

EECE 340 Project-Section 3.4 Design Insights and Discussion

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Summary of findings

In this experiment, we chose a real-life signal (voice signal) to analyse and applied to it the different tools that we had developed in previous sections.

In part 3.0:

We recorded a voice signal and analysed its time-domain and frequency-domain characteristics. By extracting a meaningful 2-second segment from the recording and computing its Fourier transform, we identified that most of the signal's energy was concentrated below 1000 Hz, with additional higher-frequency components present. This analysis highlighted that the voice signal is primarily low-frequency but contains higher-frequency details originating from noise that may be susceptible to aliasing. The frequency spectrum provided insight into which frequencies are most important for preserving intelligibility and guided the later selection of the filter cutoff frequency.

In part 3.1:

We designed and implemented a low-pass FIR filter using a windowed sinc function. The impulse response and frequency response of the filter were analysed, confirming that the filter effectively attenuated high-frequency components while preserving frequencies important for speech. Applying the filter to the extracted voice segment successfully reduced high-frequency noise when applied in part 3.3 and minimized aliasing risk. The filtered signal retained the main characteristics of the original signal while eliminating frequencies above the cutoff frequency specified.

In part 3.2:

We investigated how different sampling rates affect the reconstruction of both the original and filtered voice signals. We sampled and reconstructed the signals using sinc interpolation at 600 Hz, 1800 Hz, and 5000 Hz. At the highest sampling rate (5000 Hz), the reconstructed signals closely matched the original, preserving fine details with minimal distortion. However, as the sampling rate decreased, we observed increasing loss of important information, particularly at 600 Hz, where the reconstructed signals showed visible distortion and missed high-frequency content. The original signal at high sampling rates was missing information due to the Nyquist limit, as some frequencies could not be captured, while the filtered signal eliminated these unreachable frequencies and provided a more accurate reconstruction. At lower sampling rates, both the original and filtered reconstructions were far from the true signal, showing noticeable distortion. But at high sampling rates, both reconstructions were very close to the real signal, confirming the importance of sufficient sampling and filtering to achieve accurate reconstruction.

In part 3.3:

We analyzed how noise affects the reconstruction of both the original and filtered signals. After sampling at 1800 Hz, we added white Gaussian noise to the sampled points and then reconstructed the signals using sinc interpolation. We observed that noise degraded the reconstruction quality in both cases, introducing fluctuations and reducing fidelity. The original signal was more affected by noise, especially in higher-frequency regions, because no filter had removed those components prior to sampling. The filtered signal showed better resilience, as the filter reduced high-frequency content where noise tends to have greater impact, resulting in a cleaner reconstruction despite noise. Noise effects were particularly noticeable during low-energy regions like pauses or soft speech, where the signal was weaker and noise dominated. Overall, filtering before sampling helped reduce the influence of noise on the reconstructed signal, while the original unfiltered signal remained more sensitive to noise contamination.

Broader implications:

The findings from this analysis have important implications across audio, biomedical, and communications systems. In audio processing, effective filtering and appropriate sampling rates are crucial to preserve sound quality and avoid distortion, especially in applications like speech recognition or audio compression. In biomedical signal processing, such as ECG or EEG analysis, filtering high-frequency noise before sampling ensures that critical physiological information is retained without contamination from artifacts, leading to more accurate diagnoses. In communications systems, understanding the relationship between sampling, filtering, and noise resilience is key for reliable data transmission and signal reconstruction, particularly in noisy channels. Overall, this experiment demonstrates the importance of balancing sampling rates, filter design, and noise mitigation strategies to achieve accurate, efficient, and robust signal representation in real-world systems.