Alexandria University
Faculty of Engineering
Computer and Communication Department

Subject

Final Project (Audio Equalizer)

Done By

Name	Id	Group	Section
Omar Hossam Eldin Ebraheem	6667	1	1
Karim Mohamed El Baroudy	6424	2	1
Loay Hesham Mohamed	6439	3	1

Supervisor

Prof Dr. Hassan El Raggal

2020 – 2021

Introduction:

In this Final Project, it is required to develop an Audio Equalizer using MATLAB Software. The main function of the Audio Equalizer is to vary the gain of each specific band as the user prefers. If the user likes base he will increase the gain of the low frequencies. In this Final Project (Audio Equalizer), it is expected from the user to input the following: (The wave file name, Gain of each of the frequency band Filters in dB, Type of filters used (FIR-IIR), Output Sample Rate). In the main implementation of the Audio Equalizer it is required to develop and implement the frequency band filters in the following bands: [filter 1: (0-170 Hz), filter 2: (170-310 Hz), filter 3: (310-600 Hz), filter 4: (600-1000 Hz), filter 5: (1-3 KHz), filter 6: (3-6 KHz), filter 7: (6-12 KHz), filter 8: (12-14 KHz), filter 9: (14-16 KHz)]. It has been concluded that filter 1 is a Low Pass Filter, while filters (2, 3, 4, 5, 6, 7, 8, 9) are Band Bass Filters. After that it is required to analyse and export the following specifications of the developed filters: (magnitude, phase, impulse and step response, order, and gain/poles/zeros). Then Filter the wave file using the filters developed for both types (IIR, FIR). Then Draw the output signals from filters in Time and frequency domains. Then Amplify the output signals using the user defined gain. Then Add the amplified - output signals in time domain to form composite signal. Then Draw and compare the composite signal with the original signal in both domains (Time Domain and Frequency Domain). Then Play the output wave signal. In this project the output signal will be played and displayed on MATLAB Figures with the desired output sample rate entered by the user. It has been chosen three test wav files to be tested after implementing the project, also all the required specifications will be applied to those three test wav files. After that it is required to submit only ONE PDF REPORT containing: (Copy of Code, Different sample runs of code including the following cases: [If design is using FIR filter, If design is using IIR filters, Output signal in case if doubling output sample rate or decreasing it to half], the analysis of each filter and the exported outputs, All figures of signals in time and frequency domain).

Discussion:

All the required frequency band filters with the given frequency bands, have been implemented in both FIR and IIR types, according to the choice of the user. If the user wants to filter his wave file using filters of type FIR, he just simply enters the type of the desired filters to be FIR. If he wants to filter his wav file using filters of type IIR, also he just simply enters the type of the desired filters to be IIR.

1. The Analysis of each filter

- In this section, the analysis of each filter and the exported outputs (magnitude, phase, impulse and step response, order, and gain/poles/zeros), will be showed for each filter for the two types (FIR – IIR).
  - The analysis of each filter and the exported outputs for the filters of type FIR of order **(N=25)**, as following:
    - Magnitude and Phase of the Developed filters of type FIR as shown in the figure below *(Figure 1.1)*:

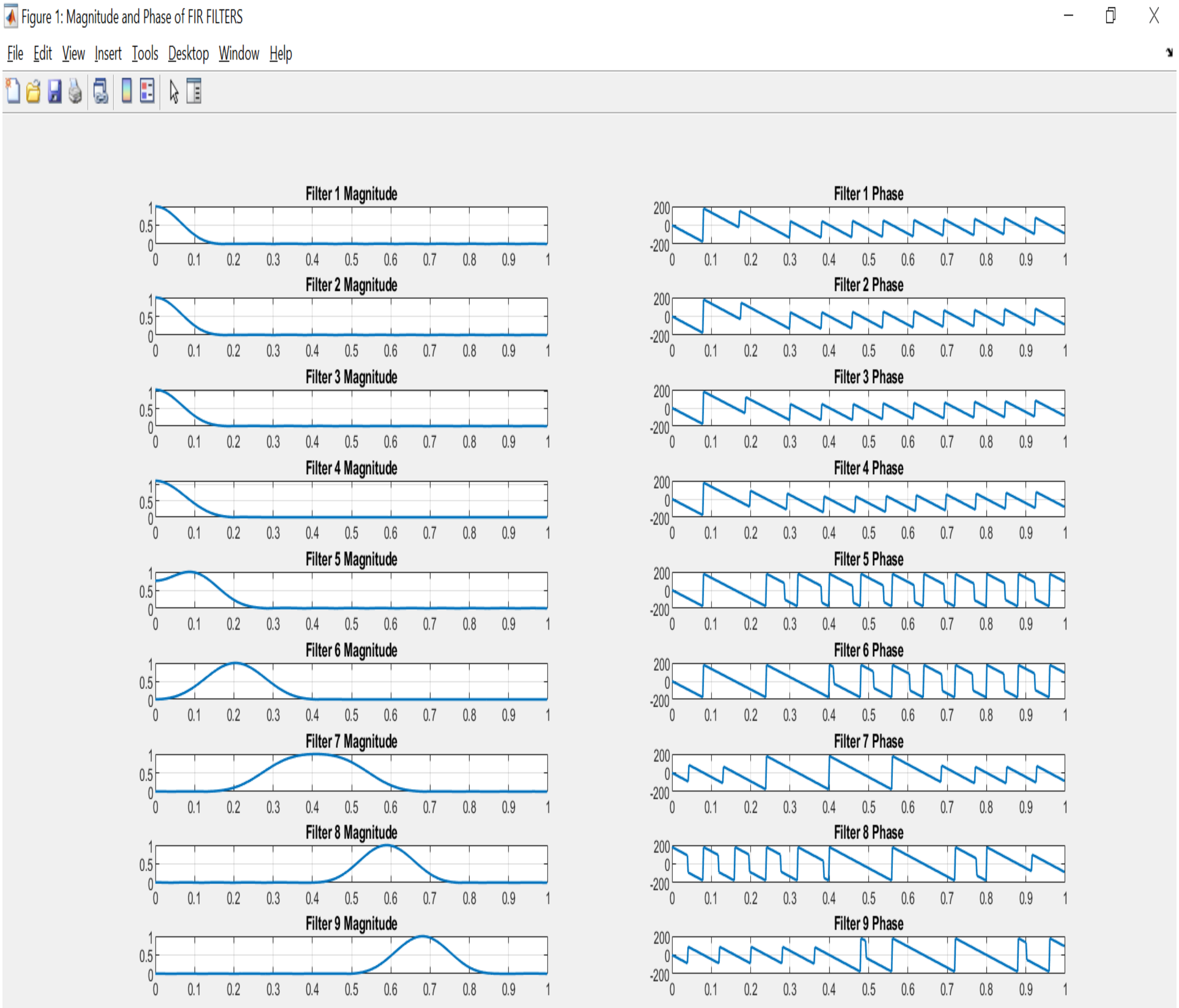


Figure 1.1

- Impulse Response and Step Response of the Developed filters of type FIR as shown in the figure below *(Figure 1.2)*:

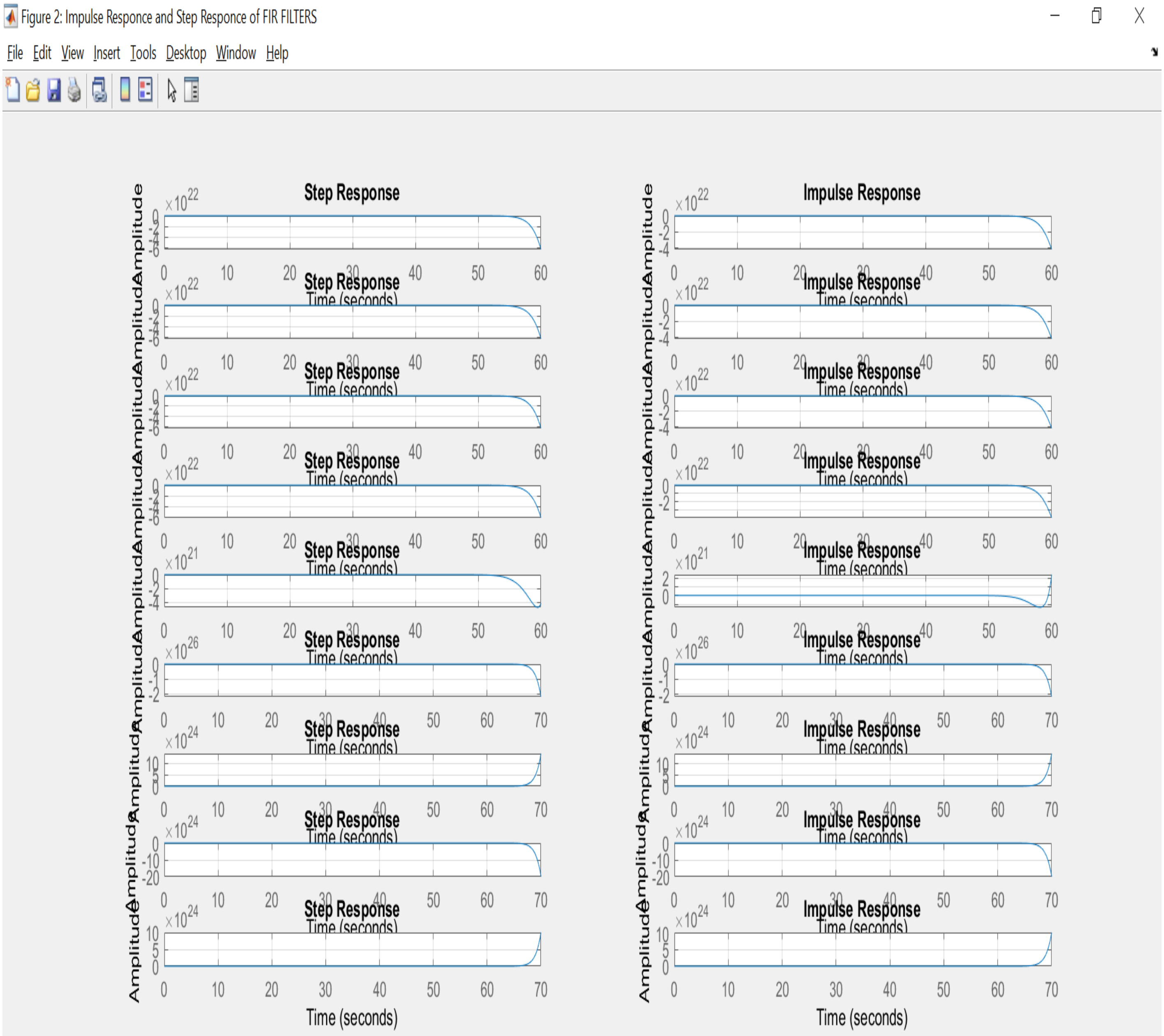


Figure 1.2

- Note that the Impulse Response and the Step Response of the developed filters shown in the figure above *(Figure 1.2)*, are arranged ascendingly (i.e., Impulse Response and the Step Response of filter 1 are shown in the first row, of filter 2 are shown in the second row and so on).

- Poles and Zeros of the Developed filters of type FIR as shown in the figure below *(Figure 1.3)*:

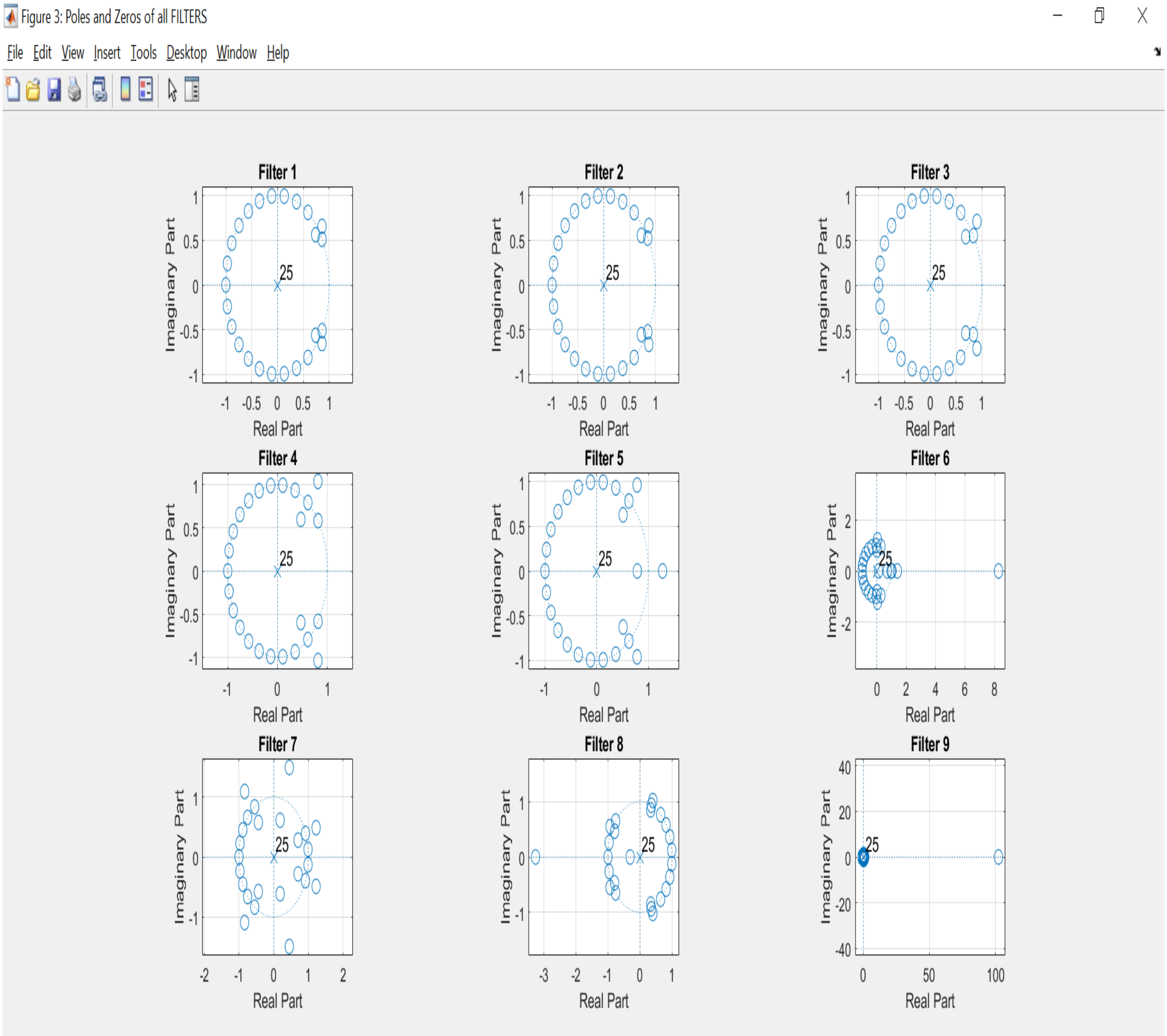


Figure 1.3

- Note that from the figure above *(Figure 1.3)*, all filters of type FIR are stable systems, because FIR system is always stable.

- The analysis of each filter and the exported outputs for the filters of type IIR of order **(N=4)**, as following:
    - Magnitude and Phase of the Developed filters of type IIR as shown in the figure below *(Figure 1.4):*

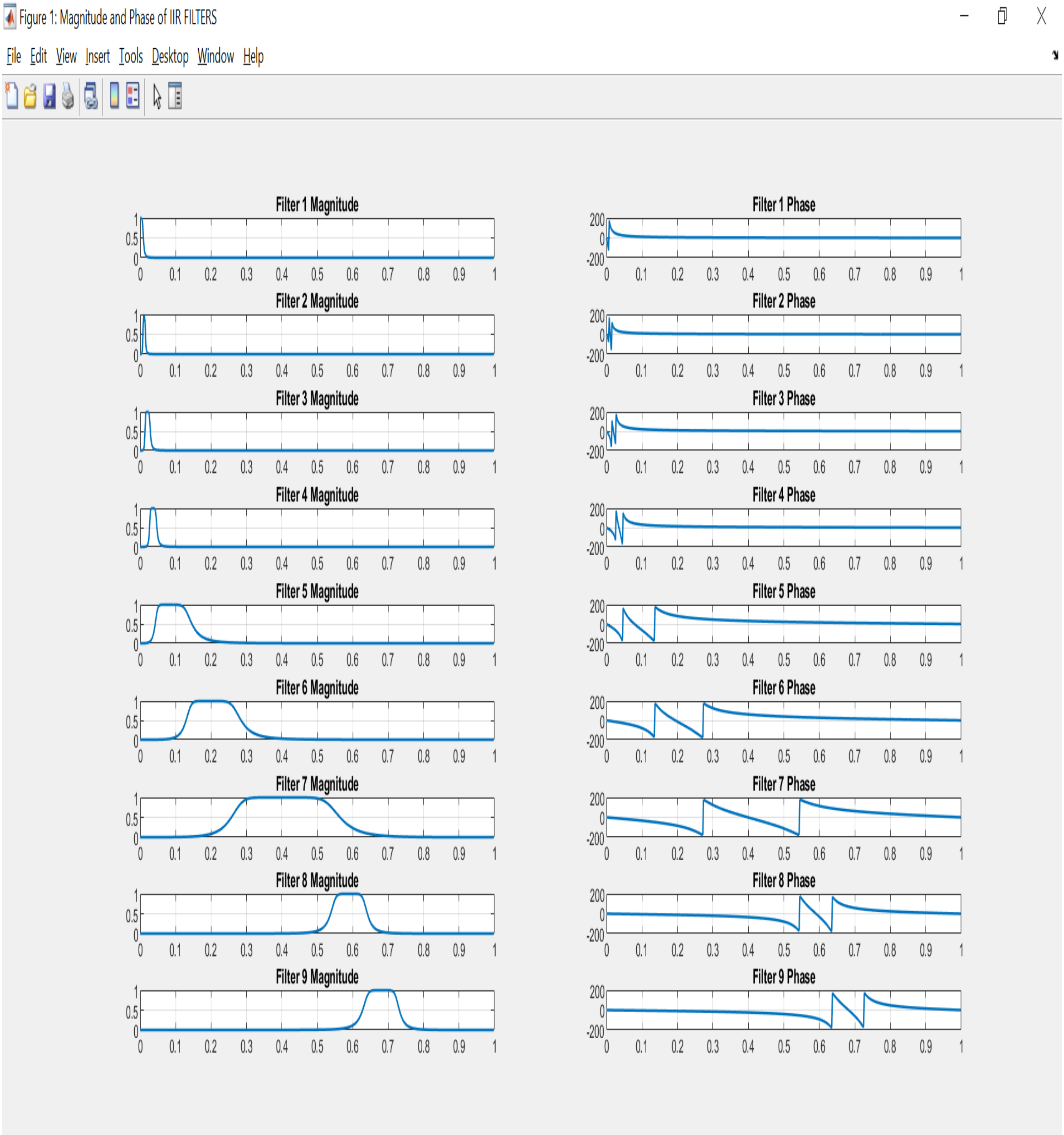


Figure 1.4

- Impulse Response and Step Response of the Developed filters of type IIR as shown in the figure below *(Figure 1.5)*:

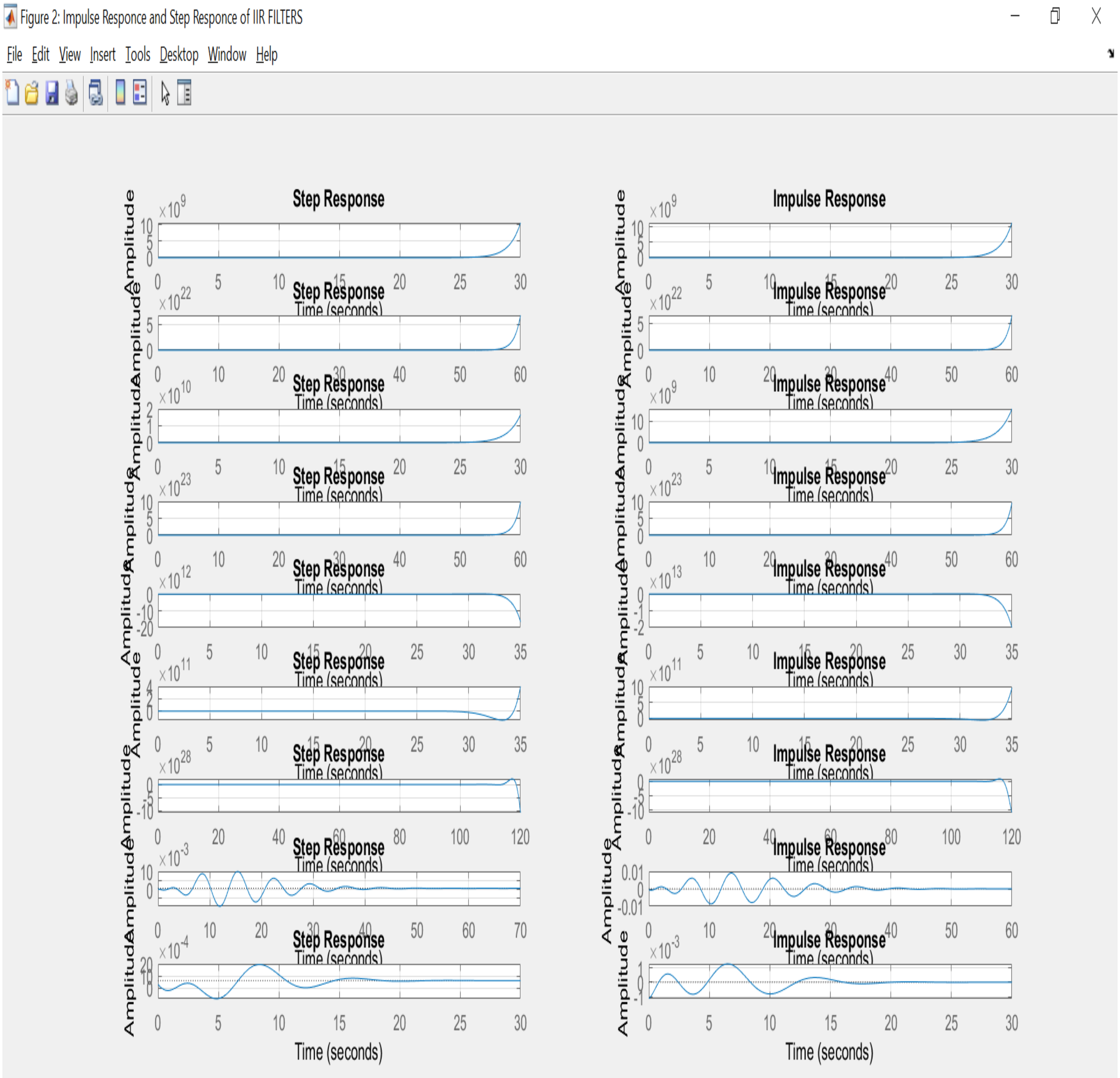


Figure 1.5

- Note that the Impulse Response and the Step Response of the developed filters shown in the figure above *(Figure 1.5)*, are arranged ascendingly (i.e., Impulse Response and the Step Response of filter 1 are shown in the first row, of filter 2 are shown in the second row and so on).

- Poles and Zeros of the Developed filters of type IIR as shown in the figure below *(Figure 1.6):*

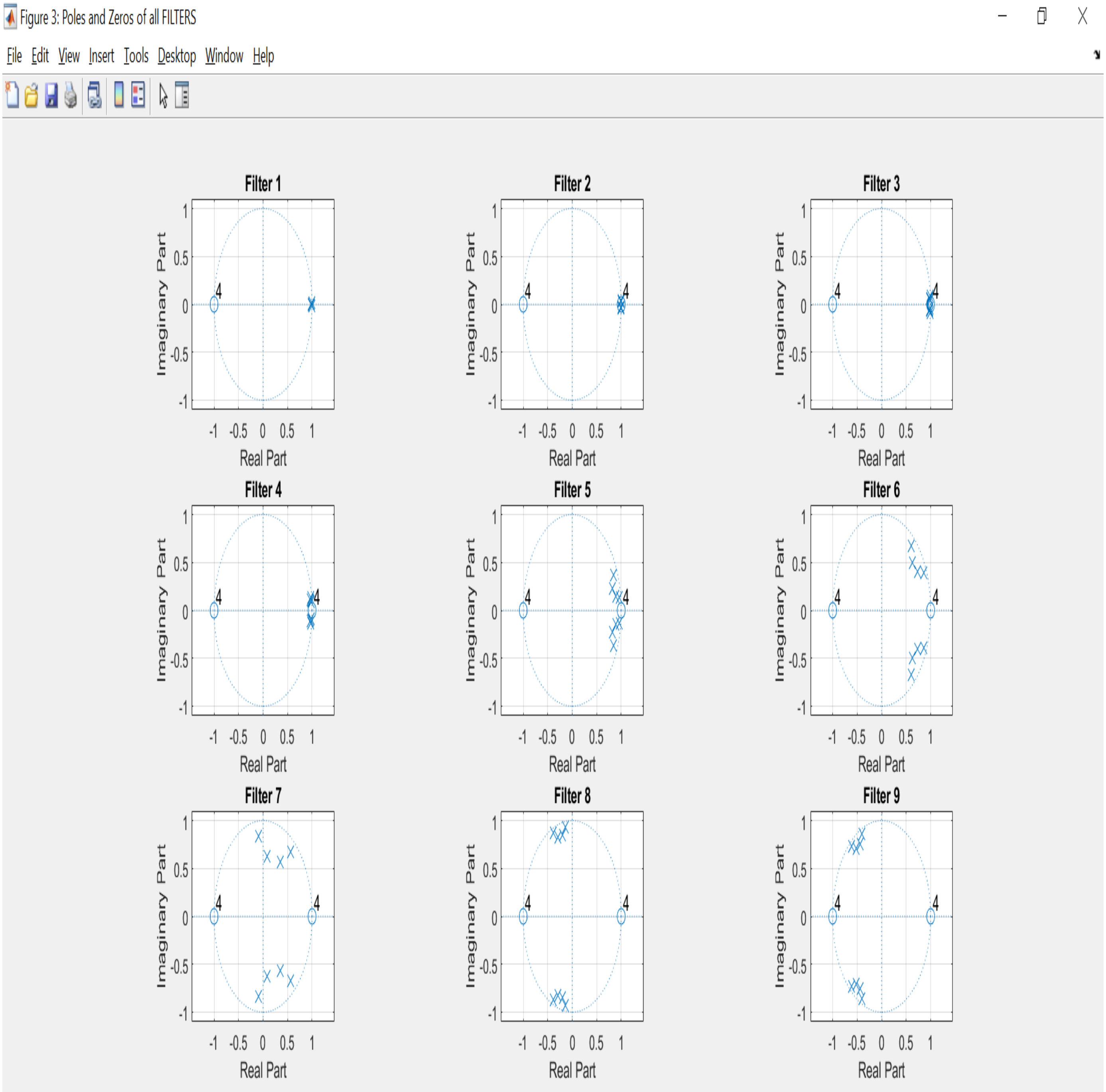


Figure 1.6

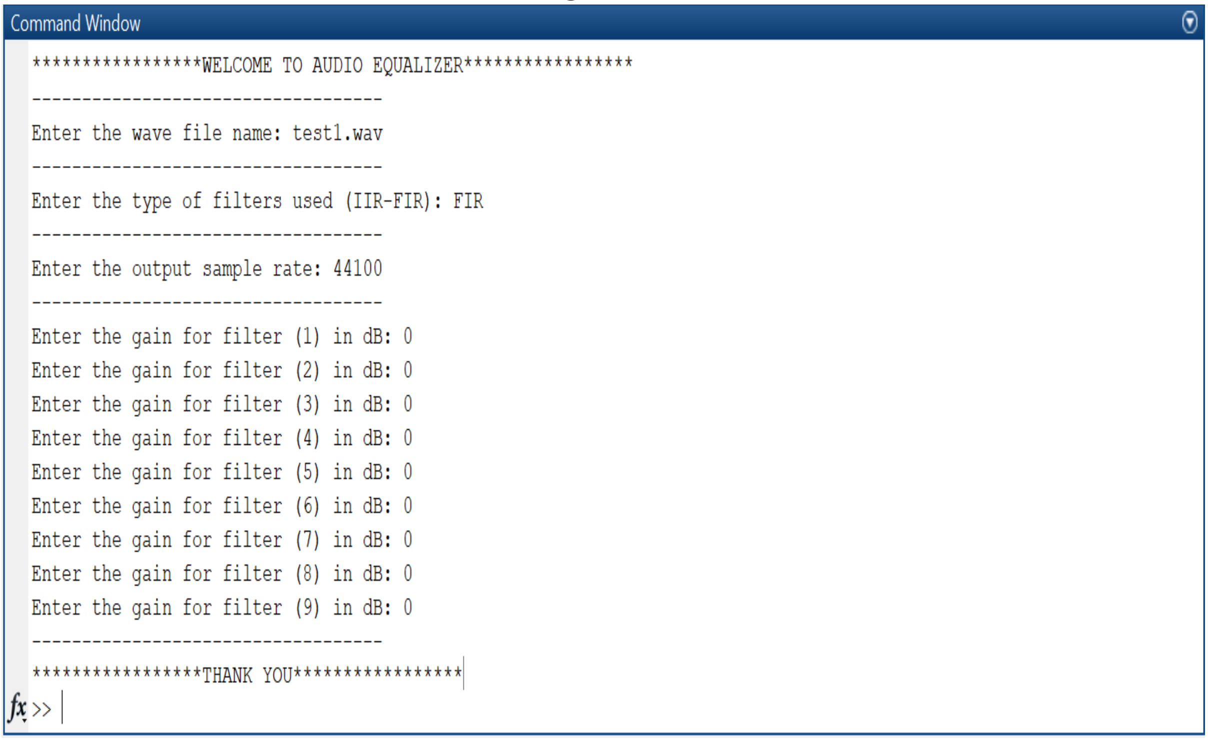
➢ Note that from the figure above *(Figure 1.6)*, filters of type IIR may be stable and may be not, because IIR systems may be stable and may be not, depends on the locations of their poles.

2. Different Sample Runs of Code

a) If the design is using FIR Filters:

➢ Applying the test1 wav file to the filters of type FIR:

- The user interface is as shown in the figure below *(Figure 2.a.1):*



```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
----------------------------------
Enter the wave file name: test1.wav
----------------------------------
Enter the type of filters used (IIR-FIR): FIR
----------------------------------
Enter the output sample rate: 44100
----------------------------------
Enter the gain for filter (1) in dB: 0
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 0
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 0
----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.a.1

- The output signals in time domain and frequency domain:

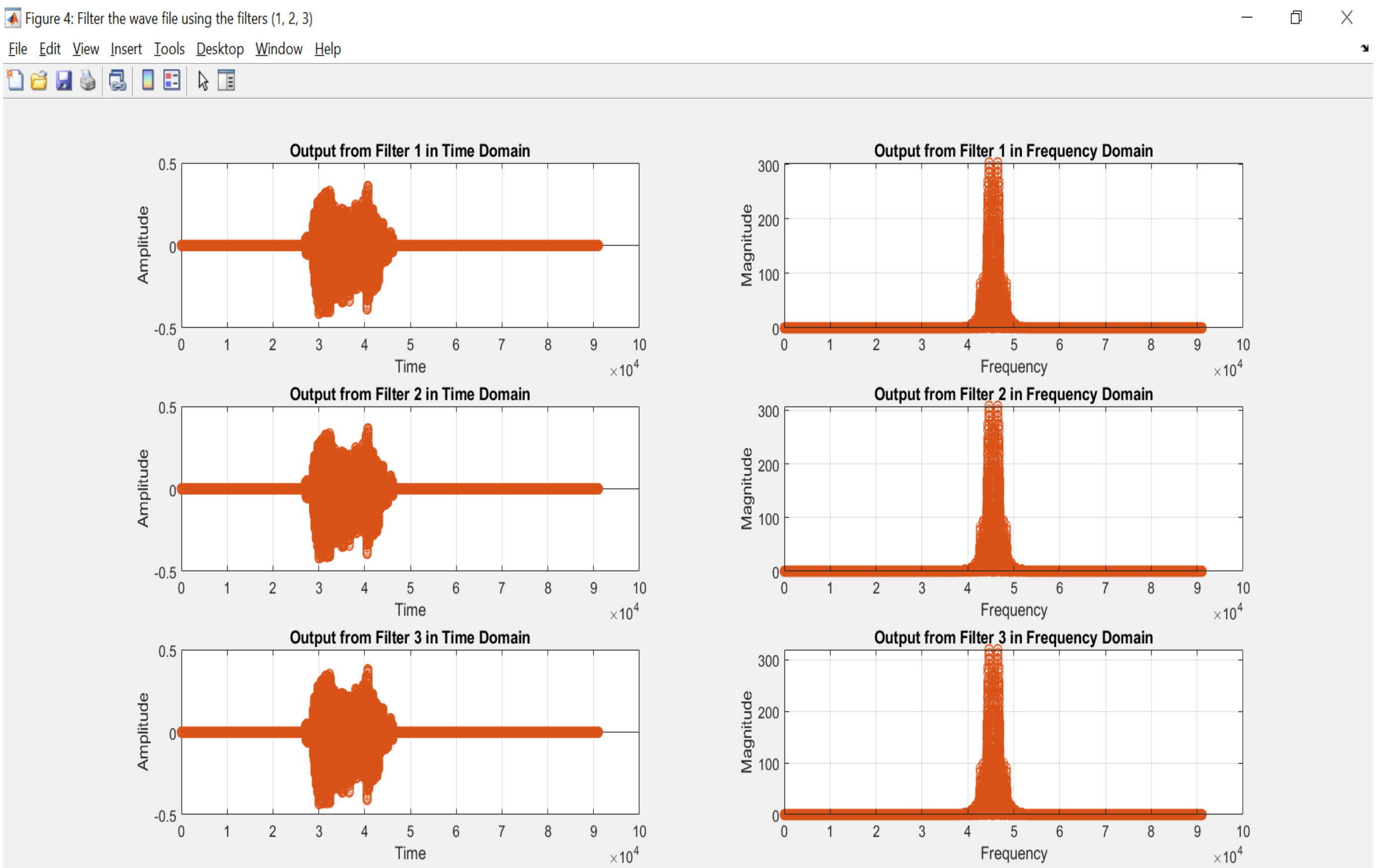


Figure 2.a.2

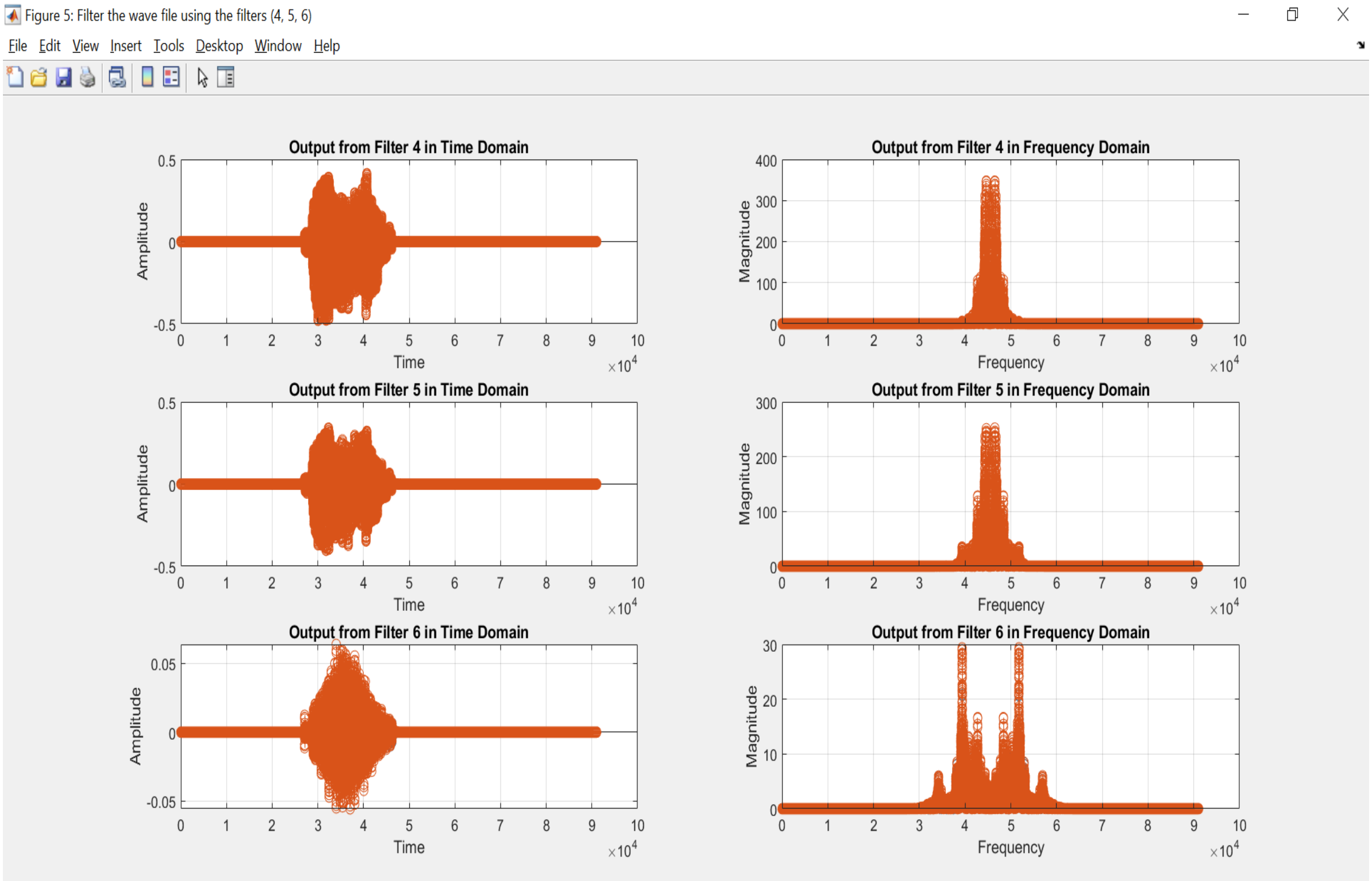


Figure 2.a.3

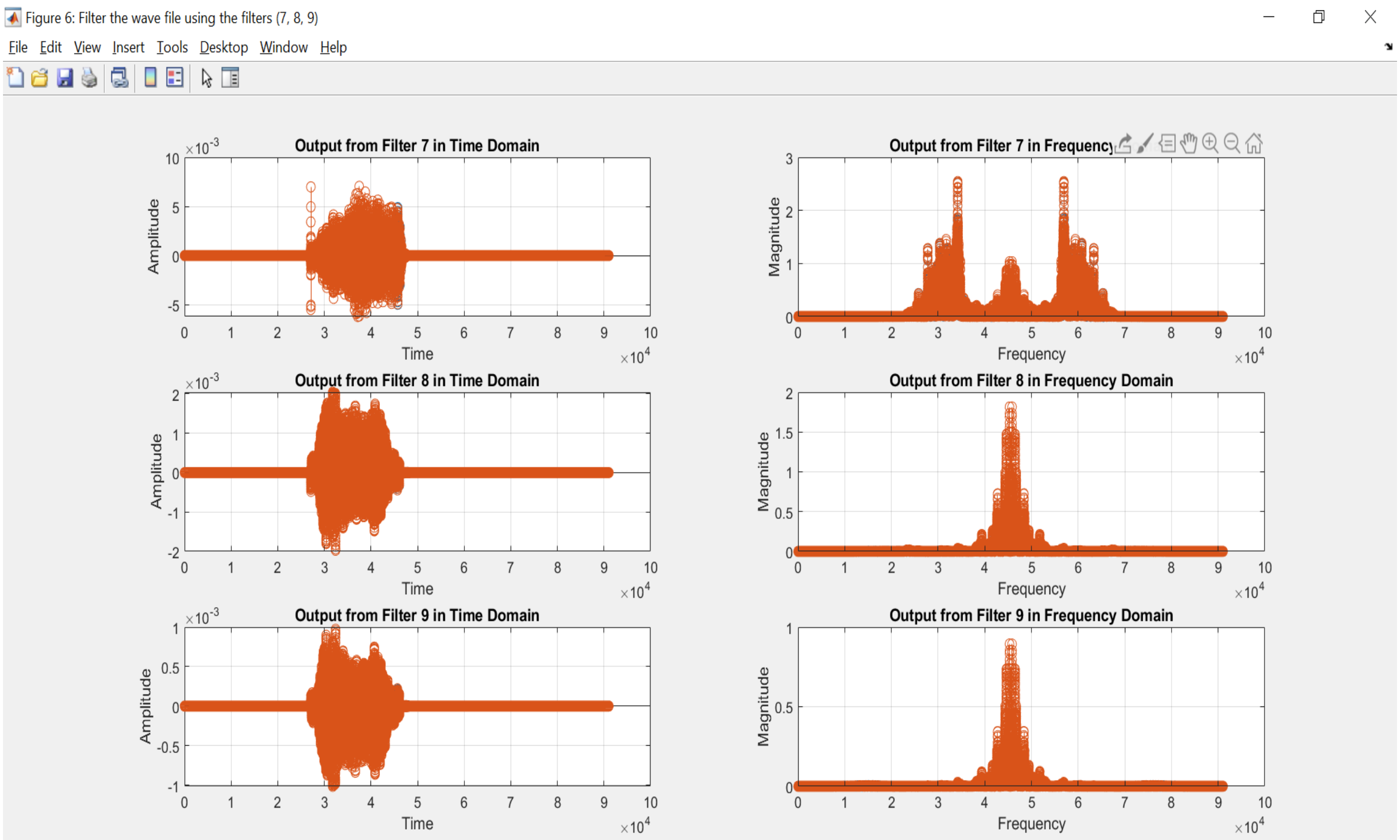


Figure 2.a.4

- Comparison between the Composite Signal amplified with gain (0 dB ➔ 1) with the Original Signal in Time Domain:

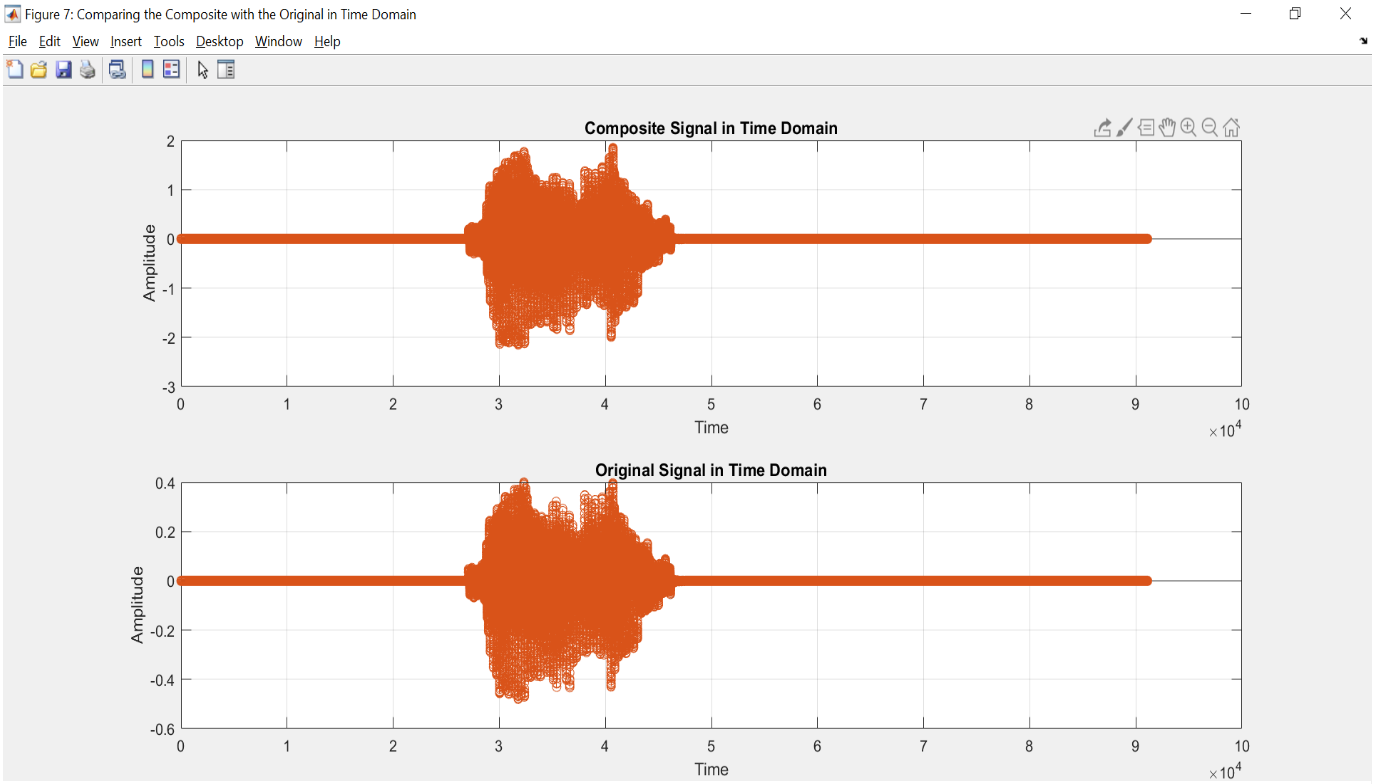


Figure 2.a.5

- Comparison between the Composite Signal amplified with gain (0 dB ➔ 1) with the Original Signal in Frequency Domain:

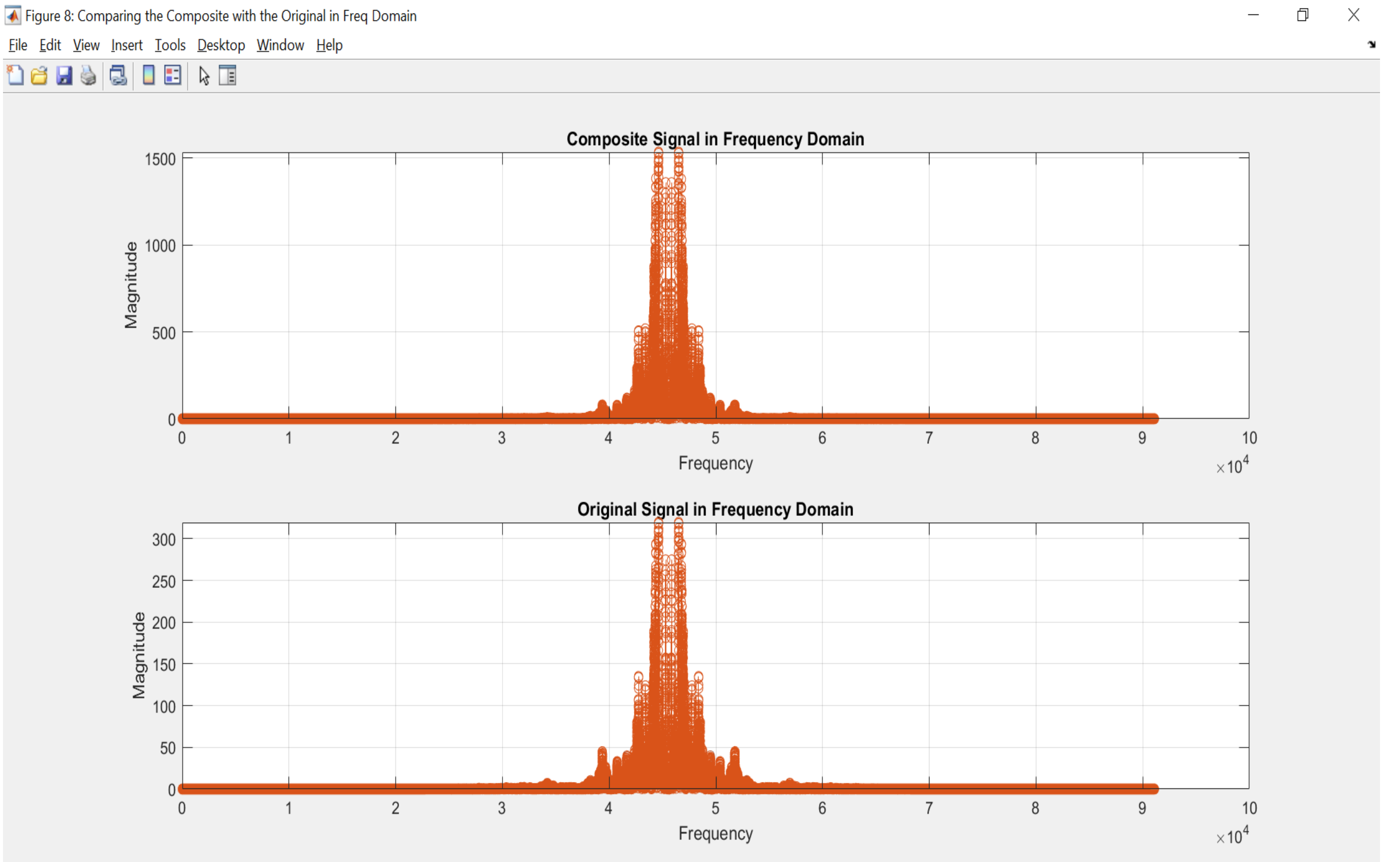


Figure 2.a.6

- Playing the output wav signal with the desired output sample rate:

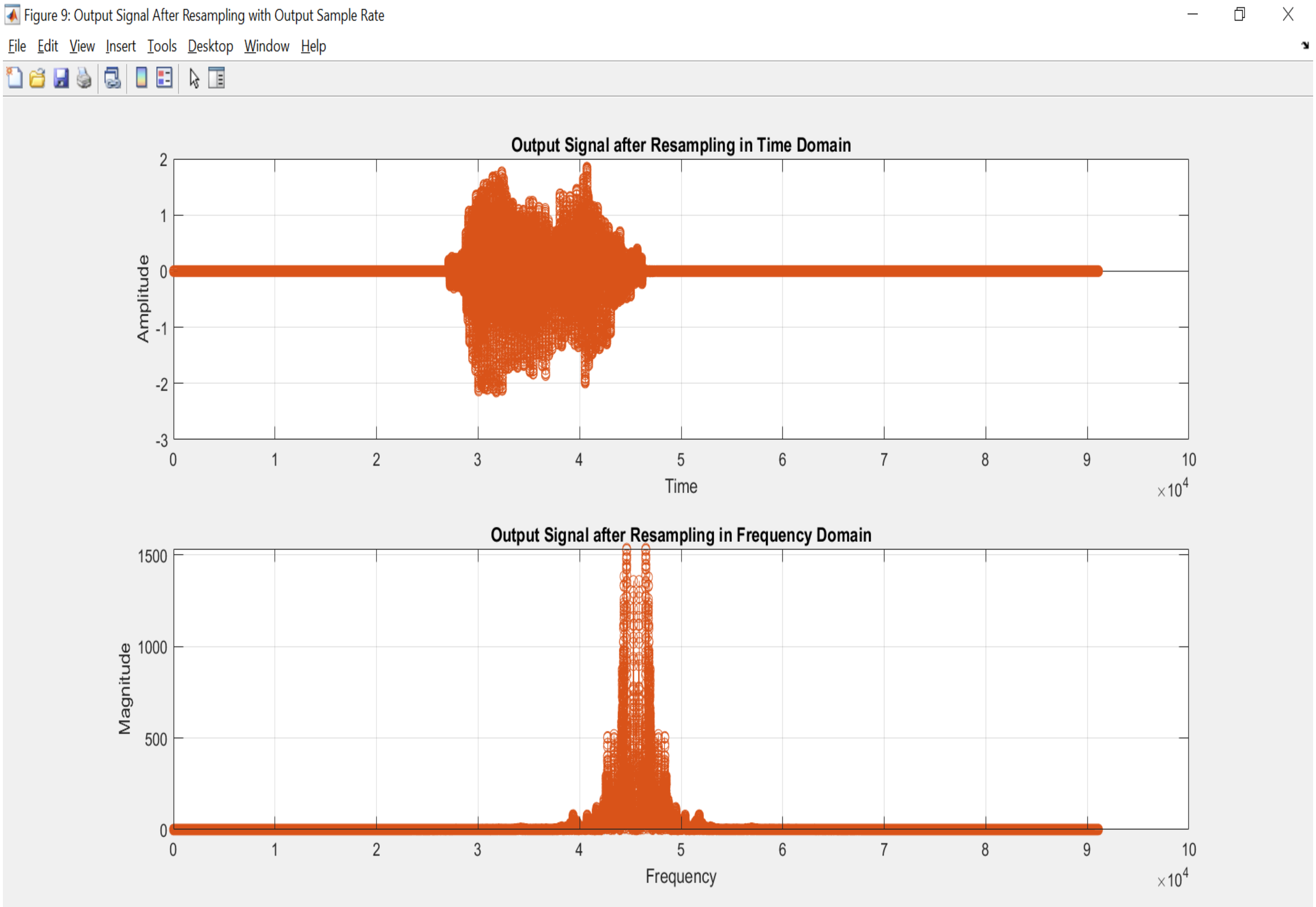


Figure 2.a.7

- Note that the output wav signal has been played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal.

➢ Output signal in case if doubling output sample rate:

- The user interface is as shown in the figure below *(Figure 2.a.8):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test1.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): FIR
-----------------------------------
Enter the output sample rate: 88200
-----------------------------------
Enter the gain for filter (1) in dB: 0
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 0
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 0
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.a.8

- Playing the output wav signal in case of doubling the output sample rate:

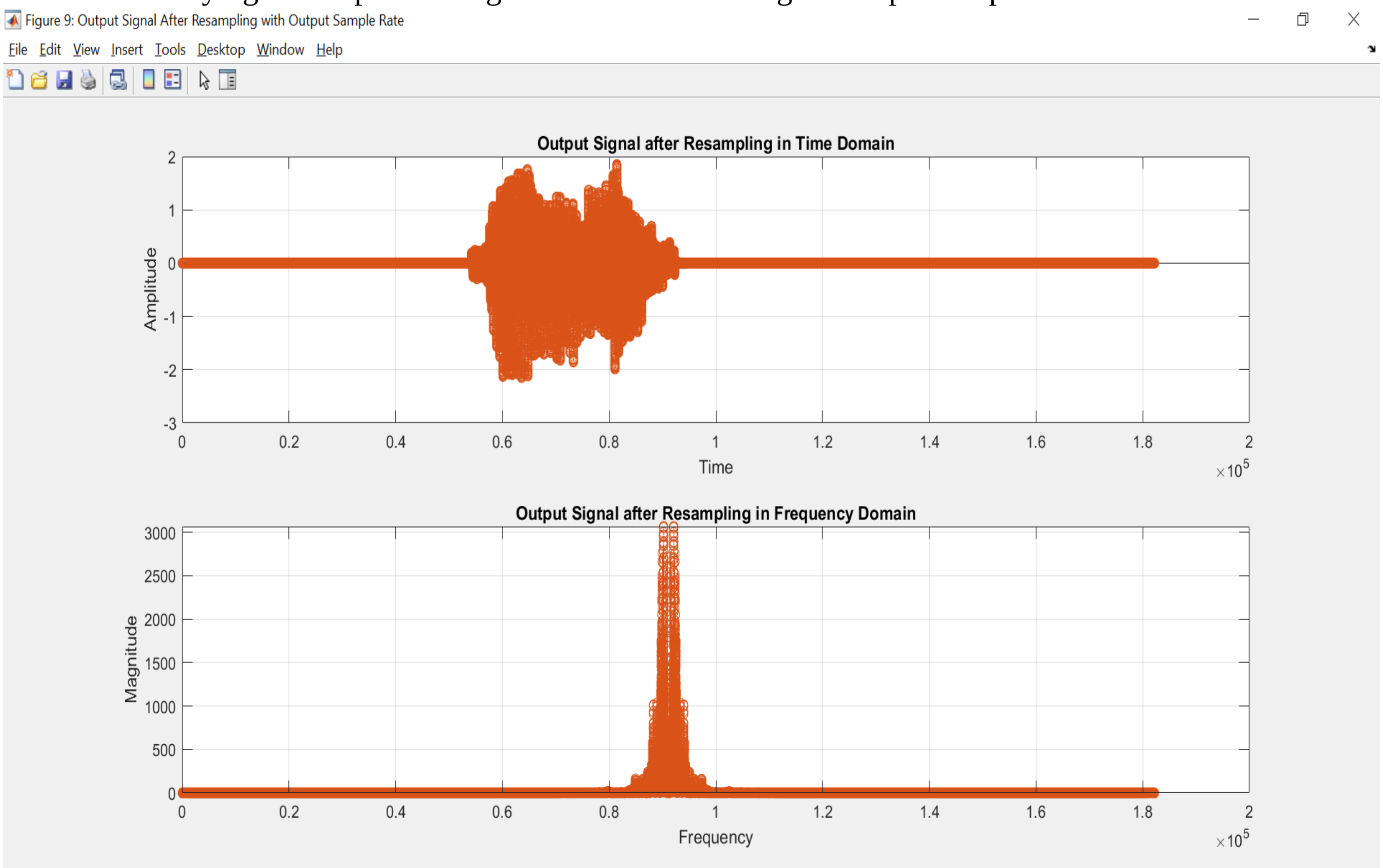


Figure 2.a.9

- Note that the output wav signal has been resampled with the double output sample rate and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with double output sample rate with those before resampling (Mentioned above).

- ➢ Output signal in case if decreasing output sample rate to half:
  - The user interface is as shown in the figure below *(Figure 2.a.10)*:

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test1.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): FIR
-----------------------------------
Enter the output sample rate: 22050
-----------------------------------
Enter the gain for filter (1) in dB: 0
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 0
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 0
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2. a.10

- Playing the output wav signal in case of decreasing the output sample rate to half:

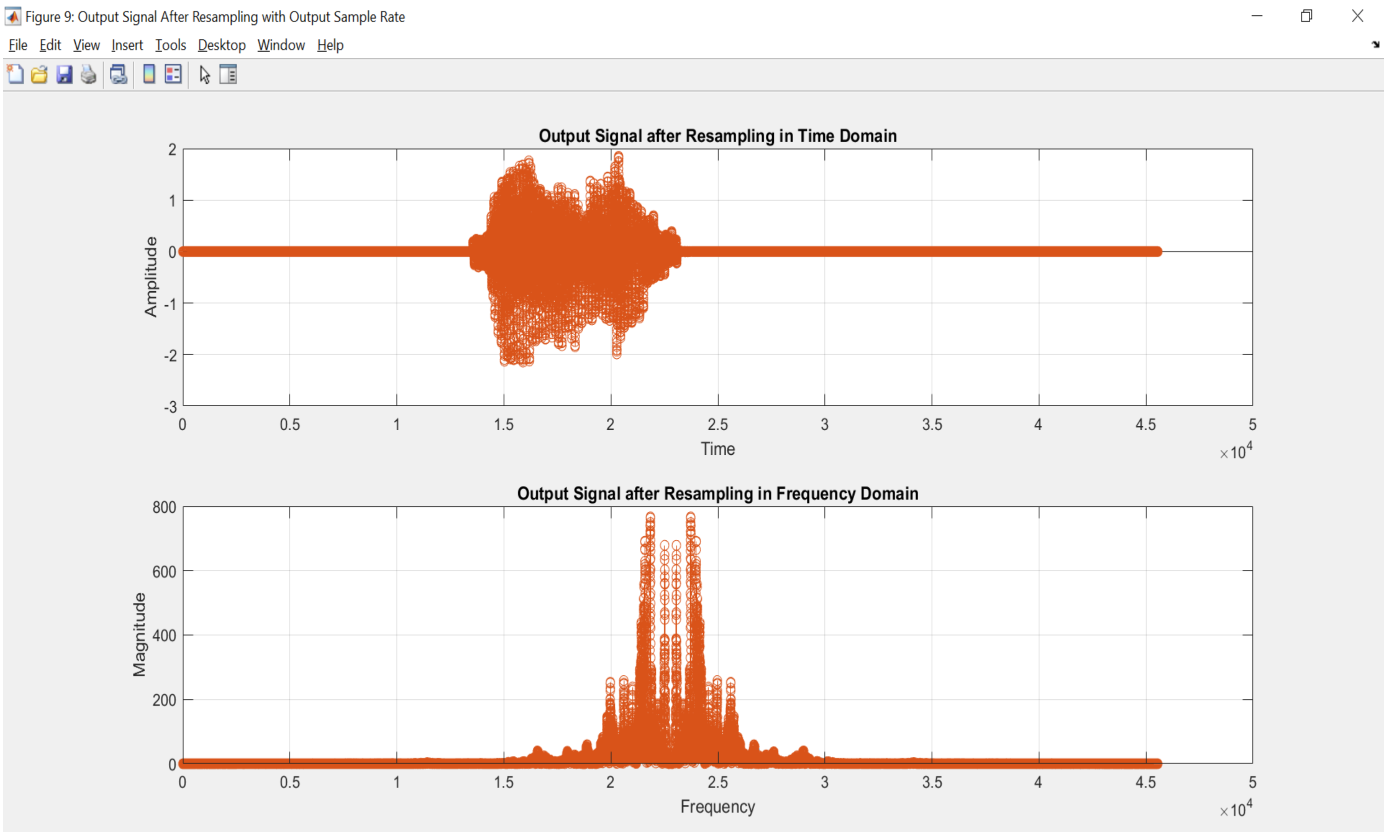


Figure 2.a.11

- Note that the output wav signal has been resampled with decreasing the output sample rate to half and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with decreasing the output sample rate to half with those before resampling (Mentioned above).

- Applying the test1 wav file to the filters of type FIR:
  - The user interface is as shown in the figure below *(Figure 2.a.12):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test1.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): FIR
-----------------------------------
Enter the output sample rate: 44100
-----------------------------------
Enter the gain for filter (1) in dB: 20
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 20
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 20
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.a.12

  - The output signals in time domain and frequency domain :

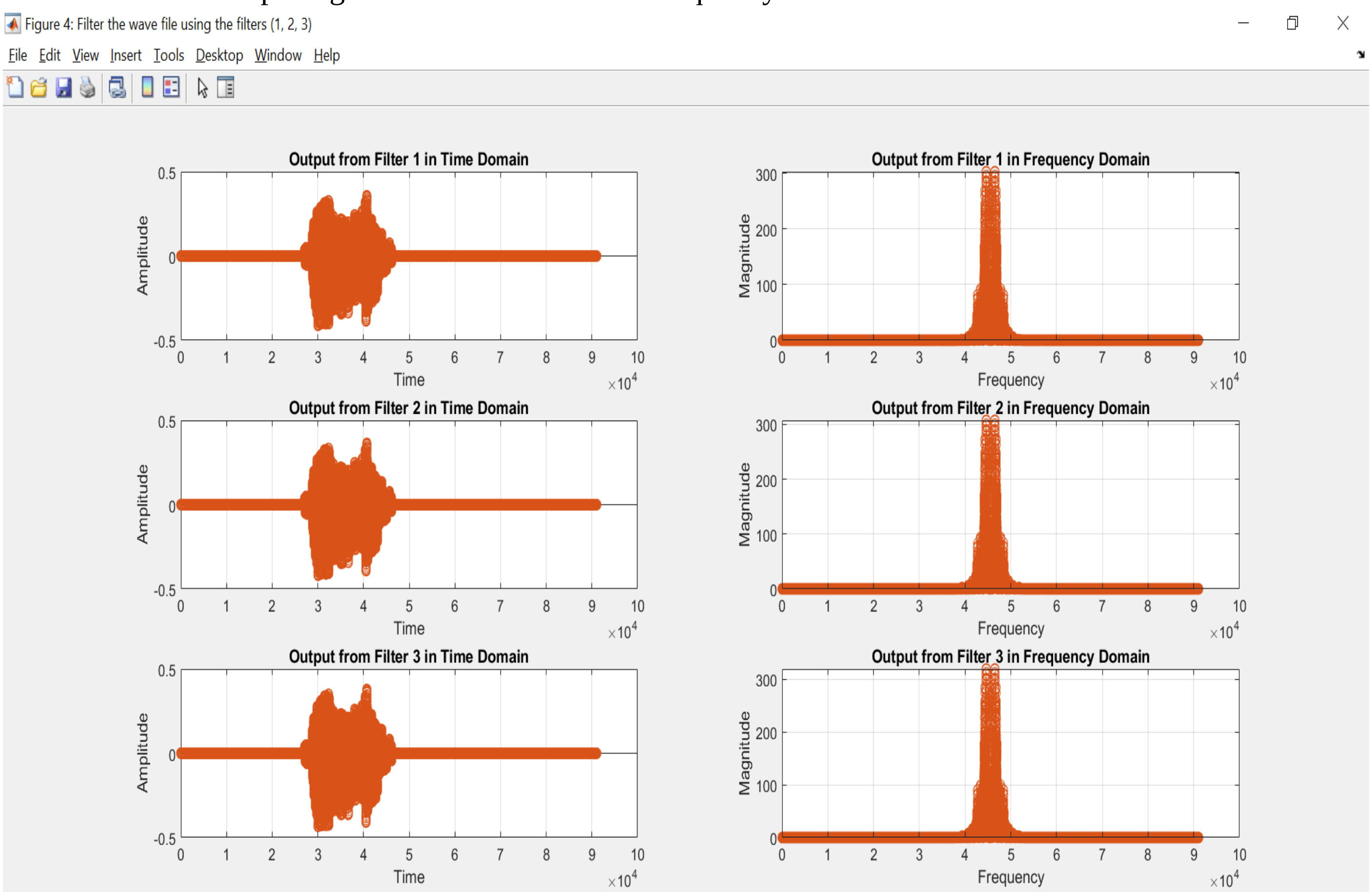


Figure 2.a.13

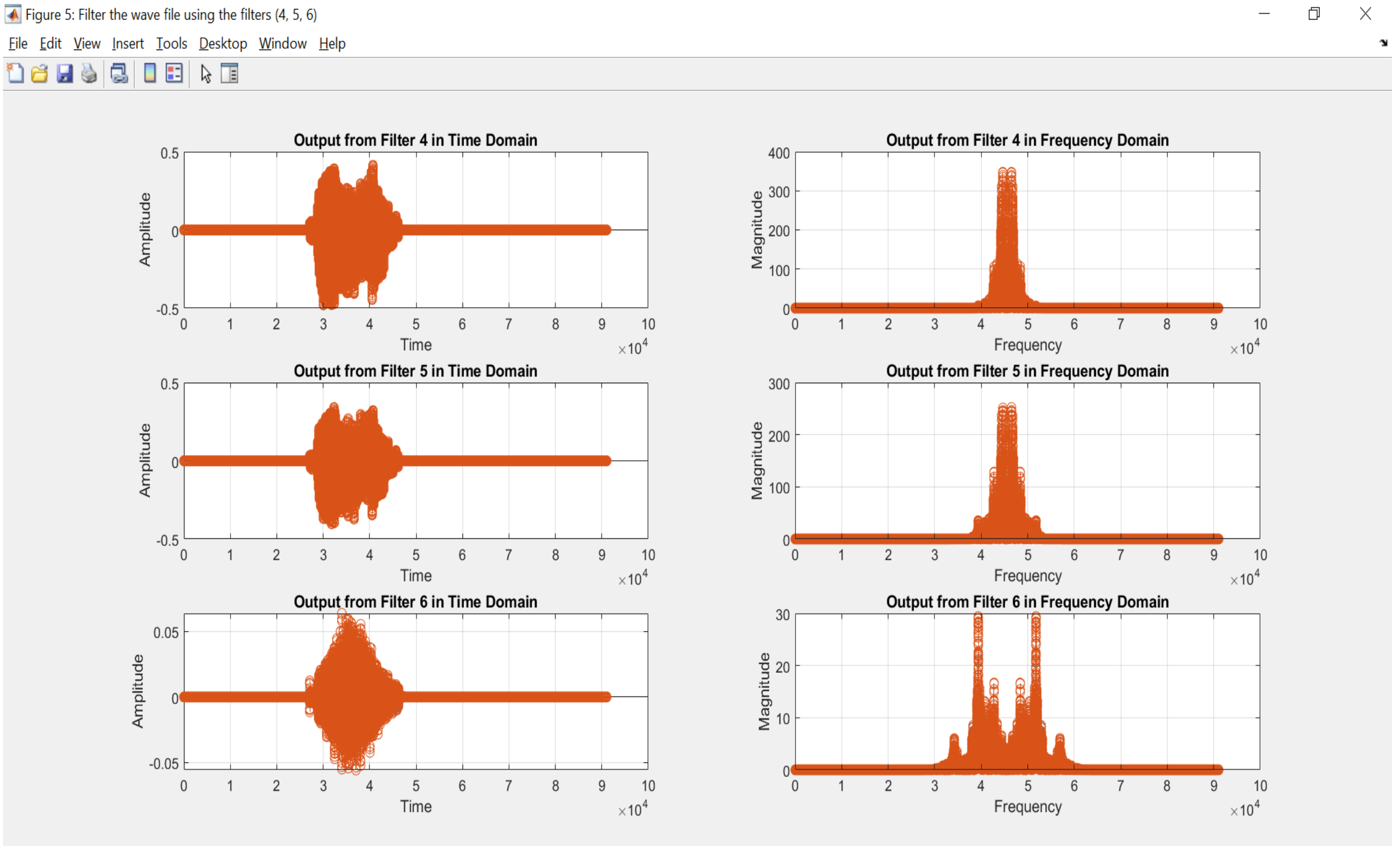


Figure 2.a.14

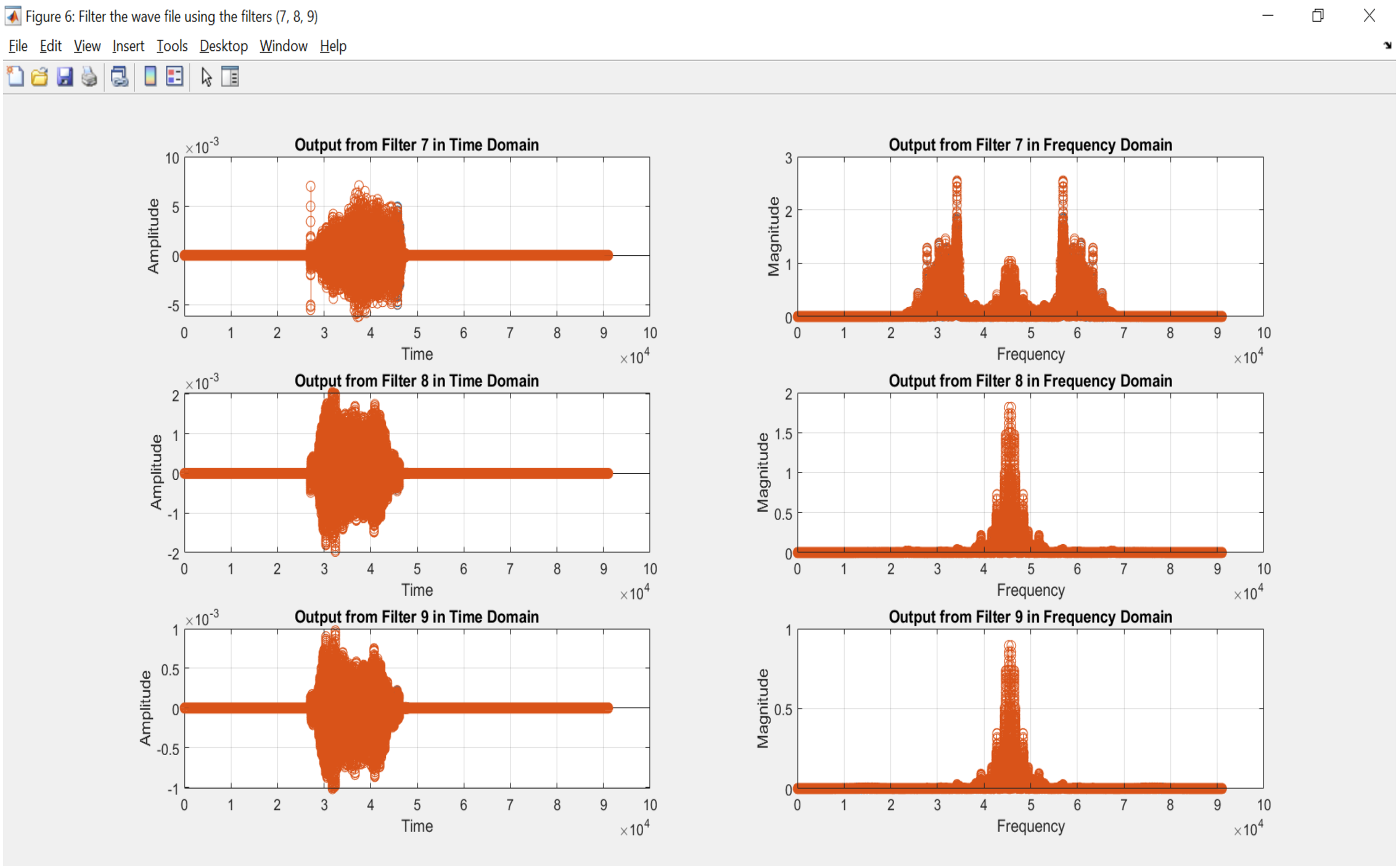


Figure 2.a.15

- Comparison between the Composite Signal amplified with the gains showed in the figure above *(Figure 2.a.12)* with the Original Signal in Time Domain:

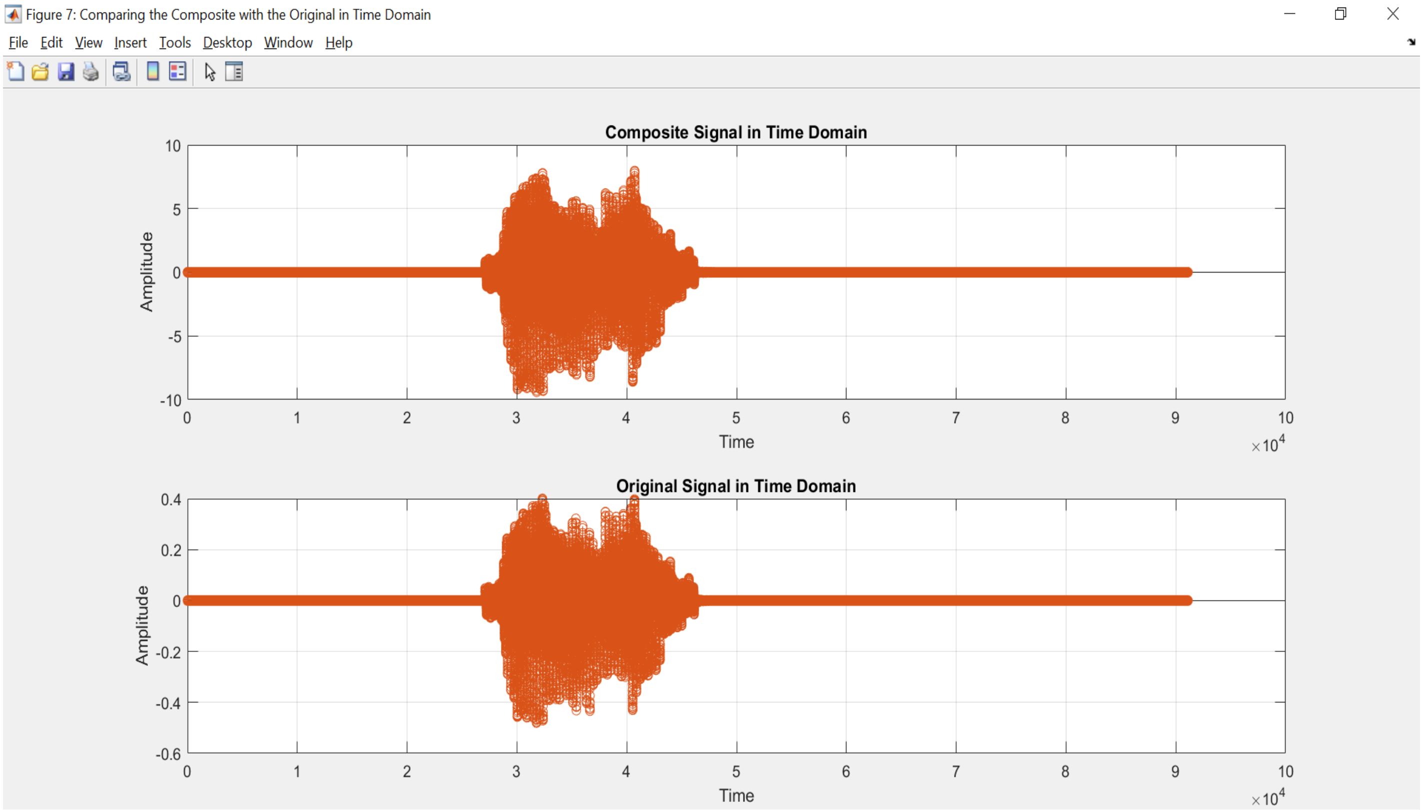


Figure 2.a.16

- Comparison between the Composite Signal amplified with the gains showed in the figure above *(Figure 2.a.12)* with the Original Signal in Frequency Domain:

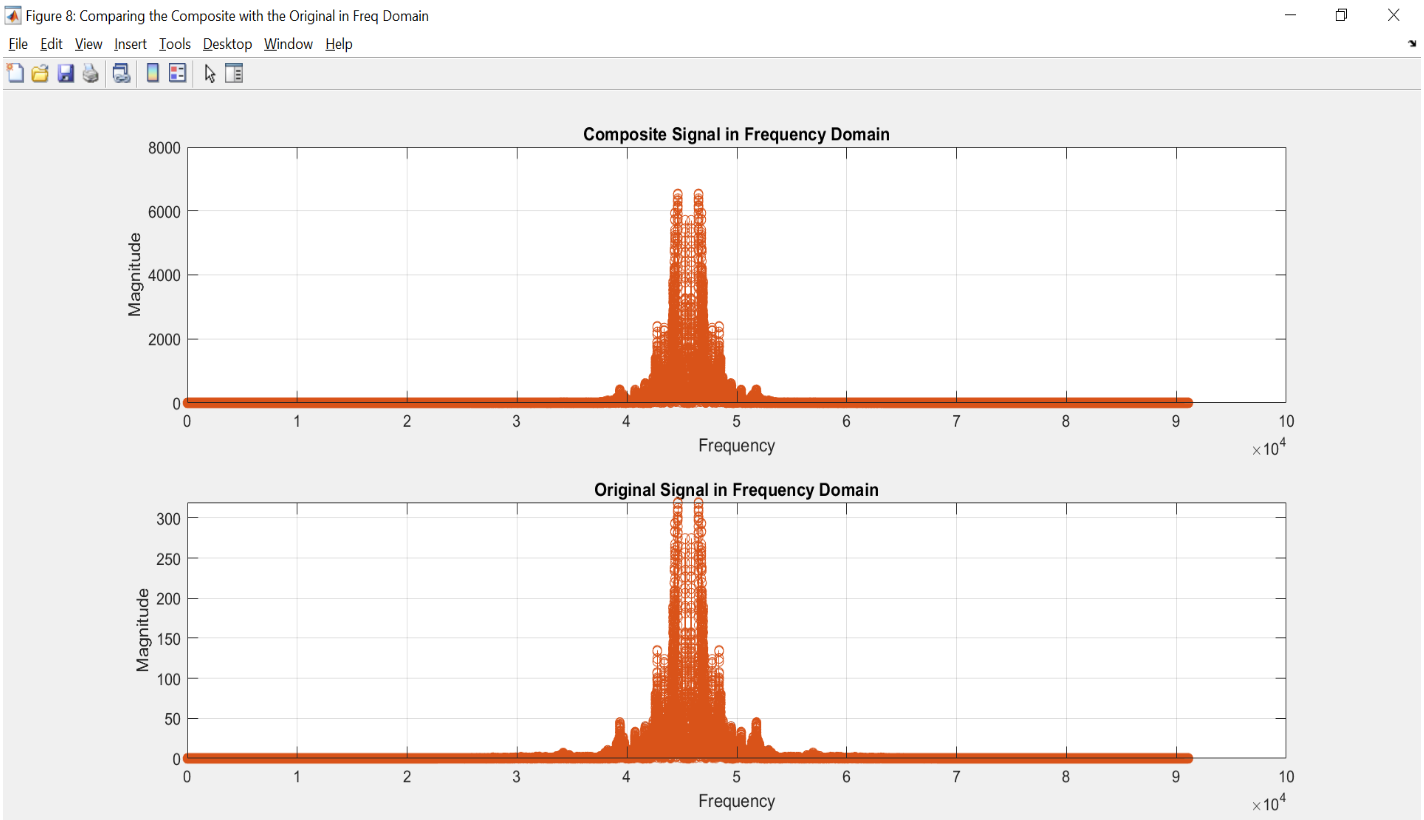


Figure 2.a.17

- Playing the output wav signal with the desired output sample rate:

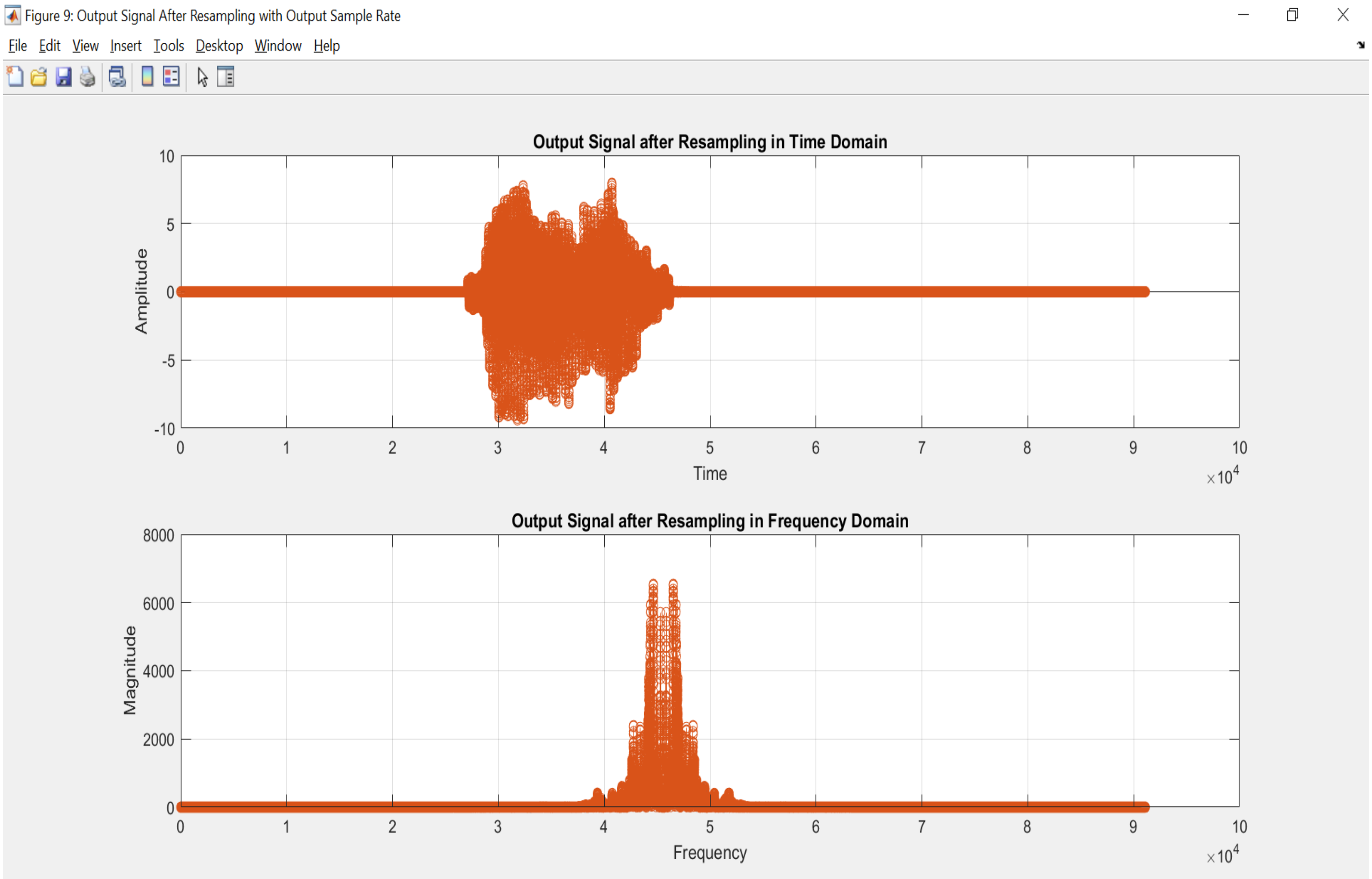


Figure 2.a.18

- Note that the output wav signal has been played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal.

➢ Output signal in case if doubling output sample rate:

- The user interface is as shown in the figure below *(Figure 2.a.19):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test1.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): FIR
-----------------------------------
Enter the output sample rate: 88200
-----------------------------------
Enter the gain for filter (1) in dB: 20
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 20
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 20
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.a.19

- Playing the output wav signal in case of doubling the output sample rate:

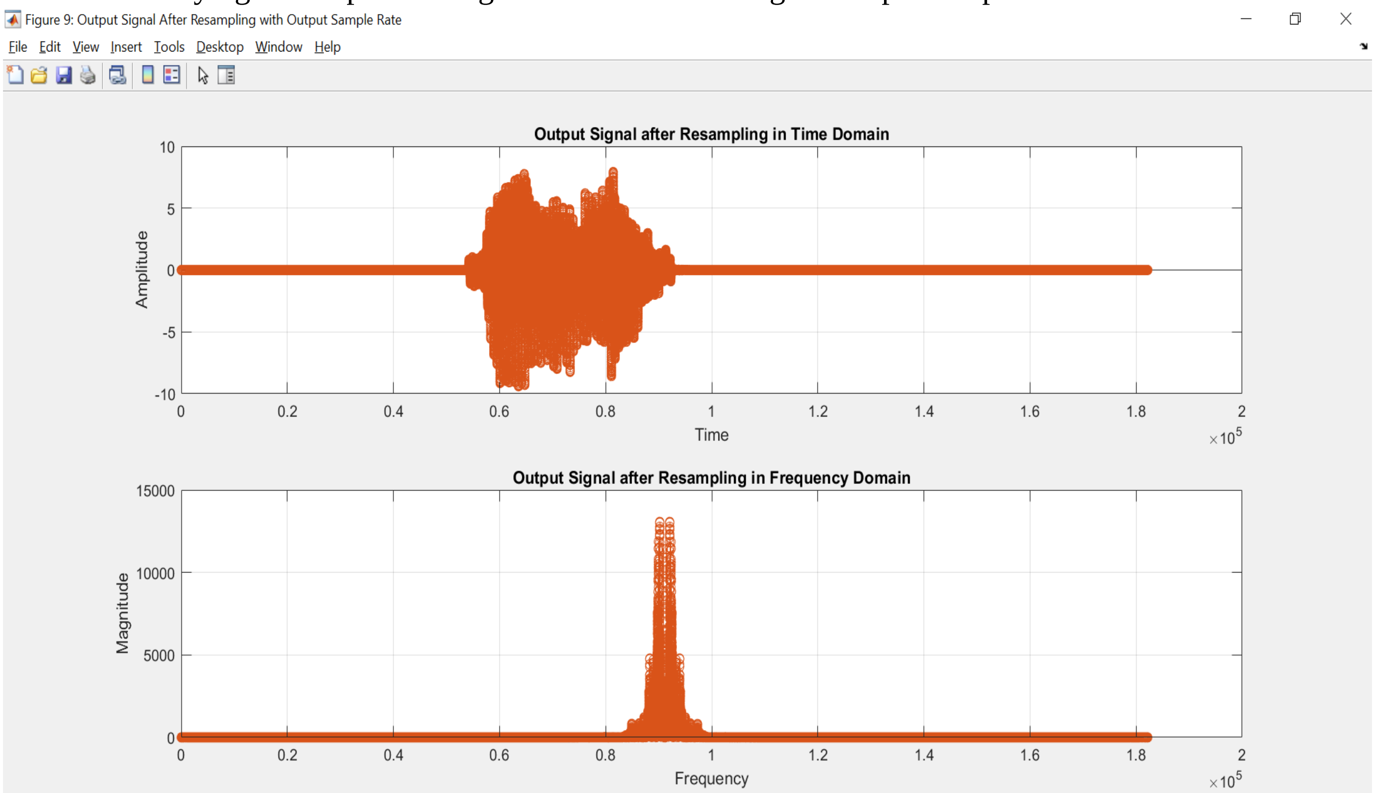


Figure 2.a.20

- Note that the output wav signal has been resampled with the double output sample rate and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with double output sample rate with those before resampling (Mentioned above).

- ➢ Output signal in case if decreasing output sample rate to half:
  - The user interface is as shown in the figure below *(Figure 2.a.21):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test1.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): FIR
-----------------------------------
Enter the output sample rate: 22050
-----------------------------------
Enter the gain for filter (1) in dB: 20
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 20
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 20
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.a.21

- Playing the output wav signal in case of decreasing the output sample rate to half:

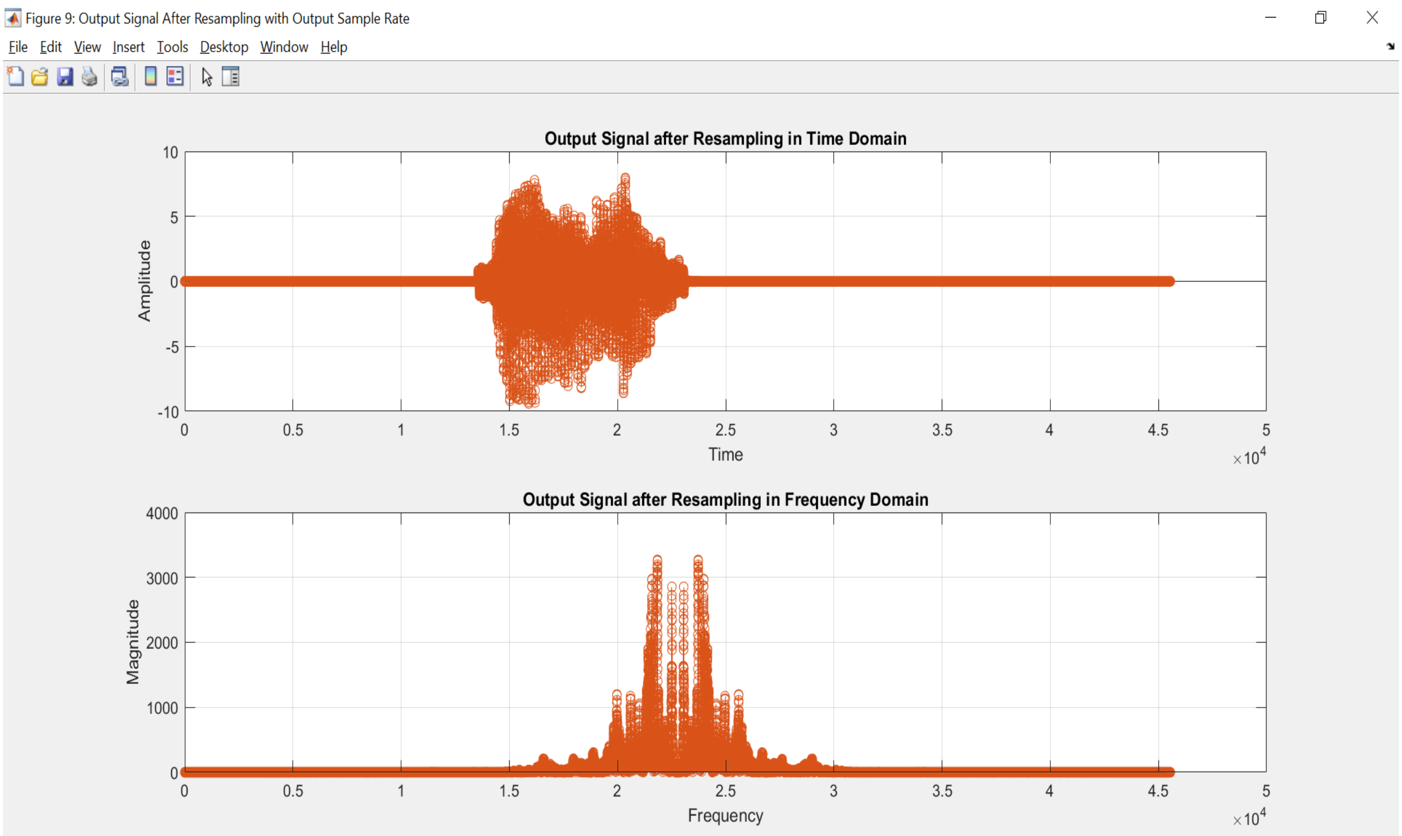


Figure 2.a.22

- Note that the output wav signal has been resampled with decreasing the output sample rate to half and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with decreasing the output sample rate to half with those before resampling (Mentioned above).

- Applying the test2 wav file to the filters of type FIR:
  - The user interface is as shown in the figure below *(Figure 2.a.23):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test2.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): FIR
-----------------------------------
Enter the output sample rate: 44100
-----------------------------------
Enter the gain for filter (1) in dB: 0
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 0
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 0
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.a.23

  - The output signals in time domain and frequency domain:

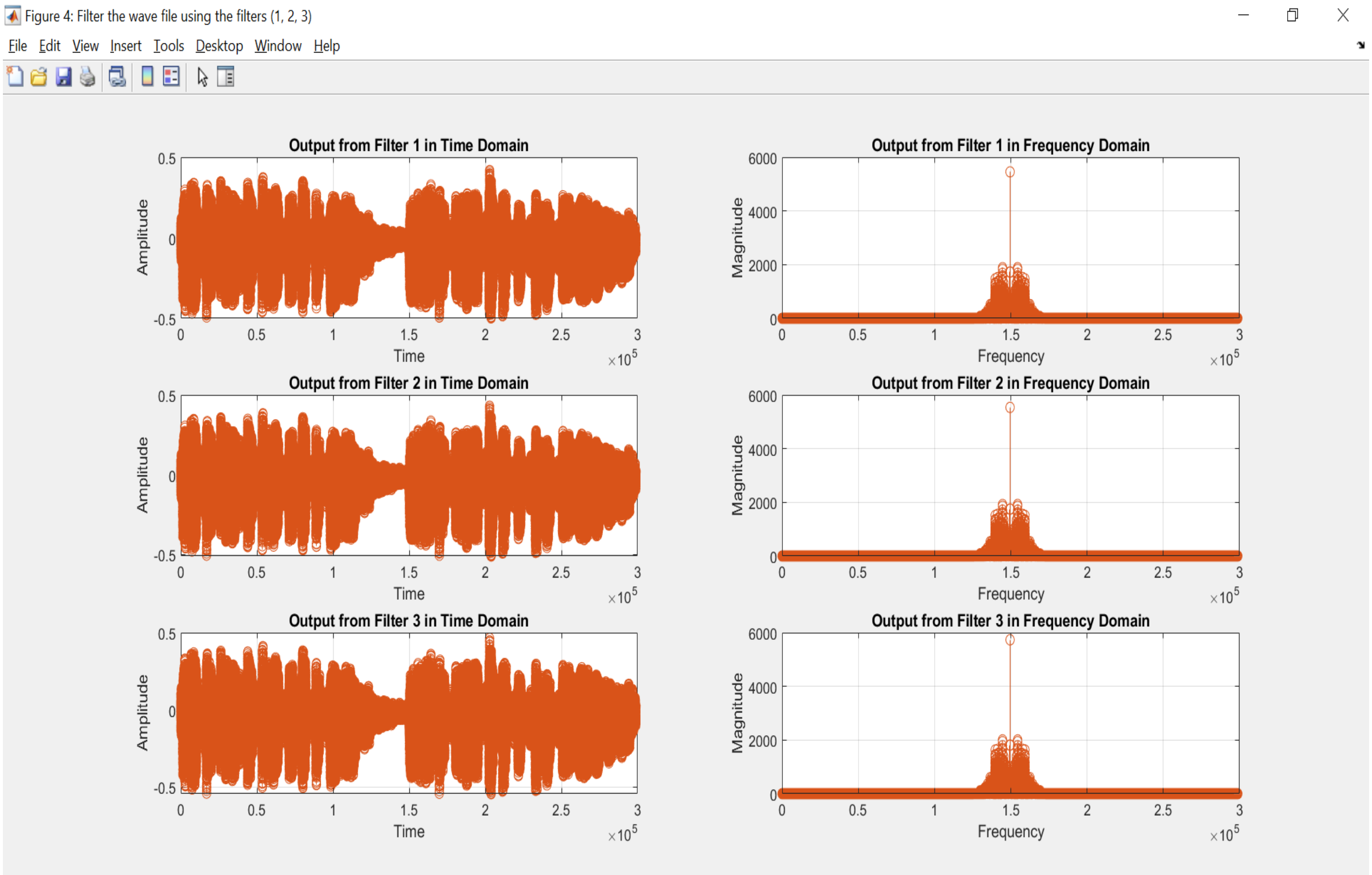


Figure 2.a.24

Fiaure 2.a.25

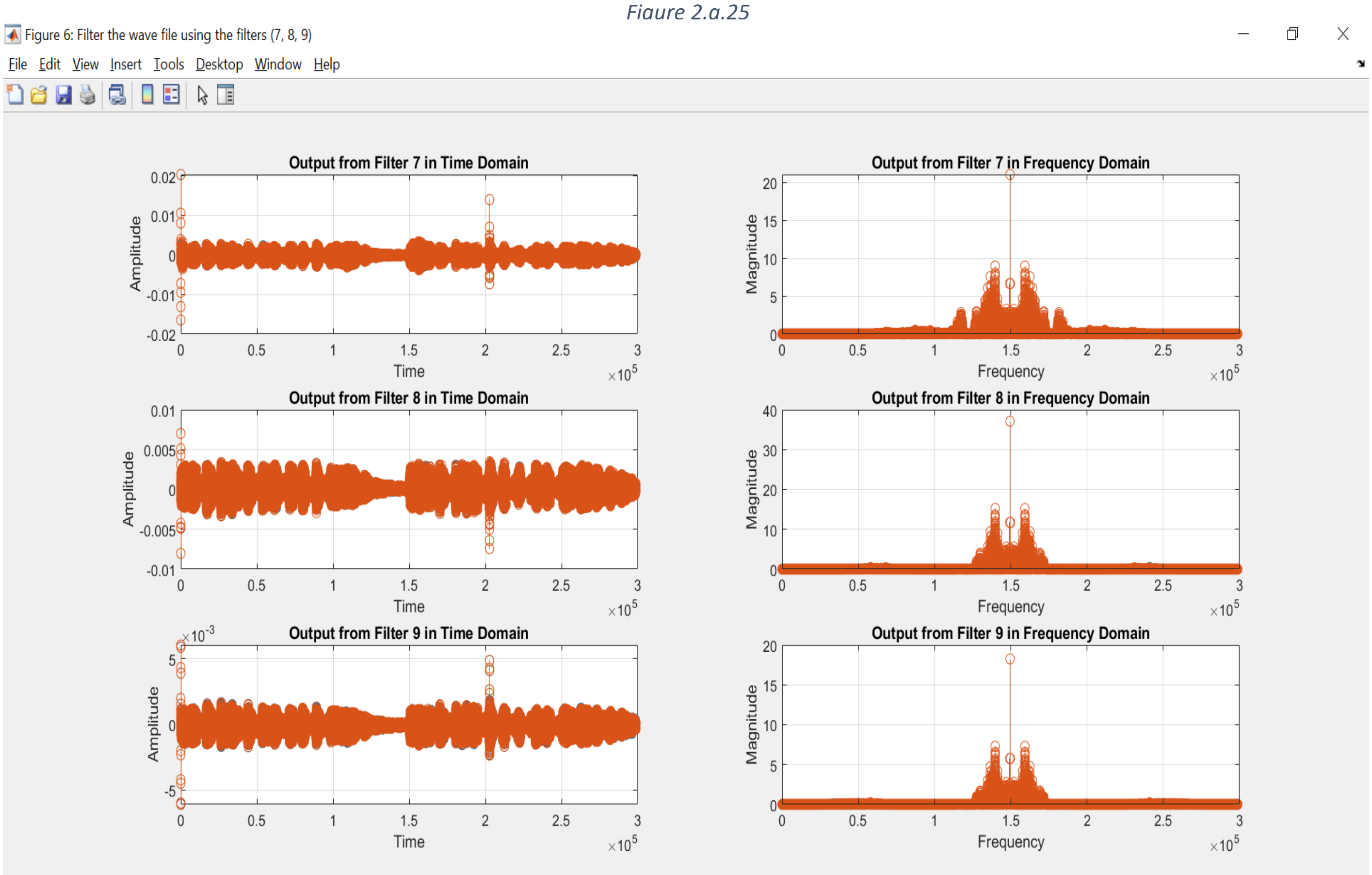


Figure 2.a.26

- Comparison between the Composite Signal amplified with gain (0 dB ➔ 1) with the Original Signal in Time Domain:

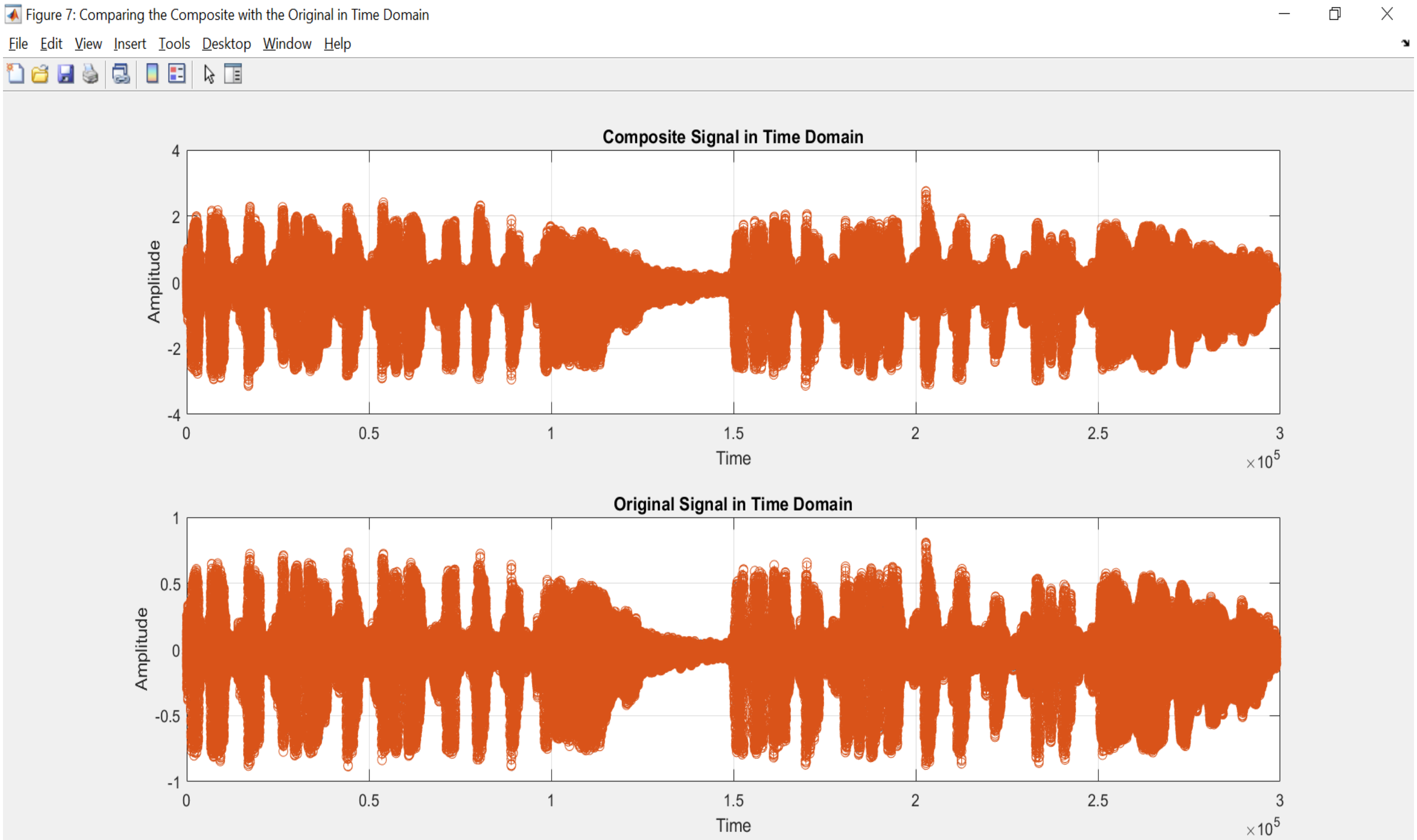


Figure 2.a.27

- Comparison between the Composite Signal amplified with gain (0 dB ➔ 1) with the Original Signal in Frequency Domain:

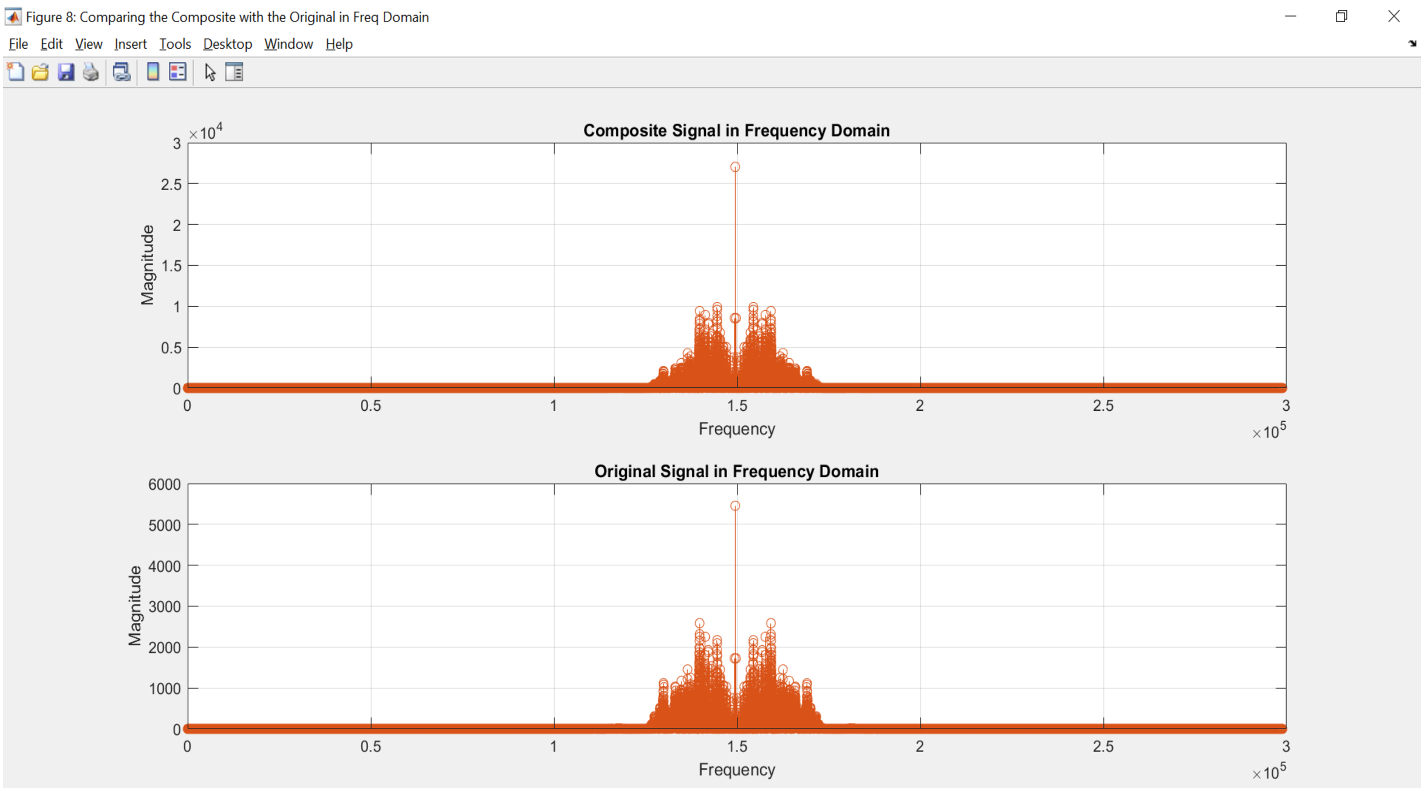


Figure 2.a.28

- Playing the output wav signal with the desired output sample rate:

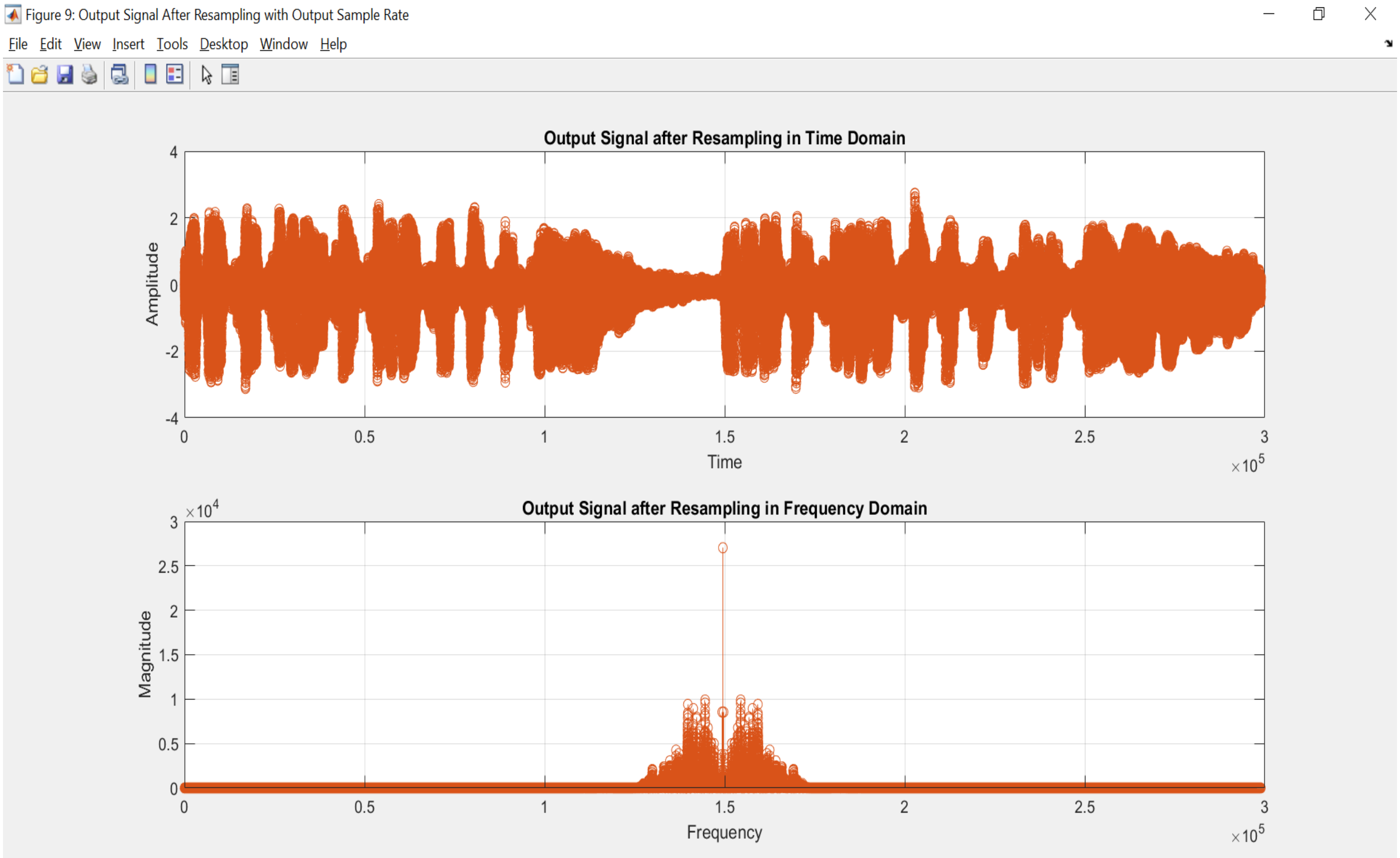


Figure 2.a.29

- Note that the output wav signal has been played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal.

➢ Output signal in case if doubling output sample rate:

- The user interface is as shown in the figure below *(Figure 2.a.30):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test2.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): FIR
-----------------------------------
Enter the output sample rate: 88200
-----------------------------------
Enter the gain for filter (1) in dB: 0
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 0
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 0
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.a.30

- Playing the output wav signal in case of doubling the output sample rate:

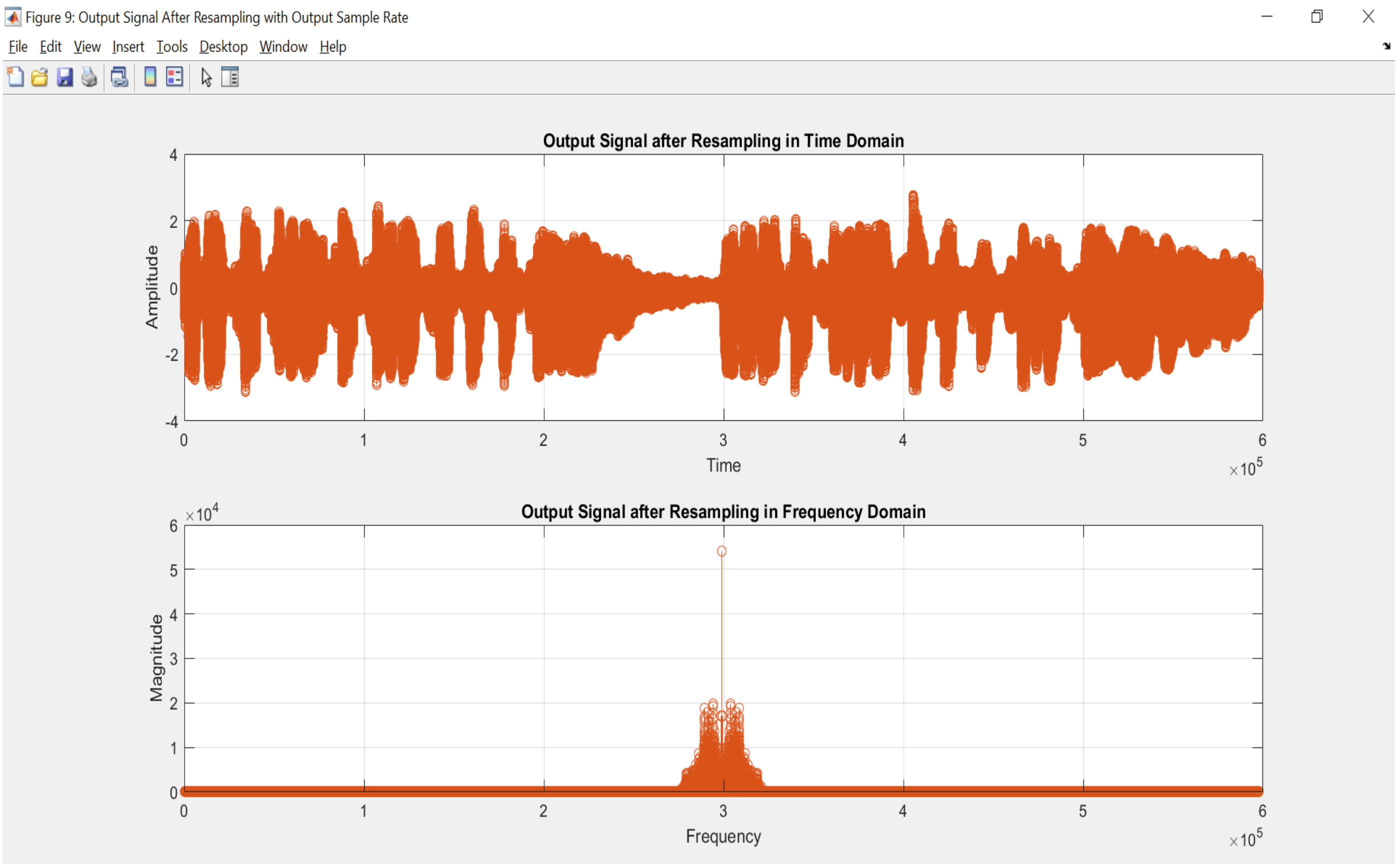


Figure 2.a.31

- Note that the output wav signal has been resampled with the double output sample rate and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with double output sample rate with those before resampling (Mentioned above).

- Output signal in case if decreasing output sample rate to half:
  - The user interface is as shown in the figure below *(Figure 2.a.32):*

```
Command Window
*****************WELCOME TO AUDIO EQUALIZER*****************
-----------------------------------
Enter the wave file name: test2.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): FIR
-----------------------------------
Enter the output sample rate: 22050
-----------------------------------
Enter the gain for filter (1) in dB: 0
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 0
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 0
-----------------------------------
*****************THANK YOU*****************
fx >>
```

Figure 2.a.32

  - Playing the output wav signal in case of decreasing the output sample rate to half:

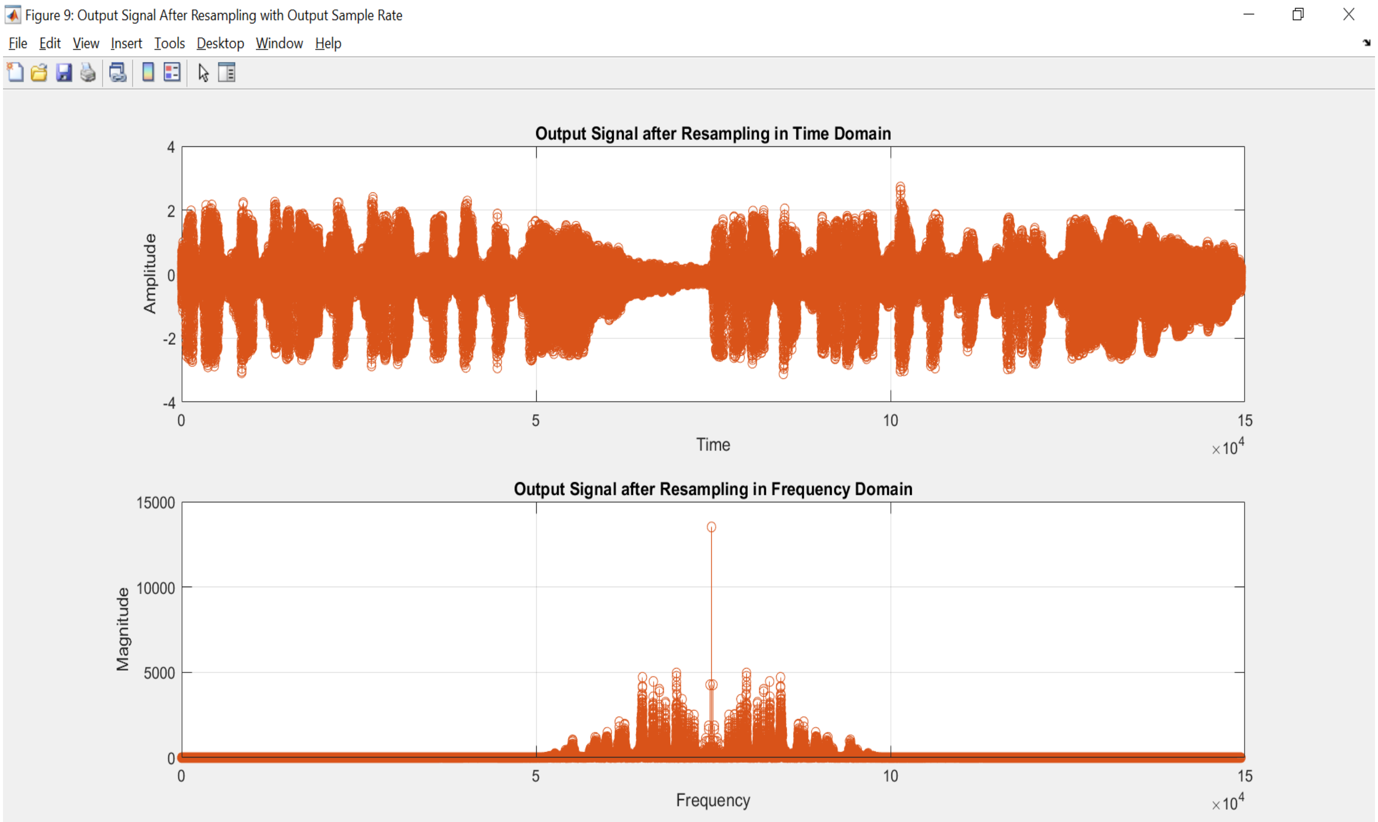


Figure 2.a.33

  - Note that the output wav signal has been resampled with decreasing the output sample rate to half and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with decreasing the output sample rate to half with those before resampling (Mentioned above).

➢ Applying the test2 wav file to the filters of type FIR:

- The user interface is as shown in the figure below *(Figure 2.a.34):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test2.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): FIR
-----------------------------------
Enter the output sample rate: 44100
-----------------------------------
Enter the gain for filter (1) in dB: 20
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 20
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 20
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.a.34

- The output signals in time domain and frequency domain:

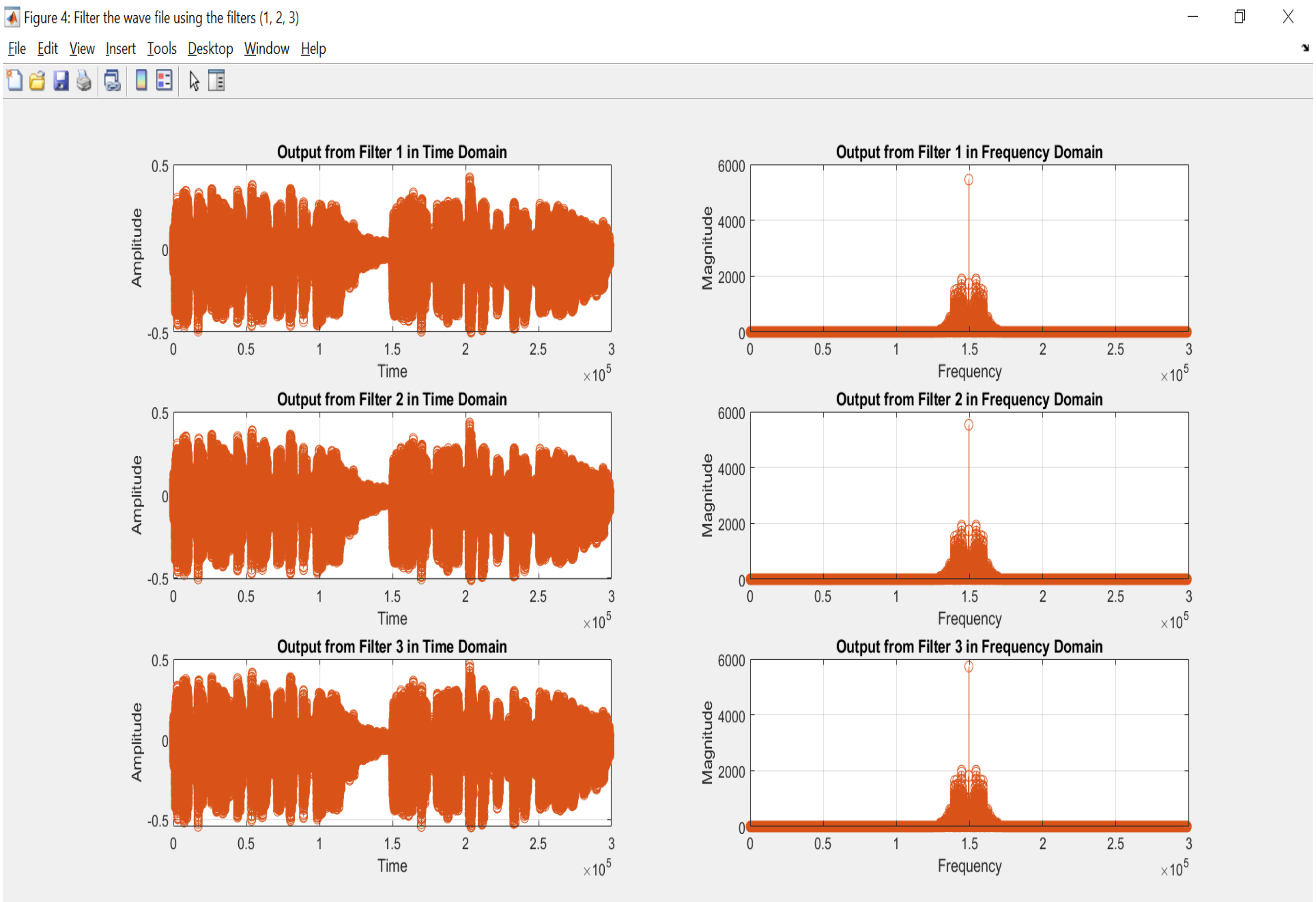


Figure 2.a.35

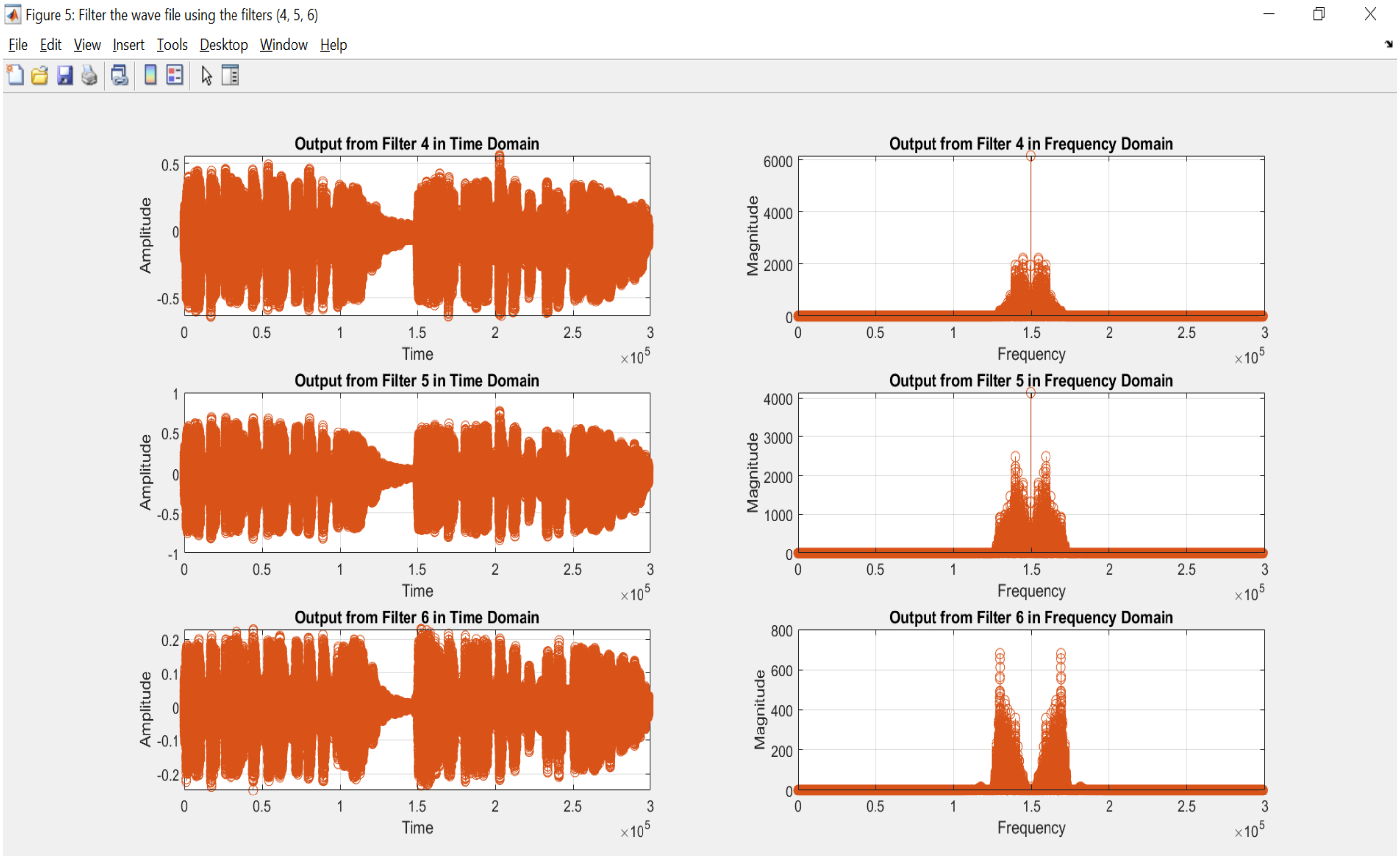


Figure 2.a.36

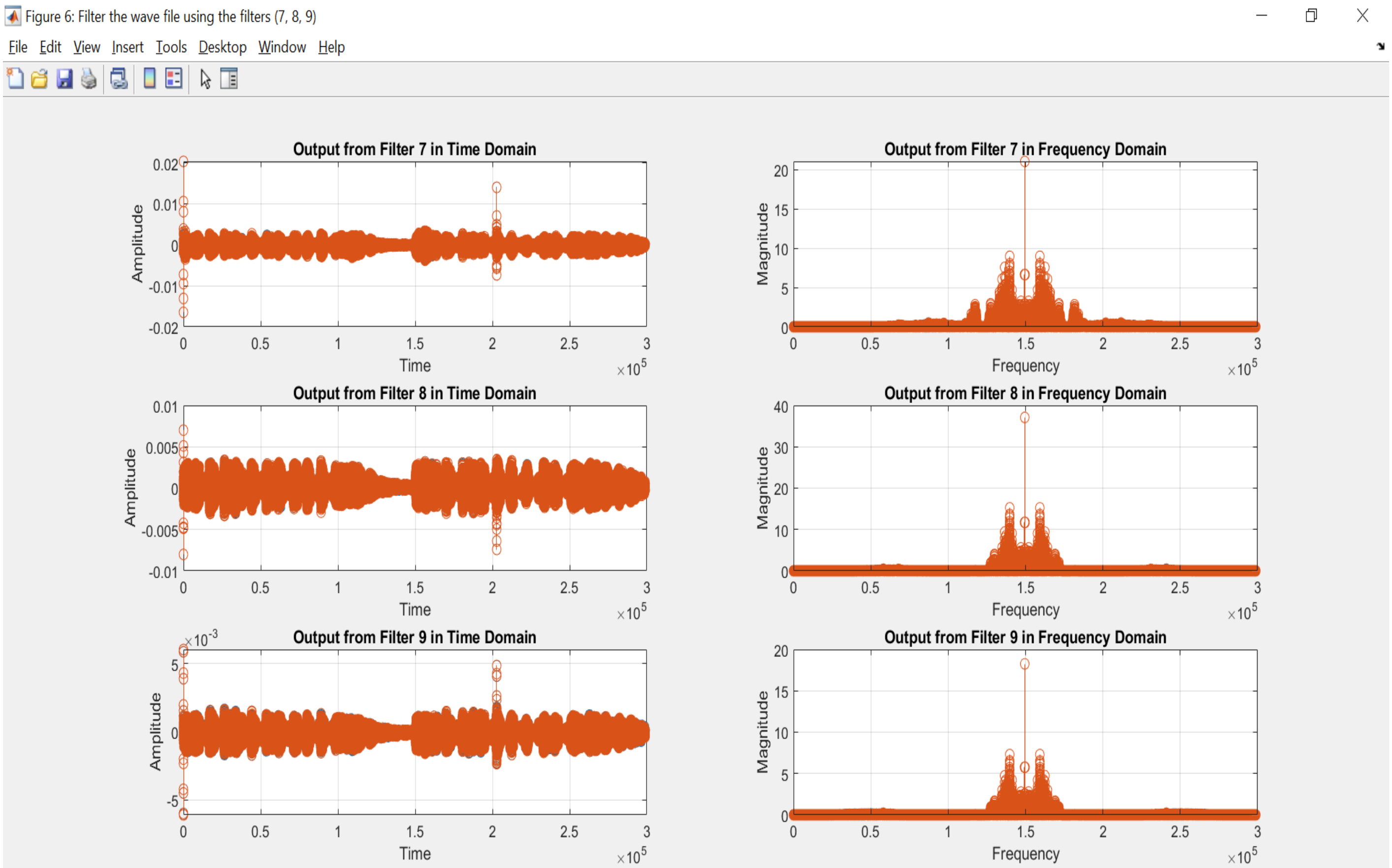


Figure 2.a.37

- Comparison between the Composite Signal amplified with the gains showed in the figure above *(Figure 2.a.34)* with the Original Signal in Time Domain:

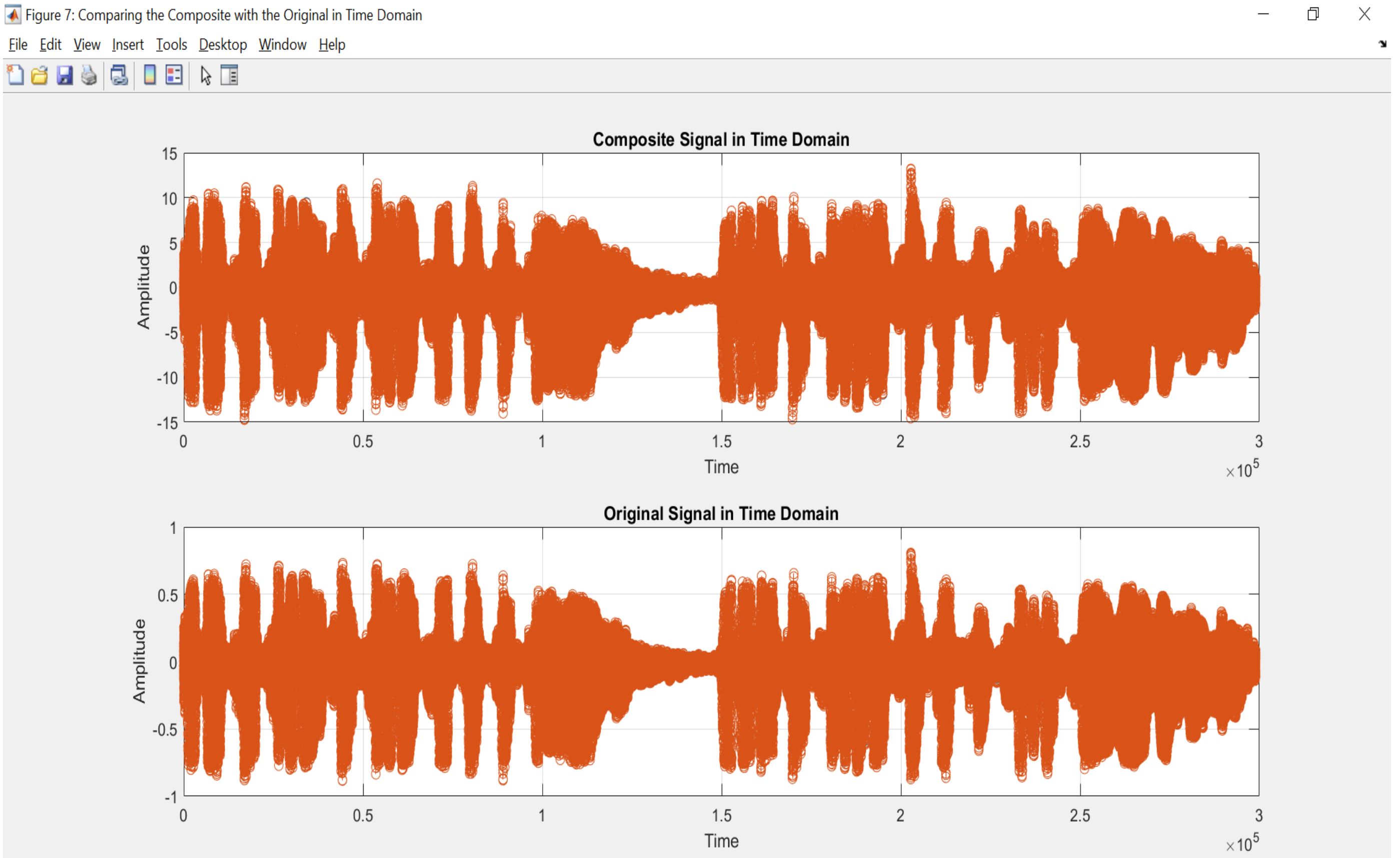


Figure 2.a.38

- Comparison between the Composite Signal amplified with the gains showed in the figure above *(Figure 2.a.34)* with the Original Signal in Frequency Domain:

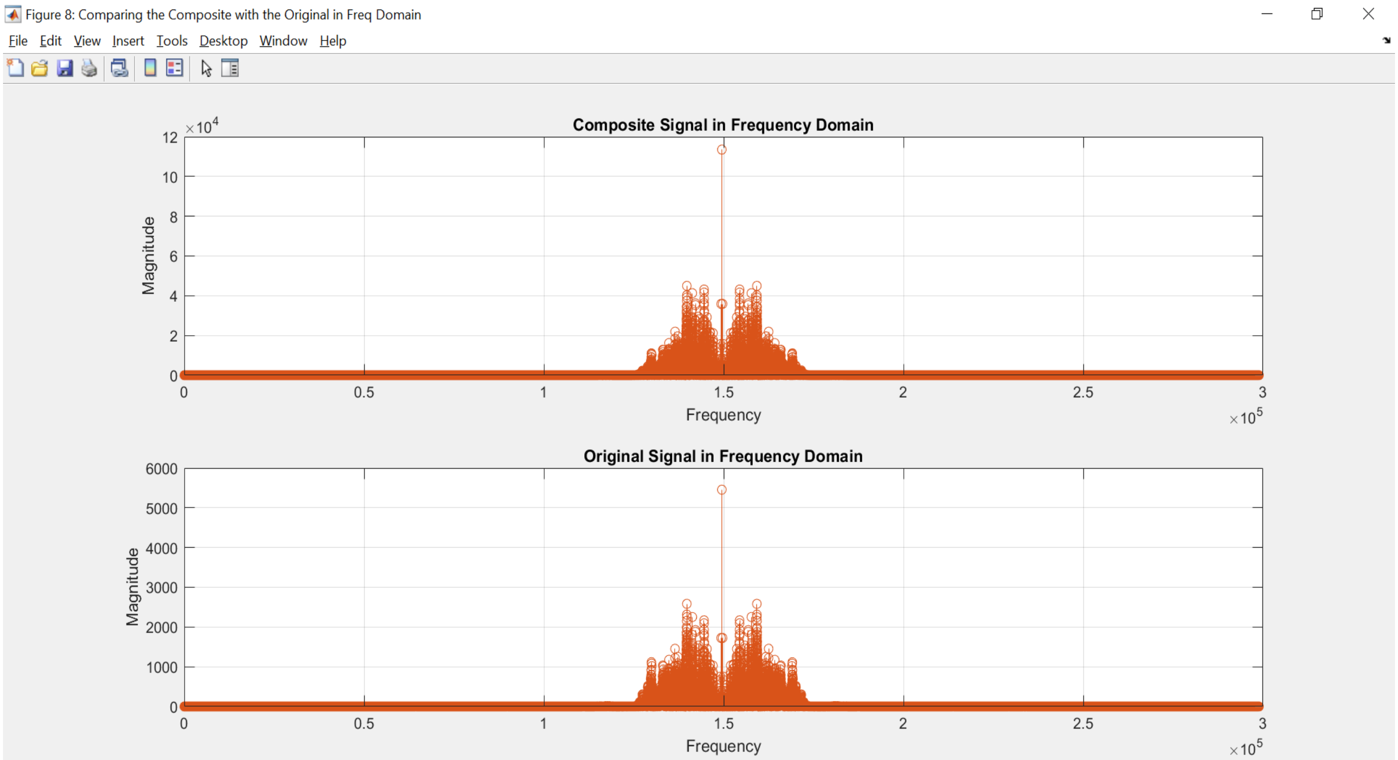


Figure 2.a.39

- Playing the output wav signal with the desired output sample rate:

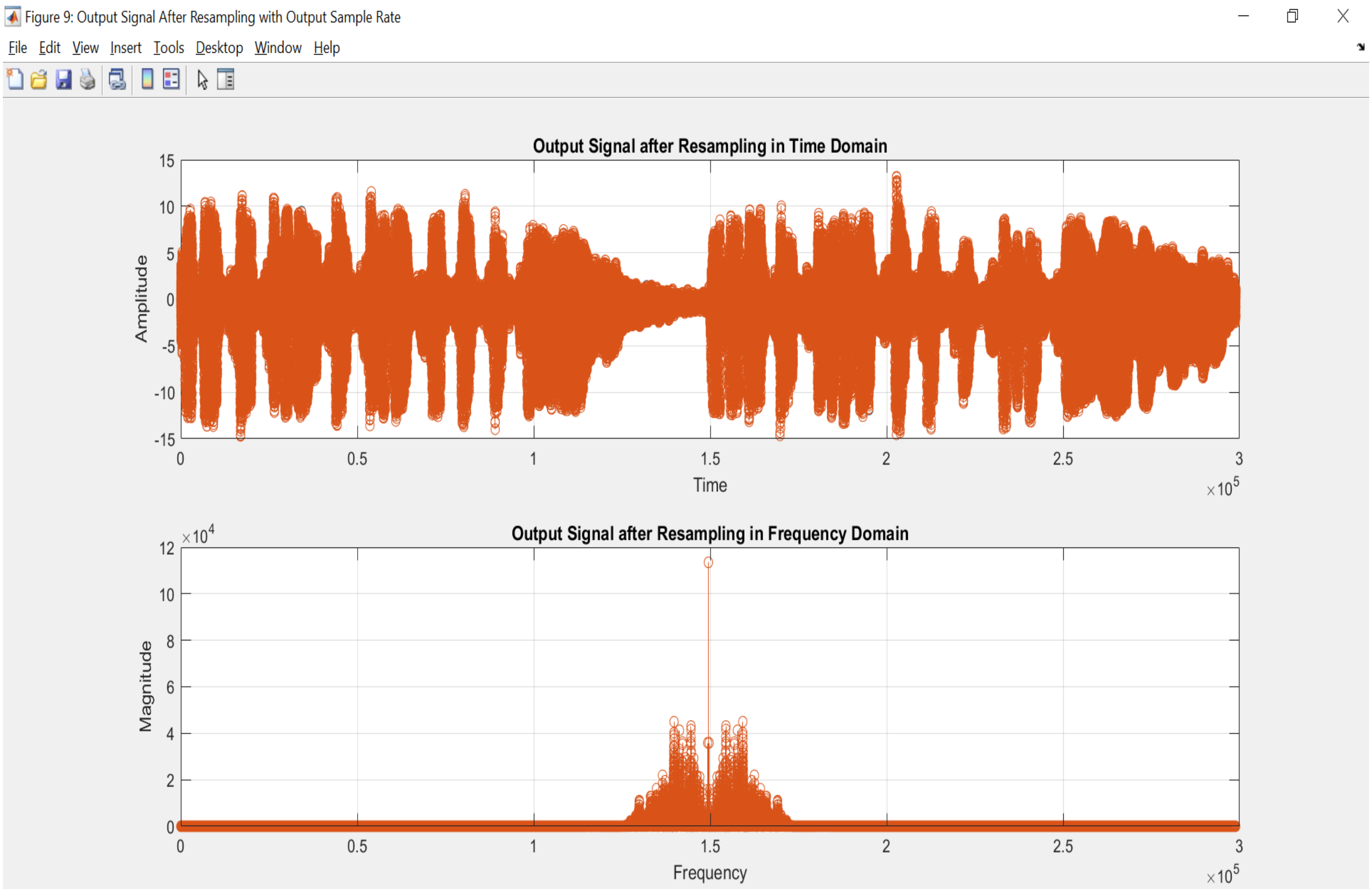


Figure 2.a.40

- Note that the output wav signal has been played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal.

➢ Output signal in case if doubling output sample rate:

- The user interface is as shown in the figure below *(Figure 2.a.41)*:

Command Window

```
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test2.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): FIR
-----------------------------------
Enter the output sample rate: 88200
-----------------------------------
Enter the gain for filter (1) in dB: 20
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 20
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 20
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.a.41

- Playing the output wav signal in case of doubling the output sample rate:

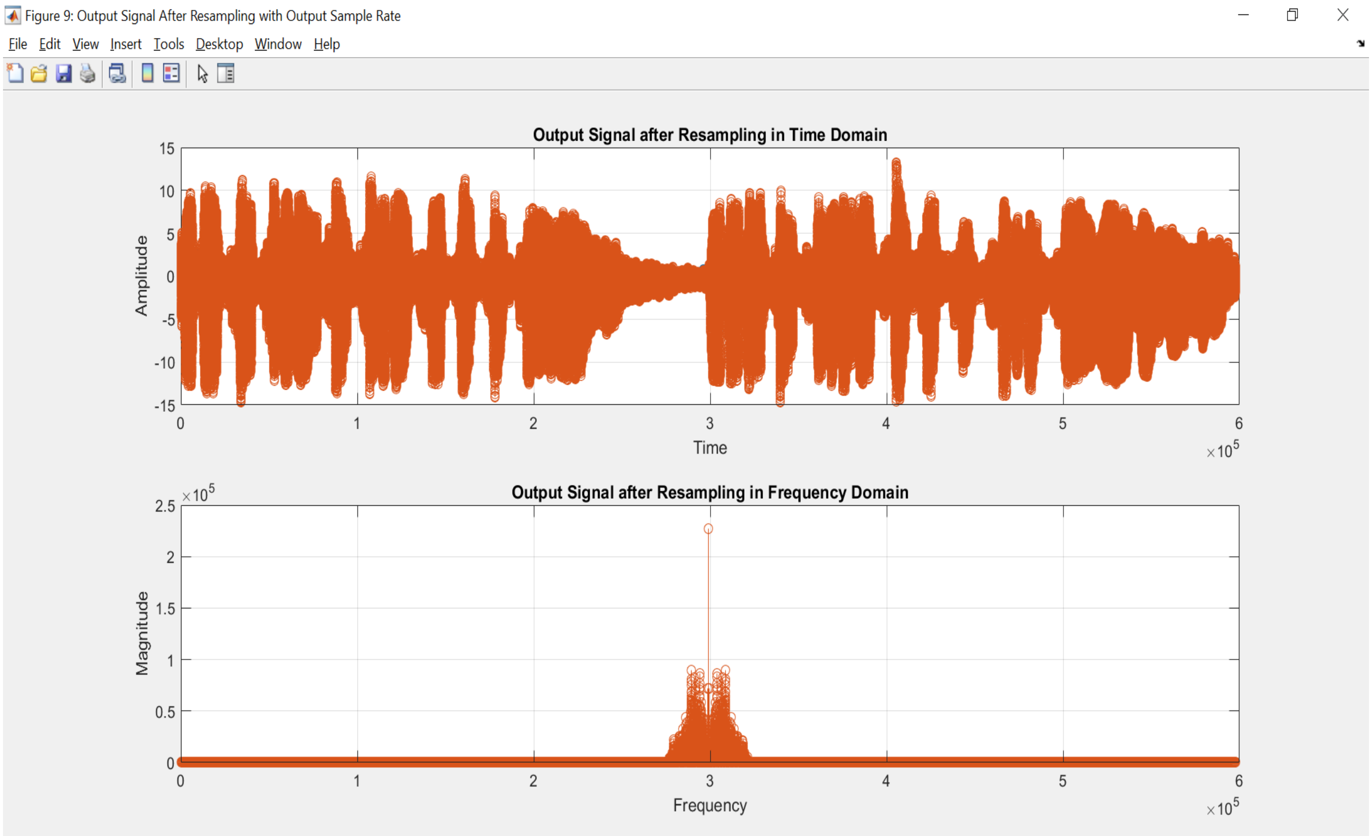


Figure 2.a.42

- Note that the output wav signal has been resampled with the double output sample rate and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with double output sample rate with those before resampling (Mentioned above).

➢ Output signal in case if decreasing output sample rate to half:

- The user interface is as shown in the figure below *(Figure 2.a.43):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test2.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): FIR
-----------------------------------
Enter the output sample rate: 22050
-----------------------------------
Enter the gain for filter (1) in dB: 20
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 20
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 20
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.a.43

- Playing the output wav signal in case of decreasing the output sample rate to half:

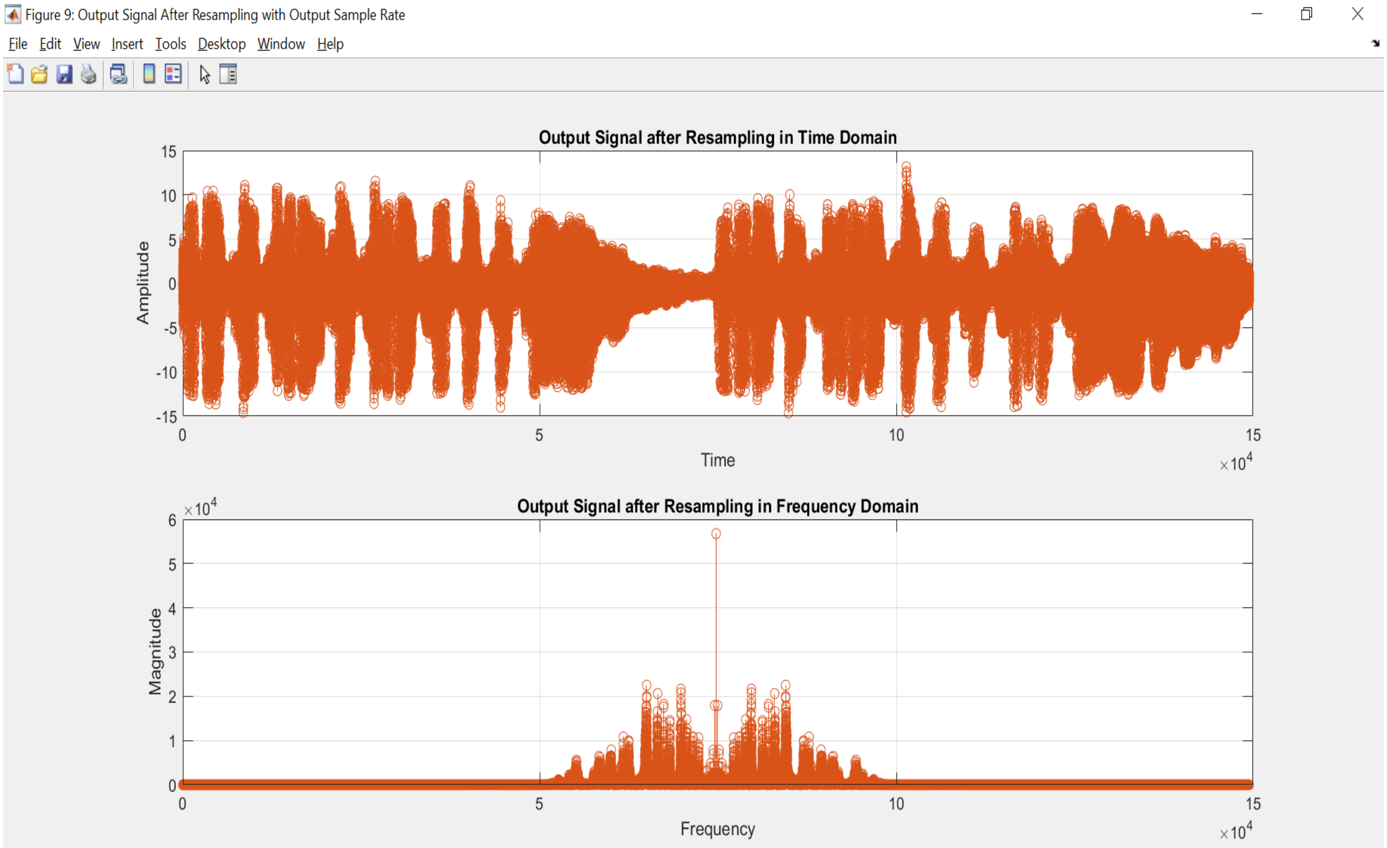


Figure 2.a.44

- Note that the output wav signal has been resampled with decreasing the output sample rate to half and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with decreasing the output sample rate to half with those before resampling (Mentioned above).

➢ Applying the test3 wav file to the filters of type FIR:

- The user interface is as shown in the figure below *(Figure 2.a.45):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
----------------------------------
Enter the wave file name: test3.wav
----------------------------------
Enter the type of filters used (IIR-FIR): FIR
----------------------------------
Enter the output sample rate: 44100
----------------------------------
Enter the gain for filter (1) in dB: 0
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 0
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 0
----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.a.45

- The output signals in time domain and frequency domain:

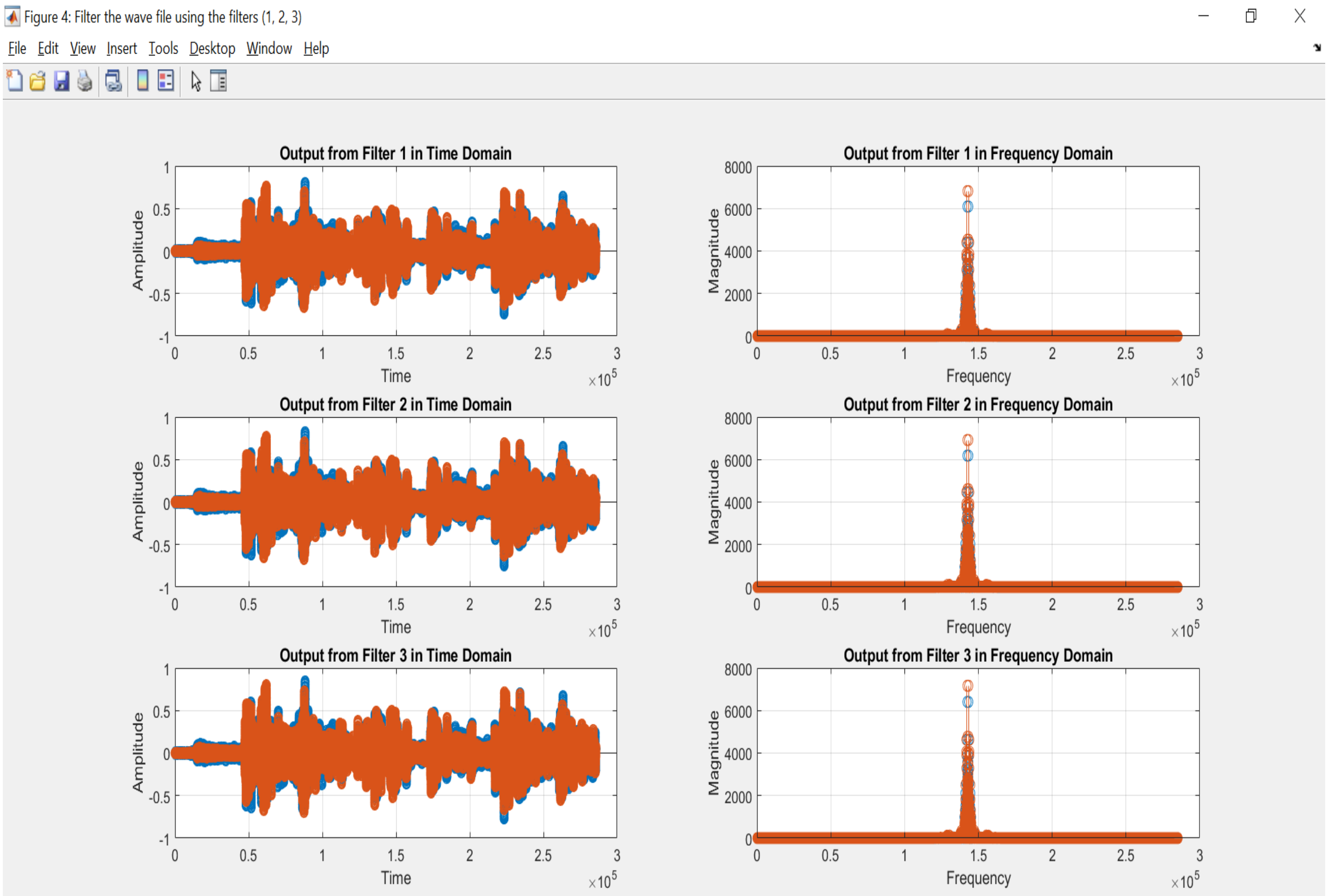


Figure 2.a.46

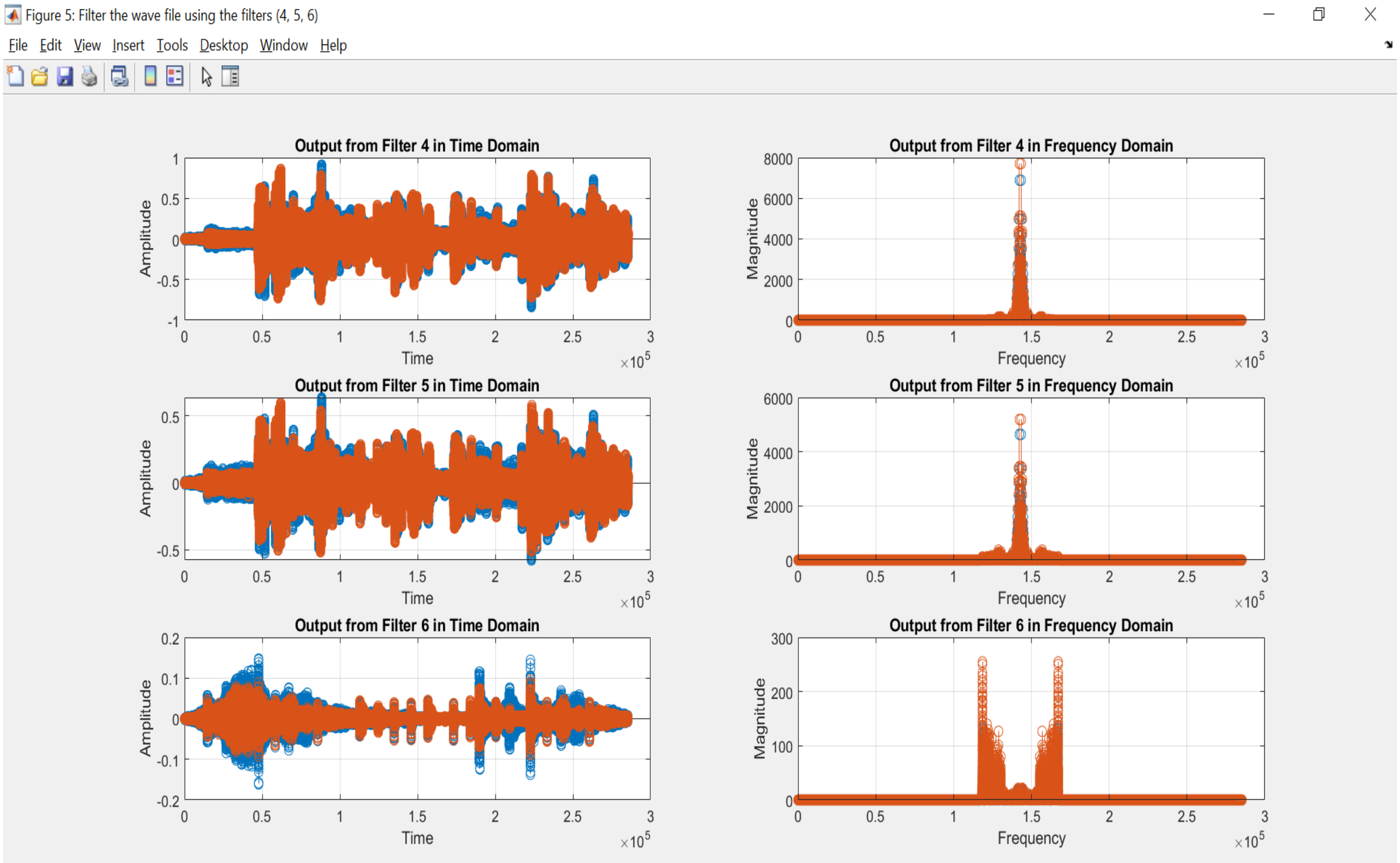


Figure 2.a.47

Figure 2.a.48

- Comparison between the Composite Signal amplified with gain (0 dB ➔ 1) with the Original Signal in Time Domain:

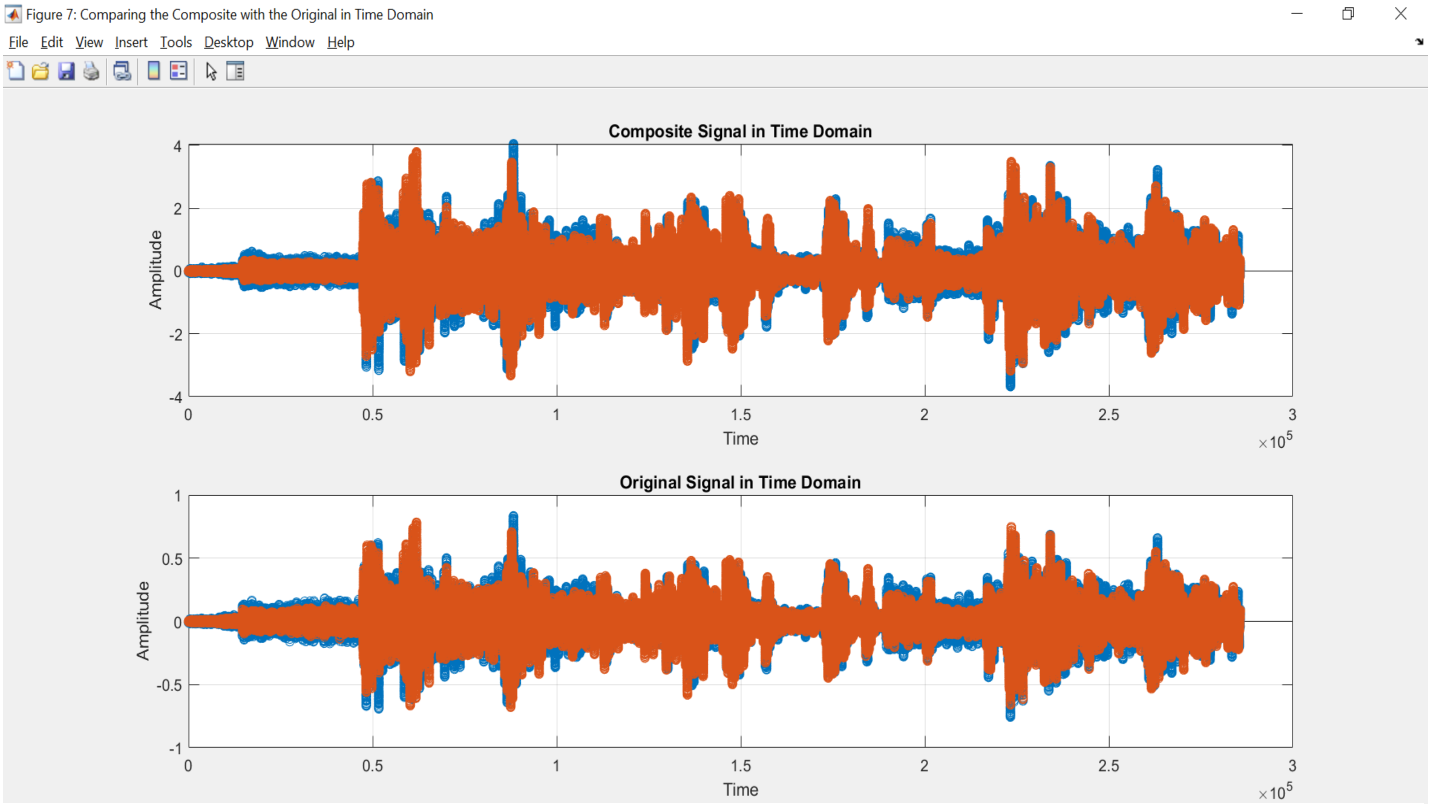


Figure 2.a.49

- Comparison between the Composite Signal amplified with gain (0 dB ➔ 1) with the Original Signal in Frequency Domain:

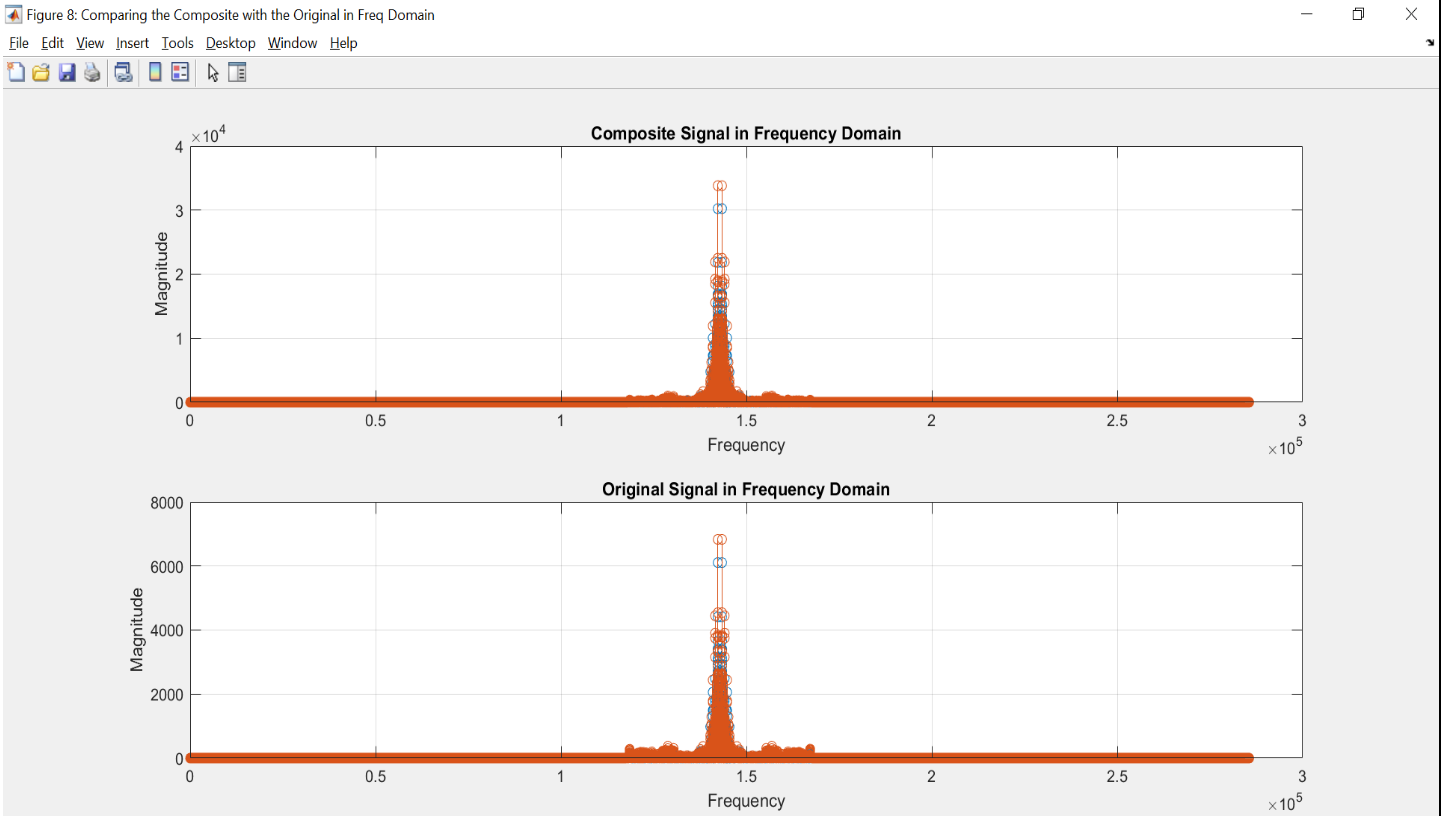


Figure 2.a.50

- Playing the output wav signal with the desired output sample rate:

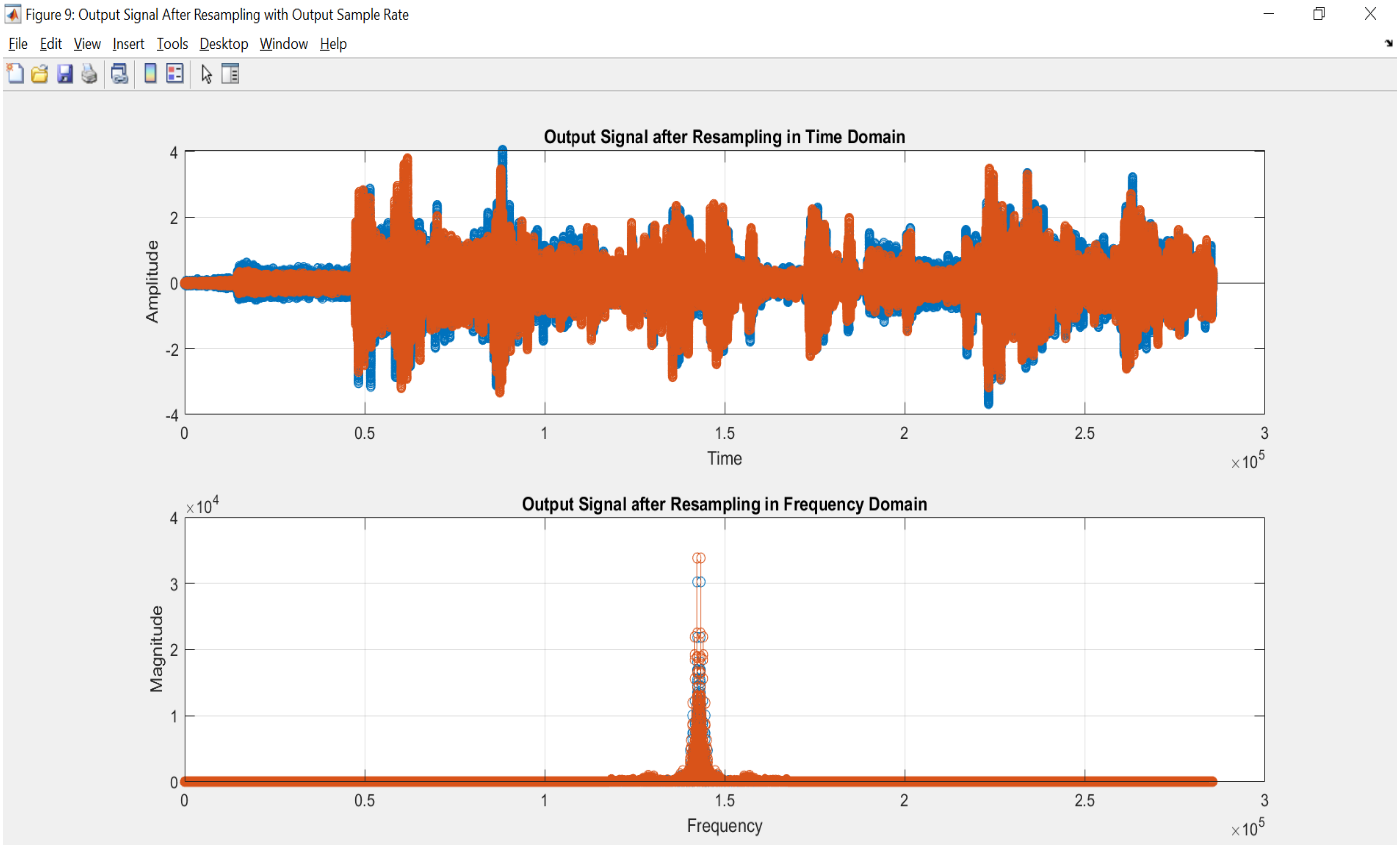


Figure 2.a.51

- Note that the output wav signal has been played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal.

➢ Output signal in case if doubling output sample rate:

- The user interface is as shown in the figure below *(Figure 2.a.52):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test3.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): FIR
-----------------------------------
Enter the output sample rate: 88200
-----------------------------------
Enter the gain for filter (1) in dB: 0
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 0
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 0
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.a.52

- Playing the output wav signal in case of doubling the output sample rate:

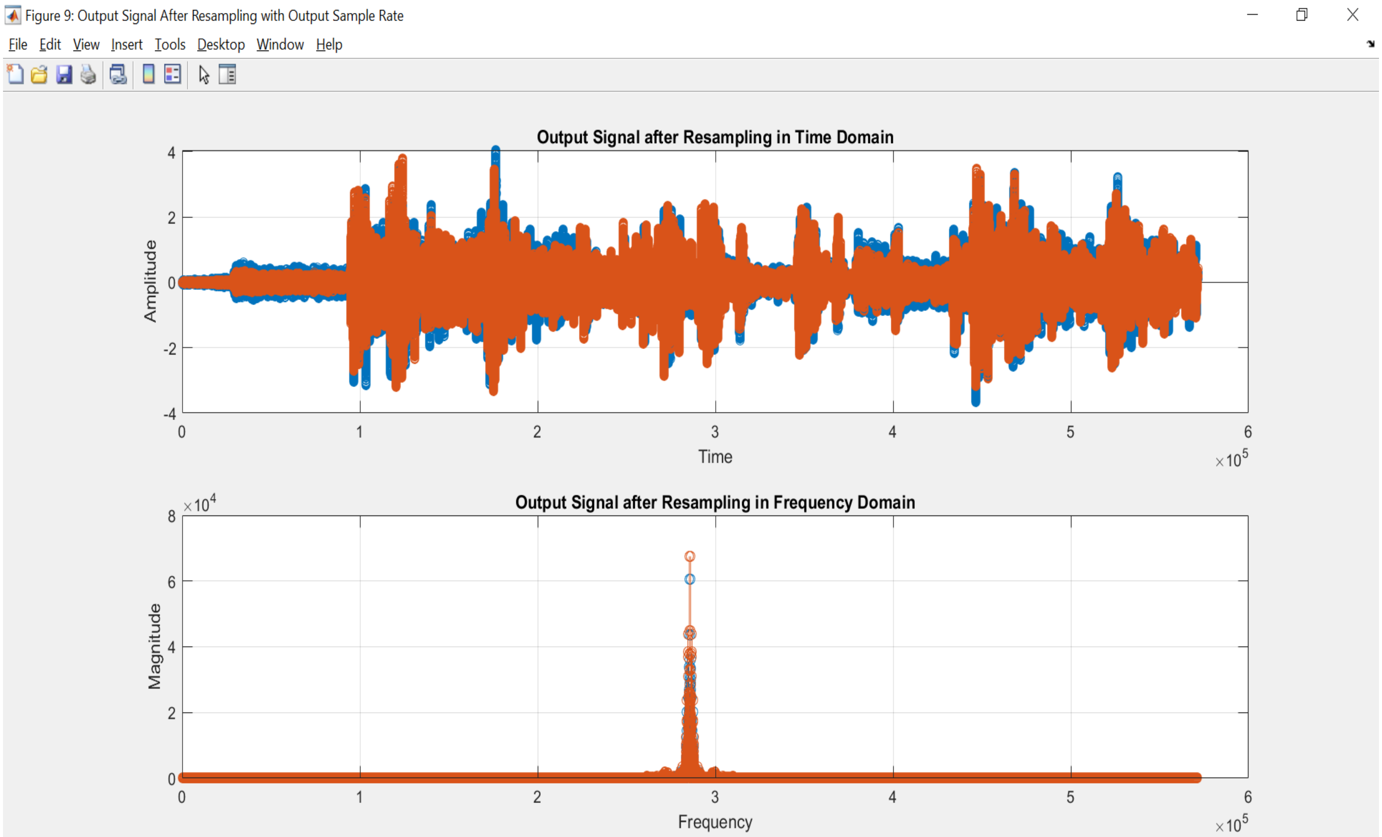


Figure 2.a.53

- Note that the output wav signal has been resampled with the double output sample rate and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with double output sample rate with those before resampling (Mentioned above).

➢ Output signal in case if decreasing output sample rate to half:

- The user interface is as shown in the figure below *(Figure 2.a.54):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test3.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): FIR
-----------------------------------
Enter the output sample rate: 22050
-----------------------------------
Enter the gain for filter (1) in dB: 0
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 0
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 0
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.a.54

- Playing the output wav signal in case of decreasing the output sample rate to half:

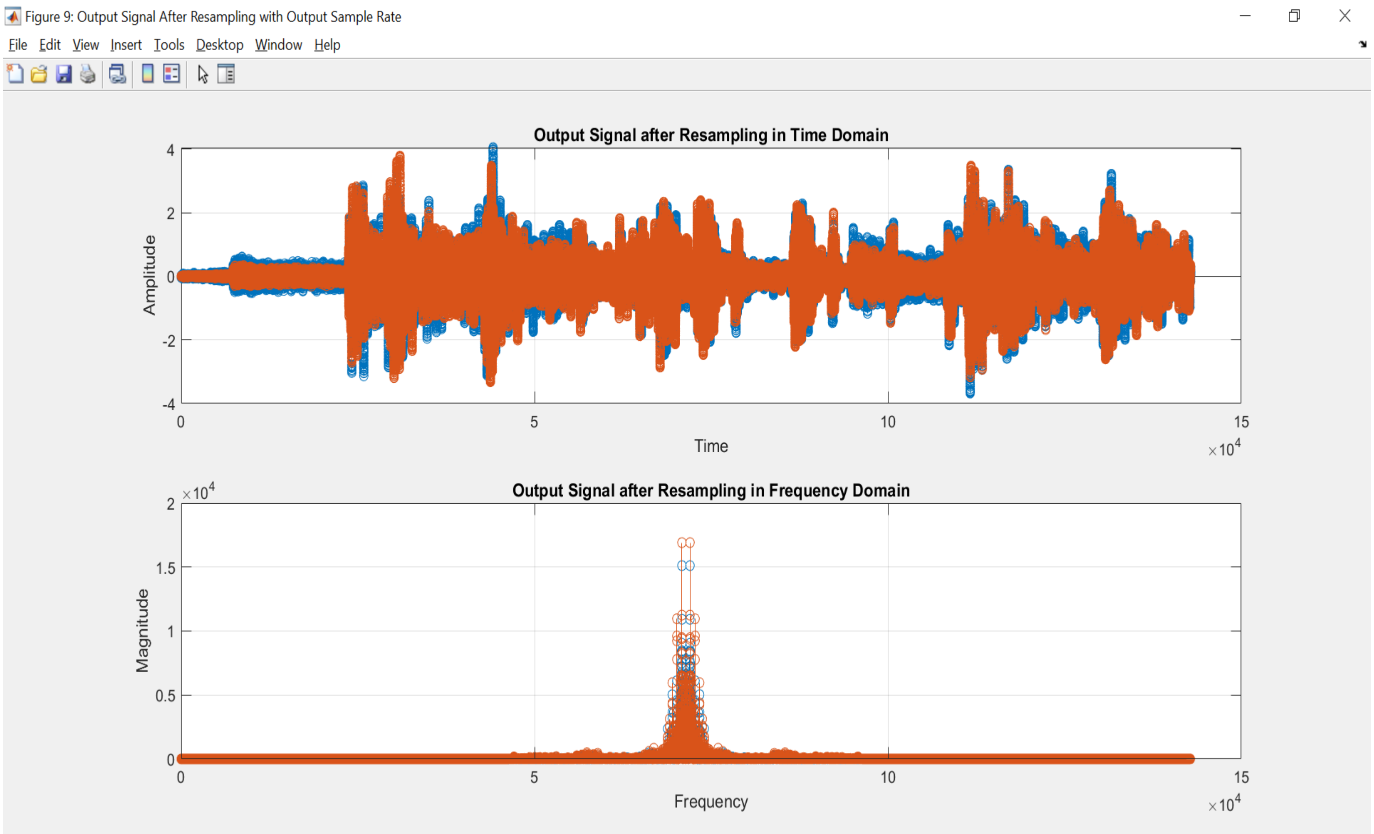


Figure 2.a.55

- Note that the output wav signal has been resampled with decreasing the output sample rate to half and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with decreasing the output sample rate to half with those before resampling (Mentioned above).

➢ Applying the test3 wav file to the filters of type FIR:

- The user interface is as shown in the figure below *(Figure 2.a.56):*

Command Window

```
****************WELCOME TO AUDIO EQUALIZER****************
----------------------------------
Enter the wave file name: test3.wav
----------------------------------
Enter the type of filters used (IIR-FIR): FIR
----------------------------------
Enter the output sample rate: 44100
----------------------------------
Enter the gain for filter (1) in dB: 20
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 20
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 20
----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.a.56

- The output signals in time domain and frequency domain:

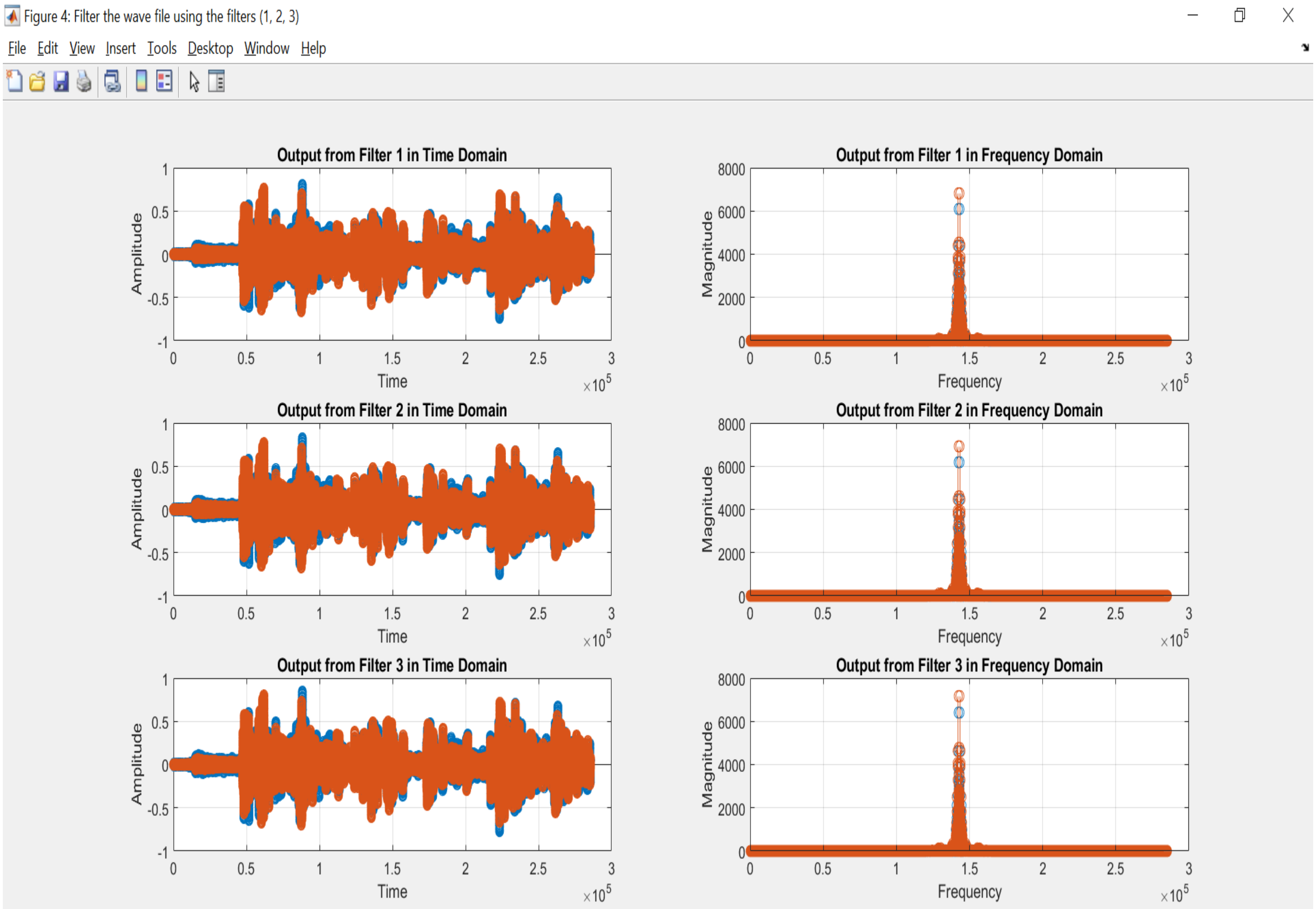


Figure 2.a.57

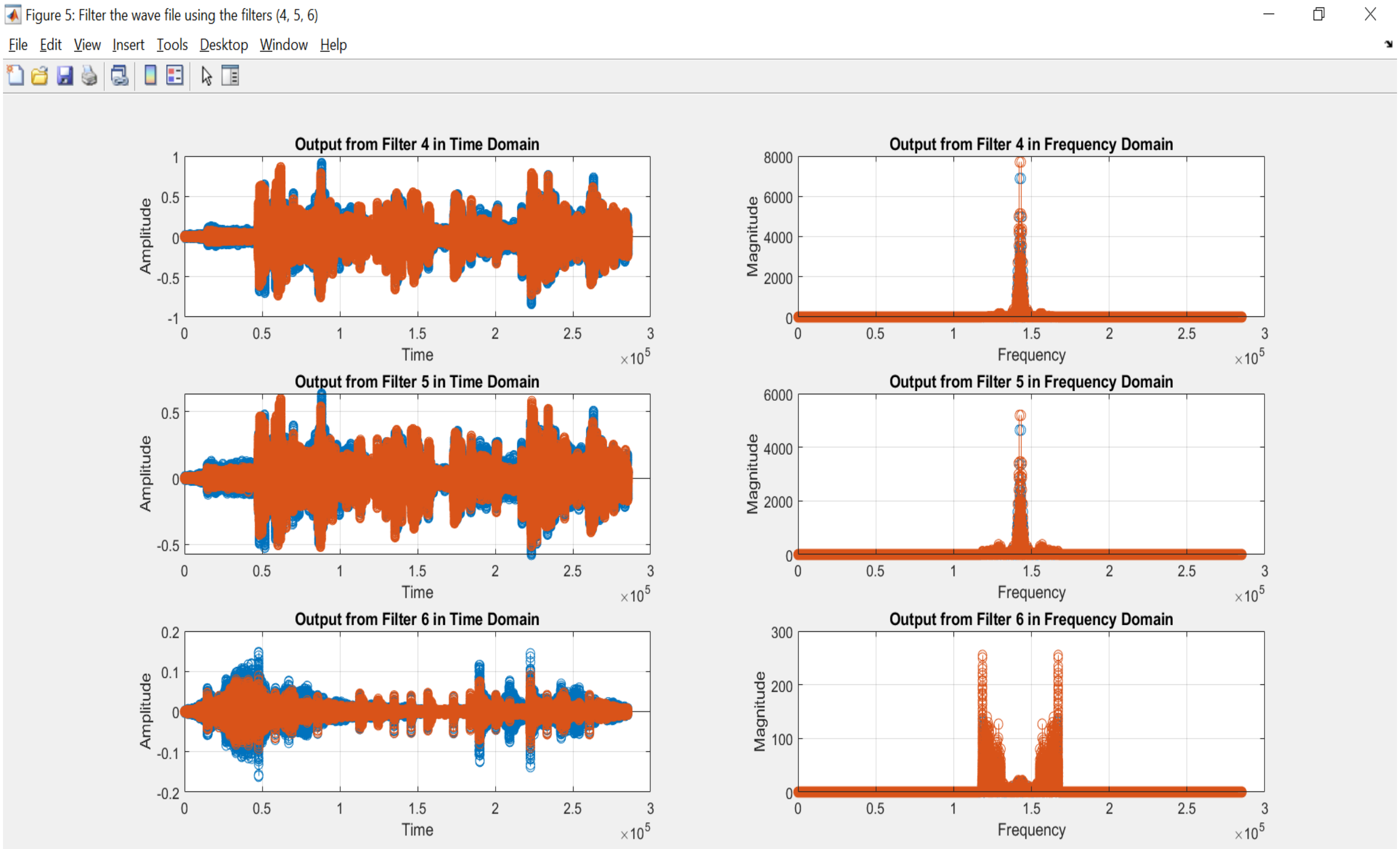


Figure 2.a.58

Figure 2.a.59

- Comparison between the Composite Signal amplified with the gains showed in the figure above *(Figure 2.a.56)* with the Original Signal in Time Domain:

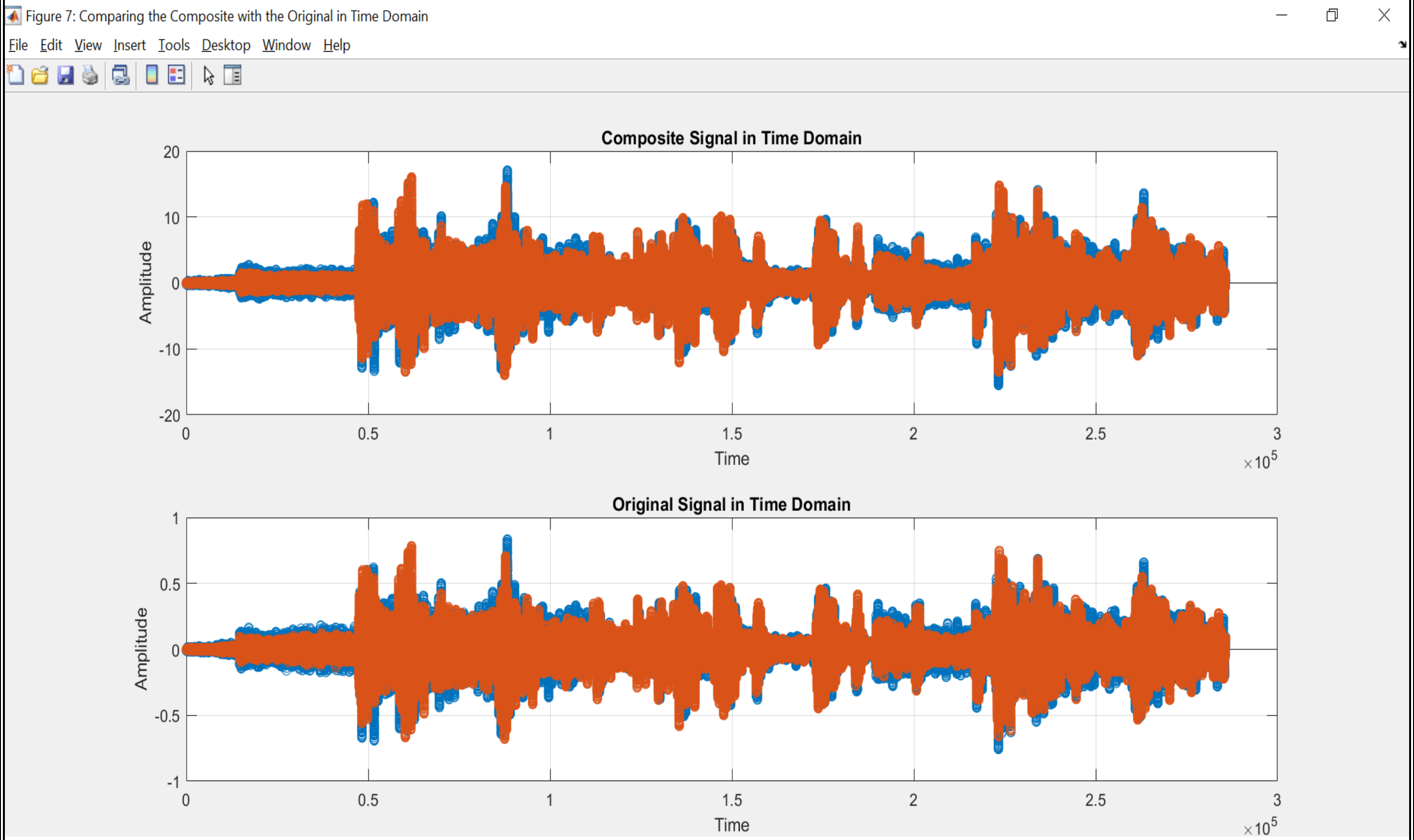


Figure 2.a.60

- Comparison between the Composite Signal amplified with the gains showed in the figure above *(Figure 2.a.56)* with the Original Signal in Frequency Domain:

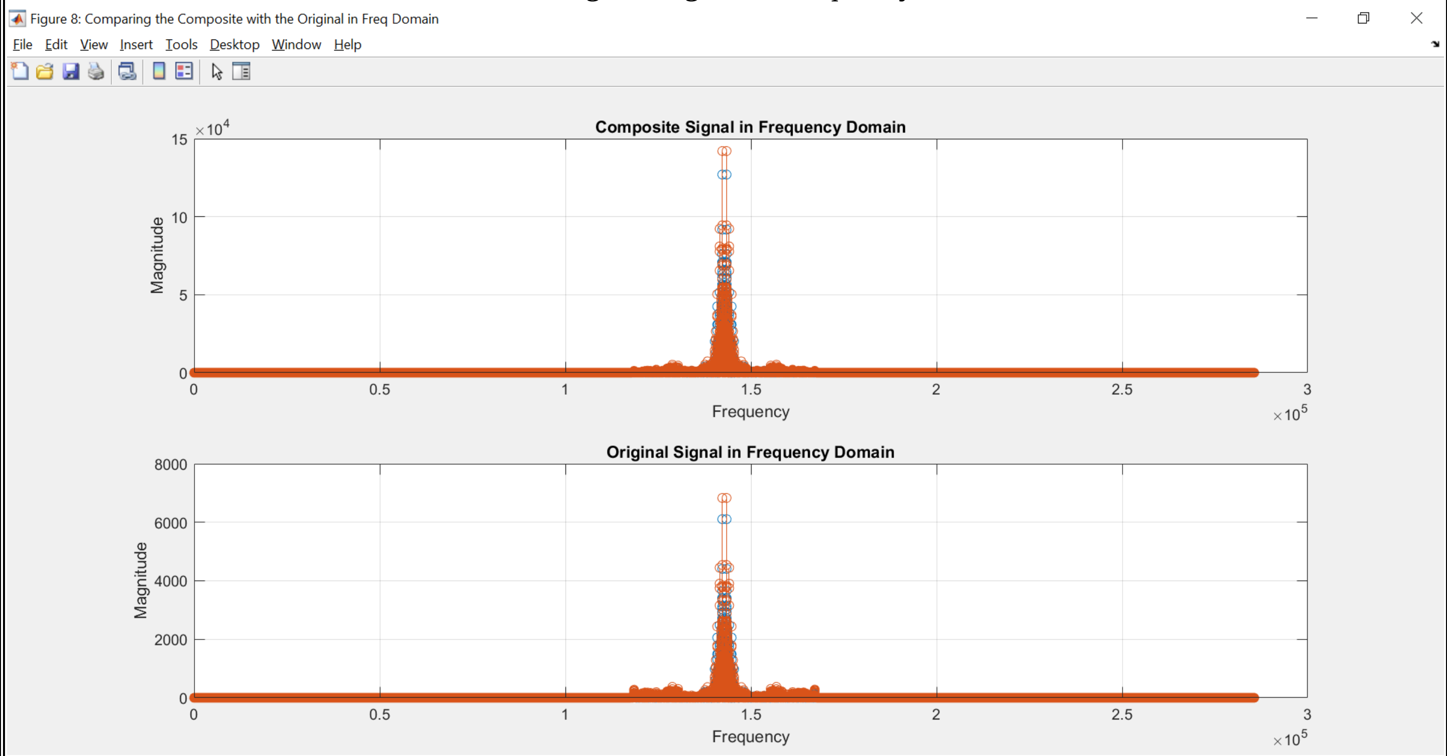


Figure 2.a.61

- Playing the output wav signal with the desired output sample rate:

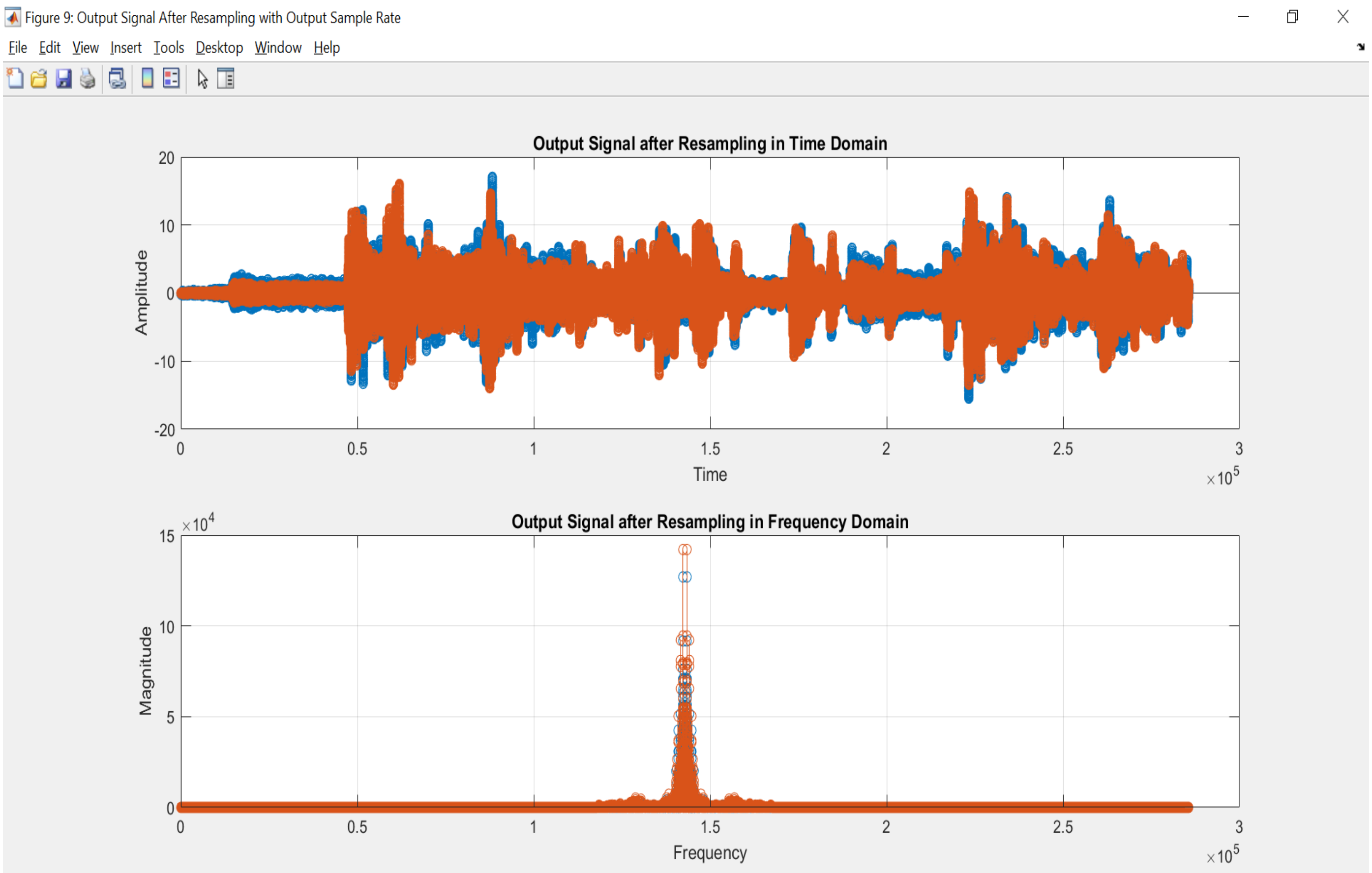


Figure 2.a.62

- Note that the output wav signal has been played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal.

- Output signal in case if doubling output sample rate:
  - The user interface is as shown in the figure below *(Figure 2.a.63):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test3.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): FIR
-----------------------------------
Enter the output sample rate: 88200
-----------------------------------
Enter the gain for filter (1) in dB: 20
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 20
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 20
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.a.63

  - Playing the output wav signal in case of doubling the output sample rate:

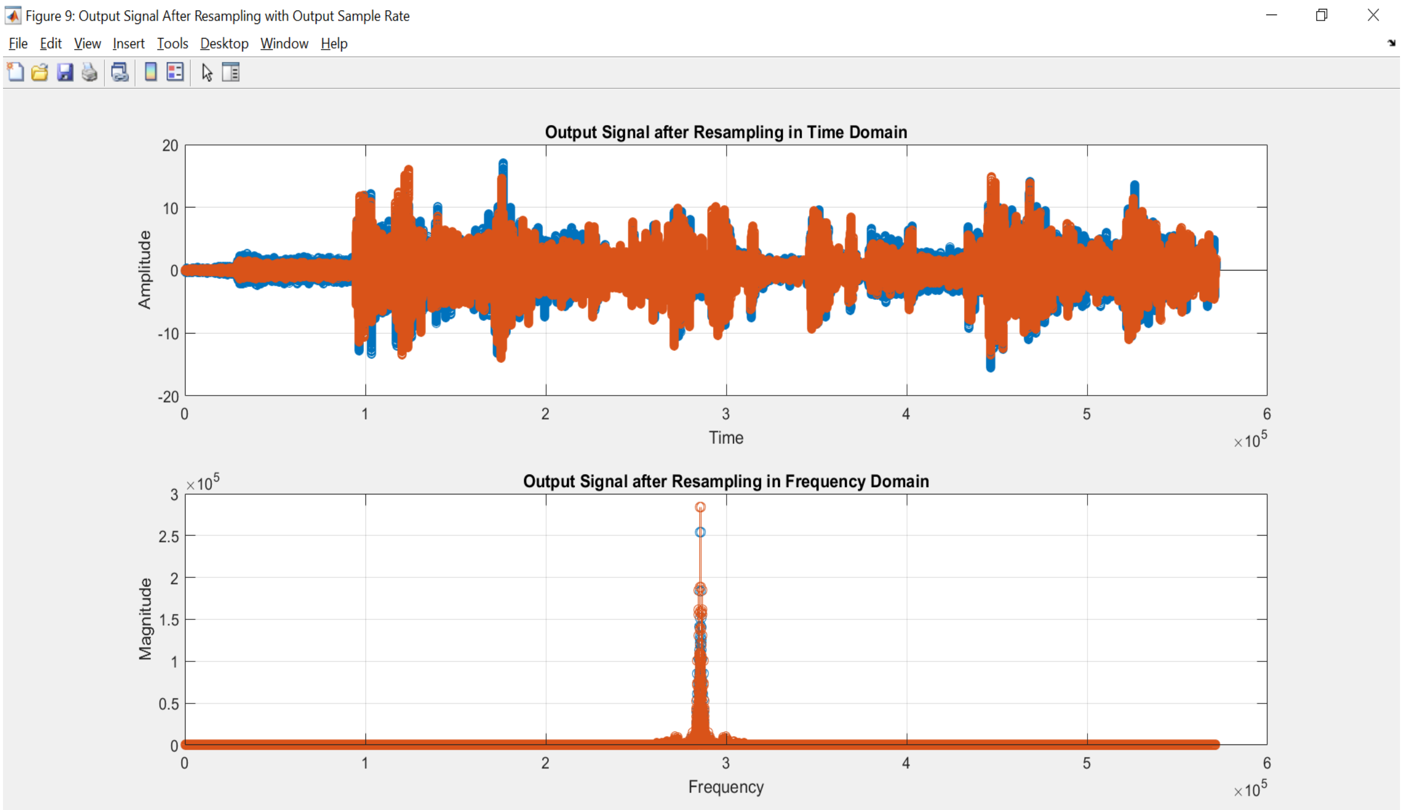


Figure 2.a.64

  - Note that the output wav signal has been resampled with the double output sample rate and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with double output sample rate with those before resampling (Mentioned above).

➢ Output signal in case if decreasing output sample rate to half:

- The user interface is as shown in the figure below *(Figure 2.a.65):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test3.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): FIR
-----------------------------------
Enter the output sample rate: 22050
-----------------------------------
Enter the gain for filter (1) in dB: 20
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 20
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 20
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.a.65

- Playing the output wav signal in case of decreasing the output sample rate to half:

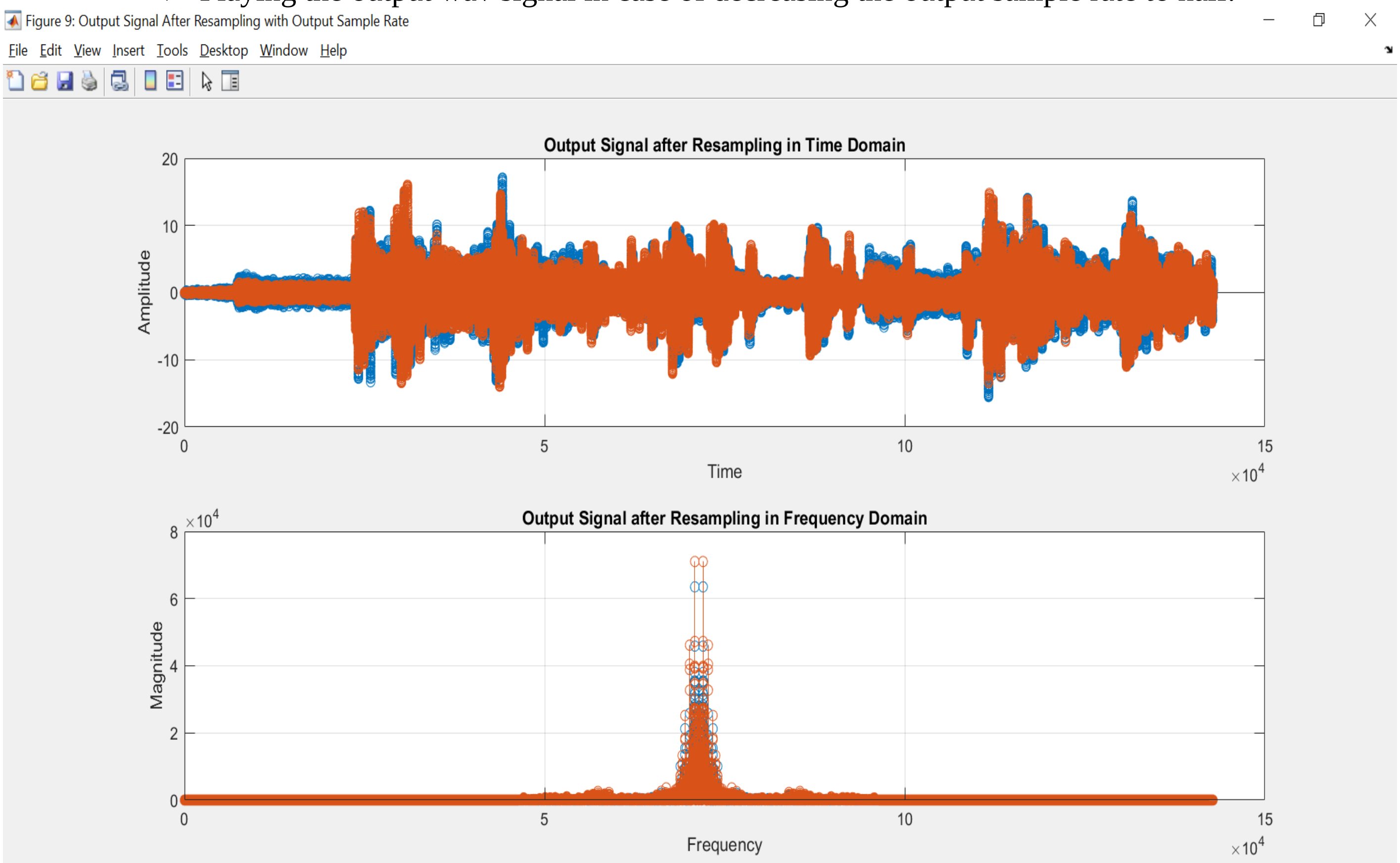


Figure 2.a.66

- Note that the output wav signal has been resampled with decreasing the output sample rate to half and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with decreasing the output sample rate to half with those before resampling (Mentioned above).

b) *If the design is using IIR Filters:*

- Applying the test1 wav file to the filters of type IIR:
  - The user interface is as shown in the figure below *(Figure 2.b.1):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test1.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): IIR
-----------------------------------
Enter the output sample rate: 44100
-----------------------------------
Enter the gain for filter (1) in dB: 0
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 0
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 0
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.b.1

- The output signals in time domain and frequency domain:

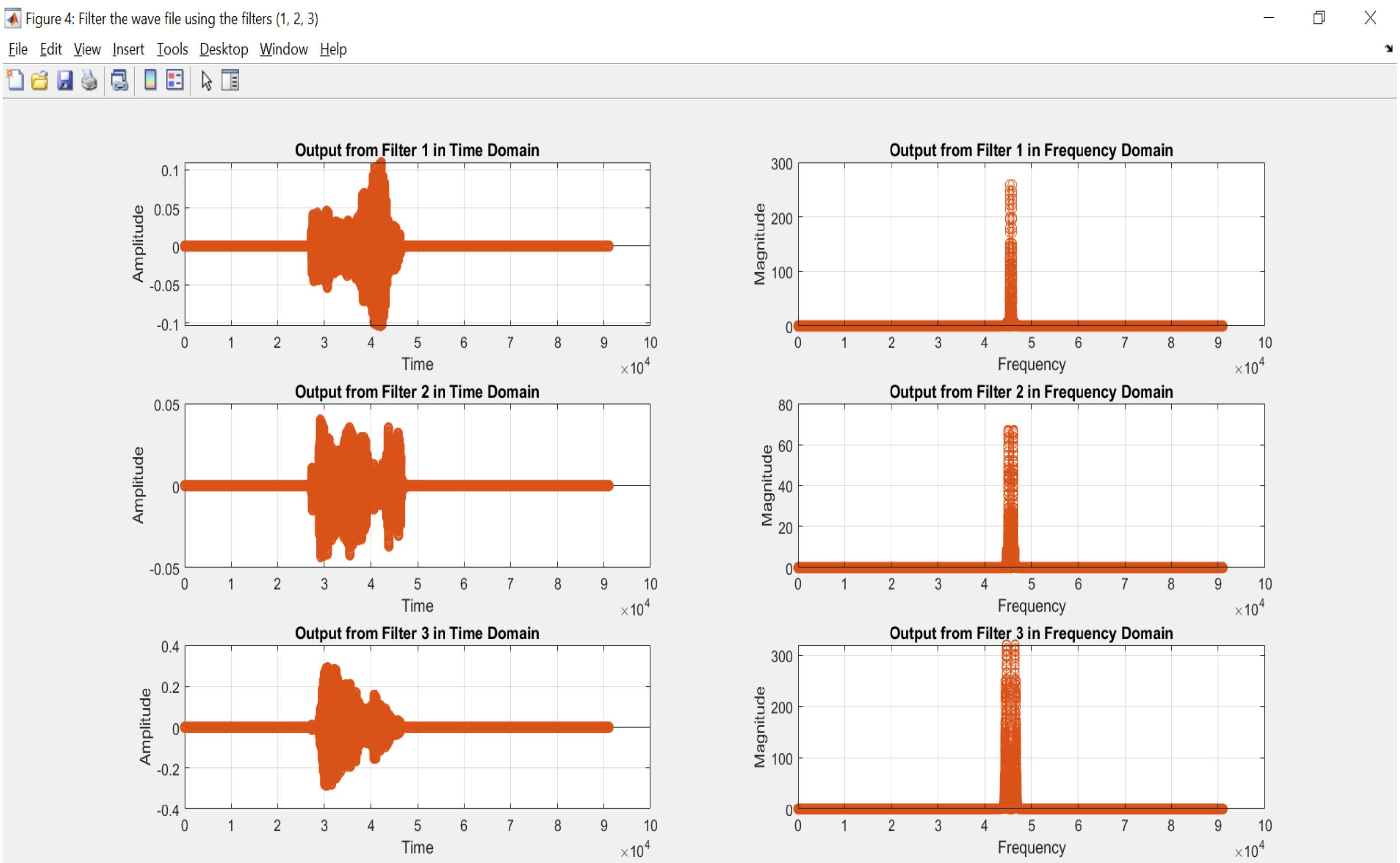


Figure 2.b.2

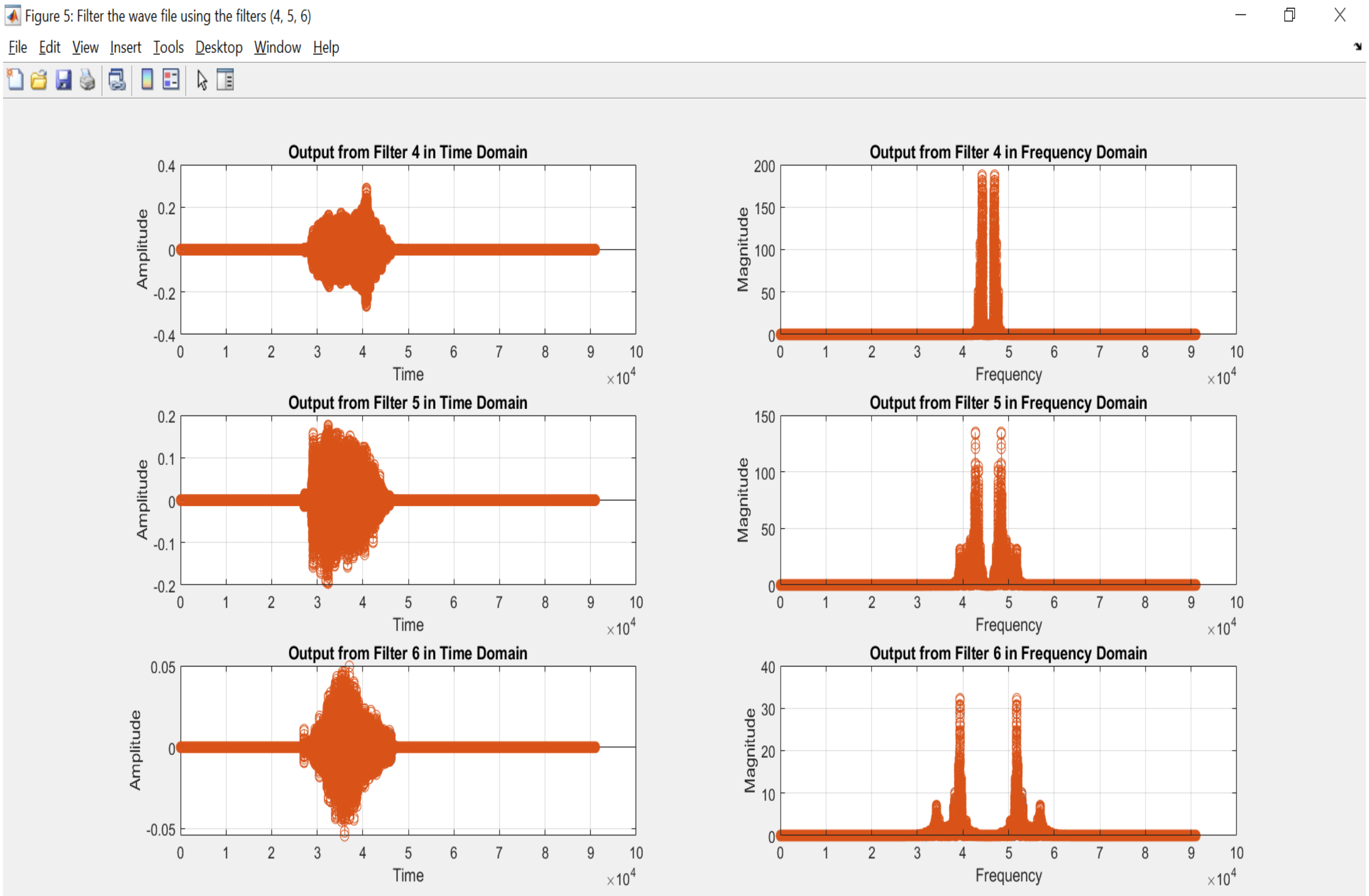


Figure 2.b.3

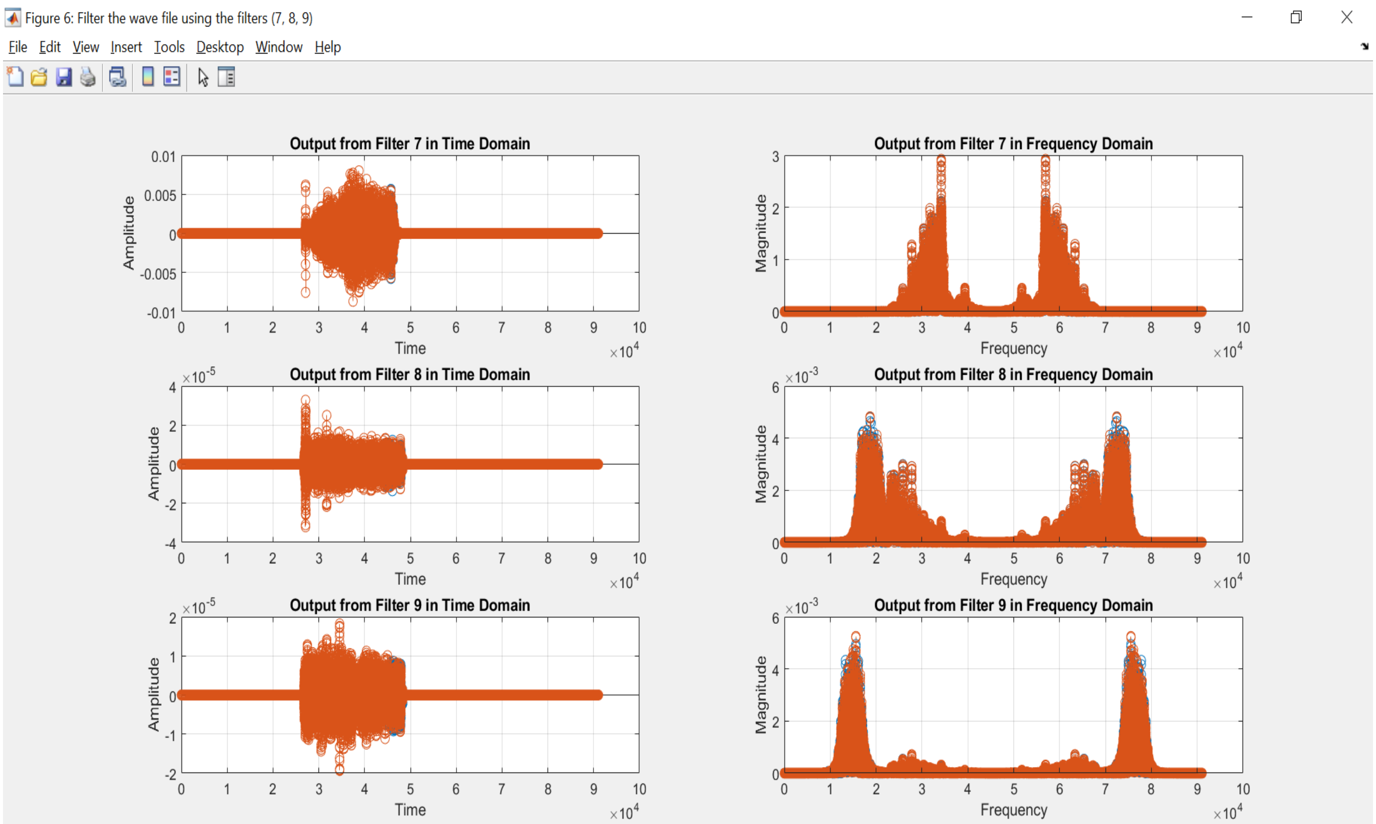


Figure 2.b.4

- Comparison between the Composite Signal amplified with gain (0 dB ➔ 1) with the Original Signal in Time Domain:

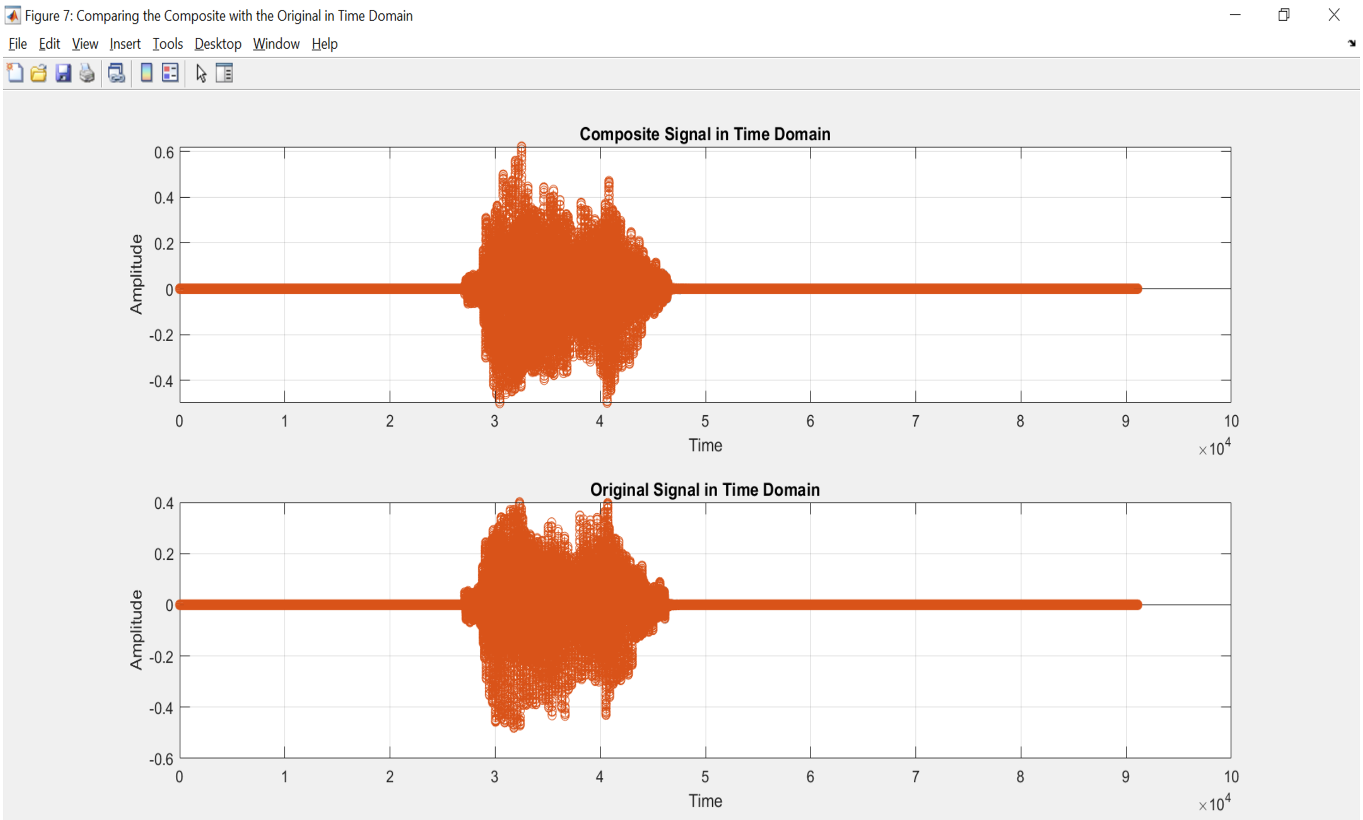


Figure 2.b.5

- Comparison between the Composite Signal amplified with gain (0 dB ➔ 1) with the Original Signal in Frequency Domain:

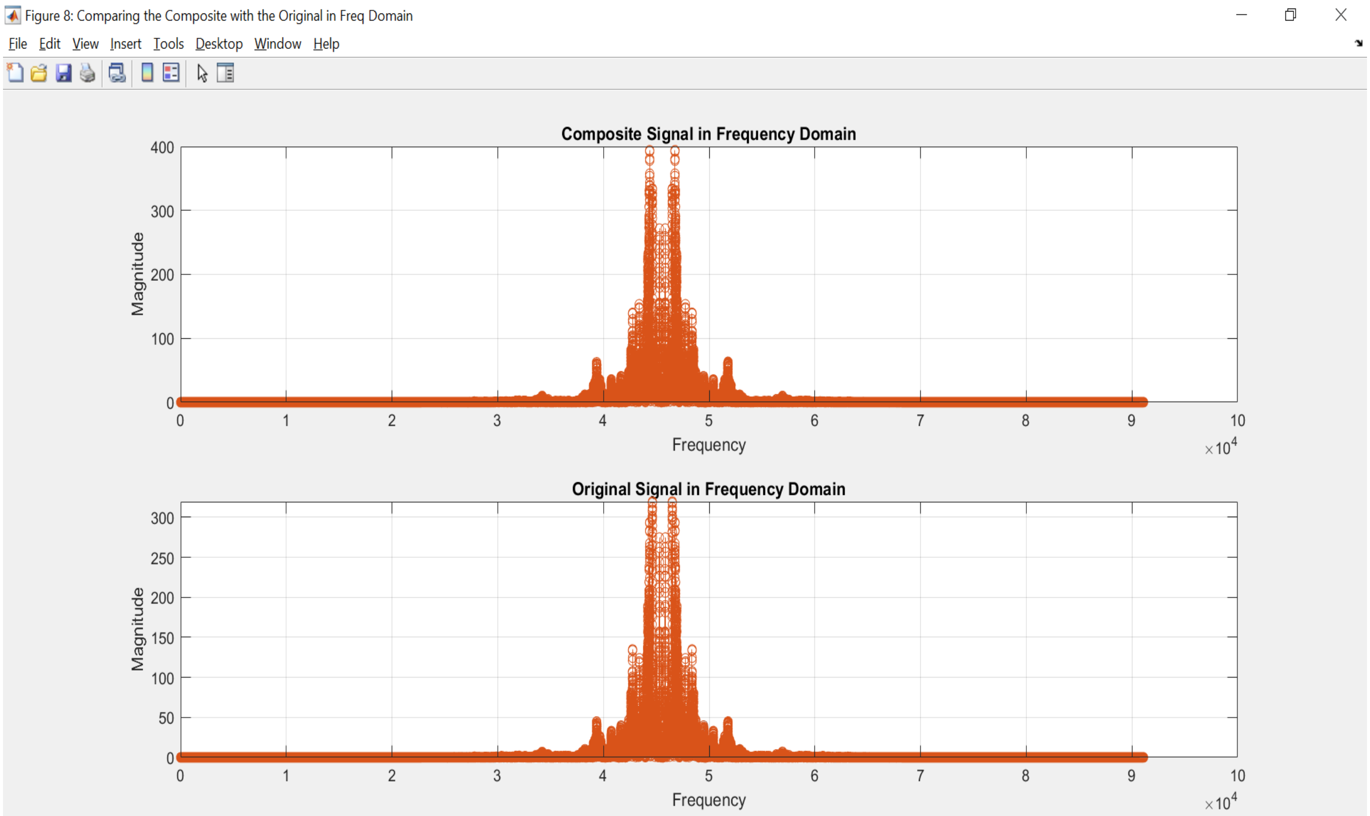


Figure 2.b.6

- Playing the output wav signal with the desired output sample rate:

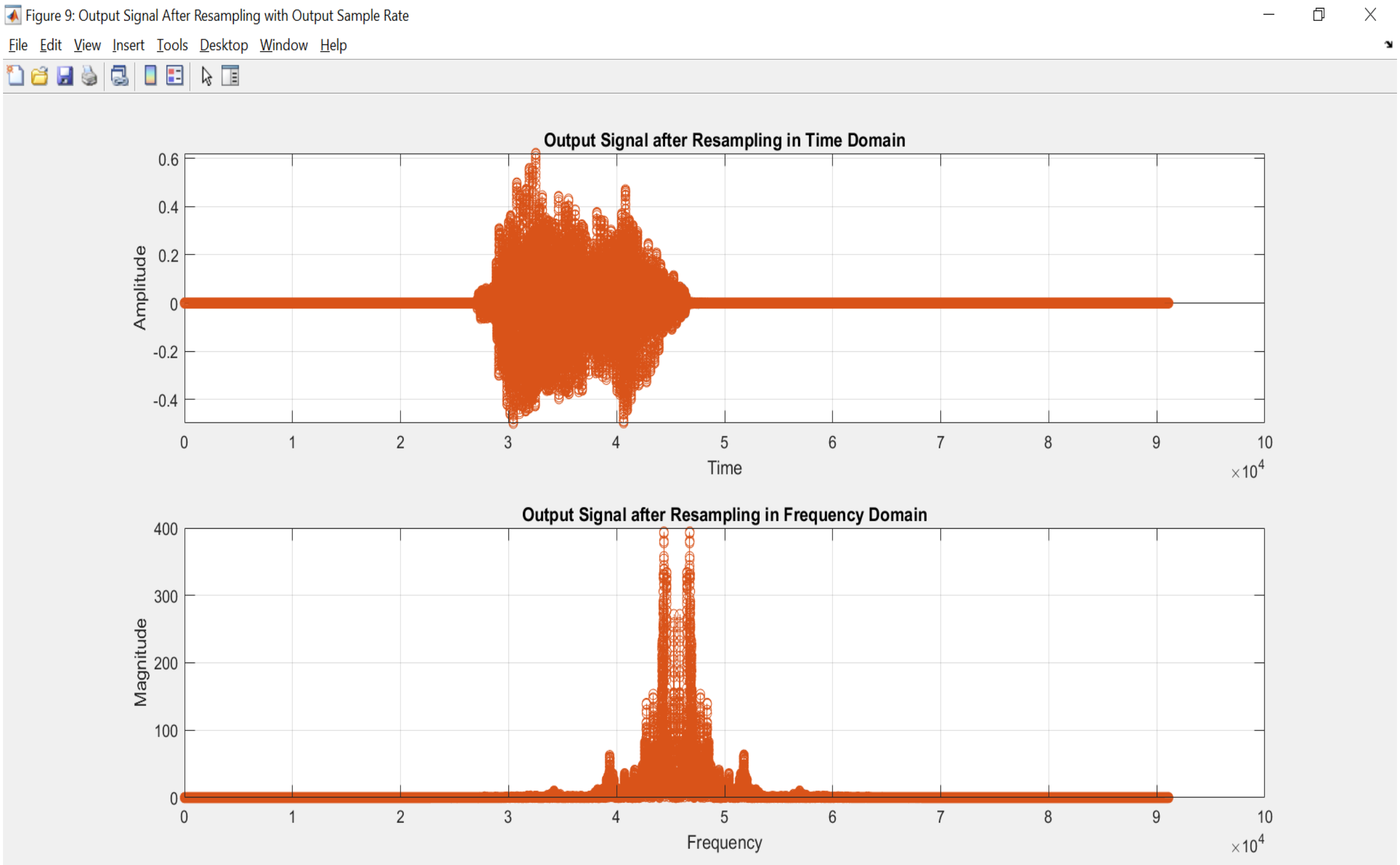


Figure 2.b.7

- Note that the output wav signal has been played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal.

➢ Output signal in case if doubling output sample rate:

- The user interface is as shown in the figure below *(Figure 2.b.8):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
----------------------------------
Enter the wave file name: test1.wav
----------------------------------
Enter the type of filters used (IIR-FIR): IIR
----------------------------------
Enter the output sample rate: 88200
----------------------------------
Enter the gain for filter (1) in dB: 0
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 0
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 0
----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.b.8

- Playing the output wav signal in case of doubling the output sample rate:

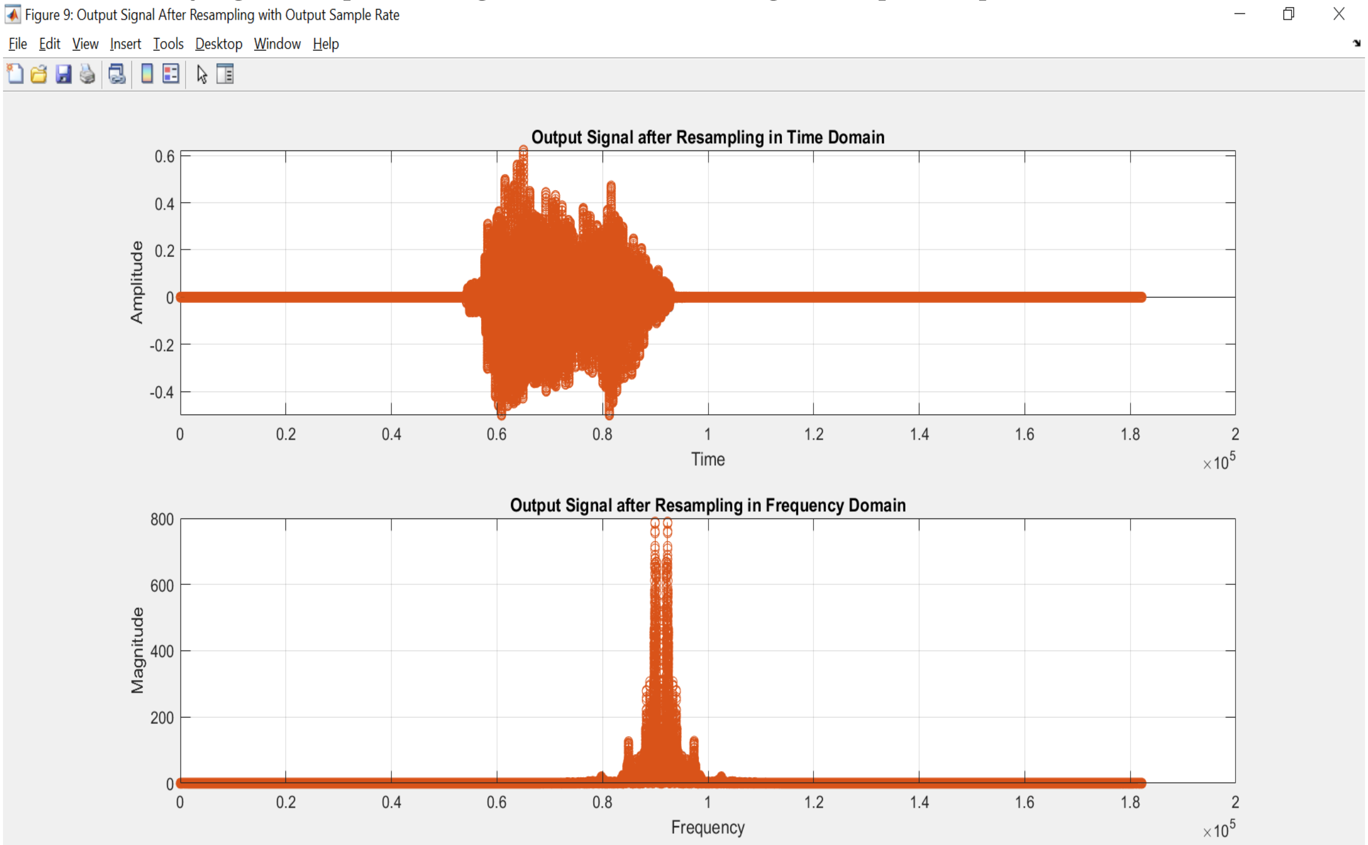


Figure 2.b.9

- Note that the output wav signal has been resampled with the double output sample rate and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with double output sample rate with those before resampling (Mentioned above).

- Output signal in case if decreasing output sample rate to half:
    - The user interface is as shown in the figure below *(Figure 2.b.10):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
----------------------------------
Enter the wave file name: test1.wav
----------------------------------
Enter the type of filters used (IIR-FIR): IIR
----------------------------------
Enter the output sample rate: 22050
----------------------------------
Enter the gain for filter (1) in dB: 0
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 0
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 0
----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.b.10

- Playing the output wav signal in case of decreasing the output sample rate to half:

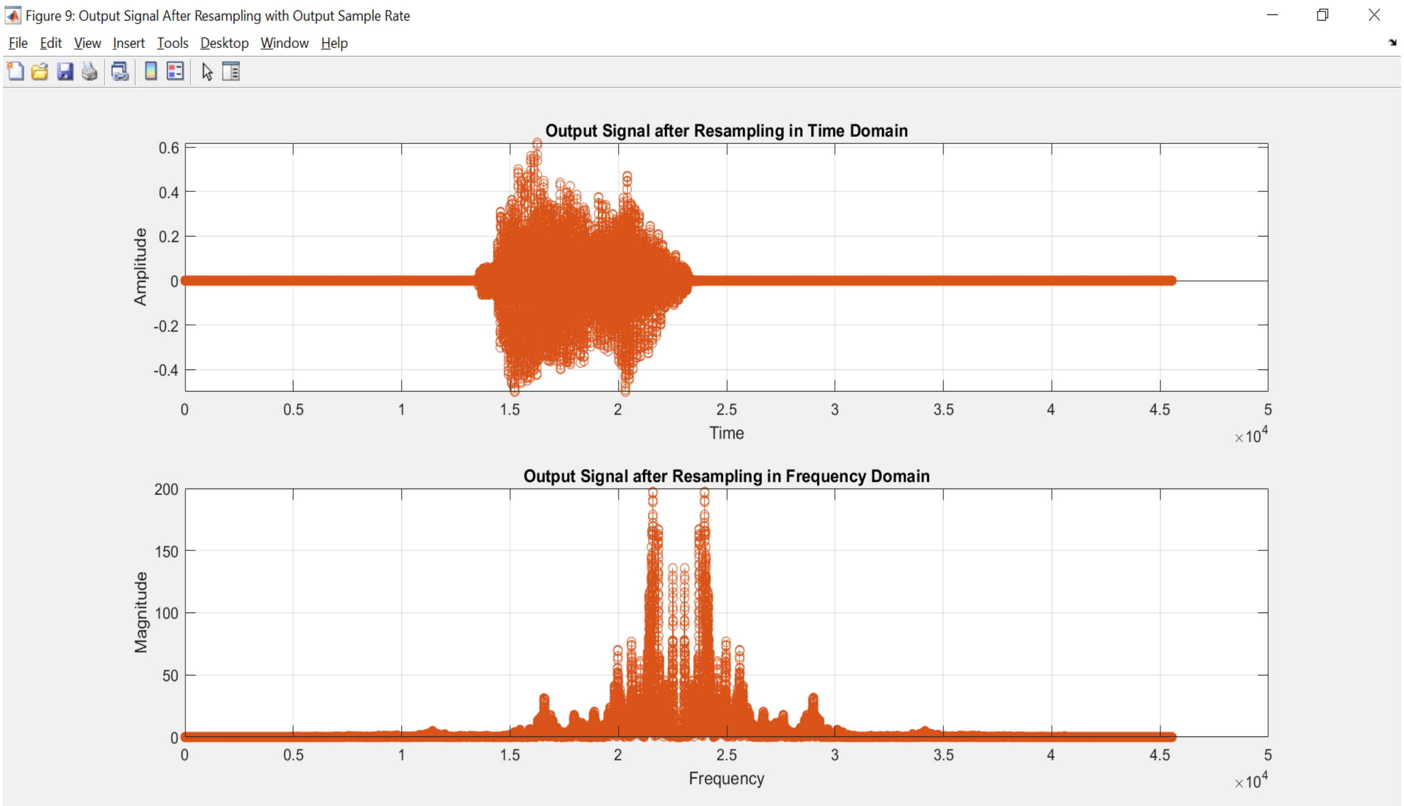


Figure 2.b.11

- Note that the output wav signal has been resampled with decreasing the output sample rate to half and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with decreasing the output sample rate to half with those before resampling (Mentioned above).

➢ Applying the test1 wav file to the filters of type IIR:

- The user interface is as shown in the figure below *(Figure 2.b.12):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
---------------------------------
Enter the wave file name: test1.wav
---------------------------------
Enter the type of filters used (IIR-FIR): IIR
---------------------------------
Enter the output sample rate: 44100
---------------------------------
Enter the gain for filter (1) in dB: 20
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 20
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 20
---------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.b.12

- The output signals in time domain and frequency domain:

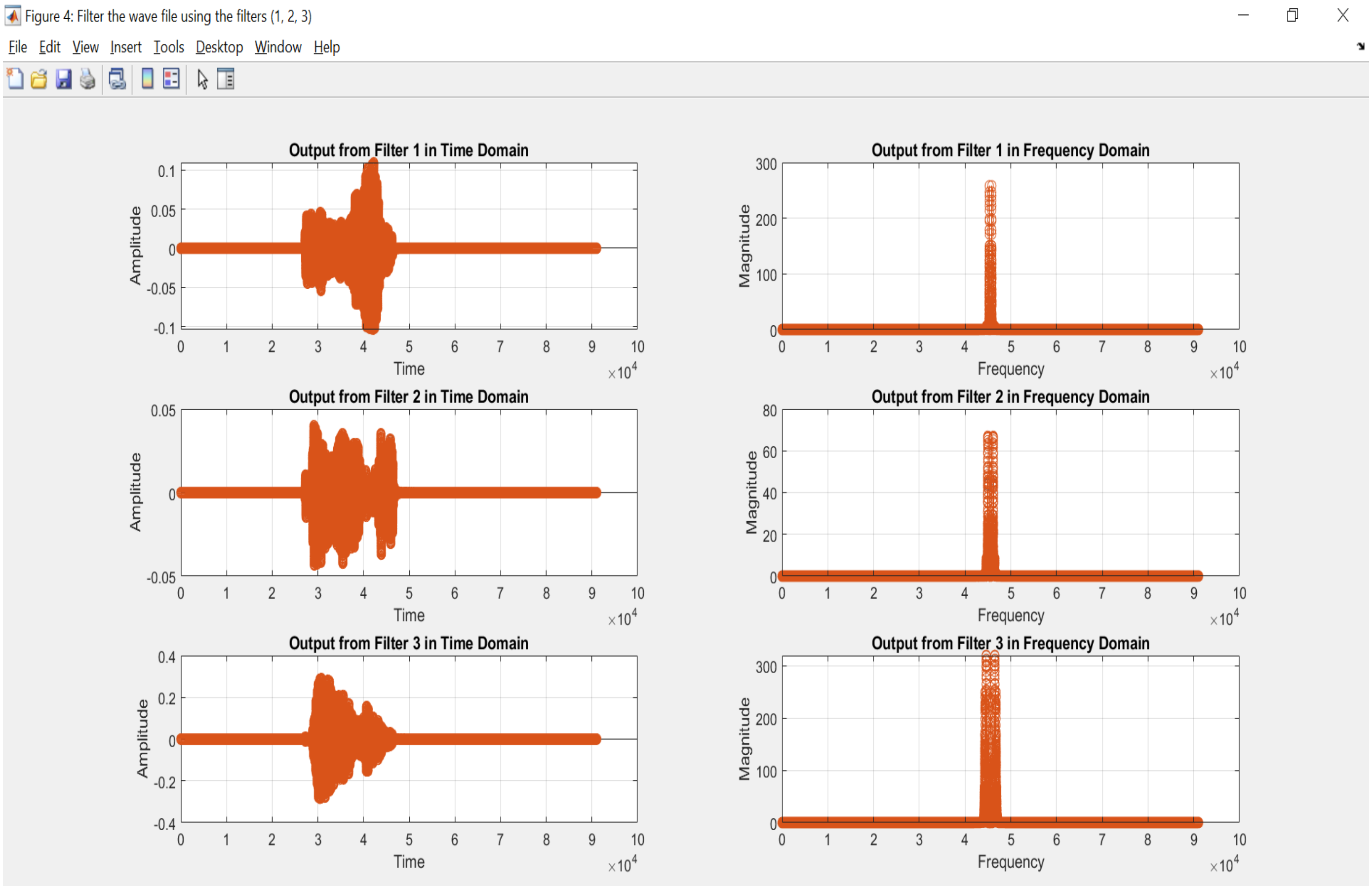


Figure 2.b.13

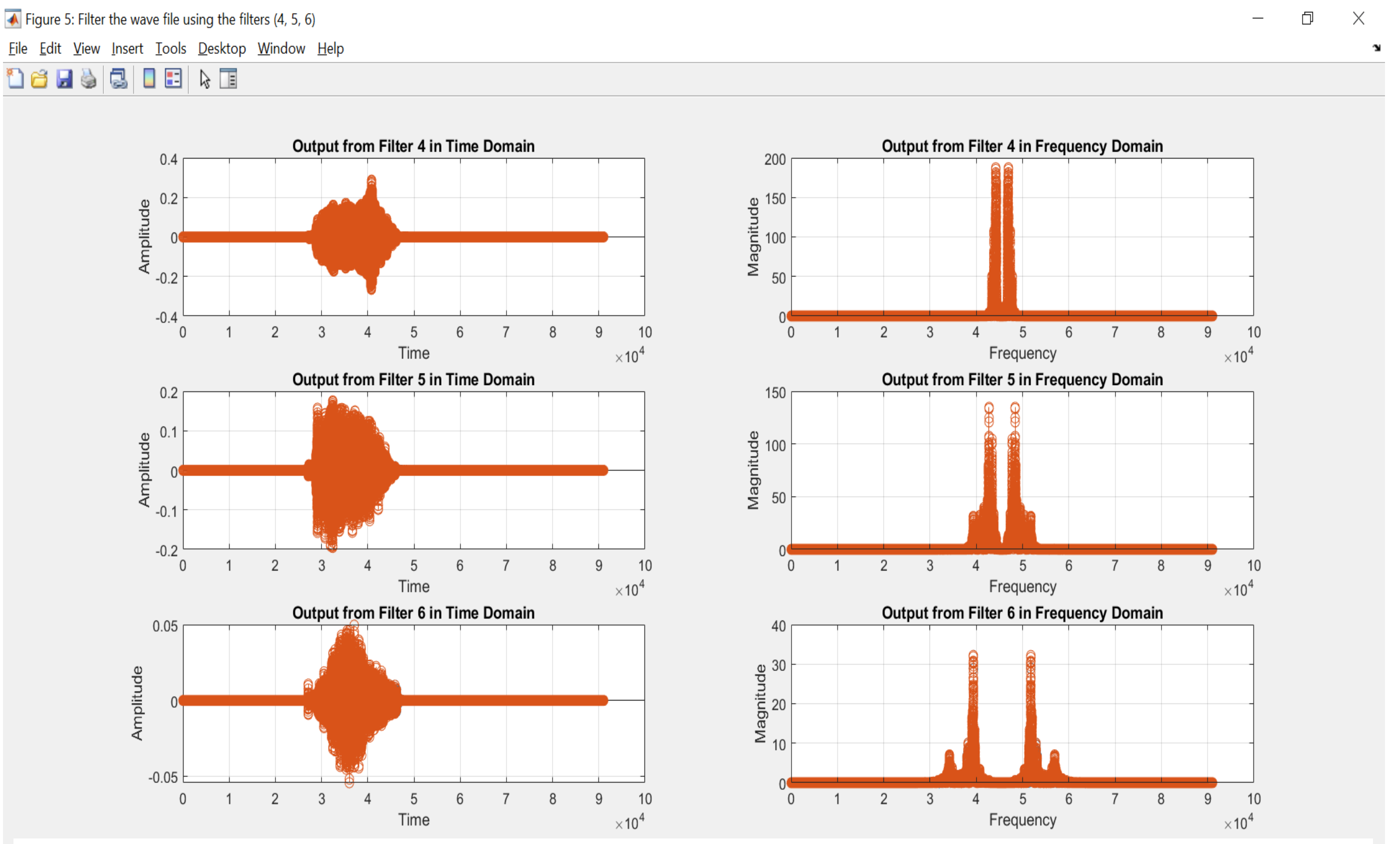


Figure 2.b.14

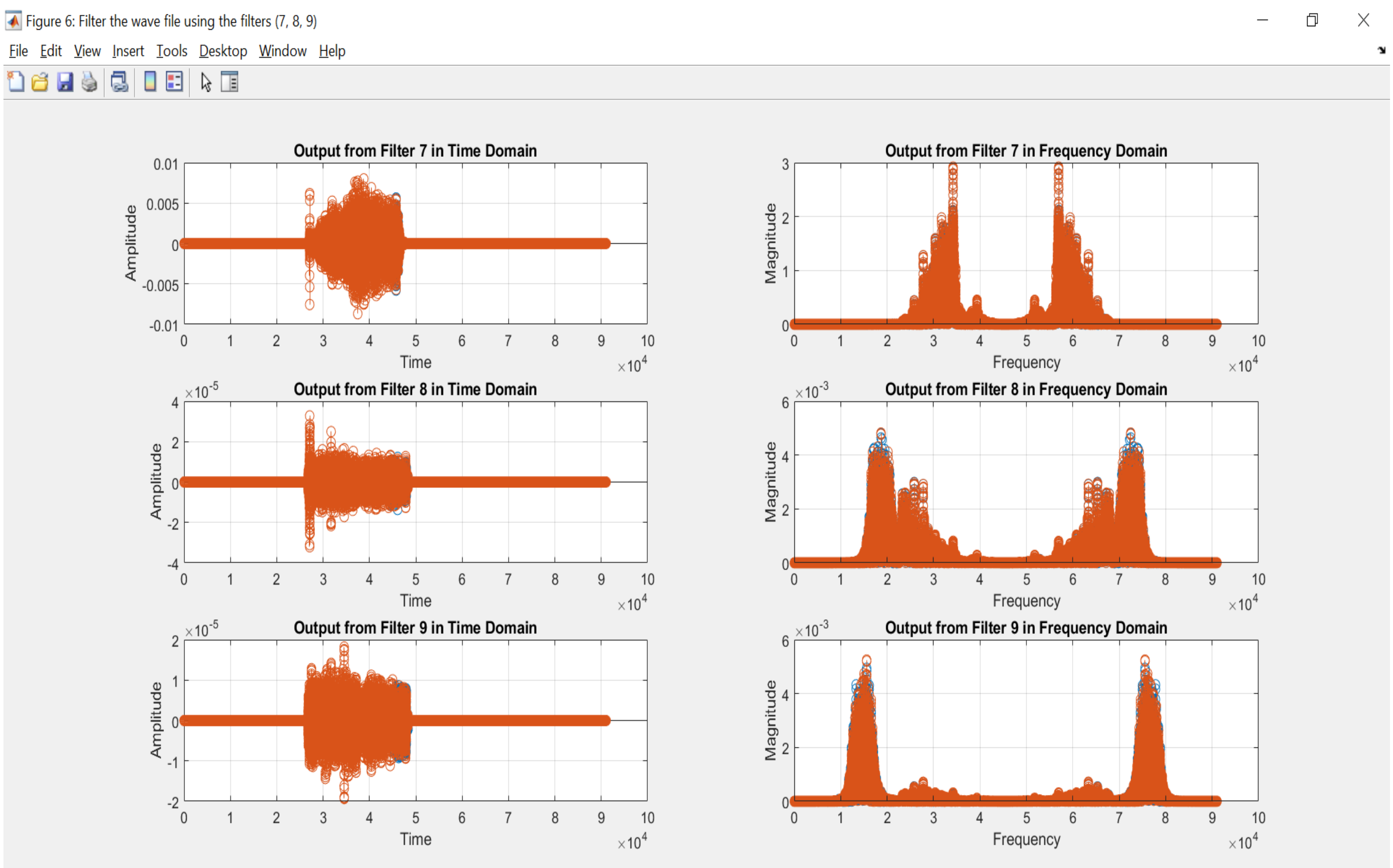


Figure 2.b.15

Comparison between the Composite Signal amplified with the gains showed in the figure above *(Figure 2.b.12)* with the Original Signal in Time Domain:

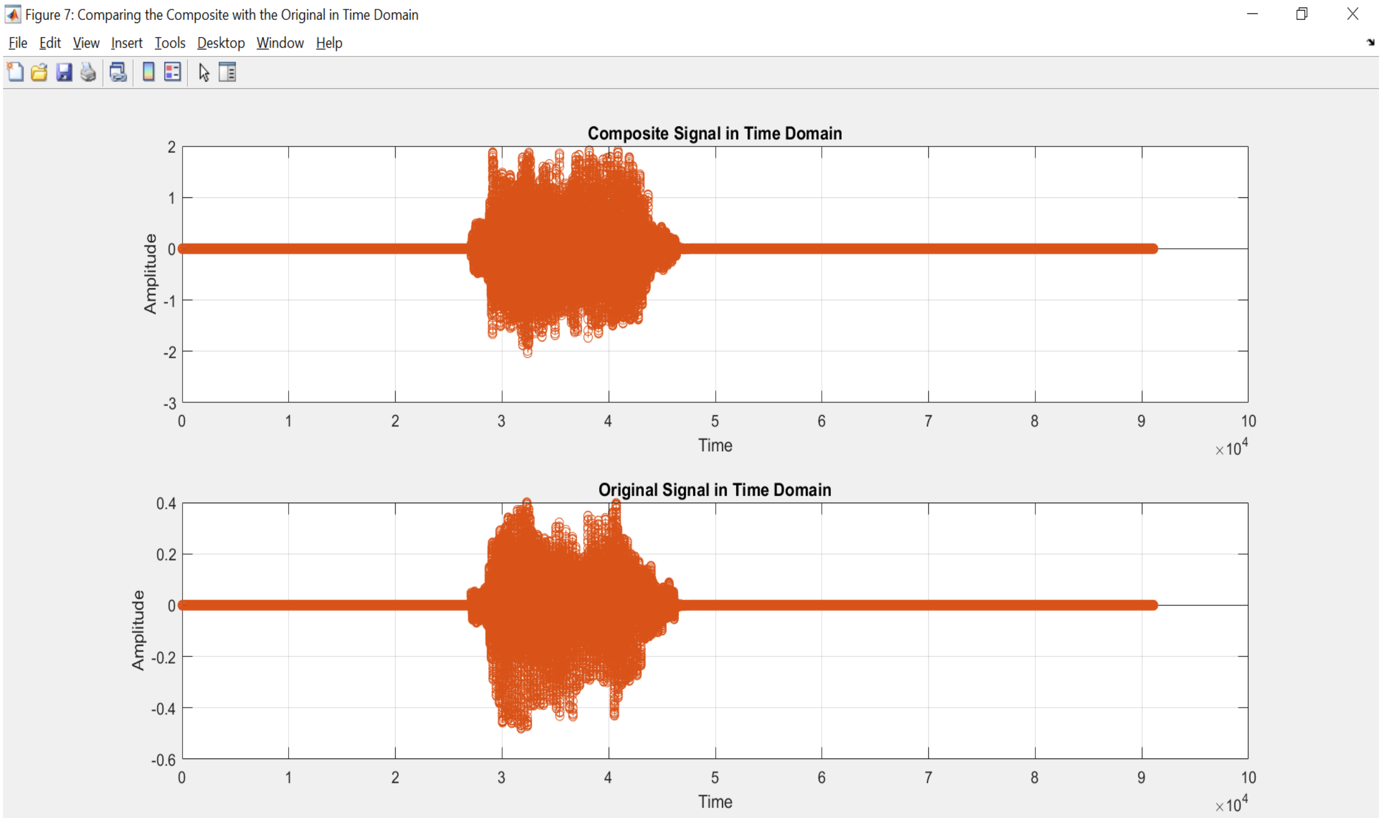


Figure 2.b.16

Comparison between the Composite Signal amplified with the gains showed in the figure above *(Figure 2.b.12)* with the Original Signal in Frequency Domain:

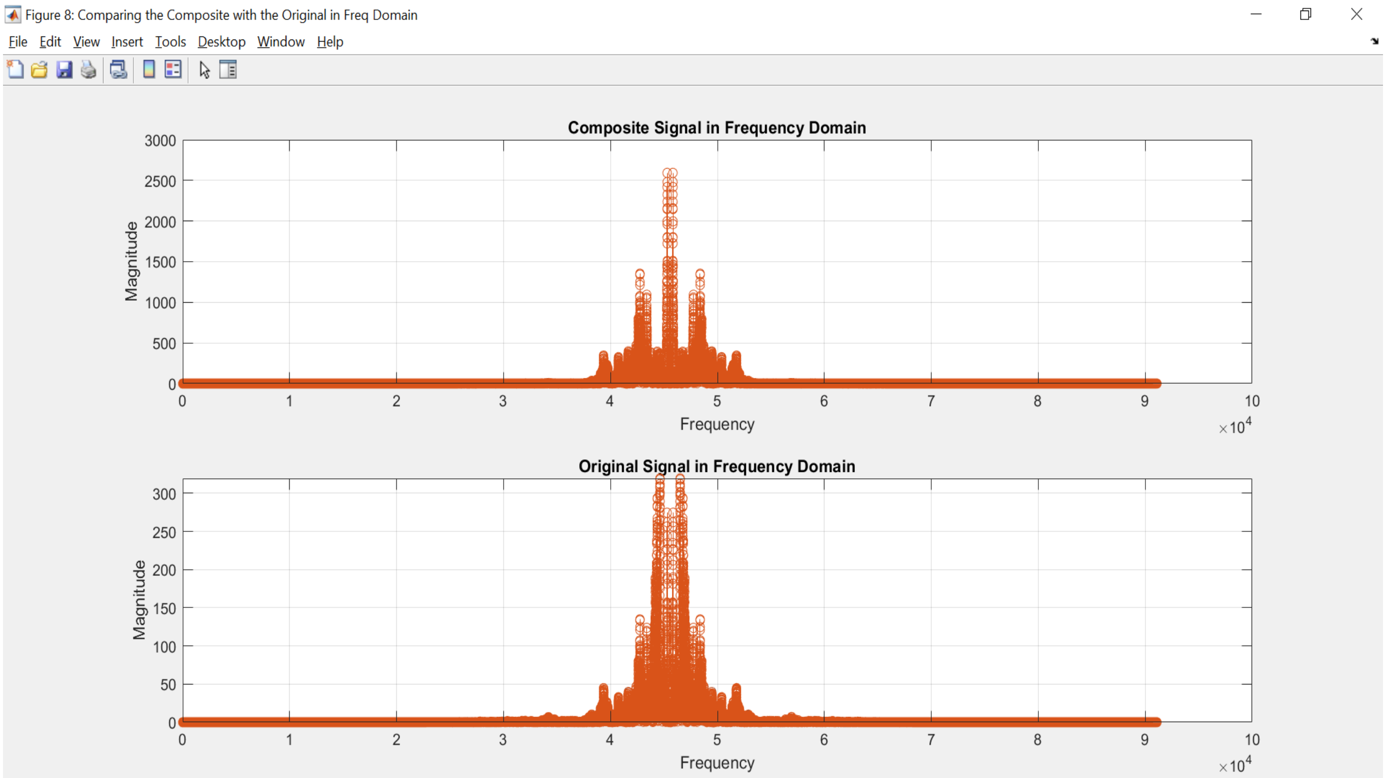


Figure 2.b.17

- Playing the output wav signal with the desired output sample rate:

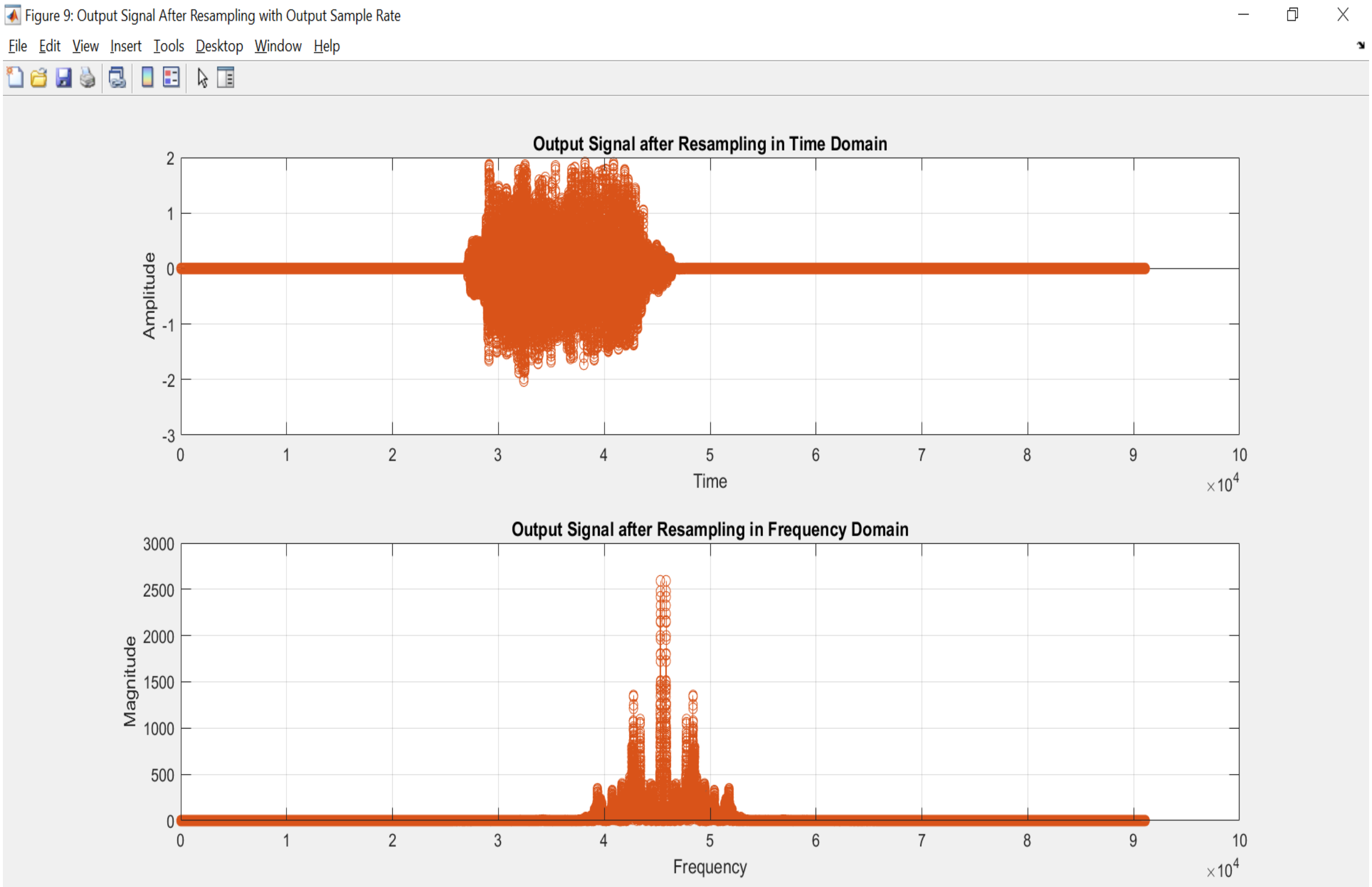


Figure 2.b.18

- Note that the output wav signal has been played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal.

➢ Output signal in case if doubling output sample rate:

- The user interface is as shown in the figure below *(Figure 2.b.19):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test1.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): IIR
-----------------------------------
Enter the output sample rate: 88200
-----------------------------------
Enter the gain for filter (1) in dB: 20
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 20
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 20
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.b.19

- Playing the output wav signal in case of doubling the output sample rate:

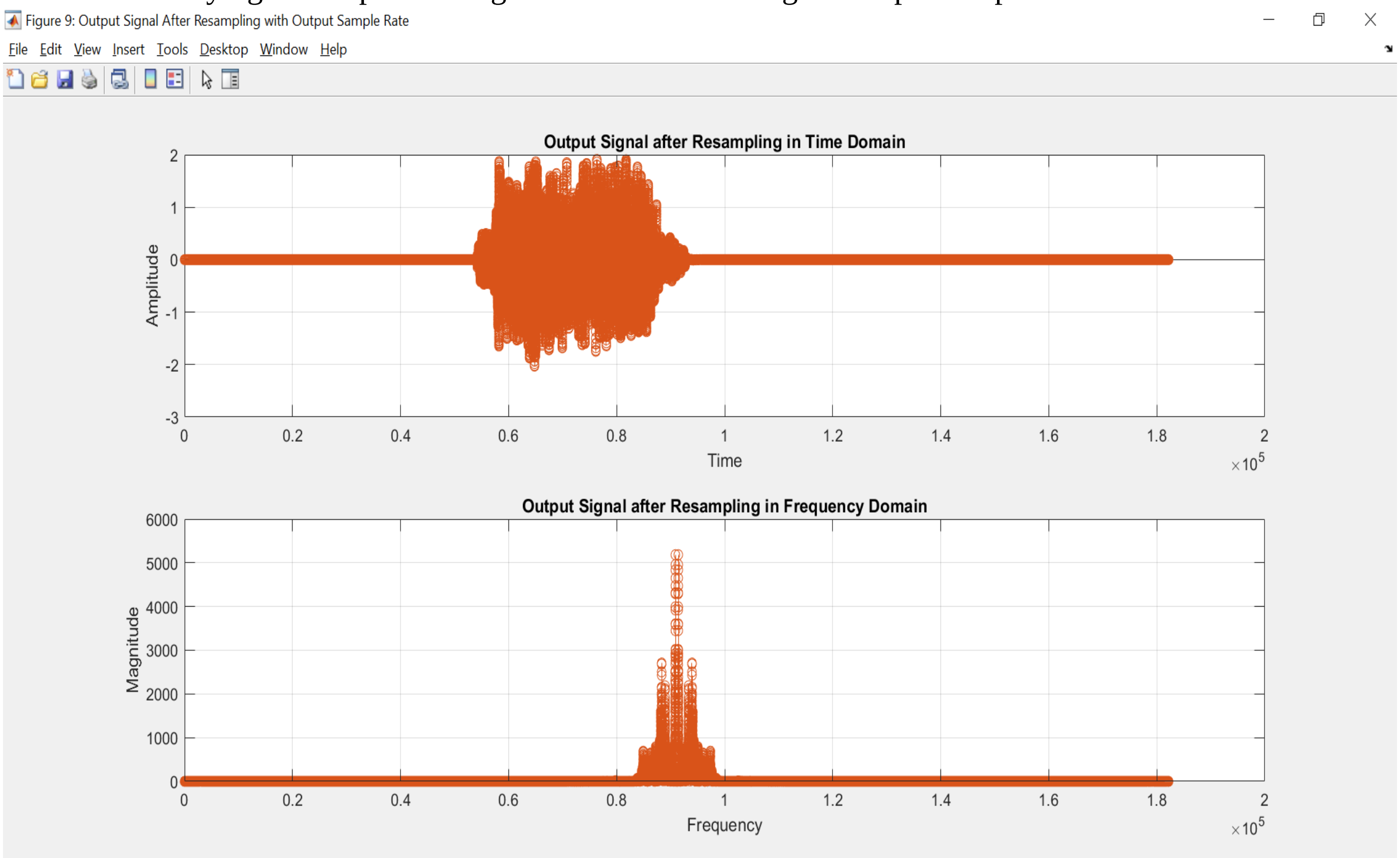


Figure 2.b.20

- Note that the output wav signal has been resampled with the double output sample rate and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with double output sample rate with those before resampling (Mentioned above).

- Output signal in case if decreasing output sample rate to half:
  - The user interface is as shown in the figure below *(Figure 2.b.21):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test1.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): IIR
-----------------------------------
Enter the output sample rate: 22050
-----------------------------------
Enter the gain for filter (1) in dB: 20
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 20
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 20
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.b.21

  - Playing the output wav signal in case of decreasing the output sample rate to half:

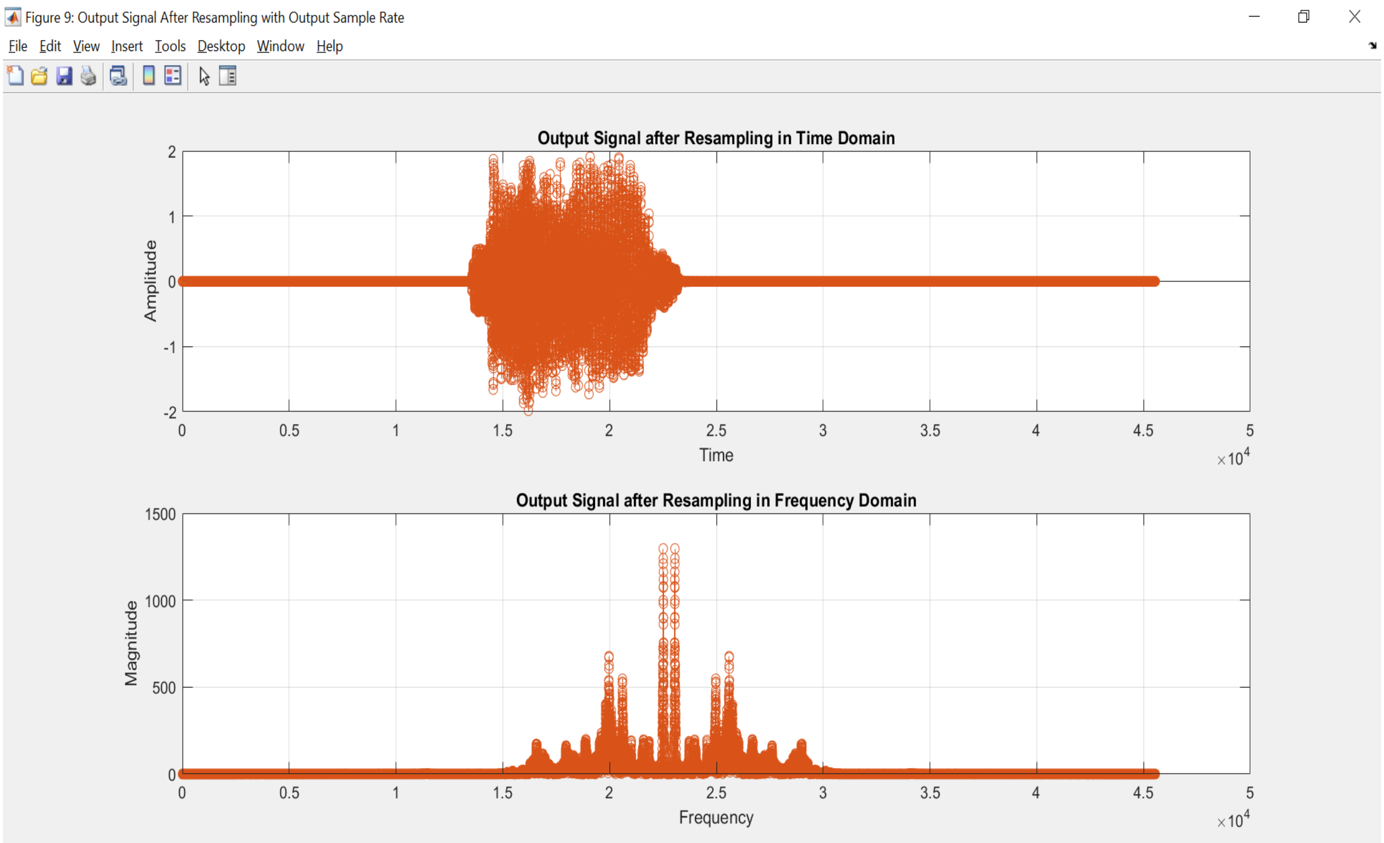


Figure 2.b.22

  - Note that the output wav signal has been resampled with decreasing the output sample rate to half and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with decreasing the output sample rate to half with those before resampling (Mentioned above).

➢ Applying the test2 wav file to the filters of type IIR:

- The user interface is as shown in the figure below *(Figure 2.b.23):*

Command Window
```
****************WELCOME TO AUDIO EQUALIZER****************
----------------------------------
Enter the wave file name: test2.wav
----------------------------------
Enter the type of filters used (IIR-FIR): IIR
----------------------------------
Enter the output sample rate: 44100
----------------------------------
Enter the gain for filter (1) in dB: 0
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 0
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 0
----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.b.23

- The output signals in time domain and frequency domain:

Figure 2.b.24

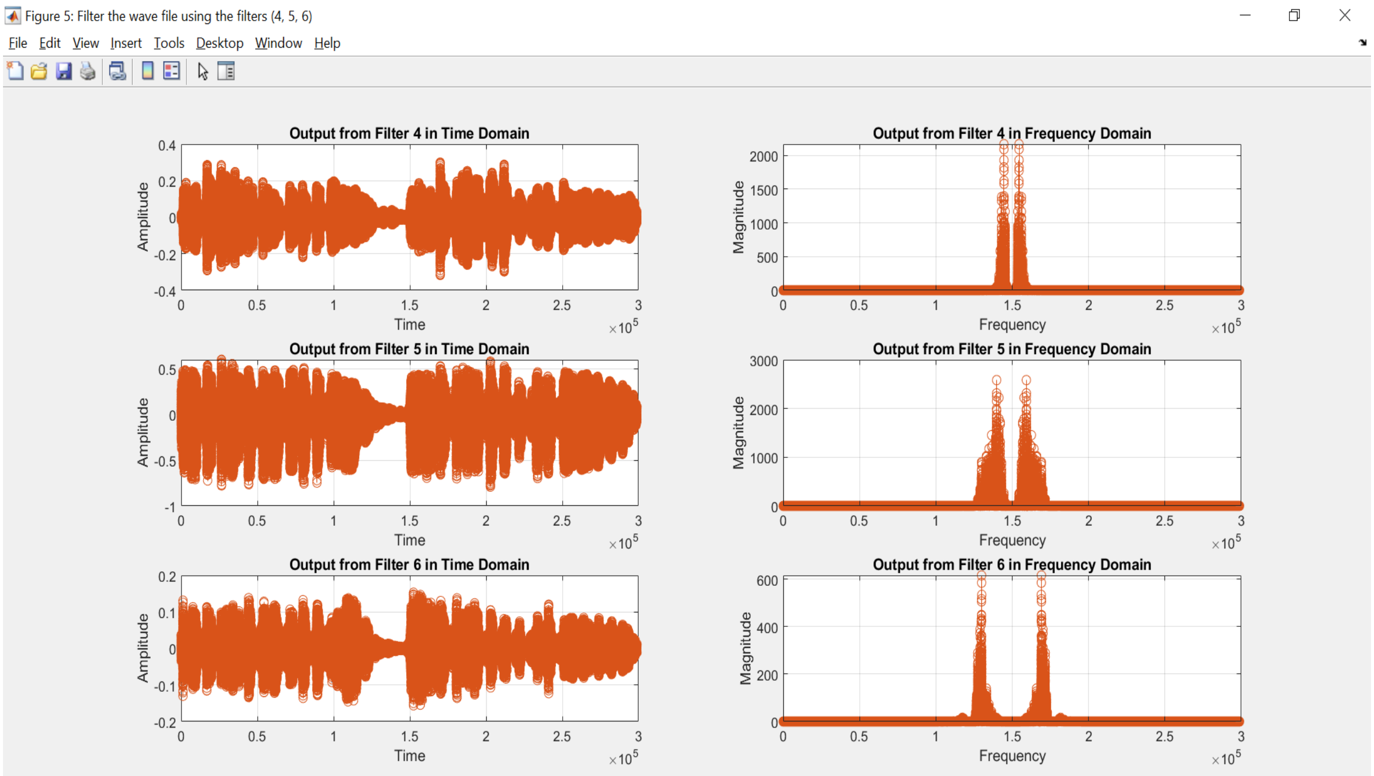


Figure 2.b.25

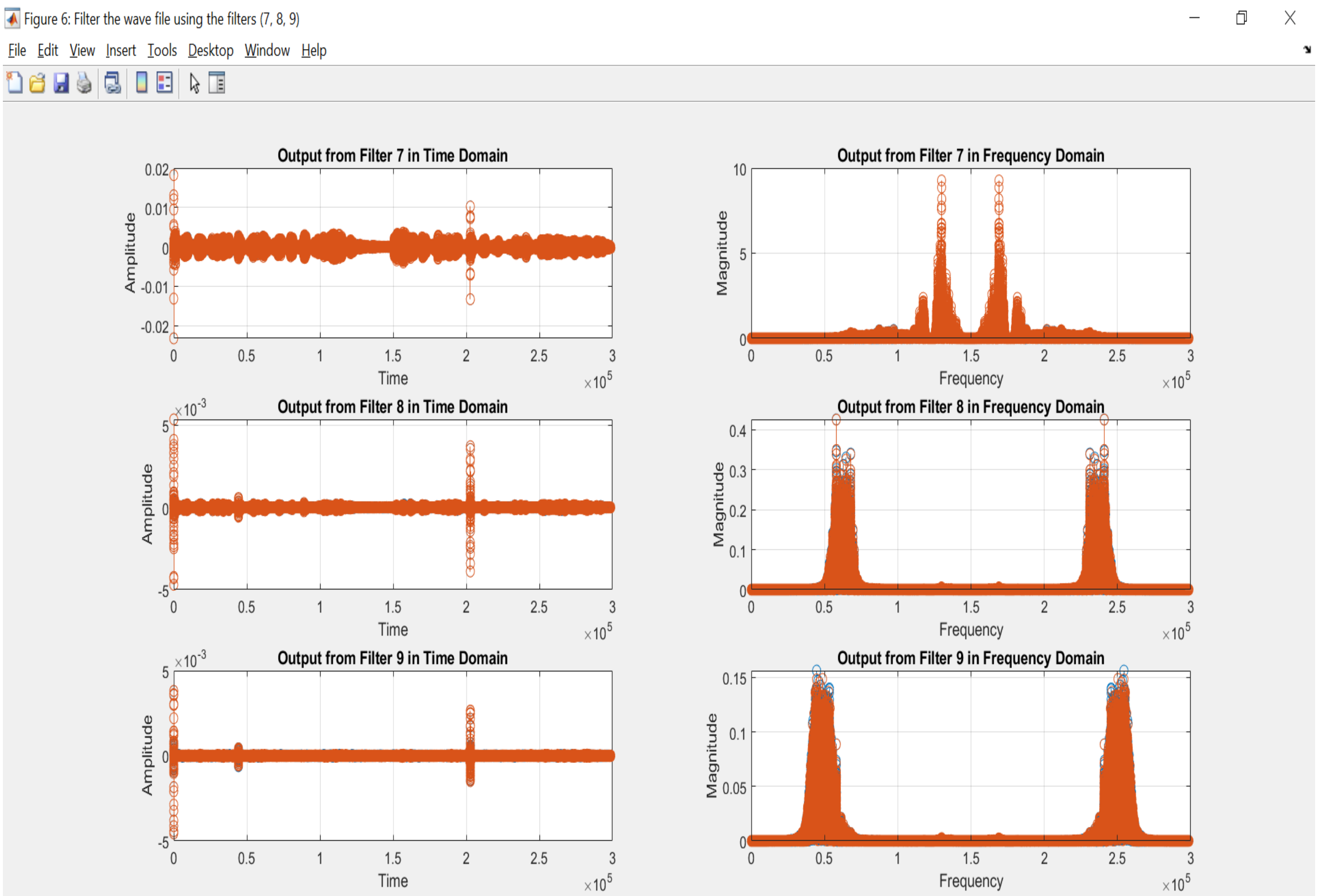


Figure 2.b.26

- Comparison between the Composite Signal amplified with gain (0 dB ➔ 1) with the Original Signal in Time Domain:

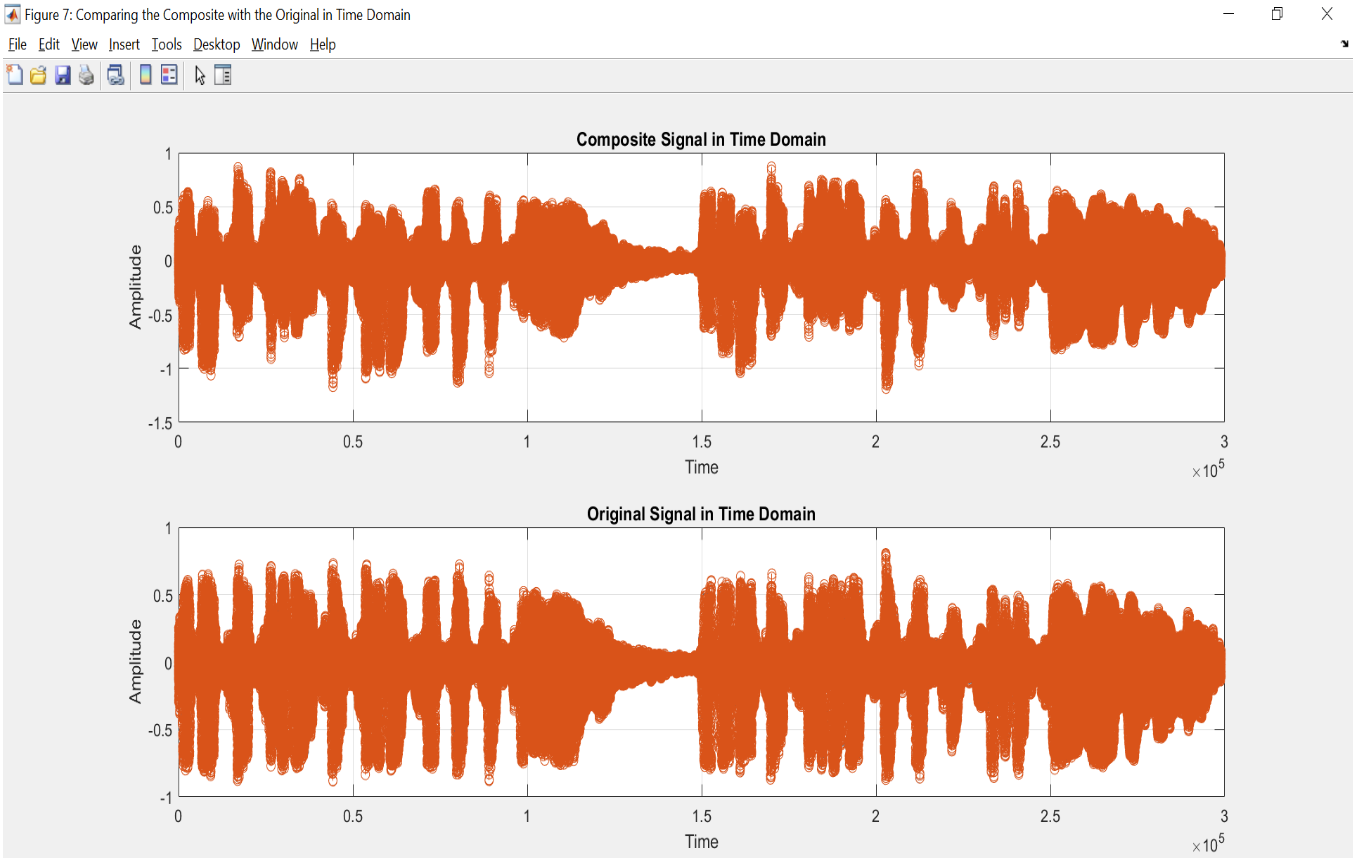


Figure 2.b.27

- Comparison between the Composite Signal amplified with gain (0 dB ➔ 1) with the Original Signal in Frequency Domain:

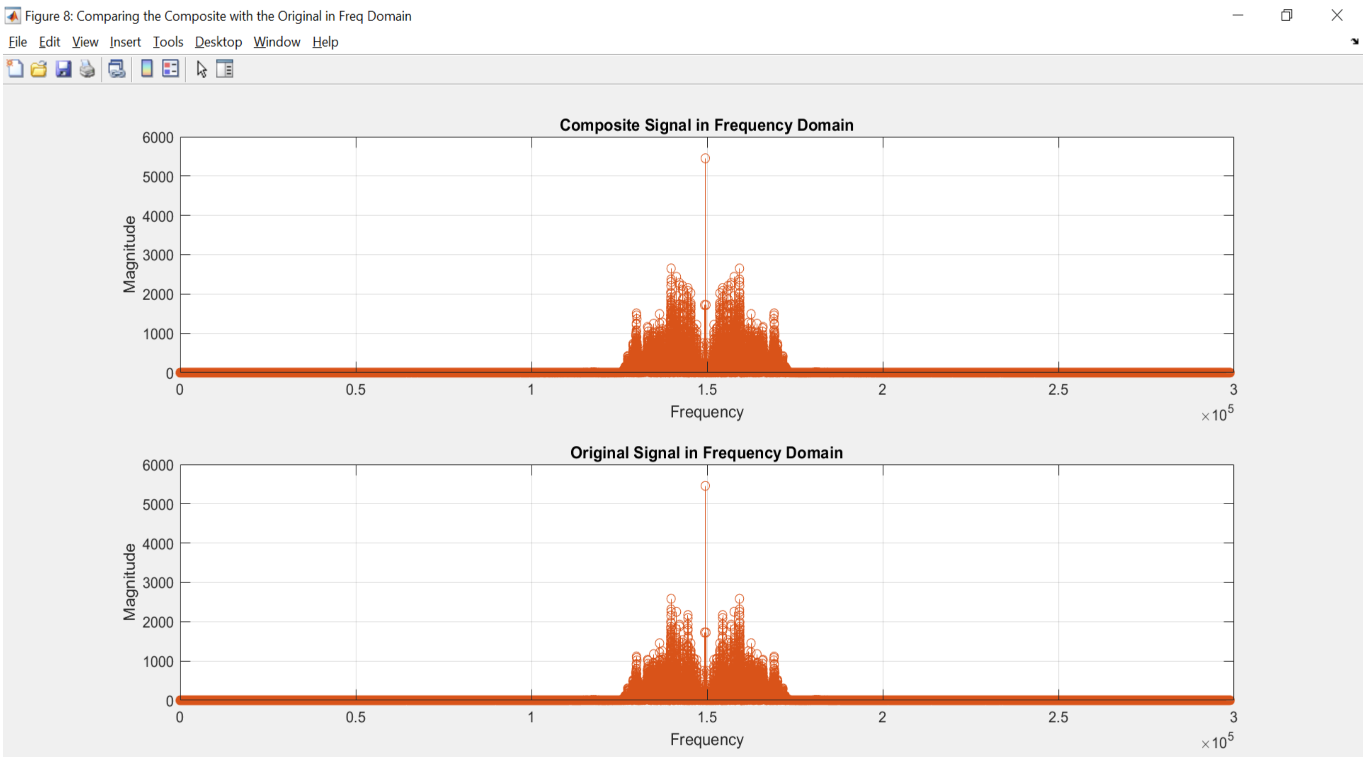


Figure 2.b.28

- Playing the output wav signal with the desired output sample rate:

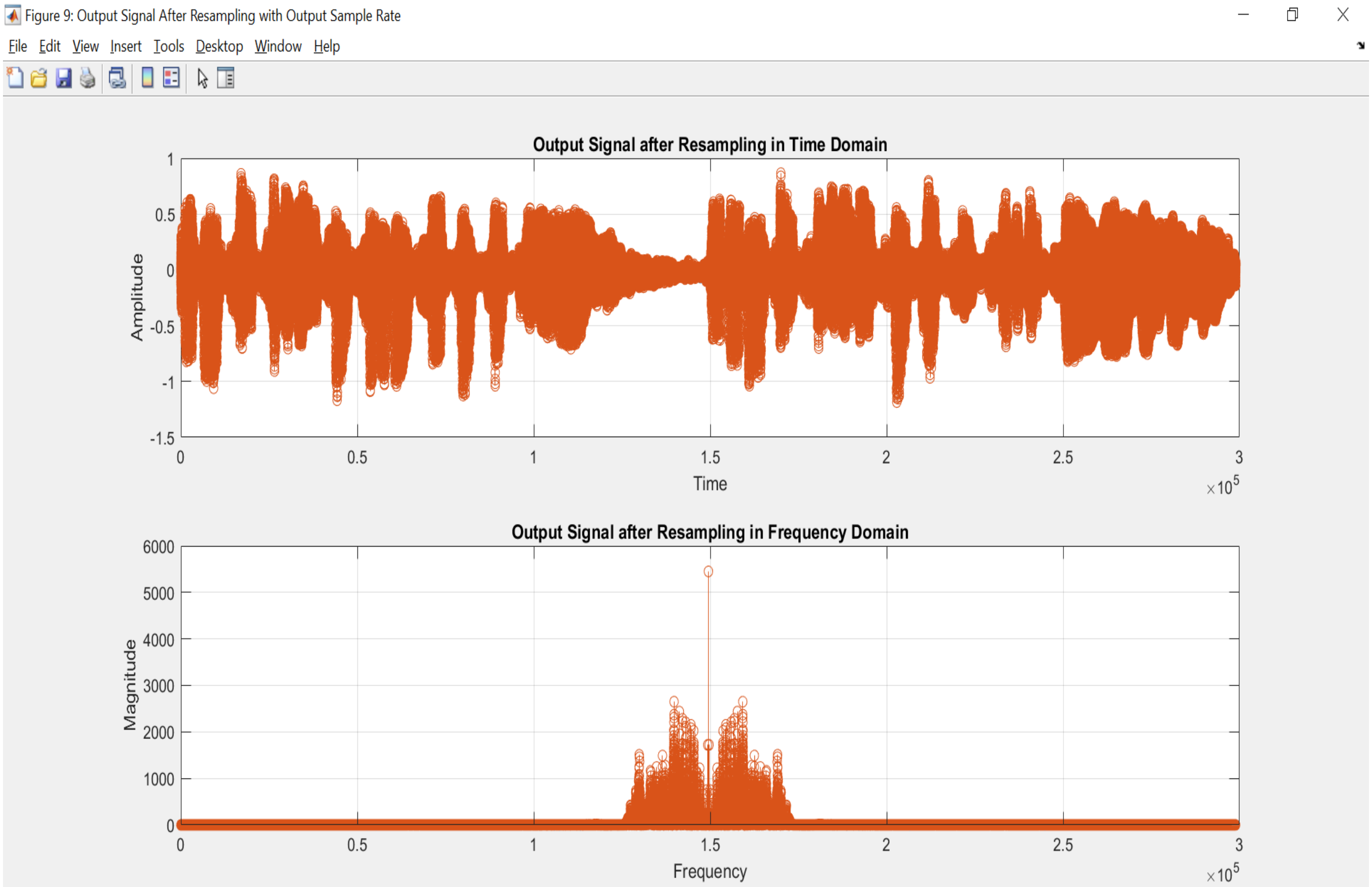


Figure 2.b.29

- Note that the output wav signal has been played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal.

➢ Output signal in case if doubling output sample rate:

- The user interface is as shown in the figure below *(Figure 2.b.30):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test2.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): IIR
-----------------------------------
Enter the output sample rate: 88200
-----------------------------------
Enter the gain for filter (1) in dB: 0
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 0
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 0
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.b.30

- Playing the output wav signal in case of doubling the output sample rate:

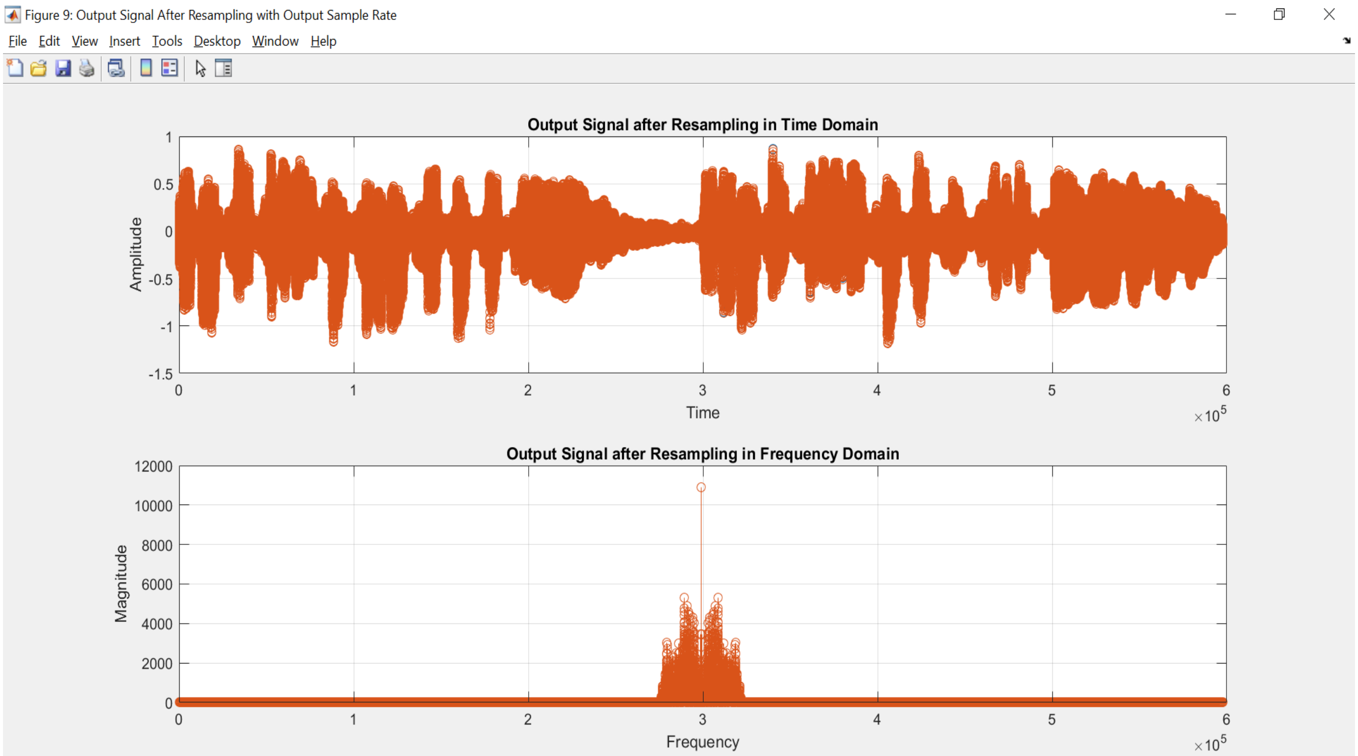


Figure 2.b.31

- Note that the output wav signal has been resampled with the double output sample rate and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with double output sample rate with those before resampling (Mentioned above).

- Output signal in case if decreasing output sample rate to half:
  - The user interface is as shown in the figure below *(Figure 2.b.32):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
----------------------------------
Enter the wave file name: test2.wav
----------------------------------
Enter the type of filters used (IIR-FIR): IIR
----------------------------------
Enter the output sample rate: 22050
----------------------------------
Enter the gain for filter (1) in dB: 0
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 0
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 0
----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.b.32

  - Playing the output wav signal in case of decreasing the output sample rate to half:

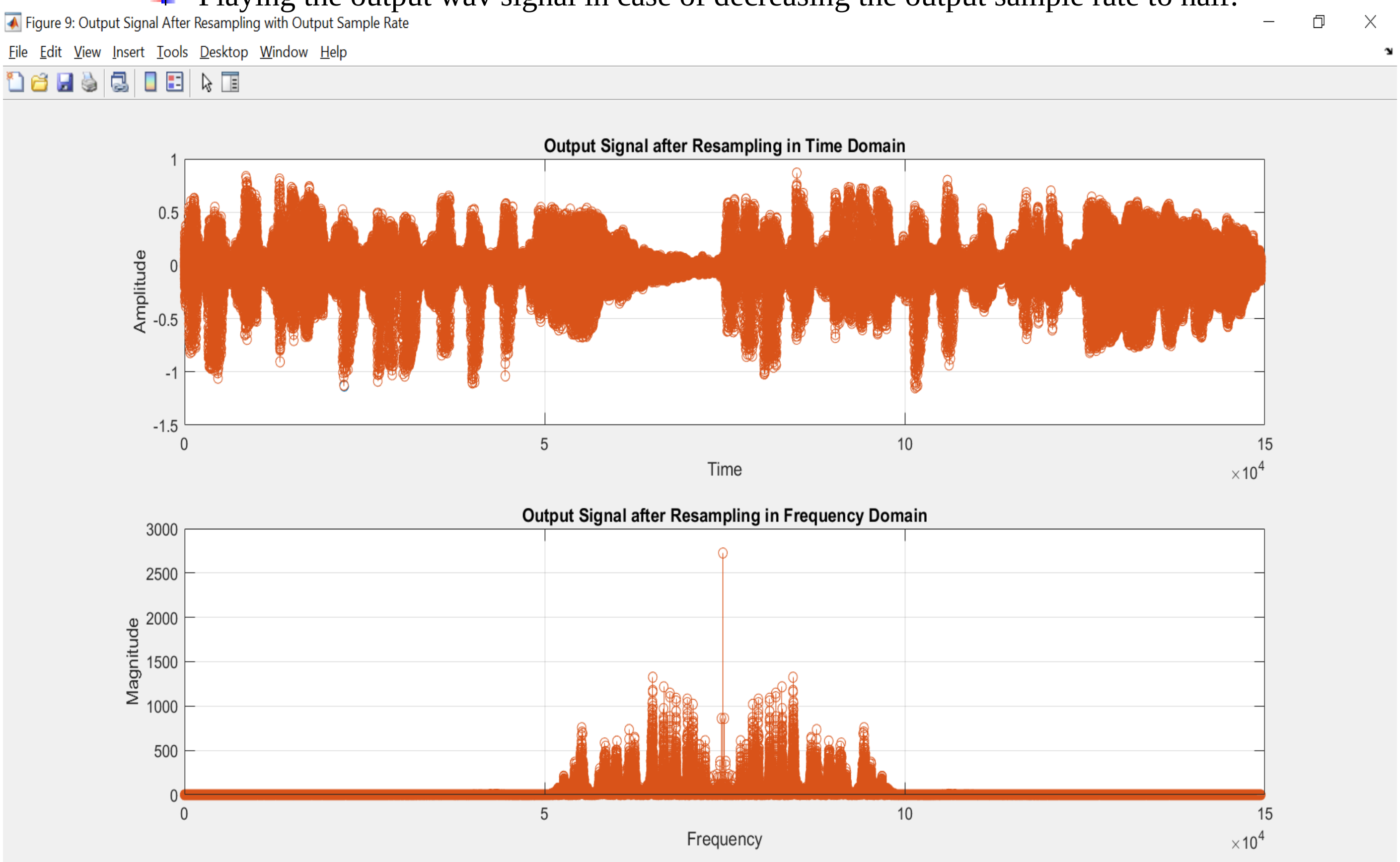


Figure 2.b.33

  - Note that the output wav signal has been resampled with decreasing the output sample rate to half and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with decreasing the output sample rate to half with those before resampling (Mentioned above).

➢ Applying the test2 wav file to the filters of type IIR:

- The user interface is as shown in the figure below *(Figure 2.b.34):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test2.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): IIR
-----------------------------------
Enter the output sample rate: 44100
-----------------------------------
Enter the gain for filter (1) in dB: 20
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 20
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 20
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.b.34

- The output signals in time domain and frequency domain:

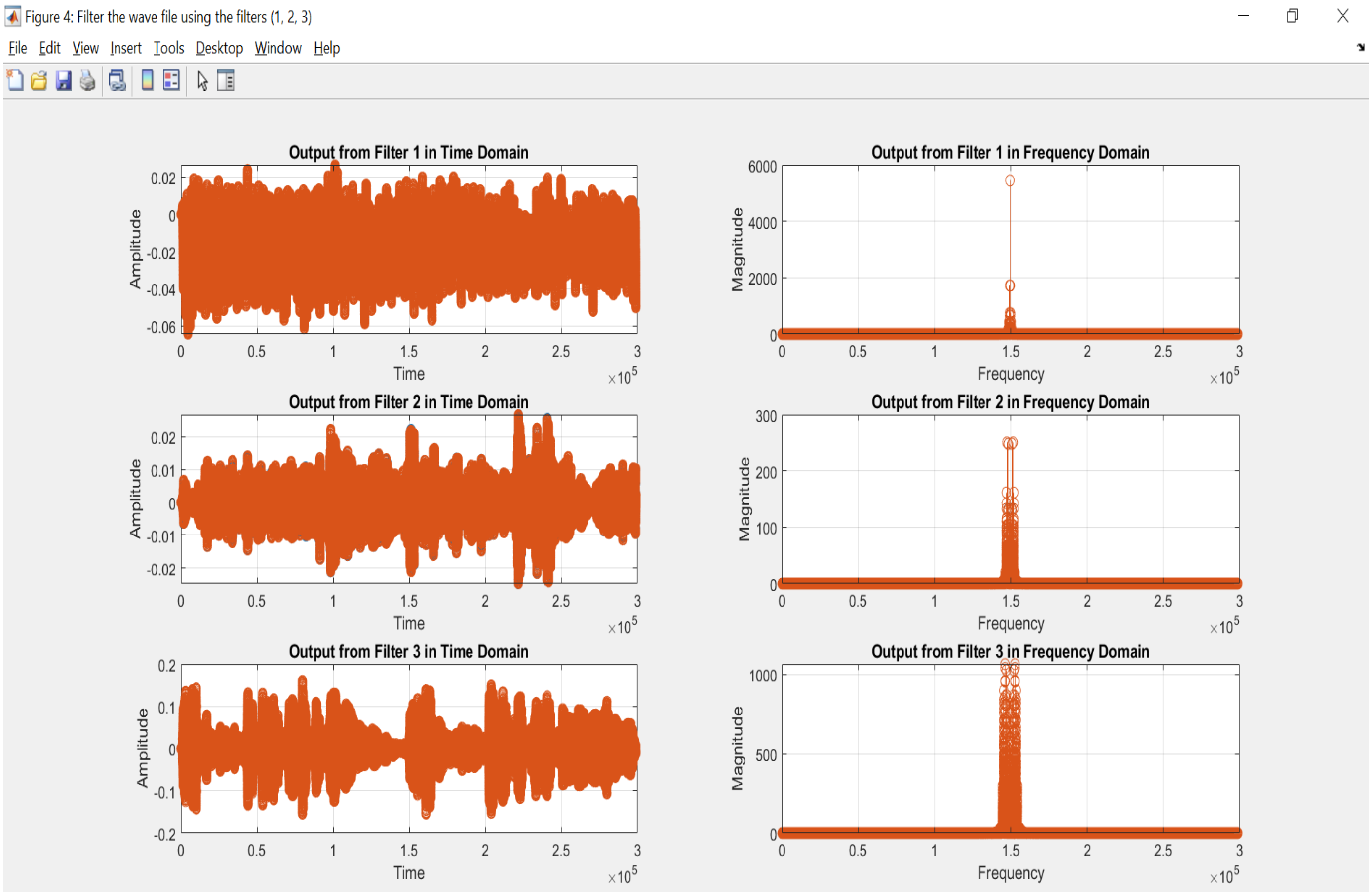


Figure 2.b.35

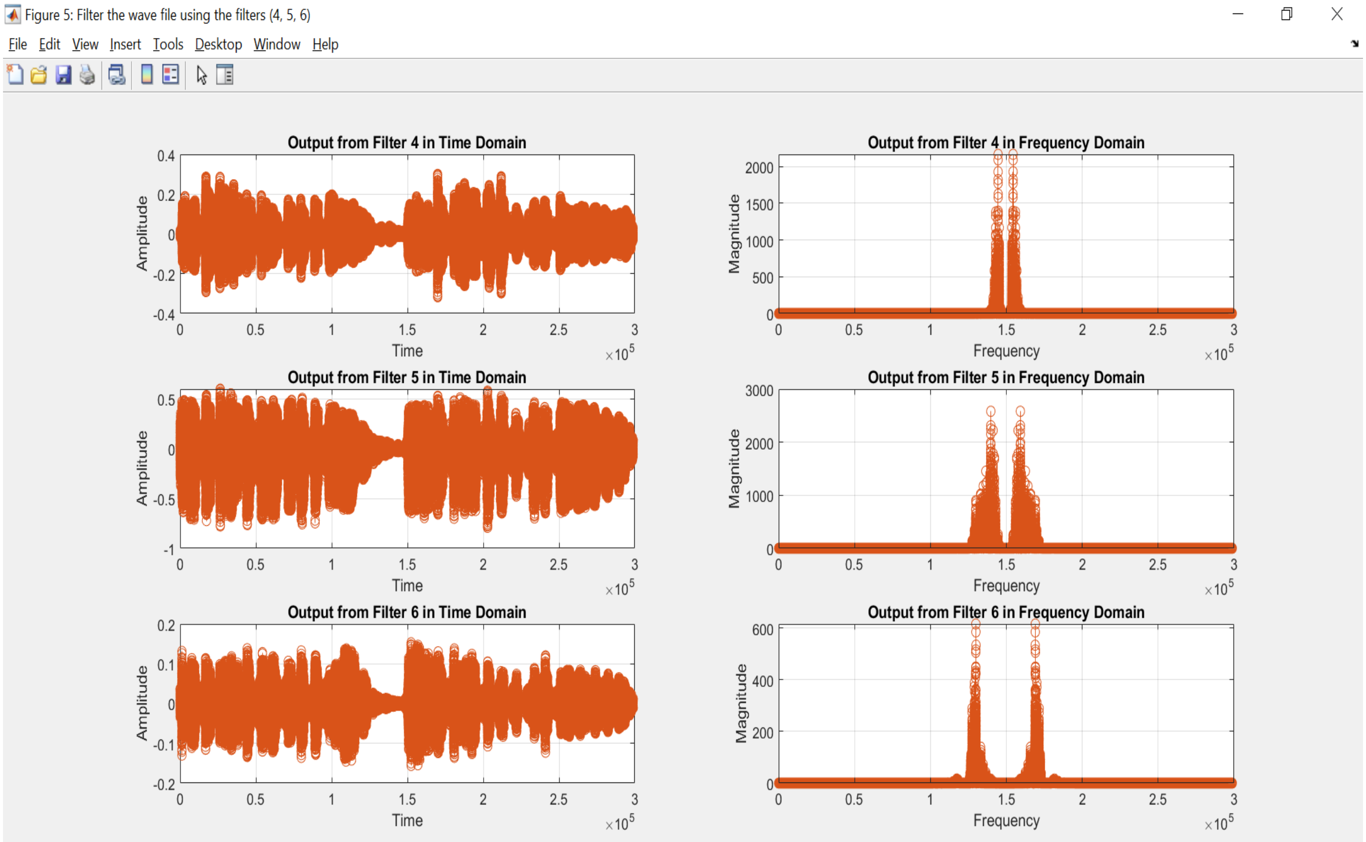


Figure 2.b.36

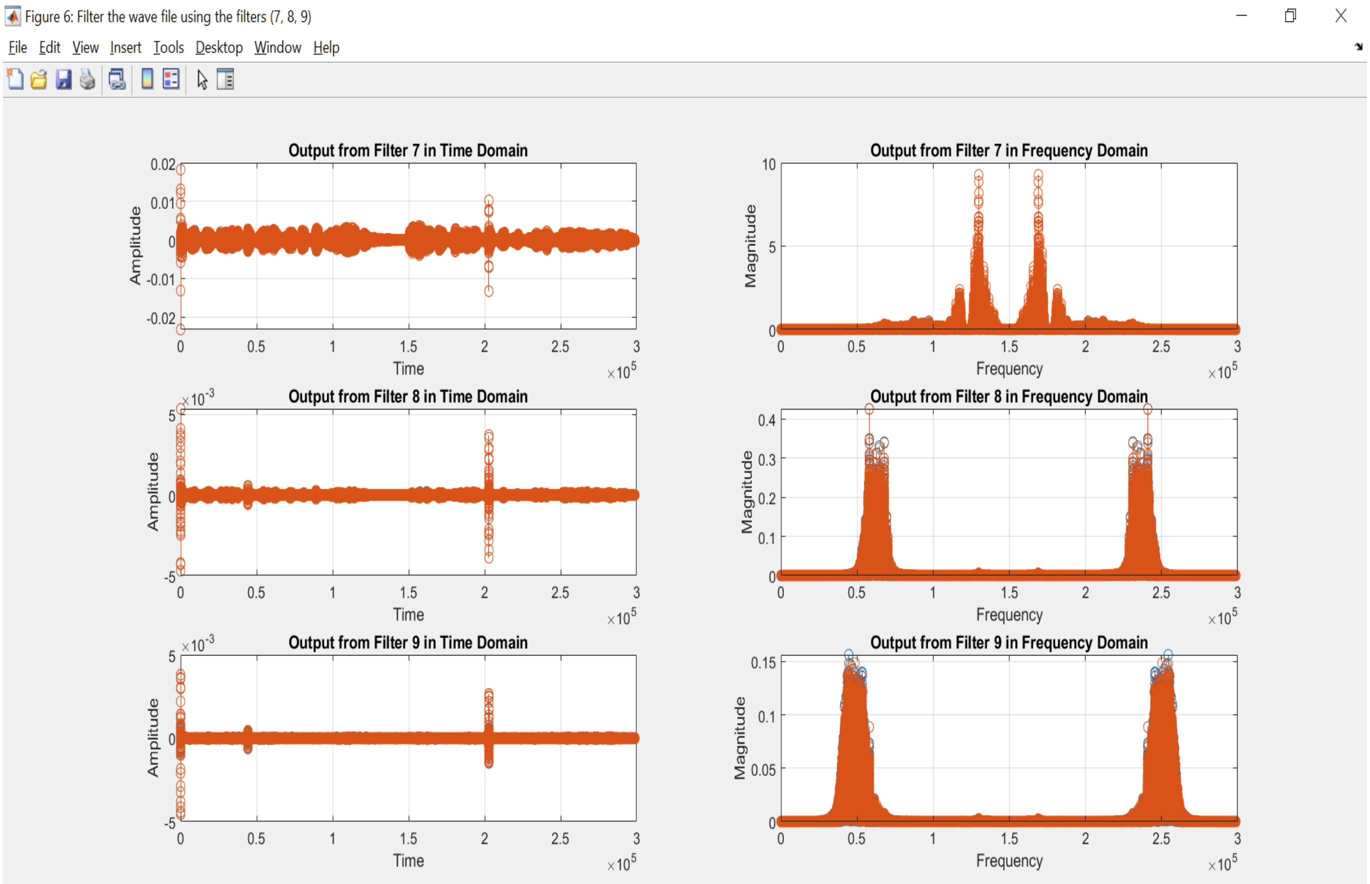


Figure 2.b.37

- Comparison between the Composite Signal amplified with the gains showed in the figure above *(Figure 2.b.34)* with the Original Signal in Time Domain:

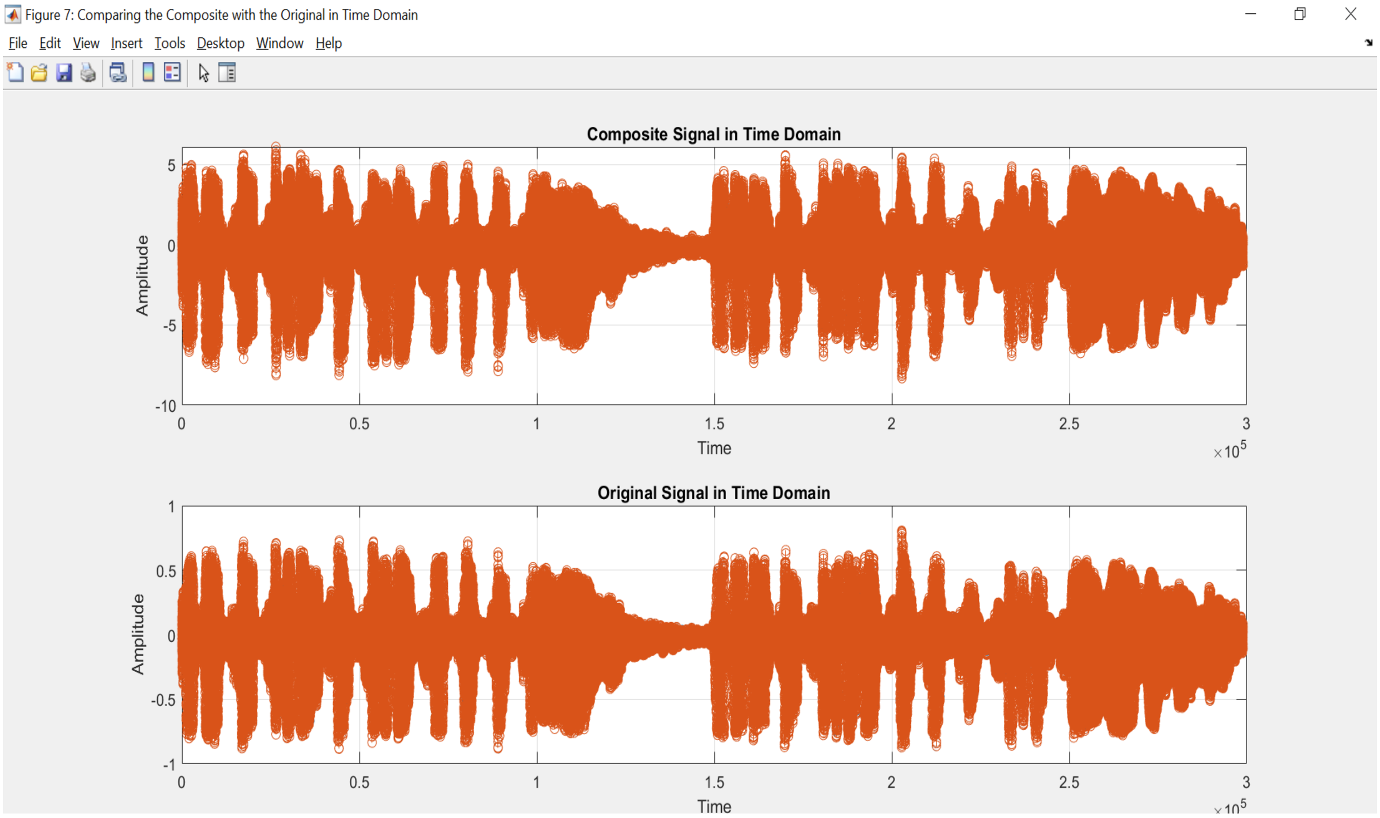


Figure 2.b.38

- Comparison between the Composite Signal amplified with the gains showed in the figure above *(Figure 2.b.34)* with the Original Signal in Frequency Domain:

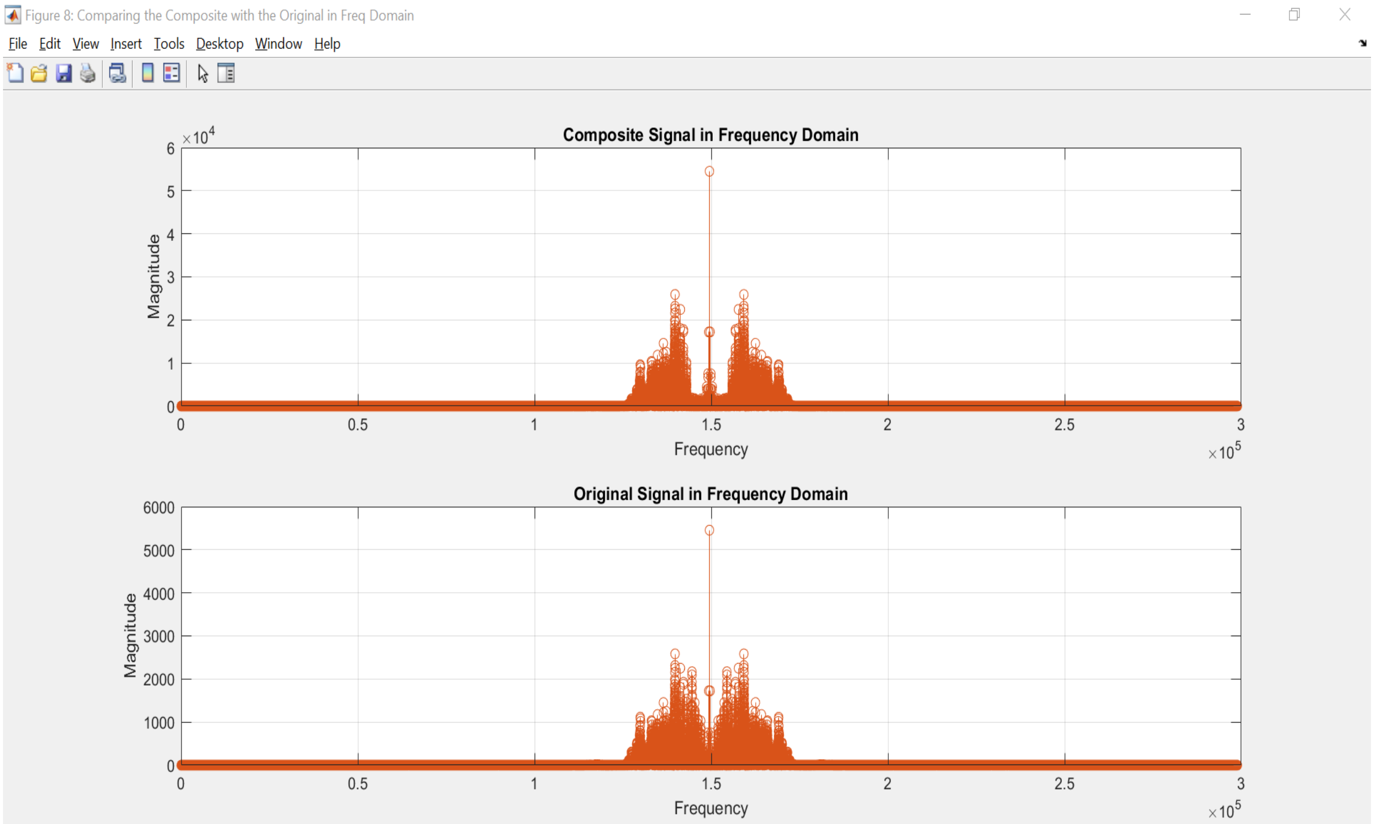


Figure 2.b.39

- Playing the output wav signal with the desired output sample rate:

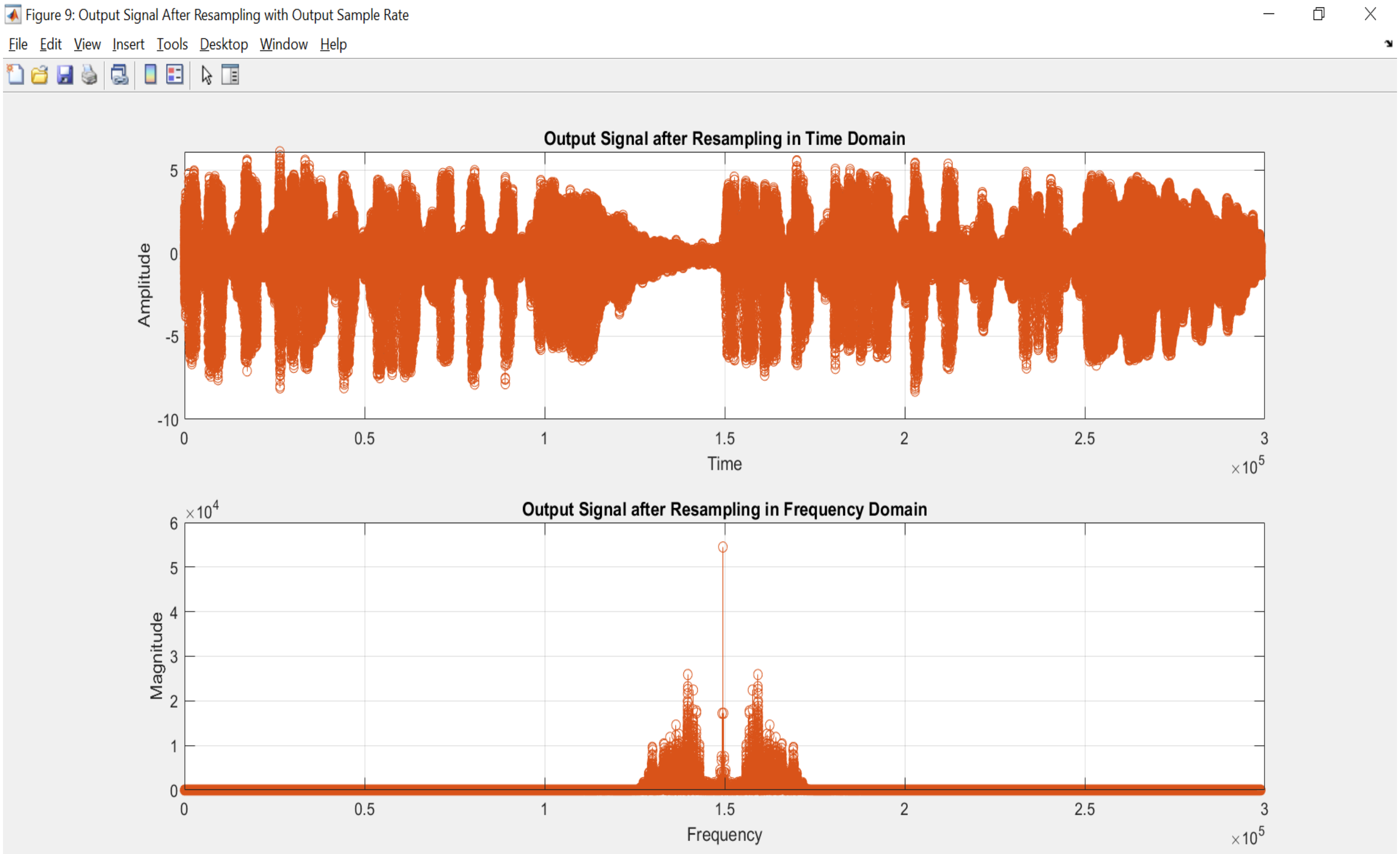


Figure 2.b.40

- Note that the output wav signal has been played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal.

➢ Output signal in case if doubling output sample rate:

- The user interface is as shown in the figure below *(Figure 2.b.41):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test2.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): IIR
-----------------------------------
Enter the output sample rate: 88200
-----------------------------------
Enter the gain for filter (1) in dB: 20
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 20
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 20
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.b.41

- Playing the output wav signal in case of doubling the output sample rate:

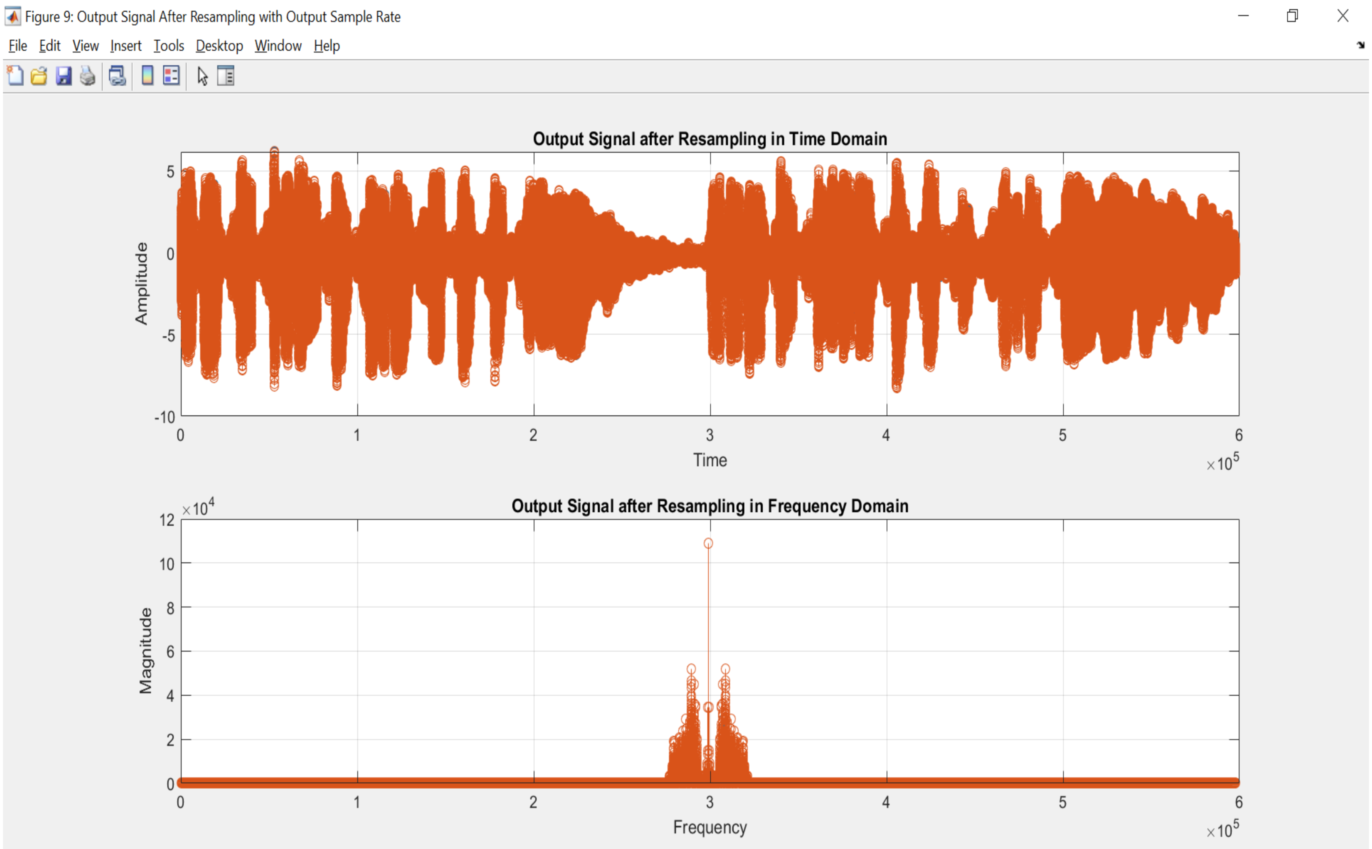


Figure 2.b.42

- Note that the output wav signal has been resampled with the double output sample rate and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with double output sample rate with those before resampling (Mentioned above).

- Output signal in case if decreasing output sample rate to half:
  - The user interface is as shown in the figure below *(Figure 2.b.43):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test2.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): IIR
-----------------------------------
Enter the output sample rate: 22050
-----------------------------------
Enter the gain for filter (1) in dB: 20
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 20
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 20
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.b.43

  - Playing the output wav signal in case of decreasing the output sample rate to half:

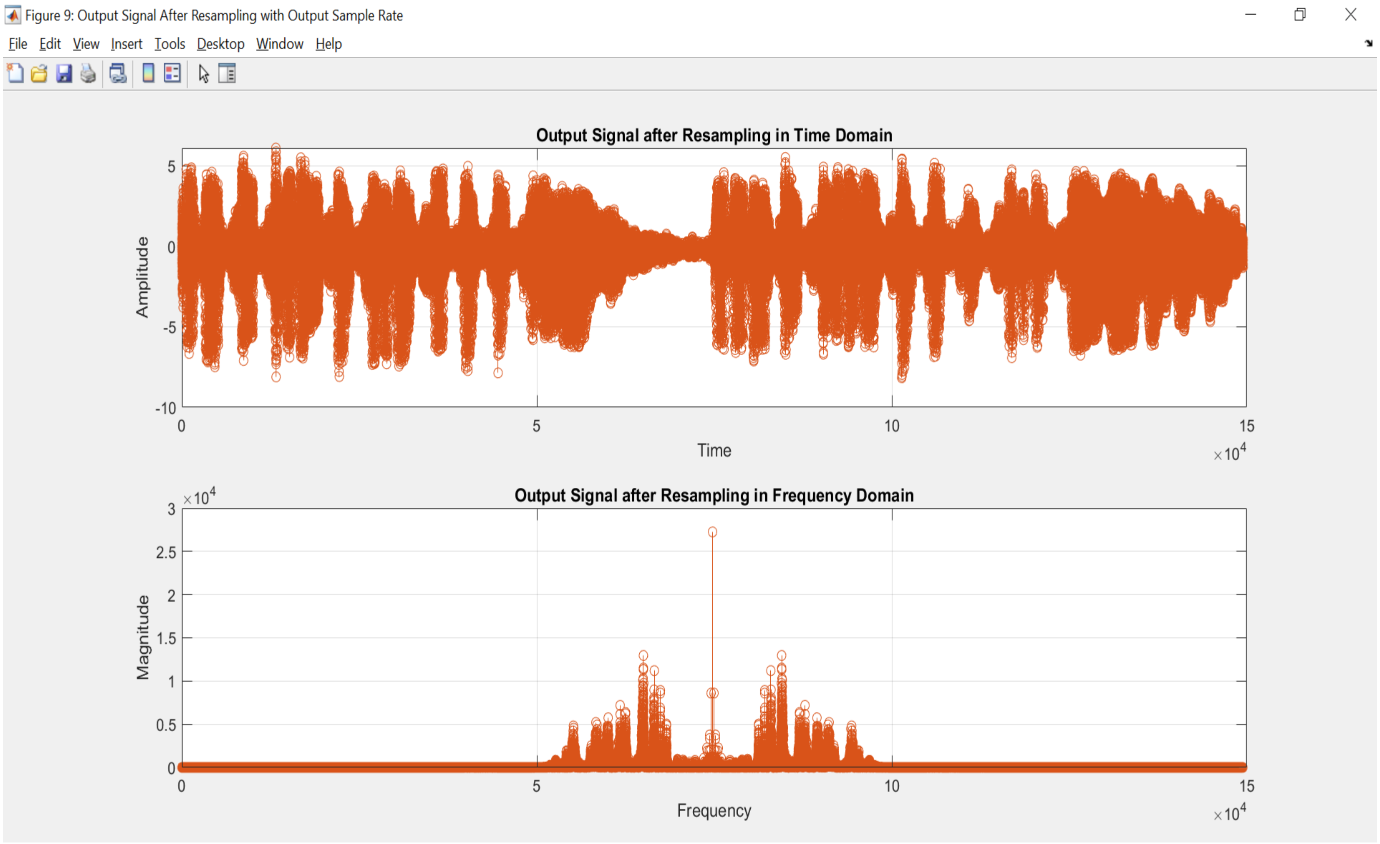


Figure 2.b.44

  - Note that the output wav signal has been resampled with decreasing the output sample rate to half and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with decreasing the output sample rate to half with those before resampling (Mentioned above).

➢ Applying the test3 wav file to the filters of type IIR:

- The user interface is as shown in the figure below *(Figure 2.b.45):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
---------------------------------
Enter the wave file name: test3.wav
---------------------------------
Enter the type of filters used (IIR-FIR): IIR
---------------------------------
Enter the output sample rate: 44100
---------------------------------
Enter the gain for filter (1) in dB: 0
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 0
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 0
---------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.b.45

The output signals in time domain and frequency domain:

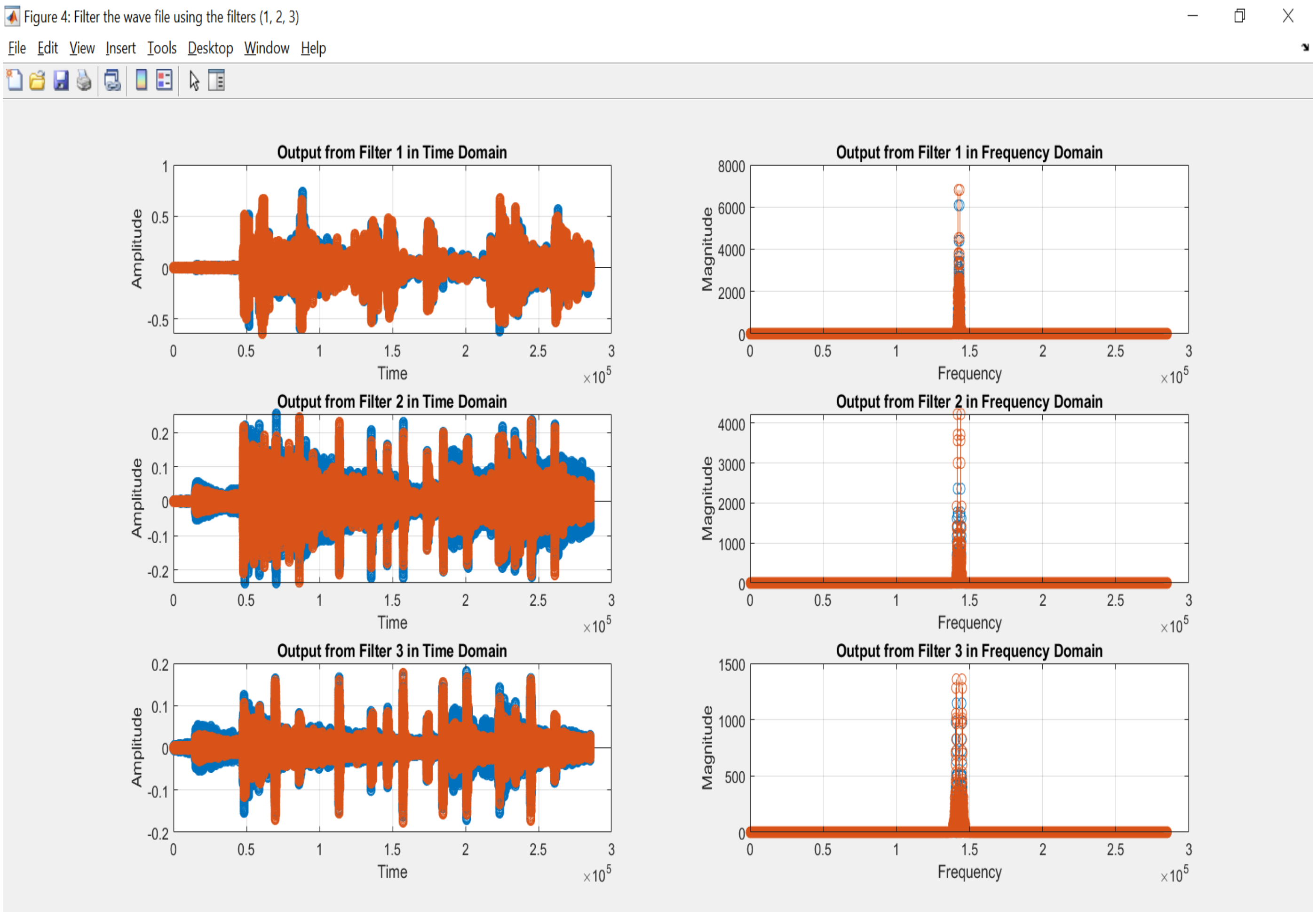


Figure 2.b.46

Figure 2.b.47

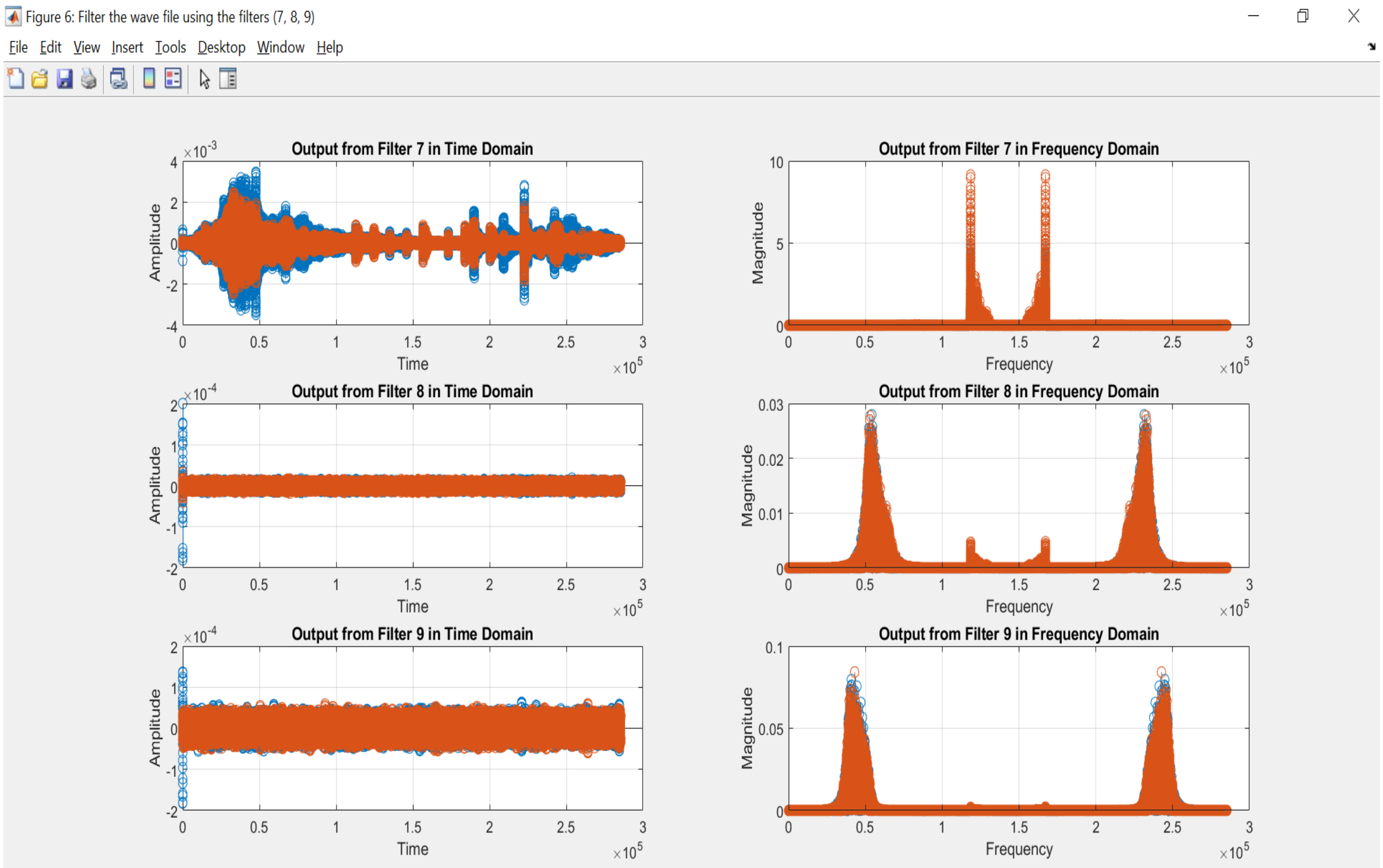


Figure 2.b.48

- Comparison between the Composite Signal amplified with gain (0 dB ➔ 1) with the Original Signal in Time Domain:

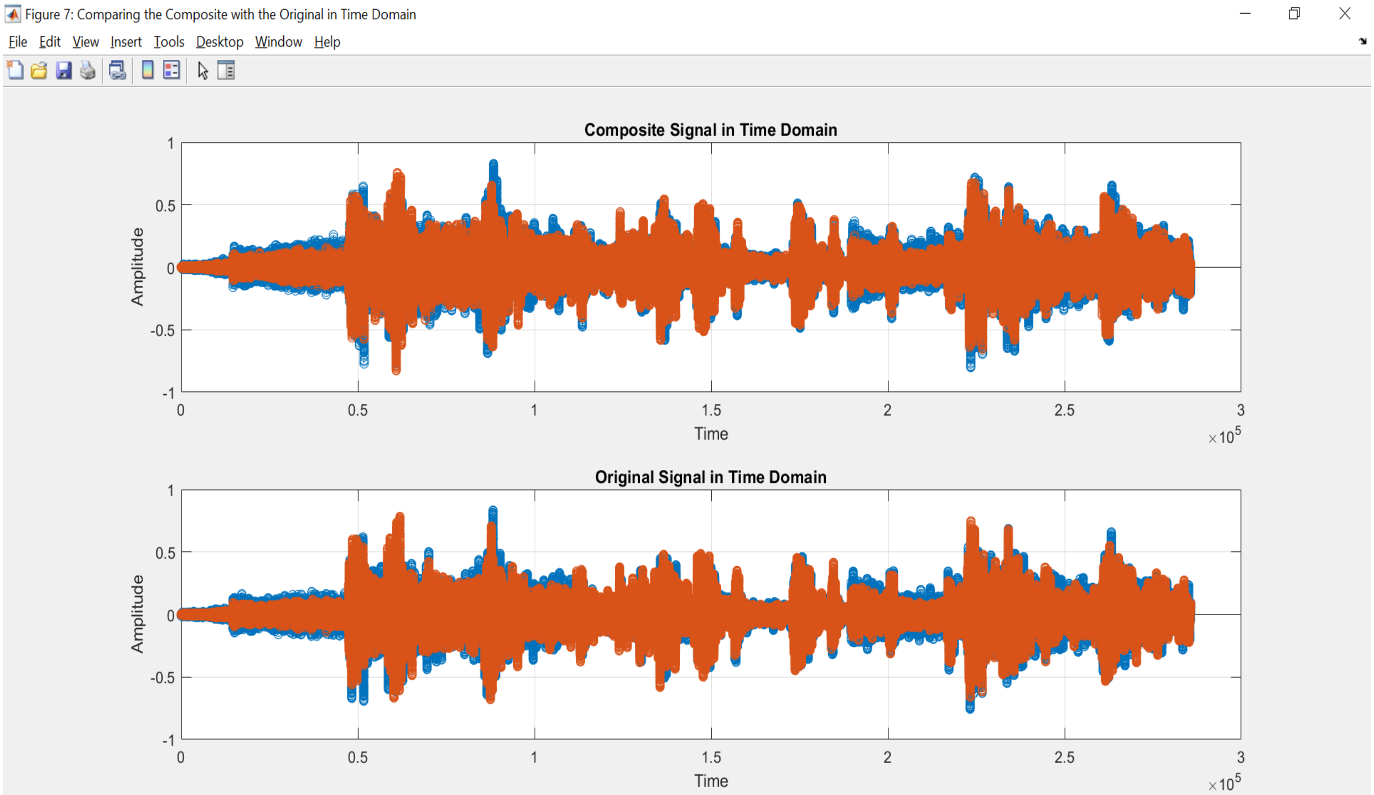


Figure 2.b.49

- Comparison between the Composite Signal amplified with gain (0 dB ➔ 1) with the Original Signal in Frequency Domain:

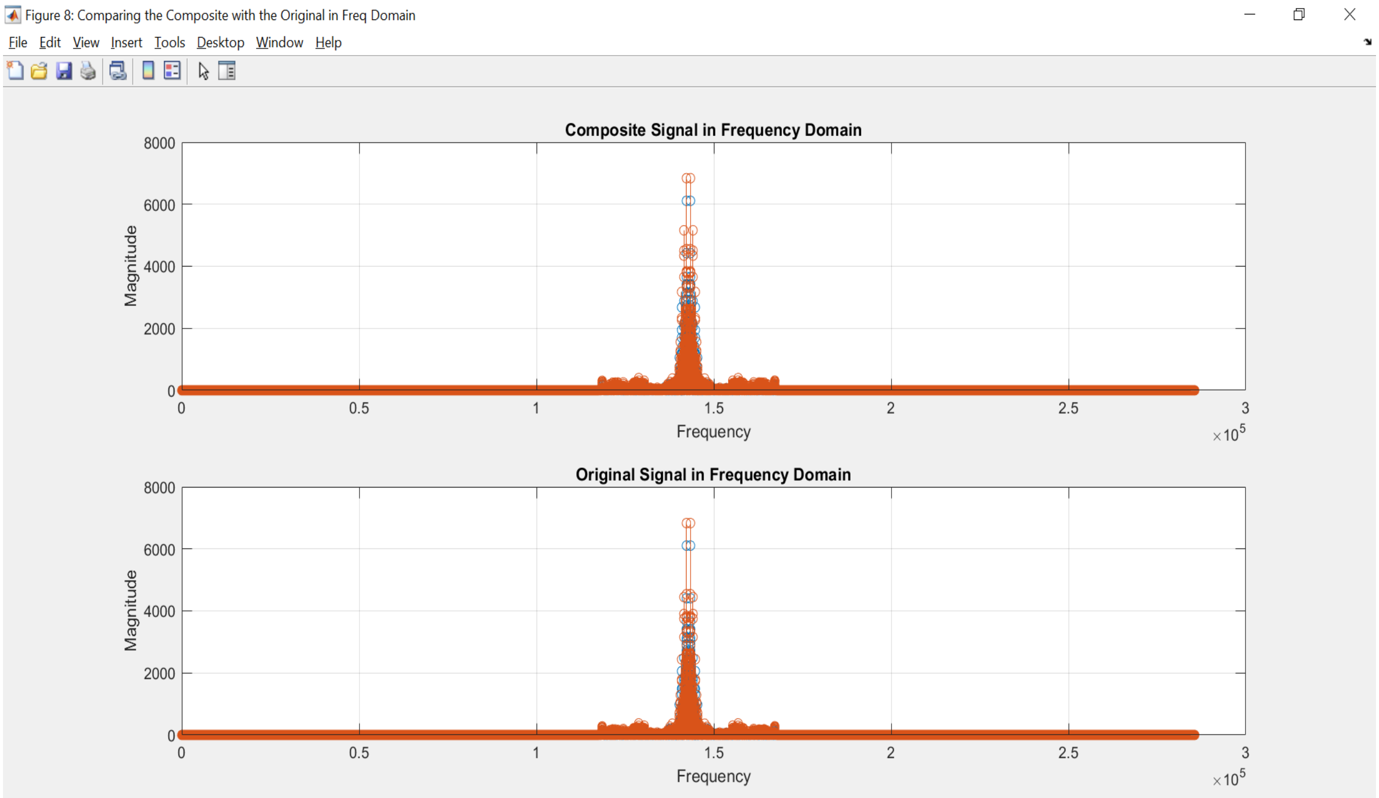


Figure 2.b.50

Playing the output wav signal with the desired output sample rate:

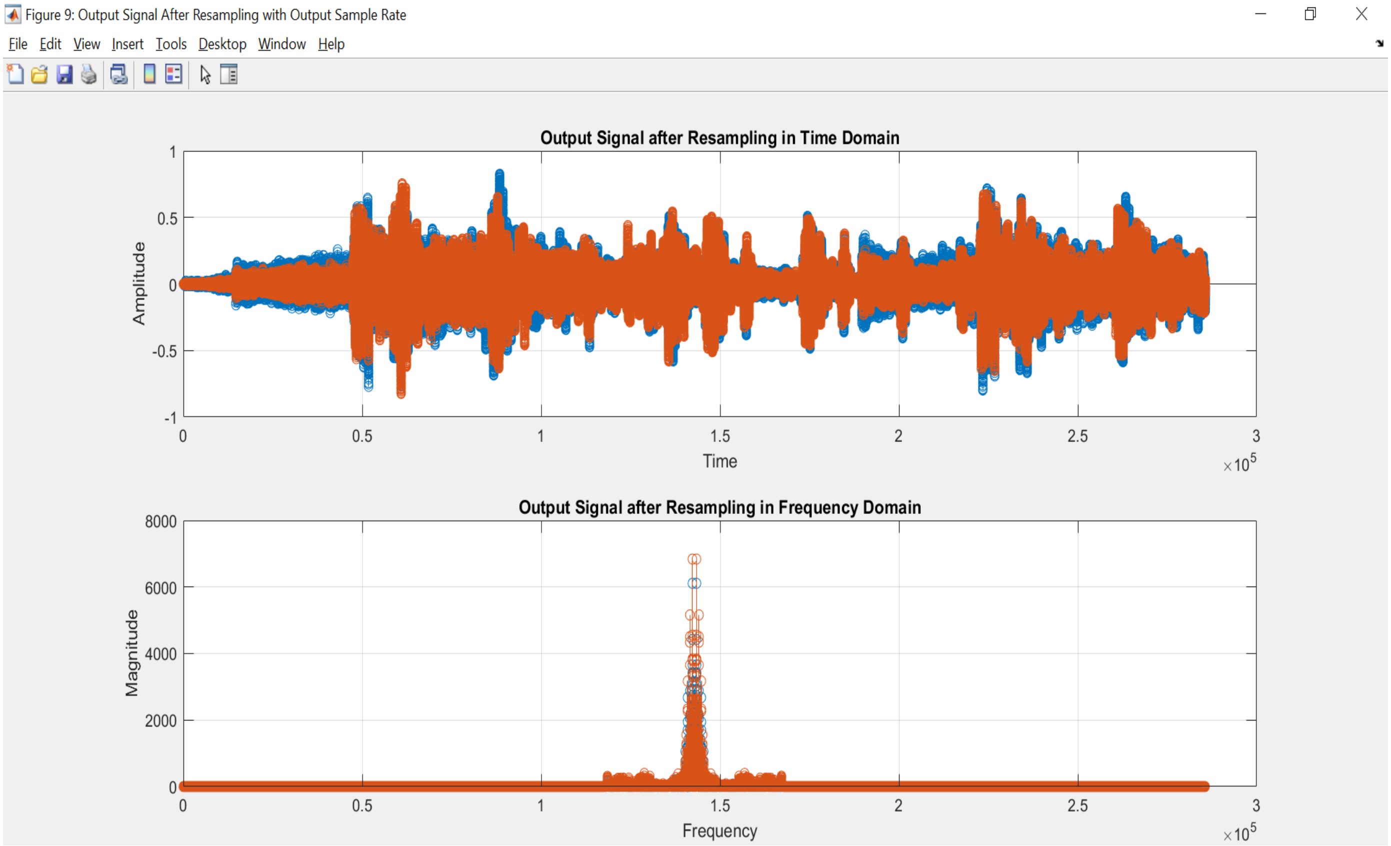


Figure 2.b.51

Note that the output wav signal has been played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal.

➢ Output signal in case if doubling output sample rate:

- The user interface is as shown in the figure below *(Figure 2.b.52):*

```
Command Window
*****************WELCOME TO AUDIO EQUALIZER*****************
-----------------------------------
Enter the wave file name: test3.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): IIR
-----------------------------------
Enter the output sample rate: 88200
-----------------------------------
Enter the gain for filter (1) in dB: 0
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 0
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 0
-----------------------------------
*****************THANK YOU*****************
fx >>
```

Figure 2.b.52

- Playing the output wav signal in case of doubling the output sample rate:

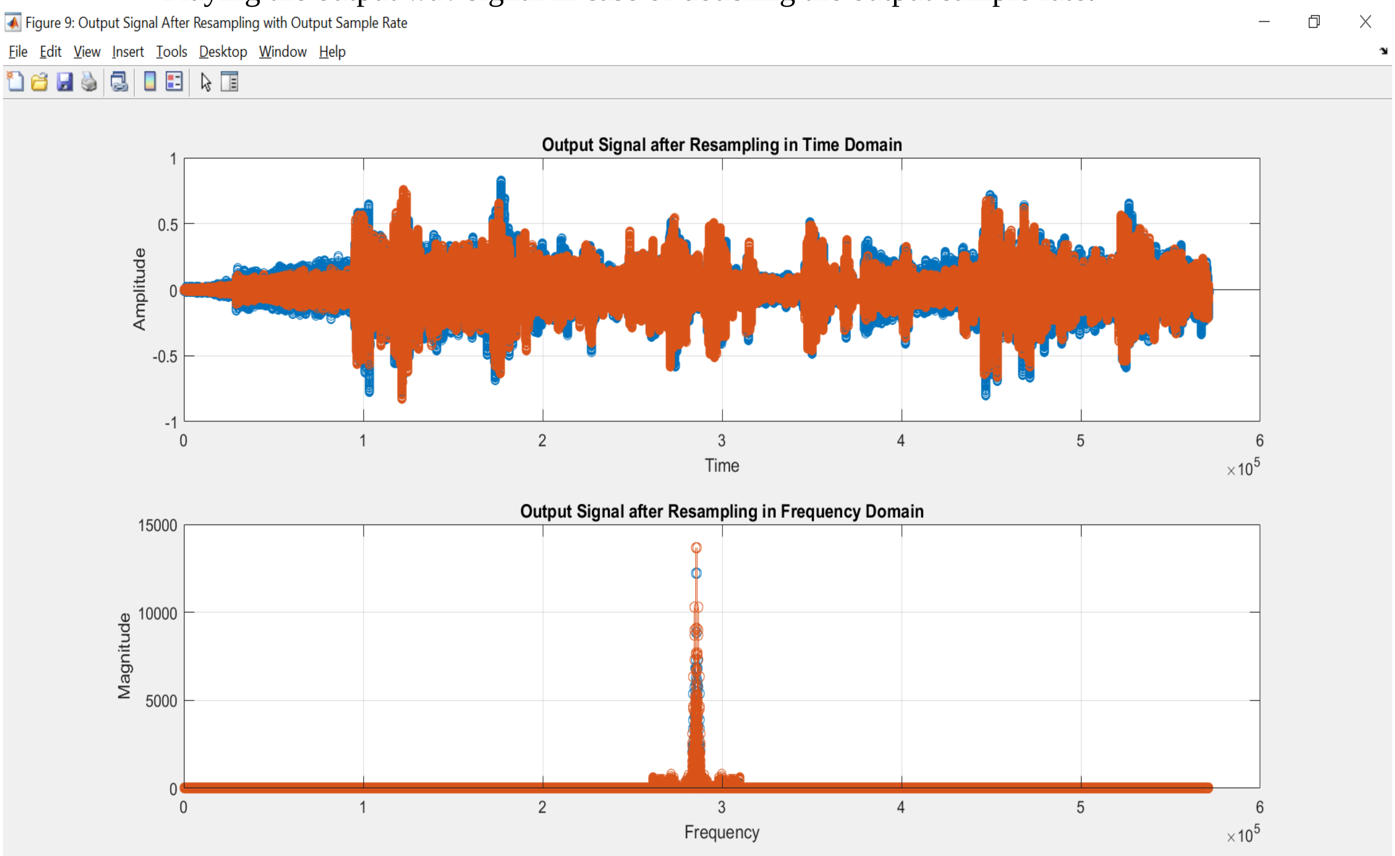


Figure 2.b.53

- Note that the output wav signal has been resampled with the double output sample rate and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with double output sample rate with those before resampling (Mentioned above).

➢ Output signal in case if decreasing output sample rate to half:

- The user interface is as shown in the figure below *(Figure 2.b.54):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test3.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): IIR
-----------------------------------
Enter the output sample rate: 22050
-----------------------------------
Enter the gain for filter (1) in dB: 0
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 0
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 0
-----------------------------------
****************THANK YOU****************
fx >> |
```

Figure 2.b.54

- Playing the output wav signal in case of decreasing the output sample rate to half:

Figure 2.b.55

- Note that the output wav signal has been resampled with decreasing the output sample rate to half and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with decreasing the output sample rate to half with those before resampling (Mentioned above).

- Applying the test3 wav file to the filters of type IIR:
  - The user interface is as shown in the figure below *(Figure 2.b.56):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test3.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): IIR
-----------------------------------
Enter the output sample rate: 44100
-----------------------------------
Enter the gain for filter (1) in dB: 20
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 20
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 20
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.b.56

  - The output signals in time domain and frequency domain:

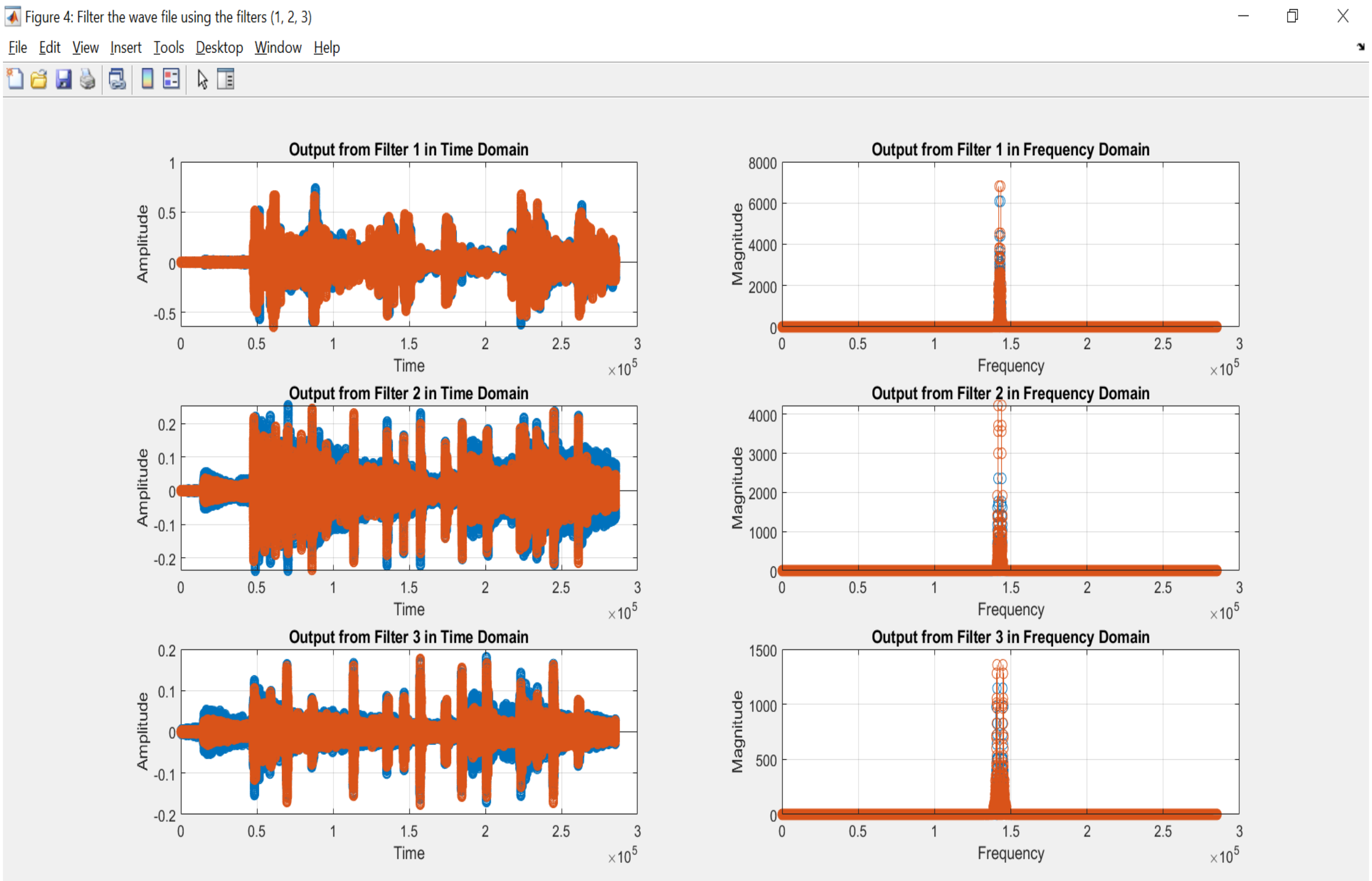


Figure 2.b.57

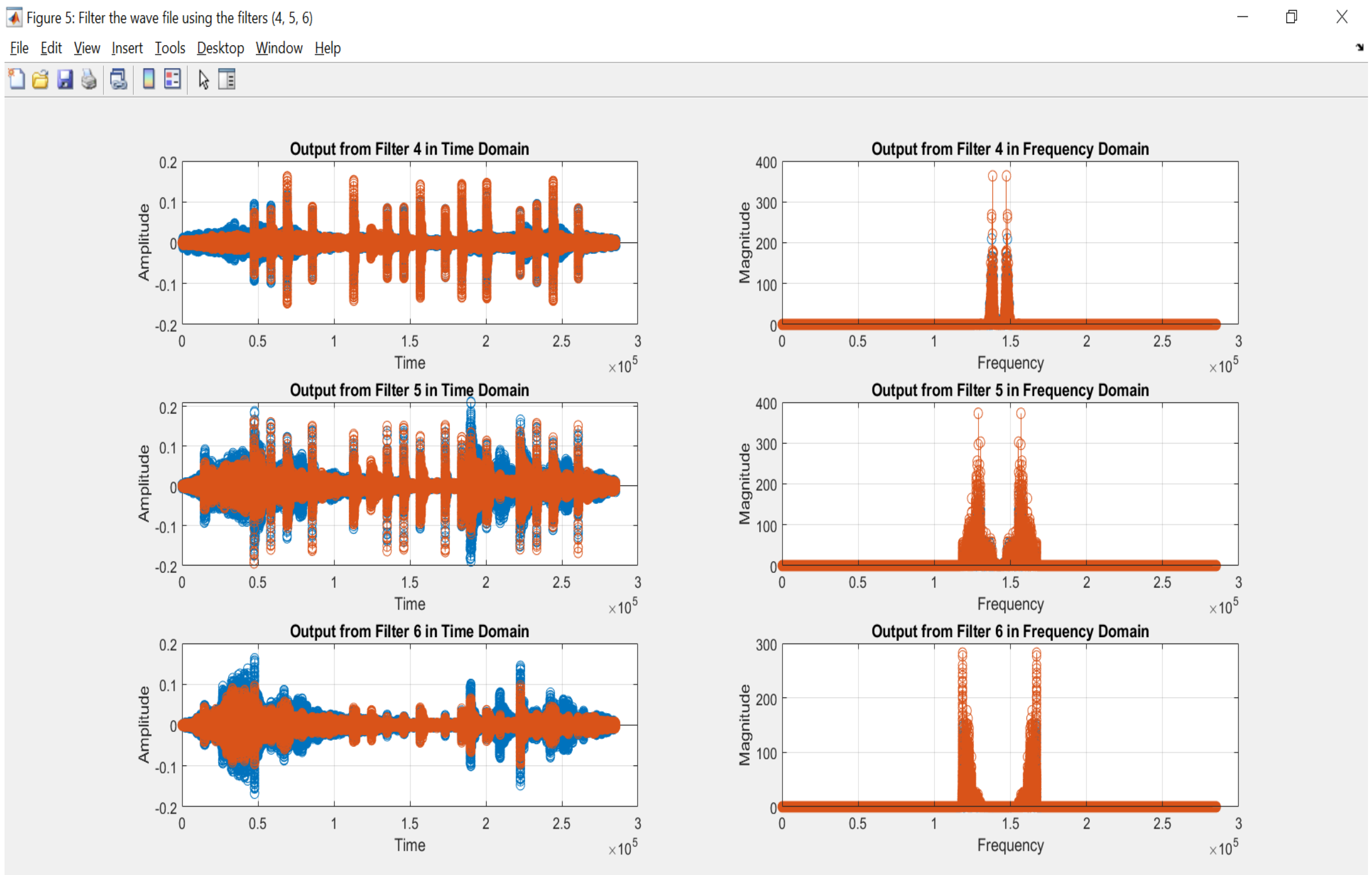


Figure 2.b.58

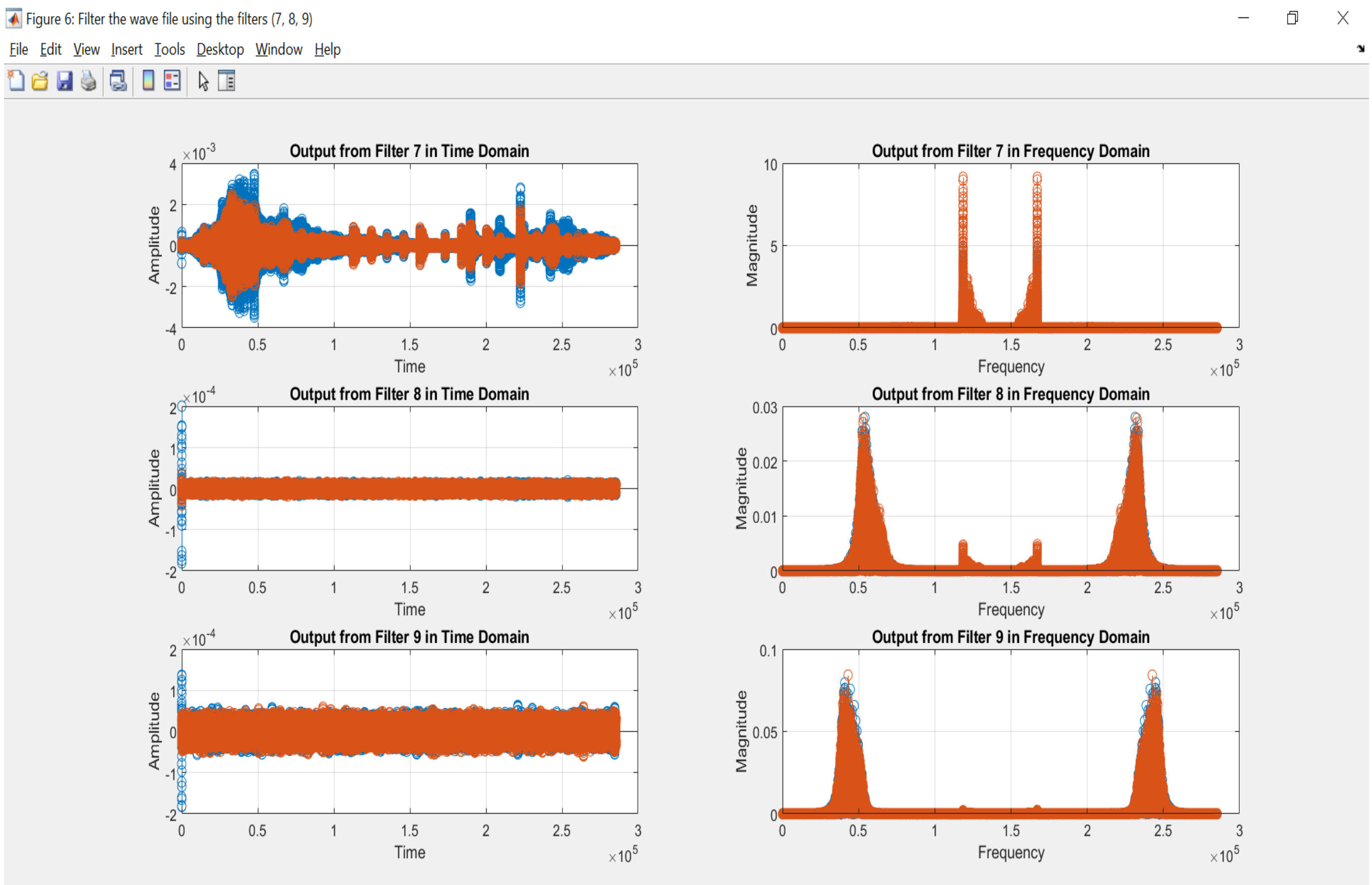


Figure 2.b.59

- Comparison between the Composite Signal amplified with the gains showed in the figure above *(Figure 2.b.56)* with the Original Signal in Time Domain:

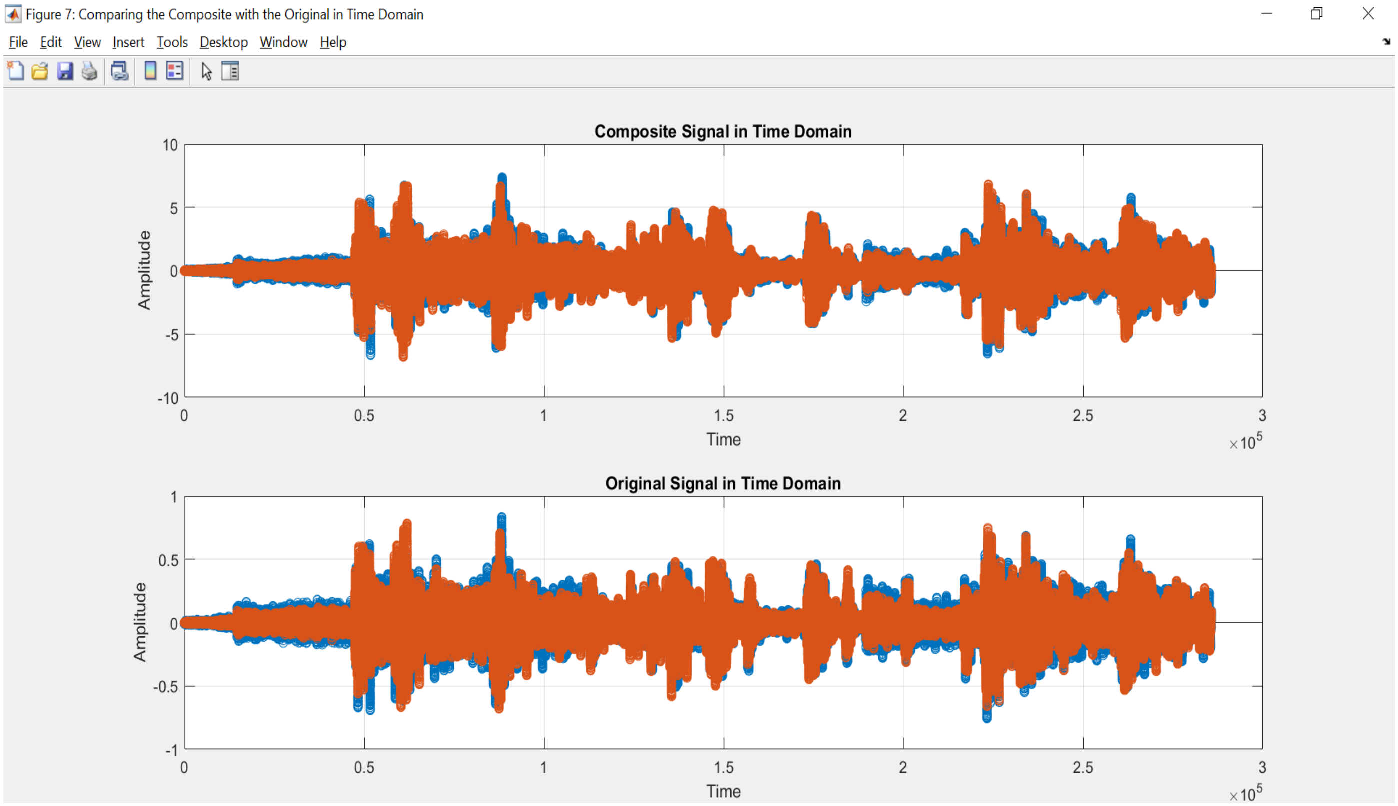


Figure 2.b.60

Comparison between the Composite Signal amplified with the gains showed in the figure above *(Figure 2.b.56)* with the Original Signal in Frequency Domain:

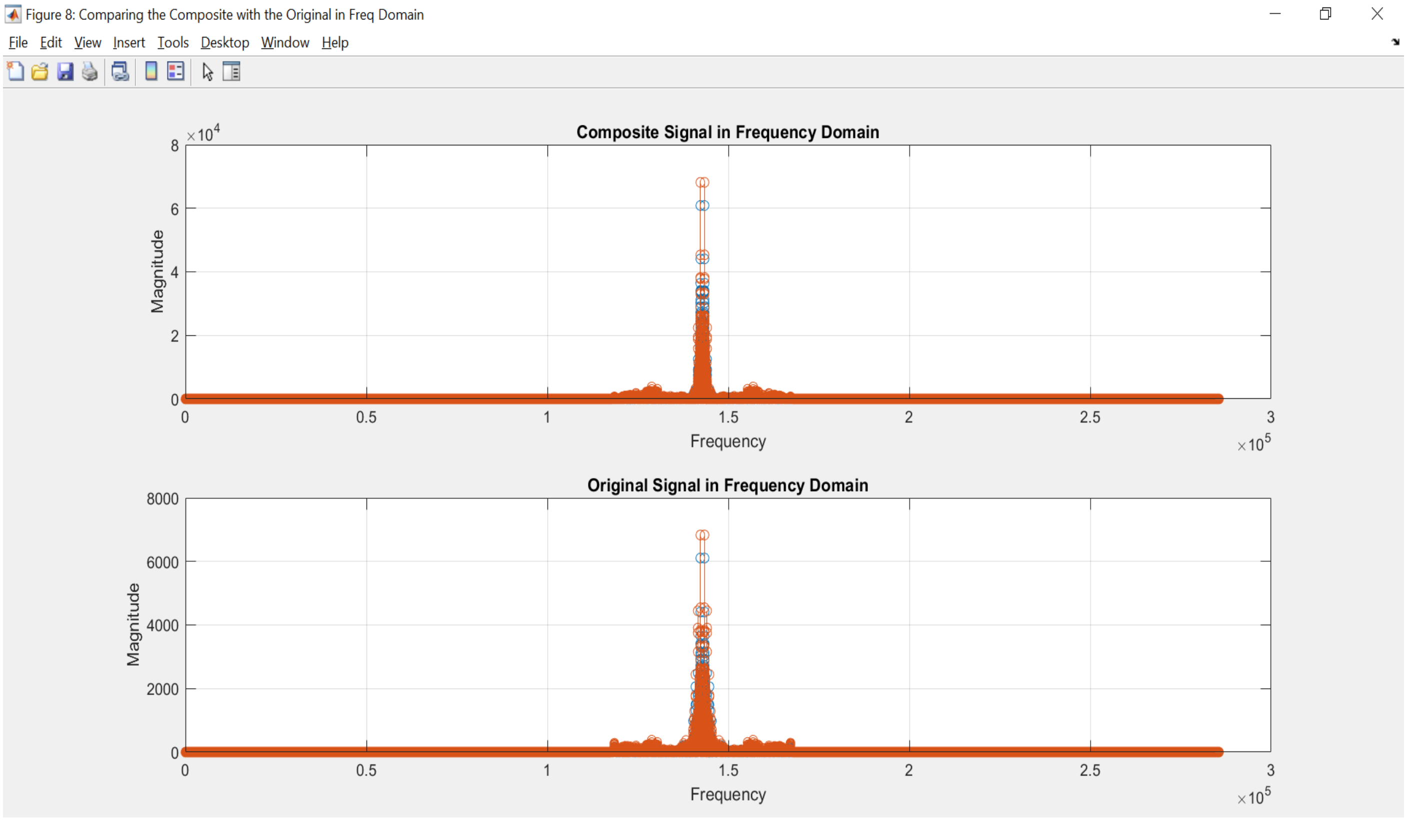


Figure 2.b.61

- Playing the output wav signal with the desired output sample rate:

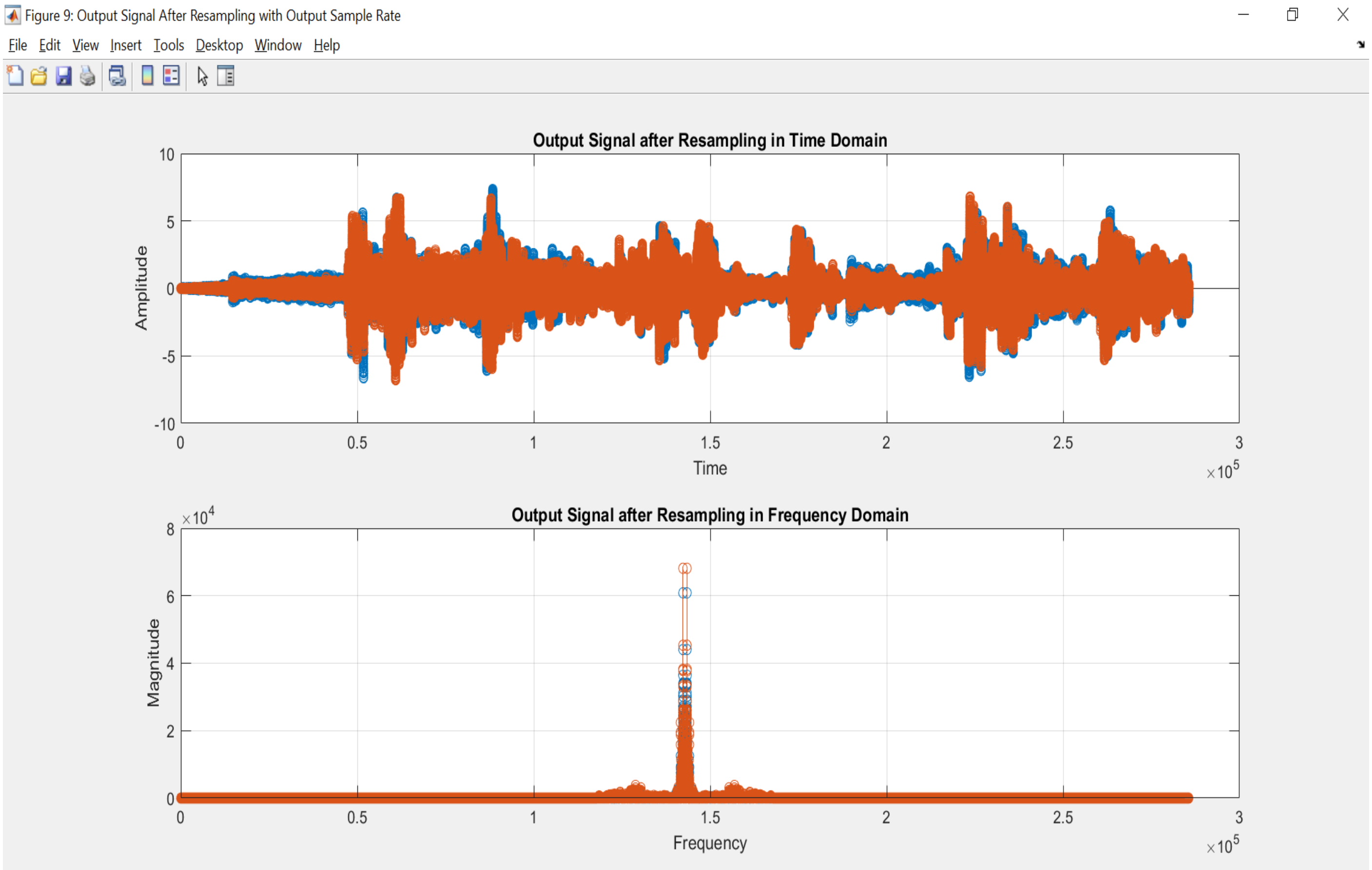


Figure 2.b.62

- Note that the output wav signal has been played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal.

➢ Output signal in case if doubling output sample rate:

- The user interface is as shown in the figure below *(Figure 2.b.63):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test3.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): IIR
-----------------------------------
Enter the output sample rate: 88200
-----------------------------------
Enter the gain for filter (1) in dB: 20
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 20
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 20
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.b.63

- Playing the output wav signal in case of doubling the output sample rate:

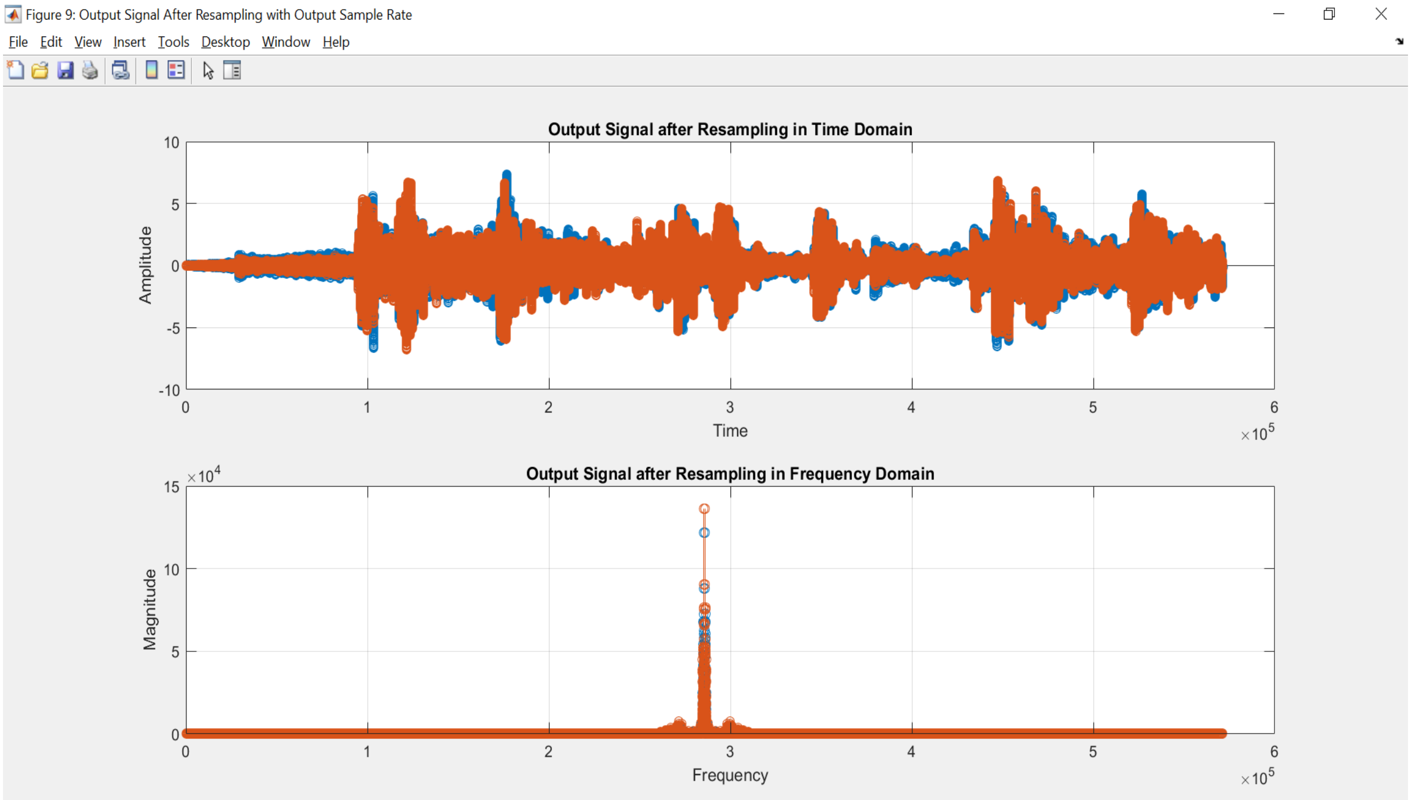


Figure 2.b.64

- Note that the output wav signal has been resampled with the double output sample rate and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with double output sample rate with those before resampling (Mentioned above).

- Output signal in case if decreasing output sample rate to half:
  - The user interface is as shown in the figure below *(Figure 2.b.65):*

```
Command Window
****************WELCOME TO AUDIO EQUALIZER****************
-----------------------------------
Enter the wave file name: test3.wav
-----------------------------------
Enter the type of filters used (IIR-FIR): IIR
-----------------------------------
Enter the output sample rate: 22050
-----------------------------------
Enter the gain for filter (1) in dB: 20
Enter the gain for filter (2) in dB: 0
Enter the gain for filter (3) in dB: 0
Enter the gain for filter (4) in dB: 0
Enter the gain for filter (5) in dB: 20
Enter the gain for filter (6) in dB: 0
Enter the gain for filter (7) in dB: 0
Enter the gain for filter (8) in dB: 0
Enter the gain for filter (9) in dB: 20
-----------------------------------
****************THANK YOU****************
fx >>
```

Figure 2.b.65

  - Playing the output wav signal in case of decreasing the output sample rate to half:

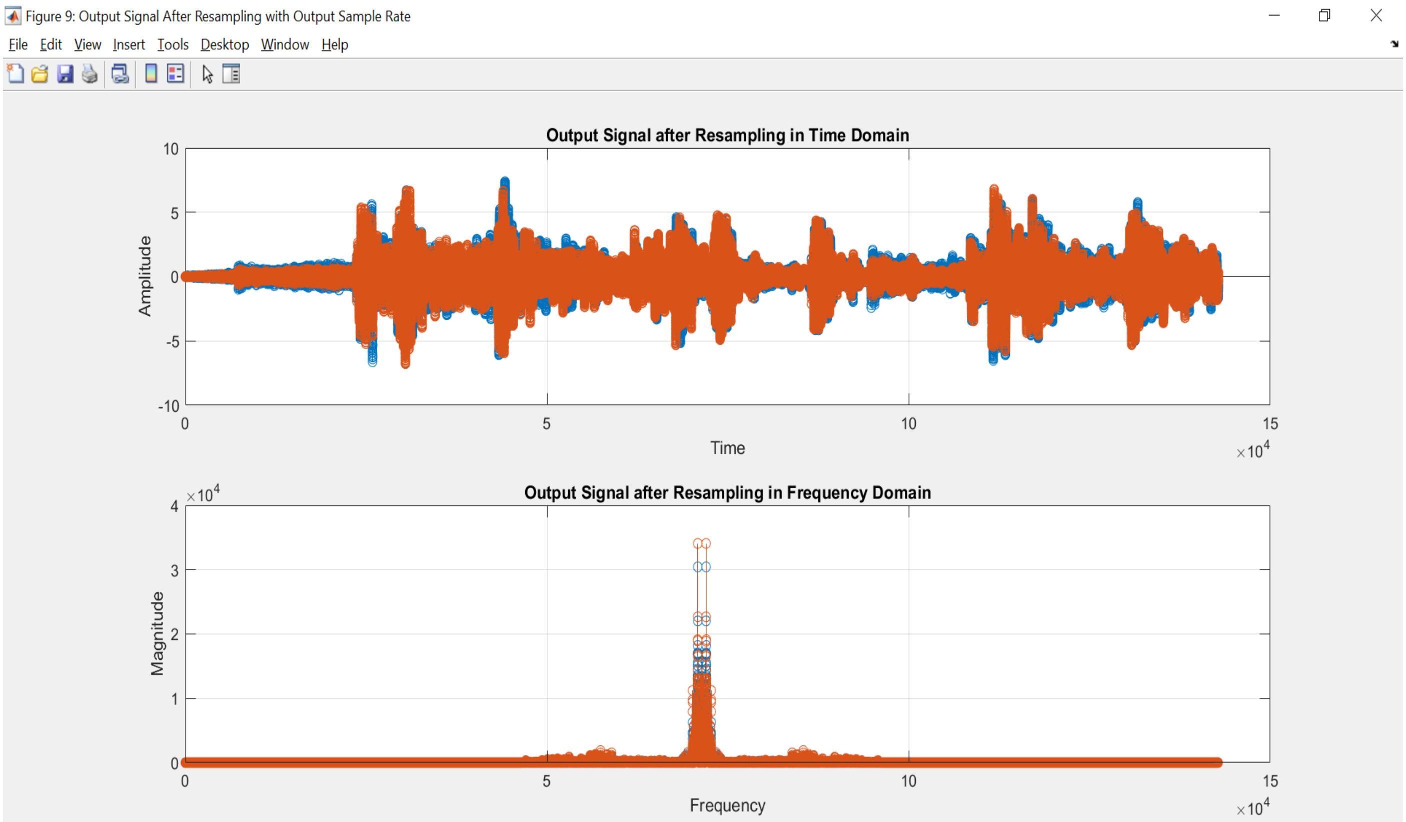


Figure 2.b.66

  - Note that the output wav signal has been resampled with decreasing the output sample rate to half and played in MATLAB Software using the sound() command, so to enjoy listening to the output wav signal, you can run the implemented Audio Equalizer with your desired song or any wav file, and enjoy playing and listening to the output wav signal. Also you can compare the output wav signal after resampling with decreasing the output sample rate to half with those before resampling (Mentioned above).

Conclusion:

In this Final Project, it is required to develop an Audio Equalizer using MATLAB Software. The main function of the Audio Equalizer is to vary the gain of each specific band as the user prefers. If the user likes base he will increase the gain of the low frequencies. In this Final Project (Audio Equalizer), it is expected from the user to input the following: (The wave file name, Gain of each of the frequency band Filters in dB, Type of filters used (FIR-IIR), Output sample rate). In the main implementation of the Audio Equalizer it is required to develop and implement the frequency band filters in the following bands: [filter 1: (0-170 Hz), filter 2: (170-310 Hz), filter 3: (310-600 Hz), filter 4: (600-1000 Hz), filter 5: (1-3 KHz), filter 6: (3-6 KHz), filter 7: (6-12 KHz), filter 8: (12-14 KHz), filter 9: (14-16 KHz)]. It has been concluded that filter 1 is a Low Pass Filter, while filters (2, 3, 4, 5, 6, 7, 8, 9) are Band Bass Filters. After that it is required to analyse and export the following specifications of the developed filters: (magnitude, phase, impulse and step response, order, and gain/poles/zeros). Then Filter the wave file using the filters developed for both types (IIR, FIR). Then Draw the output signals from filters in Time and frequency domains. Then Amplify the output signals using the user defined gain. Then Add the amplified - output signals in time domain to form composite signal. Then Draw and compare the composite signal with the original signal in both domains (Time Domain and Frequency Domain). Then Play the output wave signal. In this project the output signal will be played and displayed on MATLAB Figures with the desired output sample rate entered by the user. It has been chosen three test wav files to be tested after implementing the project, also all the required specifications will be applied to those three test wav files. After that it is required to submit only ONE PDF REPORT containing: (Copy of Code, Different sample runs of code including the following cases: [If design is using FIR filter, if design is using IIR filters, Output signal in case if doubling output sample rate or decreasing it to half], the analysis of each filter and the exported outputs, All figures of signals in time and frequency domain). Finally, the Audio Equalizer has been implemented successfully with all the specifications and criteria required. Also in the PDF REPORT, it has been included all the required sample runs of the code including all the mentioned and required cases. In addition to, it has been included the analysis of each filter and the exported outputs for the both types (FIR – IIR) filters. Also, it has been included all the figures required, in addition to the figures of signals in both time domain and frequency domain. Also, it has been included a copy of all code, the code is more than well documented. Also, the code file has been uploaded to google drive with the link below, in addition to the (test1, test2, test3) wav files. In addition to a copy of the well documented code itself.

Source Code (.m File) Link

Final Project:
https://drive.google.com/file/d/1jvE63h03UbSPUWusHgMwnDKPFuMhfyAq/view?usp=sharing

test1 wav file:
https://drive.google.com/file/d/1yYmPM9RZq_IS2cwoLUJmcQOMh_pW7oo4/view?usp=sharing

test2 wav file:
https://drive.google.com/file/d/1-RGL20nCUkb_mK790eqNuw7cCVQE0v41/view?usp=sharing

test3 wav file:
https://drive.google.com/file/d/1_j9S34pXQLu0jFTgDyWsiH6ERvewABvV/view?usp=sharing

Source Code (.m File)

```
close all; clc; clear variables;
fprintf('****************WELCOME TO AUDIO EQUALIZER****************\n');
fprintf('-----------------------------------\n');
filename=input('Enter the wave file name: ','s');
[y,Fs]=audioread(filename);
fprintf('-----------------------------------\n');
type=input('Enter the type of filters used (IIR-FIR): ','s');
fprintf('-----------------------------------\n');
fouts=input('Enter the output sample rate: ');
fprintf('-----------------------------------\n');
G1=input('Enter the gain for filter (1) in dB: ');
G2=input('Enter the gain for filter (2) in dB: ');
G3=input('Enter the gain for filter (3) in dB: ');
G4=input('Enter the gain for filter (4) in dB: ');
G5=input('Enter the gain for filter (5) in dB: ');
G6=input('Enter the gain for filter (6) in dB: ');
G7=input('Enter the gain for filter (7) in dB: ');
G8=input('Enter the gain for filter (8) in dB: ');
G9=input('Enter the gain for filter (9) in dB: ');
fprintf('-----------------------------------\n');
%Converting the gain from (dB) to ordinary unit in order to be easily multiplied
G1=10^(G1/20);
G2=10^(G2/20);
G3=10^(G3/20);
G4=10^(G4/20);
G5=10^(G5/20);
G6=10^(G6/20);
G7=10^(G7/20);
G8=10^(G8/20);
G9=10^(G9/20);
%In Case the Type of filters used is (IIR)
if (strcmpi(type,'IIR'))==1
   n=4;    %Order of the used IIR Filters
   %Filter 1 (Low Pass Filter)
   Fc=170;
   [num1,den1]=butter(n,Fc/(Fs/2));
   [Hd1,wd1]=freqz(num1,den1);
   mag1=abs(Hd1);
   phase1=angle(Hd1)*180/pi;
   [z1,p1,k1]=butter(n,Fc/(Fs/2));
   sys1=tf(num1,den1);
   %Filter 2 (Band Pass Filter)
   fc=[170,310]/(Fs/2);
   [num2,den2]=butter(n,fc);
   [Hd2,wd2]=freqz(num2,den2);
   mag2=abs(Hd2);
   phase2=angle(Hd2)*180/pi;
   [z2,p2,k2]=butter(n,fc);
```

```
sys2=tf(num2,den2);
%Filter 3 (Band Pass Filter)
fc=[310,600]/(Fs/2);
[num3,den3]=butter(n,fc);
[Hd3,wd3]=freqz(num3,den3);
mag3=abs(Hd3);
phase3=angle(Hd3)*180/pi;
[z3,p3,k3]=butter(n,fc);
sys3=tf(num3,den3);
%Filter 4 (Band Pass Filter)
fc=[600,1000]/(Fs/2);
[num4,den4]=butter(n,fc);
[Hd4,wd4]=freqz(num4,den4);
mag4=abs(Hd4);
phase4=angle(Hd4)*180/pi;
[z4,p4,k4]=butter(n,fc);
sys4=tf(num4,den4);
%Filter 5 (Band Pass Filter)
fc=[1000,3000]/(Fs/2);
[num5,den5]=butter(n,fc);
[Hd5,wd5]=freqz(num5,den5);
mag5=abs(Hd5);
phase5=angle(Hd5)*180/pi;
[z5,p5,k5]=butter(n,fc);
sys5=tf(num5,den5);
%Filter 6 (Band Pass Filter)
fc=[3000,6000]/(Fs/2);
[num6,den6]=butter(n,fc);
[Hd6,wd6]=freqz(num6,den6);
mag6=abs(Hd6);
phase6=angle(Hd6)*180/pi;
[z6,p6,k6]=butter(n,fc);
sys6=tf(num6,den6);
%Filter 7 (Band Pass Filter)
fc=[6000,12000]/(Fs/2);
[num7,den7]=butter(n,fc);
[Hd7,wd7]=freqz(num7,den7);
mag7=abs(Hd7);
phase7=angle(Hd7)*180/pi;
[z7,p7,k7]=butter(n,fc);
sys7=tf(num7,den7);
%Filter 8 (Band Pass Filter)
fc=[12000,14000]/(Fs/2);
[num8,den8]=butter(n,fc);
[Hd8,wd8]=freqz(num8,den8);
mag8=abs(Hd8);
phase8=angle(Hd8)*180/pi;
[z8,p8,k8]=butter(n,fc);
sys8=tf(num8,den8);
```

```
%Filter 9 (Band Pass Filter)
fc=[14000,16000]/(Fs/2);
[num9,den9]=butter(n,fc);
[Hd9,wd9]=freqz(num9,den9);
mag9=abs(Hd9);
phase9=angle(Hd9)*180/pi;
[z9,p9,k9]=butter(n,fc);
sys9=tf(num9,den9);

%Plotting Magnitude and Phase of the developed filters (IIR) Type
figure('Name','Magnitude and Phase of IIR FILTERS');
subplot(9,2,1);plot(wd1/pi,mag1,'linewidth',1.25);grid on;title('Filter 1 Magnitude');
subplot(9,2,2);plot(wd1/pi,phase1,'linewidth',1.25);grid on;title('Filter 1 Phase');
subplot(9,2,3);plot(wd2/pi,mag2,'linewidth',1.25);grid on;title('Filter 2 Magnitude');
subplot(9,2,4);plot(wd2/pi,phase2,'linewidth',1.25);grid on;title('Filter 2 Phase');
subplot(9,2,5);plot(wd3/pi,mag3,'linewidth',1.25);grid on;title('Filter 3 Magnitude');
subplot(9,2,6);plot(wd3/pi,phase3,'linewidth',1.25);grid on;title('Filter 3 Phase');
subplot(9,2,7);plot(wd4/pi,mag4,'linewidth',1.25);grid on;title('Filter 4 Magnitude');
subplot(9,2,8);plot(wd4/pi,phase4,'linewidth',1.25);grid on;title('Filter 4 Phase');
subplot(9,2,9);plot(wd5/pi,mag5,'linewidth',1.25);grid on;title('Filter 5 Magnitude');
subplot(9,2,10);plot(wd5/pi,phase5,'linewidth',1.25);grid on;title('Filter 5 Phase');
subplot(9,2,11);plot(wd6/pi,mag6,'linewidth',1.25);grid on;title('Filter 6 Magnitude');
subplot(9,2,12);plot(wd6/pi,phase6,'linewidth',1.25);grid on;title('Filter 6 Phase');
subplot(9,2,13);plot(wd7/pi,mag7,'linewidth',1.25);grid on;title('Filter 7 Magnitude');
subplot(9,2,14);plot(wd7/pi,phase7,'linewidth',1.25);grid on;title('Filter 7 Phase');
subplot(9,2,15);plot(wd8/pi,mag8,'linewidth',1.25);grid on;title('Filter 8 Magnitude');
subplot(9,2,16);plot(wd8/pi,phase8,'linewidth',1.25);grid on;title('Filter 8 Phase');
subplot(9,2,17);plot(wd9/pi,mag9,'linewidth',1.25);grid on;title('Filter 9 Magnitude');
subplot(9,2,18);plot(wd9/pi,phase9,'linewidth',1.25);grid on;title('Filter 9 Phase');

%Plotting Impulse Responce and step responce of the developed filters (IIR) Type
figure('Name','Impulse Responce and Step Responce of IIR FILTERS');
subplot(9,2,1);step(sys1);grid on;
subplot(9,2,2);impulse(sys1);grid on;
subplot(9,2,3);step(sys2);grid on;
subplot(9,2,4);impulse(sys2);grid on;
subplot(9,2,5);step(sys3);grid on;
subplot(9,2,6);impulse(sys3);grid on;
subplot(9,2,7);step(sys4);grid on;
subplot(9,2,8);impulse(sys4);grid on;
subplot(9,2,9);step(sys5);grid on;
subplot(9,2,10);impulse(sys5);grid on;
subplot(9,2,11);step(sys6);grid on;
subplot(9,2,12);impulse(sys6);grid on;
subplot(9,2,13);step(sys7);grid on;
subplot(9,2,14);impulse(sys7);grid on;
subplot(9,2,15);step(sys8);grid on;
subplot(9,2,16);impulse(sys8);grid on;
subplot(9,2,17);step(sys9);grid on;
subplot(9,2,18);impulse(sys9);grid on;
```

```
%In Case the Type of filters used is (FIR)
else
    den1=1;
    den2=1;
    den3=1;
    den4=1;
    den5=1;
    den6=1;
    den7=1;
    den8=1;
    den9=1;
    n=25;  %Order of the used FIR Filters
    %Filter 1 (Low Pass Filter)
    Fc=170;
    fc=Fc/(Fs/2);
    num1=fir1(n,fc);  %provide filter coefficients
    [H1,w1]=freqz(num1,1,512); %determine the frequency response
    phase1=angle(H1)*180/pi;
    [z1,p1,k1]=tf2zpk(num1,1);
    sys1=tf(num1,ones(1,51));
    %Filter 2 (Band Pass Filter)
    Fc=[170,310];
    fc=Fc/(Fs/2);
    num2=fir1(n,fc); %provide filter coefficients
    [H2,w2]=freqz(num2,1,512); %determine the frequency response
    phase2=angle(H2)*180/pi;
    [z2,p2,k2]=tf2zpk(num2,1);
    sys2=tf(num2,ones(1,51));
    %Filter 3 (Band Pass Filter)
    Fc=[310,600];
    fc=Fc/(Fs/2);
    num3=fir1(n,fc); %provide filter coefficients
    [H3,w3]=freqz(num3,1,512); %determine the frequency response
    phase3=angle(H3)*180/pi;
    [z3,p3,k3]=tf2zpk(num3,1);
    sys3=tf(num3,ones(1,51));
    %Filter 4 (Band Pass Filter)
    Fc=[600,1000];
    fc=Fc/(Fs/2);
    num4=fir1(n,fc); %provide filter coefficients
    [H4,w4]=freqz(num4,1,512); %determine the frequency response
    phase4=angle(H4)*180/pi;
    [z4,p4,k4]=tf2zpk(num4,1);
    sys4=tf(num4,ones(1,51));
    %Filter 5 (Band Pass Filter)
    Fc=[1000,3000];
    fc=Fc/(Fs/2);
    num5=fir1(n,fc); %provide filter coefficients
    [H5,w5]=freqz(num5,1,512); %determine the frequency response
    phase5=angle(H5)*180/pi;
```

```
[z5,p5,k5]=tf2zpk(num5,1);
sys5=tf(num5,ones(1,51));
%Filter 6 (Band Pass Filter)
Fc=[3000,6000];
fc=Fc/(Fs/2);
num6=fir1(n,fc); %provide filter coefficients
[H6,w6]=freqz(num6,1,512); %determine the frequency response
phase6=angle(H6)*180/pi;
[z6,p6,k6]=tf2zpk(num6,1);
sys6=tf(num6,ones(1,51));
%Filter 7 (Band Pass Filter)
Fc=[6000,12000];
fc=Fc/(Fs/2);
num7=fir1(n,fc); %provide filter coefficients
[H7,w7]=freqz(num7,1,512); %determine the frequency response
phase7=angle(H7)*180/pi;
[z7,p7,k7]=tf2zpk(num7,1);
sys7=tf(num7,ones(1,51));
%Filter 8 (Band Pass Filter)
Fc=[12000,14000];
fc=Fc/(Fs/2);
num8=fir1(n,fc); %provide filter coefficients
[H8,w8]=freqz(num8,1,512); %determine the frequency response
phase8=angle(H8)*180/pi;
[z8,p8,k8]=tf2zpk(num8,1);
sys8=tf(num8,ones(1,51));
%Filter 9 (Band Pass Filter)
Fc=[14000,16000];
fc=Fc/(Fs/2);
num9=fir1(n,fc); %provide filter coefficients
[H9,w9]=freqz(num9,1,512); %determine the frequency response
phase9=angle(H9)*180/pi;
[z9,p9,k9]=tf2zpk(num9,1);
sys9=tf(num9,ones(1,51));
den1=1;den2=1;den3=1;den4=1;den5=1;den6=1;den7=1;den8=1;den9=1;

%Plotting Magnitude and Phase of the developed filters (FIR) Type
figure('Name','Magnitude and Phase of FIR FILTERS');
subplot(9,2,1);plot(w1/pi,abs(H1),'linewidth',1.25);grid on;title('Filter 1 Magnitude');
subplot(9,2,2);plot(w1/pi,phase1,'linewidth',1.25);grid on;title('Filter 1 Phase');
subplot(9,2,3);plot(w2/pi,abs(H2),'linewidth',1.25);grid on;title('Filter 2 Magnitude');
subplot(9,2,4);plot(w2/pi,phase2,'linewidth',1.25);grid on;title('Filter 2 Phase');
subplot(9,2,5);plot(w3/pi,abs(H3),'linewidth',1.25);grid on;title('Filter 3 Magnitude');
subplot(9,2,6);plot(w3/pi,phase3,'linewidth',1.25);grid on;title('Filter 3 Phase');
subplot(9,2,7);plot(w4/pi,abs(H4),'linewidth',1.25);grid on;title('Filter 4 Magnitude');
subplot(9,2,8);plot(w4/pi,phase4,'linewidth',1.25);grid on;title('Filter 4 Phase');
subplot(9,2,9);plot(w5/pi,abs(H5),'linewidth',1.25);grid on;title('Filter 5 Magnitude');
subplot(9,2,10);plot(w5/pi,phase5,'linewidth',1.25);grid on;title('Filter 5 Phase');
subplot(9,2,11);plot(w6/pi,abs(H6),'linewidth',1.25);grid on;title('Filter 6 Magnitude');
subplot(9,2,12);plot(w6/pi,phase6,'linewidth',1.25);grid on;title('Filter 6 Phase');
```

```
    subplot(9,2,13);plot(w7/pi,abs(H7),'linewidth',1.25);grid on;title('Filter 7 Magnitude');
    subplot(9,2,14);plot(w7/pi,phase7,'linewidth',1.25);grid on;title('Filter 7 Phase');
    subplot(9,2,15);plot(w8/pi,abs(H8),'linewidth',1.25);grid on;title('Filter 8 Magnitude');
    subplot(9,2,16);plot(w8/pi,phase8,'linewidth',1.25);grid on;title('Filter 8 Phase');
    subplot(9,2,17);plot(w9/pi,abs(H9),'linewidth',1.25);grid on;title('Filter 9 Magnitude');
    subplot(9,2,18);plot(w9/pi,phase9,'linewidth',1.25);grid on;title('Filter 9 Phase');

    %Plotting Impulse Responce and step responce of the developed filters (FIR) Type
    figure('Name','Impulse Responce and Step Responce of FIR FILTERS');
    subplot(9,2,1);step(sys1);grid on;
    subplot(9,2,2);impulse(sys1);grid on;
    subplot(9,2,3);step(sys2);grid on;
    subplot(9,2,4);impulse(sys2);grid on;
    subplot(9,2,5);step(sys3);grid on;
    subplot(9,2,6);impulse(sys3);grid on;
    subplot(9,2,7);step(sys4);grid on;
    subplot(9,2,8);impulse(sys4);grid on;
    subplot(9,2,9);step(sys5);grid on;
    subplot(9,2,10);impulse(sys5);grid on;
    subplot(9,2,11);step(sys6);grid on;
    subplot(9,2,12);impulse(sys6);grid on;
    subplot(9,2,13);step(sys7);grid on;
    subplot(9,2,14);impulse(sys7);grid on;
    subplot(9,2,15);step(sys8);grid on;
    subplot(9,2,16);impulse(sys8);grid on;
    subplot(9,2,17);step(sys9);grid on;
    subplot(9,2,18);impulse(sys9);grid on;
end
%plotting poles&zeros for all filters
figure('Name','Poles and Zeros of all FILTERS');
subplot(3,3,1);zplane(z1,p1);grid on;title('Filter 1');
subplot(3,3,2);zplane(z2,p2);grid on;title('Filter 2');
subplot(3,3,3);zplane(z3,p3);grid on;title('Filter 3');
subplot(3,3,4);zplane(z4,p4);grid on;title('Filter 4');
subplot(3,3,5);zplane(z5,p5);grid on;title('Filter 5');
subplot(3,3,6);zplane(z6,p6);grid on;title('Filter 6');
subplot(3,3,7);zplane(z7,p7);grid on;title('Filter 7');
subplot(3,3,8);zplane(z8,p8);grid on;title('Filter 8');
subplot(3,3,9);zplane(z9,p9);grid on;title('Filter 9');
%Filter the wave file using the filters developed
%Draw the output signals in Time and frequency domains
figure('Name','Filter the wave file using the filters (1, 2, 3)');
%Filter 1 (Low Pass Filter)
x1=filter(num1,den1,y);
subplot(3,2,1);stem(x1);grid on;title('Output from Filter 1 in Time
Domain');ylabel('Amplitude');xlabel('Time');
X1=fftshift(fft(x1));
subplot(3,2,2);stem(abs(X1));grid on;title('Output from Filter 1 in Frequency
Domain');ylabel('Magnitude');xlabel('Frequency');
```

```
%Filter 2 (Band Pass Filter)
x2=filter(num2,den2,y);
subplot(3,2,3);stem(x2);grid on;title('Output from Filter 2 in Time
Domain');ylabel('Amplitude');xlabel('Time');
X2=fftshift(fft(x2));
subplot(3,2,4);stem(abs(X2));grid on;title('Output from Filter 2 in Frequency
Domain');ylabel('Magnitude');xlabel('Frequency');
%Filter 3 (Band Pass Filter)
x3=filter(num3,den3,y);
subplot(3,2,5);stem(x3);grid on;title('Output from Filter 3 in Time
Domain');ylabel('Amplitude');xlabel('Time');
X3=fftshift(fft(x3));
subplot(3,2,6);stem(abs(X3));grid on;title('Output from Filter 3 in Frequency
Domain');ylabel('Magnitude');xlabel('Frequency');
%Filter 4 (Band Pass Filter)
figure('Name','Filter the wave file using the filters (4, 5, 6)');
x4=filter(num4,den4,y);
subplot(3,2,1);stem(x4);grid on;title('Output from Filter 4 in Time
Domain');ylabel('Amplitude');xlabel('Time');
X4=fftshift(fft(x4));
subplot(3,2,2);stem(abs(X4));grid on;title('Output from Filter 4 in Frequency
Domain');ylabel('Magnitude');xlabel('Frequency');
%Filter 5 (Band Pass Filter)
x5=filter(num5,den5,y);
subplot(3,2,3);stem(x5);grid on;title('Output from Filter 5 in Time
Domain');ylabel('Amplitude');xlabel('Time');
X5=fftshift(fft(x5));
subplot(3,2,4);stem(abs(X5));grid on;title('Output from Filter 5 in Frequency
Domain');ylabel('Magnitude');xlabel('Frequency');
%Filter 6 (Band Pass Filter)
x6=filter(num6,den6,y);
subplot(3,2,5);stem(x6);grid on;title('Output from Filter 6 in Time
Domain');ylabel('Amplitude');xlabel('Time');
X6=fftshift(fft(x6));
subplot(3,2,6);stem(abs(X6));grid on;title('Output from Filter 6 in Frequency
Domain');ylabel('Magnitude');xlabel('Frequency');
%Filter 7 (Band Pass Filter)
figure('Name','Filter the wave file using the filters (7, 8, 9)');
x7=filter(num7,den7,y);
subplot(3,2,1);stem(x7);grid on;title('Output from Filter 7 in Time
Domain');ylabel('Amplitude');xlabel('Time');
X7=fftshift(fft(x7));
subplot(3,2,2);stem(abs(X7));grid on;title('Output from Filter 7 in Frequency
Domain');ylabel('Magnitude');xlabel('Frequency');
%Filter 8 (Band Pass Filter)
x8=filter(num8,den8,y);
subplot(3,2,3);stem(x8);grid on;title('Output from Filter 8 in Time
Domain');ylabel('Amplitude');xlabel('Time');
X8=fftshift(fft(x8));
```

```
subplot(3,2,4);stem(abs(X8));grid on;title('Output from Filter 8 in Frequency
Domain');ylabel('Magnitude');xlabel('Frequency');
%Filter 9 (Band Pass Filter)
x9=filter(num9,den9,y);
subplot(3,2,5);stem(x9);grid on;title('Output from Filter 9 in Time
Domain');ylabel('Amplitude');xlabel('Time');
X9=fftshift(fft(x9));
subplot(3,2,6);stem(abs(X9));grid on;title('Output from Filter 9 in Frequency
Domain');ylabel('Magnitude');xlabel('Frequency');
%Amplify the output signals using the user defined gain
x1=G1*x1;
x2=G2*x2;
x3=G3*x3;
x4=G4*x4;
x5=G5*x5;
x6=G6*x6;
x7=G7*x7;
x8=G8*x8;
x9=G9*x9;
%Add the amplified - output signals in time domain to form composite signal
xt=x1+x2+x3+x4+x5+x6+x7+x8+x9;
% Draw and compare the composite signal with the original signal in time domain
figure('Name','Comparing the Composite with the Original in Time Domain');
subplot(2,1,1);stem(xt);grid on;title('Composite Signal in Time
Domain');ylabel('Amplitude');xlabel('Time');
subplot(2,1,2);stem(y);grid on;title('Original Signal in Time
Domain');ylabel('Amplitude');xlabel('Time');
xf=fftshift(fft(xt));
yf=fftshift(fft(y));
% Draw and compare the composite signal with the original signal in frequency domain
figure('Name','Comparing the Composite with the Original in Freq Domain');
subplot(2,1,1);stem(abs(xf));grid on;title('Composite Signal in Frequency
Domain');ylabel('Magnitude');xlabel('Frequency');
subplot(2,1,2);stem(abs(yf));grid on;title('Original Signal in Frequency
Domain');ylabel('Magnitude');xlabel('Frequency');
%Playing the output wave signal
sound(xt,fouts);
xt=resample(xt,fouts,Fs);
figure('Name','Output Signal After Resampling with Output Sample Rate');
subplot(2,1,1);stem(xt);grid on;title('Output Signal after Resampling in Time
Domain');ylabel('Amplitude');xlabel('Time');
xf=fftshift(fft(xt));
subplot(2,1,2);stem(abs(xf));grid on;title('Output Signal after Resampling in Frequency
Domain');ylabel('Magnitude');xlabel('Frequency');
fprintf('****************THANK YOU****************\n');
```

References

1. MATLAB Software
2. Final Project Manual
3. Lab Videos
4. Lab Manuals