Identification of Stress Speech in Marathi Language using AEMD Algorithm

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Abstract - Stress recognition is one of the important and interested research works for human being. This paper deals with the stress level and on that we define as the person is under low stress or high stress. Stressful Speech is a non-linear and non-stationary signal and termed as a physical quantity which can be measured through MATLAB Software. This paper presents a new method called Adaptive Empirical Mode Decomposition (AEMD) applied to speech in Marathi Language datasets.

Index Terms - AEMD, MATLAB, FFT, FPGA, IMF.

I. INTRODUCTION

In the 21st century, stress is a very fast growing problem around the world that affects not only human health but also the productivity of organization where this human is working. Also high Stressful speech is a pressure which is acting on individual’s body in form of tremor, anxiety, depression, greater heart beats, and sleeping difficulties. So it is very important to identify the person who is in high stress or low stress. Today’s world is filled with less physical movements and has induced mental stress levels which introduced new field of research as analysis of speech. Stress recognition from speech signal is the identification of speech as normal or stressed. Again stress speech will be further classified as high level stress or low level stress depending on its amplitude level. These levels can be measured by the changes or fluctuations in muscles of vocal chords named as microtremor frequency in Stressful speech. Adaptive Mode Decomposition [2] was applied for decomposing speech into Intrinsic Mode Functions (IMF) [3][4].

II. IDENTIFICATION OF STRESS USING EMPIRICAL MODE DECOMPOSITION (EMD)

An Empirical Mode Decomposition algorithm is developed using MATLAB to detect microtremor in the voice for stress detection or Lie Detection [1]. The EMD process is a “Sifting Process” that extracts intrinsicmode functions (IMF) from raw input signal until the microtremor is been extracted [2]. The general AEMD process is expressed in the flowchart as shown in Fig. 1 [1] [5].

III. ADAPTIVE EMD PROCEDURES

AEMD is an iterative or “sifting” process described as follows [1]:
1) Upper and lower envelopes of the unstressed voice signal \( h(x) \) are constructed with its maxima and minima using cubic spline function.
2) Mean of the envelopes \( m_i \) is subtracted from \( h(x) \) to obtain a new signal \( h_i(x) \).
3) Determine if \( h_i(x) \) is an IMF using the criteria described above.
4) If \( h_i(x) \) is an IMF, it is subtracted from the original signal \( h(x) \), and the resulted new signal \( h(x) \) goes through the above procedures until another IMF is obtained.
5) Each IMF is checked if it is in the microtremor frequency band (8 – 12 Hz). If not, the algorithm adaptively adjusts the stopping criteria until the in-band IMF representing a microtremor is detected. This IMF is used as the reference.
6) The stopping criteria consist of several important parameters including the absolute amplitude of the remaining signal, the mean value of the envelope, the cross-correlation coefficient between the remaining signal and the original signal, and the Standard Deviation (SD) between two consecutive results in the sifting process.
AEMD is a method of breaking down a signal without leaving the time domain [6]. The empirical mode decomposition generalizes the Fourier analysis. It decomposes a signal as the sum of intrinsic mode functions [4]. AEMD procedure can be applied to decompose the time series into a set of IMFs and a residue [7]. By applying the Hilbert transform to each IMF signal can be further analyzed to calculate the instantaneous frequency and amplitude of each IMF [8]. The whole process is called Hilbert Huang Transform. In this study, we implement an iterative algorithm to find the intrinsic mode functions for Marathi real time database.

IV. DATABASE

In this paper we use real time database recording in Marathi Language. This database is developed by our self. Database recordings are considered for sampling rate greater than or equal to 44.1 kHz because a signal with value less than this sampling rate will not decompose properly [9]. The data samples are imported in MATLAB software and AEMD algorithm is applied to this Marathi Language.

The sentences we recorded into Audacity Software and taken from different persons of age group from 25-58 yrs. The Normal Audio Signal recorded is “I am Happy”, “मी आनंदात आहे”. The Low Stress Audio Signal sentence is “I am very late for the office”, “मला ऑफिसला जाण्यासाठी खूप उशीर झाला आहे”. The High Stress Audio Signal sentence is “Stupid, you don’t know such silly things”, “मूख तुला एवढी साधी गोष्ट कठन नाही”.

V. RESULTS

Concentrating on the levels of stress we have used AEMD Algorithm to detect the Microtremours (8-12 Hz). In this research we define frequency from 8-10 Hz as low stress audio signal and a value more than 10 Hz to be considered as High Stress as Audio Signal [13]. We have added one more classification as Normal speech audio signal for the study analysis. Following MATLAB Figures explains the Algorithm. Figure 2 shows that the audio is Normal speech into Matlab window.

Above figure shows the Empirical Mode Decomposition Components from AEMD Algorithm where X-label is amplitude and Y-label is time of Low Audio.

Above figure displays the FFT of EMD components of Fig. 3

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Above figure provides the frequency response of the input signal along with the filtered frequency response of low stress audio signal. The filter here we have designed is a Bandpass filter and Butterworth filter. For this filter we have founded the amplitude and phase values of it.

From fig. 6 we get the low stress audio magnitude in (DB) and on Y-label we consider frequency in hertz.

Above figure illustrates the MATLAB window for proving that the audio was a low Stress Signal with Bandpass filter coefficients in Hexadecimal format.

Above figure shows the Empirical Mode Decomposition Components from AEMD Algorithm where X-label is amplitude and Y-label is time of High Stress Audio signal.

From above figure we get the high stress audio magnitude in (DB) and on Y-label we consider frequency in hertz.

Above figure provides the frequency response of the input signal along with the filtered frequency response of High Stress audio signal.
VI. CONCLUSION

From the above results the AEMD Algorithm is applicable for decomposition of real audio into IMF. We had used this Algorithm for identification of stress in the Marathi Language. Throughout this research we have been concerned primarily with the basic principles behind the design of Bandpass filter through AEMD algorithm and implementation of stress speech identification by MATLAB Software.

VII. FUTURE SCOPE

The AEMD algorithm for Marathi Language was totally applied In MATLAB software. The key part will be used in coordination with VHDL for designing total stress speech Identification system.

REFERENCES