SIMULINK HEARING AID

1. Introduction

In modern digital systems, speech and audio enhancement play a vital role in improving communication quality. Background noise, distortion, and poor frequency response often degrade signals in real-world conditions. This project demonstrates a MATLAB-based audio enhancement pipeline that mimics the functionality of hearing aids and speech processing systems. The pipeline adds noise to a clean signal, applies filtering and frequency shaping, compresses amplitudes, and finally evaluates the result using spectrograms.

2. Objectives

The key objectives of this project are:

- To simulate real-world noisy environments by adding Gaussian noise to a clean audio input.
- To design digital filters (low-pass and band-pass) for noise suppression and frequency shaping.
- To implement amplitude compression in the frequency domain using FFT.
- To visualize improvements using spectrogram analysis.

3. Methodology

The audio processing chain is shown in Figure 1 (placeholder for block diagram).

- **Audio Input**

  A .wav file is loaded in MATLAB and converted to mono if necessary.

- **Noise Addition**

  White Gaussian noise with 40 dB SNR is added to simulate environmental interference.

- **Noise Reduction (Low-Pass Filter)**

  A low-pass IIR filter attenuates noise above 3.5 kHz.

- **Frequency Shaping (Band-Pass Filter)**

  A band-pass filter emphasizes the 2–4 kHz region, which improves speech intelligibility.

- **Amplitude Compression (FFT Domain)**

  The FFT spectrum is clipped above a threshold to prevent sudden peaks.

  IFFT reconstructs the signal with reduced dynamic range.

- **Spectrogram Analysis**

  The spectrogram provides time-frequency visualization of the noisy and processed signals

4. Results

The project demonstrated the following:

- The original input signal was clean and distortion-free.
- Addition of Gaussian noise degraded quality, making the waveform irregular.
- The low-pass filter removed unwanted high-frequency noise.
- The band-pass filter emphasized key speech frequencies, improving intelligibility.
- FFT-based amplitude compression normalized the signal and reduced sudden peaks.
- Spectrogram analysis clearly showed improvement between the noisy input and final processed output.

5. Appendix

Figure 1: Original input waveform
Figure 2: Noisy signal waveform
Figure 3: Denoised signal waveform after low-pass filtering
Figure 4: Frequency-shaped signal (band-pass filter applied)
Figure 5: Spectrogram of noisy signal
Figure 6: Spectrogram of final processed output

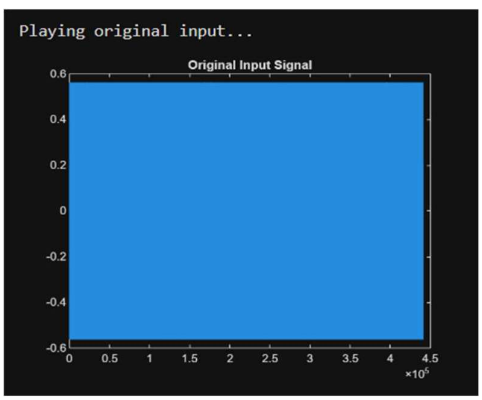


Figure-1

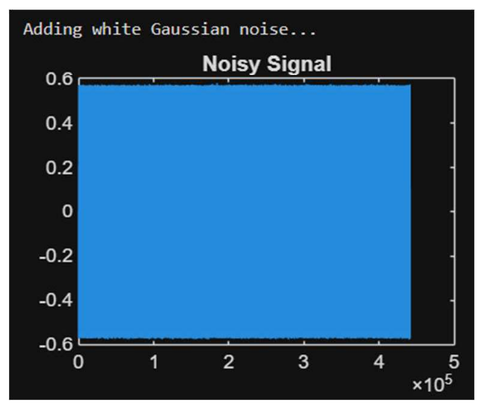
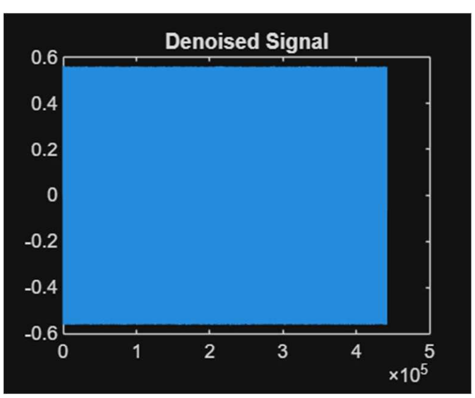


Figure-2



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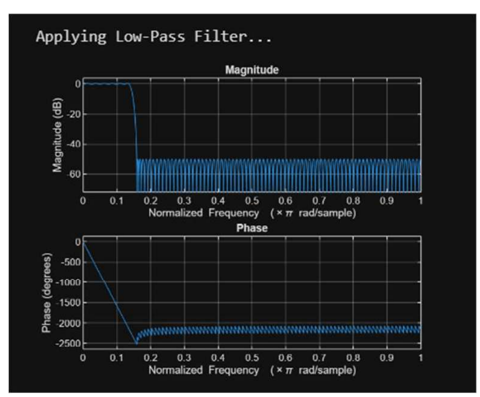

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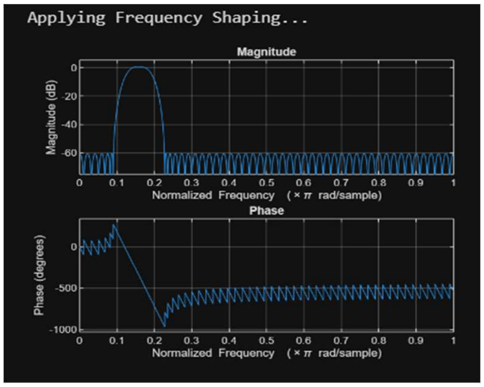

Figure-5

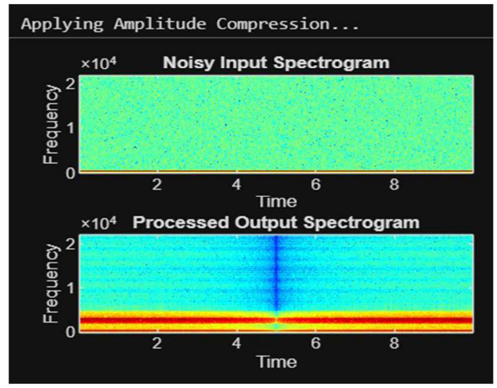

Figure-6

6. Conclusion

This project successfully demonstrates audio enhancement using MATLAB. Noise reduction, frequency shaping, and amplitude compression significantly improved clarity compared to the noisy input. Future extensions may include adaptive filtering, Wiener filtering, or real-time DSP hardware implementation.

Such methods can be applied in:

- Hearing aids for the hearing-impaired.
- Telecommunications to improve call quality.
- Speech recognition systems as preprocessing.

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