IOSR Journal of Electrical and Electronics Engineering (IOSR-JEEE)
e-ISSN: 2278-1676, p-ISSN: 2320-3331
PP 44-47
www.iosrjournals.org

Moving Average and Pre-emphasis filter design in Preprocessing of Pathological Speech Signals

Prof. Syed Mohammad Ali[1], Dr. Pradeep Tulshiram Karule[2],
Prof. Syed Akbar Ali[3], Prof. Syed Irfan Ali[4]

[1]*(Department of Electronics & Telecommunication Engineering, Anjuman College of Engineering & Technology/Nagpur University, India)*
[2]*(Department of Electronics Engineering, Yeshwantrao chavan College of Engineering, Wanadongri/Nagpur University, India)*
[3]*(Department of Electronics & Telecommunication Engineering, Anjuman College of Engineering & Technology/Nagpur University, India)*
[4]*(Department of Electronics & Telecommunication Engineering, Anjuman College of Engineering & Technology/Nagpur University, India)*

ABSTRACT: *Because of nature of job, unhealthy social habits and voice abuse, people are subjected to risk of voice problems It is well known that most of speech disorders causes changes in the acoustic voice signals [9].In the recent year the trend towards automated analysis of pathological noise signal has gain momentum. The awkwardness of analog equipment has simulated development of digital computer techniques for processing and analysis of pathological speech signal in patient care system. The above filter design techniques & prepossessing of speech signal can be used in any speech processing application.In this paper voices of patients are taken and they are preprocessed. In prepossessing Speech signal is passed through Moving Average (M.A) filter, High pass (H.P) filter for removal of noise & then framing and windowing is done. This preprocessed output can be used for pathological voice recognition, speech identification, speaker identification & many more application.*

Keywords -*Framing, High-Pass Filter, Moving Average Filter, Pathological Speech Signal,Windowing.*

I. INTRODUCTION

Pathological voice recognition has received a greater attention from researchers in the last decade. Speech processing has proved to be excellent tool for noise disorder detection [3]. Pathological voice signal of patient from Dr. Naresh Agrawal Hospital, ENT Surgeon has been taken. The signals are recorded keeping mic two inch away from mouth using voice recorder of window XP. The sampling is chosen to 11025 samples /sec.

The patient has pronounced, 'a' which is vowel, then 'ah' which is vowel with consonant and a word 'Hello' for two sec, these signals frequency are noisy & noise needs to be removed. So filters are designed. The signal is passed through filter & then framing & windowing is done. The output of window is called prepossessed output which can be used for further application like diagnosis of disease or speaker recognition. Physicians often use invasive techniques like endoscopy to diagnose symptoms of vocal fold disorders however it is possible to diagnose disease using certain feature of speech signal [3]. Speech signal is sinusoidal signal having different frequency, different amplitude & different phase. It is given by the expression given below [6].

$$\sum_{i=1}^{N} A_i(t)\sin[2\pi F_i(t)t + \theta_i(t)]$$

Where, $A_i(t)$, $F_i(t)$ &$\theta_i(t)$ are the sets of amplitudes, frequencies & phases respectively, of the sinusoids, The use of non-invasive techniques to evaluate the larynx and vocal tract helps the speech specialists to perform accurate diagnose[10]. Voice & speech production requires close cooperation of numerous organs which from the phonetic point of view may be divided into organ.

- Lungs, Bronchi, Tracheas (producing expiration air steam necessary for phonation)

- Larynx (amplifying the initial tone)
- Root of the tongue, throat, nasal cavity, oral cavity (forming tone quality & speech sound) [7].

Speech signal in non-intrusive in nature& it has potential for providing quantitative data with reasonable analysis time. So study of speech signal of pathological voice has become an important topic for research as it reduces work load in diagnoses of pathological voices [8].

II. BLOCK DIAGRAM

The Fig.1 shows block diagram of preprocessing of speech signal.

$$\left\{\begin{matrix}\text{Speech}\\ \text{Signal}\end{matrix}\right\} \rightarrow \underbrace{\boxed{\begin{matrix}\text{M.A.\&}\\ \text{H.P.}\\ \text{Filter}\end{matrix}} \rightarrow \boxed{\text{Framing}} \rightarrow \boxed{\text{Windowing}}}_{\text{Preprocessing Of Speech Signal}} \rightarrow \left\{\begin{matrix}\text{Processed}\\ \text{output}\end{matrix}\right\}$$

Figure1. Block diagram of preprocessing of speech signal

III. FILTER DESIGN

The noisy speech signal is passed through filters like moving average filter, prepossessing filter. The moving average filter takes average of samples for filtering the noise from signal. The expression for output of such filter is given below [6].

$$Y(n) = \frac{X(n) + X(n-1) + X(n-2)}{3}$$

Where, X(n) is the input speech sample.

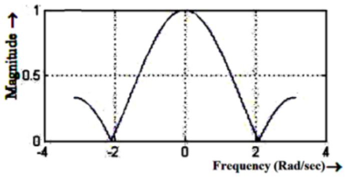


Figure 2. Magnitude v/s Frequency plot of moving average filter

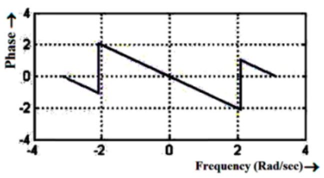


Figure 3. Phase v/s Frequency plot of moving average filter

Analysis with different averages of samples are taken it is found that with average of 3 proper filtering is done. The system function H(z) & the impulse response h(n) of filter are given below:

$$H(z) = \frac{Y(z)}{X(z)} = \frac{1}{3}[1 + z^{-1} + Z^{-2}]$$

$$\therefore \ h(n) = [\frac{1}{3}, \frac{1}{3}, \frac{1}{3}]$$

This FIR filters shown by magnitude & phase plots in fig. 2, fig. 3, fig. 4& fig. 5 are more stable.

IV. PRE-EMPHASIS FILTER

To ensure a high-quality product, diagrams and lettering MUST be either computer-drafted or drawn using India ink.

High pass Pre-emphasis is used to flatten speech signal spectrum & to make the speech signal less sensitive to finite precision effects later in speech signal processing [4].

Here we have used single coefficient FIR filter. The system function H(z) of filter is

$$H(z) = 1 - \lambda z^{-1} \qquad \ldots (1)$$

$$\frac{Y(z)}{X(z)} = 1 - \lambda z^{-1}$$

$$Y(z) = X(z) - \lambda z^{-1} X(z)$$

The time domain representation of filter will be

$$Y(n) = X(n) - \lambda X(n-1)$$

Where, λ is the filter coefficient and the value of pre-emphasis coefficient is $\lambda \; \varepsilon \; (0.9 - 1.0)$ with $\lambda = 0.9375$, best optimum result of filtering is received [5]. The pre-emphasis filter amplifies the area of spectrum. Thus improving the efficiency of spectral analysis [2].

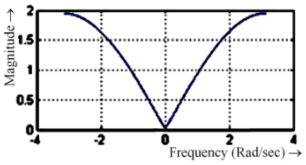


Figure 4. Magnitude v/s Frequency plot of Pre-emphasis Filter

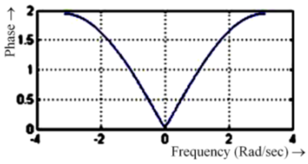


Figure 5. Phase v/s Frequency plot of Pre-emphasis Filter

V. FRAMING & WINDOWING

Any speech signal is slowly varying over time (quassi stationary) that is when the signal is examined over a short period of time (5 msec to 100 msec). The signal is fairly stationary.

And that is why, speech signals are often analyzed in short time segment, which are referred as short-term spectral analysis. This practically means the signal is blocked in frames of typically 20-30 msec. Adjacent frames typically overlap each other with 30% to 50 % This done in order not to lose any information due to the windowing. After the signal has been framed each frame is multiplied by window function W(n) with length N. The frame has N= 256 samples & adjacent frame separated by M=128 samples [4]. Where N is the length of the frame. Typically hamming window W(n) is used & given by

$$W(n) = 0.5 - 0.46\cos(2\pi n / N - 1) \quad 0 \le n \le N-1$$

Where, N is total number of samples. The windowing is done to avoid problems due to truncation of signal as window helps in smoothing of signal [1].

VI. RESULT & CONCLUSION

Fig. 6 shows that, signals are perfectly filtered using both ma and pre-emphasis filter. The output of ma filter needs to be scaled to get amplified output. From the table it is clear that this system can be used for any speech preprocessing applications. Filtered signal is framed and passed through window.

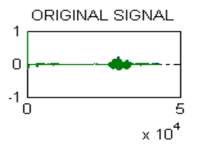
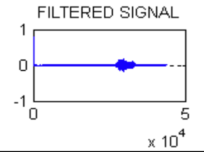
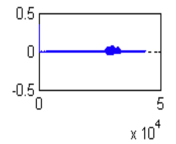
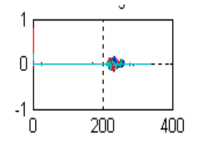
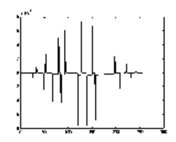
SN	Patient Samples	Disease	HP Filter Output	MA Filter Output	Framing	Windowing
1		oral Carcinoma				

Figure 6. Example showing complete preprocessing analysis of patient's speech samples

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