Name:
Class and Lab No.: EEE 304 Lab 1
Submission Date: 9-30-08

1. Write a paragraph explaining what have you learnt from this exercise.

The purpose of the exercise was to become familiar with the LabView environment by creating a VI file and front panel to load and display data from a speech file as seen in Figure 1 and Figure 2. The VI was designed to filter the speech using a low-pass filter and display the frequency spectrum of the filtered signal. I learned how to how to use LabView to play a sound file, break the sound file into frames, create a high-pass or low-pass filter, and perform spectral analysis using Fast Fourier Transform. I thought it was also beneficial to be able to control the cutoff frequency while listening to the audio file and observing the changes in the graph of the filtered signal. This showed me not only what kind of differences filtering could make on the signal, but on the audio perception of the signal as well.

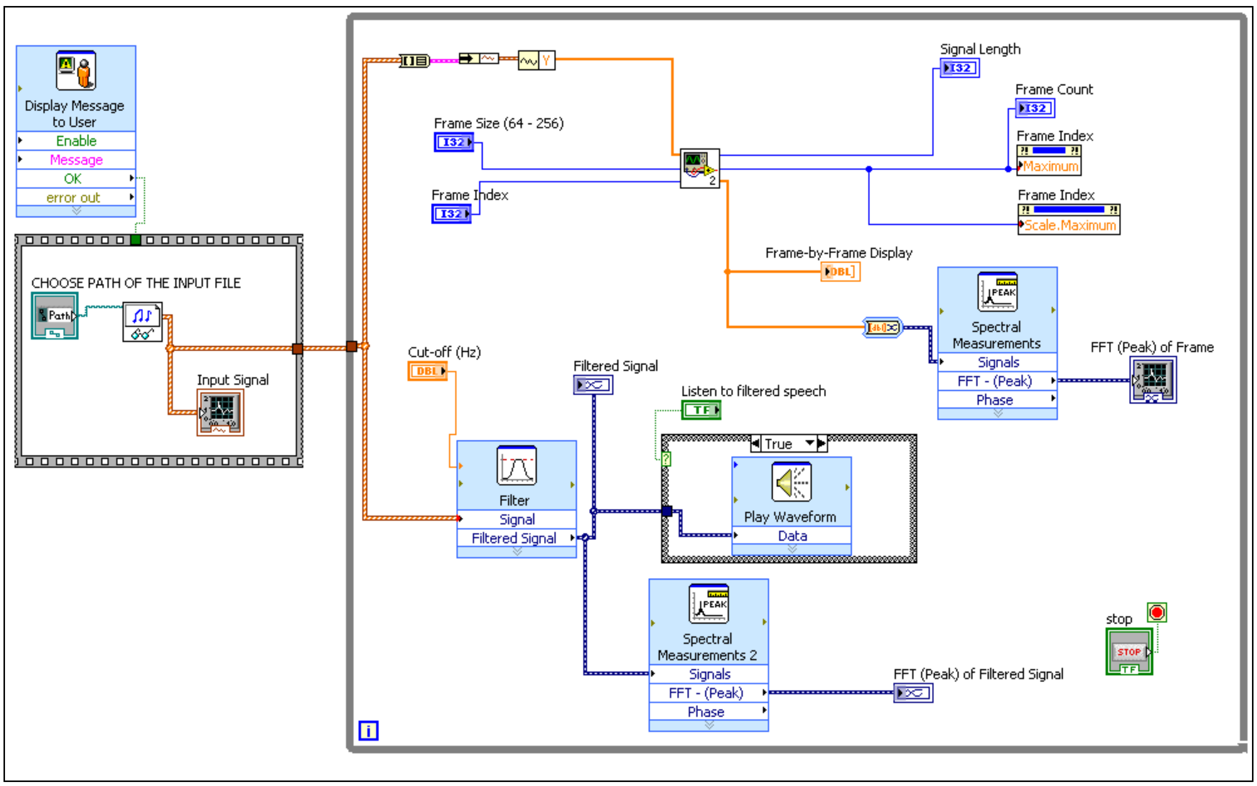

Figure 1. Block Diagram

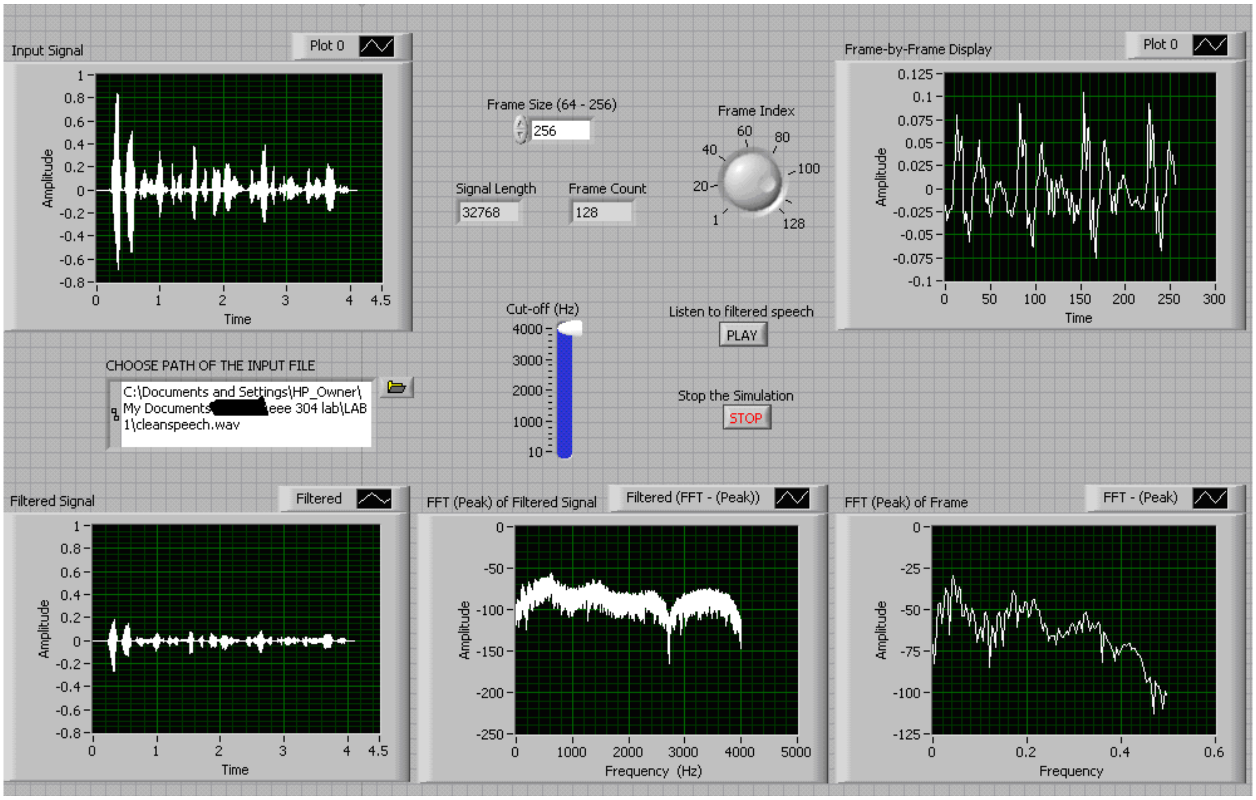

Figure 2. Front Panel

2. Write a paragraph explaining the sampling frequency, high pass and low pass filter.

The sampling frequency is the number of times per second a continuous signal is sampled, resulting in a discrete signal. The Nyquist-Shannon theorem states that a sampling frequency of twice the maximum frequency of the signal will allow the signal to be accurately reconstructed. If a sampling frequency lower than twice the maximum frequency is used, the signal will become aliased, causing the signals to become indistinguishable from each other and distorted. Equation 1 states this theorem, where f_S is the sampling frequency, and B is the bandwidth or bandlimit of the signal, for which there are no frequencies present above B.

$$f_s > 2B \qquad \text{Equation 1.}$$

This sampling frequency also dictates the cutoff frequency for high-pass and low-pass filters. As shown in Equation 2, the cutoff frequency, f_c, must be less than half of the sampling frequency.

$$0 < f_c < \frac{f_s}{2} \qquad \text{Equation 2.}$$

The low-pass filter passes frequencies up to a certain cutoff frequency and attenuates frequencies higher than the cutoff frequency. Conversely, the high pass filter passes frequencies above the cutoff frequency and attenuates frequencies below the cutoff frequency.

For this exercise, the sampling frequency of the input file is equivalent to the Nyquist frequency of 8000Hz. The bandlimit, or maximum frequency of the original signal, is 4000Hz. The maximum cutoff frequency must be 4000Hz as well.

3. What changes do you find in the speech when it is filtered? How do the characteristics vary with change in cut-off frequency?

At a cutoff frequency of 10Hz, the filtered signal graph showed that much of the original signal had been reduced. The FFT showed reduced amplitudes for the higher frequencies compared to lower frequencies. Low tones were dominant in the audio, because the high frequencies had been attenuated.

As the cutoff frequency increased, higher tones slowly became more present in the audio. The filtered signal became more similar to the input signal. At a sampling frequency of 4000Hz, the FFT showed that the most frequencies had the same amplitude, indicating that there was no attenuation caused by the cutoff frequency. This would be expected because the bandlimit of the original signal is 4000Hz.

The input signal, filtered signal, and FFT at 10Hz and 4000Hz can be seen in Figure 3 on the next page.

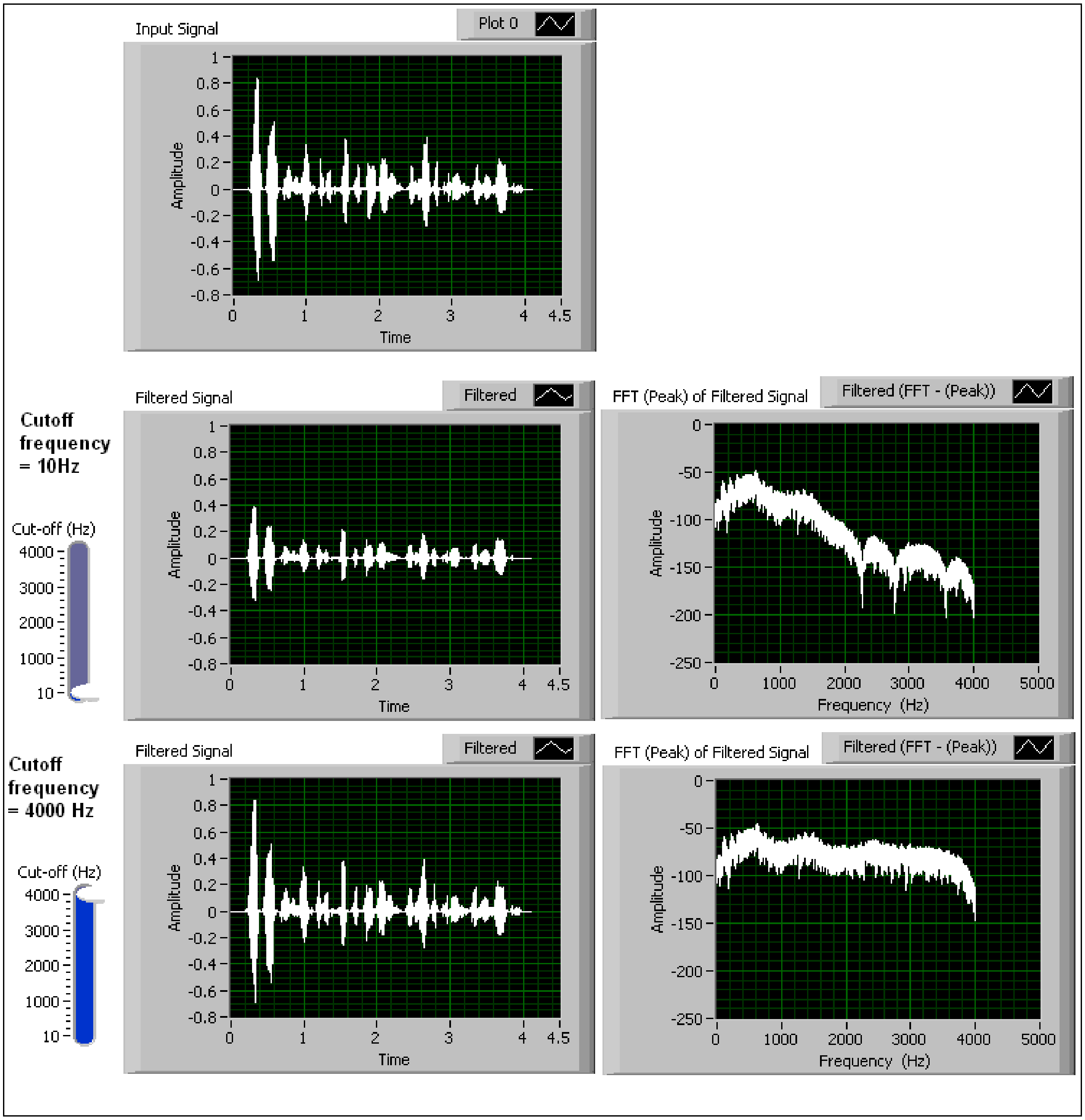

Figure 3. Low-pass filter

4. Replace the low-pass filter with the high pass filter. What difference do you observe with change in cut-off frequency?

When a high-pass filter was used, the cutoff frequency of 10Hz allowed the signal to pass through, seemingly unchanged. The filtered signal appears to be identical to the input signal. The FFT graph also shows that there was little attenuation. The high-pass filter at 10Hz is similar to the low-pass filter at 4000Hz in this case as it allows the entire speech signal to pass through unaltered.

As the cutoff frequency was increased, the filtered signal became smaller. The FFT showed that frequencies in the lower range were attenuated. This was also heard in the audio as higher tones became more dominant and the speech became more difficult to understand as the signal became smaller. The high-pass filter at high cut-off frequencies resulted in a smaller filtered signal at high frequencies than the low-pass filter at low cut-off frequencies. This is due to the frequency of male speech being closer to the lower end of the spectrum.

At a cutoff frequency of 4000Hz, the filtered signal surprisingly became larger, but still significantly smaller than the input signal. The FFT showed only slight attenuation near 2750Hz. The filtered audio was also similar to the original audio. This may have been due to an error in the filter, since the cutoff frequency of 4000Hz is also the sampling frequency and the maximum frequency present in the original signal.

The high-pass filter with cutoff frequencies of 10Hz, 3430Hz, and 4000 Hz can be seen in Figure 4 on the next page.

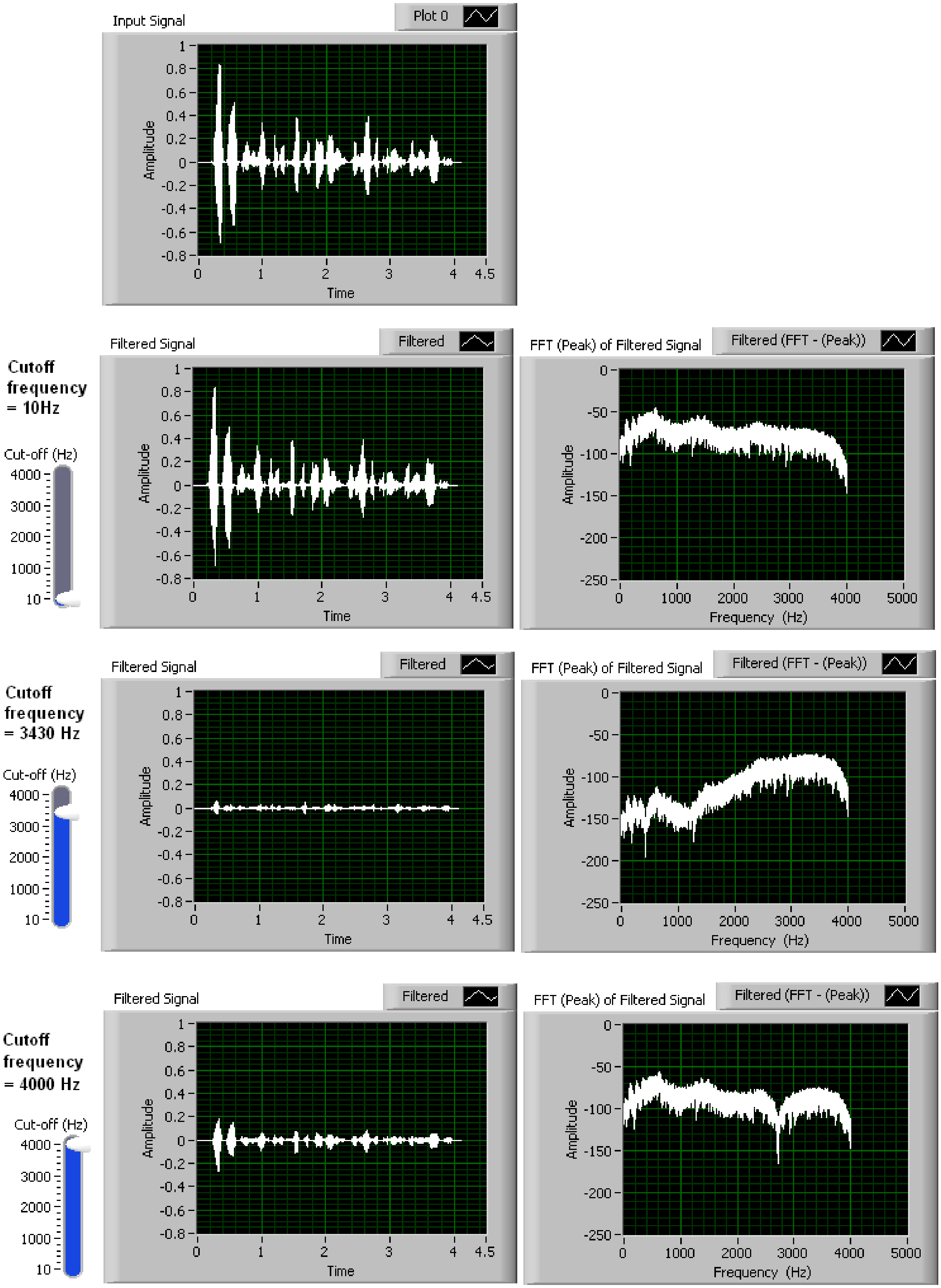

Figure 4. High-pass filter