

# Assignment 1 Speech Processing

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# 1 Question 1

Given the following specification for a single-formant resonator, obtain the transfer function of the filter  $H(z)$  from the relation between resonance frequency / bandwidth, and the pole angle / radius. Plot filter magnitude response (dB magnitude versus frequency) and impulse response.

$F1$  (formant) = 900 Hz

$B1$  (bandwidth) = 200 Hz

$F_s$  (sampling freq) = 16 kHz

$$\left. \begin{aligned} r_0 &= e^{-B_0 T} \\ \theta_0 &= 2\pi F_0 T \end{aligned} \right\} \text{Poles of form } re^{j\theta}$$

$$H(z) = \frac{k}{(1 - re^{j\theta} z^{-1})(1 - re^{-j\theta} z^{-1})}$$

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

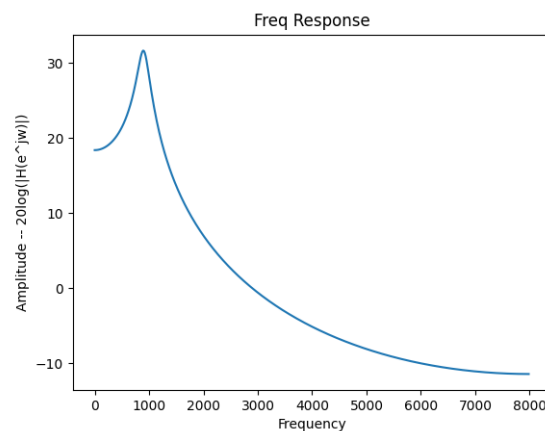
$$\begin{aligned} \text{numerator} &= [1] \\ \text{Denominator} &= [1, -2r\cos\theta, r^2] \end{aligned}$$

$$\frac{Y(z)}{X(z)} = \frac{1}{1 - 2r\cos\theta z^{-1} + r^2 z^{-2}}$$

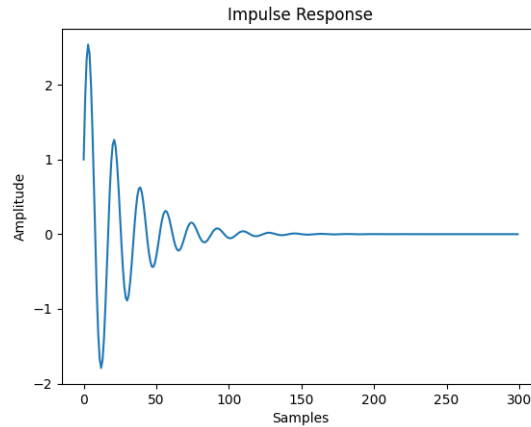
Difference equation

$$Y[n] - Y[n-1]r\cos\theta + r^2 Y[n-2] = X[n]$$

The above formula is for the Single formant resonator. Plugging in the values in the equation we observe the following frequency response



To obtain the impulse response, we give an impulse to the function at  $x=0$  and obtain the response using the difference equation.  $Y[n]$  depends only on  $Y[n-1]$ ,  $Y[n-2]$  and  $X[n]$



```

1  import numpy as np
2  import math
3  import matplotlib.pyplot as plt
4  from scipy import signal
5
6  # Input Parameters
7  F_1=900
8  B_1=200
9  F_s=16000
10
11  r_i = np.exp(-B_1*3.142/F_s)
12  theta_i = 2*3.142*F_1/F_s
13  denominator=[1,-2*r_i*math.cos(theta_i),r_i*r_i]
14  numerator=[1]
15  w, h = signal.freqz(numerator,denominator)
16
17  # plt.title('Freq Response')
18  # plt.plot(F_s*w/(2*3.142), 20 * np.log10(abs(h)))
19  # plt.ylabel('Amplitude -- 20log(|H(e^jw)|)')
20  # plt.xlabel('Frequency')
21  # plt.savefig('Q1_1.png')
22  y=[]
23  time=[]
24  for i in range(0,300):
25      y.append(0)
26      time.append(i)
27  # print(time)
28  y[0]=1 # impulse
29  y[1]= -y[0]*denominator[1]
30  for i in range(2,299):
31      y[i] = -y[i-1]*denominator[1]-y[i-2]*denominator[2]; #x[i]=0
32  plt.title('Impulse Response')
33  plt.plot(time, y)
34  plt.ylabel('Amplitude')
35  plt.xlabel('Samples')
36  plt.savefig('Q1_2.png')
37

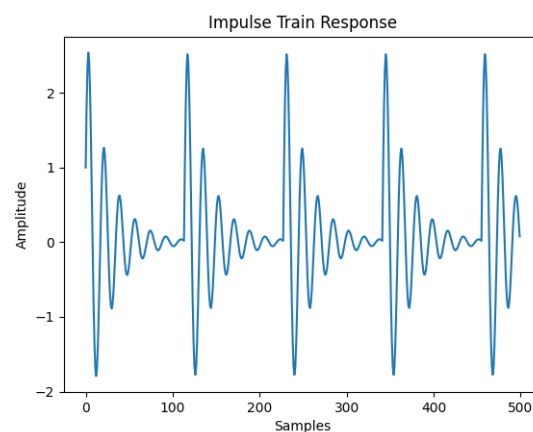
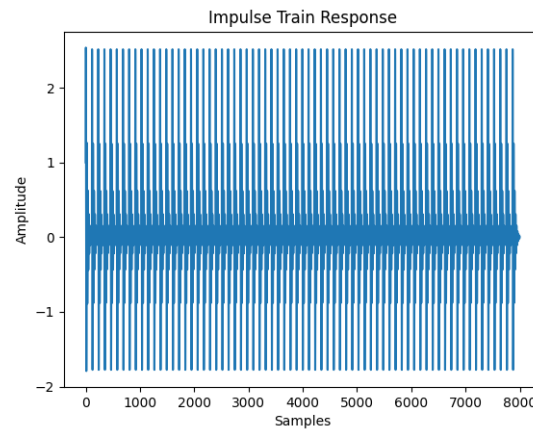
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## 2 Question 2

Excite the above resonator (“filter”) with a periodic source excitation of  $F_0 = 140$  Hz. You can approximate the source signal by narrow-triangular pulse train. Compute the output of the source-filter system over the duration of 0.5 second using the difference equation implementation of the LTI system. Plot the time domain waveform over a few pitch periods so that you can observe waveform characteristics. Play out the 0.5 sec duration sound and comment on the sound quality.

Now instead of using an impulse response, a train of impulses are used as an input. The train of inputs have a frequency of 140 Hz. With a sampling rate of 16K Hz, we have 8000 samples in 0.5 seconds. Out of those 70 samples must be impulses. Thus after every 8000/70 samples there is an impulse.

Using Python we code this in the input variable and pass the same through the Difference Equation to obtain the output response.



Since the frequency is very low its very low pitched and doesnt have such good quality

```

1  import numpy as np
2  import math
3  import matplotlib.pyplot as plt
4  from scipy import signal
5  from scipy.io.wavfile import write
6
7  # Input Parameters
8  F_1=900
9  B_1=200
10 playingtime=0.5
11 F_0=140
12 F_s=16000
13
14 r_i = np.exp(-B_1*3.142/F_s)
15 theta_i = 2*3.142*F_1/F_s
16 denominator=[1,-2*r_i*math.cos(theta_i),r_i*r_i]
17 numerator=[1]
18 w, h = signal.freqz(numerator,denominator)
19 x=[]
20 y=[]
21 time=[]
22
23 total_samples=int(F_s*playingtime)

```

```

24 for i in range(0,total_samples):
25     x.append(0)
26     y.append(0)
27     time.append(i)
28
29
30 i=0
31 count=0
32 while i<total_samples and count < int(F_0*playingtime):
33     x[i]= 1;
34     i=i+int(F_s/F_0);
35     count+=1
36
37 y[0]=1
38 y[1]= x[1]-y[0]*denominator[1]
39 for i in range(2,total_samples):
40     y[i] = x[i]-y[i-1]*denominator[1]-y[i-2]*denominator[2];
41 plt.title('Impulse Train Response')
42 # plt.plot(time[0:500], y[0:500])
43 plt.plot(time, y)
44 plt.ylabel('Amplitude')
45 plt.xlabel('Samples')
46 # plt.savefig('Q2_1.png')
47 plt.savefig('Q2_2.png')
48
49 y = np.array(y)
50 write('Q2_1.wav', F_s, y)

```

### 3 Question 3

Vary the parameters as indicated below and comment on the differences in waveform and sound quality for the different parameter combinations.

- (a)  $F_0 = 120$  Hz,  $F_1 = 300$  Hz,  $B_1 = 100$  Hz
- (b)  $F_0 = 120$  Hz,  $F_1 = 1200$  Hz,  $B_1 = 200$  Hz
- (c)  $F_0 = 180$  Hz,  $F_1 = 300$  Hz,  $B_1 = 100$  Hz

Using the above code and varying the values of the parameters  $F_0$ ,  $F_1$  and  $B_1$  we observe the following three graphs and the image sounds

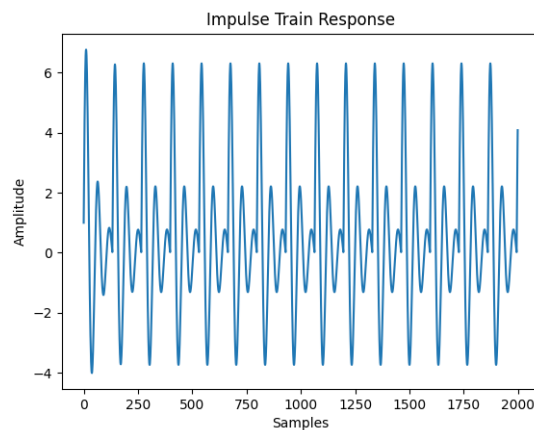


Figure 1:  $F_0 = 120$  Hz,  $F_1 = 300$  Hz,  $B_1 = 100$  Hz

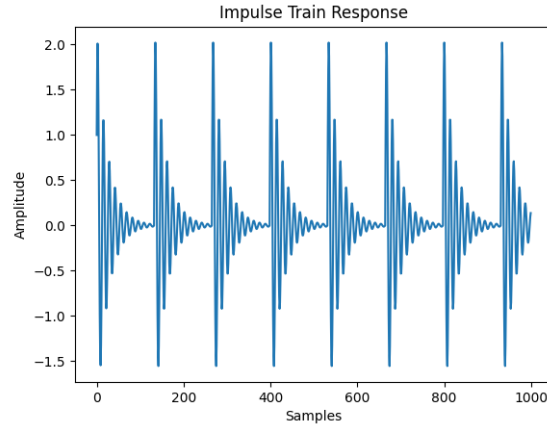


Figure 2:  $F_0 = 120$  Hz,  $F_1=1200$  Hz,  $B_1 = 200$  Hz

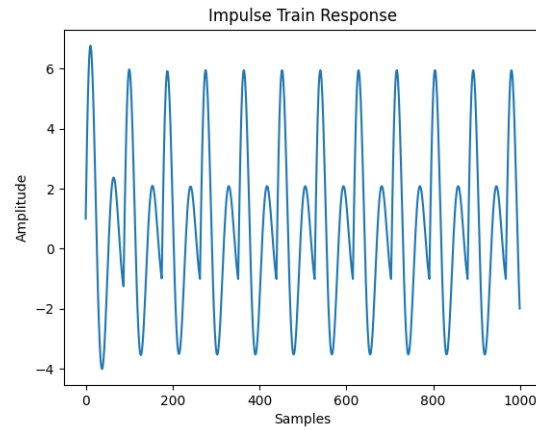


Figure 3:  $F_0 = 180$  Hz,  $F_1 = 300$  Hz,  $B_1 = 100$  Hz

The first and the third sounds have roughly the same waveforms. The 3rd one seems to have a much higher pitch the 1st one and can be easily identified when we play the sounds as well. The first and second have a different bandwidth and thus it takes longer for the 2nd waveform to die than the first one. This is also clearly visible in the number of waves in the 1000 samples. Moreover, 2nd waveform on hearing seems to have a longer decay effect too. Clearly there is a direction correlation of bandwidth and decay time. The first and second also differ in terms of the formant frequency as well. Which makes the angle of the pole different. The second waveform doesn't experience a sudden decrease. It always decreases gradually.

```

1  import numpy as np
2  import math
3  import matplotlib.pyplot as plt
4  from scipy import signal
5  from scipy.io.wavfile import write
6
7  # Input Parameters
8  F_1=300
9  B_1=100
10 playingtime=0.5
11 F_0=180
12 F_s=16000
13
14 r_i = np.exp(-B_1*3.142/F_s)
15 theta_i = 2*3.142*F_1/F_s
16 denominator=[1,-2*r_i*math.cos(theta_i),r_i*r_i]
17 numerator=[1]
18 w, h = signal.freqz(numerator,denominator)
19 x=[]

```

```

20 y=[]
21 time=[]
22
23 total_samples=int(F_s*playingtime)
24 for i in range(0,total_samples):
25     x.append(0)
26     y.append(0)
27     time.append(i)
28 i=0
29 count=0
30 while i<total_samples and count < int(F_0*playingtime):
31     x[i]= 1;
32     i=i+int(F_s/F_0);
33     count+=1
34
35 y[0]=1 # impulse
36 y[1]= x[1]-y[0]*denominator[1]
37 for i in range(2,total_samples):
38     y[i] = x[i]-y[i-1]*denominator[1]-y[i-2]*denominator[2];
39 plt.title('Impulse Train Response')
40 plt.plot(time[0:1000], y[0:1000])
41 plt.ylabel('Amplitude')
42 plt.xlabel('Samples')
43 # plt.savefig('Q2_1.png')
44 plt.savefig('Q3_3.png')
45
46 y = np.array(y)
47 write('Q3_3.wav', F_s, y)
48

```

## 4 Question 4

In place of the simple single-resonance signal, synthesize the following more realistic vowel sounds at two distinct pitches ( $F_0 = 120$  Hz,  $F_0 = 220$  Hz). Keep the bandwidths constant at 100 Hz for all formants. Duration of sound: 0.5 sec

Vowel  $F_1$ ,  $F_2$ ,  $F_3$

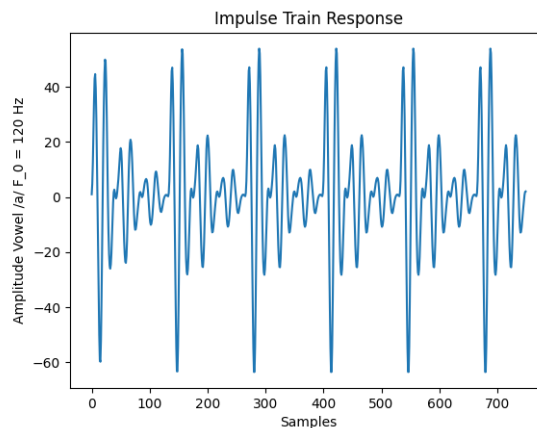
/a/ 730, 1090, 2440

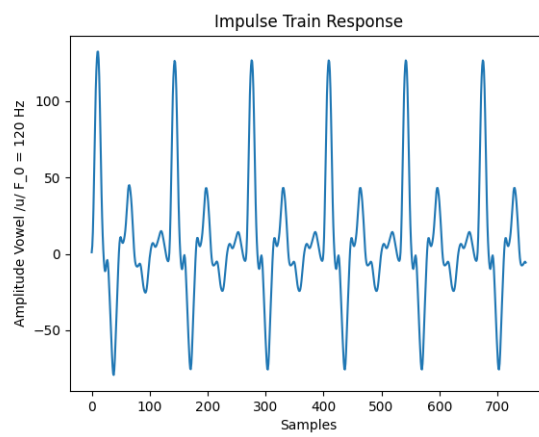
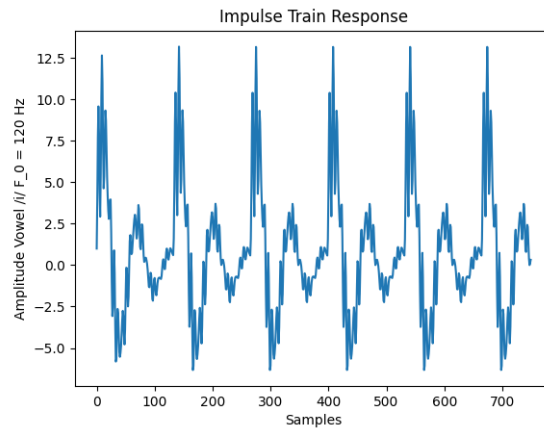
/i/ 270, 2290, 3010

/u/ 300, 870, 2240

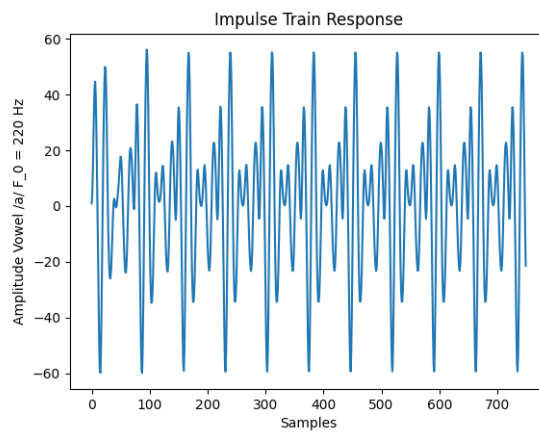
Since we use 3 formants here, We can multiply the transfer functions so as to obtain the final response of the same. To do this, let the first impulse train be bounded by  $F_0$  and then passed through the difference equation using  $F_1$  to obtain a response  $Y$ . This  $Y$  is then passed as an input to the difference equation using  $F_2$  a response  $Y_1$  is obtained. Then the  $Y_1$  is passed as an input to the difference equation using  $F_3$  and a response  $Y_2$  is obtained. This is the final response to the train of impulses.

### 4.1 120 Hz

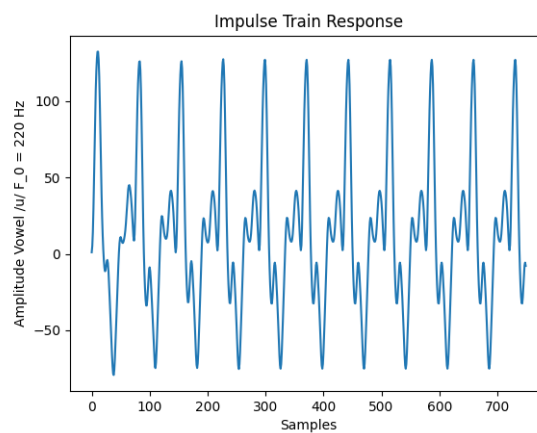
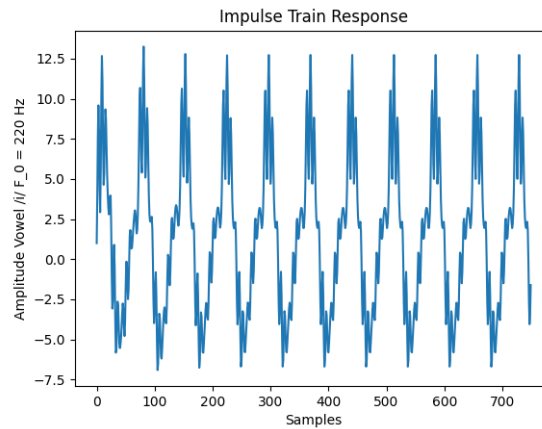




## 4.2 220 Hz







```

1  import numpy as np
2  import math
3  import matplotlib.pyplot as plt
4  from scipy import signal
5  from scipy.io.wavfile import write
6
7
8  B_1=100
9  playingtime=0.5
10 F_0=220
11 F_s=16000
12
13 x=[]
14 y=[]
15 y_1=[]
16 y_2=[]
17 time=[]
18 total_samples=int(F_s*playingtime)
19 for i in range(0,total_samples):
20     x.append(0)
21     y.append(0)
22     y_1.append(0)
23     y_2.append(0)
24     time.append(i)
25 i=0
26 count=0
27 while i<total_samples and count < int(F_0*playingtime):
28     x[i]= 1;
29     i=i+int(F_s/F_0);
30     count+=1
31
32 # Input Parameters
33 F_1=300

```

```

34 F_2=870
35 F_3=2240
36
37 def response(x,y,F):
38     r_i = np.exp(-B_1*3.142/F_s)
39     theta_i = 2*3.142*F/F_s
40     denominator=[1,-2*r_i*math.cos(theta_i),r_i*r_i]
41     y[0]=x[0] # impulse
42     y[1]= x[1]-y[0]*denominator[1]
43     for i in range(2,total_samples):
44         y[i] = x[i]-y[i-1]*denominator[1]-y[i-2]*denominator[2];
45     return y
46
47 y=response(x,y,F_1)
48 y_1=response(y,y_1,F_2)
49 y_2=response(y_1,y_2,F_3)
50
51 plt.title('Impulse Train Response')
52 plt.plot(time[0:750], y_2[0:750])
53 plt.ylabel('Amplitude Vowel /u/ F_0 = 220 Hz')
54 plt.xlabel('Samples')
55 plt.savefig('Q4_u_220.png')
56
57 y_2 = np.array(y_2)
58 write('Q4_u_220.wav', F_s, y_2)
59
60

```

The Audio files of 'A' and 'U' do sound like them, but there is a little roughness in 'I'. But the voices are a little robotic