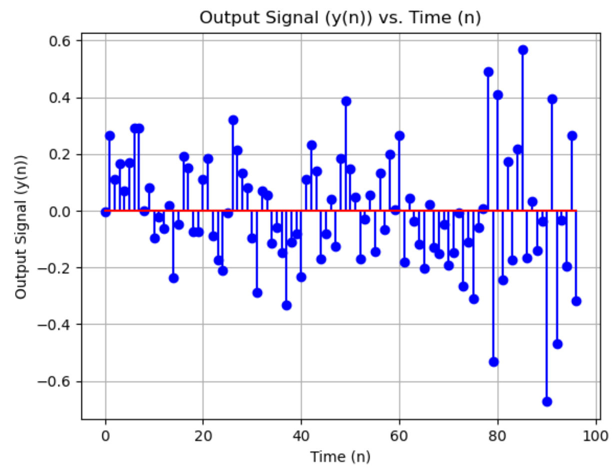

Fig. 2: Graph of y(n)

The plot of $H(\omega)$ is given as :

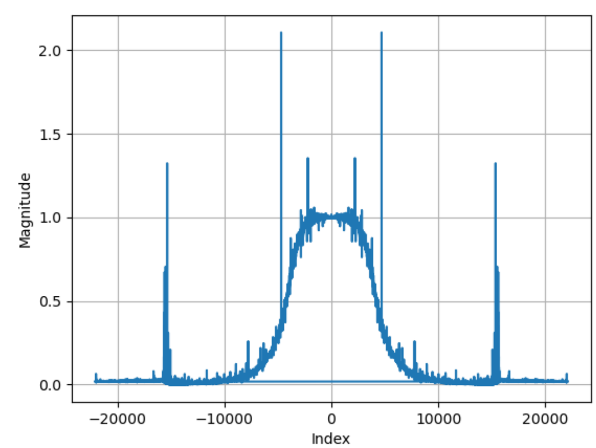

Fig. 3: Graph of $H(\omega)$

$H(\omega)$ is taken as : $\dfrac{X(\omega)}{Y(\omega)}$

$$H(\omega) \overset{\mathcal{F}^{-1}}{\longleftrightarrow} h(n) \tag{5}$$

Taking Inverse fourier transform of $H(\omega)$

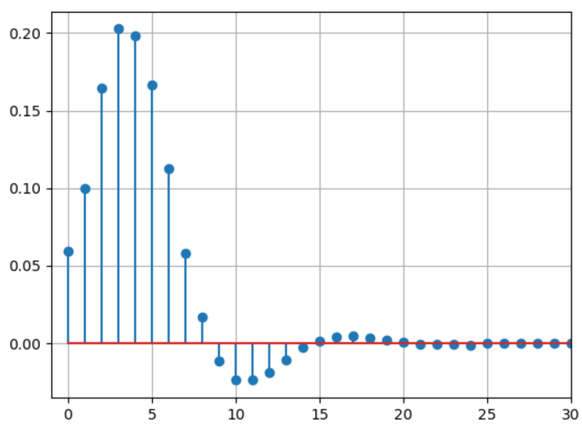

Fig. 4: Graph of h(n)