EE679 ASSIGNMENT-1

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Q1)

$Z = e^{sT}$

$z_p = e^{-B_i \pi T} e^{j 2\pi F_i T}$

$z_p = r e^{j\theta}$

$$H(z) = \frac{K}{(1 - r e^{j\theta} z^{-1})(1 - r e^{-j\theta} z^{-1})}$$

$r_i = e^{-B_i \pi T_s}$ $\quad \theta_i = 2\pi F_i T_s \quad \Rightarrow B_i = \frac{-\ln(r)}{\pi T_s}$

Given $F_1 = 900\,Hz$, $B_1 = 200\,Hz$, $F_s = 16\,kHz$.

$$r_1 = \exp\left(-200 \times \pi \times \frac{1}{16000}\right) = \exp\left(\frac{-\pi}{80}\right)$$

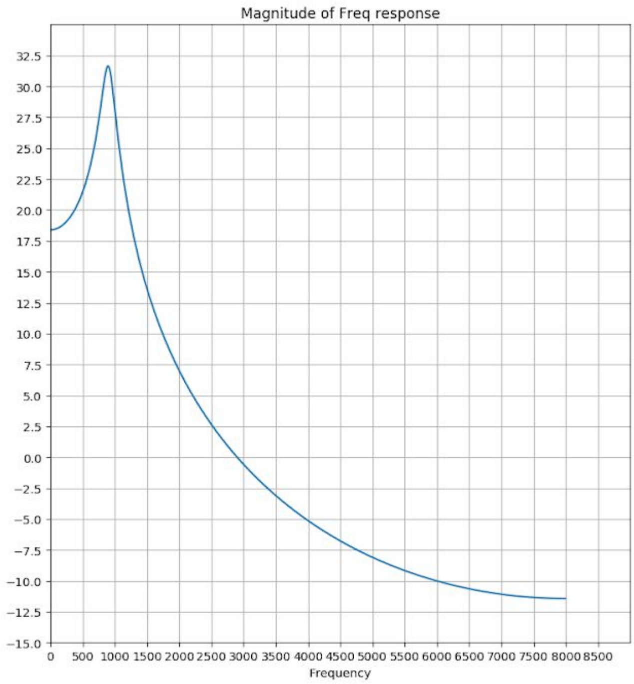
$$\theta_1 = 2\pi \times 900 \times \frac{1}{16000} = \frac{9\pi}{80}$$

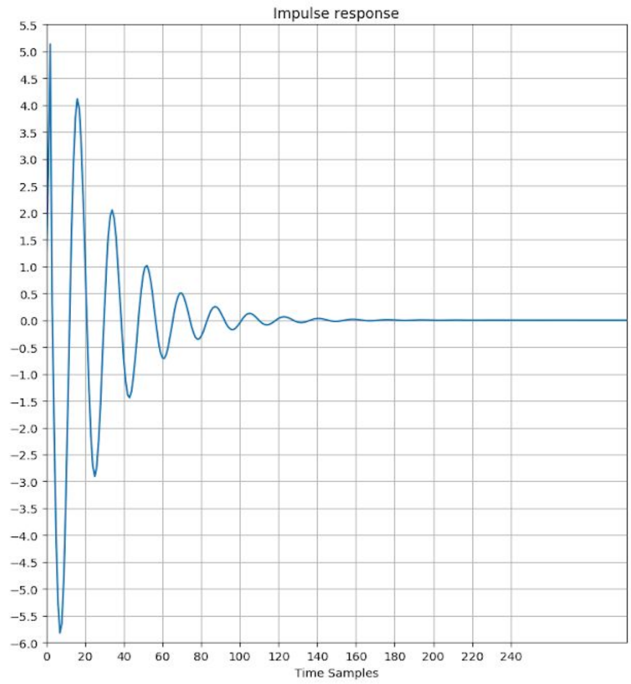
$$H(z) = \frac{1}{1 - 2 \cdot r \cdot \cos\theta \cdot z^{-1} + r^2 z^{-2}} \approx \frac{1}{1 - 1.8z^{-1} + 0.924z^{-2}}$$

To plot the magnitude of frequency, Substitute

$z = e^{j\omega}$ $\quad \omega = \left\{2\pi \times \frac{k f_0}{f_s}\right\}$

Then Substitute $z = e^{j\omega}$ in above $H(z)$ equation and then obtain $H(e^{j\omega})$. Thereafter I used {abs} function in python to plot $|H(e^{j\omega})|$

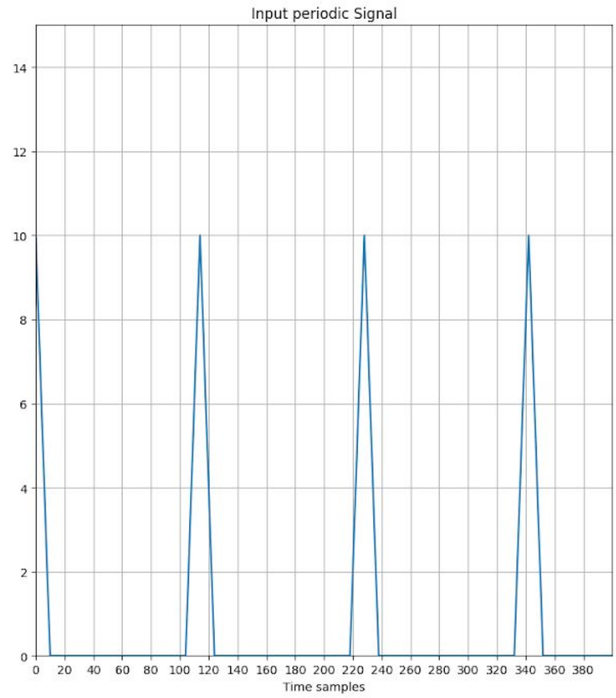




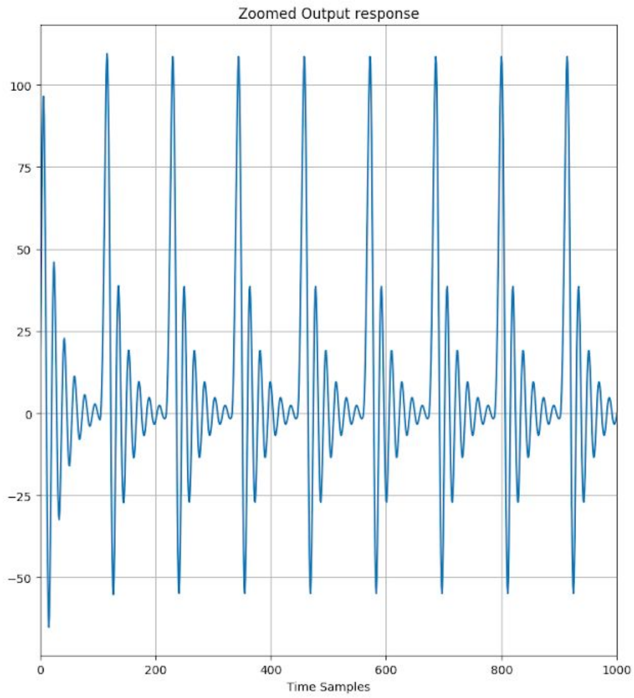
To calculate the Impulse response I used the difference equation of input and output signal in a loop with impulse signal as input (i.e 1 at n = 0 and 0 elsewhere).

Program written for Q1 : "**q1.py**"

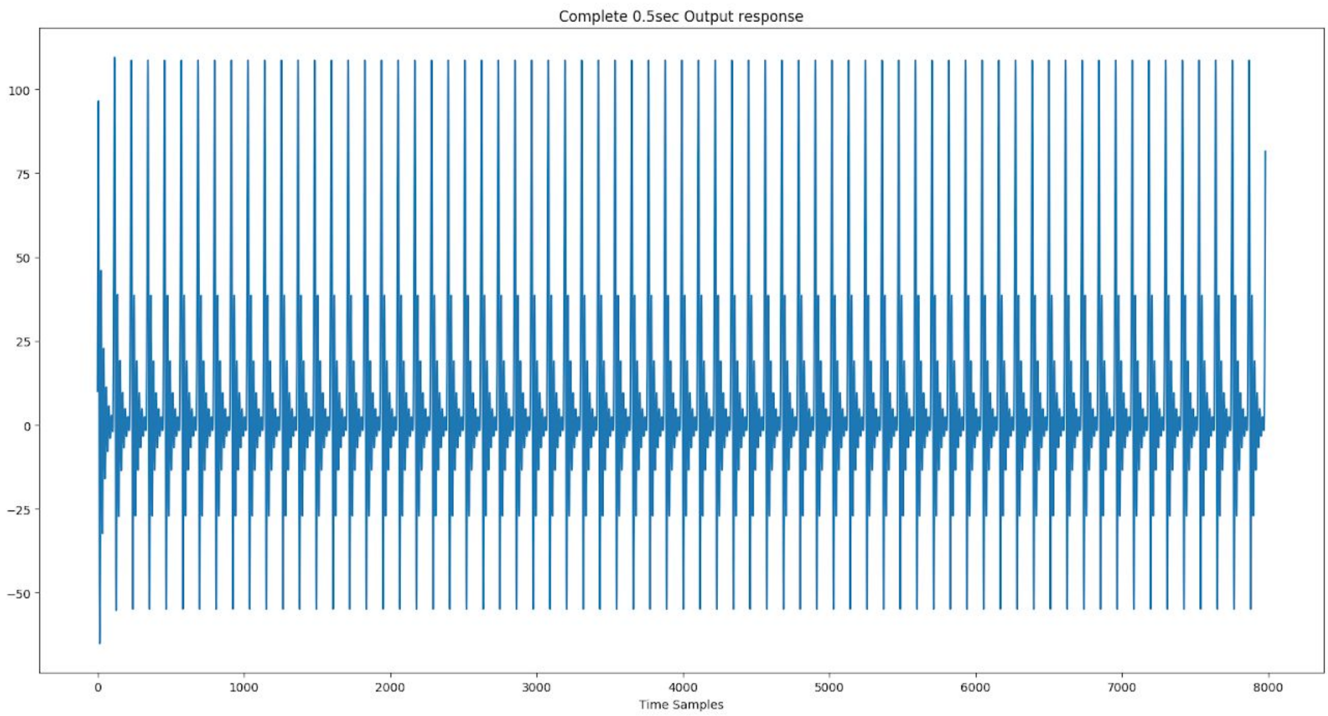
Q2)



Input periodic Signal
Time samples



Zoomed Output response
Time Samples



Complete 0.5sec Output response
Time Samples

Comments on the Quality of the Sound signal :

1. From the perceived sound it seems to be loud and focused towards some frequency because there is no significant noise in the audio signal.
   Actually we call that some frequency above as formant/ resonant frequency

2. The reason also seems bit evident because the format frequency is 900Hz and the harmonics of exciting frequency are 140, 280, 420, 560, 700, **840, 980,** 1120 Hz,,......

   So, there are frequency components close to formant frequency (**900Hz**) and other frequencies are far apart from the formant frequency..

   Thus, less noise and sound seems to be concentrated enough towards format frequency..

   While listening to the audio file clearly at a bit higher volume and for longer duration, it looks like two frequencies (840, 980) are slightly distinguishable in long duration signals.

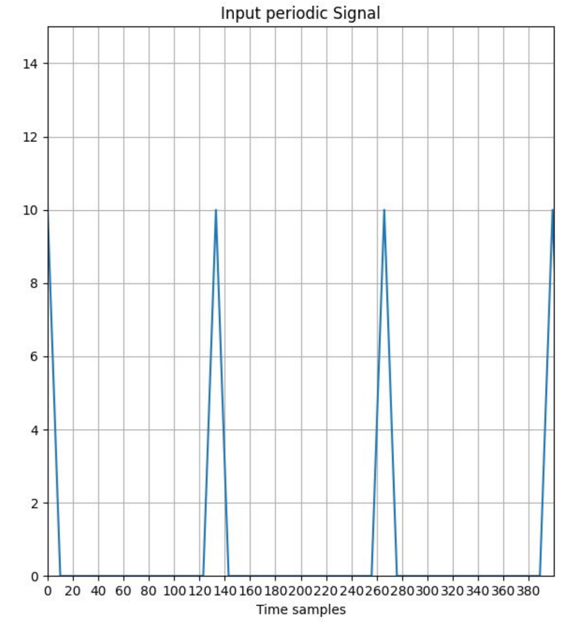
I have reduced the triangular pulse width and tested the audio file by comparing with earlier one, and was not able to find any significant difference between them perceptually.
I have also included the graphs and audio file for this narrower test…..

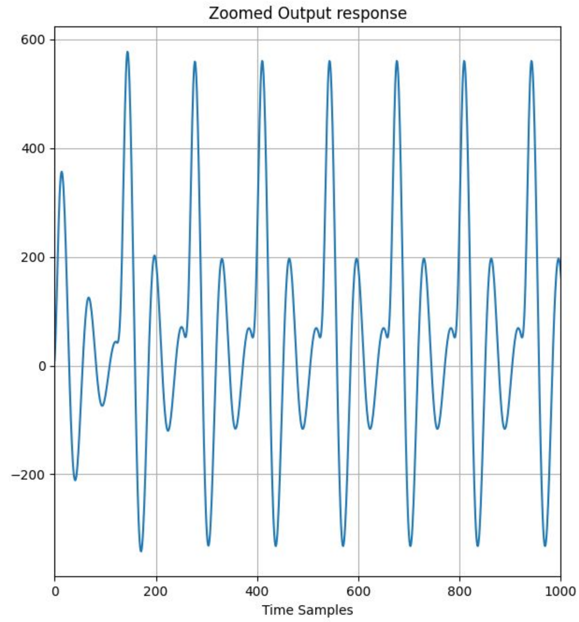
Audio file : "**q2.wav**"

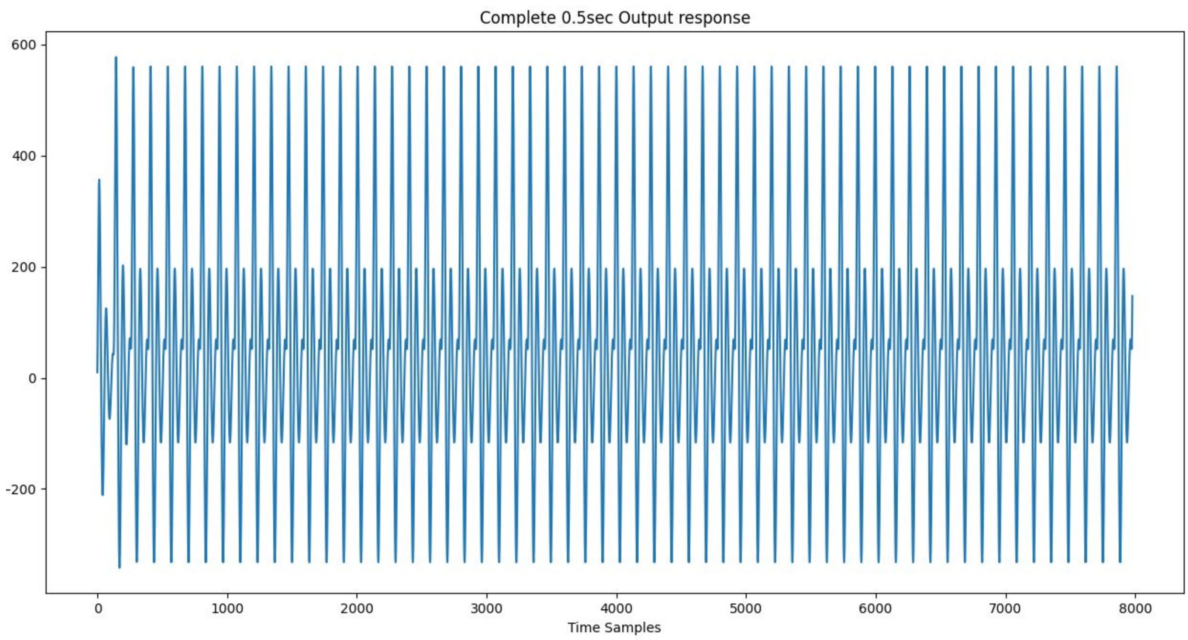
Program written for Q2 : "**q2.py"**

Q3)

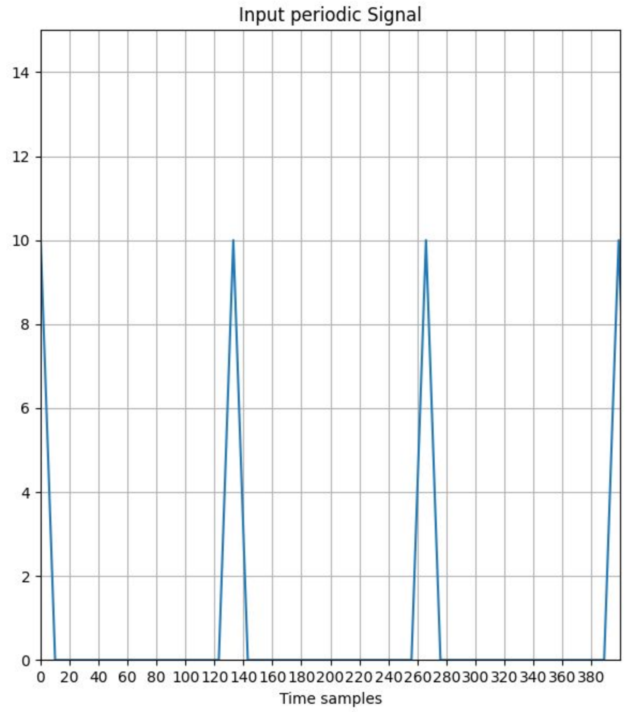
Part (a) : F0 = 120 Hz, F1 = 300 Hz, B1 = 100 Hz   Audio File : q3_a.wav

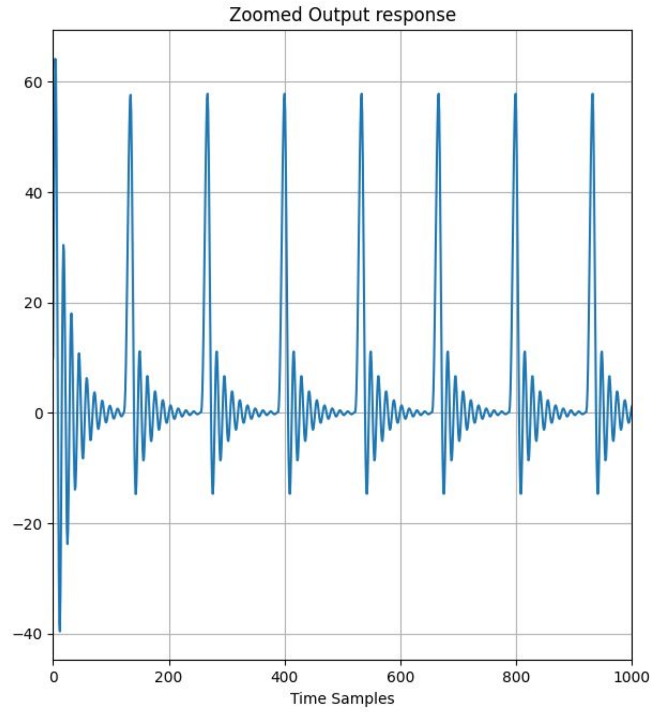


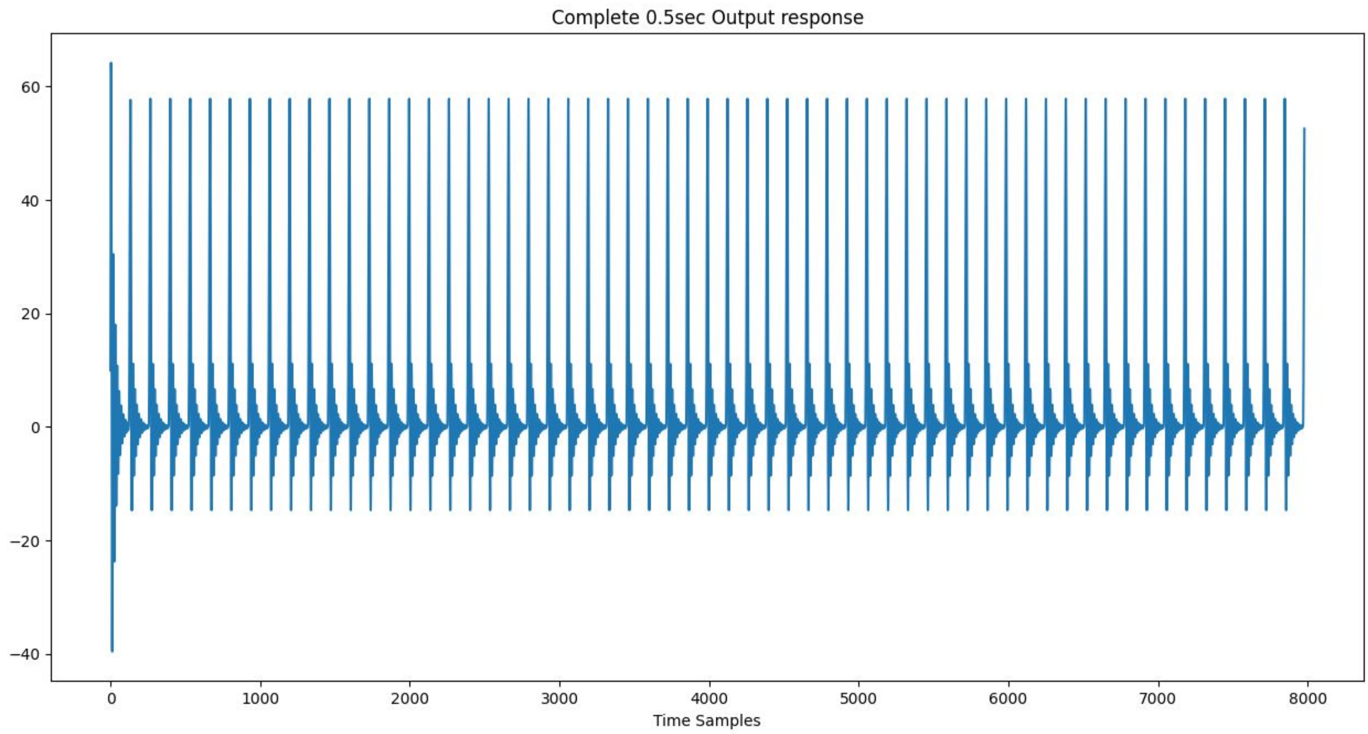




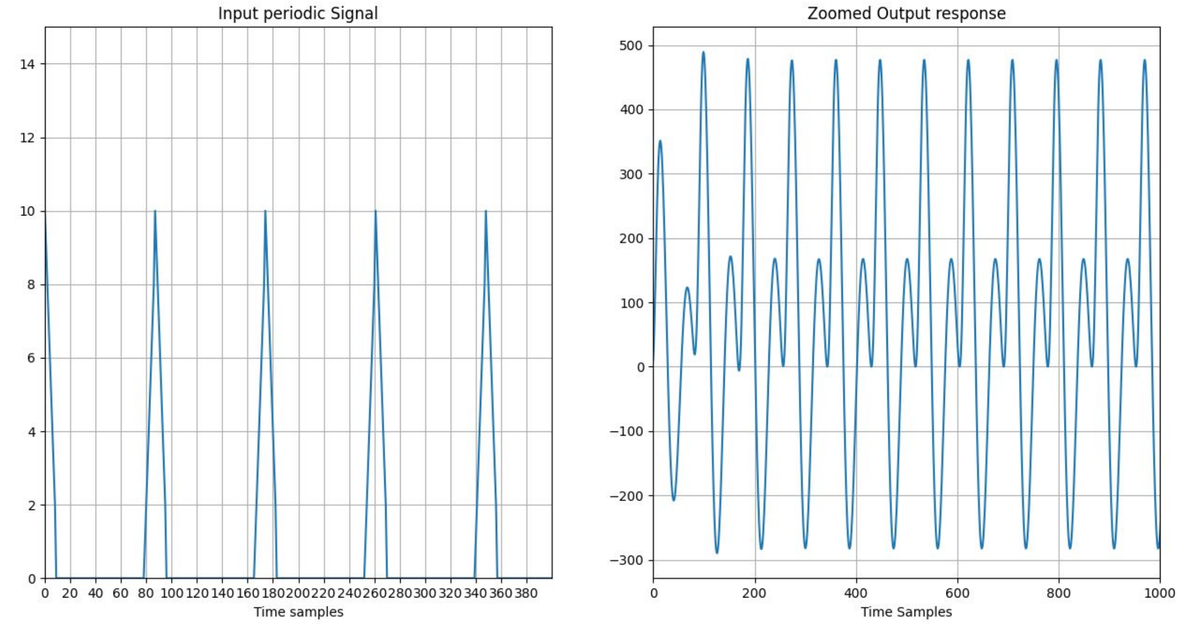
Part (b) : F0 = 120 Hz, F1=1200 Hz, B1 = 200 Hz Audio File : q3_b.wav

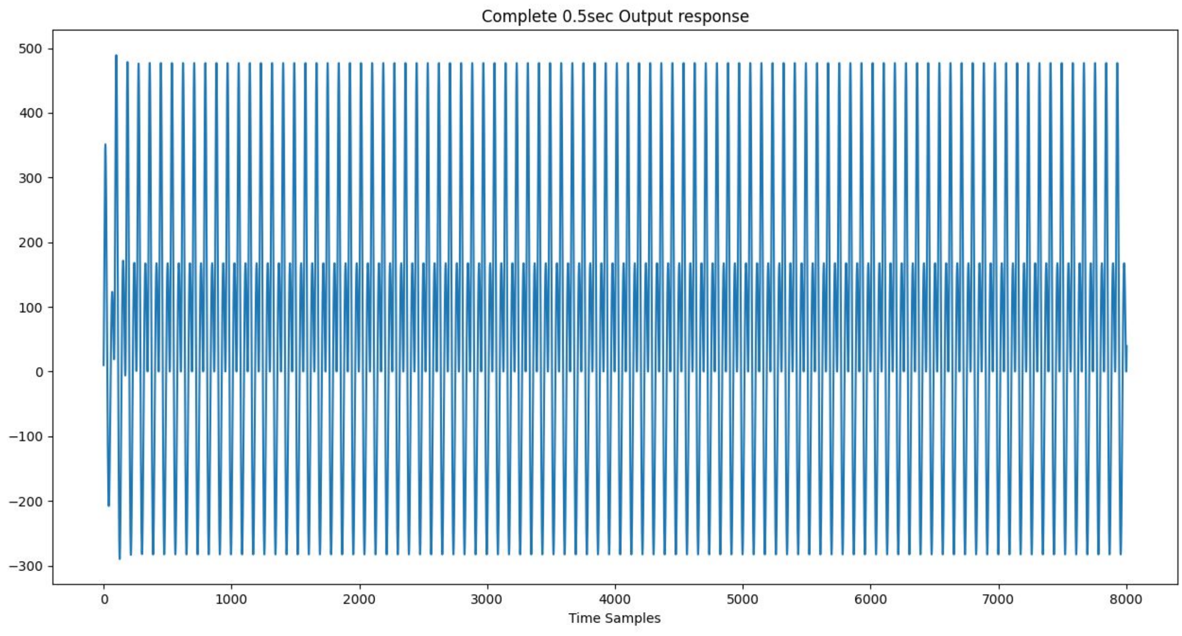






Part (c) : F0 = 180 Hz, F1 = 300 Hz, B1 = 100 Hz   Audio file : q3_c.wav

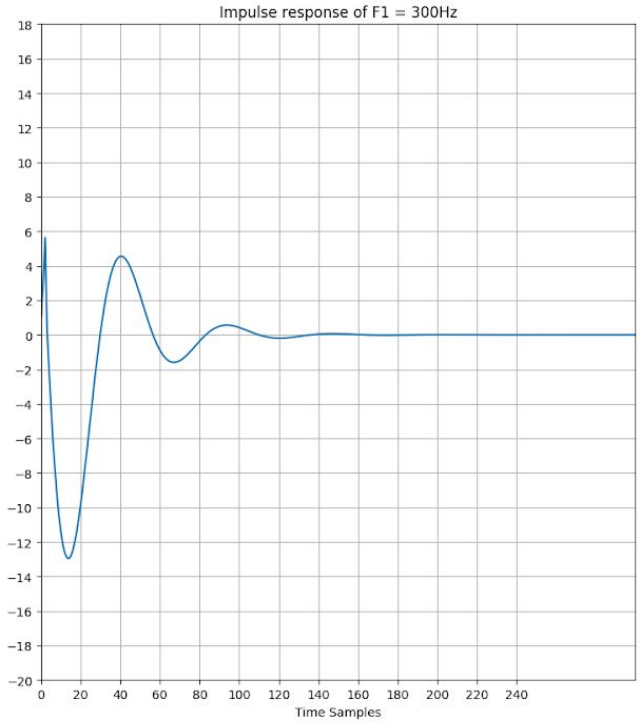


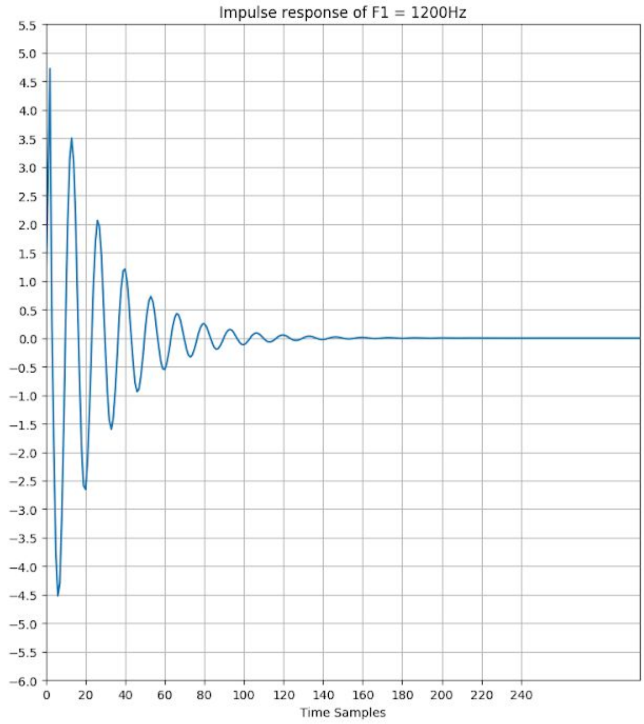


(a) F0 = 120 Hz, F1 = 300 Hz, B1 = 100 Hz
(b) F0 = 120 Hz, F1=1200 Hz, B1 = 200 Hz
(c) F0 = 180 Hz, F1 = 300 Hz, B1 = 100 Hz

Comments on the WAVEFORM shapes :

Impulse responses of F1 = 300Hz (part a,c) , F1 = 1200Hz(part b)

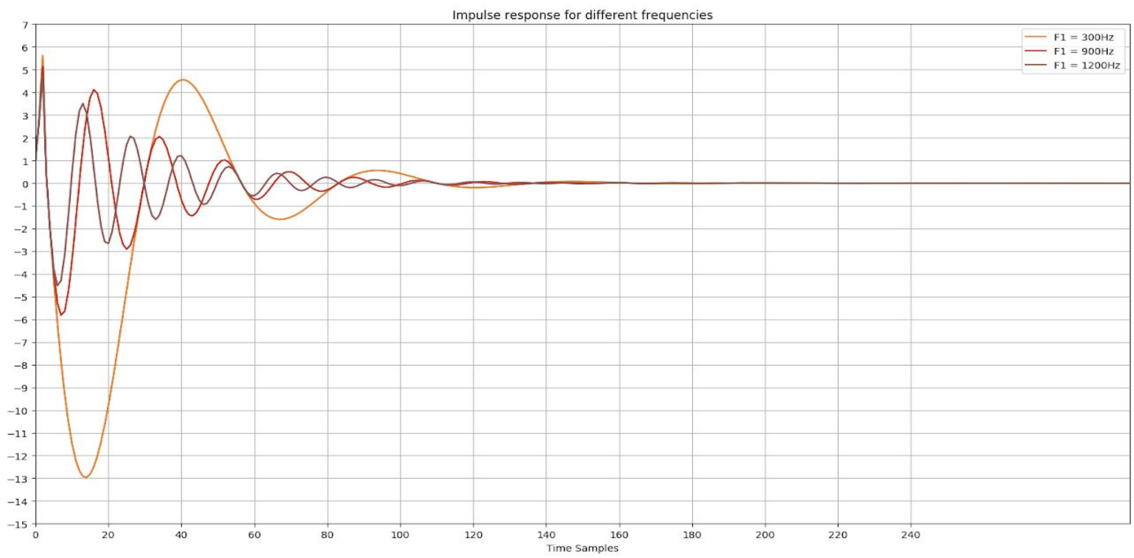




Convolving these impulse responses with the periodic single frequency thin triangular pulse signal gives output which is (closely similar with) the periodic (period = triangular signal frequency) replication of the impulse response .

So this explains why the output plots of part a,b,c look like the plots above.

NOTE : It is important to look at the amplitudes of the Impulse response. It is evident from the Fourier transform mathematical equation , that amplitude of impulse response varies inversely with the frequency .



Comments on Sound Quality :

1. Perceptually the sound generated in part b is very concentrated in terms of frequencies/pitch heard (very less blurry).

2. Then between part a & c . The sound generated by part c is faint/blurry compared to the sound generated in terms of frequencies/pitch heard..

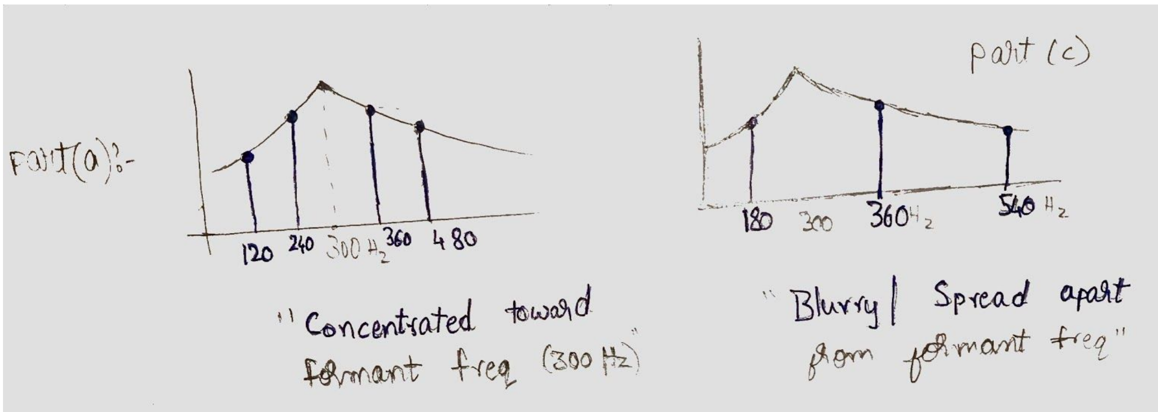
Reason :

1. In part b the formant frequency is 1200Hz , the excitation frequency is 120Hz. So the 10th harmonic of excitation source aligns exactly with formant frequency (120Hz*10 = 1200 Hz) .So the resonance occurs strongs and the 1200Hz component gets amplified more and other frequencies gets suppressed by the filter. So we listen a sound concentrated at 1200Hz..

2. Looking at part a, part c .
   Formant frequency and Bandwidth are the same in part a , c but the exciting frequency is different . 120Hz and 180Hz respectively..

   F1 = 300 Hz, B1 = 100 Hz
   The harmonics of frequency in part a : 120Hz, 240Hz, 360Hz, 480Hz
   The harmonics of frequency in part c : 180Hz, 360Hz, 540Hz, 720Hz

   In part(a) there are two frequency harmonics close to formant frequency (240Hz, 360Hz) , In case of part(c) there is only one component close to formant frequency (360Hz) and all other components are far apart . So compared to part(c) the part(a) is more concentrated at formant frequency. Thus part(c) is blurry compared to part(a).



Program for Qn3 : "**q3.py**"

Q4) In this question 3 formant frequencies are given .. F1 ,F2, F3..

Implementation :

In case of single formant frequency, we have a two pole model for discrete time filter . After designing that two pole discrete time filter we convolve it with the input signal .

So, we take advantage of this setup even in case of a multi formant filter model.Let there be n formant frequencies. First we design two pole filter models for each formant frequency and desired bandwidth for each formant .

Thereafter approximate the multi formant filter with the filter formed by serially cascading these n filters , which we designed for each formant frequency.

In case of our 3 formant frequency, the following steps are followed :

1. Design separate filters for each formant frequency (F1, F2 and F3) and their corresponding bandwidths(B1, B2 and B3) using the two pole model we have.

2. Now , generate the input signal of thin triangular pulses of frequency F0

3. First pass this **Triangular pulse** through the single formant filter designed by using two pole model for formant frequency F1 and Bandwidth B1.The output from this filter is named "**Filter1 Response**".

4. Then pass this **Filter1 Response** through the single formant filter designed by using two pole model for formant frequency F2 and Bandwidth B2.The output from this filter is named "**Filter2 Response**".

5. Then pass this **Filter2 Response** through the single formant filter designed by using two pole model for formant frequency F3 and Bandwidth B3.The output from this filter is named "**Output Response**".

NOTE :These responses mentioned above are in time domain

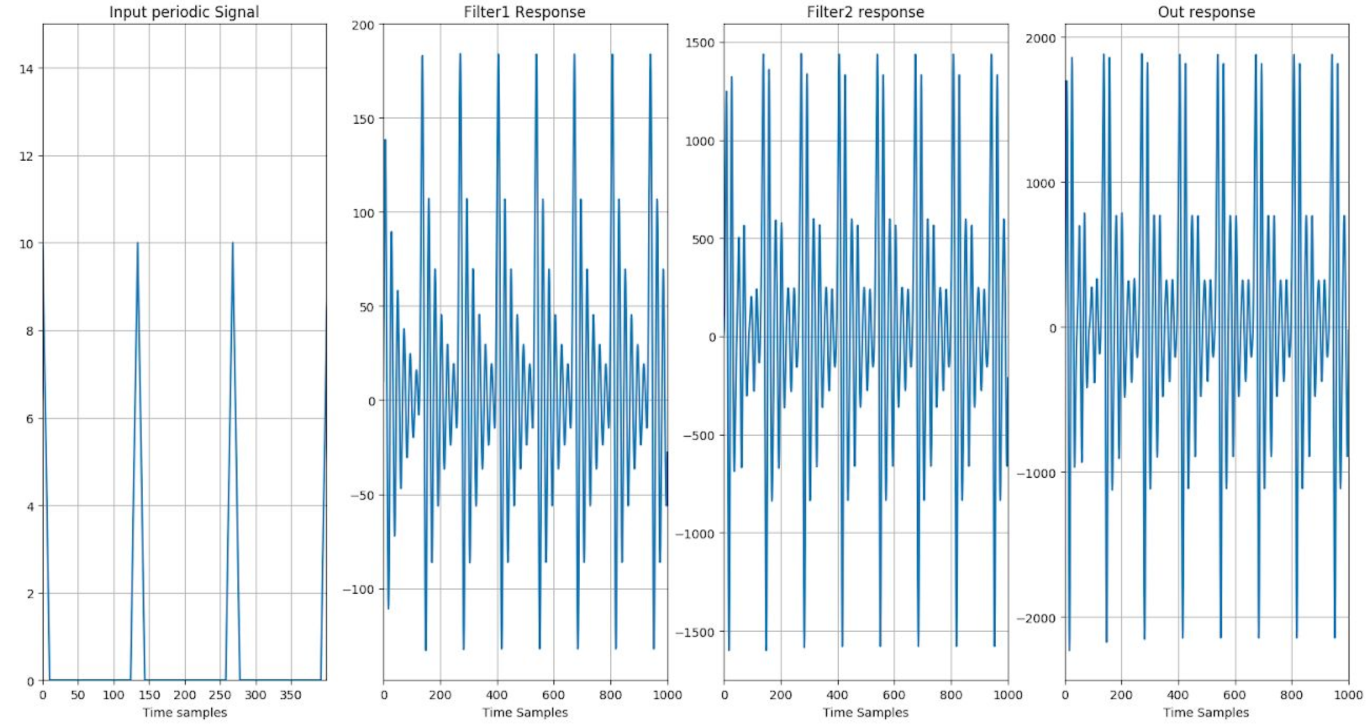
Observations :

1. Significant difference in the perception of sounds generated at 120Hz and 220Hz is noticed.The sound at 220Hz seems to be pleasant compared with the 120Hz .

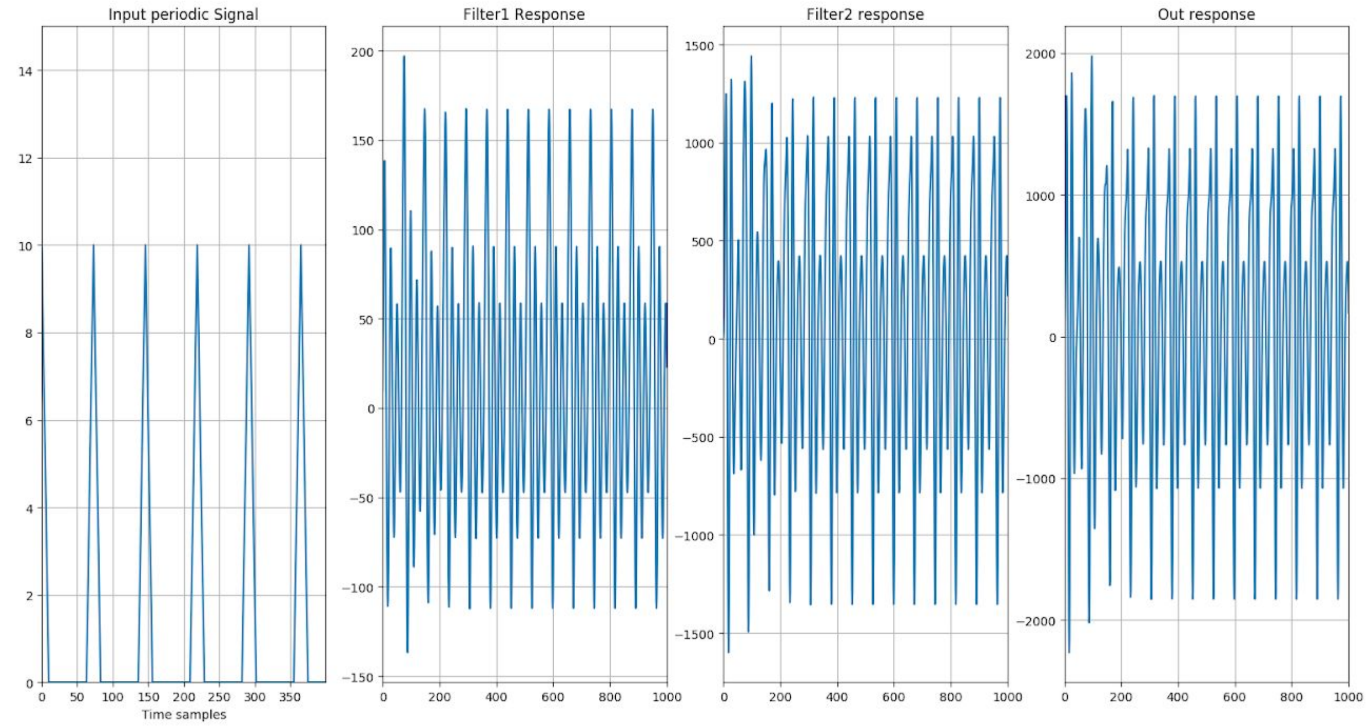
2. The perceptual feature difference (we name it to be pitch) felt between 120Hz ,220Hz sounds remain roughly the same for all vowels.

The below shown plots are waveforms corresponding to each vowel at both frequencies. Audio files are attached in the zip folder .
Program for question 4 is : "**q4.py"**

Vowel /a/ at 120Hz Audio file : q4_a_120Hz.wav ,
Waveform :

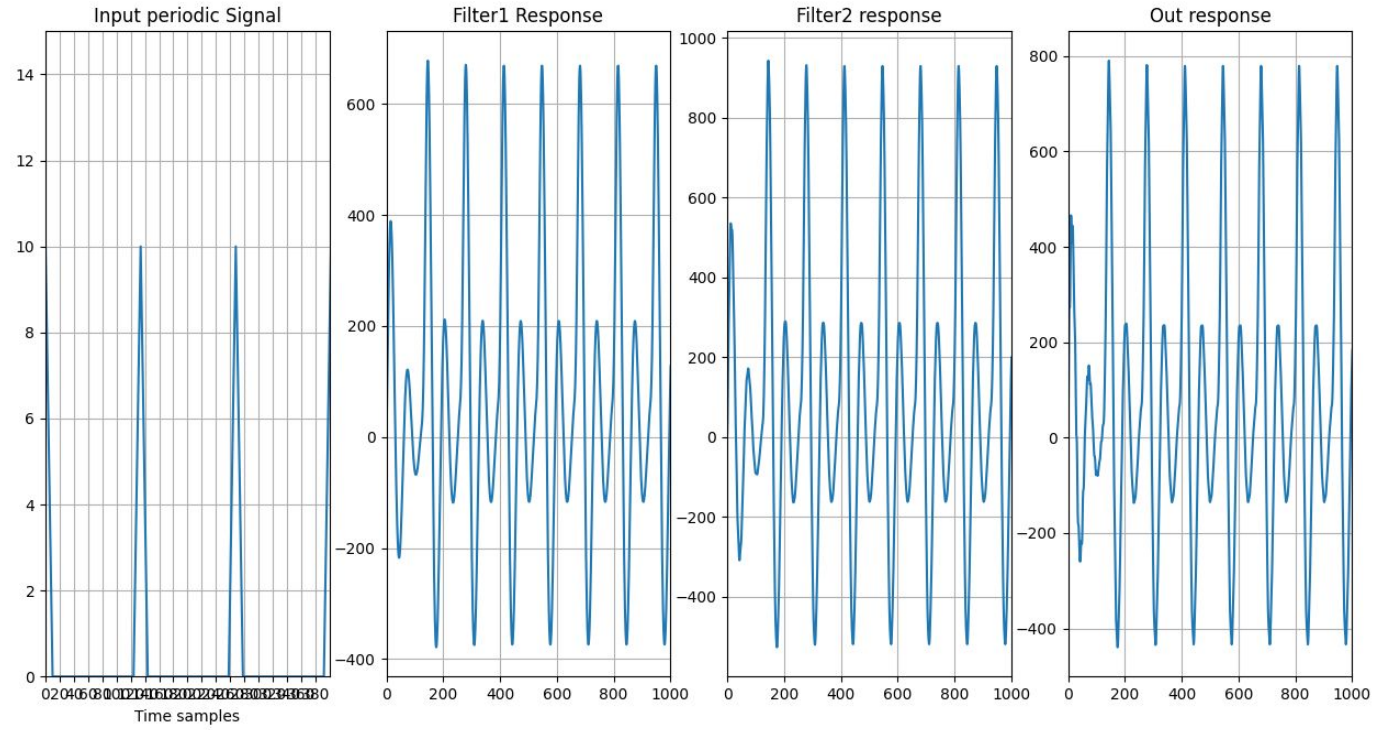


Vowel /a/ at 220Hz Audio file : q4_a_220Hz.wav ,
Waveform :



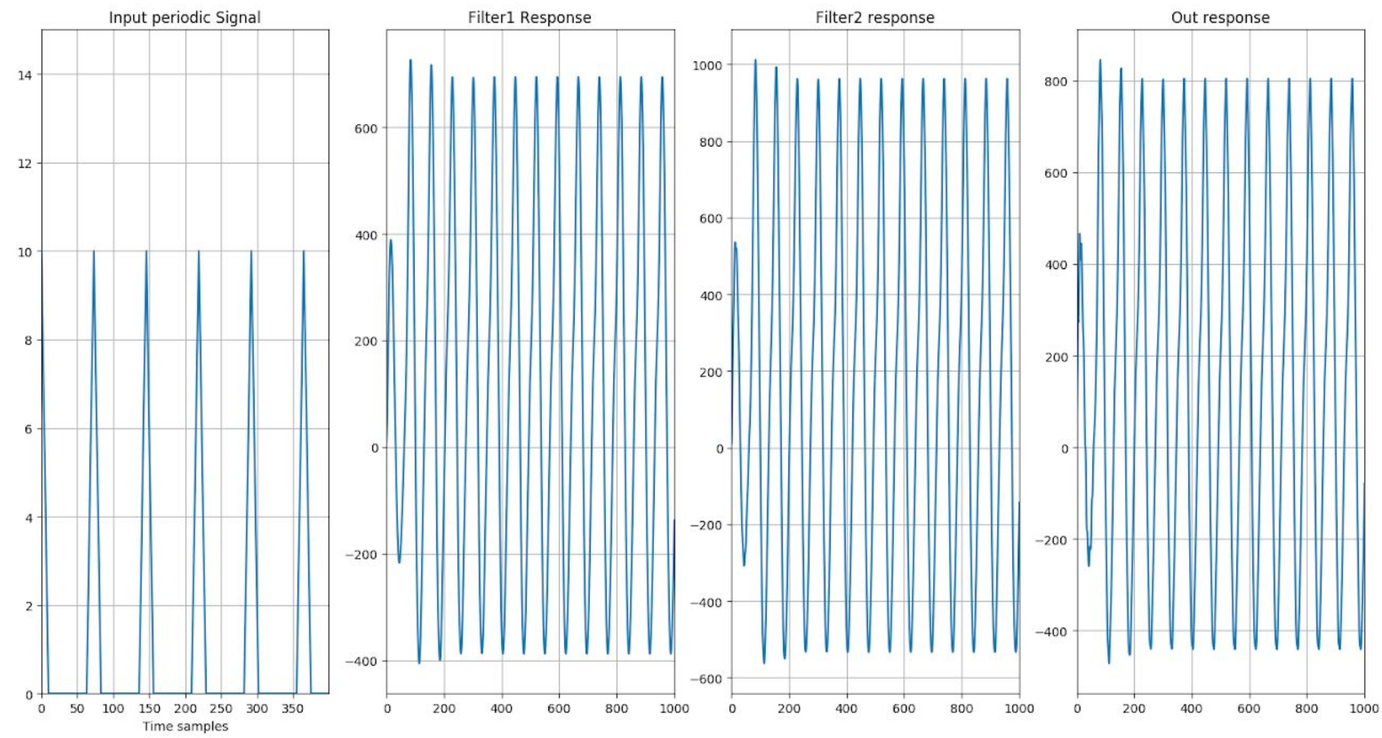
Vowel /i/ at 120Hz Audio file : q4_i_120Hz.wav ,

Waveform :

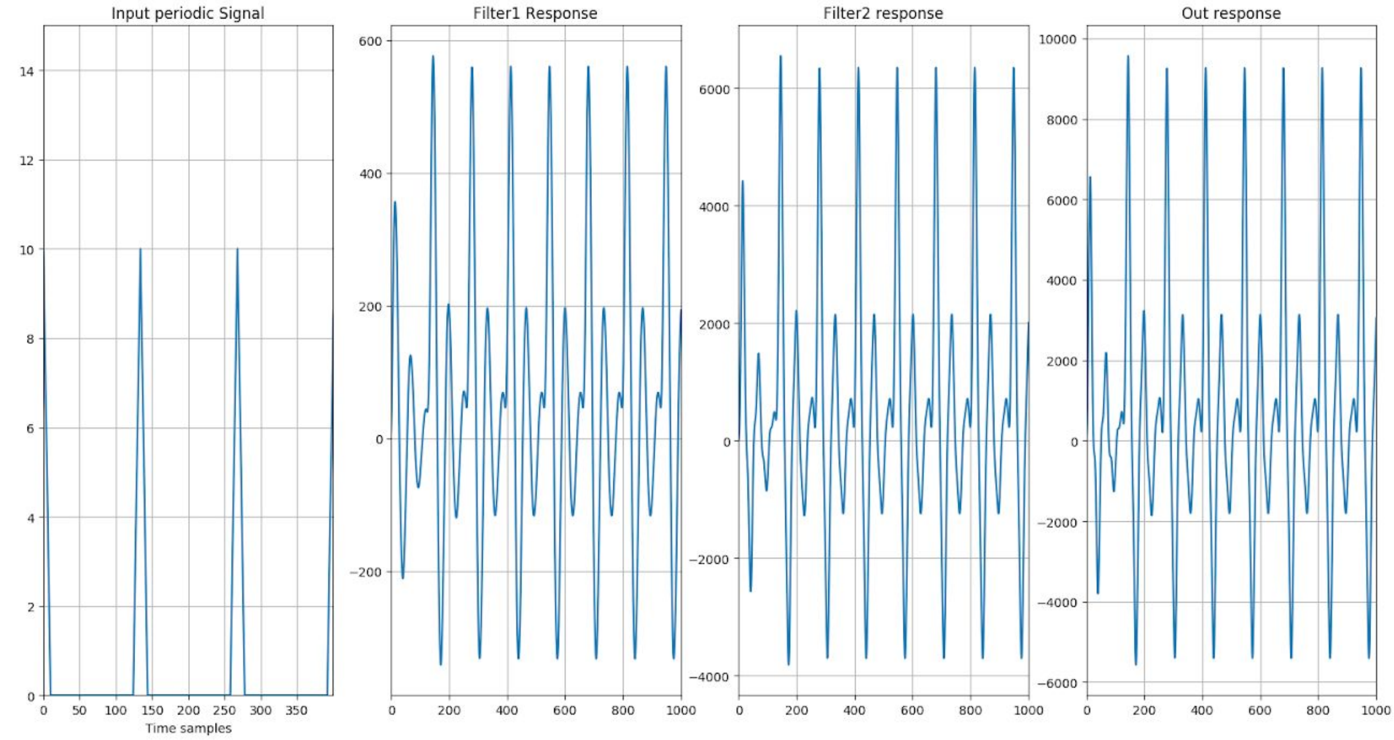


Vowel /i/ at 220Hz Audio file : q4_i_220Hz.wav ,

Waveform :



Vowel /u/ at 120Hz Audio file : q4_u_120Hz.wav ,
Waveform :



Vowel /u/ at 220Hz Audio file : q4_u_220Hz.wav ,
Waveform :