A Review on Python Programming For Speech Processing of Hearing Aid

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ABSTRACT

This paper reviews the use of python developing audio signal processing applications also it discuss the probability use of python in signal processing application for hearing impairment. The overview of python language numpy, scipy, matplotlib is also discussed. Also speechpy is open source python package used for speech pre processing technique and post processing operations.

Keyword : - *Hearing aid, Hearing loss, Python.*

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1. INTRODUCTION

Hearing loss is a partial or total inability to hear. Hearing loss may occur in one or both ears. As per World Health Organization, 466 million People at Risk of Hearing Loss. Hearing impairment can be classified, on the basis of location of the defect in the auditory system, as conductive, sensorineural, and central losses. Conductive loss occurs due to an abnormality in the middle ear leading to poor transmission of the sound to the inner ear. Sensorineural loss is caused by pathology in the cochlea and/or due to degeneration of the auditory nerves. Central loss occurs due to inability of the brain in decoding the neural firings into meaningful linguistic information.

The terms hearing impaired or hard of hearing are usually reserved for people who have relative insensitivity to sound in the speech frequencies. The severity of a hearing loss is categorized according to the increase in volume above the usual level necessary before the listener can detect it Different vowel sounds are distinguished by unique sets of these resonances, or formant frequencies. The aim of the hearing aid is to amplify sound signals in such a way that they become audible for the hearing impaired person.

There are many applications in the field of audio Signal processing. Python is a high level, interpreted and general purpose dynamic programming language that focuses on code readability. Audio signal processing libraries are available for general purpose programming languages which provides a comprehensive array of signal processing tools. However, it generally takes more time to develop applications or prototypes in C/C++ or java language.

Basic Mechanism of Hearing and hearing aid

Hearing is the process by which humans use their ears to detect and perceive sounds. Ears are important for hearing and for controlling a sense of position and balance. The basic structure is divided into three parts.

1) Outer ear
2) Middle ear and
3) Inner ear

The outer ear receives the signal and directs it towards the middle ear. The middle ear performs three basic functions. Firstly it acts as an impedance matching network, secondly it acts as an amplifier, thirdly and most importantly it splits the signal into different frequencies.

The third part which is the inner ear works like a spectrum analyzer. It encodes the signal at different frequencies, makes different nerve cells resonate and transmit short pulses to the brain.

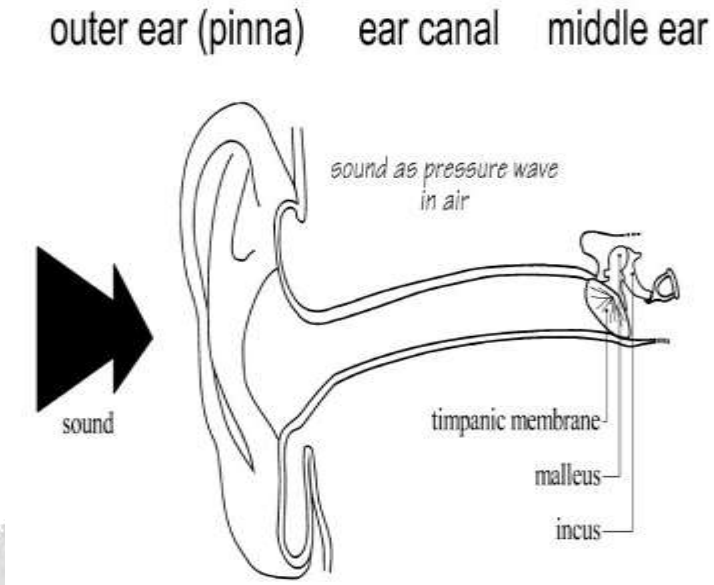


Figure: Hearing Mechanism of Human Ear

Audiogram:
The audiogram is a graph which gives a detailed description of your hearing ability and which can be described as a picture of your sense of hearing. The audiogram is a graph showing the results of a pure-tone hearing test. It will show how loud sounds need to be at different frequencies for you to hear them. The audiogram shows the type, degree, and configuration of hearing loss. When you hear a sound during a hearing test, you raise your hand or push a button. The audiogram illustrates your hearing ability by showing your hearing threshold at various frequencies. Hearing threshold is an indication of how soft a sound may get before it is inaudible. A hearing threshold of between 0 and 25 dB is considered normal. The list below outlines different hearing loss thresholds as they are determined in relation to an individual with a normal hearing threshold.

1. Mild hearing loss: 25 to 40 dB higher than normal.
2. Moderate hearing loss: 40 to 55 dB higher than normal.
3. Moderate-to-severe hearing loss: 55 to 70 dB higher than normal.
4. Severe hearing loss: 70 to 90 dB higher than normal
5. Profound loss: 90 dB or more.

How to Read an Audiogram

The vertical axis of the audiogram represents sound volume or intensity, which is measured in decibels (dB). The more one moves down the axis, the louder the sound becomes. This corresponds to turning up the volume on a radio. Zero decibel at the top of the axis represents the softest sound a person is normally able to hear and is not an indication that you cannot hear any sounds at all.

The horizontal axis of the audiogram represents sound frequency or pitch measured in Hertz (Hz). Sound frequency increases gradually the further one moves to the right along the axis. This movement can be compared to playing on the left side of a piano and gradually moving to the right side where the tone becomes more and more high-pitched. Frequencies between 500 Hz and 3000 Hz are most commonly used during ordinary conversation. During a hearing test the results are recorded on the audiogram by means of red Os for the right ear and blue Xs for the left one. The resulting red and blue lines show your hearing threshold for each ear, and the results may well differ.

Figure : Structure of audiogram

Types of Hearing Aid

1 Behind-the-ear
Behind-the-ear (BTE) hearing aids they rest behind the ear and send sound into the ear in one of three different ways, This is known as an open ear fitting. Because it blocks the ear less than an ear mould, it can give a more natural sound and is less noticeable. But it is only suitable if the hearing loss is mild or moderate.

2 With a receiver in the ear
Also known as Receiver in the ear. The sound is send via a discreet, hair-fine wire to the receiver seated in the ear canal; close to the eardrum. BTE hide behind the Auricle and have a thin clear tube that runs into the ear canal.

3 In-the-ear and in-the-canal
In-the-ear (ITE) hearing aids and in-the-canal (ITC) hearing aids have their working parts in the ear mould, so the whole aid fits into the ear. ITC aids are smaller and less visible as they fit right inside the ear canal. The custom hearing aid that fits within the outer portion of the ear. suitable for slight to moderate hearing loss.

4 Completely-in-the-canal
Completely-in-the-canal (CIC) hearing aids placed deep into the ear canal and are almost invisible. They are less visible unless someone looks closely at the ear. CIC are most suitable for people who are experiencing mild hearing loss.

5 Invisible-in-the-canal
Invisible-in-the-canal (IIC) hearing aids are fitted very deeply in the ear canal. Some IIC models stay in the ear for a few months at a time and can only be removed by the audiologist, who will maintain and clean the aid. IIC hearing aids are suitable for mild to moderately severe hearing loss.

Python

Python is one of the popular high-level programming languages used in an extensive variety of application domains. Python is a high level, interpreted and general purpose and open source programming language that runs on many platforms including Linux, Mac OS X and Windows. It is widely used and actively developed, and integrates with many other programming languages, frameworks. Python code is often said to be almost like pseudocode, since it allows to express very powerful ideas in very few lines of code while being very readable.

Some important features of the language include:

- A simple language which is easier to learn
- Large standard libraries to solve common tasks
- Simple Elegant Syntax
- Free and open source
- Portability

Python for Scientific Computing:
Three packages that are widely used for performing efficient numerical calculations and data visualization using Python. Example programs that make use of these packages are given

NumPy: Numpy is the core library for scientific computing in Python. Numpy is a library for the Python programming language, adding support for large, multi-dimensional arrays and matrices, along with a large collection of high-level mathematical functions to operate on these arrays.

SciPy: Numpy provides a high-performance multidimensional array and basic tools to compute with and manipulate these arrays. SciPy builds on this, and provides a large number of functions that operate on numpy arrays and are useful for different types of scientific and engineering applications.

Matplotlib: Matplotlib is a plotting library. Matplotlib is a library of 2-dimensional plotting functions that provides the ability to quickly visualize data from NumPy arrays, and produce publication-ready figures in a variety of formats. It can be used interactively from the Python command prompt, providing similar functionality to MATLAB or GNU Plot. It can also be used in Python scripts, web applications servers or in combination with several GUI toolkits.

The Design of Frequency Sampling of Finite Impulse Response Filter, is developed. Python platform is chosen because it is the tool of choice for most educational and research purposes and it provides powerful computation and advanced visualization tools.

II METHODOLOGY

FIR FILTER DESIGN METHOD

FIR filters are filters having a transfer function of a polynomial in *z*- and is an all-zero filter in the sense that the zeroes in the *z*-plane determine the frequency response magnitude characteristic. The *z* transform of a *N*-point FIR filter is given by

$$H(z) = \sum_{n=0}^{N-1} h(n) z^{-n}$$

FIR filters are particularly useful for applications where exact linear phase response is required. The FIR filter is generally implemented in a non-recursive way which guarantees a stable filter.
In actual procedure for designing digital FIR filters first, the desired filter responses are characterized, and the filter coefficient values are calculated for a causal FIR filter. There are different methods to find the coefficients of digital filter from frequency specifications.
a). Fourier series method
b). The window method
c). Frequency sampling method
d). Optimal filter design method

Frequency Sampling Method

In this method the given frequency response is sampled at a set of equally spaced frequencies to obtain *N* samples. Thus, sampling the continuous frequency response *Hd* (*w*) at *N* points essentially gives us the *N*-point DFT of H(2pnk/N). Thus by using the IDFT formula, the filter coefficients can be calculated using the following formula:

$$h(n)=\frac{1}{N}\sum_{n=0}^{N-1} H(k)e^{j(2\pi n/N)k}$$

There are two different set of frequencies that can be used for taking the samples. One set of frequency samples are at $fk = k/N$ where $k = 0,1,\ldots.N-1$.

The other set of uniformly spaced frequency samples can be taken at $fk = (k + ½)/N$ for $k = 0,1,\ldots.N-1$.

The second set gives us the additional flexibility to specify the desired frequency response at a second possible set of frequencies. Thus a given band edge frequency may be closer to type-II frequency sampling point that to type-I in which case a type-II design would be used in optimization procedure.

The main attraction in this method is that it allows for a recursive realization of FIR filters, which can be computationally very efficient. However, it also lacks flexibility in specifying or controlling filter parameters. The window method is basically used for the design of prototype filters like the low-pass, high-pass, band-pass etc. They are not very suitable for designing of filters with any given frequency response. On the other hand, the frequency sampling technique is suitable for designing of filters with a given magnitude response. The ideal frequency response of the filter is approximated by placing appropriate frequency samples in the *z*- plane and then calculating the filter co-efficients using the IFFT algorithm.

Implementation

Implementation of a filter is done using python. Python is a software package for high performance numerical computation, visualization and programming environment.

File input/output (scipy.io):

Provides functions for reading and writing files in many different data formats, including .wav, .csv and matlab data files (.mat).

Signal processing (scipy.signal):

Provide implementations of many useful signal processing techniques, such as waveform generation, FIR and IIR filtering and multi-dimensional convolution.

scipy.io.wavfile.read:

Open a WAV file. Return the sample rate (in samples/sec) and data from a WAV file.

scipy.io.wavfile.write(filename**,** rate**,** data**):**

Write a numpy array as a WAV file.

Scipy.signal.firwin:

This function computes the coefficients of a finite impulse response filter. The filter will have linear phase; it will be Type I if numtaps is odd and Type II if numtaps is even.

Three different functions i.e. fir1, fir2 and kaiserord are used to design FIR filters.

Fir1 function implements the classical method of windowed linear-phase FIR digital filter design. It is used for design of filters in standard lowpass, highpass, bandpass, and bandstop configurations.

Fir2 function is used for designing of frequency sampling-based digital FIR filters with arbitrarily shaped frequency response.

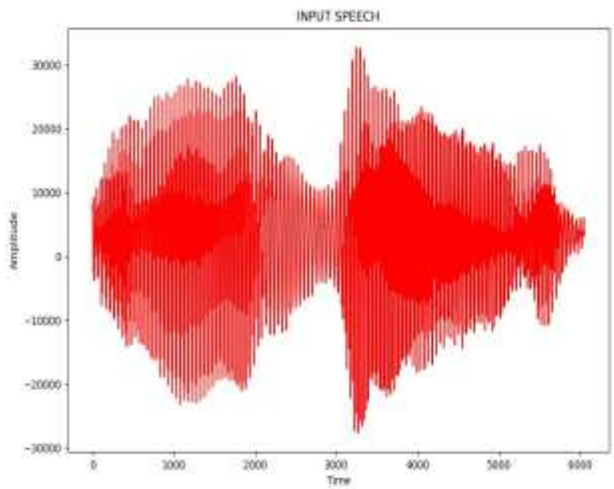
Kaiserord function returns a filter order n and beta parameter to specify a Kaiser window for use with the fir1 function. Given a set of specifications in the frequency domain, kaiserord estimates the minimum FIR filter order that will approximately meet the specifications. Kaiserord converts the given filter specifications into passband and stop band ripples and converts cutoff frequencies into the form needed for windowed FIR filter design.

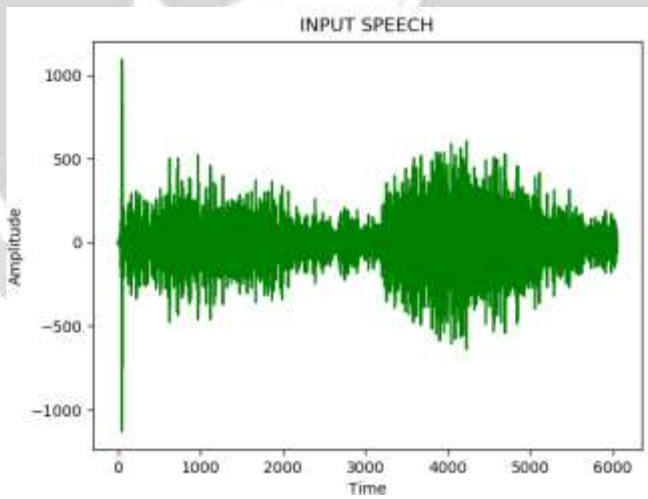
numtaps : Length of the filter

cutoff : Cutoff frequency of filter. In the latter case, the frequencies in cutoff should be positive and monotonically increasing between 0 and fs/2. The values 0 and fs/2 must not be included in cutoff.

III. RESULTS

In this work, speech signal was presented over the headphones or taking from .wav file. The original signal as well as the modified signal was presented in terms of time and amplitude spectrum and as audio file. The ABA speech original and modified signals is shown in below figure.





IV. CONCLUSION

In this work, the types of filter banks for hearing aids are studied and found that non uniform filter bank provides better solution. The frequency sampling technique has been studied for the speech signal processing algorithm. Python has been described for the use of signal processing.

The frequency sampling method is used to design non recursive FIR filters for both standard frequency selective and filters with arbitrary frequency response. The proposed algorithm will be implemented on Python software and Low pass filter implemented in python.. To get results of original ABA speech signal and modified signals .

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