JEPPIAAR ENGINEERING COLLEGE

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

Faculty Name: Mr. B. Appoon

Course Title: EC3492 – Digital Signal Processing

Academic Year: 2023 - 24

Semester/Year: IV/II

Type: Project-Based Learning in correlate to applications of Digital Signal Processing

Objective

To design and implement digital filters in MATLAB for processing audio signals, with the objective of removing unwanted noise, enhancing signal quality, and analyzing the frequency response of the filters.

Filter Design Parameters

- Sampling frequency (Fs)
- Cutoff frequency/frequencies (Fc)
- Filter order (N)
- Passband/stopband ripple
- Transition width

MATLAB Implementation

1. Reading an Audio File
2. Basic FIR Filter Design
3. IIR Filter Design
4. Frequency Response Visualization
5. Playing and Comparing Audio

Advanced Techniques

1. Equalizer Design (Multiple Band-pass Filters)
2. Noise Reduction Using Adaptive Filters
3. Real-time Audio Processing

Working:

1. The program filters a noisy audio signal using different types of Butterworth digital filters. Output is analyzed by listening to the audio and viewing the waveform and spectrum plots.
2. Low Pass Butterworth Filter: Removes high-frequency noise while preserving low-frequency components.
3. High Pass Butterworth Filter: Emphasizes high-frequency details and reduces low-frequency noise.
4. Band Pass Butterworth Filter: Isolates a specific frequency band (e.g., 1 kHz to 2.5 kHz), removing frequencies outside the band.

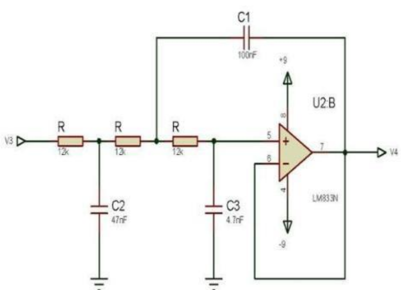

Fig.1 Simulation: (MATLAB)

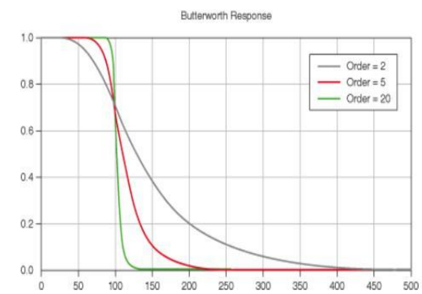

Fig.2 Signal Response

Tools Used:

1. MATLAB / GNU Octave
2. Audio (.wav) file
3. Butterworth Filter (butter, filter)
4. FFT (Fast Fourier Transform)
5. PC / Laptop
6. Microphone (optional)
7. Speakers or Headphones
8. Plotting Tools

Practical Considerations

1. **Filter stability**: Always check stability, especially for IIR filters
2. **Phase distortion**: FIR filters can be designed to have linear phase
3. **Computational complexity**: Higher order filters provide sharper transitions but require more computation
4. **Latency**: Important for real-time applications

Application:

1. Audio processing
2. Biomedical signal denoising
3. Communication systems

Conclusion:

MATLAB provides comprehensive tools for audio signal filtering, from basic filter design to advanced real-time processing. The Signal Processing Toolbox and DSP System Toolbox offer specialized functions that simplify the implementation of various filtering techniques for audio applications.

CO & PO Mapping:

To introduce the concepts of adaptive filters and its application to communication engineering	CO5
Engineering Knowledge	PO1
Modern Tool Usage	PO5
Life Long Learning	PO12

Outcome of the Activity:

The outcome of Processing Audio Signals using MATLAB improves the audio quality, feature extraction and classification, real-time processing capability, automation and efficiency, and real-time noise suppression for better hearing. Improvise the students' ability to innovate and utilize modern tools in the platform of Application-based and Project-based learning.