



ENGINEERING PROFESSIONS DEPARTMENT

Computer Systems Engineering

Digital Signal Processing

Speech Noise Reduction through FIR Bandpass Filtering with Kaiser Window Method

Prepared by:

Ahmed Marwan Jalambo 2049011002

Haytham Mohammed Khalil 2049011015

Mohammed Yousef Abu Amra 2049011027

Supervisor:

Eng. Akram Abu Garad

Abstract

- This project presents a speech noise reduction system based on a **Finite Impulse Response (FIR) bandpass filter** designed using the **Kaiser Window method**. Targeting the **80 Hz – 6 kHz** range, the filter suppresses low-frequency **hums** and high-frequency **hiss** while preserving speech clarity and intelligibility.
- The Kaiser window's adjustable **β parameter** allows fine control over **transition sharpness** and **stopband attenuation**, enabling performance tuning for various noise conditions.
- A custom GUI enhances usability by allowing users to experiment with **cutoff frequencies**, **transition widths**, and **attenuation levels**, instantly hearing and visualizing the results. The system is practical for applications such as **telephony**, **voice recording**, and **communication in noisy environments**.

Introduction

Speech signals are often affected by unwanted noise from environment, recording devices, or transmission channels.

Goal: Reduce noise while maintaining speech intelligibility.

Approach:

- Use a **FIR bandpass filter** to pass only speech frequencies (**80 Hz – 6 kHz**).
- Remove low-frequency hum (< 80 Hz) and high-frequency hiss (> 6 kHz).
- Apply **Kaiser window** for precise control over filter performance.

Theoretical Background – FIR Filters 1/2

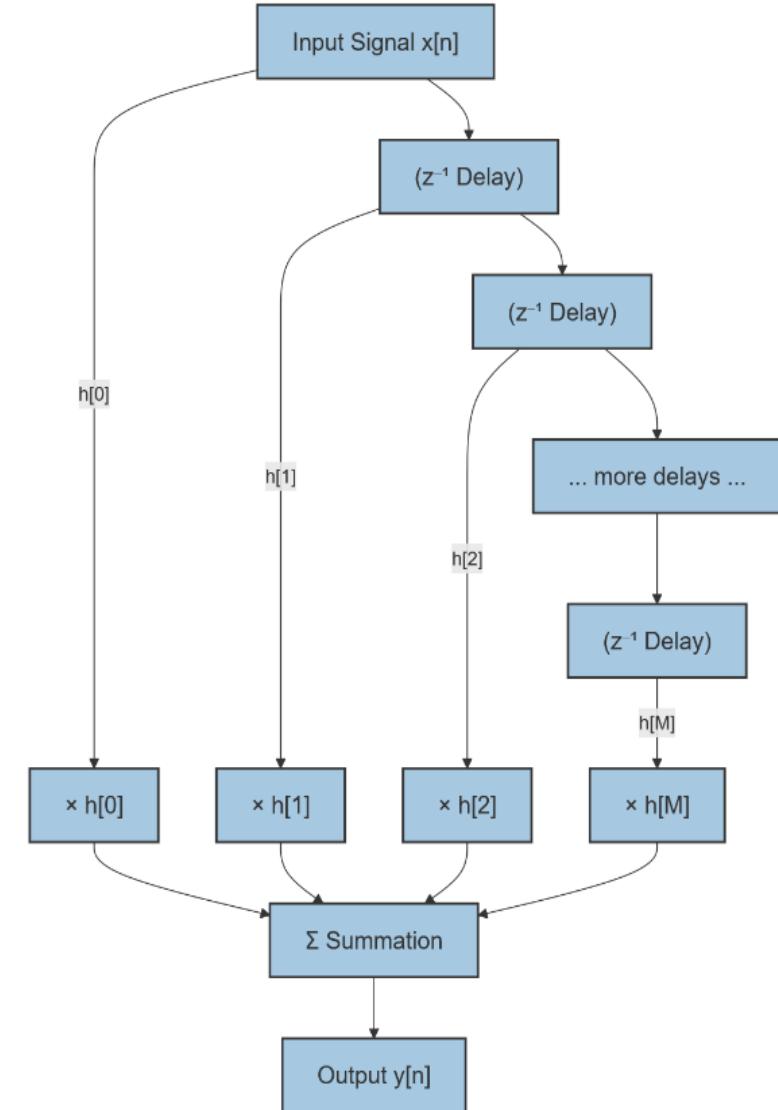
What is an FIR Filter — why FIR?

Finite Impulse Response (FIR) filters produce an output based on a *finite* set of past input samples.

- They have a **linear phase response**, preserving the shape of waveforms — important for speech processing.
- Always **stable** (no feedback loops) and straightforward to design for specific frequency bands.

How FIR Filtering Works:

- The **block diagram** shows how the input signal $x[n]$ is delayed, multiplied by filter coefficients $h[k]$, and summed to produce the output $y[n]$.



Theoretical Background – FIR Filters 2/2

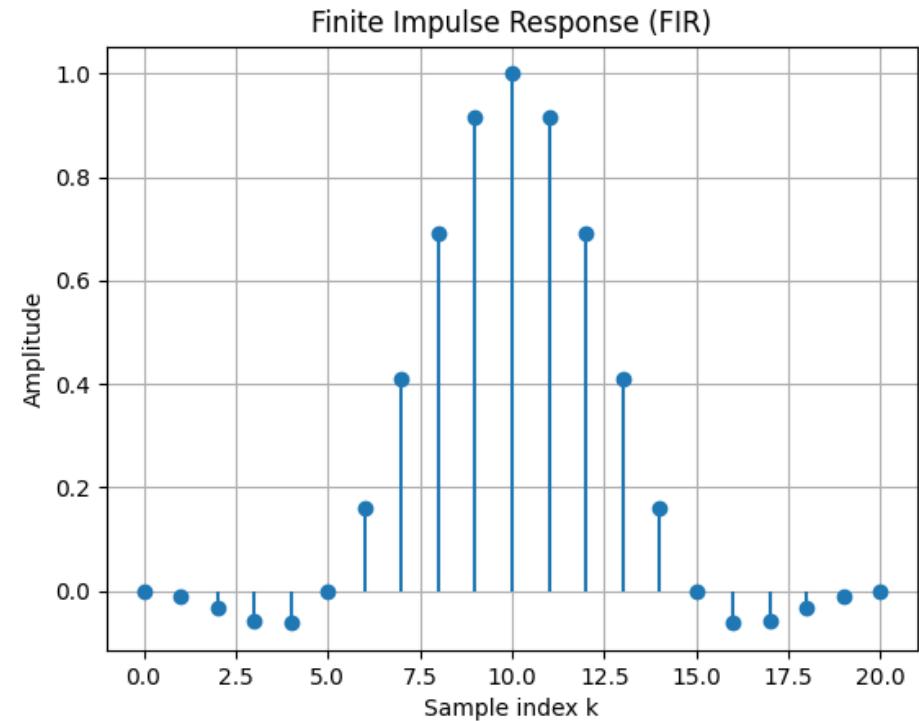
- The **impulse response plot** illustrates the set of coefficients $h[k]$ that define the filter's behavior in the time domain.
- **Mathematical Formulation:**

$$y[n] = \sum_{k=0}^M h[k] \cdot x[n - k]$$

$h[k]$: filter coefficients

M : filter order

Key Point: In speech noise reduction, the filter is designed so that the impulse response emphasizes desired speech frequencies while attenuating unwanted noise.



FIR Bandpass Filters – Principle 1/2

Concept

- A **bandpass filter** passes frequencies within a defined range and attenuates frequencies outside.
- In **speech processing**, the target range is typically **80 Hz – 6,000 Hz**.
- This removes:
 - **Low-frequency hum** (< 80 Hz) from power lines, wind, or mic handling
 - **High-frequency hiss** (> 6 kHz) from electronic noise or harsh consonants

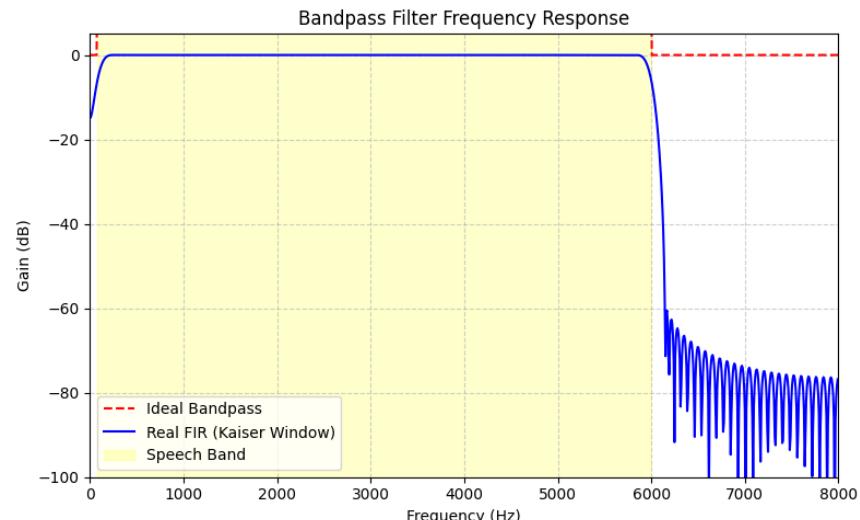
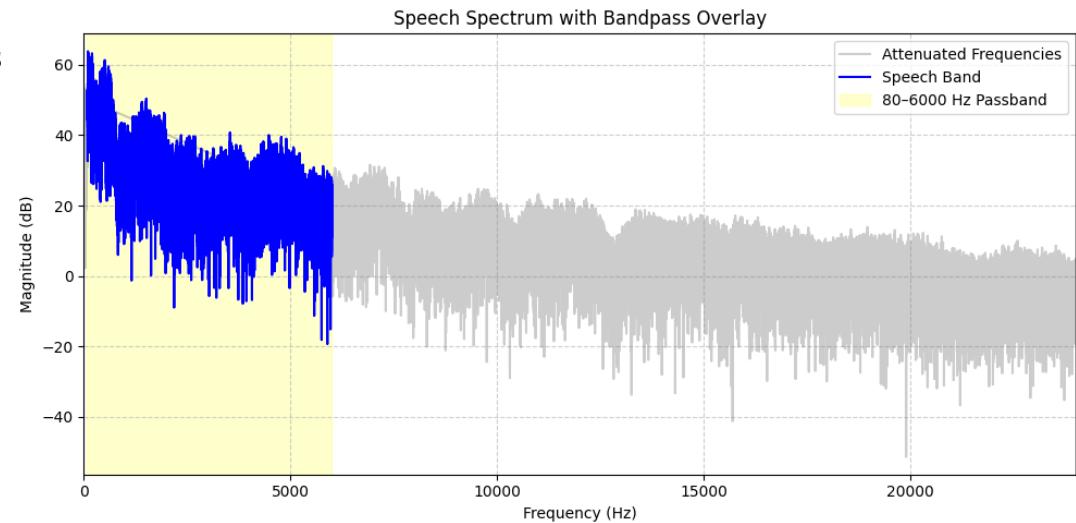
How It Works

1. Frequency Response View

- **Yellow band:** Speech range
- **Red dashed line:** Ideal bandpass
- **Blue curve:** Real FIR filter using Kaiser window — smooth transition with strong stopband attenuation

2. Speech Spectrum Overlay

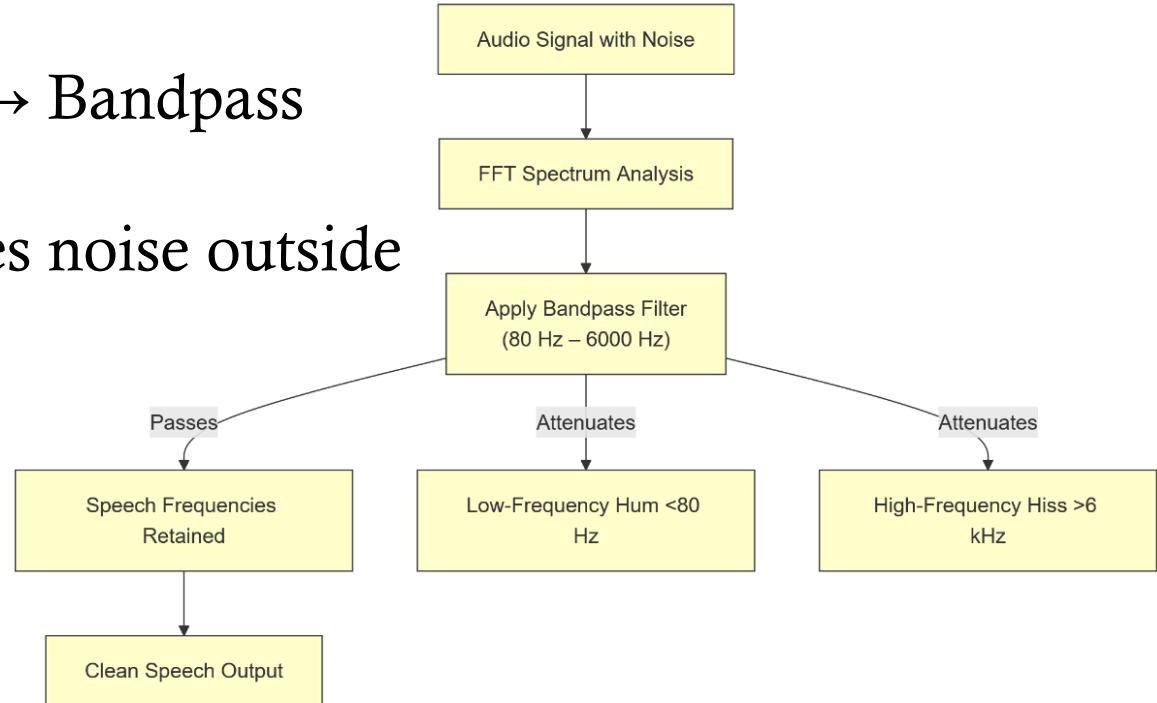
- Original speech spectrum in **grey**
- Filter passband retains speech components (blue in the yellow zone)
- Out-of-band noise is strongly reduced



FIR Bandpass Filters – Principle 2/2

3. Filtering Workflow

- Audio with noise → FFT analysis → Bandpass filtering (80 Hz–6 kHz)
- Passes speech frequencies, attenuates noise outside the band
- Produces **clean speech output**



Key Design Parameters

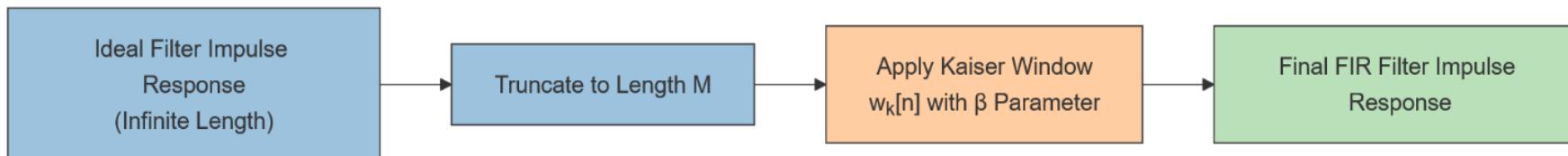
- **Lower cutoff (f_L), Upper cutoff (f_H)** — define the speech band
- **Transition width** — controls filter sharpness
- **Stopband attenuation (α_s)** — determines noise reduction level

Windowing FIR Filters – Why Kaiser? 1/2

Purpose of Windowing

- Ideal filters have **infinite impulse responses**, which are impractical to implement.
- **Window method:**
 - Start with ideal filter impulse response
 - Truncate to a finite length (M)
 - Multiply by a **window function** to control side lobes and transition width

Goal: Reduce leakage and unwanted ripples in the frequency response.



Windowing FIR Filters – Why Kaiser? 2/2

Kaiser Window Advantages

- **Adjustable performance via β (beta) parameter:**
 - Low $\beta \rightarrow$ Narrow main lobe, higher side lobes (less attenuation)
 - High $\beta \rightarrow$ Wider main lobe, much lower side lobes (better attenuation)
- **Trade-off control between:**
 - **Main lobe width** → affects transition sharpness
 - **Stopband attenuation** → affects noise suppression
- Capable of **very high stopband attenuation**, ideal for **speech noise reduction**.

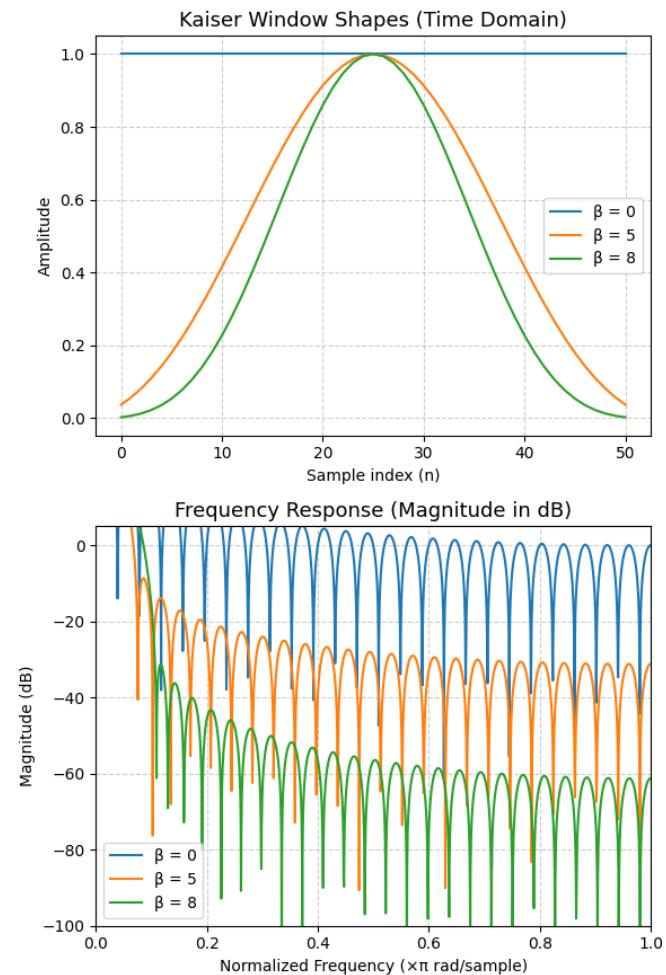
Kaiser Window Equation

$$w_k[n] = \frac{I_0\left(\beta \sqrt{1 - \left(\frac{n - \alpha}{\alpha}\right)^2}\right)}{I_0(\beta)}, 0 \leq n \leq M$$

I_0 : Zero-order modified Bessel function

$$\alpha = M/2$$

Kaiser Window: Time & Frequency Domain Views

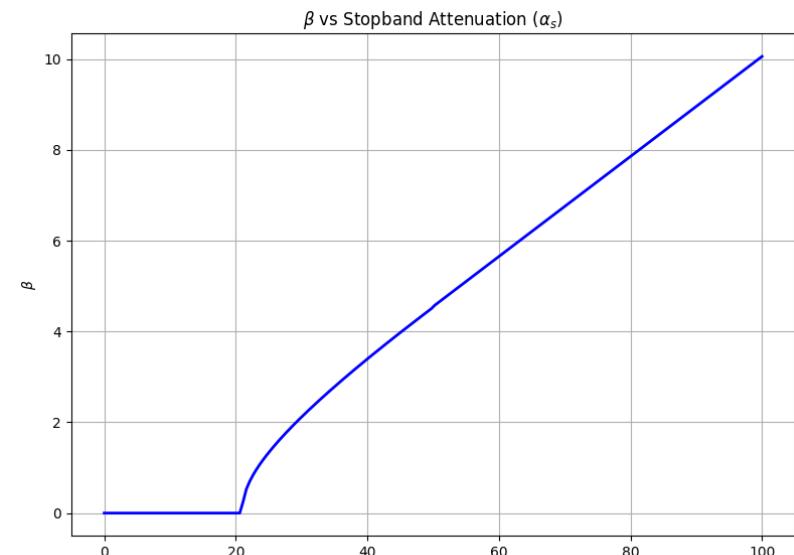


Kaiser Window Parameter β Selection 1/2

Attenuation to Beta Mapping

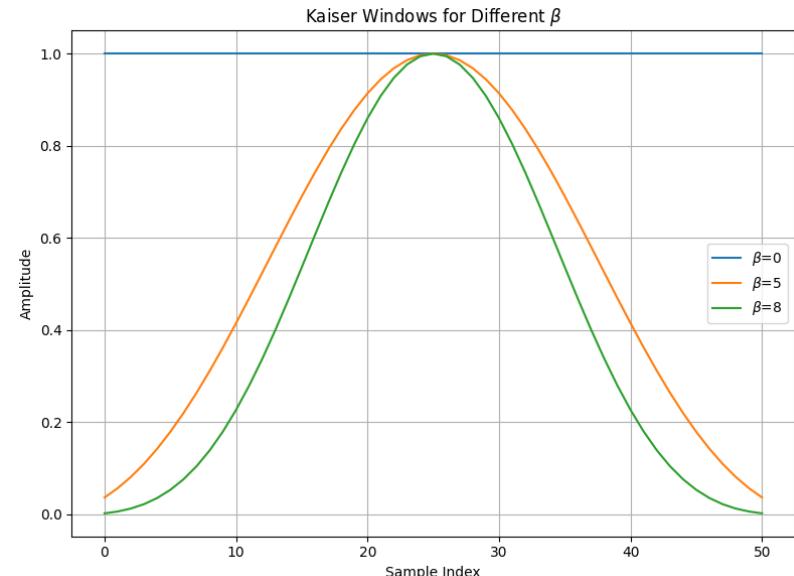
- Stopband attenuation (α_s) determines the β parameter.
- Higher $\alpha_s \rightarrow$ higher $\beta \rightarrow$ Lower side lobes (better noise suppression).
- Mapping is defined by standard equations:

$$\beta = \begin{cases} 0 & , \alpha_s \leq 21 \\ 0.5842(\alpha_s - 21)^{0.4} + 0.07886(\alpha_s - 21) & , 21 < \alpha_s \leq 50 \\ 0.1102(\alpha_s - 8.7) & , \alpha_s > 50 \end{cases}$$



Kaiser Windows for Different β

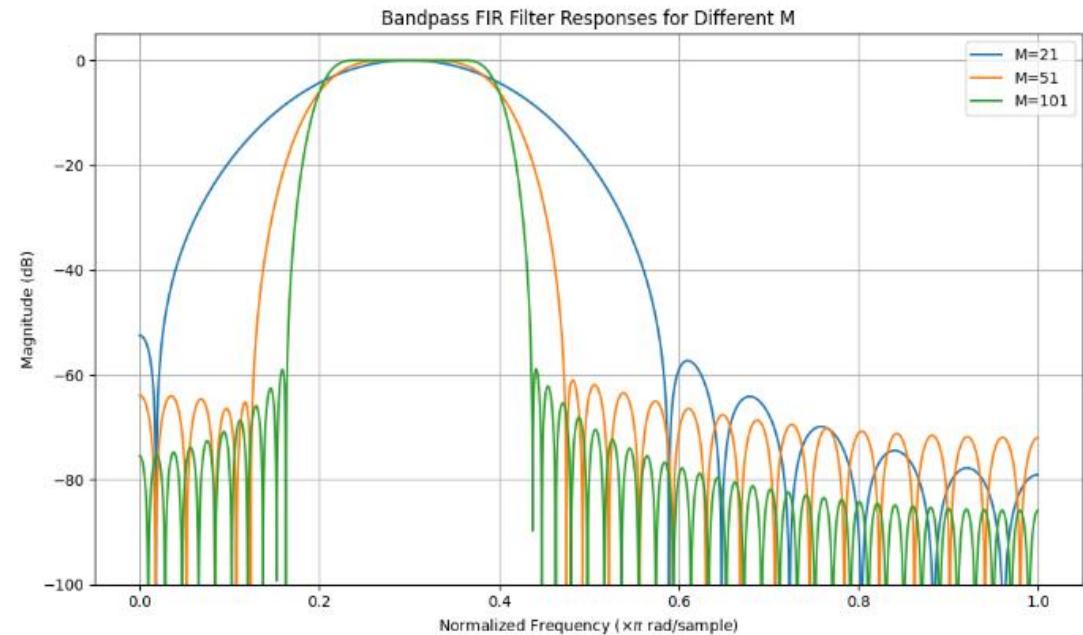
- Increasing β widens the **main lobe** and greatly reduces **side lobes** in the frequency response.
- Low β (near 0) \rightarrow Sharp transitions but poor attenuation.
- High β \rightarrow Smoother transitions, excellent attenuation.



Kaiser Window Parameter β Selection 2/2

Filter Order (M) Effect

- Higher M (more filter taps) →
 - **Sharper transitions** in frequency response.
 - **Better selectivity** between passband and stopband.
 - **Higher computational cost.**
- Typical trade-off: Choose M to balance performance and real-time feasibility.



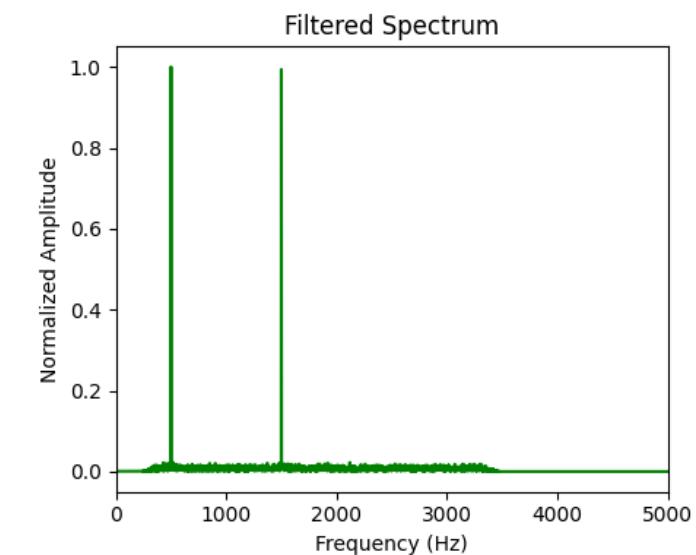
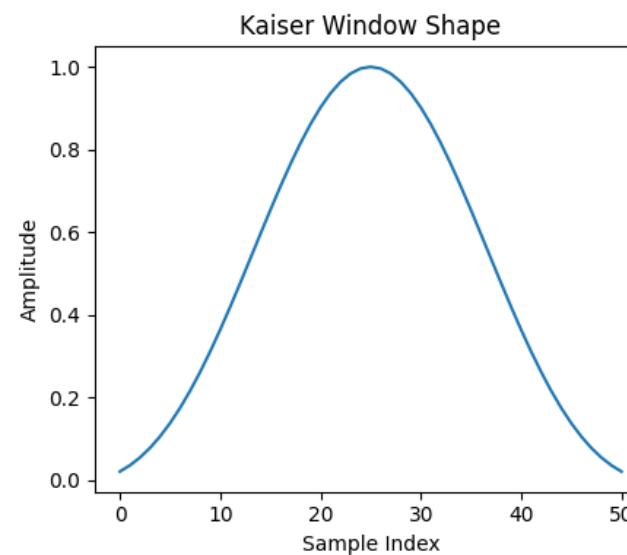
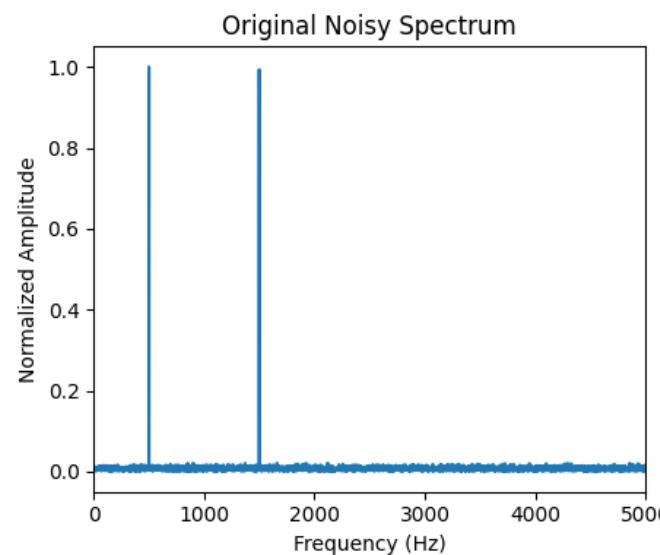
Practical Design Notes

- Select α_s based on required **noise suppression level**.
- Set **transition width** according to the desired **frequency resolution**.
- Both β and M can be fine-tuned to meet **DSP project constraints** (latency, complexity, and attenuation).

Bandpass Filter Design Using Kaiser Window – Workflow 1/2

Visual Overview

- **Spectrum of the noisy input signal:** contains target speech frequencies plus unwanted noise.
- **Kaiser window shape:** applied to the truncated ideal impulse response.
- **Filtered spectrum:** passband preserved, noise outside removed.



Bandpass Filter Design Using Kaiser Window – Workflow

2/2

Design Steps

1. **Define normalized frequency edges (f_L, f_H):** Convert desired cutoff frequencies (e.g., 80 Hz, 6,000 Hz) to normalized form (0 – 1) based on sampling rate.
2. **Set performance goals**
 - Required stopband attenuation α_s (in dB)
 - Desired transition width (Hz)
3. **Compute β (Beta1):** Use standard formulas to map $\alpha_s \rightarrow \beta$.
4. **Determine filter order (M):** Based on α_s and transition width to achieve sharpness and attenuation targets.
5. **Generate FIR filter coefficients**
6. **Apply filter to audio:** Convolve the designed FIR filter with the noisy speech signal.

Implementation Note:

- These steps are implemented in **design_kaiser_bandpass_filter** in the project codebase — automating the calculation of β, M and filter coefficients.

Demo

Conclusion

- **Result:** Noise outside the speech band was effectively removed, improving clarity.
- **Advantages:**
 - Flexible parameter control (cutoff frequencies, transition width, attenuation)
 - No phase distortion
 - User-friendly GUI for quick testing and comparison
- **Applications:** Telephony, broadcasting, voice recording in noisy environments

The End