EECE 340 Project - Section 3.0 Application and Analysis

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Introduction

In this step, we extract a meaningful time-domain segment from the recorded speech signal. The signal is a real-world voice recording sampled at 4000 Hz, representing a continuous-time speech waveform converted into discrete samples. We isolate a portion between 3 to 5 seconds. We compute the Fourier Transform of this segment to explore its frequency content and then sample and reconstruct the signal using our custom functions to observe the effects of sampling and reconstruction on a real-world voice signal.

MATLAB file description

voiceplot_ftr.m:

This script records a 6-second voice signal at 4000 Hz sampling rate using MATLAB's audiorecorder function. It plots the complete recorded signal and extracts a 2-second time-domain segment from 3 to 5 seconds for analysis. It visualizes the extracted segment and computes its frequency spectrum using our Fourier Transform function (ftr). The script saves the entire recorded signal and its sampling frequency to a .mat file ('my_recording.mat') for later processing.

reconstruct_voiceplot.m:

This script loads the full recorded voice signal and sampling frequency from 'my_recording.mat'. It extracts the same 3 to 5 second segment from the loaded recording, then applies the sample function (sample). It reconstructs the original signal from the sampled points via the reconstruct function (reconstruct), and visualizes the original segment, sampled points, and reconstructed signal together to compare sampling and reconstruction quality with the original signal.

Figures description

Figure 1: original voice sample

This plot shows the full 6-second recorded voice signal sampled at 4000 Hz. The amplitude varies over time, with distinct speech bursts and pauses corresponding to spoken words or syllables, demonstrating the dynamic and transient nature of human speech.

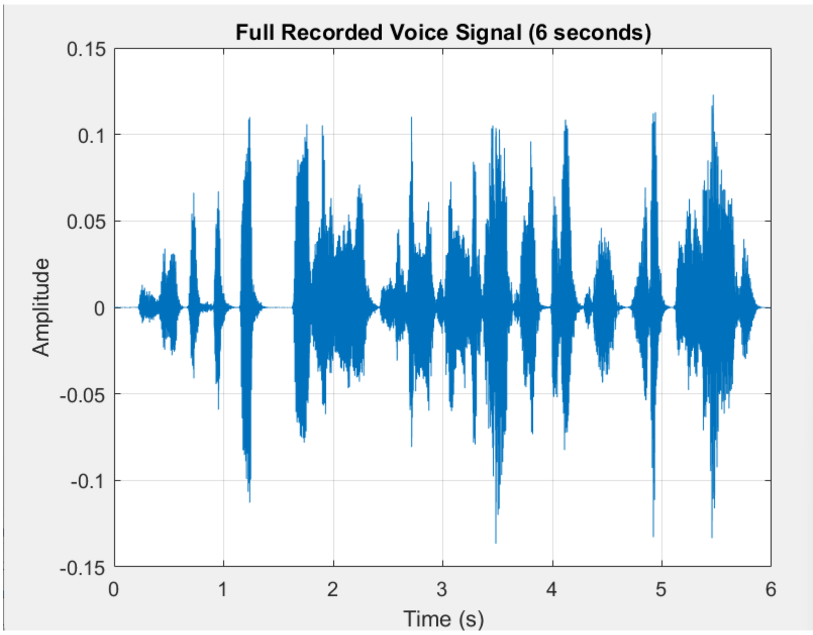


Figure 2: segment of voice sample

This plot shows the extracted 2-second voice segment from **3 to 5 seconds** of the original 6-second voice recording. The signal represents the amplitude of the recorded speech over time, sampled at **4000 Hz**. You can see bursts of higher amplitude, separated by lower- amplitude or near-silent regions (pauses or softer speech sounds).

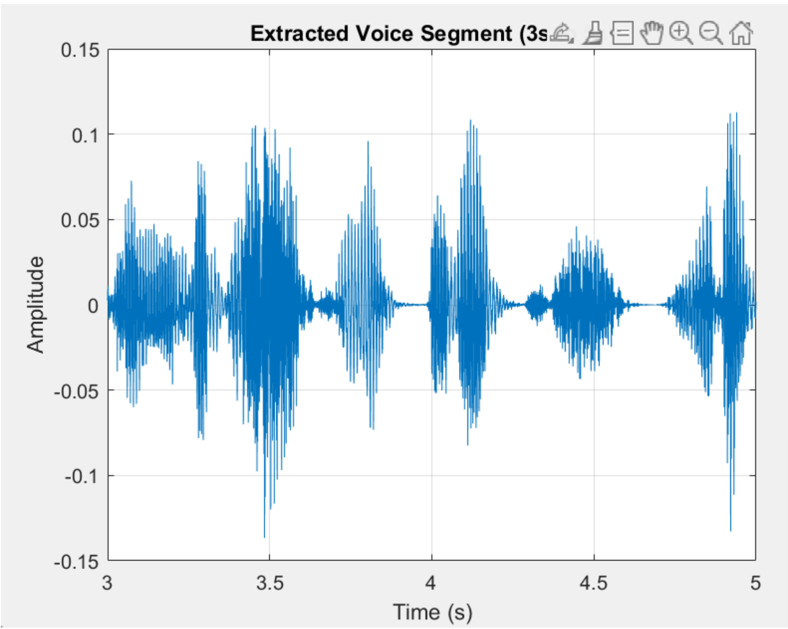


Figure 3: frequency spectrum

This plot shows the frequency spectrum of the extracted 2-second voice segment (from 3s to 5s), computed using the Fourier Transform and normalized in magnitude. The spectrum displays prominent peaks around **100 Hz, 200 Hz, 400 Hz and 600 Hz**. Most energy is concentrated below **1200 Hz**, with the higher frequency components (above ~1300 Hz) being relatively weaker and likely contributed by **background noise or recording artifacts** rather than meaningful speech content. Most of the **intelligible and important speech content** is found **below 1000 Hz.**

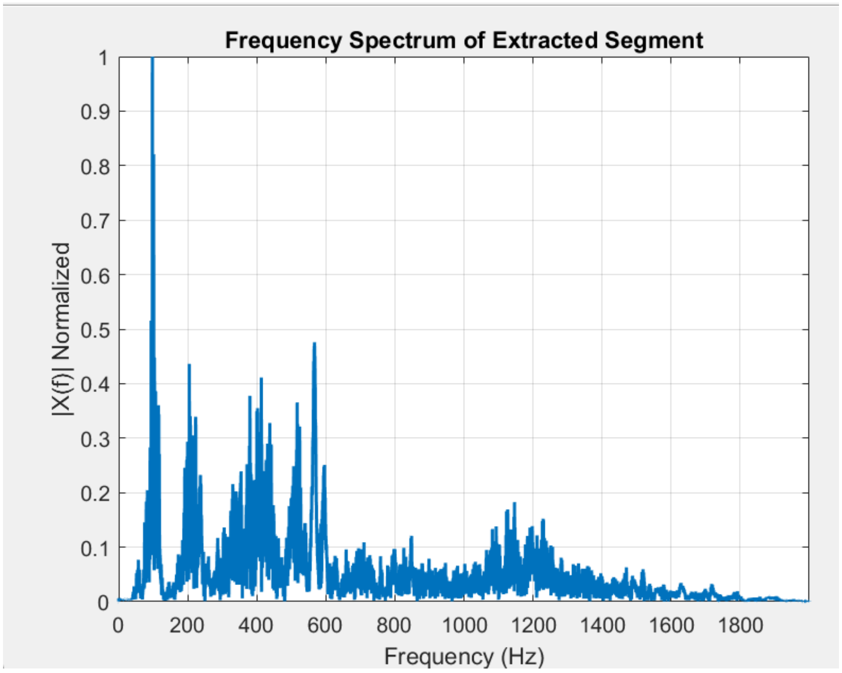
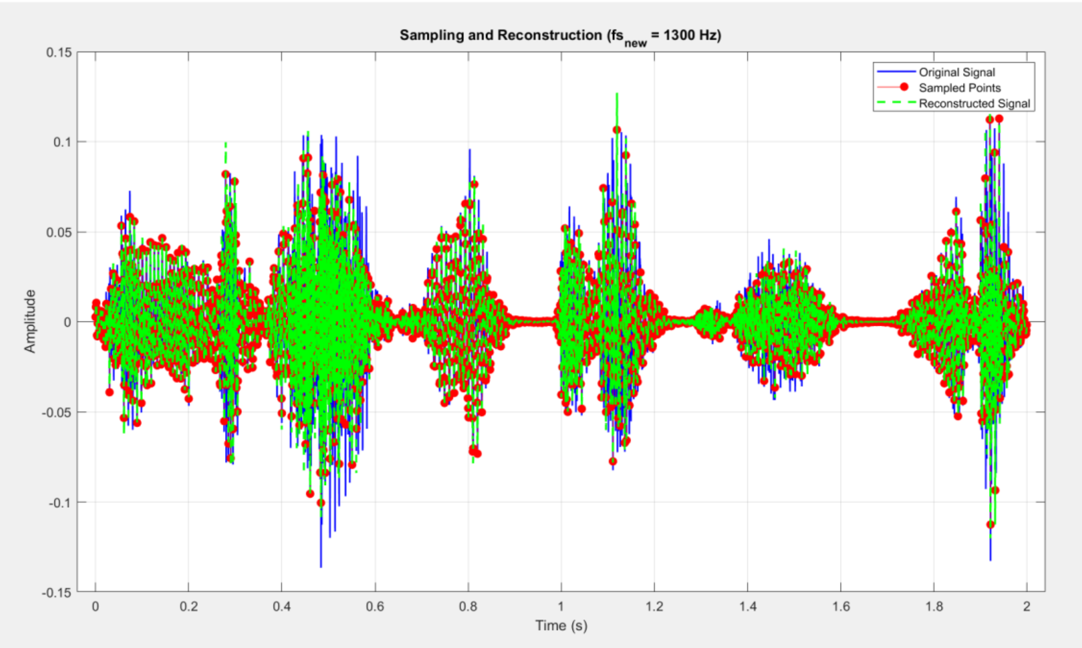


Figure 4: Sampling and reconstruction

This plot shows the original extracted signal (blue line), the sampled points at a reduced sampling rate of **fs_new = 1300 Hz** (red circles), and the reconstructed signal using sinc interpolation (green dashed line). The reconstructed signal closely follows the sampled points and approximates the original waveform, although some fine details and high-frequency content are lost due to the sampling rate. If we were to increase it further, then the reconstruction would become even more detailed (as seen in part 3.2 later).



Reconstruction errors:

The reconstructed signal, obtained using sinc interpolation from the sampled points, successfully approximated the overall shape of the original waveform but exhibited a noticeable loss of fine, high-frequency details. This reconstruction error arises from the inherent limitations of the sampling process: any frequency components in the original signal that exceed the Nyquist frequency (half the sampling rate) are not captured during sampling. As a result, these higher frequencies are either lost or folded (aliased) into lower frequencie. Since sinc interpolation can only reconstruct frequencies present in the sampled data, it is unable to recover information that was never captured or was distorted due to aliasing. Therefore, the reconstruction preserves the main features of the signal but lacks the full high-frequency content of the original continuous-time signal.

Aliasing:

In this experiment, the voice signal was recorded at a sampling frequency of **4000 Hz**, corresponding to a Nyquist frequency of **2000 Hz**. This sampling rate is sufficient to capture frequencies up to 2000 Hz without aliasing, based on the Nyquist-Shannon sampling theorem. The frequency spectrum of the recorded signal showed that most of the energy was concentrated below 1200 Hz, with additional weaker components extending toward 1600 Hz. Since the original signal’s frequency content did not exceed the Nyquist frequency, no significant aliasing is expected during the recording process at this sampling rate.

Impact on system design:

This experiment highlights how aliasing and reconstruction errors affect decisions in **sampling rate and bandwidth selection**. To avoid aliasing, the sampling rate must be at least twice the highest frequency present in the signal, following the Nyquist theorem. Lower sampling rates reduce data size but risk losing important frequency content and degrading reconstruction quality. Therefore, system designers must balance the trade-off between signal fidelity and system constraints based on the application's frequency requirements.