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Python based Frequency Sampling Technique Algorithm for Hearing Impairment

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Abstract: Hearing is taken into account because the most vital senses of humans connects us to the world always and the communication between humans are mostly by speaking and listening. As per World Health Organization, 466 million People at Risk of the hearing impairment across the world. It adversely affects physical, behavioral, and social functions, as well as the general quality of live. The most task of the hearing aid is used to by selection amplify the audio sounds such the processed sound can be audible for hearing impairment.

In this work, frequency sampling method is used to design a filter for hearing impairment. The proposed algorithm is implemented using python programming. The sound is categorized into two ways vowels and consonants. Vowels have low frequencies that means high pitch and are easier to hear. Consonants have higher frequencies and harder to hear. The changes in formants and pitch is observed which would be useful speech for hearing impaired.

Keywords: Hearing loss; Python software; PRAAT software: Specch words;

I. INTRODUCTION

Hearing loss occurs in each ears. As per World Health Organization, 466 million People at risk of Hearing Loss across the world. Hearing loss can be classified into three types such as conductive, Sensorineural and mixed hearing loss on the basis of location of severity.

Hearing impaired is inability to hear or sound in the speech frequencies. The main function of the hearing aid is used to by selection amplify the audio sounds such the processed sound can be audible for the hearing impaired person. The classification of sounds are vowels and consonant. Vowels have low frequencies that means high pitch and are easier to hear. Consonants have higher frequencies that means low pitch and harder to hear.

The speech is a random in nature generated and which consists of complicated and easy resonant frequencies called Formants. The importance of calculating formant frequencies gives an idea to finding out utterances in speech and also the voice quality.

we are analyzing the formant frequencies, pitch from the speech samples that means nonsense speech words and analyzed using PRAAT spectrogram technique. The pitch and formant frequencies are detected for different speech words or signals for original and modified signals.

A. Basic Mechanism of Hearing and Hearing aid

Hearing is the process by which humans use their ears to detect the sounds. Ears are important organ. The basic structure is divided into three parts which is.

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

The outer ear receives the signal and forwards to the middle ear. We hear a sound depends on the power of the sound as well as on the frequency of the vibration.

The cochlea is main organ of hearing which is in the inner ear. They receive sound waves and passes them on to the brain. Then the brain analyzes this Sound waves are created when an object moves towards the pinna. The cochlea is the principal functional unit of the inner ear. The cochlea's basilar membrane divides the input signal into different frequencies. The location of internal hair cells along the basil membrane determines the optimum response of the hair cells to totally different frequencies. Once the sound signal is distributed within the sort of travel wave in tube, the hair cells on the apex react to low frequencies, whereas the hair cells at the lowest answer high frequencies.

B. What is Hearing loss or Hearing Impairment

Hearing impairment is inability to hearing or exhausting of hearing. hearing impairment is also permanent or temporary, and already gift at birth, or any unwellness. Figure 1 shows the degree of hearing impairment. A hearing threshold of between 0 and 25 decibels (dB) sound unit is taken into account as normal hearing. [9]

Degree of hearing impairment is following vary.

- Mild hearing loss: Above 21 to 40 decibels.
- Moderate hearing loss: Above 41 to 60 decibels.
- Severe hearing loss: Above 61 to 80 decibels.
- Profound hearing loss or deafness: Above 90 decibels

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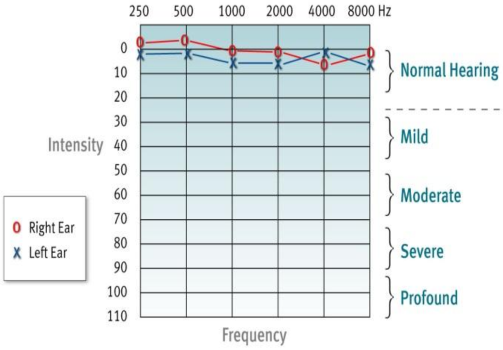

Figure1: Degree of hearing impairment

C. Audiogram

The representation is graph that is showing our hearing ability, a sign of soft sound might get before it's supersonic is Hearing threshold. Audiometry tests will sight whether or not you have got sensorineural hearing impairment (damage to the nerve or cochlea) or semiconducting hearing impairment (damage to the tissue layer or the little bones). The vertical axis represents sound volume or intensity, and also the horizontal axis represents sound frequency or pitch measured in Hertz (Hz). it'll show however loud sounds got to be at totally different frequencies for you to listen to them [9].

An representation report is employed to see the following:

-The presence of hearing impairment

-The sort of hearing impairment.

-The configuration of hearing impairment (which frequencies square measure specifically affected).

-The degree of hearing impairment (the severity)

-Whether or not hearing impairment is unilateral(one side) or bilateral (both sides).

D. Types of hearing impairment

1) Conductive hearing impairment

In conducting hearing impairment there is a tangle with the Outer or cavity. It is caused by an excessive amount of cerumen, Ear Infections[1].

2) Sensorineural hearing impairment

In Sensorineural hearing impairment happens once the hearing organ, the tube, and/or the eighth cranial nerve is broken and might not send the electrical info to the brain. Sensorineural hearing impairment is often permanent. It is genetic or caused by the natural aging method, diseases, accidents or exposure to loud noises [1].

3) Mixed hearing impairment

Mixed hearing impairment is a combination of conducting hearing impairment and Sensorineural hearing impairment. The Sensorineural hearing impairment part is permanent, whereas the conducting hearing impairment part will either be permanent or temporary [1].

E. Hearing Aid

Hearing aid is to amplify the sound signals in such how that they become perceptible for the hearing impaired person. Hearing aids square measure utilized by the part deaf however not the utterly deaf. The hearing aid is placed either in and or round the ear. Digital hearing aids square measure different; they convert sound waves to digital sounds, manufacturing a definite duplication of every sound, rather than simply amplifying it. The digital technology wont to improve performance in sure environments, reducing ground noise and racket. They need four major parts: the electro-acoustic transducer, battery, amplifier, and receiver.

Types of Hearing Aid

1 Behind-the-ear
2 With a receiver in the ear
3 In-the-ear and in-the-canal
4 Completely-in-the-canal
5 Invisible-in-the-canal

II. METHODOLOGY

A. Fir Filter Design Technique

FIR filters are helpful for applications wherever actual linear section response is needed. The FIR filter is largely implemented in associate extremely non recursive technique that guarantees a stable filter.

There are alternative ways to realize the coefficients of digital filter from frequency specifications.

a). Fourier series technique
b).The window technique
c). Frequency sampling technique
d). Optimal technique

B. Frequency Sampling Technique

FIR filters for each customary frequency selective and filters with capricious frequency response.

In the frequency sampling technique, the specified frequency response Hd(e^{jw}) is sampled at a collection of uniformly spaced frequencies,

$$w=2\pi k/N \qquad (1)$$

where k= 1,2,....N-1/2 for N odd
k=1,2,...N/2-1 for N even.

Thus by using the IDFT formula, the filter coefficients may be calculated using the formula:

$$h(n)=1/N \sum_{n=0}^{N-1} H(k)\, e^{(2\pi n/N)k} \qquad n = 0; 1,2….N-1 \qquad (2)$$

The frequency response Hd(e^{jw}) calculated using N-point Finite Impulse Response (FIR) h(n), will coincide with Hd(e^{jw}) at w = 2πk/N and is given as

$$H(e^{jw}) = \sum_{n=0}^{N-1} h(n)\, e^{-jwn} \qquad (3)$$

Two completely different set of frequencies that is employed for taking the samples. One set of frequency samples are taken at fk=k/N, wherever k/N, wherever k=0,1,.N-1. And different set of uniformly spaced frequency samples may be taken at fk=(k+1)/N for k = 0,1,.N-1.

The second set which is the extra flexibility to specify the specified frequency response at a second doable set of frequencies. we get the unit impulse response,

$$h(n)= IDFT[H(k)],$$

Wherever IDFT is the Inverse distinct Fourier rework.

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The inverse DFT then yields an impulse response which can result in a filter whose frequency response constant as that of the specification specifically at the placement of the frequency samples. The advantage of this technique is that we are able to style filters directly within the frequency domain, however

the disadvantage is that the frequency will solely be number solely be number times of 2/N, and that we cannot guarantee a random cutoff frequency.[15]

III. IMPLEMENTATION

Implementation is done into two parts

1) Implementation using MATLAB
2) Implementation using Python programming

1) Implementation using MATLAB

The MATLAB program is implemented for the processing of speech signals using frequency sampling algorithm. The various steps used for programming are as follows;

-Get an input signal.
-Apply frequency sampling method on it.
-Break up samples into frames.
-Apply the windowing function to each frames.
-Apply Firwin2 function. Apply convolution to get modified signal.

2) Implementation using Python programming

Implementation of a filter for hearing impairment is completed using python programming. Python may be a package for top performance programming atmosphere. Python programming may be a high level, and open source programing language that runs on several platforms. Python code is specific terribly powerful ideas in only a few lines of code while being very readable. Some necessary options of python language are following:

-A straightforward language
-Large customary libraries
-Portability
-Free and open source.

Python for Scientific Computing

Three packages that square measure wide used for playing economical numerical calculations and information visual image mistreatment Python and that build use of those packages square measure following

NumPy

NumPy is that the core library for scientific computing for the Python programing language, adding support for multi-dimensional arrays and matrices, together with an oversized assortment of high-level mathematical functions to work on these arrays.

SciPy

NumPy is provides a superior two-dimensional array and basic tools to calculate with and manipulate these arrays. SciPy provides an oversized variety of functions that treat numpy arrays and square measure helpful for various sorts of scientific and engineering applications.

Matplotlib

Matplotlib may be a plotting library in python programming. Matplotlib may be a library of 2-dimensional plotting functions that gives the power to quickly visualize information from NumPy arrays, and manufacture publication-ready figures in a very sort of formats. It may be used interactively from the Python prompt, providing similar practicality to MATLAB or wildebeest Plot. It can even be employed in Python scripts, net applications servers or together with many GUI toolkits.

The filter is designed using the Frequency Sampling method of Finite Impulse Response Filter. Python platform is chosen as a result of it is the tool of alternative for many academic and analysis functions and it provides powerful computation and advanced visual image tools.

File input/output (Scipy.io):

Provides functions for reading and writing files in many various information formats like .wav, and MATLAB (.mat) information files. File input/output (scipy.io) Provide implementations of the various useful signal method technique, like wave FIR and IIR filtering and multi-dimensional convolution.

Scipy.io.wavfile.read:

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

Scipy.io.wavfile.write:

Write a numpy array as a WAV file.

Scipy.signal.firwin:

This operate computes the coefficients of a finite impulse response filter. The filter will have linear phase, it will be kind I if numtaps is odd and kind II if numtaps is even. Three different functions i.e. fir1, fir2 and kaiserord square measure accustomed vogue FIR filters.

Fir1 operate implements the classical methodology of windowed linear-phase FIR digital filter vogue. It is used for type of filters in traditional lowpass, highpass, bandpass, and bandstop configurations.

Fir2 operate is used for arising with of frequency sampling based digital FIR filters with each that method fashioned frequency response.

Kaiserord operate returns a filter order n and beta parameter to specify a Emperor window to be used with the fir1 operate.

Numtaps:

Length of the filter.

Cutoff:

Cutoff frequency of filter, the frequencies in cutoff need to be positive and monotonically increasing between zero and fs/2. The values zero and fs/2 shouldn't be enveloped in cutoff.

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IV. RESULT AND DISCUSSION

Initially we design filter in MATLAB and then it convert into Python language according to python libraries or command.

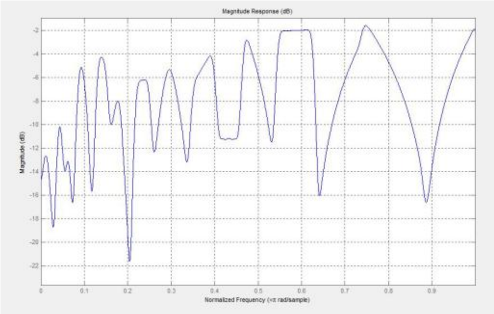

Figure 2: Original ABA speech word

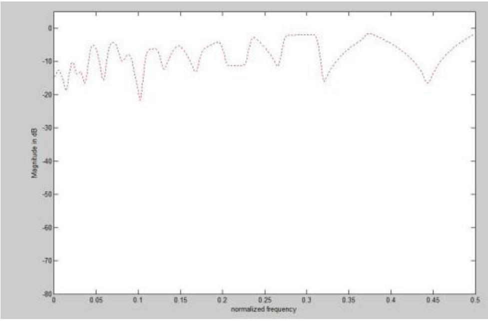

Figure 3: Modified ABA speech word

In this work, speech signal was taking from .wav file. the first signal furthermore because of the modified signal was given in terms of amplitude and time. The ABA and APA and ASA speech signal of original and changed signals is shown in below.

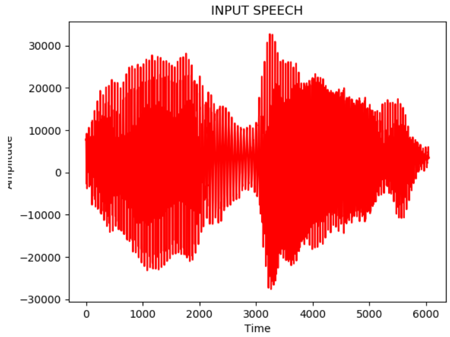

Figure 4: Original ABA speech word

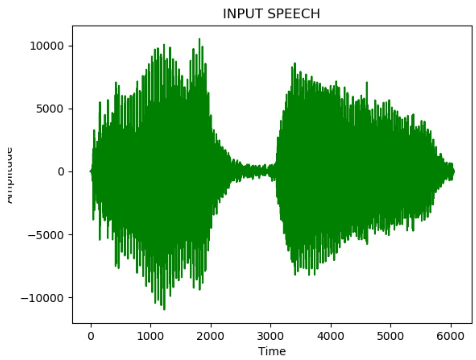

Figure 5: Modified ABA speech word

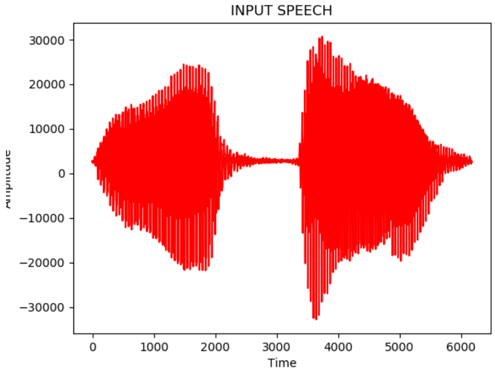

Figure 6: Original APA speech word

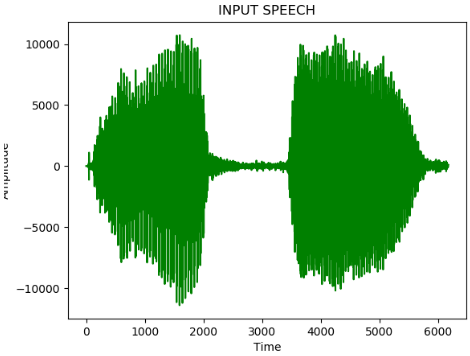

Figure 7: Modified APA speech word

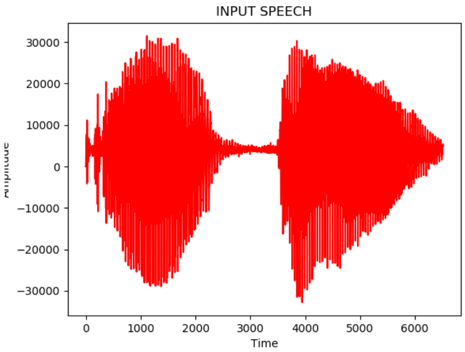

Figure 8: Original ASA speech word

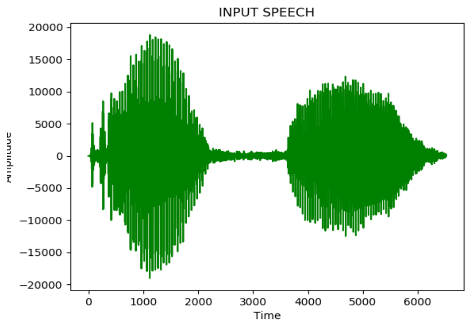

Figure 9: Modified ASA speech word

We have also use the ***PRAAT*** software to plot spectrograms of our waveforms, Figure 10, 11, 12 shows the spectrogram of ABA speech signals. Using PRAAT we can calculate formants and pitch frequency at vowels and also at consonants.

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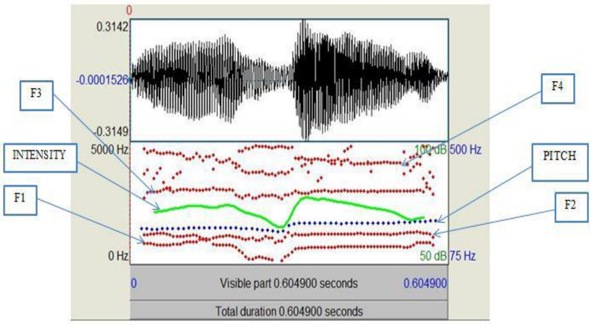

Figure 10: Spectrogram of ABA speech

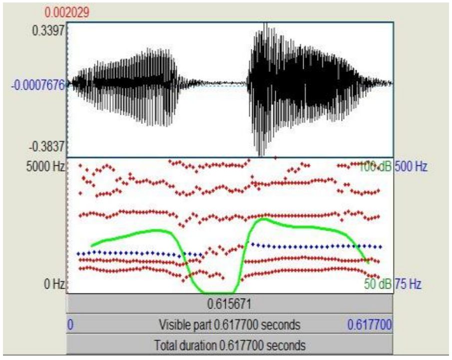

Figure 11: Spectrogram of APA speech

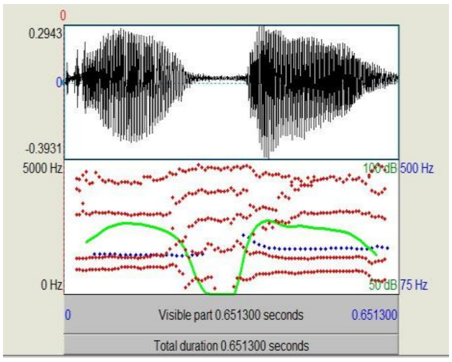

Figure 12: Spectrogram of ASA speech

Table 1. shows the Formant and Pitch Frequency for original (Like ABA, ADA) and modified signals (ABA_M, ADA_M) at vowels at time = 0.14 sec. and Table 2 shows the Formant and Pitch Frequencies for original and modified signals at consonants at time t = 0.40 sec. From this tables we can see the difference between the changing the frequency.

Table 1: Formant and Pitch Frequencies for original and modified signals at vowels

t=0.14	F1	F2	F3	F4	PITCH
ABA	998.80	1308.35	3115.26	4405.34	200.12
ABA_M	996.33	1023.25	3009.25	4213.79	200.08
ADA	939.66	1311.63	2860.86	4020.33	195.68
ADA_M	815.35	1257.72	2863.62	3922.14	195.91
AGA	638.77	1358.42	2804.29	4190.38	198.91
AGA_M	648.89	1380.01	2780.11	4001.34	198.87
APA	994.50	1277.93	3000.18	4217.21	198.45
APA_M	877.16	1157.63	2974.75	4086.10	198.50
ATA	942.84	1294.66	3141.11	4032.71	195.16
ATA_M	682.34	1249.53	2861.24	3900.66	194.56
AKA	1007.71	1294.07	3111.60	4239.61	197.18
AKA_M	1010.16	1189.25	3048.57	4117.02	197.23
ASA	987.54	1373.96	2951.69	4272.00	196.64
ASA_M	967.19	1328.34	2929.43	4132.71	196.54
AFA	1007.33	1124.33	3129.05	4372.22	200.62
AFA_M	816.39	1133.29	3017.98	4176.37	201.03

Table 2: Formant and Pitch Frequencies for original and modified signals at consonants

t=0.40	F1	F2	F3	F4	PITCH
ABA	720.47	1207.51	3009.78	4282.58	214.825
ABA_M	699.95	1211.78	2960.08	4133.76	214.8028
ADA	762.22	1968.98	2699.58	3473.09	222.1045
ADA_M	810.58	2038.30	2815.74	3695.15	222.21
AGA	563.38	1301.87	2409.43	3783.93	212.49
AGA_M	648.06	1364.63	2501.35	3691.62	211.74
APA	829.08	1255.53	2869.82	4014.51	218.96
APA_M	806.28	1251.83	2849.65	4045.32	219.00
ATA	855.06	1472.99	2691.53	4198.36	217.15
ATA_M	846.82	1497.74	2685.18	4004.69	217.15
AKA	744.18	1249.44	2660.60	3155.02	223.71
AKA_M	726.90	1321.84	2711.30	3669.70	226.43
ASA	766.13	1258.05	1942.25	3223.64	218.66
ASA_M	737.82	1419.08	2592.68	3166.35	220.03
AFA	872.67	1878.86	2749.53	3623.09	226.16
AFA_M	837.52	1851.45	2832.33	3765.31	227.90

Changes in Formants F1/F2/F3/F4 is observed. The original speech words is not audible to hearing impairment hence the changes in pitch and Formants makes the words audible to hearing impairment.

V. CONCLUSION

The results obtained are verified with VCV words. The presentation of a simple acoustic cues can be restored by changing the magnitudes of specific frequencies for speech words. The changes in formants and pitch is observed which would be useful speech for hearing impaired.

Future scope

Present system utilize only one algorithm, the enhance system may uses multiple algorithm to resolve the problem of hearing impairment and also more hearing impairment can be tested.

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