

# DSP Project Report

## Names:

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## Code:

We created 10 Filters [7 FIR Filters and 3 IIR Filters],

Types of FIR:

- 1) Constr\_band\_equiripple
- 2) Constr\_Least\_squares
- 3) Equiripple
- 4) Generalized\_equiripple
- 5) Least\_squares
- 6) Window\_Chebyshev
- 7) Window\_Rectangular

Types of IIR:

- 1) Butterworth\_
- 2) Chebyshev\_type\_1
- 3) Chebyshev\_type\_2

As well as, we created three main functions  
[initialize\_data(), amplify\_signal() and add\_filterd\_signal()]

In `initialize_data()`, we convert original signal from time domain to frequency domain then we filter the original signal and create 9 output signals in time domain then convert them to frequency domain.

In `amplify_signal()`, we multiply the 9 filtered signals in gains that user enters then convert 9 amplified signals from time domain to frequency domain.

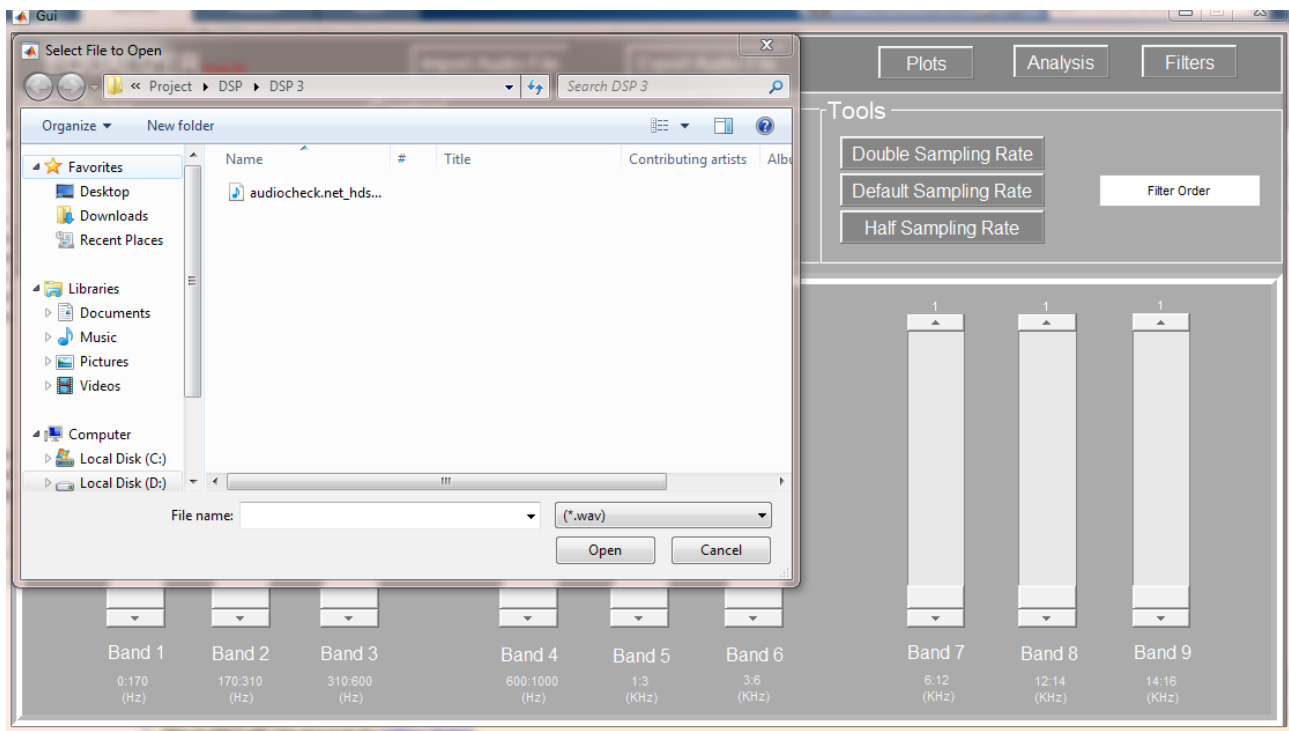
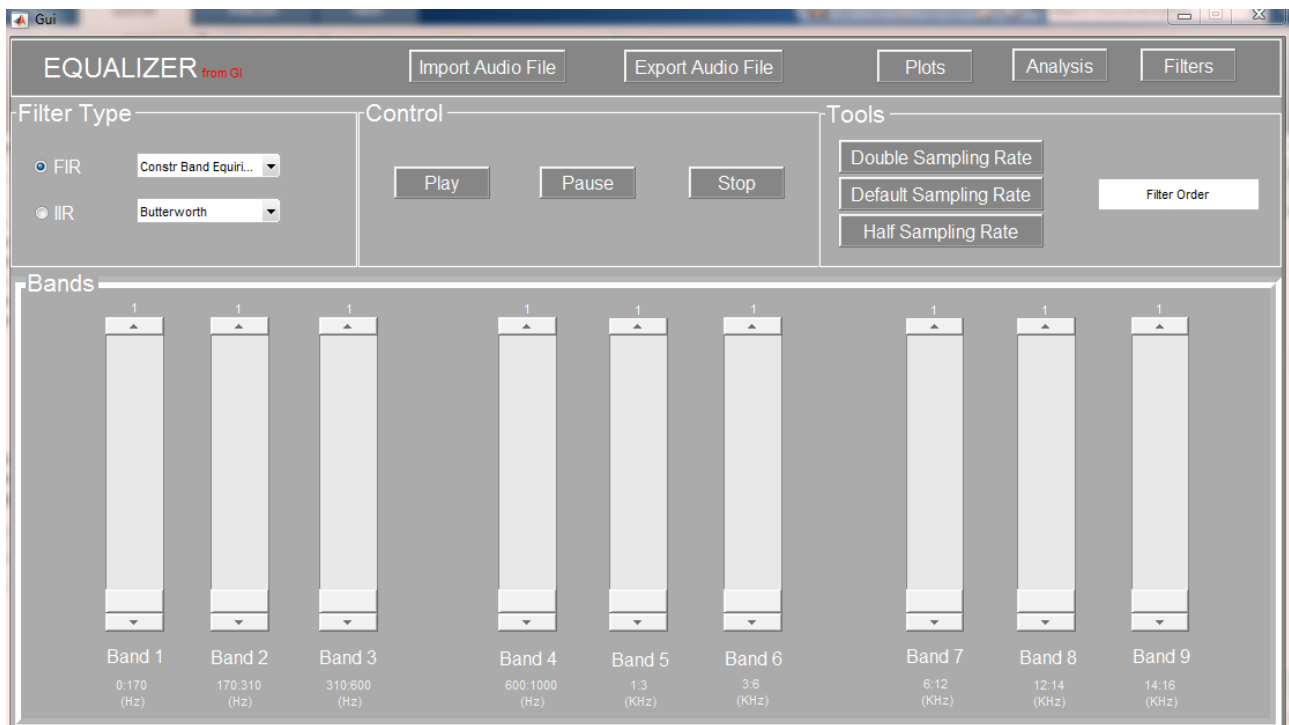
In `add_filterd_signal()`, we add 9 amplified signals to create the composite signal then convert it from time domain to frequency domain.

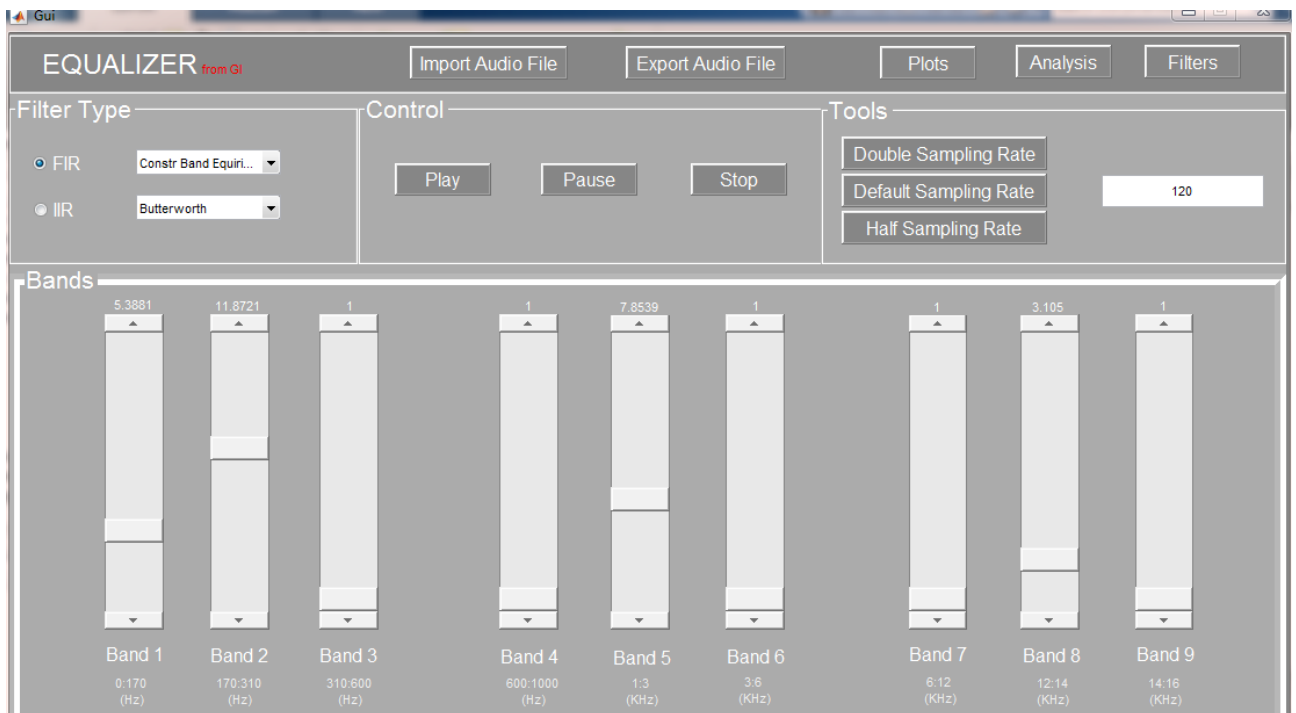
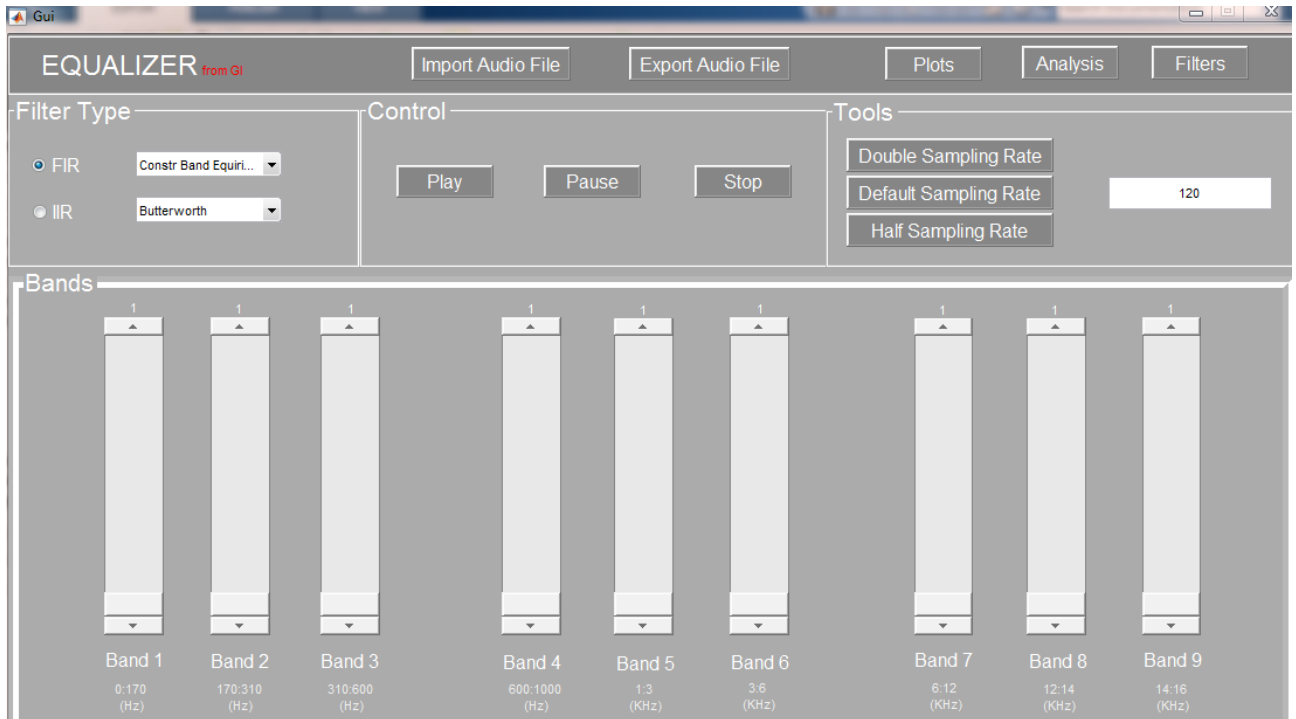
## GUI:

In our GUI, we give user the ability to:

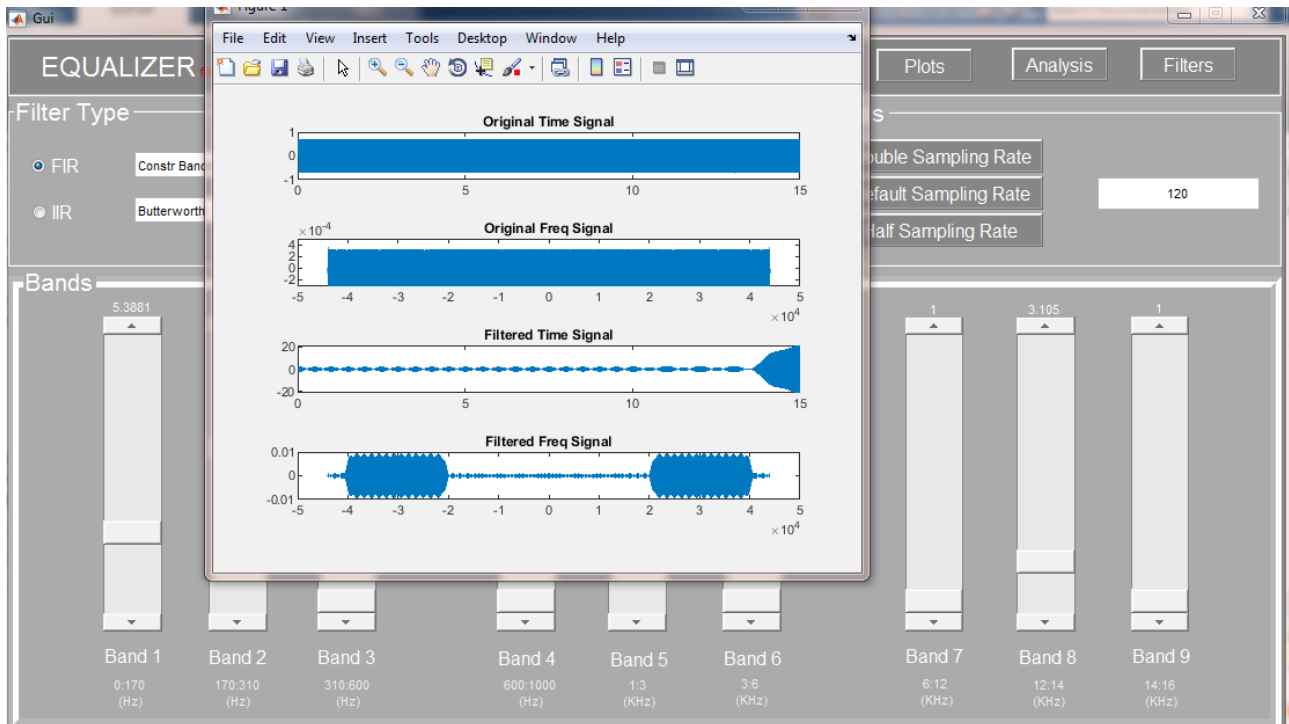
- 1) import any audio as input.
- 2) export or save the audio after filteration and amplification.
- 3) choose the filter type.
- 4) enter 9 gains for 9 filters.
- 5) play or stop the audio.
- 6) choose the sampling ratio.
- 7) enter filter order.
- 8) plot (input and composite signals) or (analysis of each filter) or (output signals from filters).

## GUI Screenshots:





## Original and Composite signals:



## Filters' Analysis:



# Signal Filtration:

