DSP Project

Music Equalizer

- **Mohamed Ramadan Mohamed - 56**
- **Hassan Ahmed Hassan - 26**
- **Omar Mohamed Eissa - 47**

About Project:

A music equalizer using matlab filters.

Outline:

- Project Explanation.
- Filters Plots .
    - Ideal Filters.
    - IIR - ButterWorth.
    - FIR.
        - Equiripple.
        - Least-square.
        - Rectangular.
        - Chebyshev.
        - Hamming .
        - Blackman-harris.
- Ideal Filters.
- GUI Exploration.
- Sample Runs.
- Analysis.

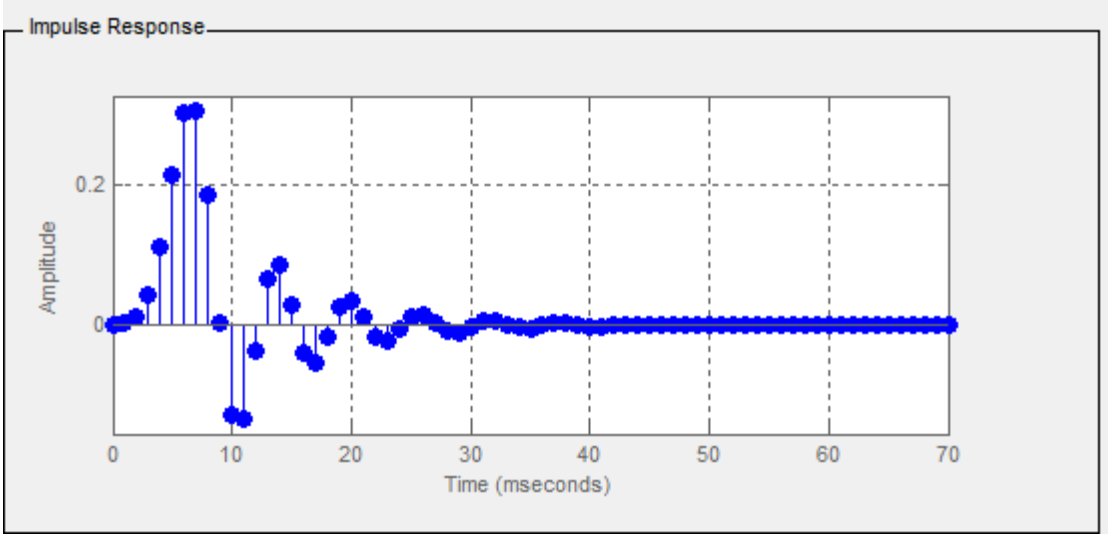
Project Explanation

It is required to develop audio equalizer using MATLAB. One common Audio equalizer is the Winamp program as shown in figure. The equalizer function is to vary the gain of each specific band as the user prefers (i.e. if the user likes base he will increase the gainof low frequencies).

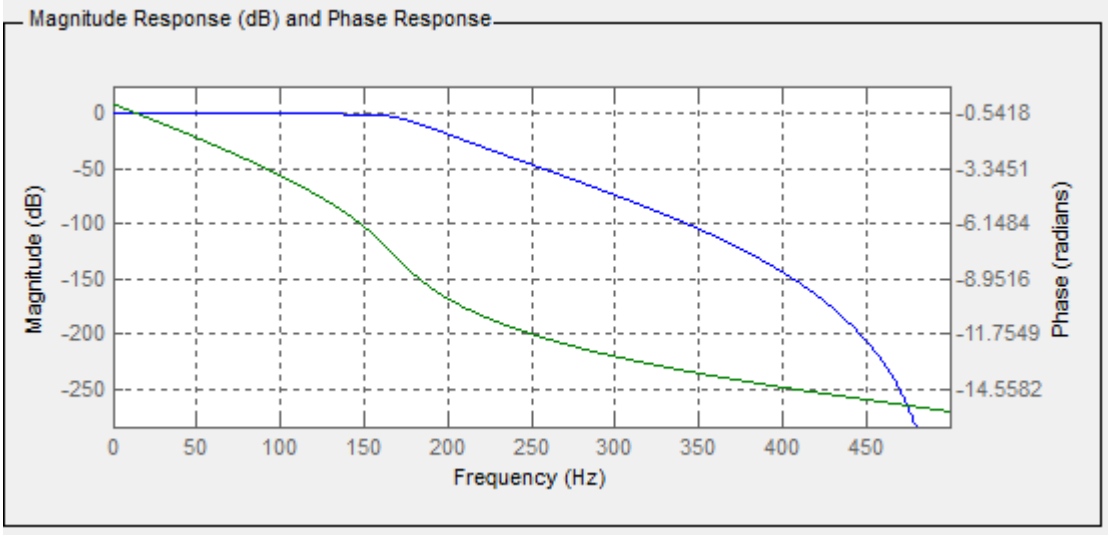
IIR - ButterWorth

As Low Pass Filter:

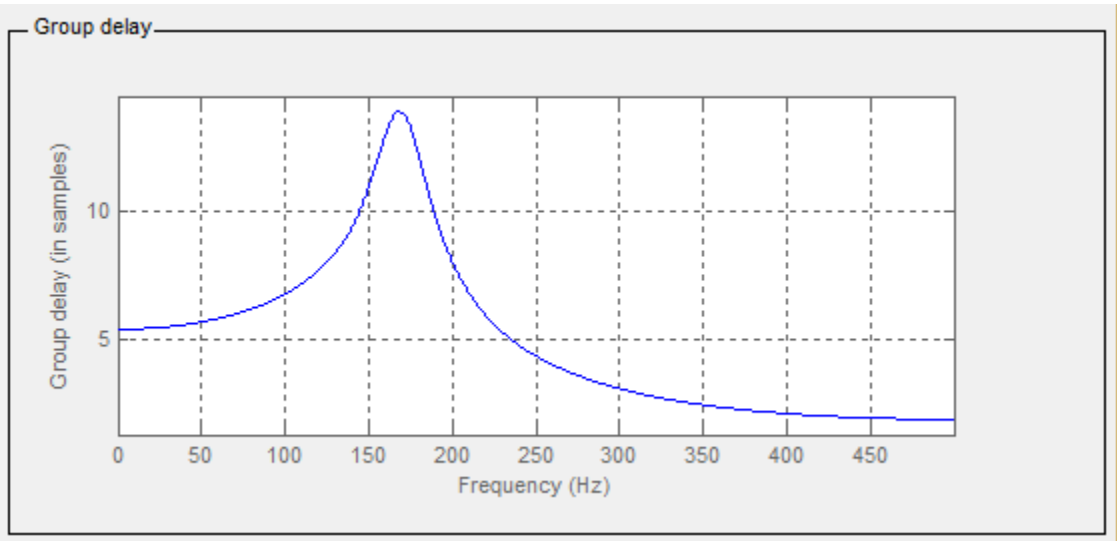
Impulse Response:



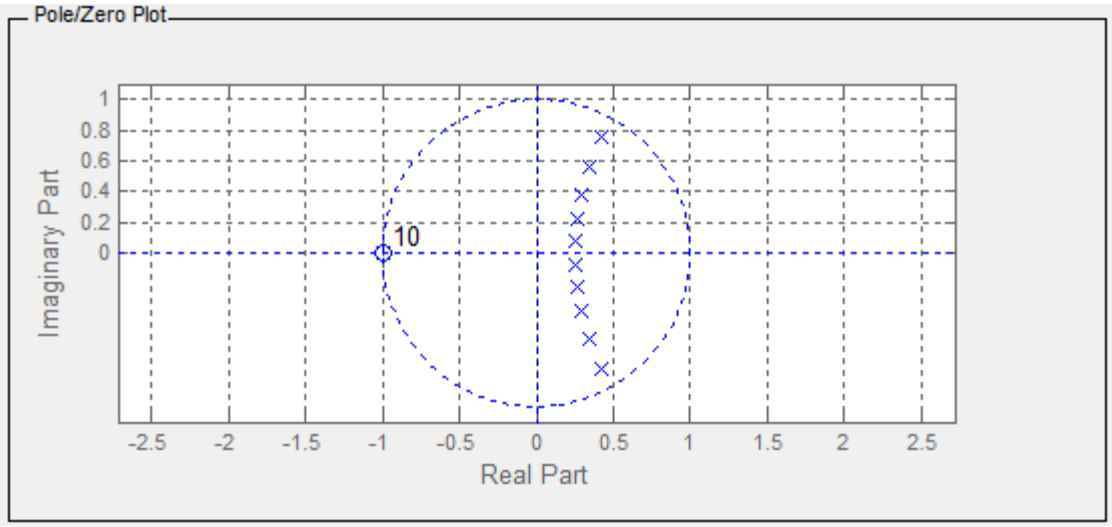
Frequency Response:



Group Delay:

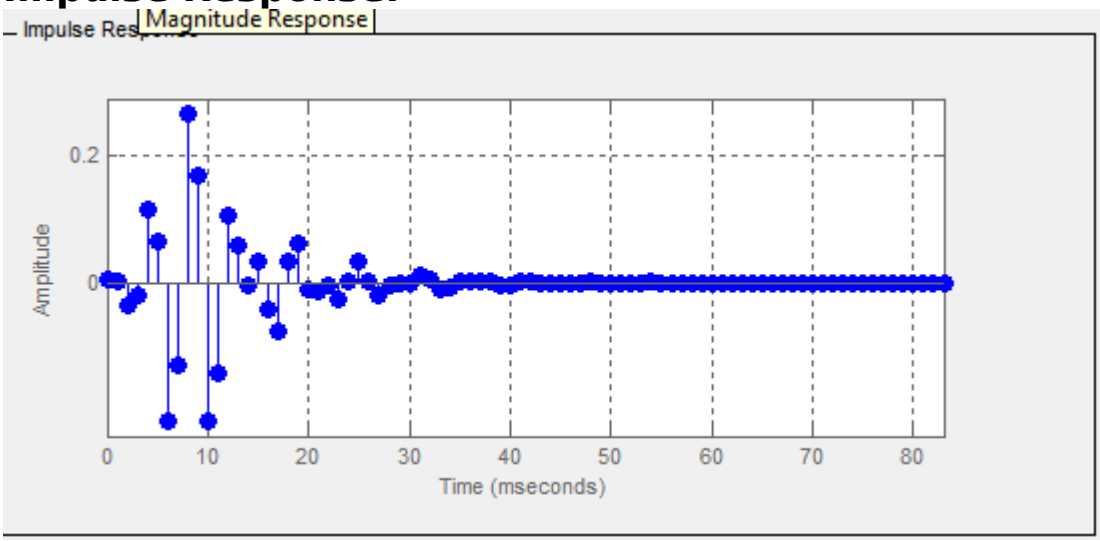


Poles and Zeros:

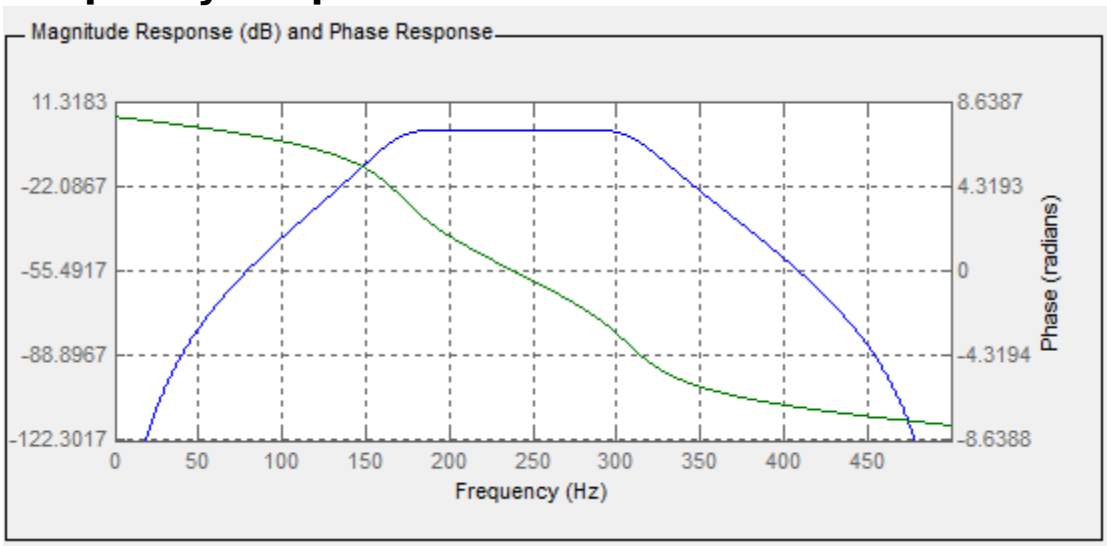


Band Pass(170-310):

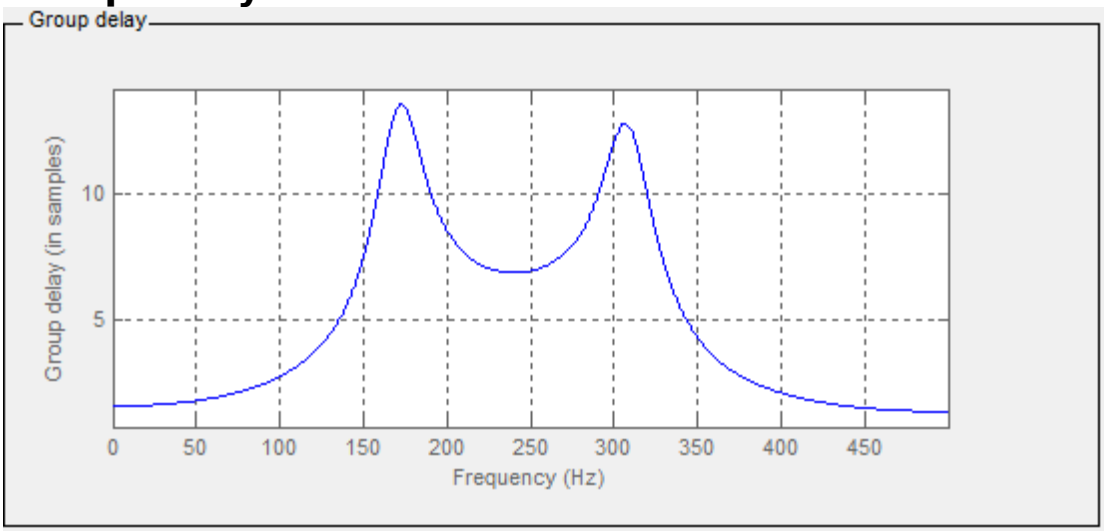
Impulse Response:



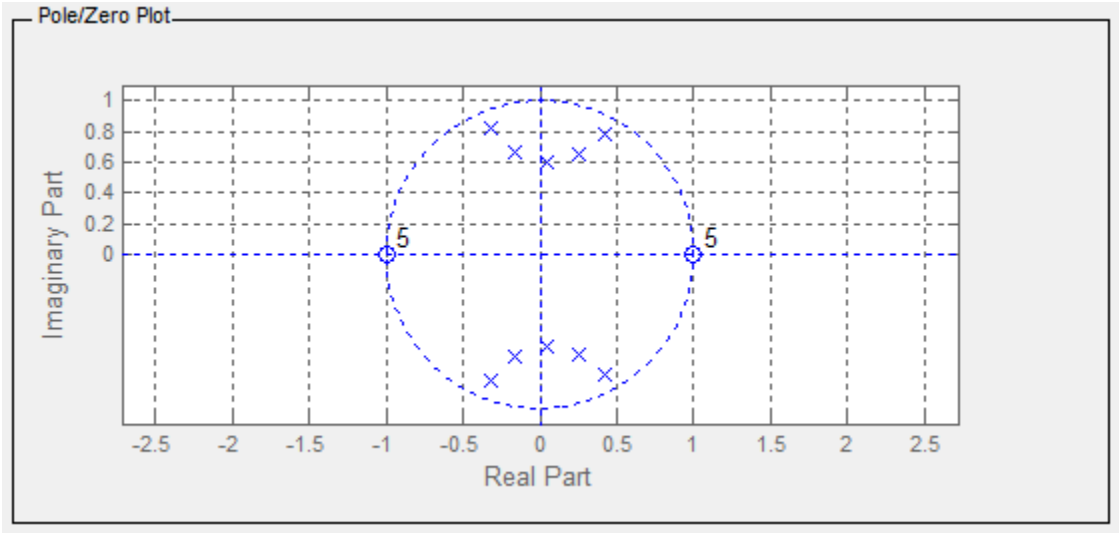
Frequency Response:



Group Delay:

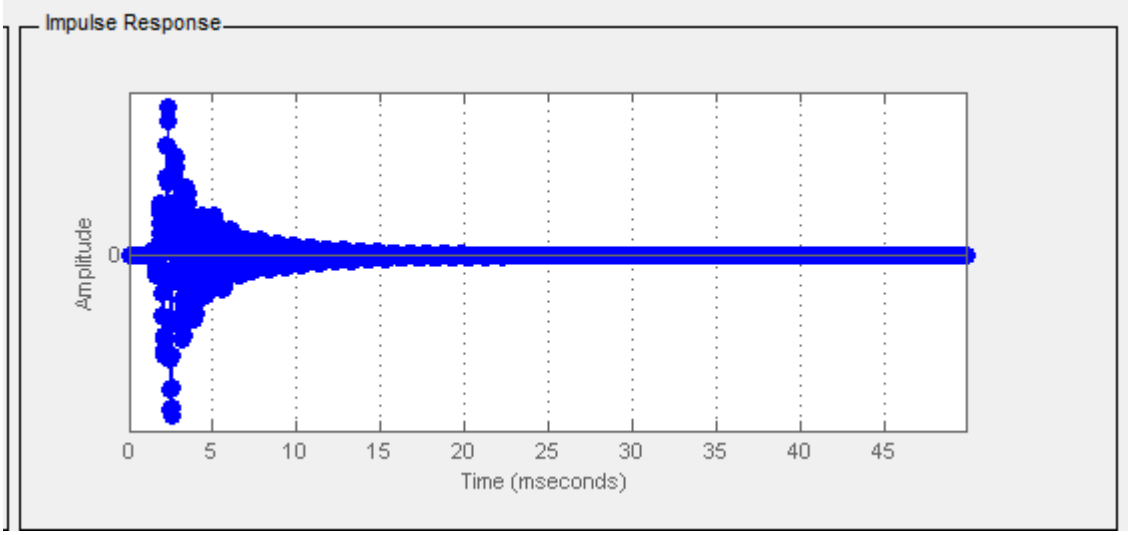


Poles And Zeros:

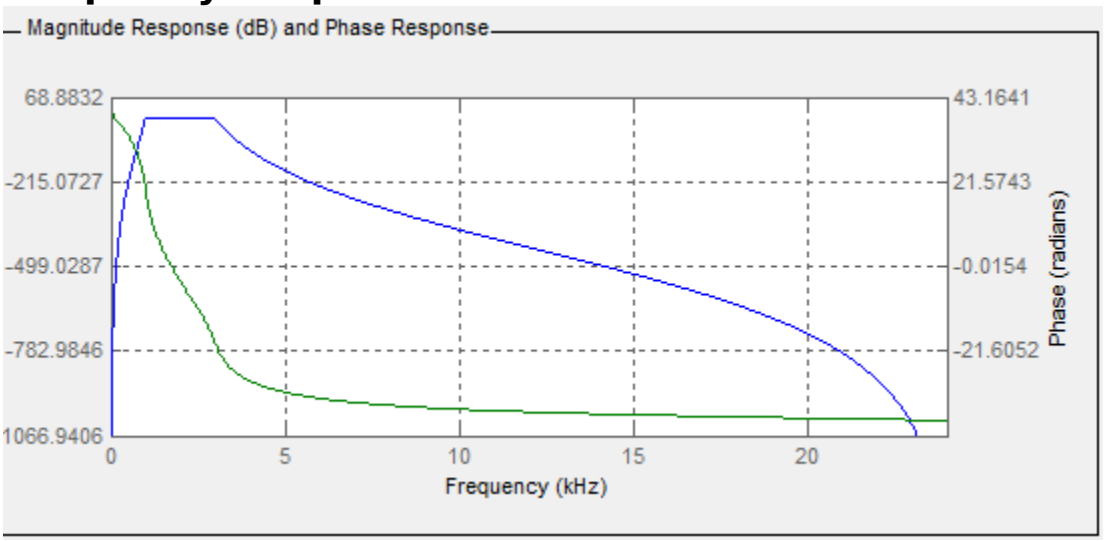


Band Pass(1k-3k):

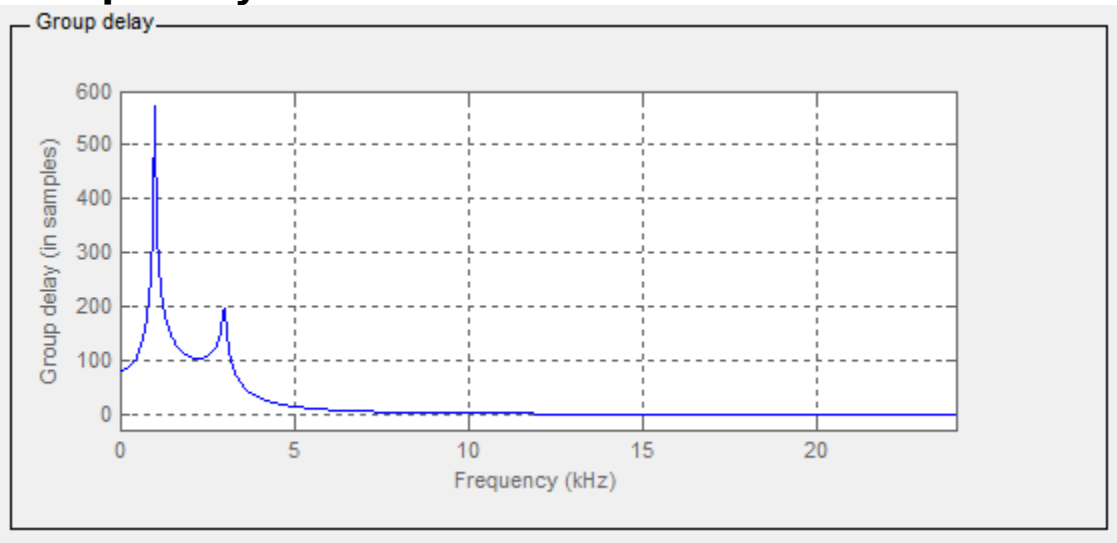
Impulse Response:



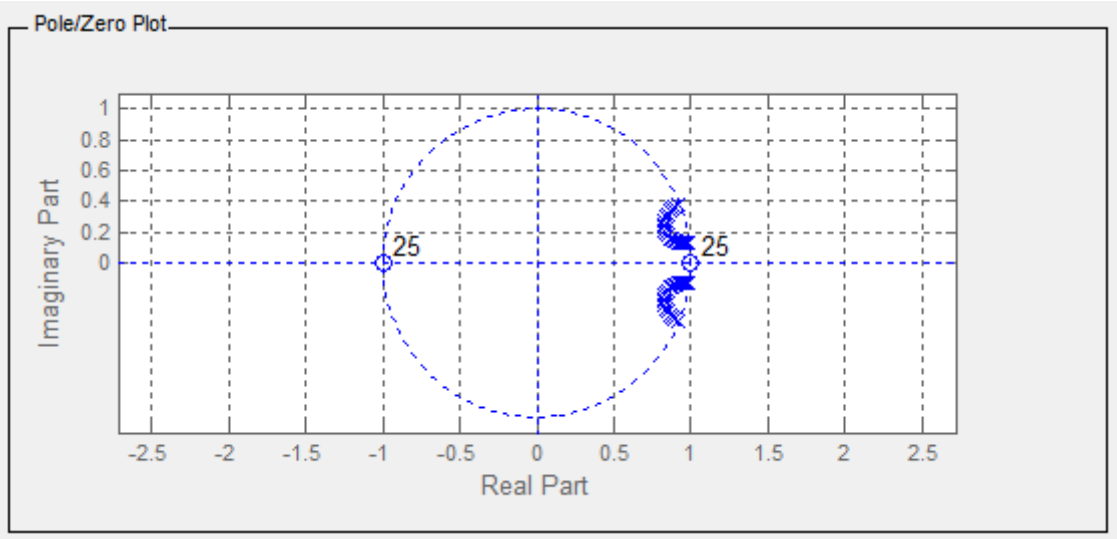
Frequency Response:



Group Delay:



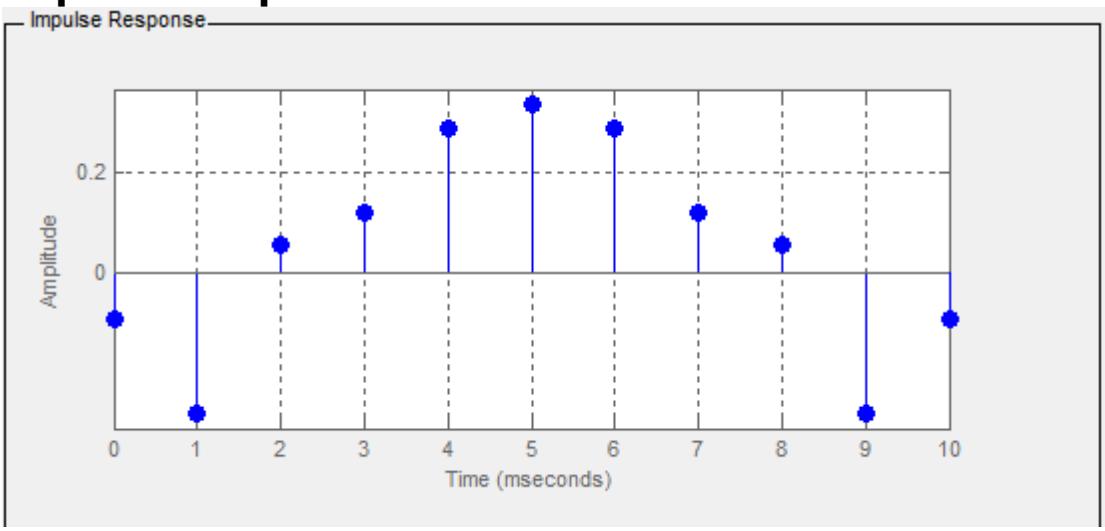
Poles And Zeros:



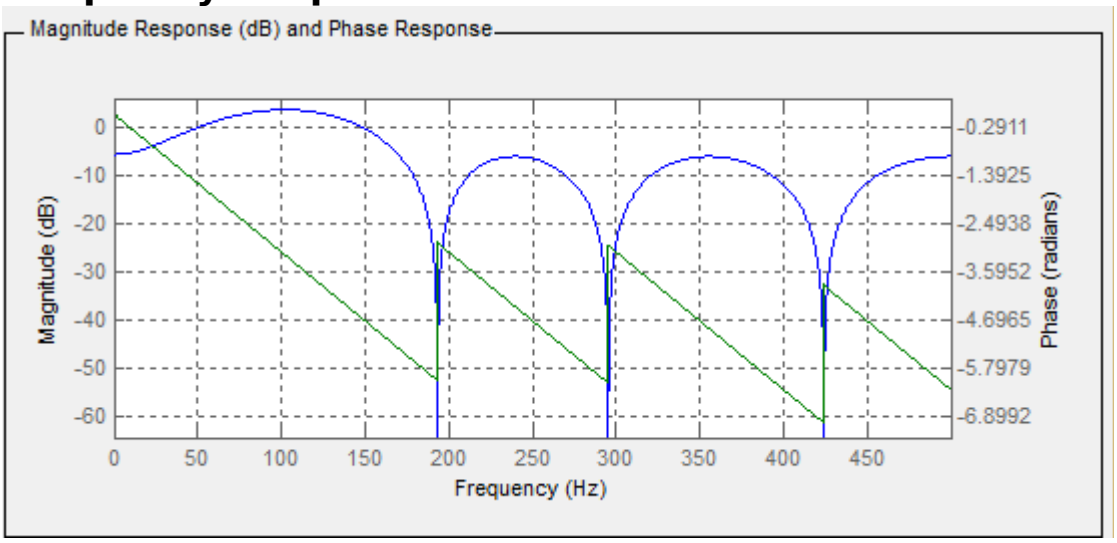
FIR
Equiribble

As Low Pass Filter:

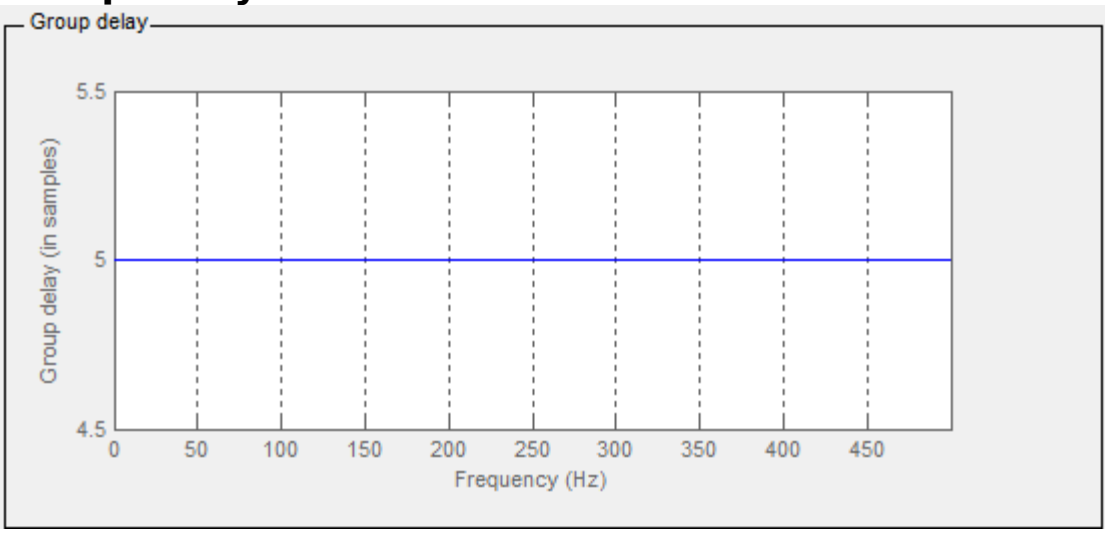
Impulse Response:



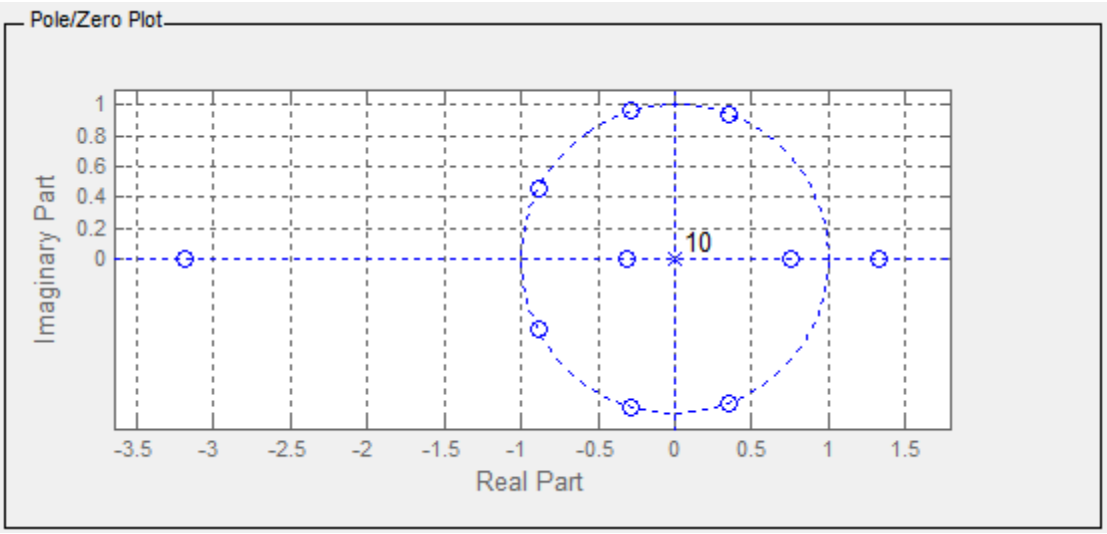
Frequency Response:



Group Delay:

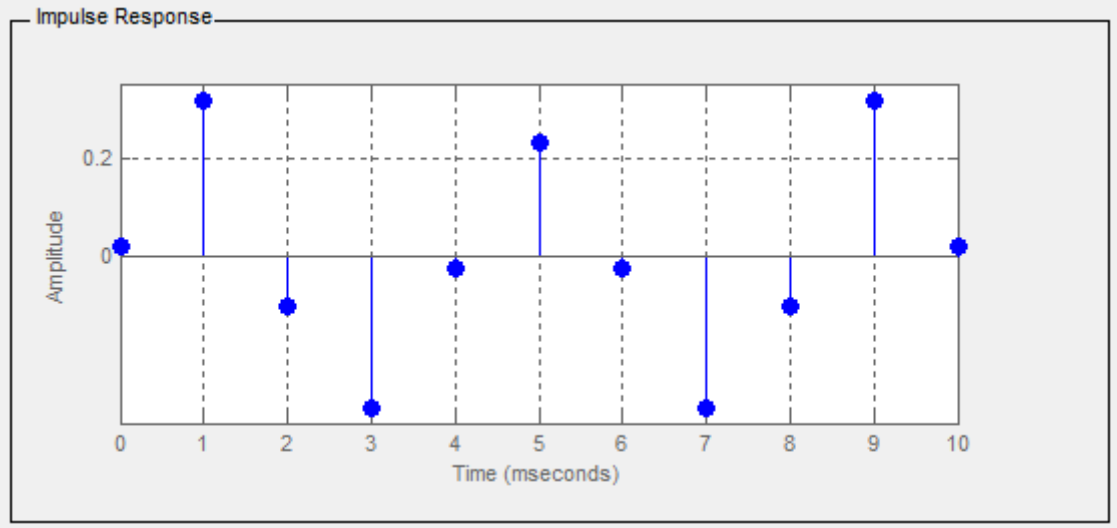


Poles and Zeros:

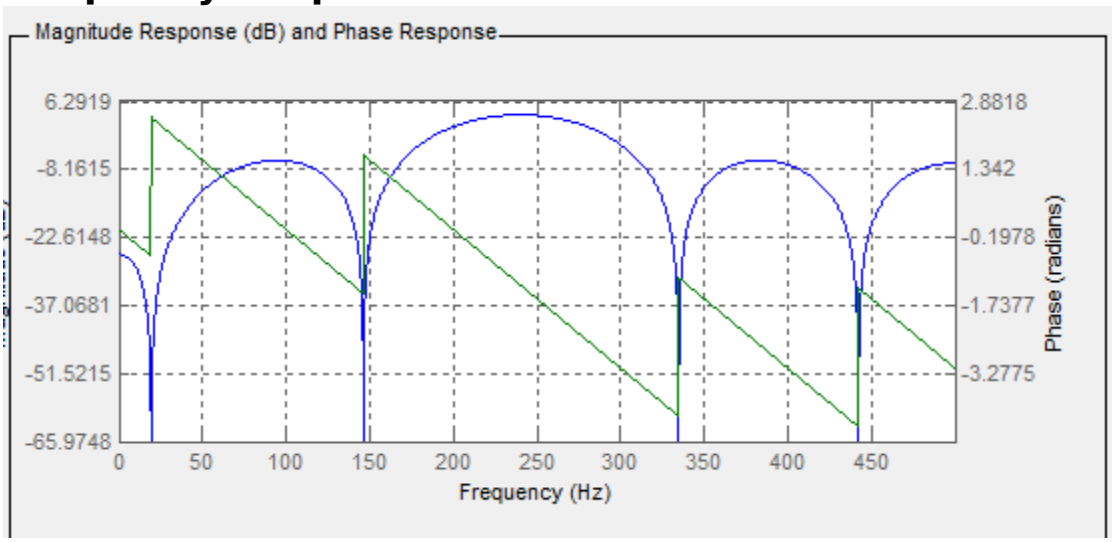


Band Pass(170-310):

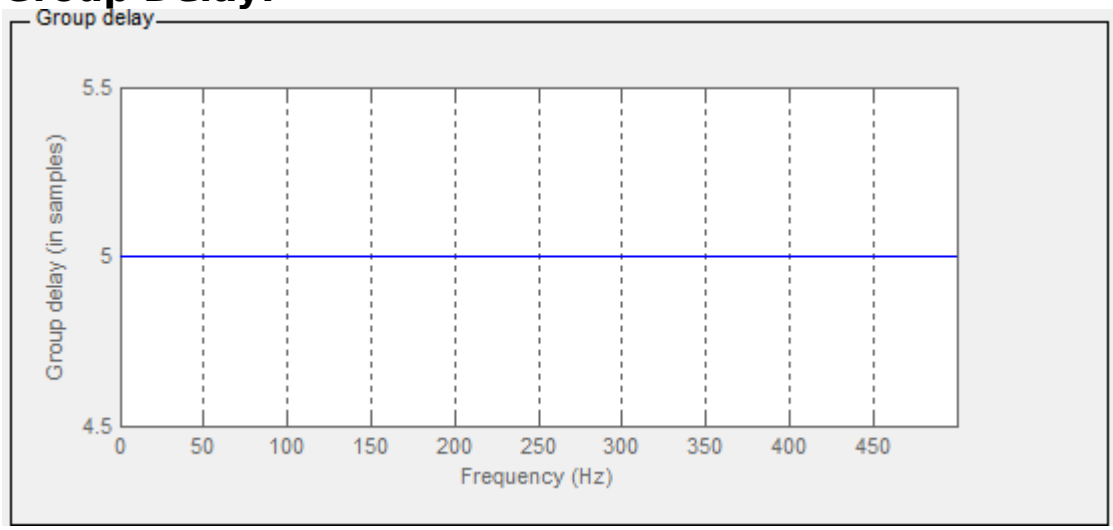
Impulse Response:



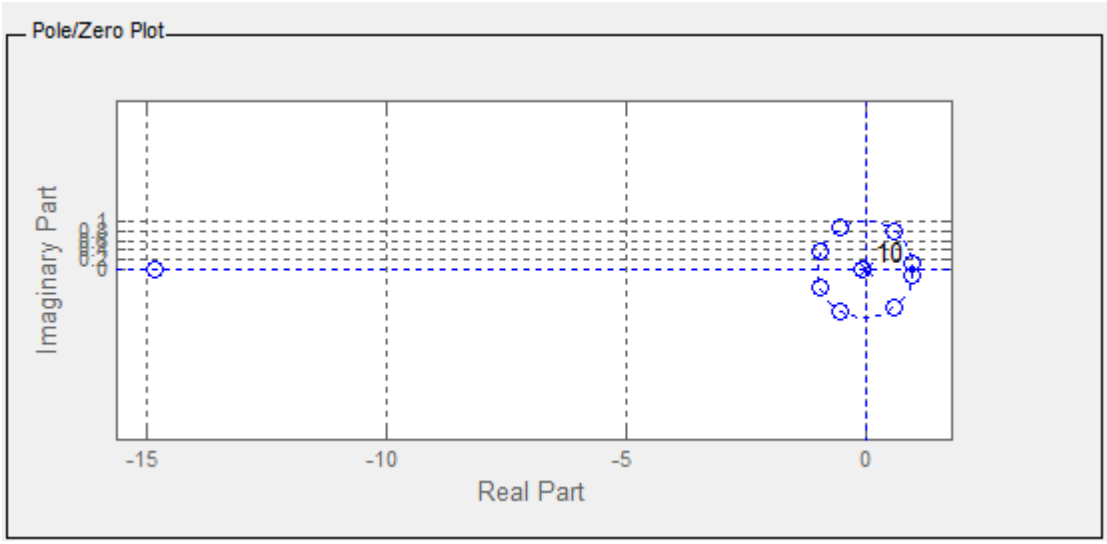
Frequency Response:



Group Delay:

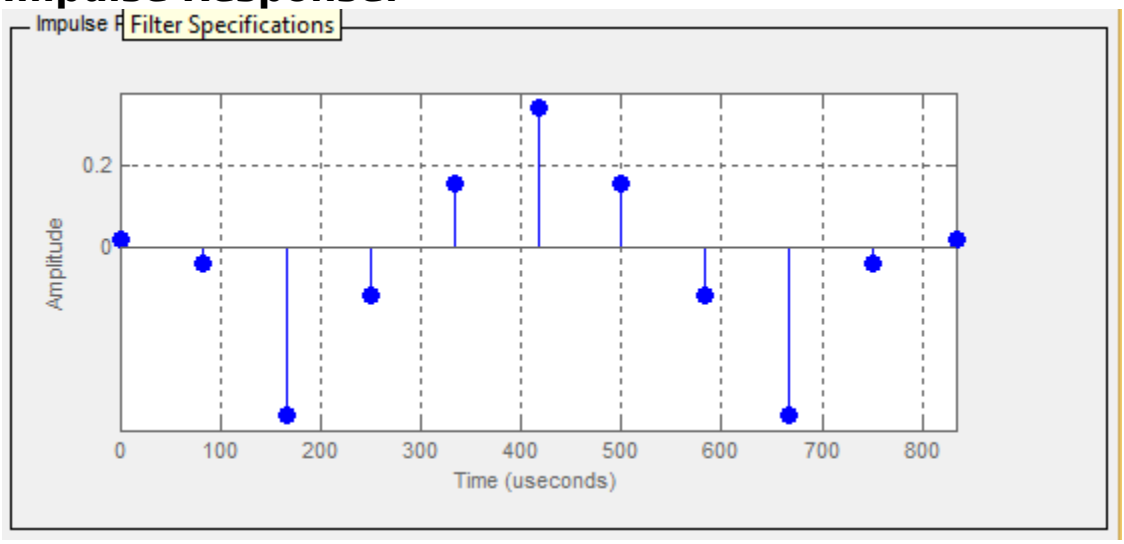


Poles And Zeros:

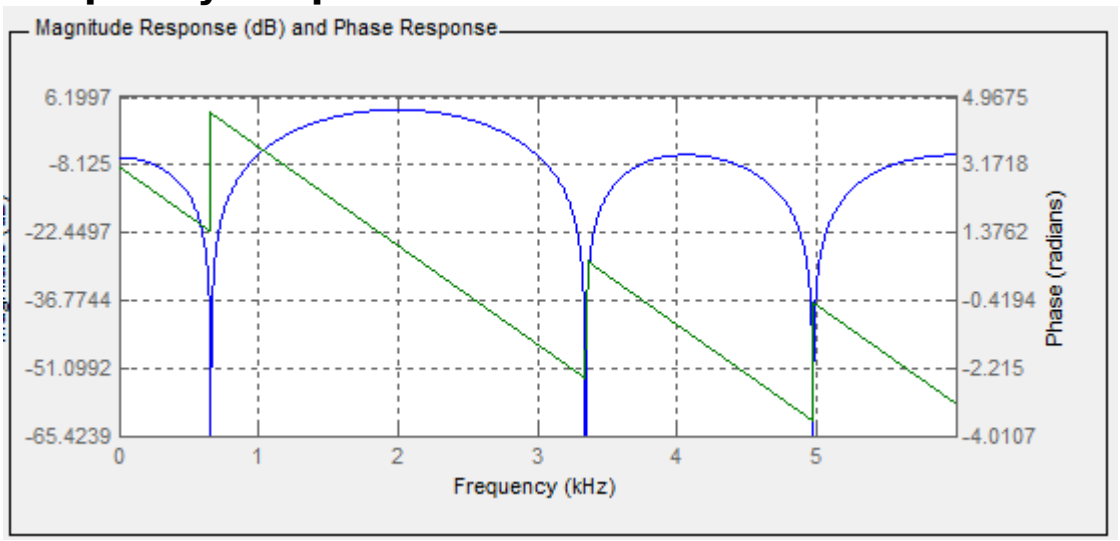


Band Pass(1000-3000):

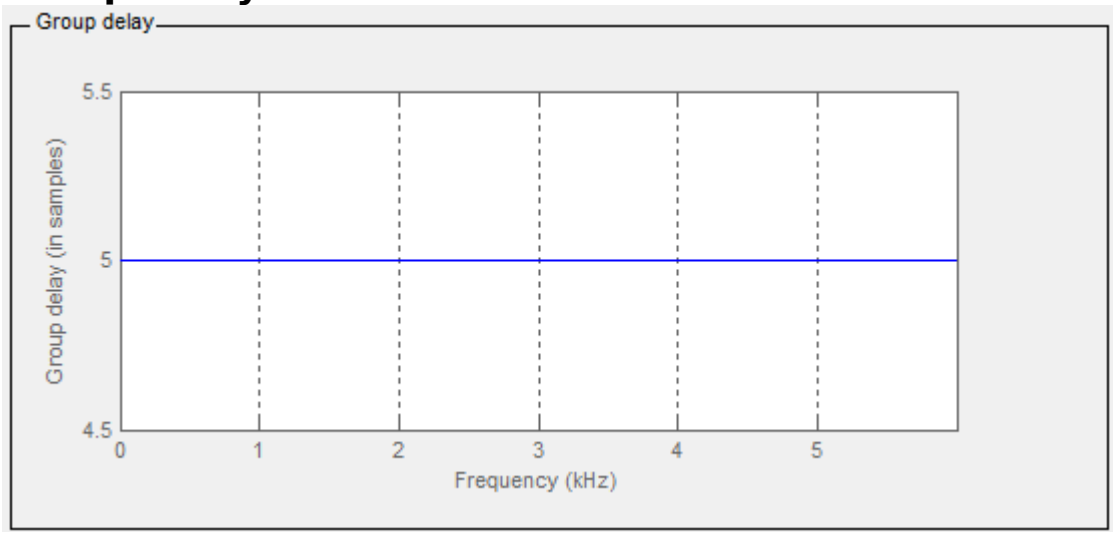
Impulse Response:



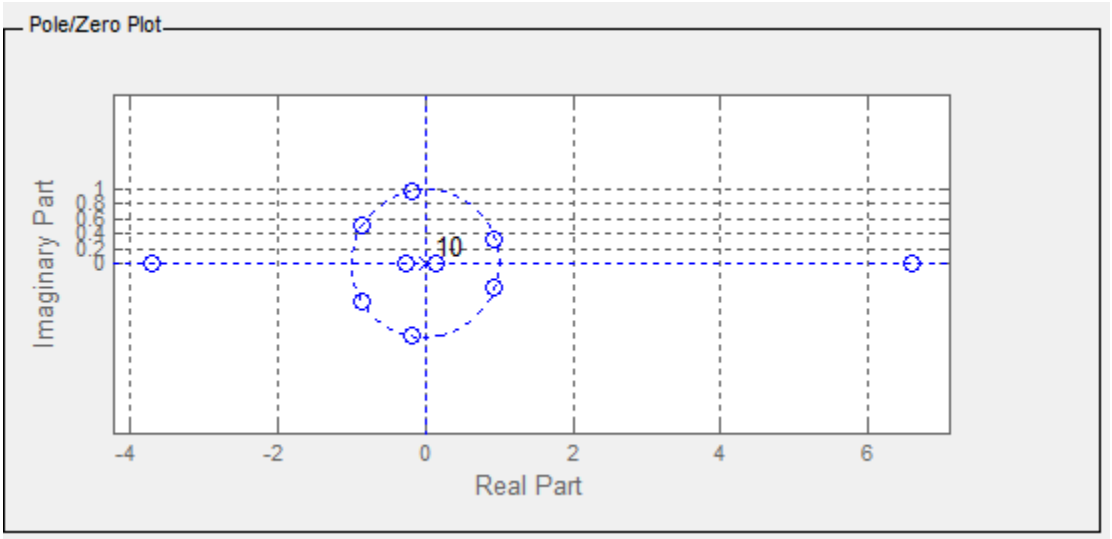
Frequency Response:



Group Delay:



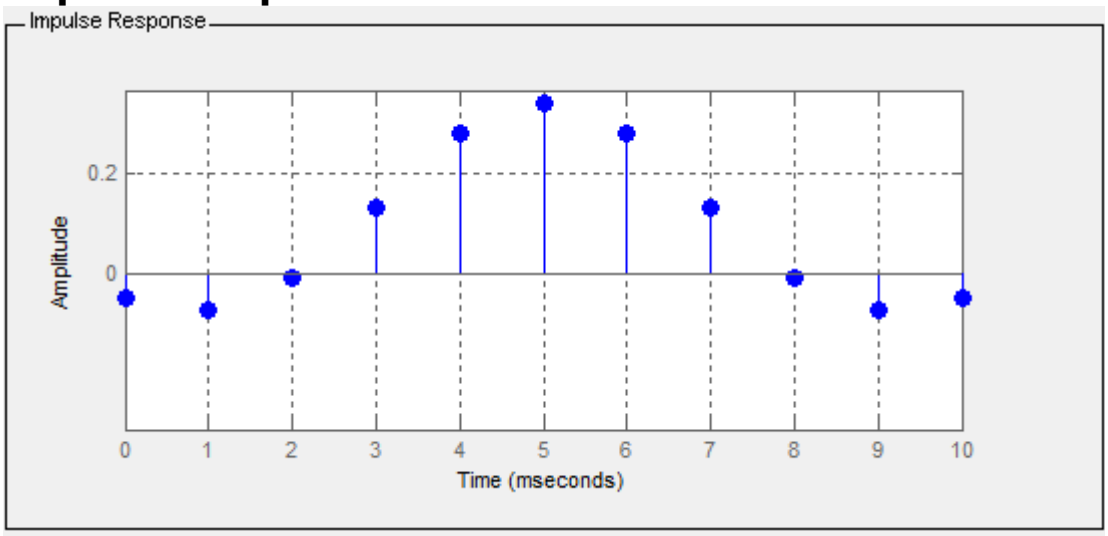
Poles And Zeros:



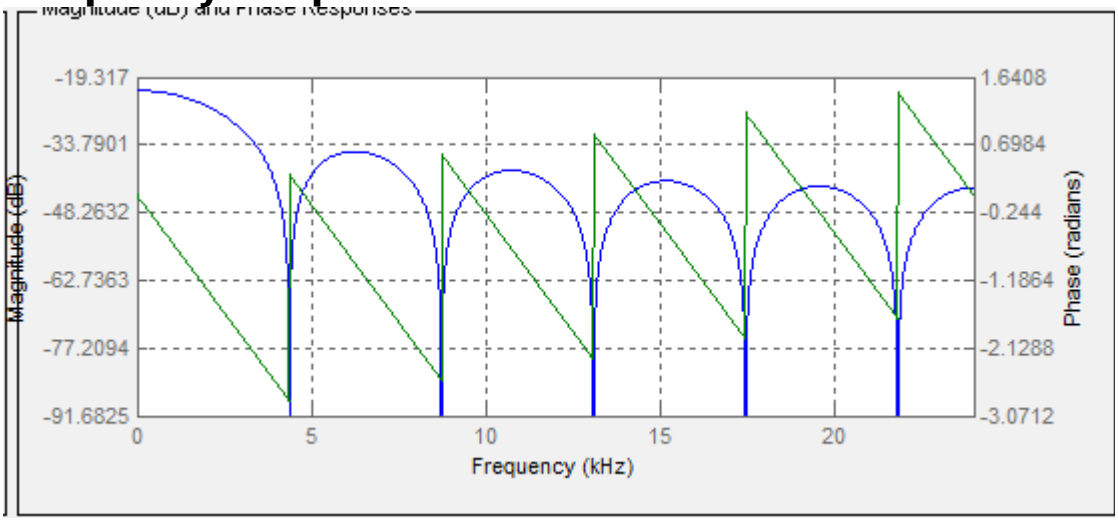
Least Square

As Low Pass Filter:

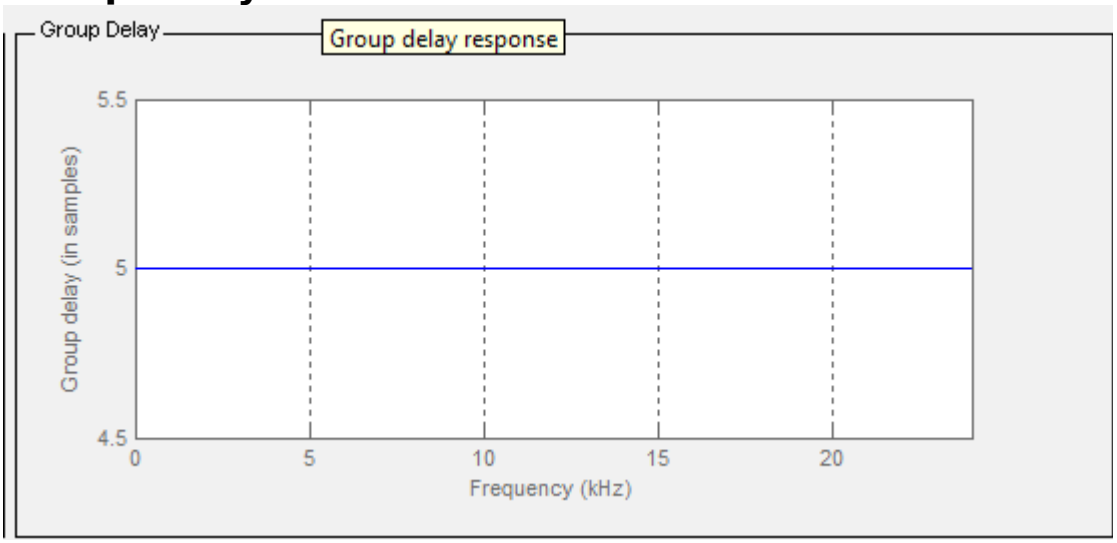
Impulse Response:



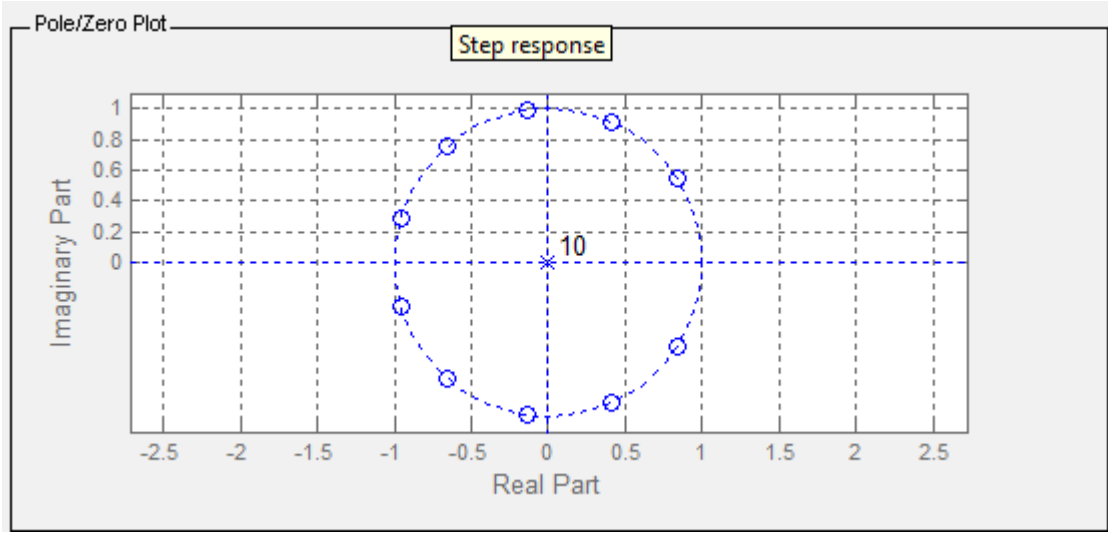
Frequency Response:



Group Delay:

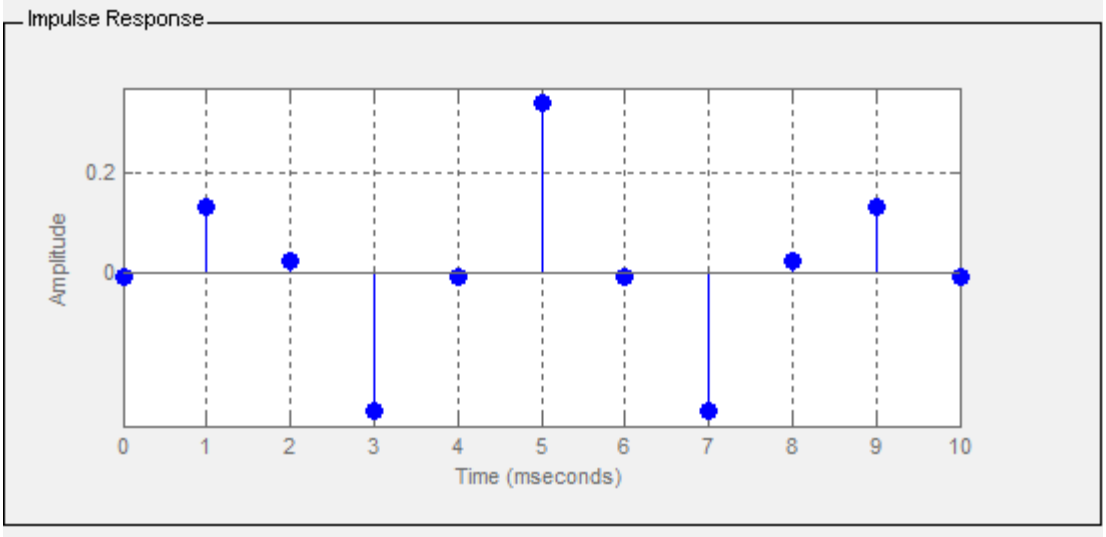


Poles and Zeros:

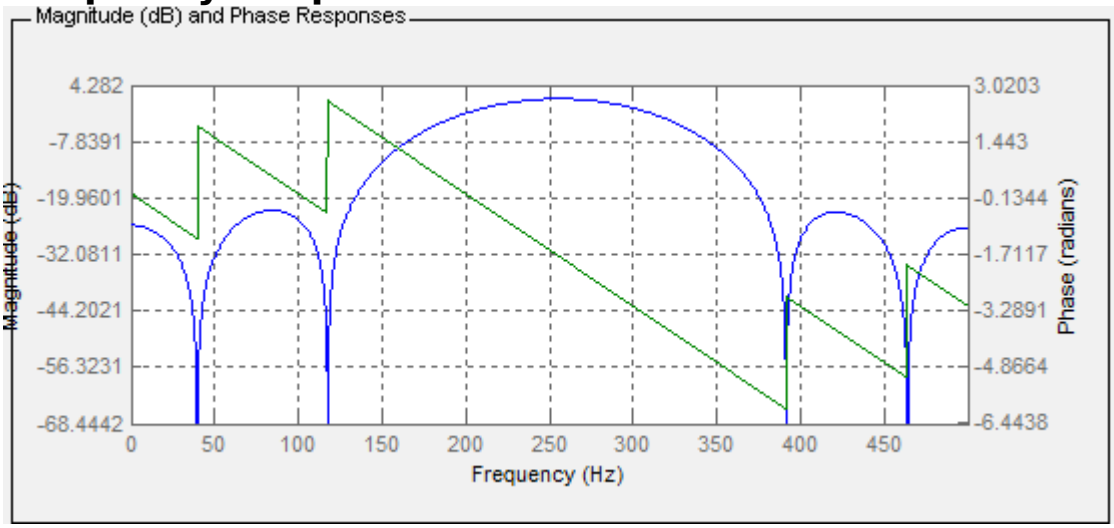


Band Pass(170-310):

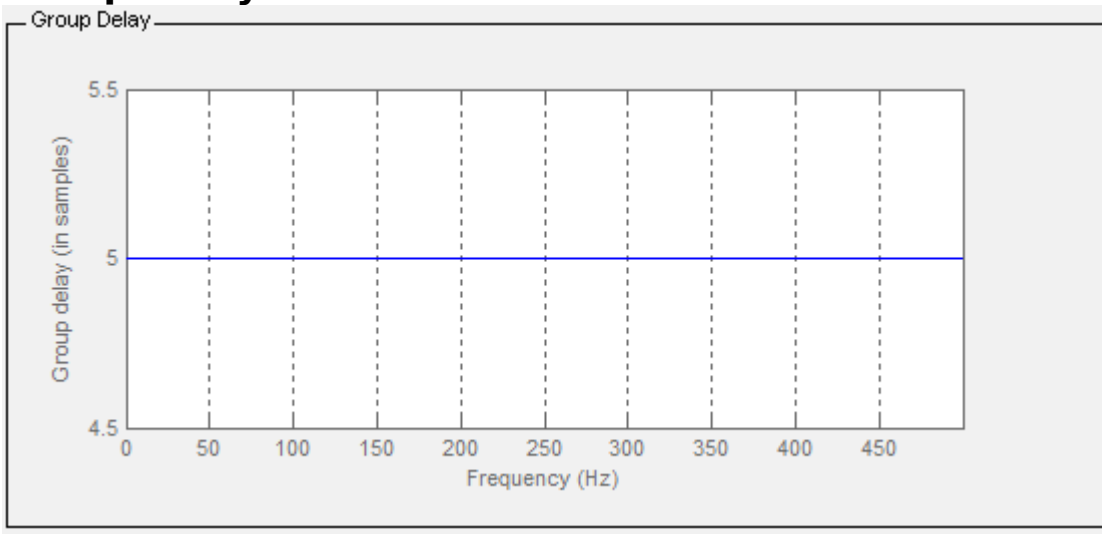
Impulse Response:



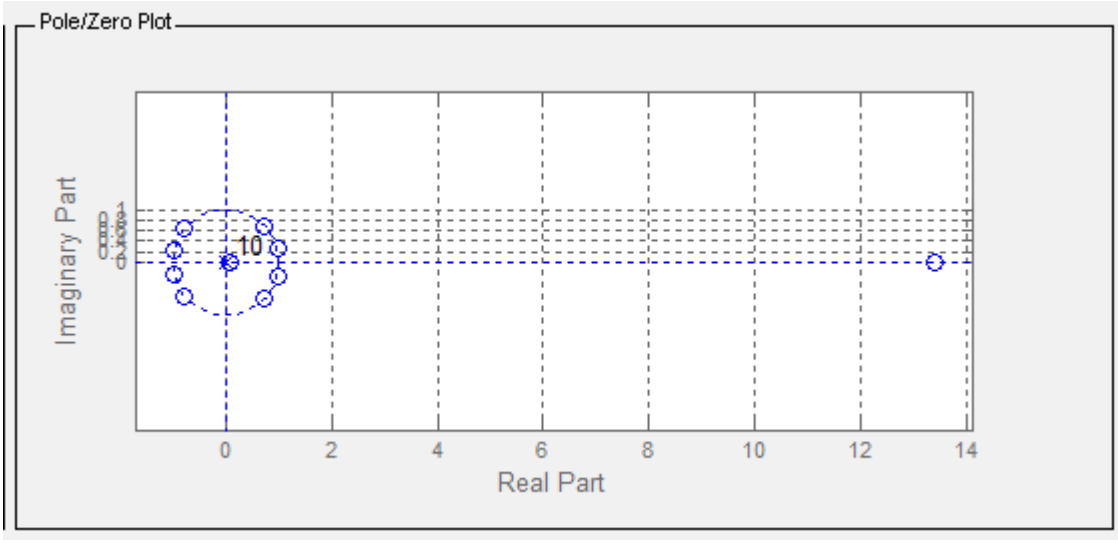
Frequency Response:



Group Delay:

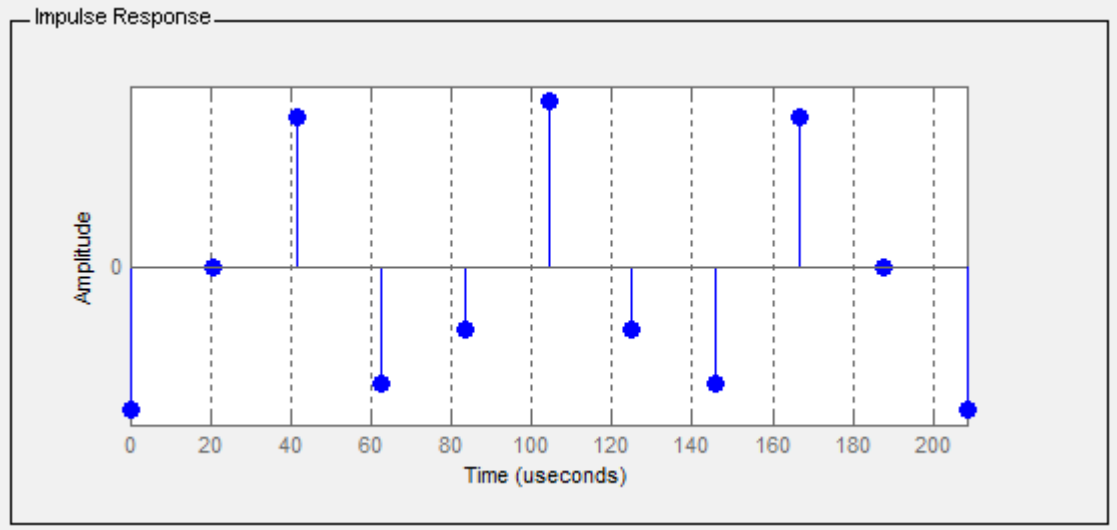


Poles And Zeros:

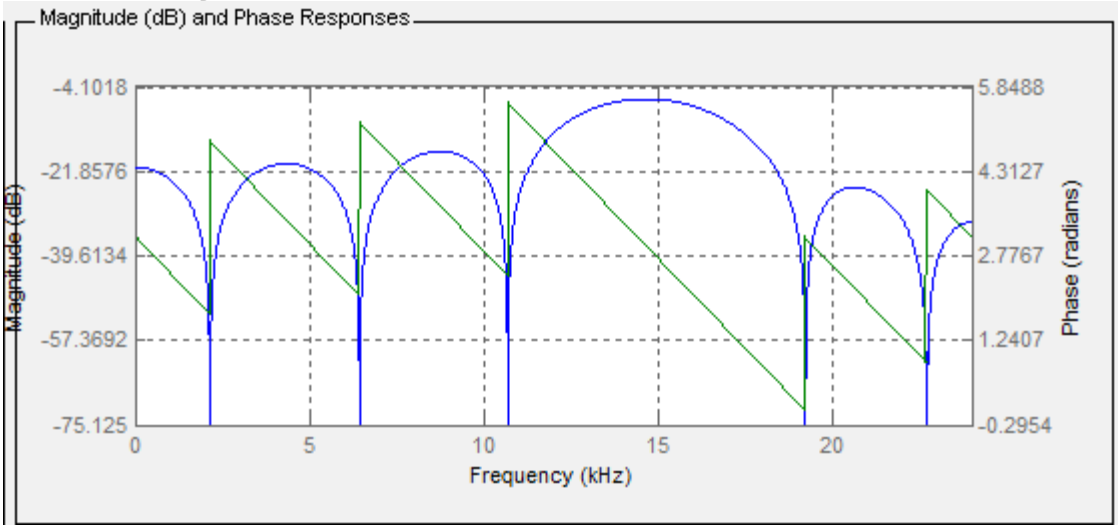


Band Pass(14k-16k):

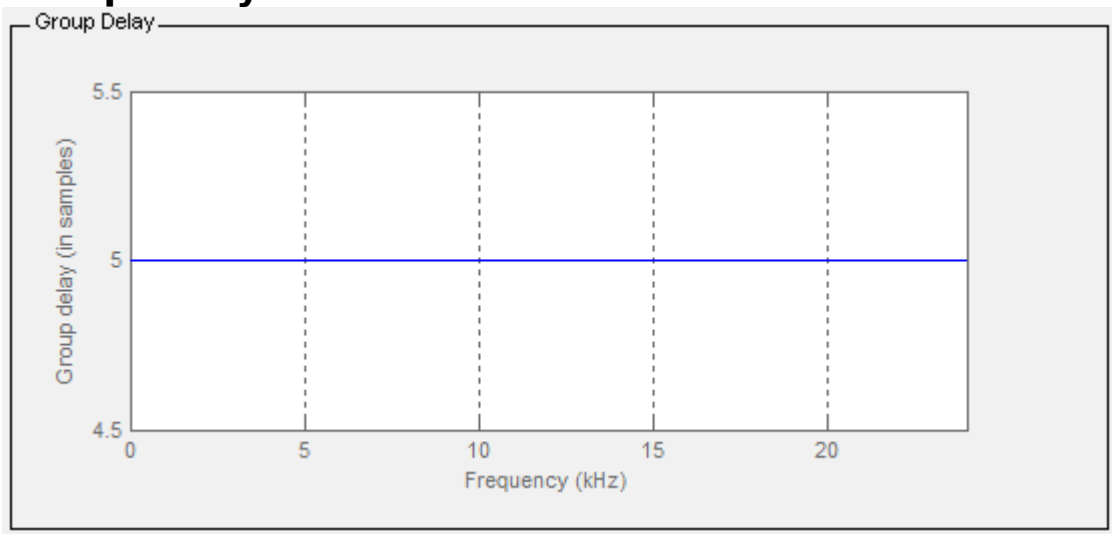
Impulse Response:



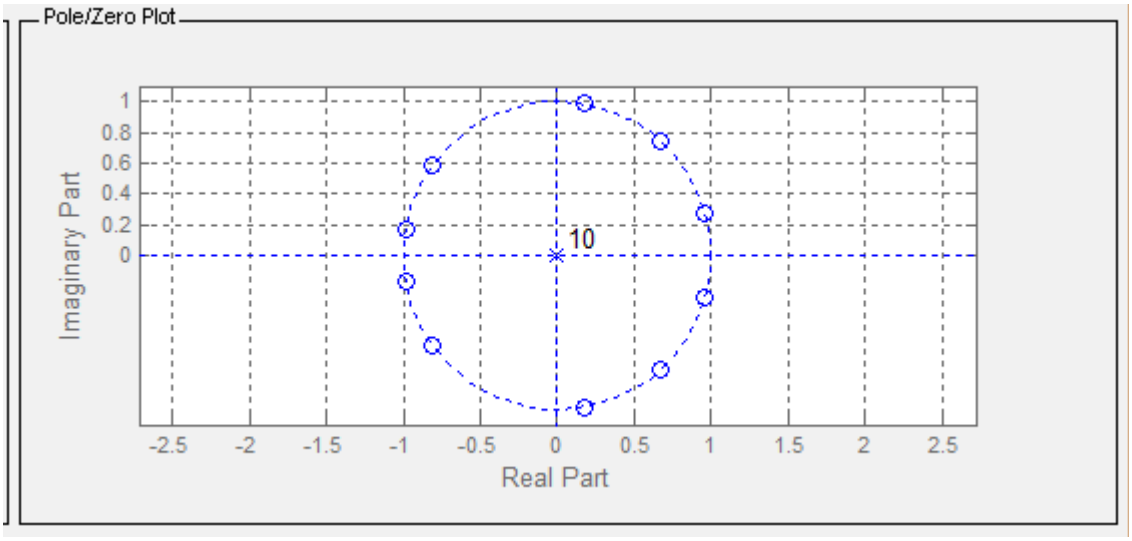
Frequency Response:



Group Delay:



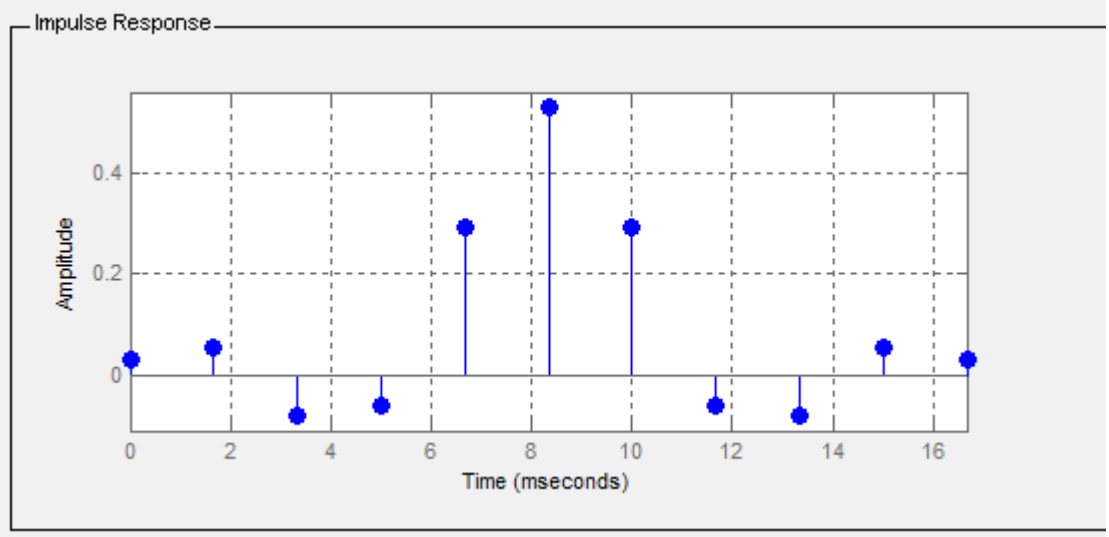
Poles And Zeros:



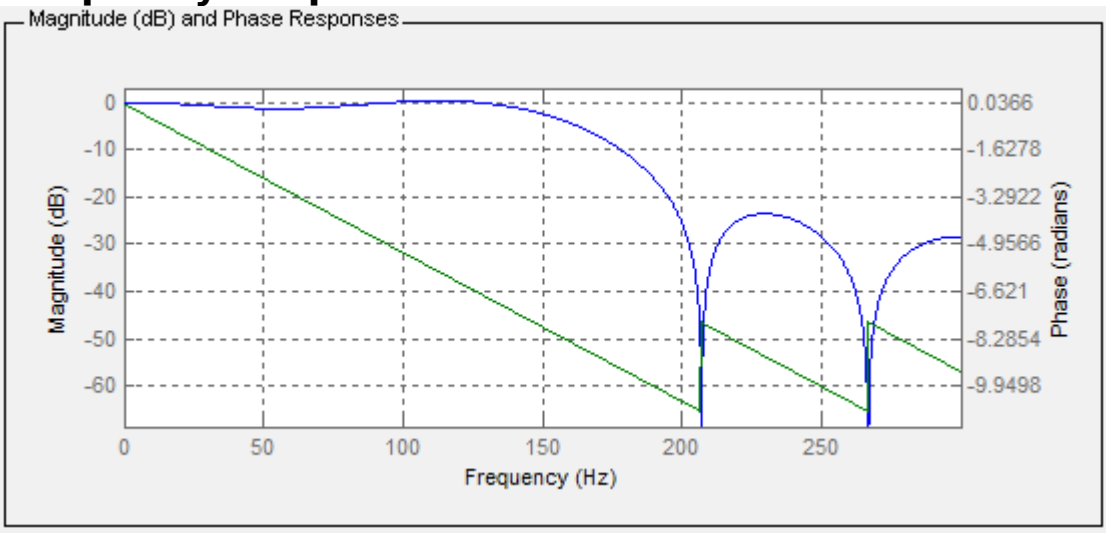
Rectangle

As Low Pass Filter:

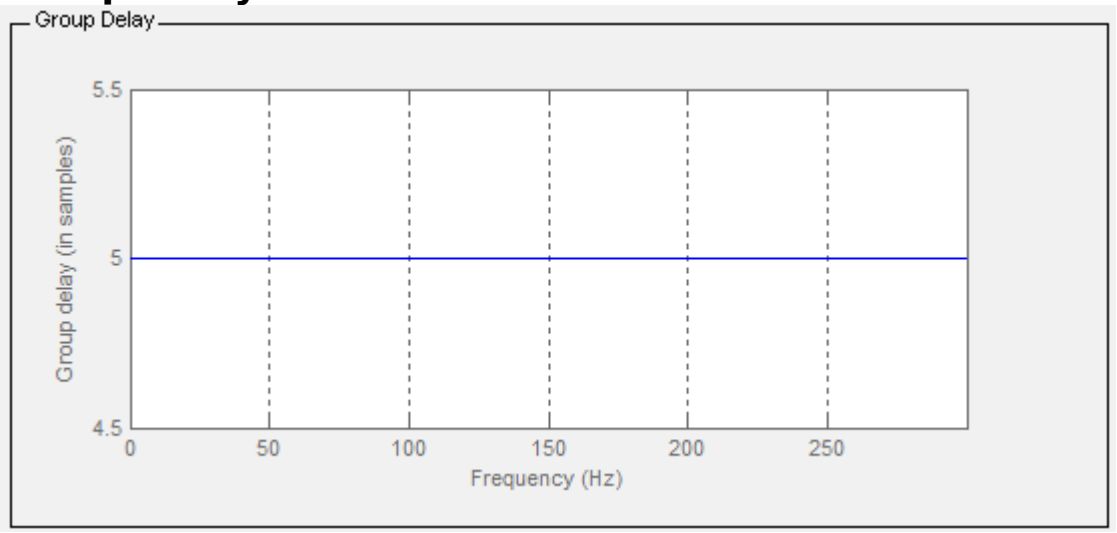
Impulse Response:



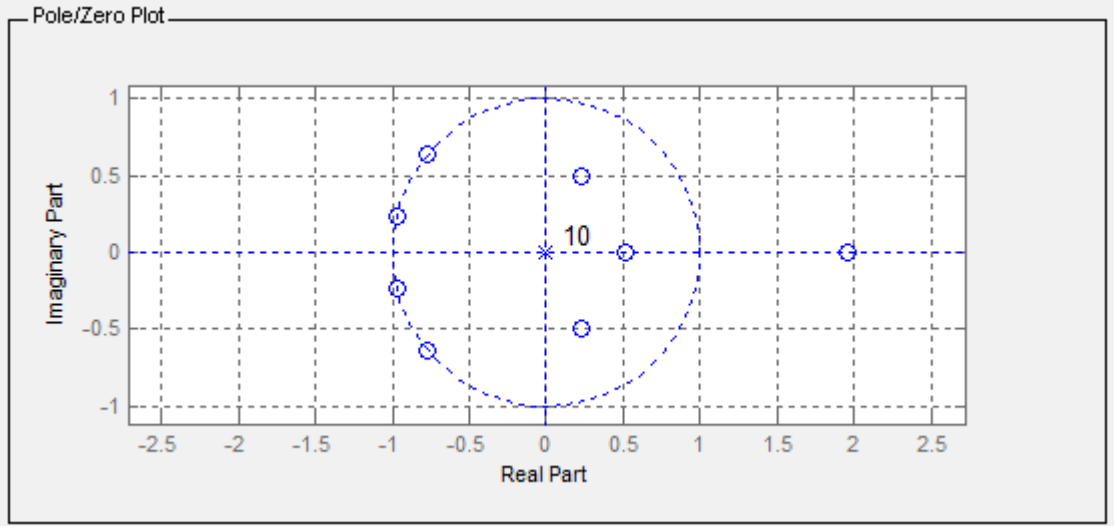
Frequency Response:



Group Delay:

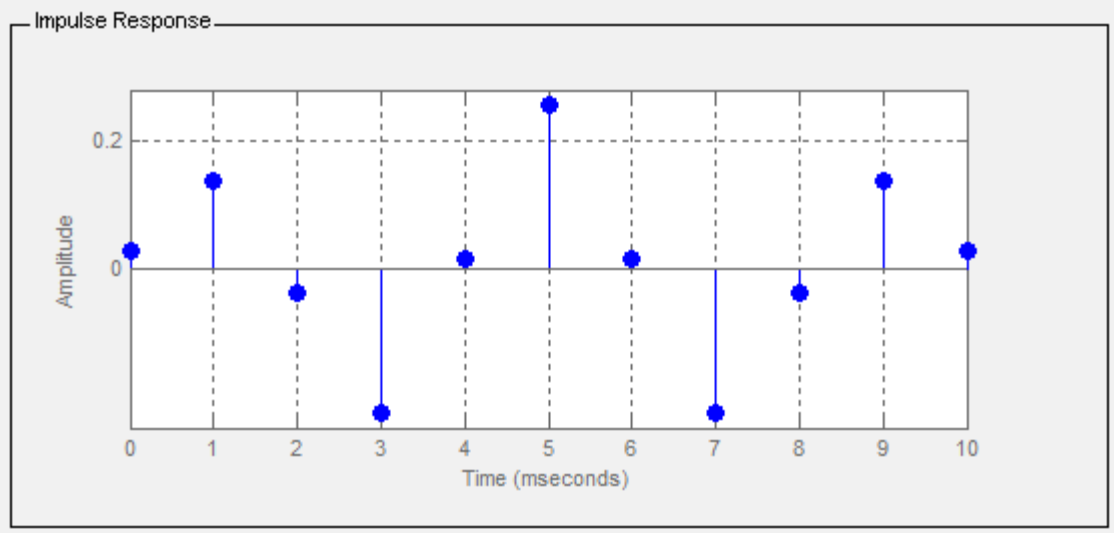


Poles and Zeros:

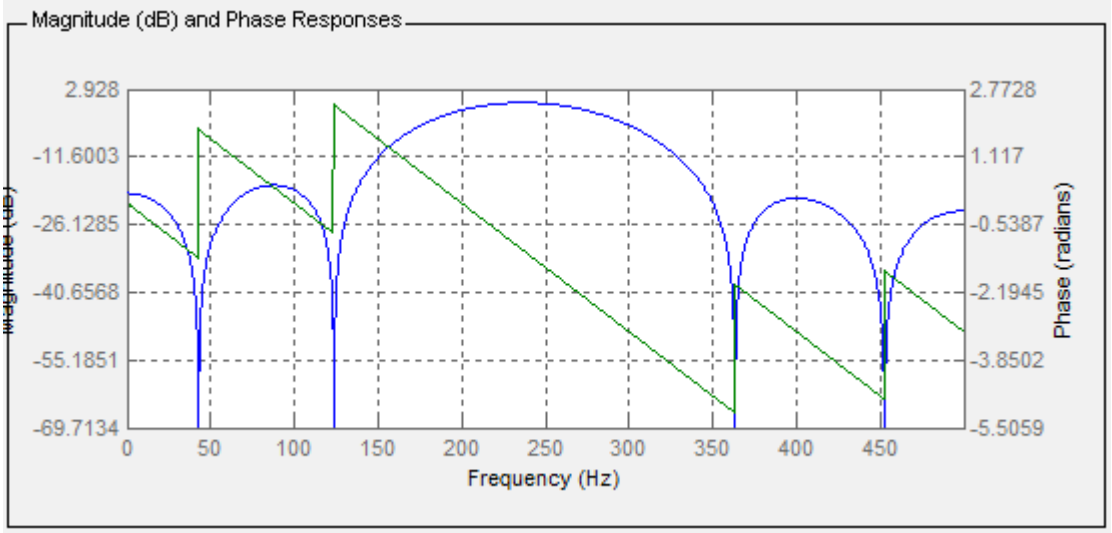


Band Pass(170-310):

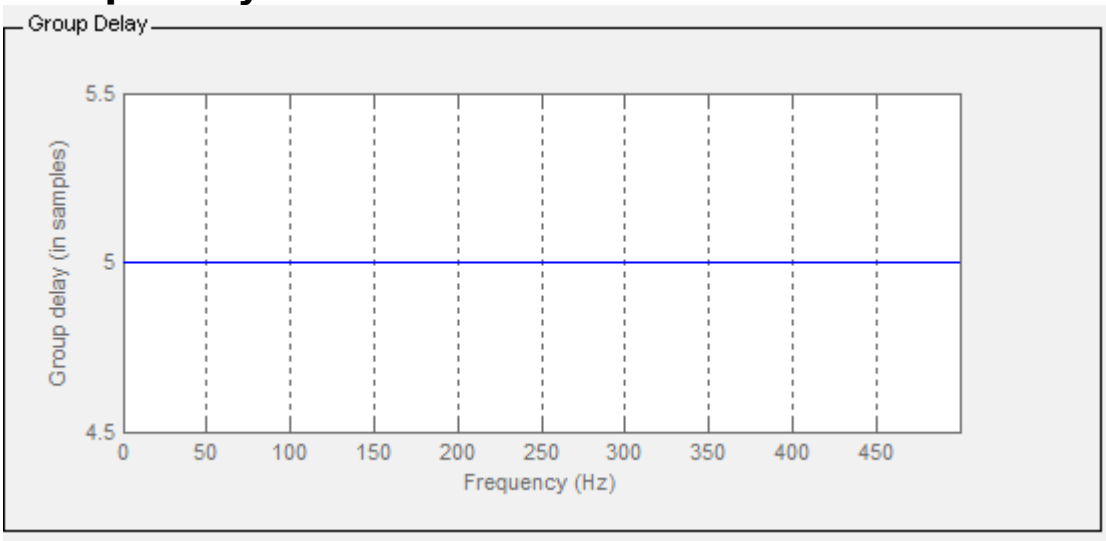
Impulse Response:



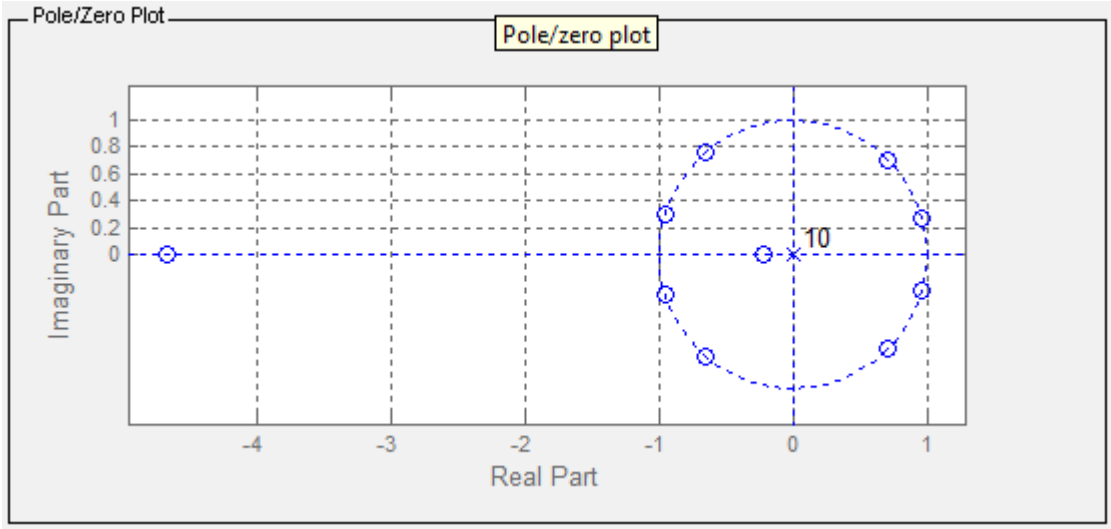
Frequency Response:



Group Delay:

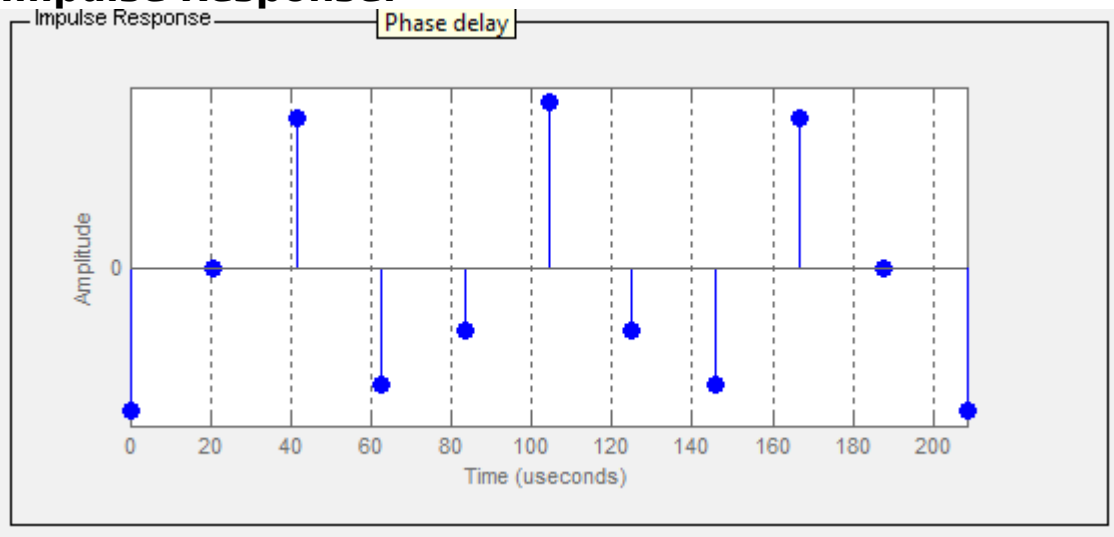


Poles And Zeros:

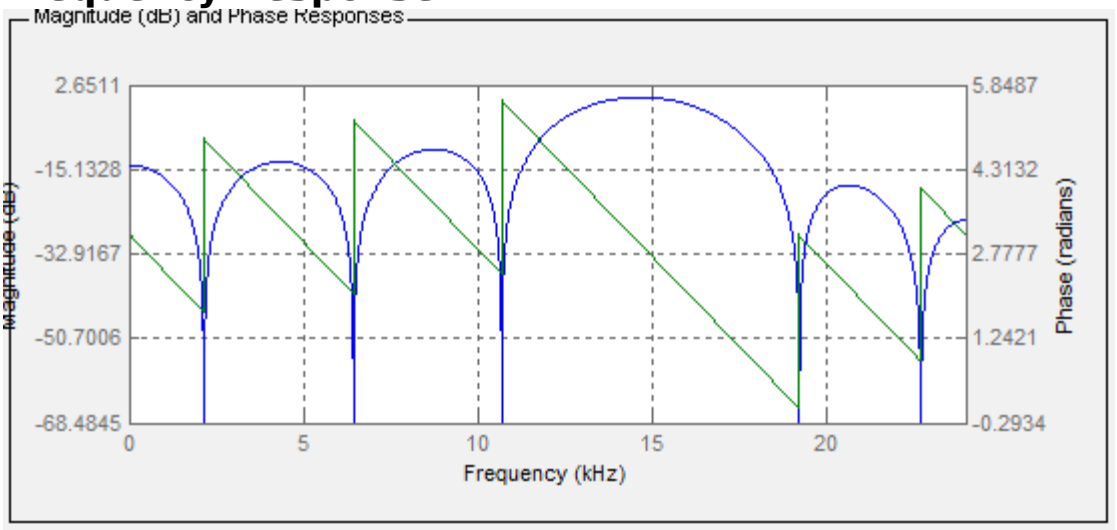


Band Pass(14k-16k):

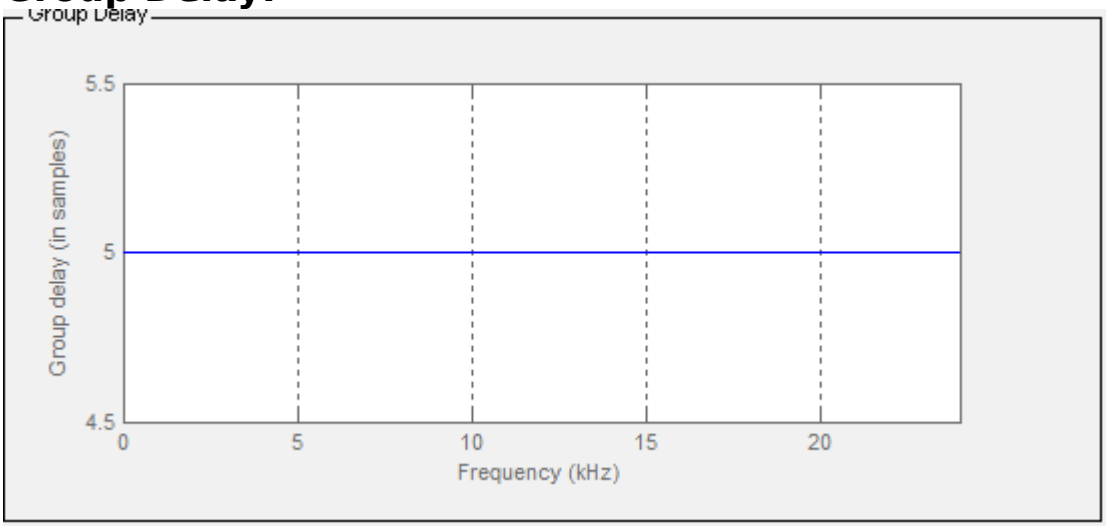
Impulse Response:



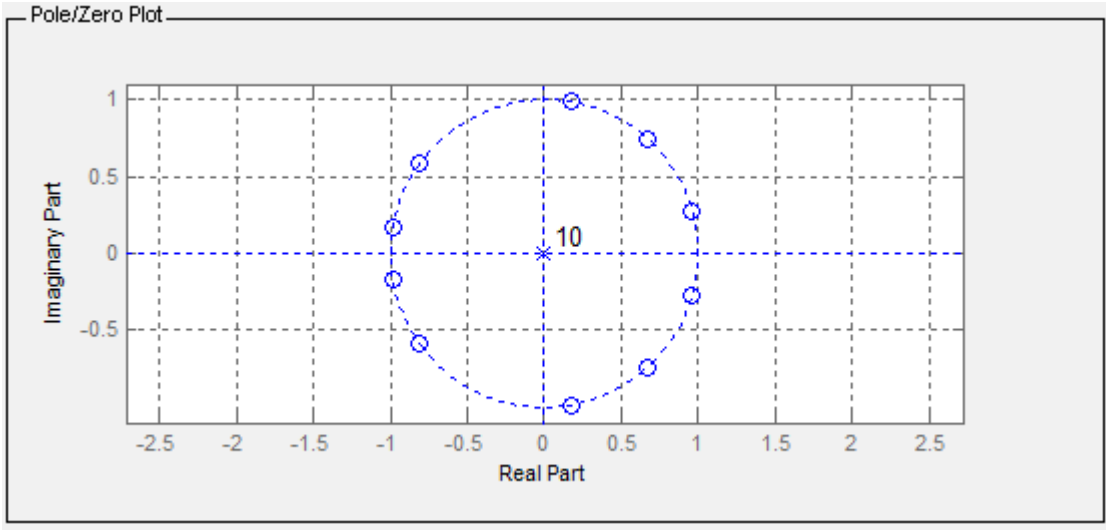
Frequency Response:



Group Delay:



Poles And Zeros:

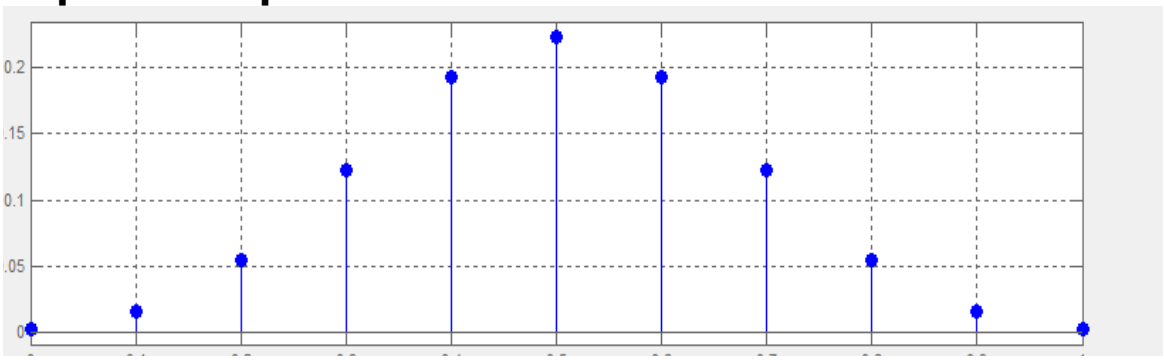


Chebyshev

cutoff at 6 dB

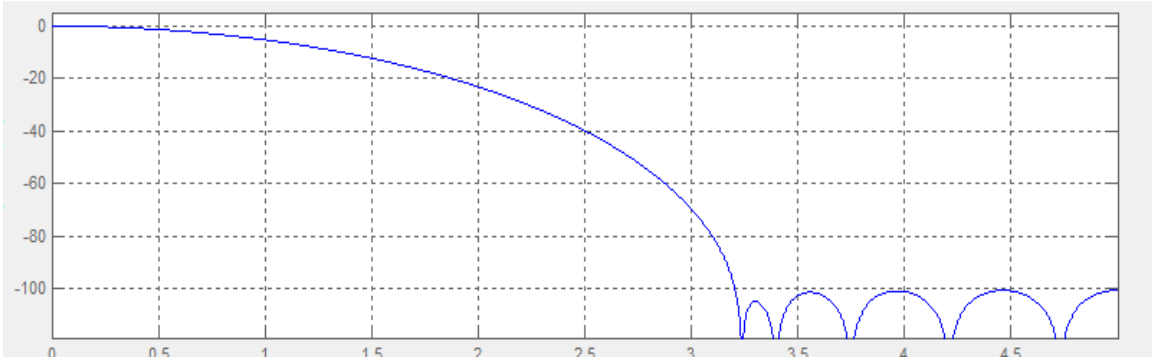
As Low Pass Filter:

Impulse Response:

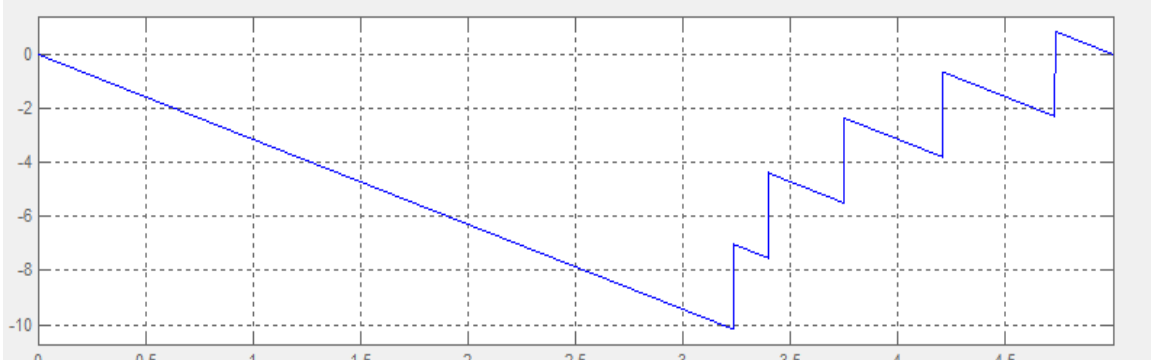


Frequency Response:

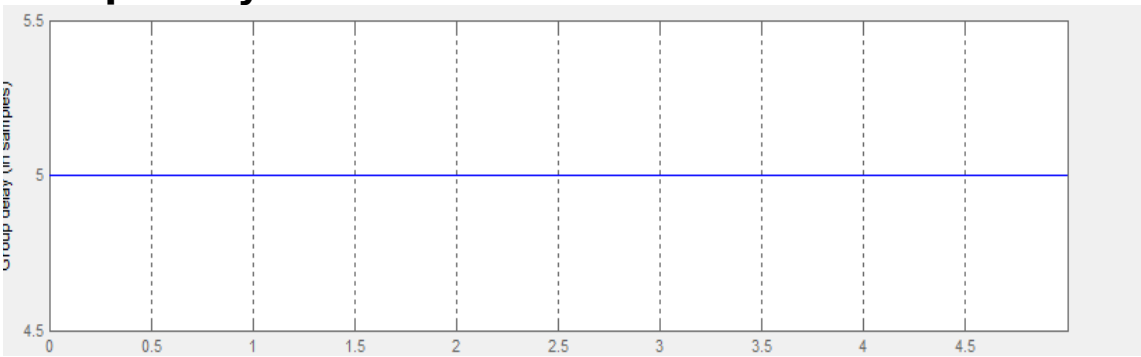
Magnitude:



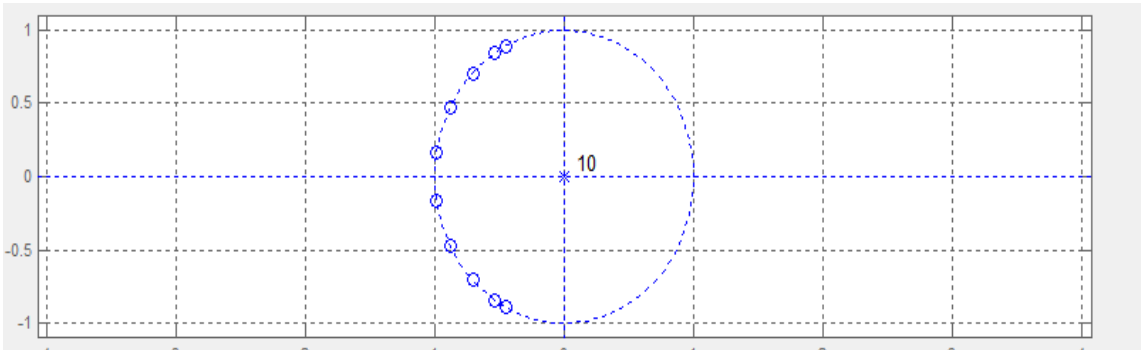
Phase:



Group Delay:

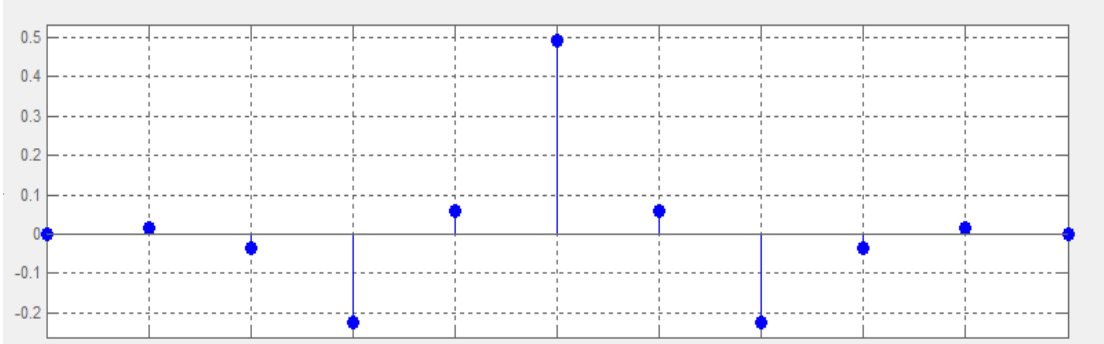


Poles and Zeros:



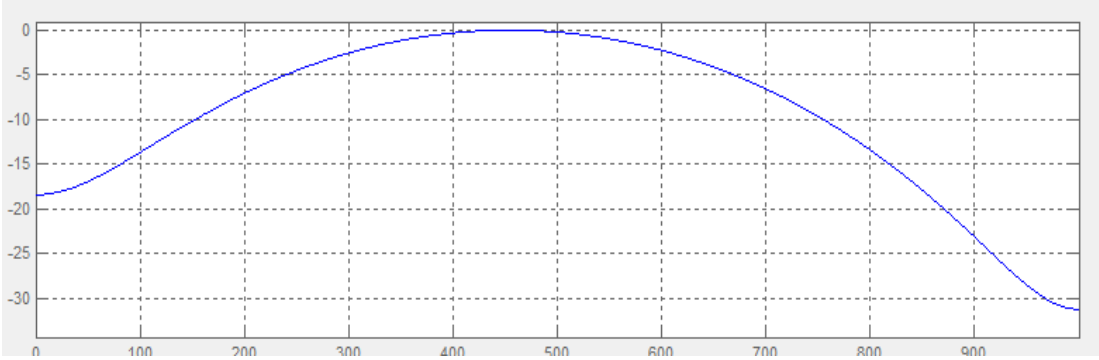
Band Pass(310-600):

Impulse Response:

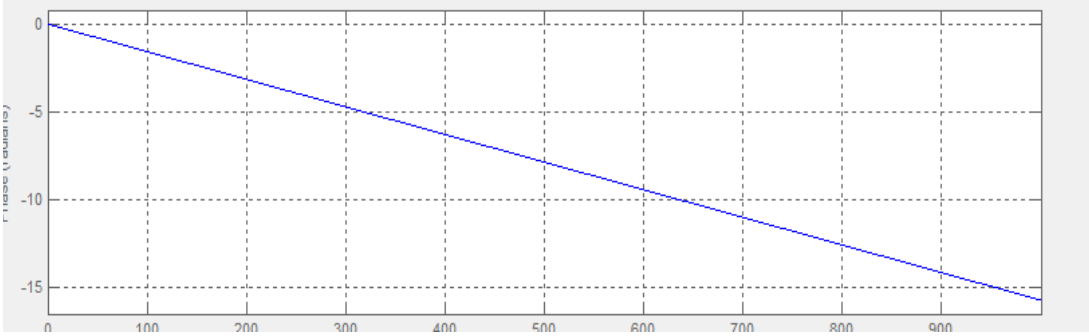


Frequency Response:

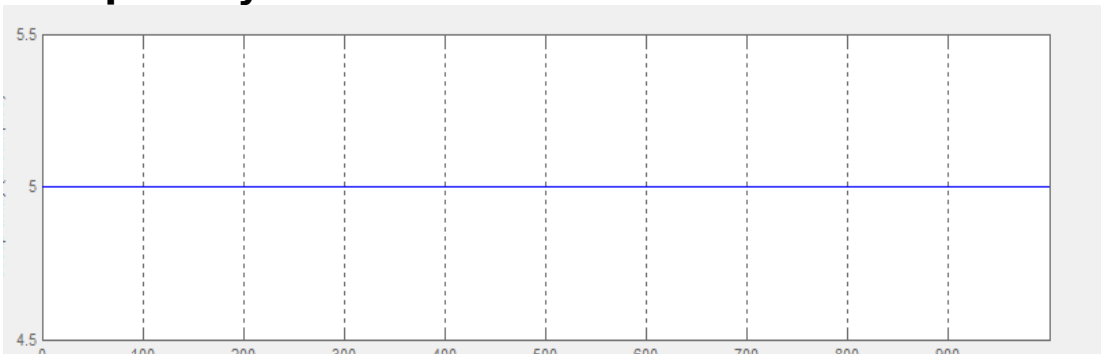
Magnitude:



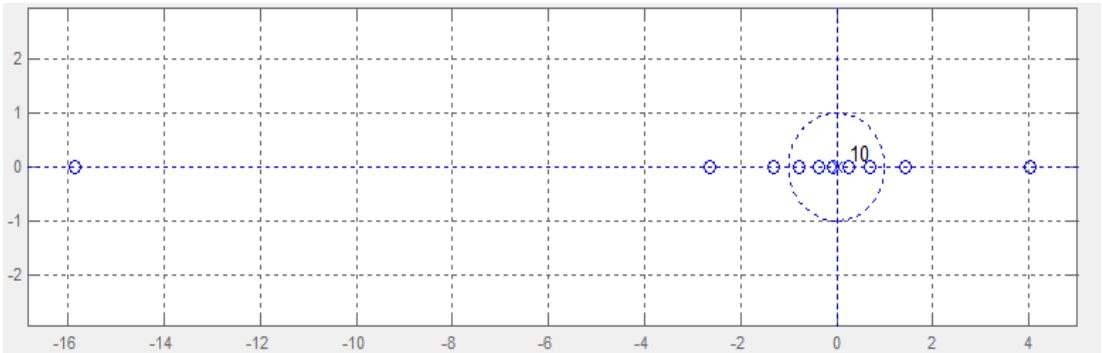
Phase:



Group Delay:

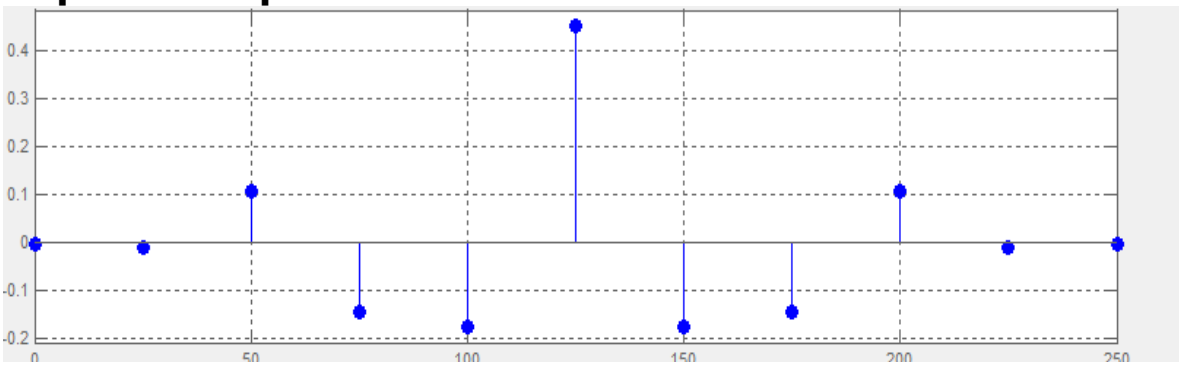


Poles And Zeros:



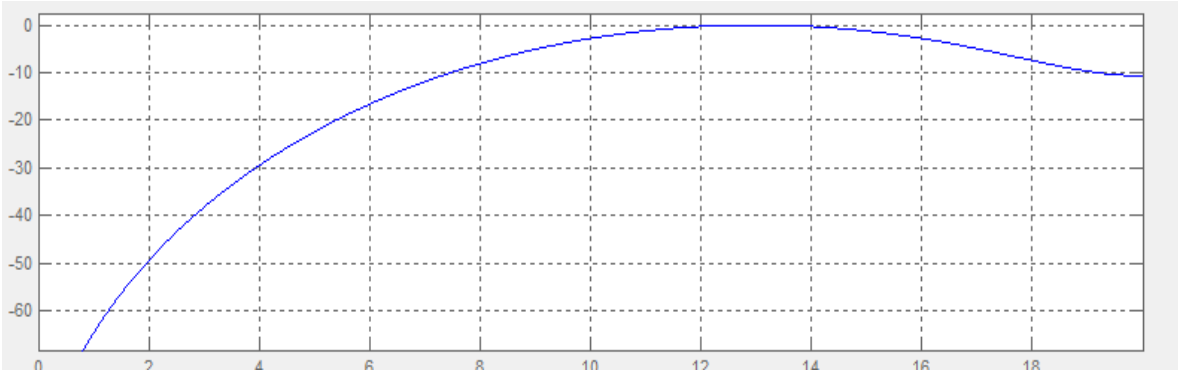
Band Pass(12k-14k):

Impulse Response:

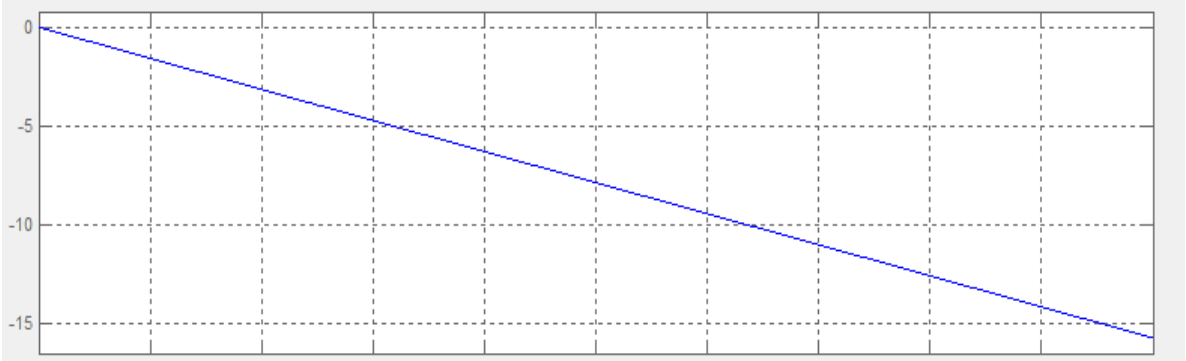


Frequency Response:

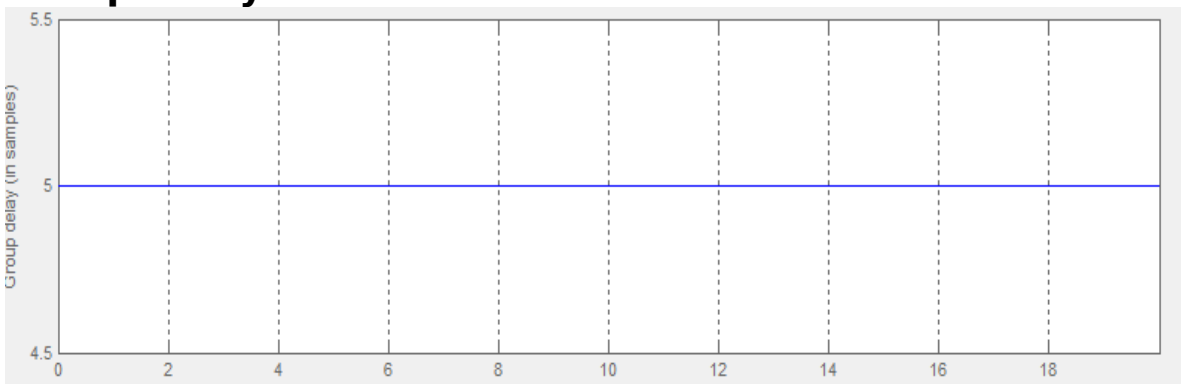
Magnitude:



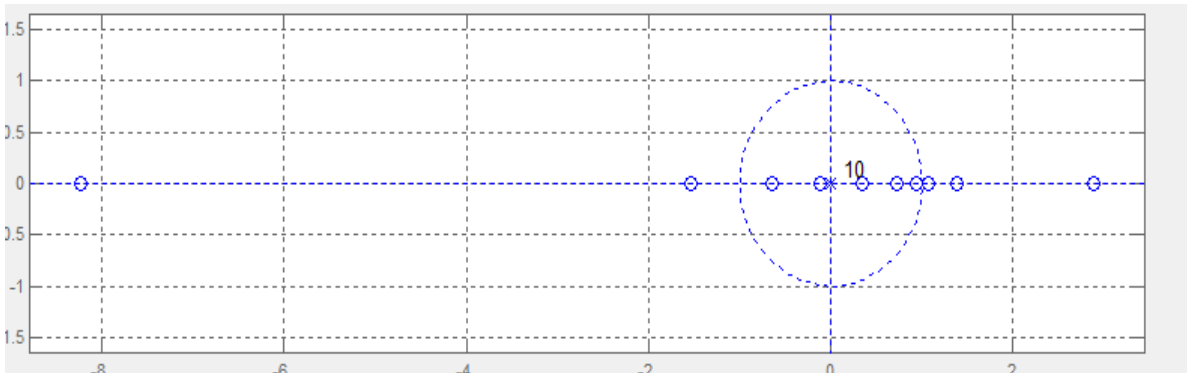
Phase:



Group Delay:



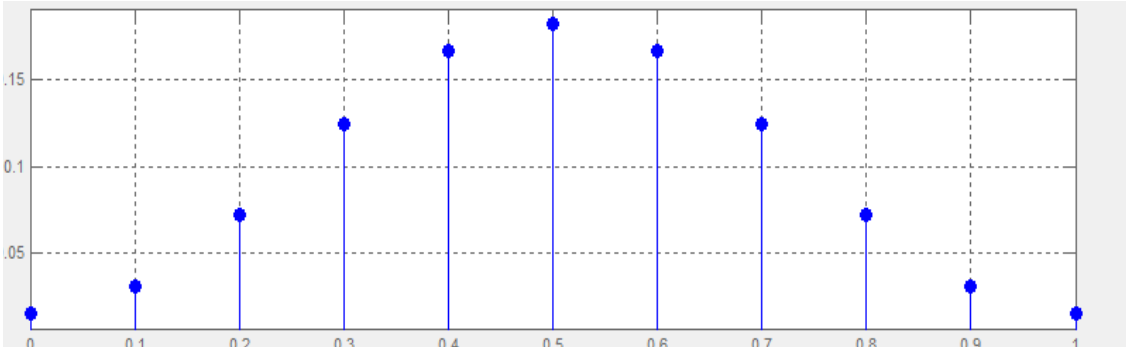
Poles And Zeros:



Hamming

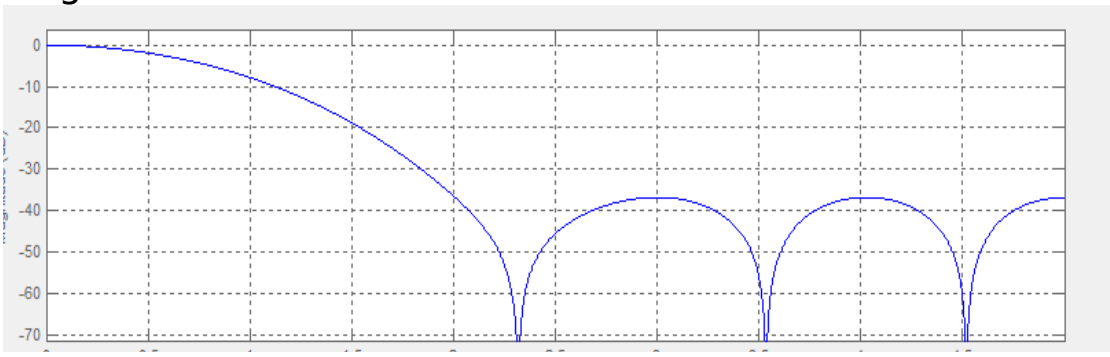
As Low Pass Filter:

Impulse Response:

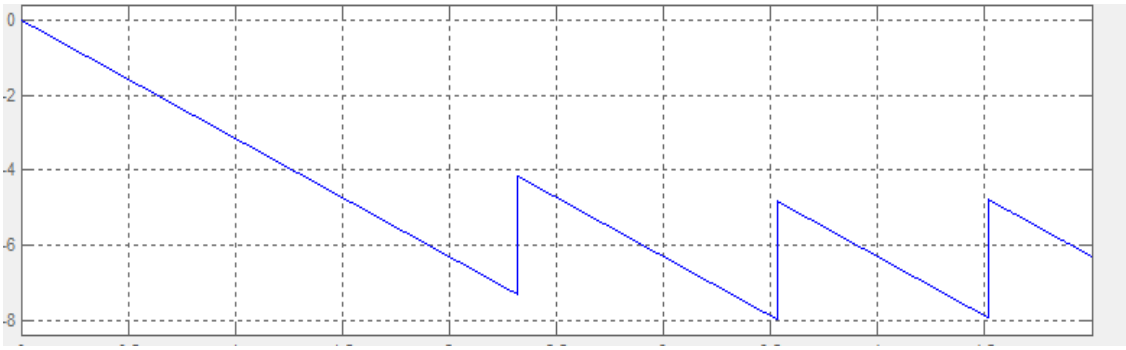


Frequency Response:

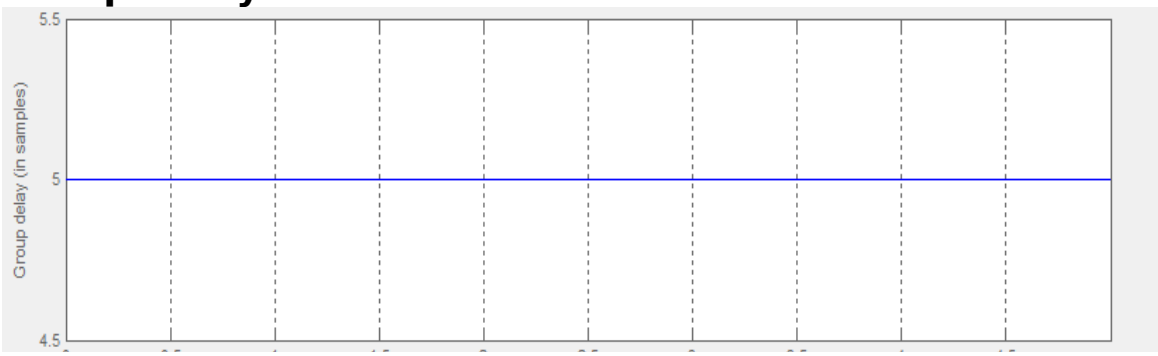
Magnitude:



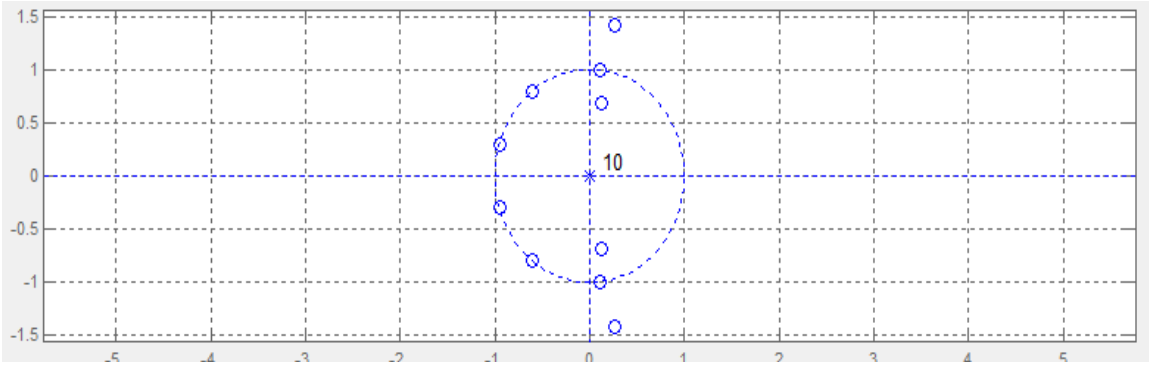
Phase:



Group Delay:

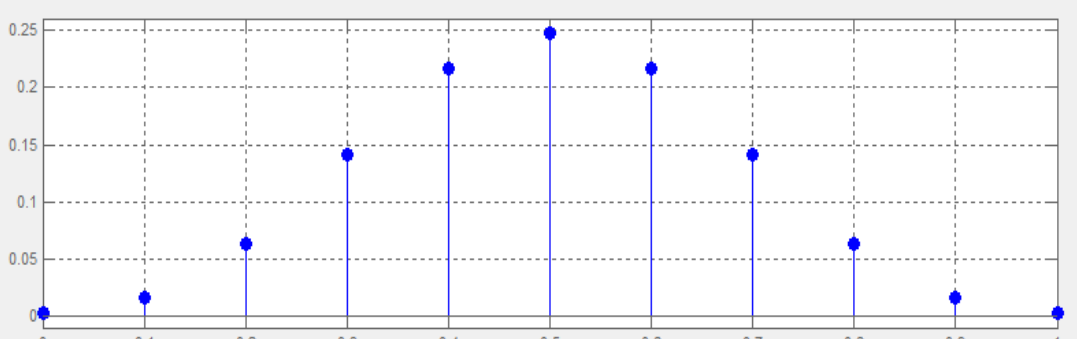


Poles and Zeros:



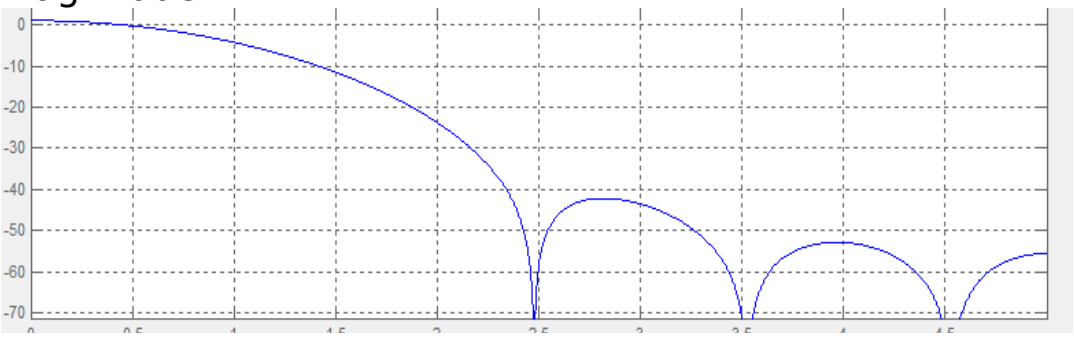
Band Pass(310-600):

Impulse Response:

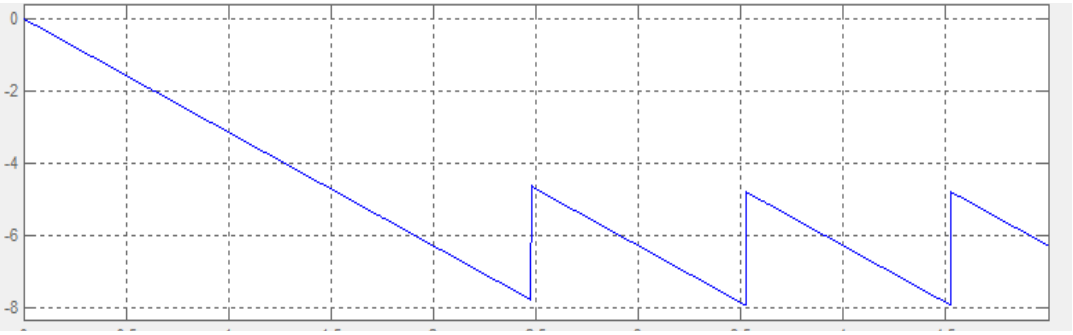


Frequency Response:

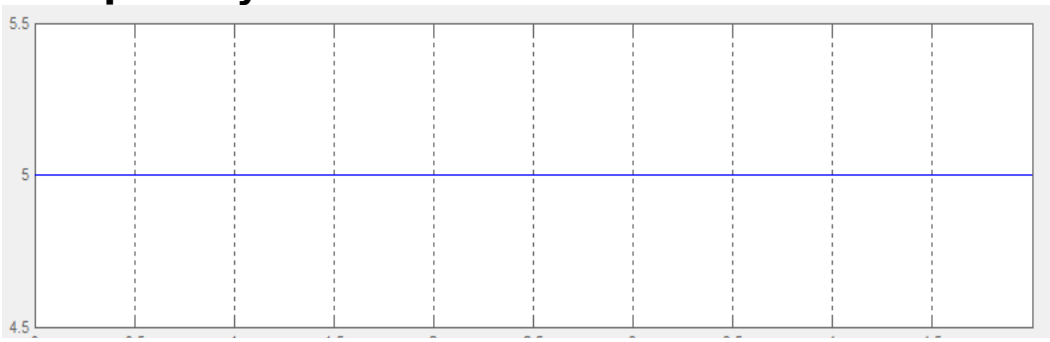
Magnitude:



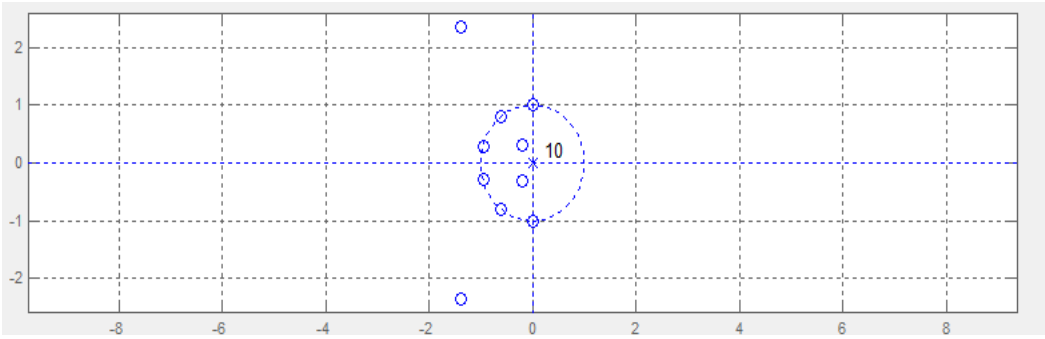
Phase:



Group Delay:

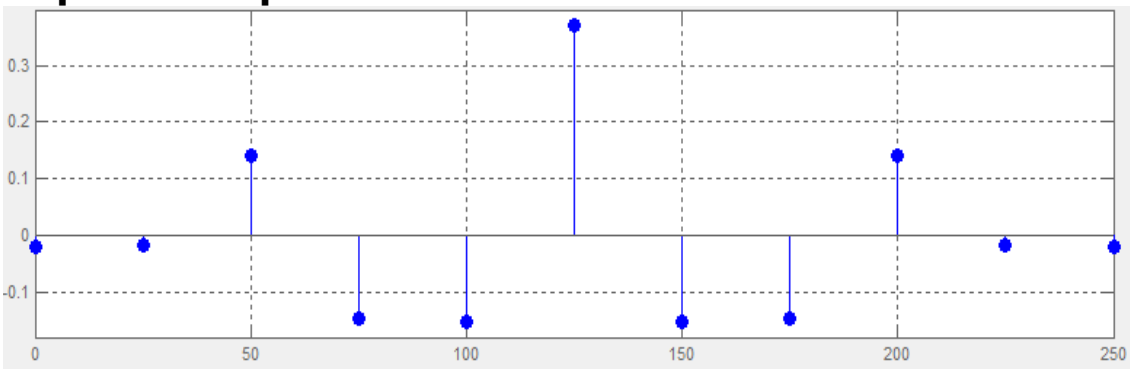


Poles And Zeros:



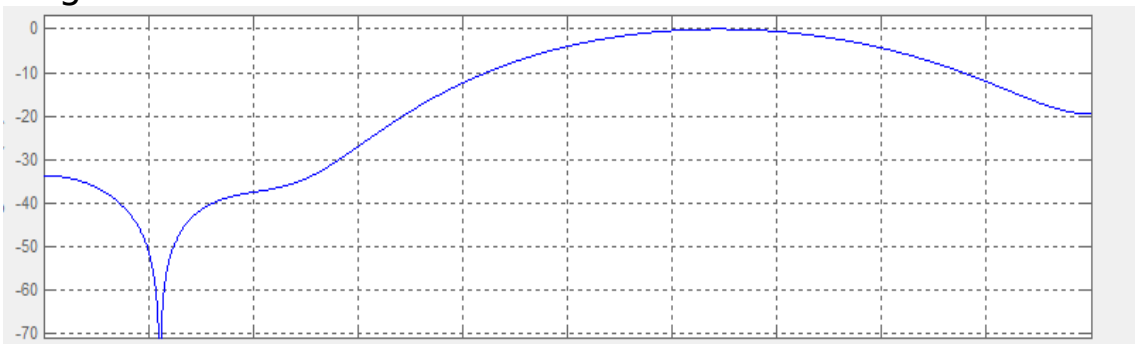
Band Pass(12000-14000):

Impulse Response:

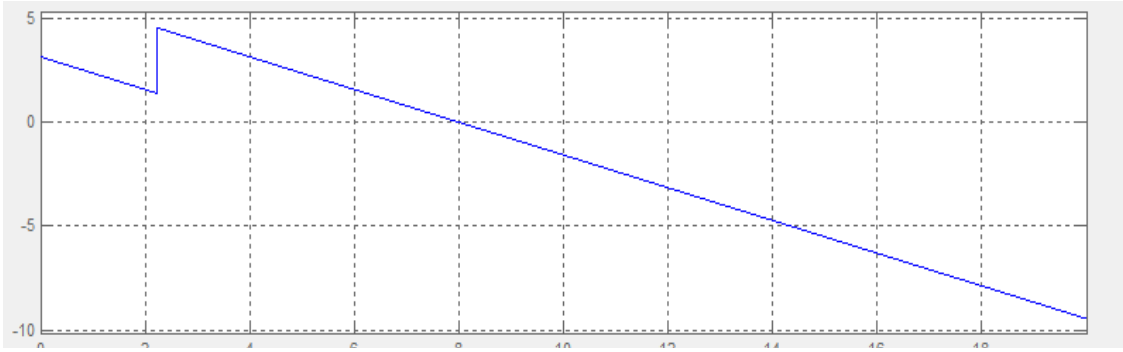


Frequency Response:

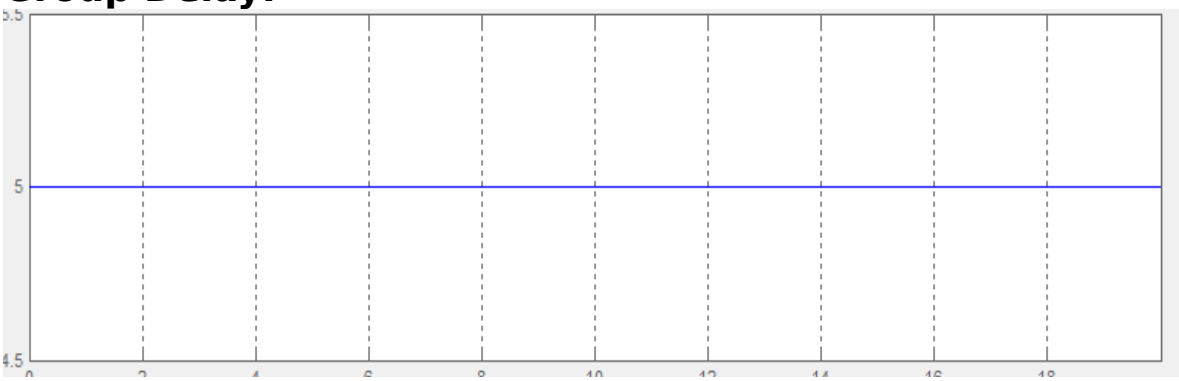
Magnitude:



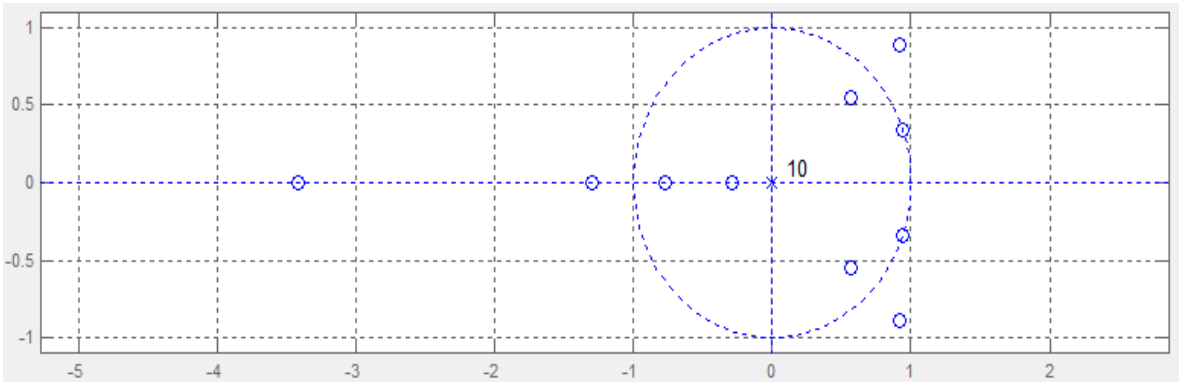
Phase:



Group Delay:



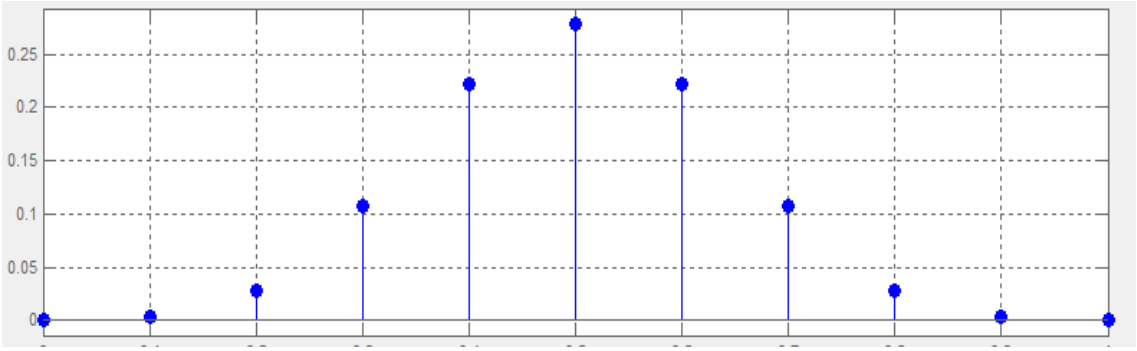
Poles And Zeros:



Blackman-harris

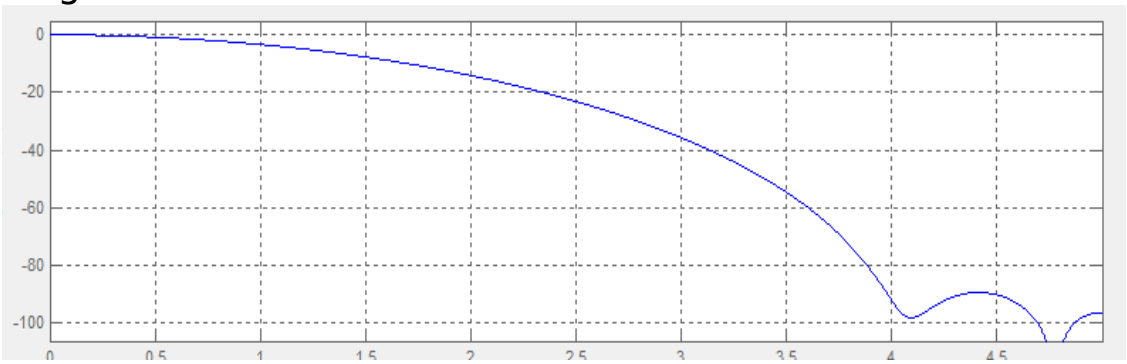
As Low Pass Filter:

Impulse Response:

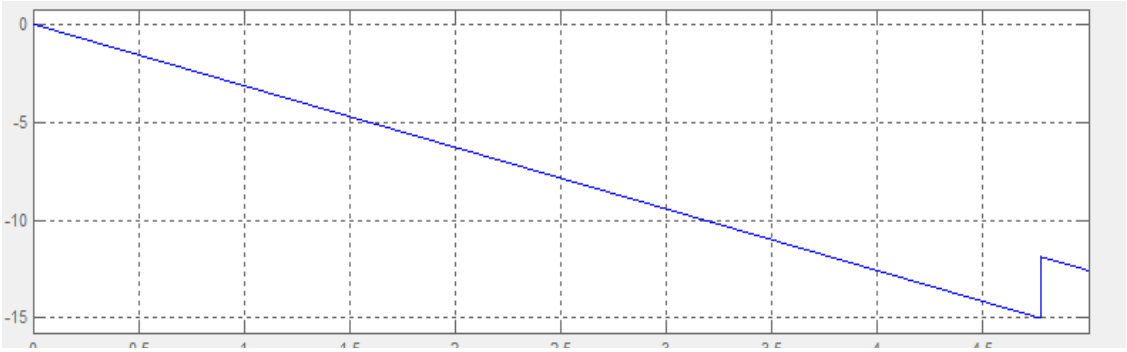


Frequency Response:

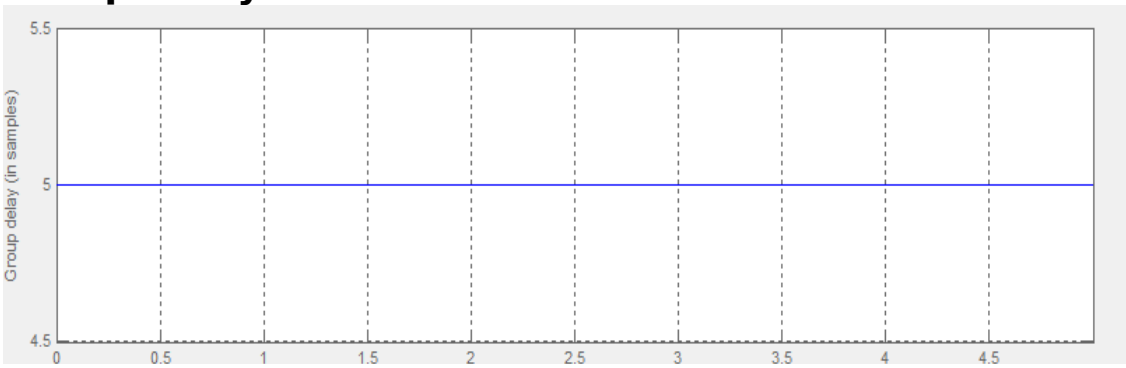
Magnitude:



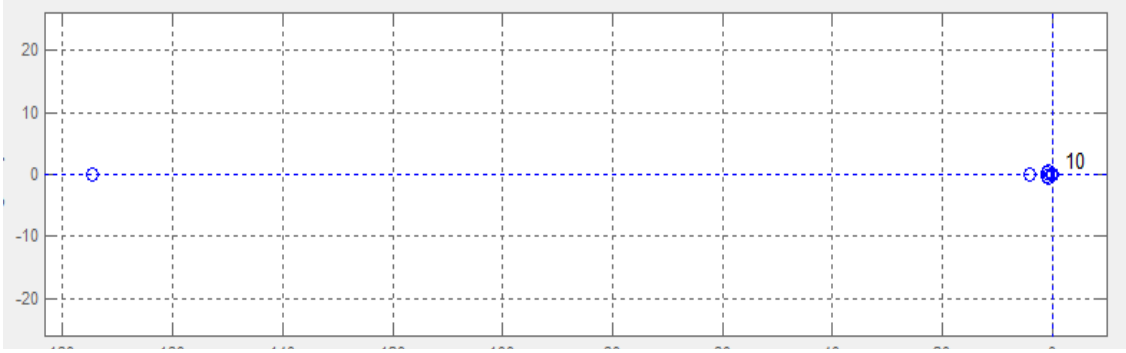
Phase:



Group Delay:

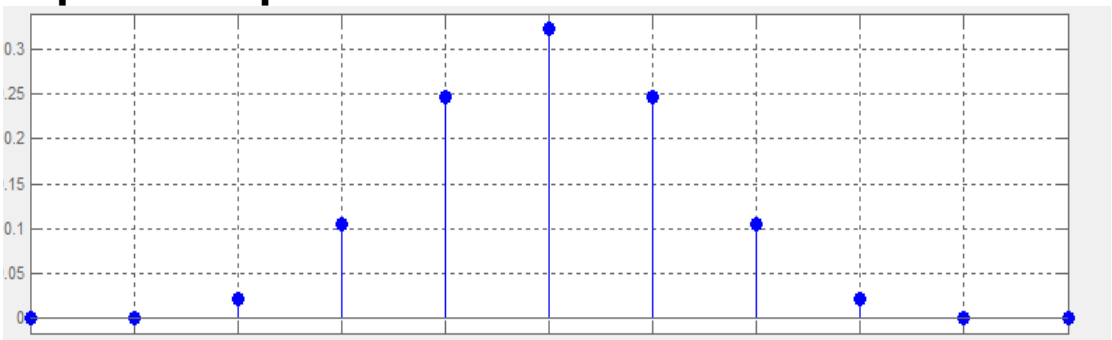


Poles and Zeros:



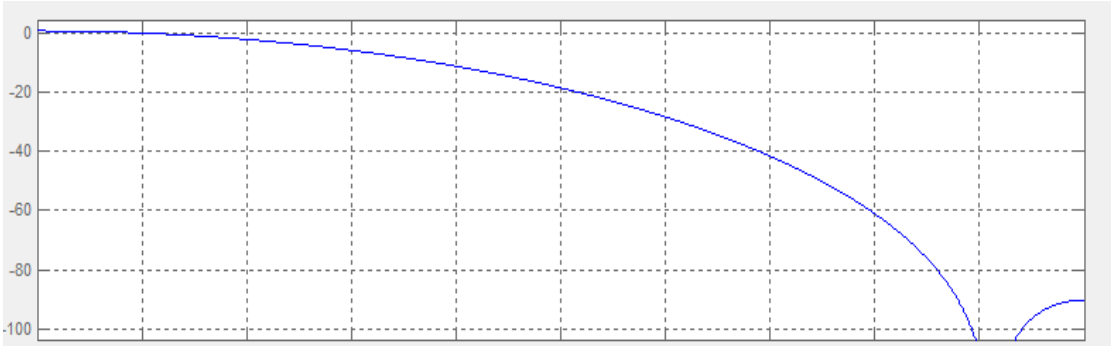
Band Pass(310-600):

Impulse Response:

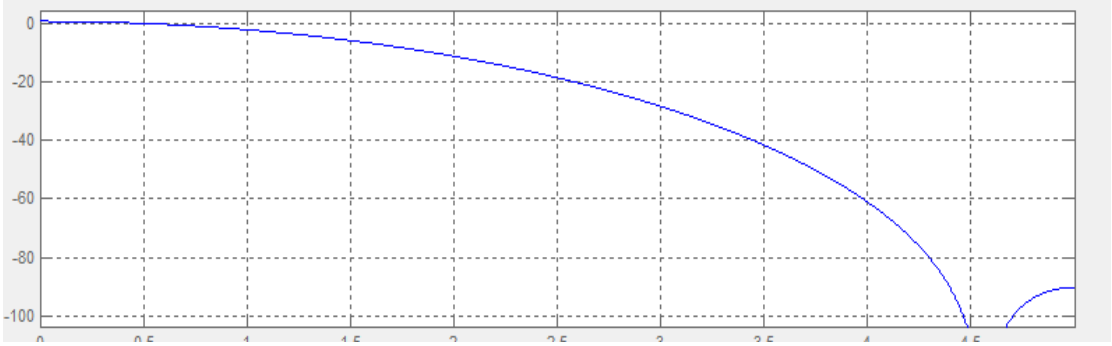


Frequency Response:

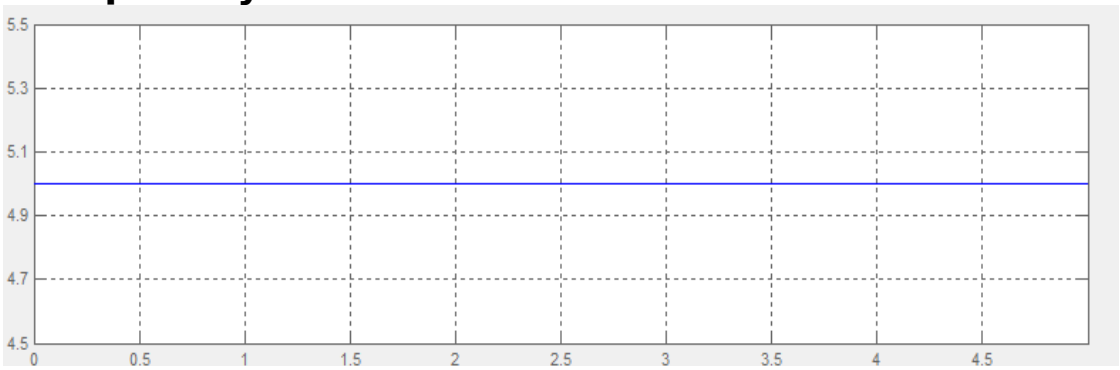
Magnitude:



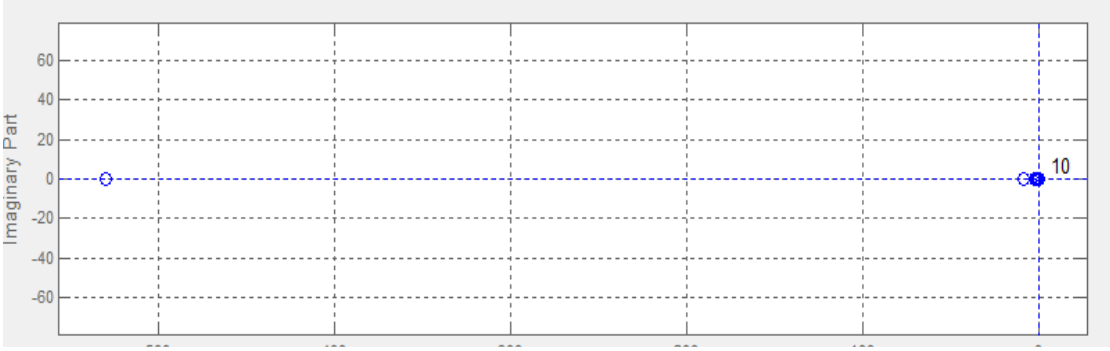
Phase:



Group Delay:

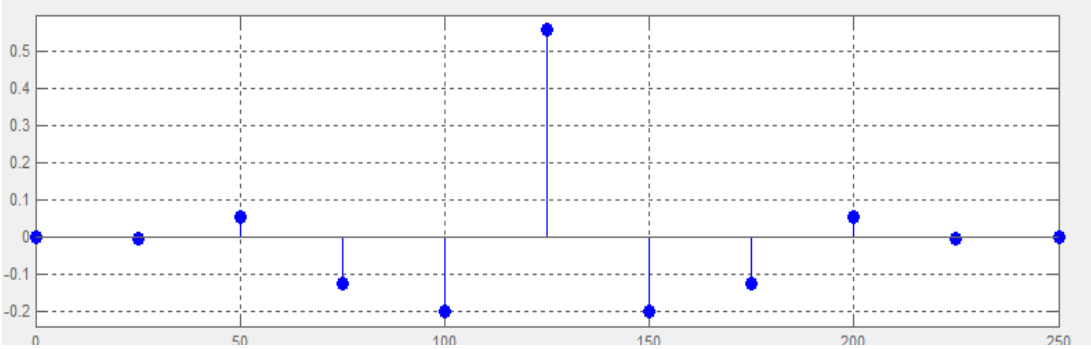


Poles And Zeros:



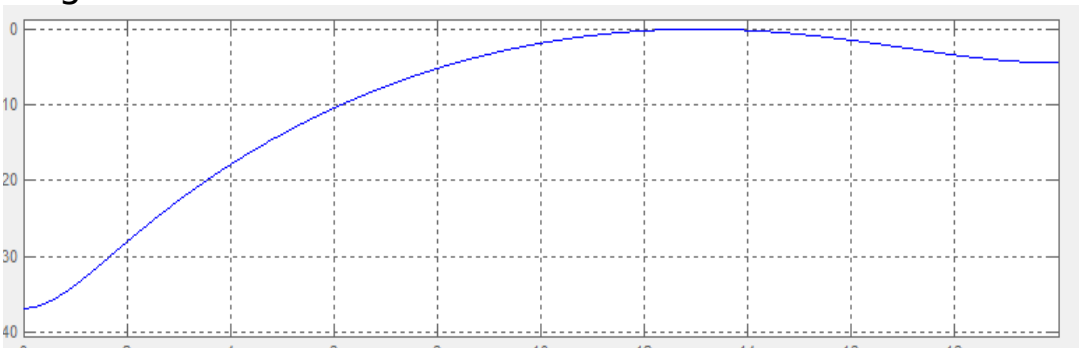
Band Pass(12k-14k):

Impulse Response:

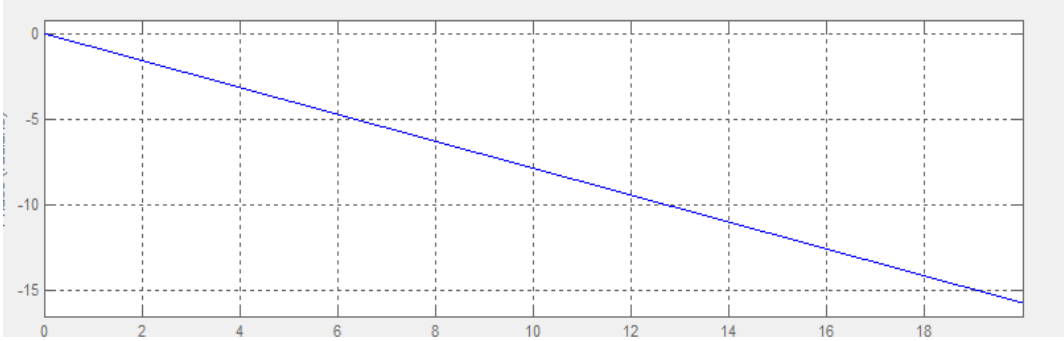


Frequency Response:

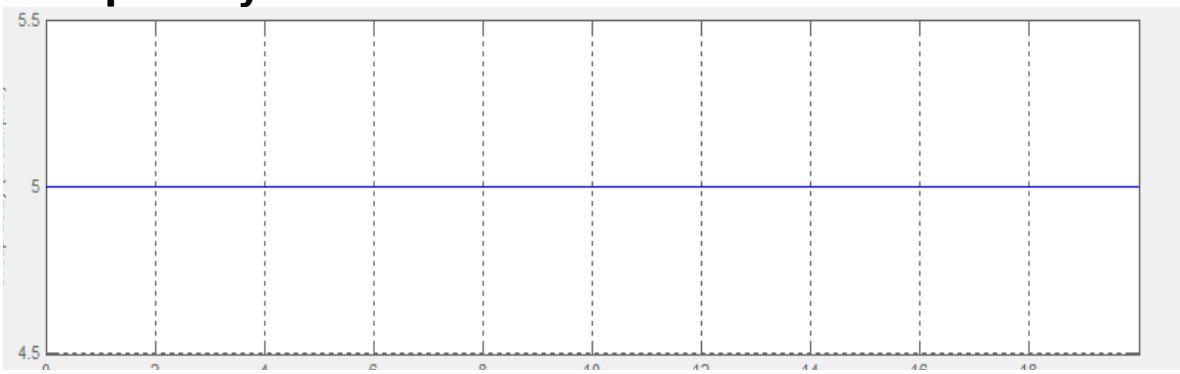
Magnitude:



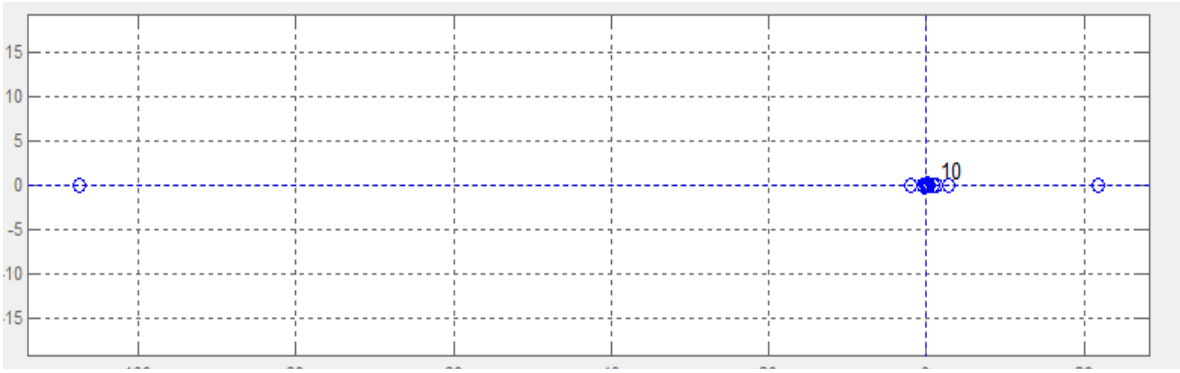
Phase:



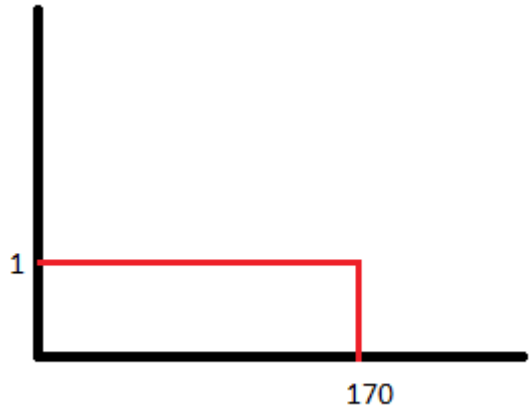
Group Delay:

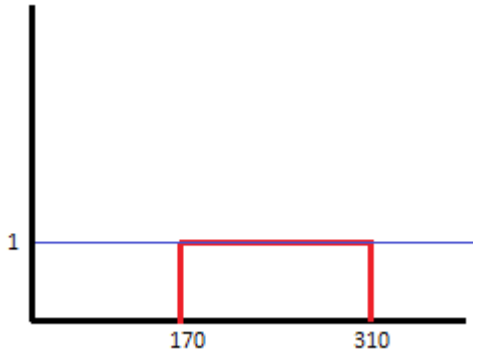


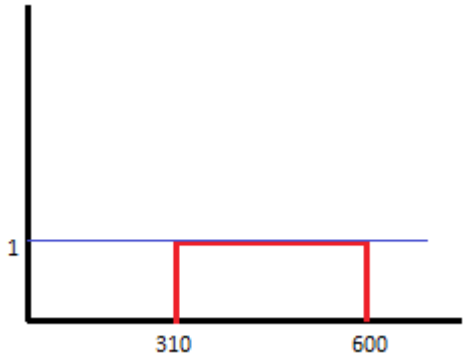
Poles And Zeros:

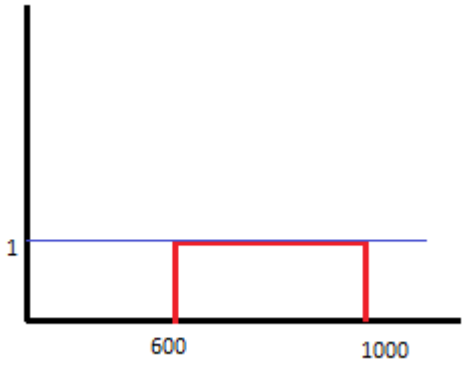


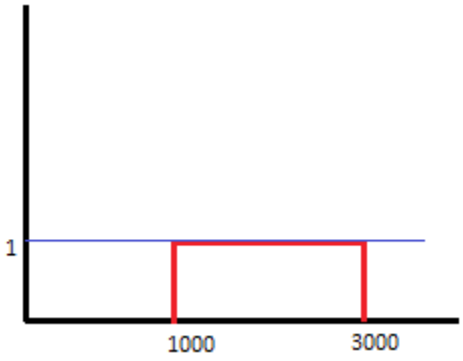
Ideal Filters (rect):

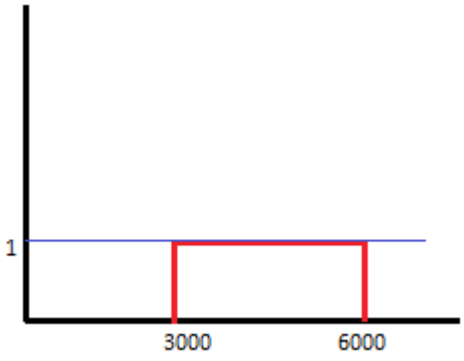


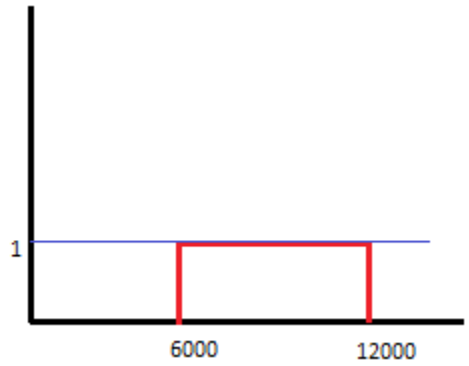


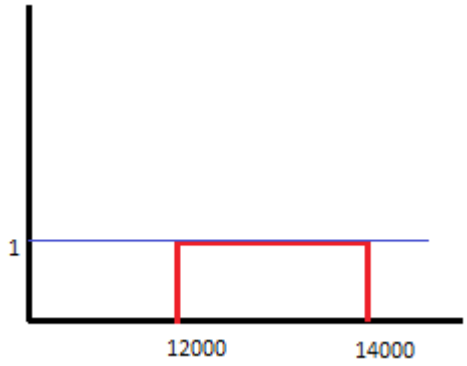


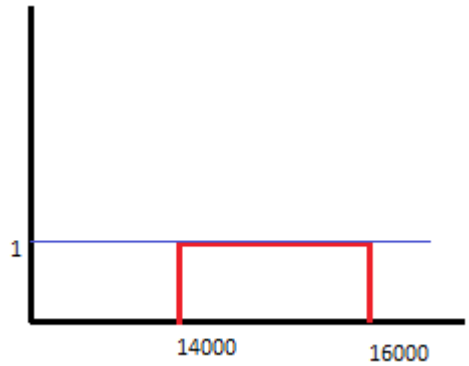




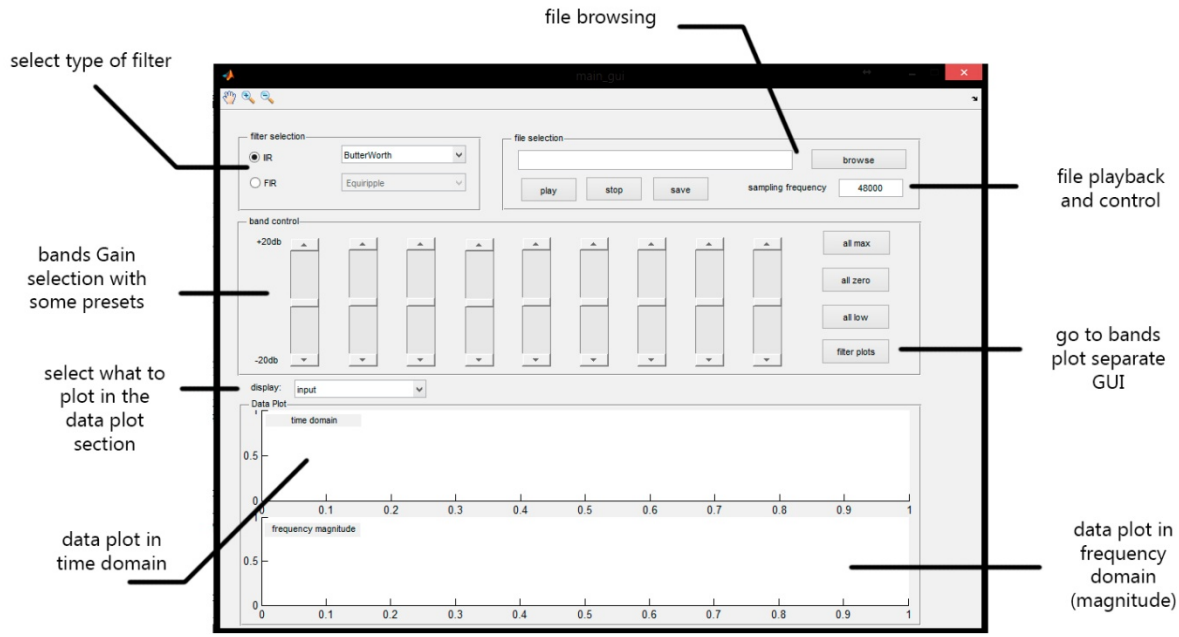


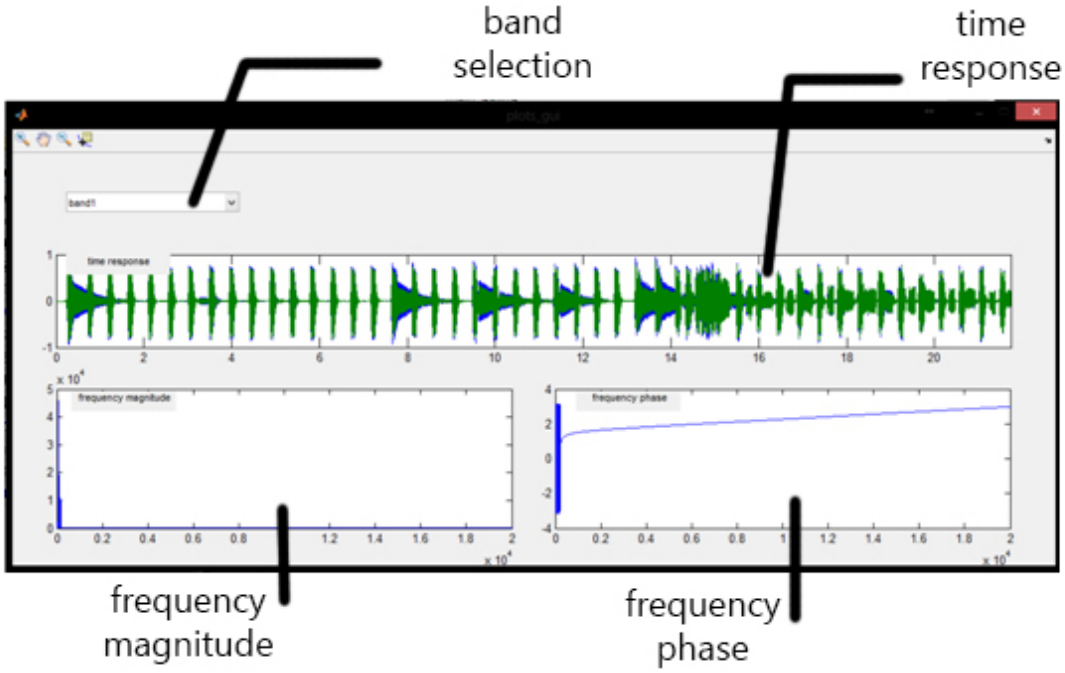

1
6000
12000


1
12000
14000

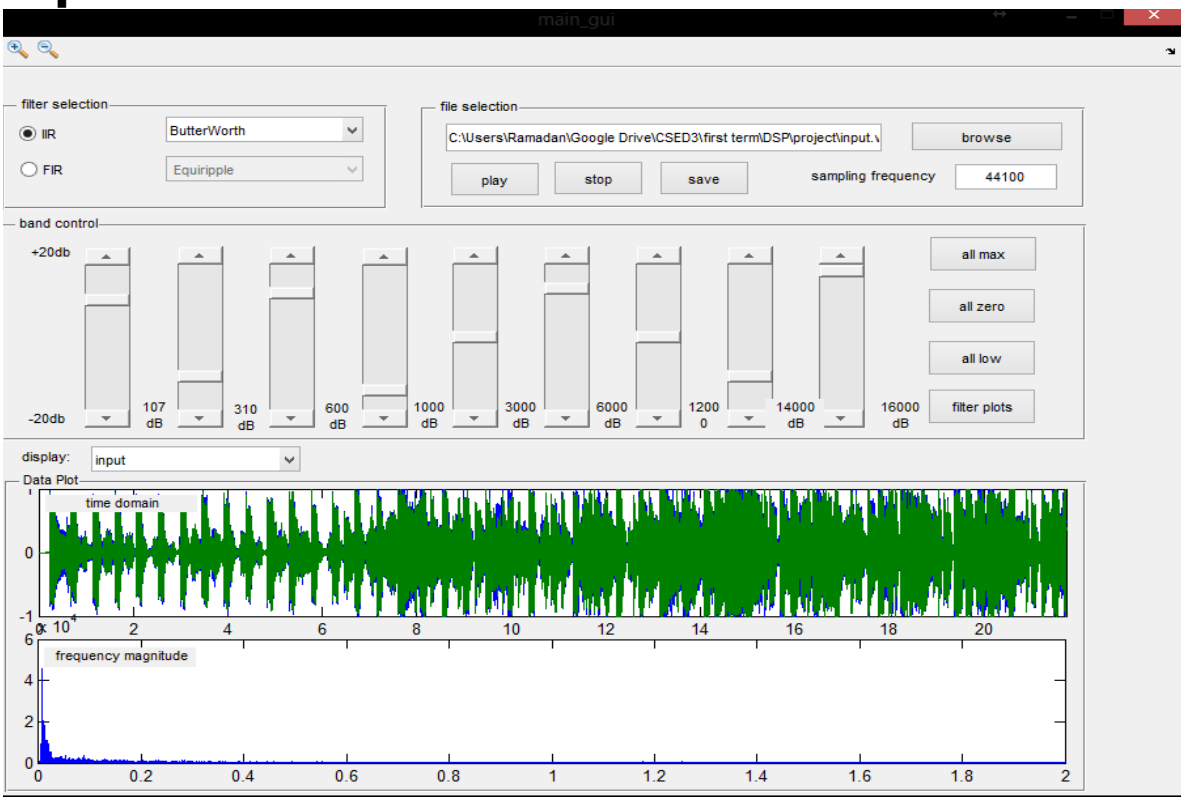

1
14000
16000

GUI Exploration

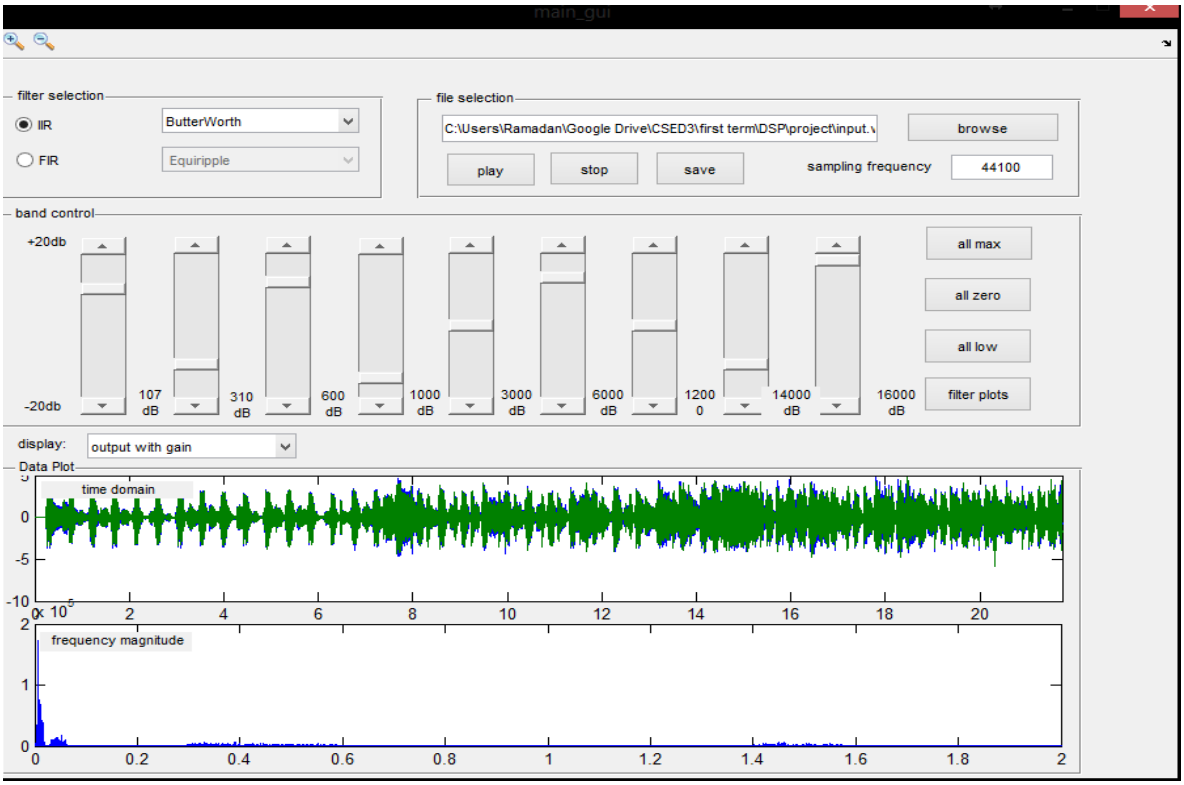




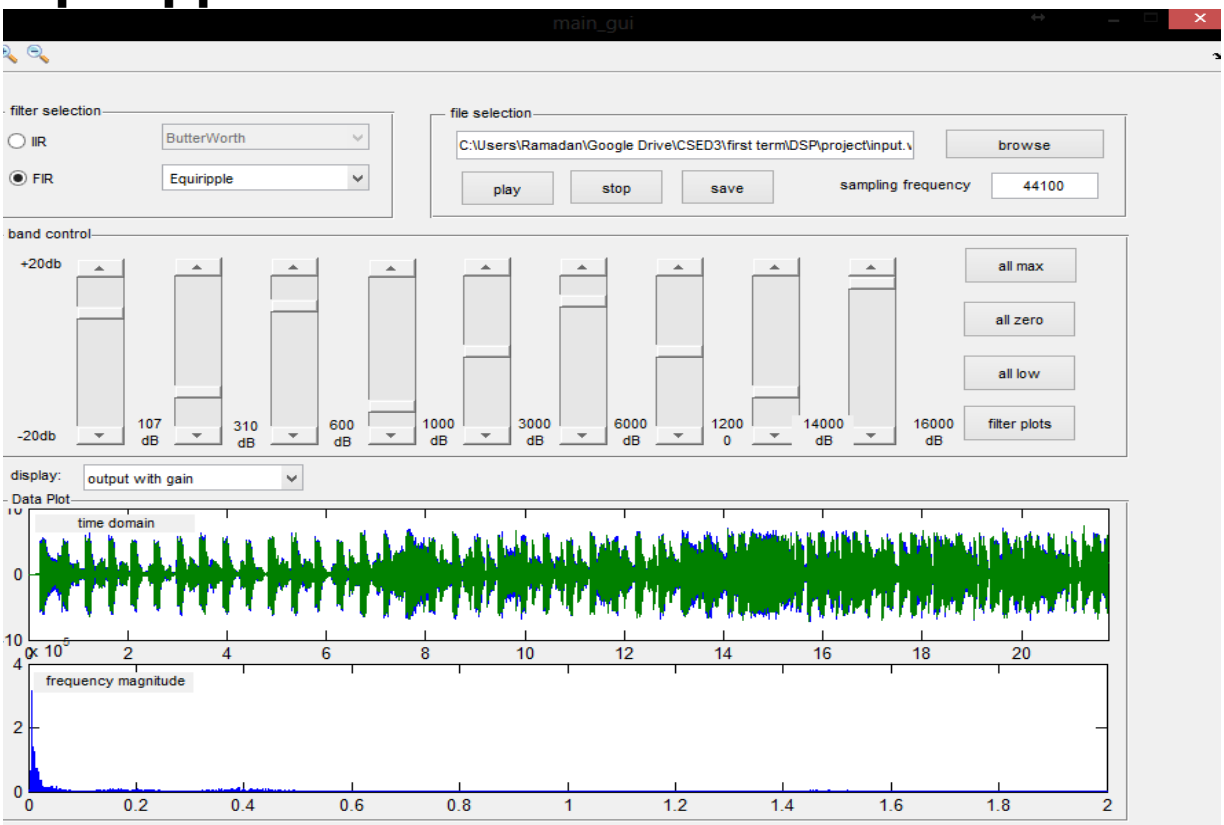
Sample Runs:
input:



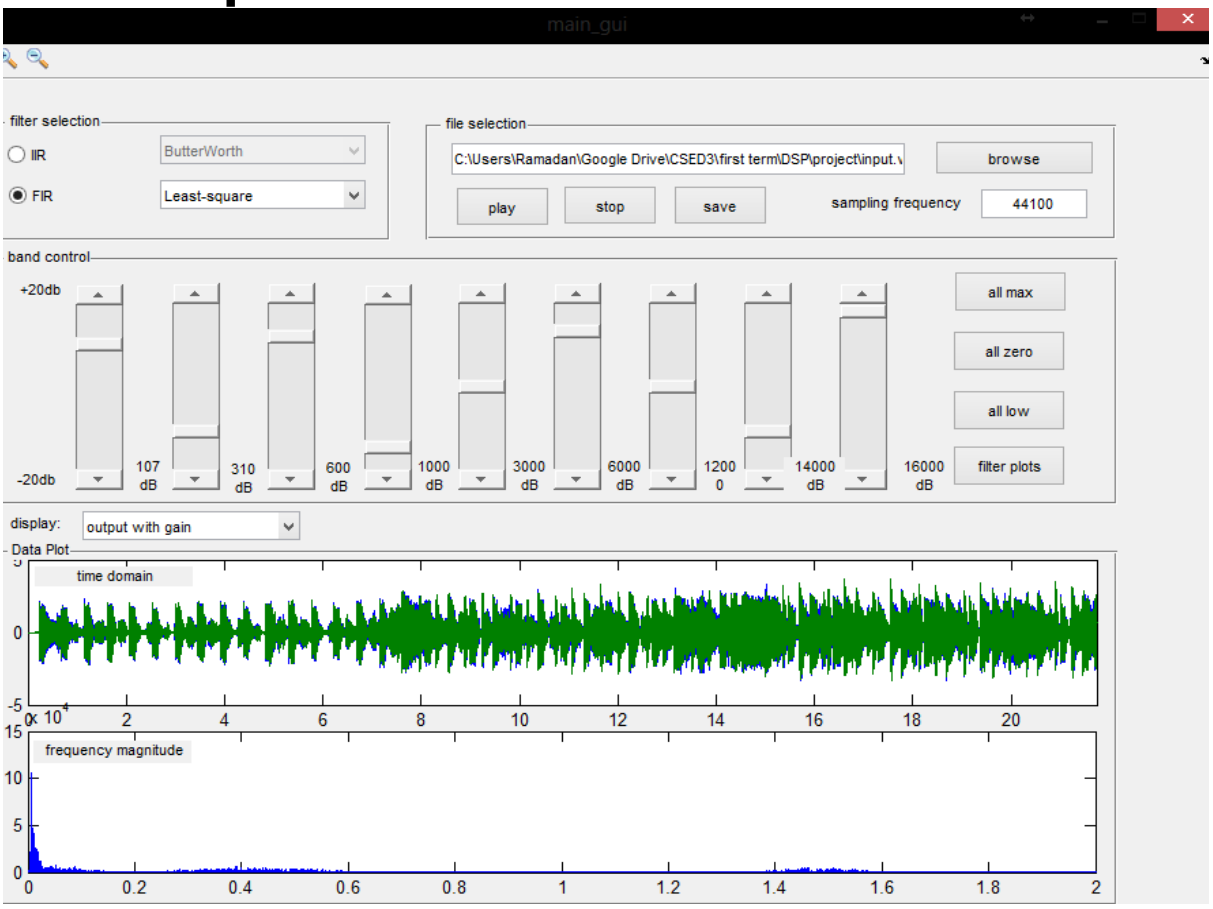
output:
IIR:


main_gui
filter selection
IIR
FIR
ButterWorth
Equiripple
file selection
C:\Users\Ramadan\Google Drive\CSED3\first term\DSP\project\input.
browse
play
stop
save
sampling frequency
44100
band control
+20db
-20db
107 dB
310 dB
600 dB
1000 dB
3000 dB
6000 dB
1200 0
14000 dB
16000 dB
all max
all zero
all low
filter plots
display:
output with gain
Data Plot
time domain
frequency magnitude

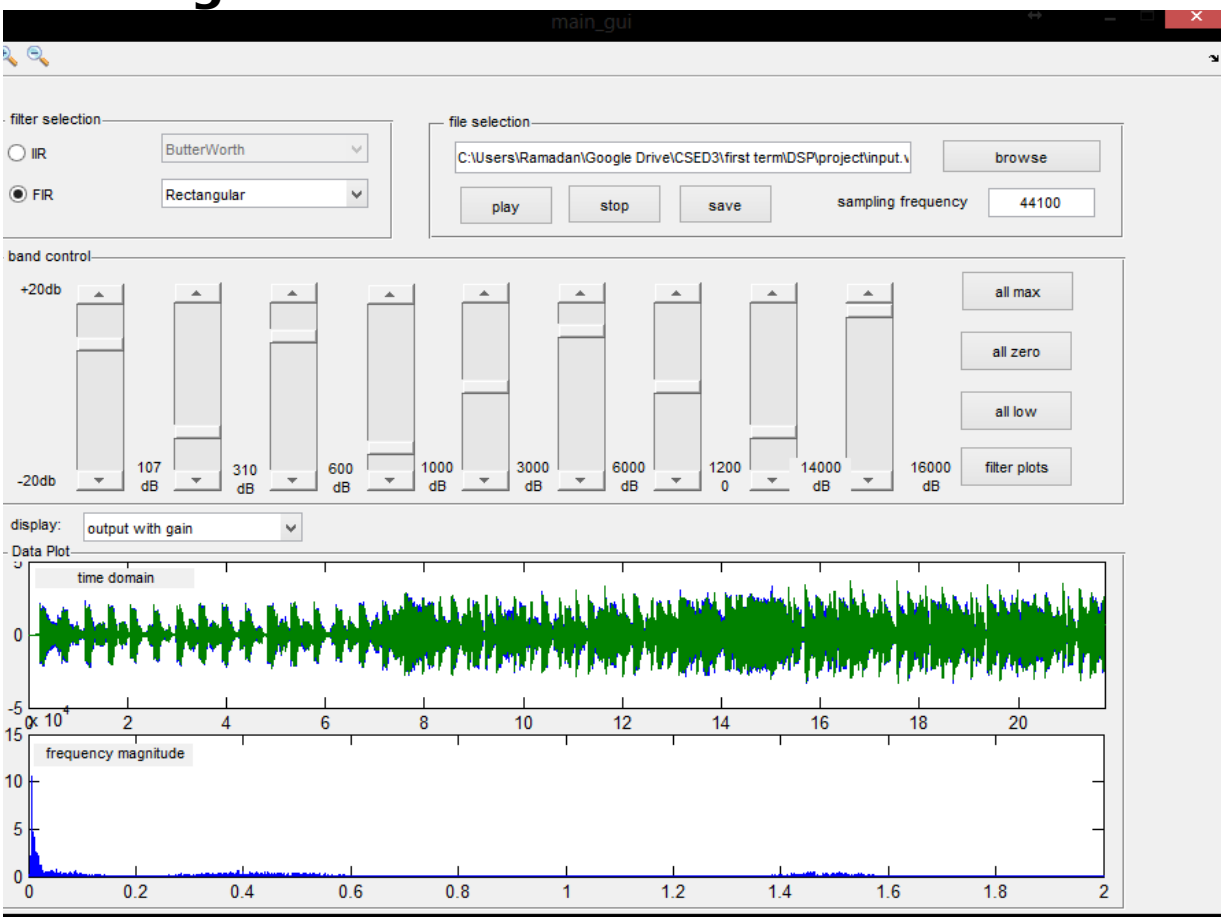
FIR:
Equiripple:



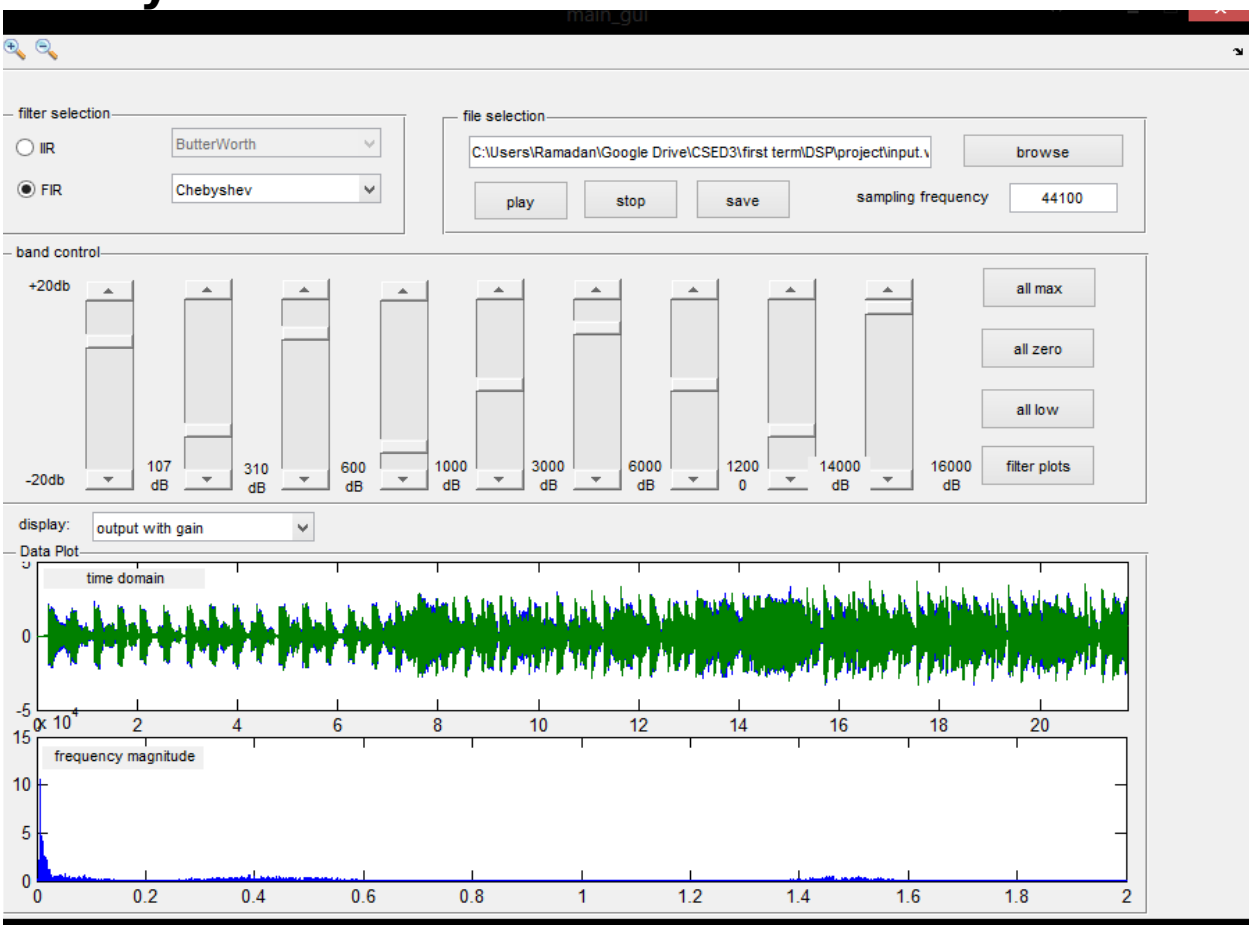
Least-Square:



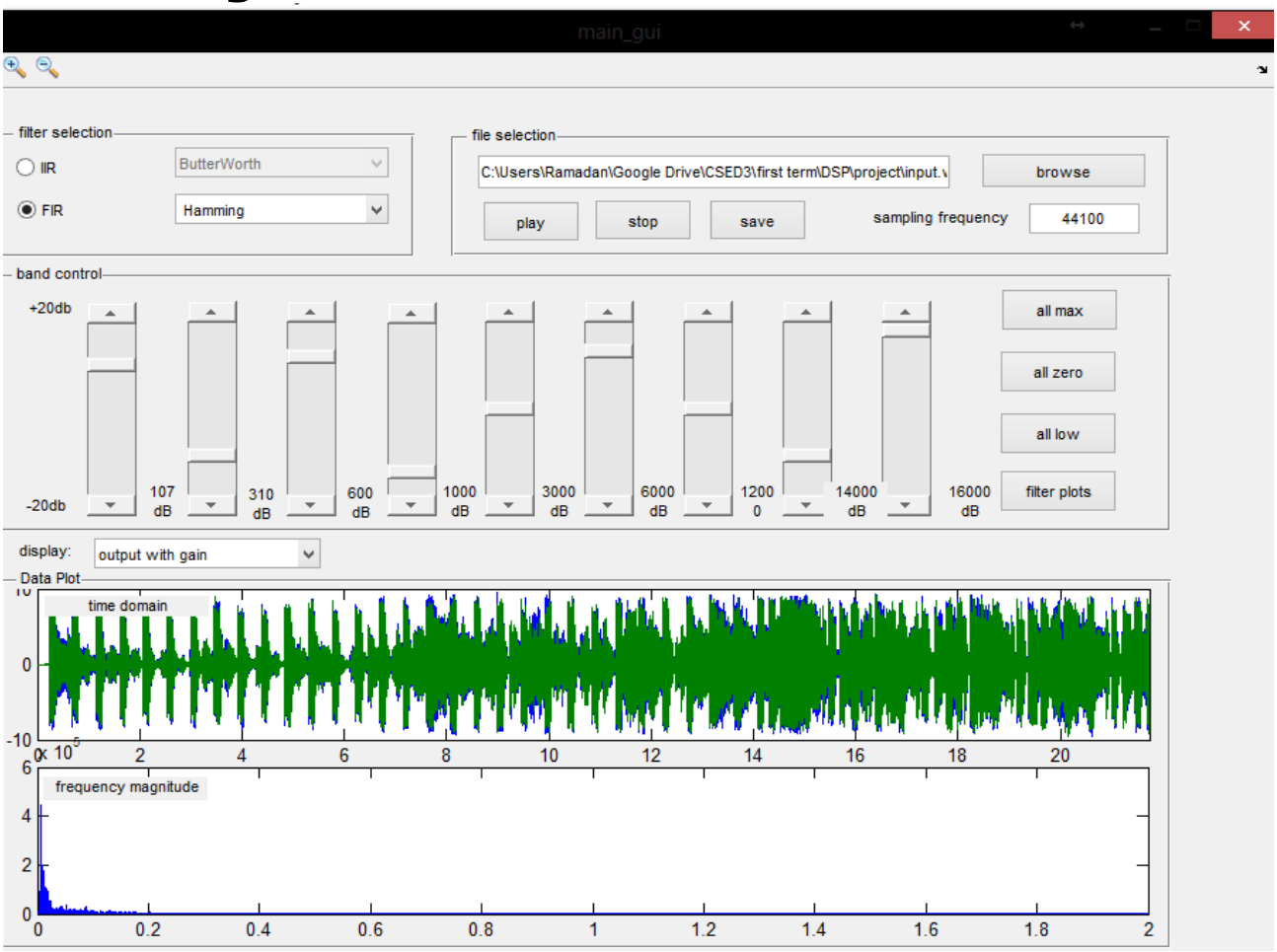
Rectangular:



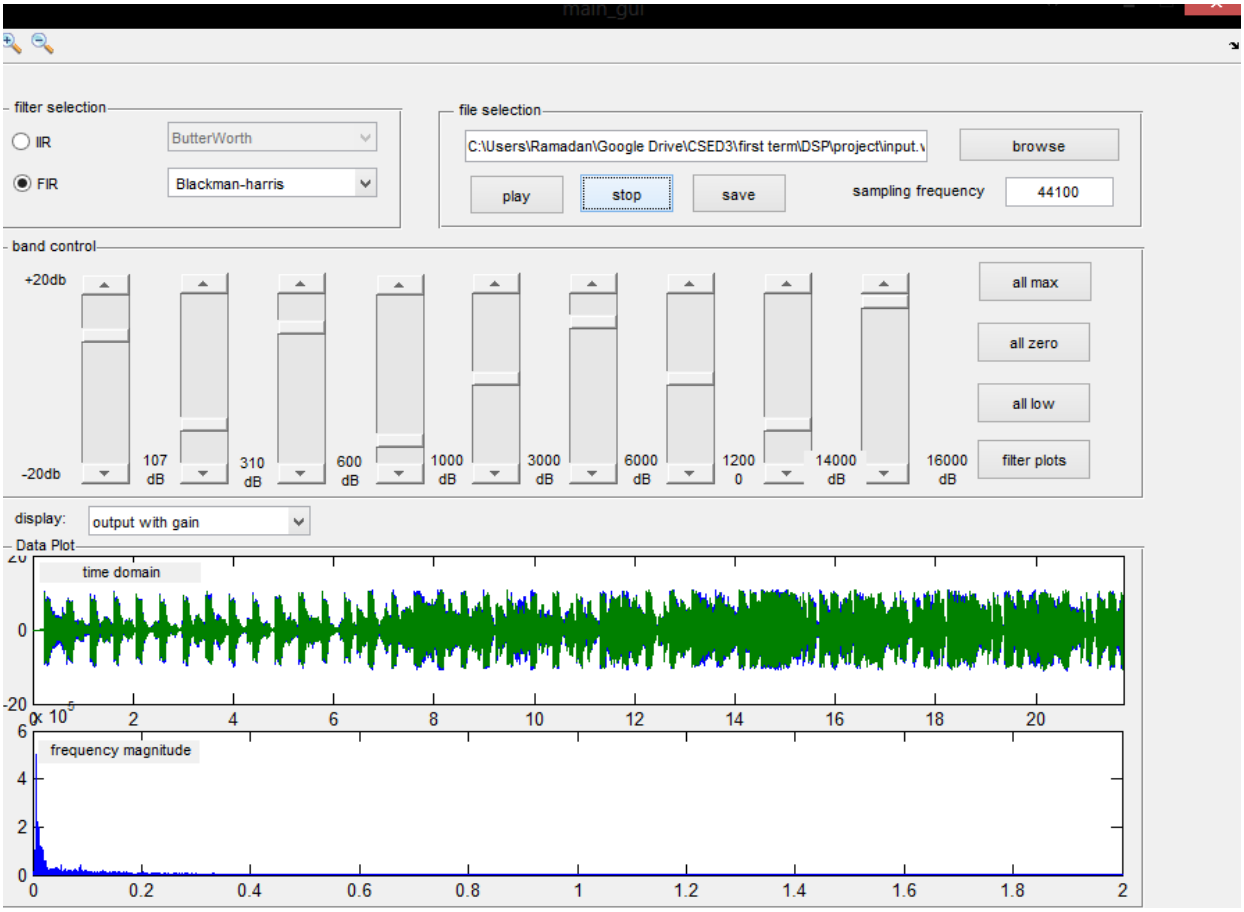
Chebyshev


filter selection
IIR
ButterWorth
FIR
Chebyshev
file selection
C:\Users\Ramadan\Google Drive\CSED3\first term\DSP\project\input.\
browse
play
stop
save
sampling frequency
44100
band control
+20db
-20db
107 dB
310 dB
600 dB
1000 dB
3000 dB
6000 dB
1200 0
14000 dB
16000 dB
all max
all zero
all low
filter plots
display:
output with gain
Data Plot
time domain
frequency magnitude

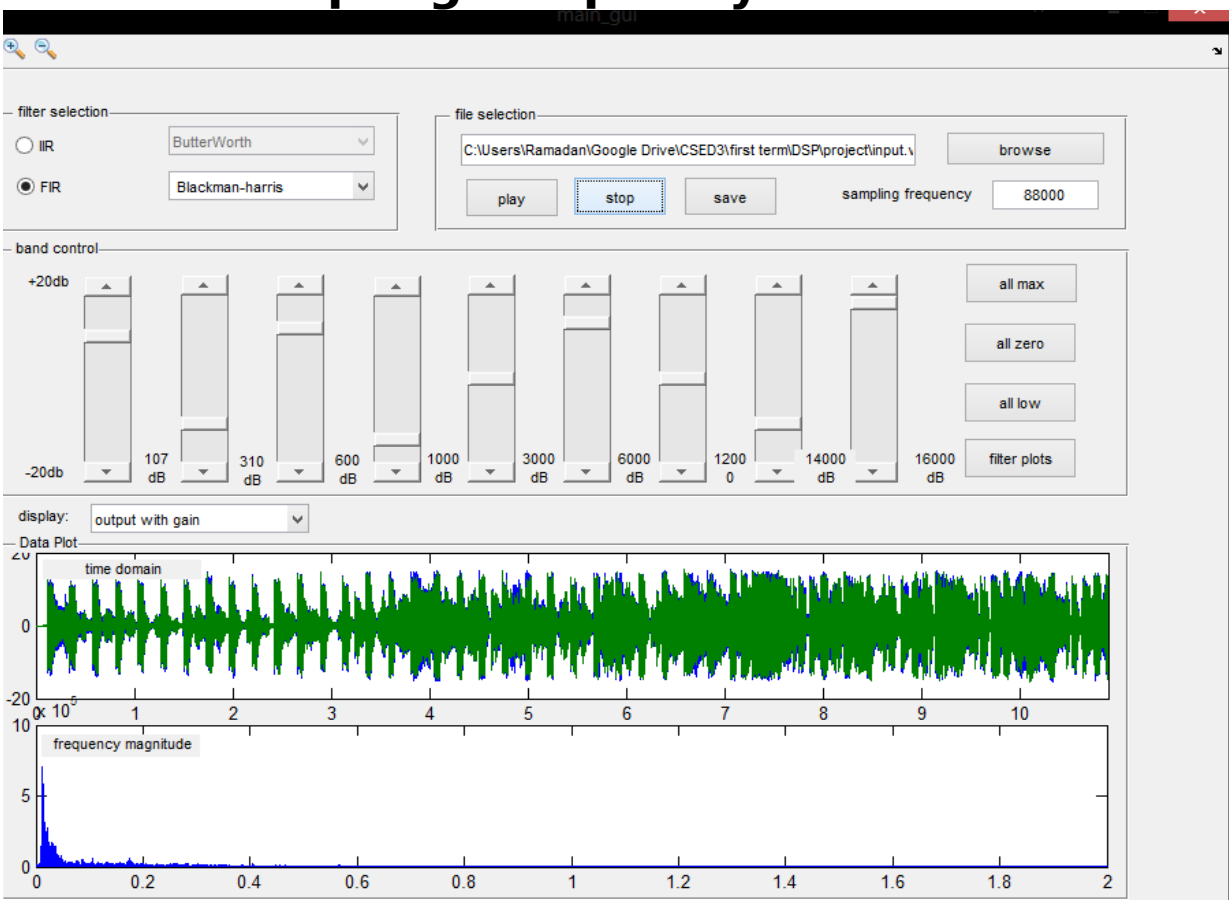
Hamming:


main_gui
filter selection
IIR
ButterWorth
FIR
Hamming
file selection
C:\Users\Ramadan\Google Drive\CSED3\first term\DSP\project\input.
browse
play
stop
save
sampling frequency
44100
band control
+20db
-20db
107 dB
310 dB
600 dB
1000 dB
3000 dB
6000 dB
12000
14000 dB
16000 dB
all max
all zero
all low
filter plots
display:
output with gain
Data Plot
time domain
frequency magnitude

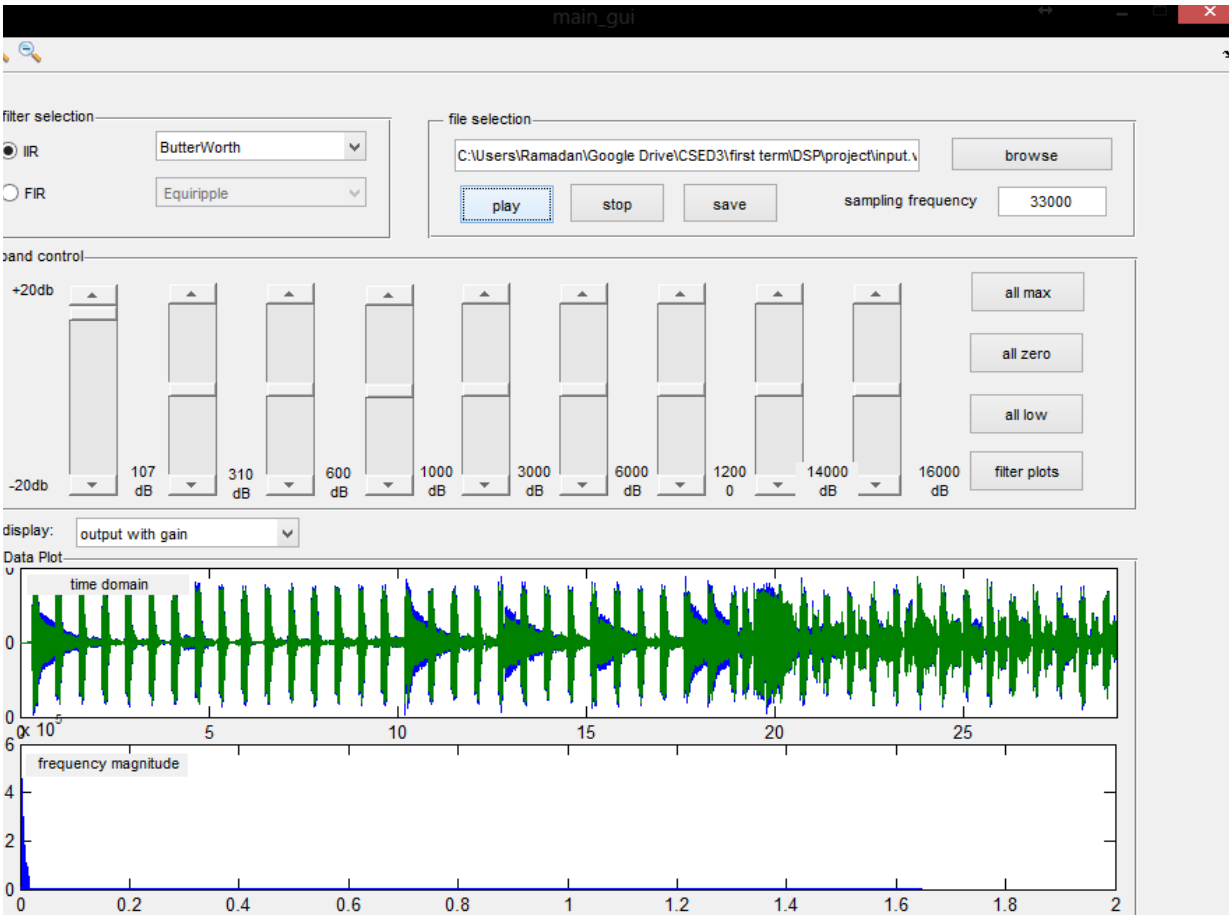
Blackman-Harris:



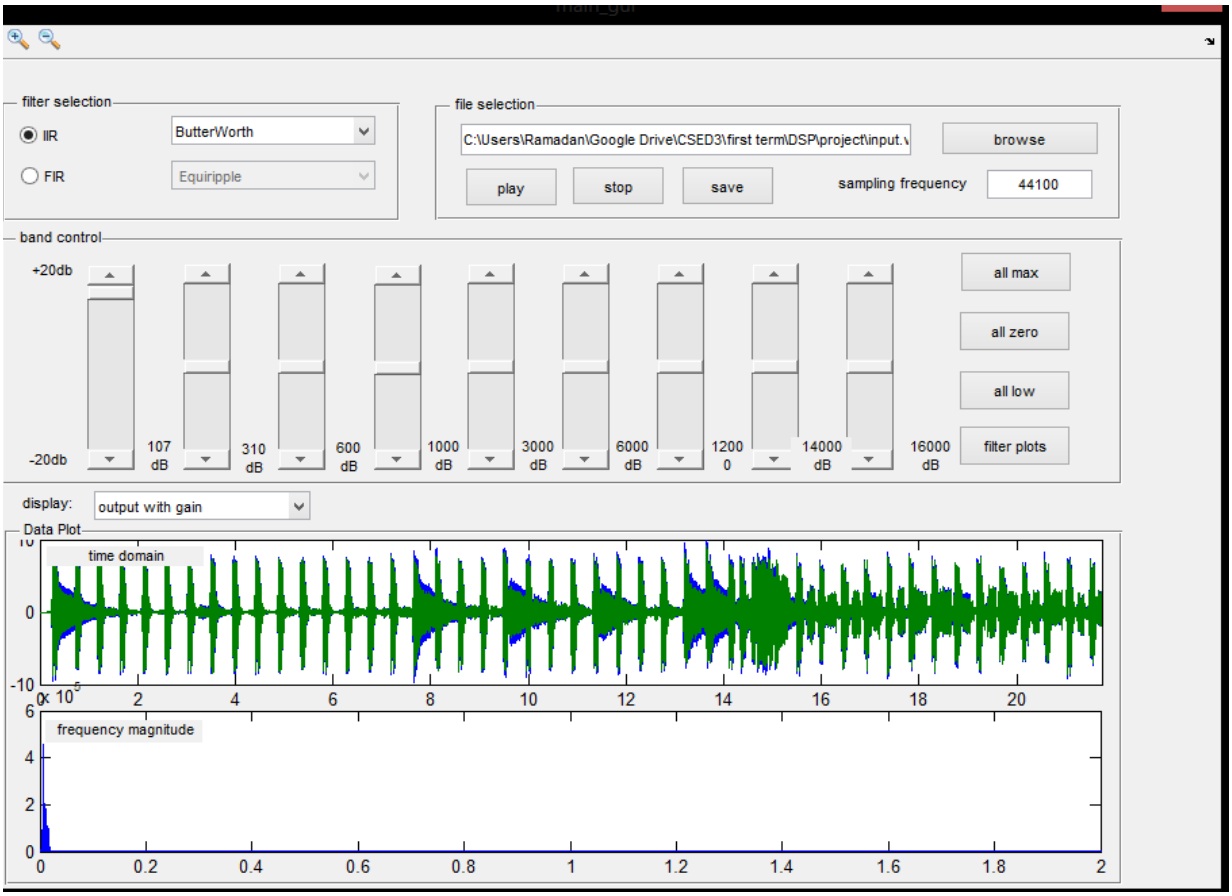
Double Sampling Frequency:



Less Frequency than original sampling frequency:



Base dominant:


filter selection
IIR
ButterWorth
FIR
Equiripple
file selection
C:\Users\Ramadan\Google Drive\CSED3\first term\DSP\project\input.
browse
play
stop
save
sampling frequency
44100
band control
+20db
-20db
107 dB
310 dB
600 dB
1000 dB
3000 dB
6000 dB
1200 0
14000 dB
16000 dB
all max
all zero
all low
filter plots
display:
output with gain
Data Plot
time domain
frequency magnitude

Filter Analysis:

Order:

Increasing the order makes the average error decreases but the max error is bounded by 9% as in gibbs phenomena. Increasing the order also increases the group delay and the cost. So we should make the order not very large and not very small. We makes it around 25.

Attenuation:

Attenuation is the ration between the output and the input.

Increasing A stop makes better performance and gets closer to the ideal filter but increases complexity.

Fpass, Fstop:

making them closer gets closer to the ideal filter as there is no intermediate band in the ideal filter. But increasing the distance between them makes the output become closer 1 at pass band.