EECE 340 Project-Section 3.2 Sampling and Reconstruction

Prepared by: Carl Wakim and Joseph Chahine

Introduction

In this section, we investigate the effects of sampling and reconstruction on both the original and filtered versions of the voice signal. We first apply a low-pass filter to the extracted signal segment to reduce high-frequency components. Then, we sample both the original and filtered signals at various sampling rates and reconstruct them using sinc interpolation. By comparing the reconstructed signals to the original signal, we aim to assess the impact of filtering and sampling rate on reconstruction fidelity.

MATLAB file description

Sampling_reconstruction:

This script loads a pre-recorded voice signal from my_recording.mat and extracts a 2-second segment from 3 to 5 seconds. It applies a low-pass filter to the segment using an external apply_lpf function to obtain a filtered version of the signal. (Note that in this section we decreased cutoff frequency to 800Hz instead of the original 1300Hz that was used in 3.1. This was done to amplify the effect of the LPF)

The script then samples both the original and filtered segments at multiple sampling rates (600 Hz, 1800 Hz, 5000 Hz). For each sampling rate, it reconstructs the signal using sinc interpolation from the sampled points.

For each case, it plots:

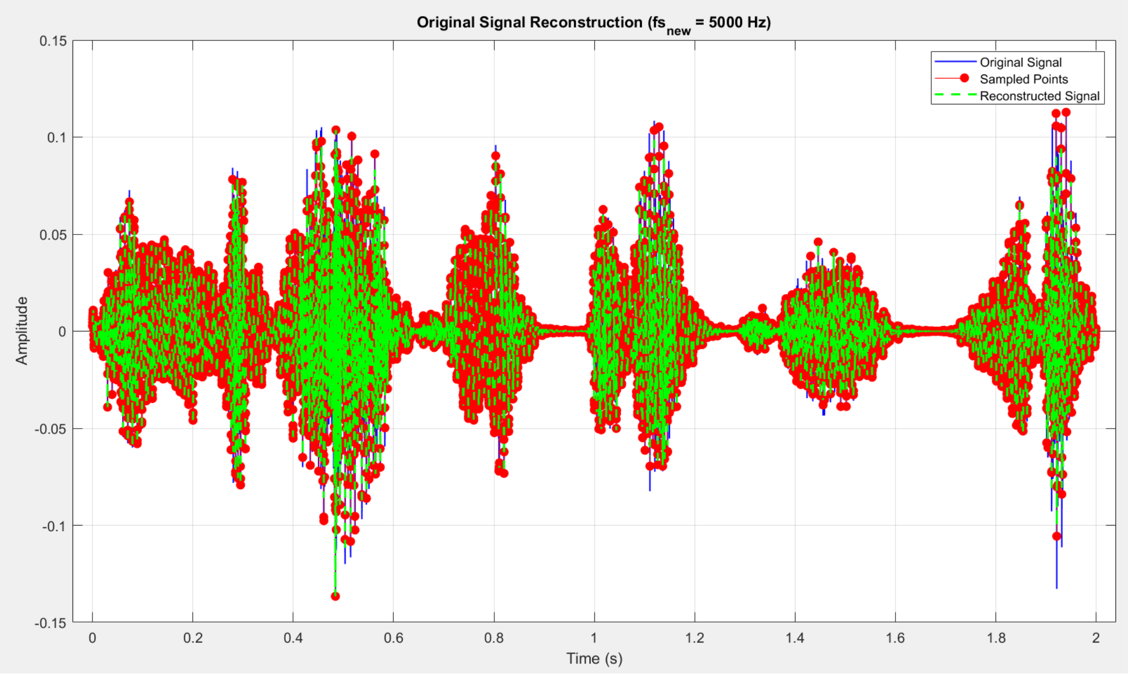
1. The original segment with its sampled points and reconstructed signal.
2. The filtered segment with its sampled points and reconstructed signal.

Finally, it computes the reconstruction error for both the original and filtered signals (comparing each reconstruction to the clean version it came from) and prints the error values for each sampling rate.

Figures description

Figure 1: Sampling at f=5000Hz

The figure shows two separate plots comparing the reconstruction results for the original and filtered voice signal, both sampled and reconstructed at a sampling rate of **5000 Hz**. The **bottom plot** shows the filtered signal, while the **top plot** shows the original (unfiltered) signal. At this sampling rate, both reconstructions closely follow the original waveforms, with the reconstructed signals mostly overlapping the originals. The filtered signal shows that the high-frequency content has been reduced by the low-pass filter. Both signals exhibit high reconstruction fidelity at 5000 Hz, with minimal visible aliasing or distortion. This confirms that at a sampling rate well above the Nyquist frequency for speech, both the original and filtered signals can be sampled and reconstructed with high accuracy, although filtering provides some noise suppression and smoothness



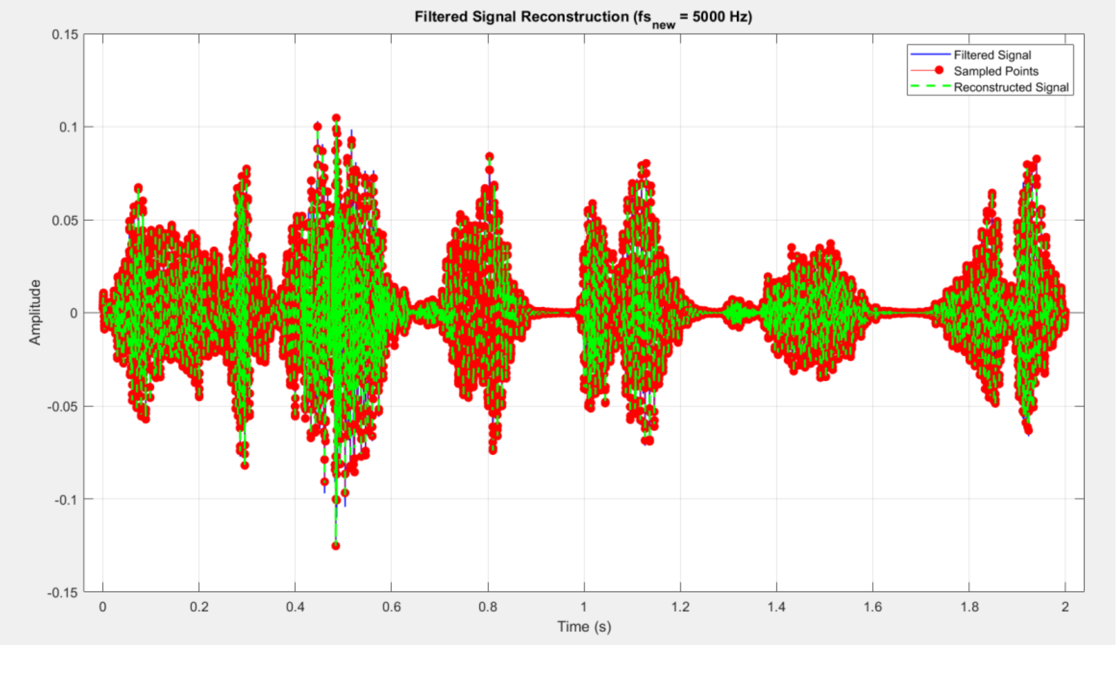


Figure 2: Sampling at f=1800Hz

This figure shows the reconstruction of both the filtered signal (bottom) and the original signal (top), sampled and reconstructed at a lower sampling rate of **1800 Hz**. Similar to the previous case at higher sampling frequency, the reconstruction still generally follows the original signal shape. However, because of the lower sampling rate, some of the higher frequency components are no longer captured as effectively, and finer details begin to blur or smooth out. Despite this reduction in captured frequency content, the overall waveform remains recognizable, and the reconstruction retains the primary features of the original signal, indicating acceptable reconstruction fidelity at this sampling rate. The filtered signal reconstruction shows slightly better performance because high-frequency content was already removed by the low-pass filter, reducing aliasing effects. Overall, both reconstructions remain recognizable, but fidelity is reduced at this lower sampling rate

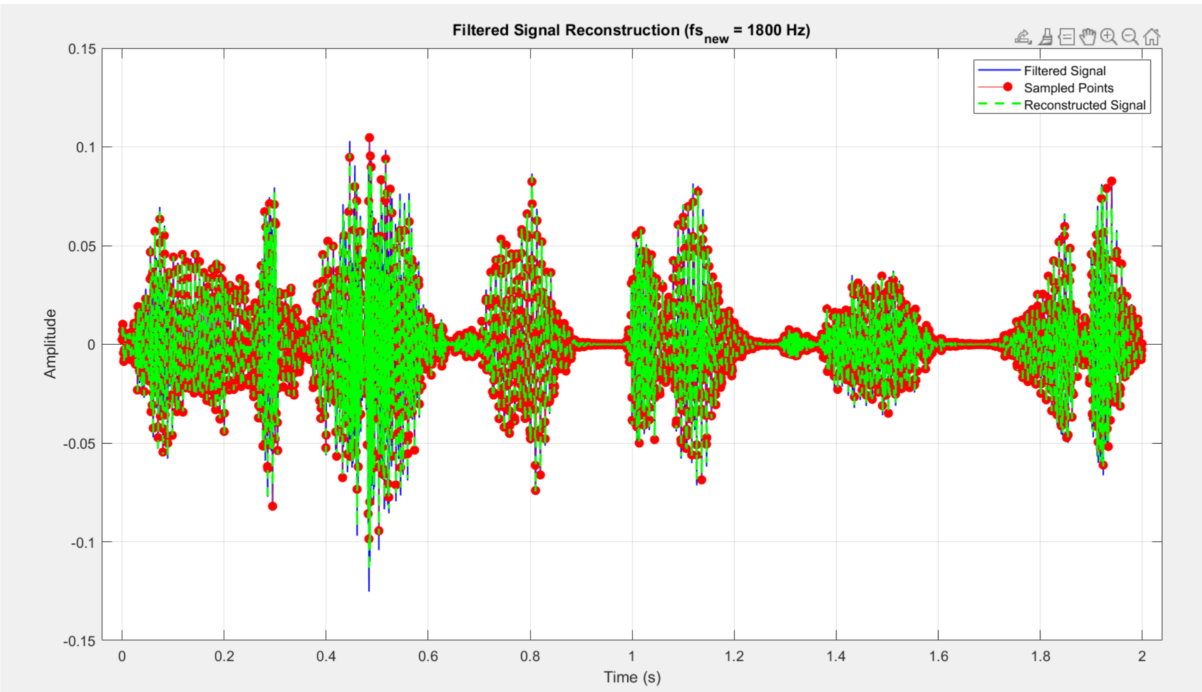
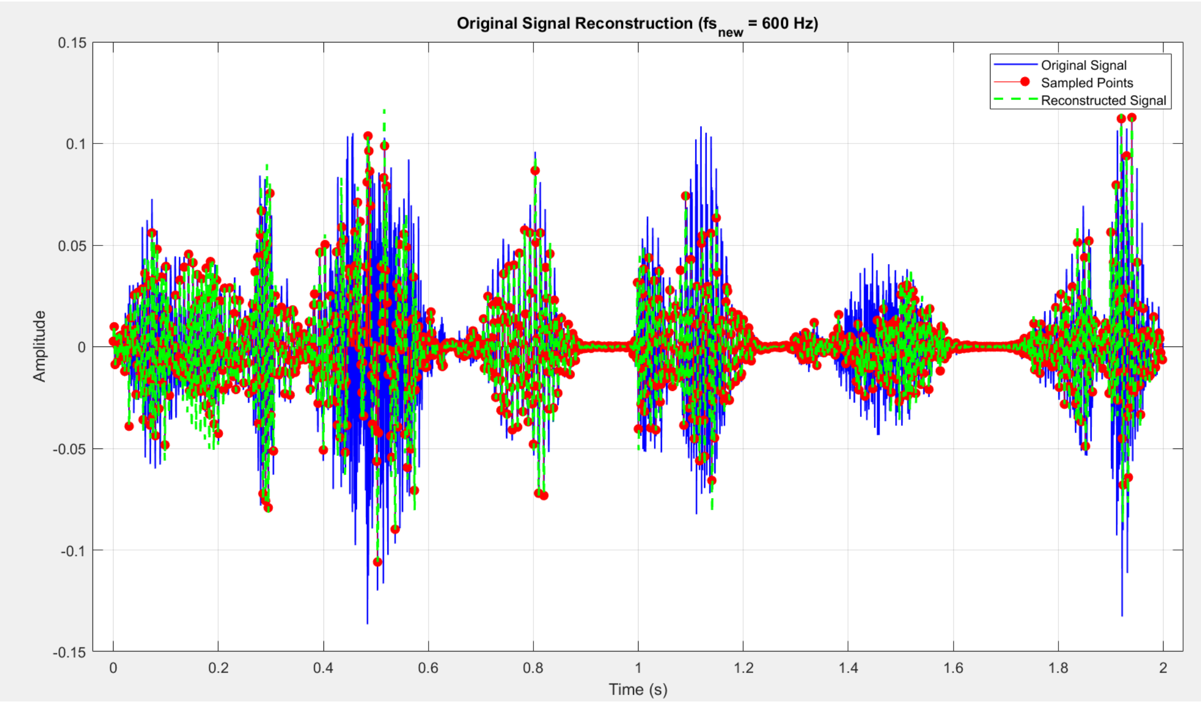
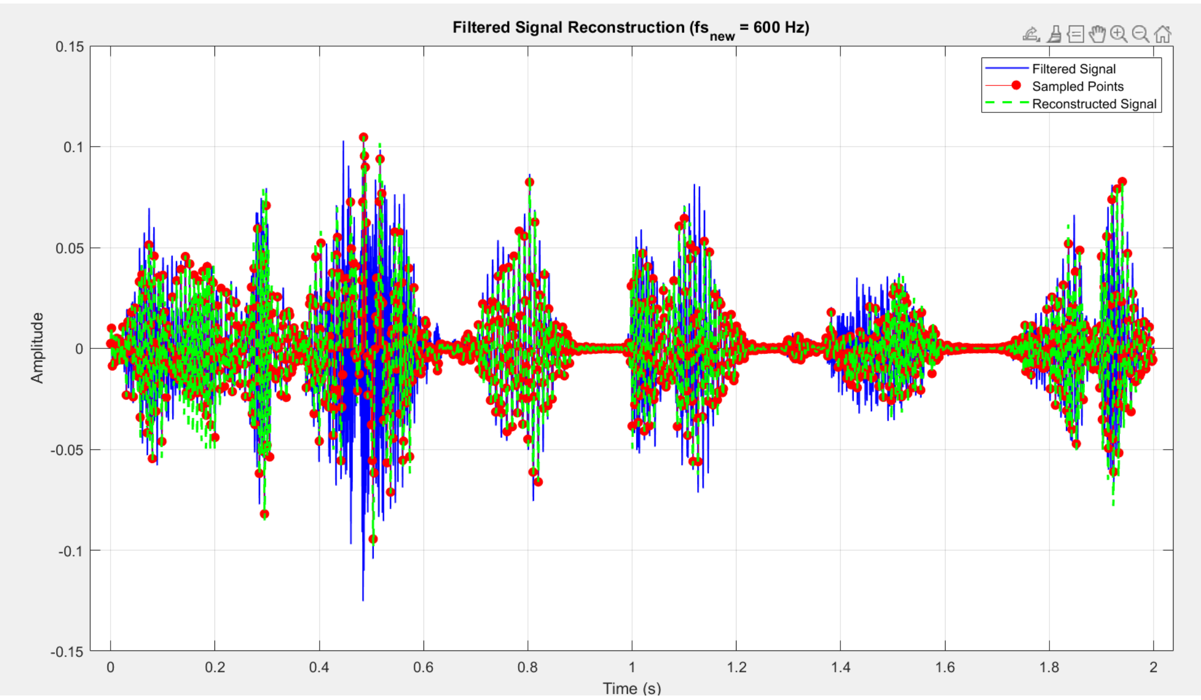


Figure 3: Sampling at f=600Hz

This figure shows the reconstruction of both the filtered signal (bottom) and the original signal (top), sampled and reconstructed at a sampling rate of **600 Hz**.At this low sampling rate, the reconstruction visibly **loses a lot of important detail** from the original waveform. The signal appears **missing fine variations**, especially in regions where the original waveform changes rapidly.We also observe that while the filtered version performs slightly better in retaining the general shape (because high frequencies were removed before sampling), **both signals suffer from aliasing** and undersampling effects. High-frequency components are not captured because the sampling rate is too low to meet the Nyquist criterion.This demonstrates that at **600 Hz**, the sampling rate is insufficient to preserve the full frequency content of the voice signal, leading to **loss of critical information and distortion** in the reconstruction. In speech or audio applications, such degradation would result in **loss of clarity and intelligibility**.





Conclusion on aliasing and error:

At fs=5000Hz:
At this high sampling rate, both the filtered and original signals are sampled well above the Nyquist rate for the signal's frequency content. The reconstructed signals closely follow the original signals in both cases, preserving fine details and waveform shape. The error is minimal, and no significant aliasing is observed. This demonstrates that sampling at a sufficiently high rate ensures high fidelity reconstruction.

At fs=1800Hz:
With a sampling rate of 1800 Hz, reconstruction still maintains a good approximation of the original signals, although slight distortions and smoothing effects become more noticeable, especially in the unfiltered signal. Aliasing begins to appear due to the reduced ability to capture frequency components near and above the Nyquist limit (900 Hz). The filtered signal performs better under this condition because the high frequencies were already attenuated, reducing aliasing artifacts and reconstruction error.

At fs=600Hz:
At this low sampling rate, significant aliasing occurs because the Nyquist frequency is only 300 Hz, well below the important frequency content of the original signal. As a result, the reconstructed signal loses many high-frequency details and exhibits visible distortion and smoothing. The filtered signal shows improved behavior relative to the unfiltered case, as high-frequency components that would alias were already suppressed by the low-pass filter. However, the loss of information is still evident, and reconstruction accuracy is notably reduced