Signal conditioning refers to change in an input signal to meet the requirements of a system for which the input is being used. The accepted forms for these inputs include: voltages, frequencies or charge from different types of sensors. This project uses a microphone which records a voice signal and deals with the frequency of human speech. The output from conditioning systems can be a voltage, frequency , resistance or any specialized output.

The signal conditioning system includes amplification , filtering, range matching and several other processes to modify the input signal.

This project requires an individual to speak into a microphone and the audio signal is then conditioned according to the needs of the project. The audio signal will include background noise, channel distortion and the valuable speech which is of meaning to the researchers. The positioning of the microphone i.e below or above the head and the distance of the microphone from the mouth also contribute to the quality of the signal. This audio signal will be amplified and then filtered to obtain a healthy signal.

The amplification is done to improve the measurement resolution of the signal and increase the Signal to Noise Ratio(SNR). This also assists in compensating for any weak voice of an individual or a microphone with a very low output signal. The output from the amplifier is then fed into the filter.

The process of removing the unwanted frequencies from the audio signal is called filtering. The human speech typically has a frequency response of 300 Hz to 3 kHz. This frequency band rejects most part of the noise it also rejects the plosive consonants like “p” and “t” which require a higher frequency to be correctly differentiated. This reason contributes to the selection of a higher frequency band which ranges over 300 Hz to 8 kHz[1]. Once the required frequency bandwidth is decided a filter is chosen. The filter selected for this project is a bandpass filter which will eliminate the frequencies before 300 Hz and beyond 8 kHz.

The output from the filter is then led into the signal processing system.

[1] Stevens, K. N. (1998). Acoustic Phonetics. Cambridge, MA: The MIT Press.