DIGITAL SIGNAL PROCESSING LABORATORY
E&ECE DEPARTMENT
IIT KHARAGPUR

EXPERIMENT No: 4

This experiment is slightly different than the rest of the experiments in that it requires slightly more involvement from your side and requires some qualitative measures by having someone else listen to the processed audio signal. Further it requires you to read a scientific paper and understand at least the main points of the signal processing performed.

Please read the paper: 1) https://www.ncbi.nlm.nih.gov/pubmed/7569981 available at: http://www.utdallas.edu/~assmann/hcs6367/shannon_zeng_kamath_wygonski_ekelid95.pdf, Shannon et al 1995, Speech recognition with primarily temporal cues, Science. 1995 Oct 13;270(5234):303-4.

A further interesting paper on slightly different aspects:
https://www.ncbi.nlm.nih.gov/pubmed/11882898 available at:
https://www.ee.columbia.edu/~dpwe/e6820/papers/SmithDO02-chimaeric.pdf Smith et al 2002, Chimaeric sounds reveal dichotomoies in auditory perception, Nature. 2002 Mar 7;416(6876):87-90.

The goal of the assignment is to implement a part of the paper (1) Shannon et al 1995 and try to gain understanding of relative importance of low frequency temporal structure of speech and frequency content of speech in speech perception. This work is the most important basis of processing done for cochlear implants. The details will be discussed in the introduction to the experiment.

Use the methods described in the paper (see Note 7 in the paper) to modify the provided speech signal fivewo.wav. Have 6 cases: 1 band, 2 bands, 3 bands, 4 bands and 8 bands and 16 bands– the filters should be logarithmically spaced and should span 90 Hz to 5.76 kHz (6 octaves). Thus for 1 band case the lower and higher cut-offs should be 90 Hz and 5.76 kHz, and for the 2 band case the 2 filters should be 90 Hz to 720 Hz and 720 Hz to 5.76 kHz. Use fourth order bandpass Butterworth (MATLAB function butter) filters instead of elliptic IIR and no need for the preemphasis filter. For the filtering operation use the filtfilt or filter function. For extraction of envelope use the Hilbert transform
(https://in.mathworks.com/help/signal/ug/envelope-extraction-using-theanalytic-signal.html) and then low pass filter (again use butter for the low pass filter) with cutoff of 240 Hz.

Create the new sounds and have someone who has not heard the sentence tell you what they hear (give comments). There is no need to do elaborate statistics with multiple speech sounds as in the paper. Get a qualitative idea whether intelligibility increases or not and by how many bands is the sound clearly understood.