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SMOOTHING DIGITAL SIGNALS

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ABSTRACT: This article discusses some of the techniques of smoothing speech sounds. As a result of research how much the number of points involved in smoothing are learned and based. The issue of smoothing sounds was investigated by the non-orator speech recognition problem.

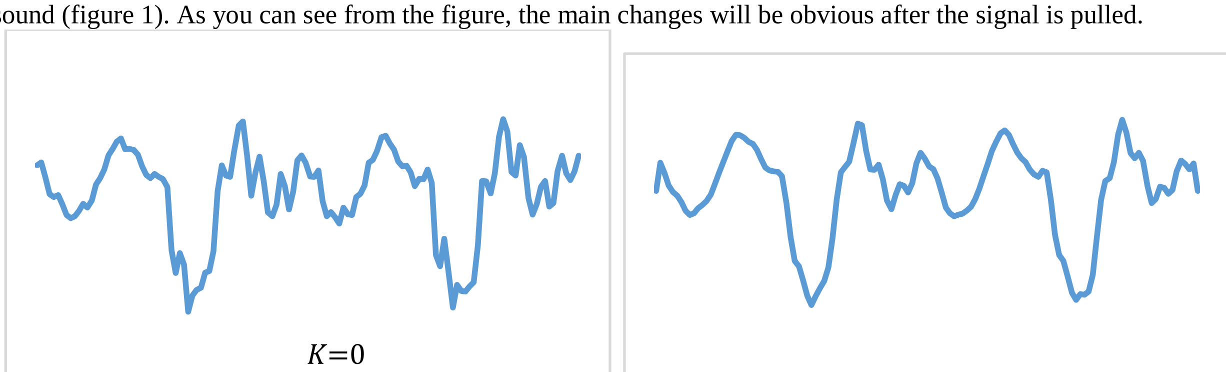
KEY WORDS: speech sounds, phoneme, discretion signals, recognition of symbols, trend, smoothing signals.

I.INTRODUCTION

The highest peak of modern computer technology is definitely related to the identification of symbols and cybernetics. Solving identification of symbols issues will give many benefits such as decreasing extra human labor, automating extremely complex processes and in some cases, avoiding human factors. However, the identification of symbols issues are relevance, as well as they are complex. Specifically, recognition of speech sounds is one of the most difficult aspects of recognizing symbols issue. In particular, the researcher needs not only to know programming, but also to have high mathematical knowledge, physical properties of sound waves and even linguistic knowledge to develop speech recognition algorithms. Initial processing of audio signals is one of the most important steps for effective solutions in the development of speech recognition algorithms. Specifically, recognition of symbols by first cultivating of data has direct impact on key performance indicators, such as precision algorithms for data recognition, stagnation against data received in different circumstances, simplification of algorithm development and program performance. It is appropriate to pay attention to the first cultivating issues of speech sounds in recognition of speech sounds algorithms producing. First cultivating process of speech sounds recognition encompasses normal processing of signals, quantization, discretion, different filtering, smoothing and so on. Specifically, in this article we focus on the use of algorithms of speech sounds recognition in the problem of smoothing of speech sounds.
The issue of smoothing digital sounds is widely used not only in speech sounds, but also on musical sounds, economic indicators, monitoring and analysis of air temperature, humidity changes in hydrometric observations and many other practical issues. What are the results achieved by smoothing signals on variety of topics?

- By smoothing signals, trend will be identified
- Random snooze signals or noise are eliminated by smoothing signals
- We can use smoothing signals in order to filtering purpose

Trend shows the main indicator changes (tendency) of signals. That is, it gives a better understanding of signals. Trend is the changes in the time series. For example, let’s take any the smallest of period of phoneme signal in the speech sound (figure 1). As you can see from the figure, the main changes will be obvious after the signal is pulled.



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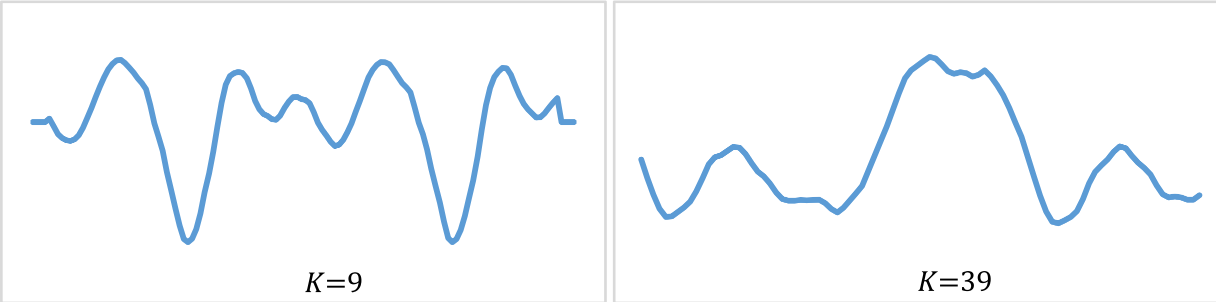

Figure 1

Nowadays many methods and algorithms such as slider medium values, different weight value and medians were produced in smoothing signals issues.
According to medium value method, the smoothing signals are done like that
Digital signals values are given at each N point.
$\{f_1, f_2, f_3, \dots f_N\}$ ёки $f_i; i = 1,2,3, \dots N.$

$i -$ to calculat slider medium value, i pint for before and next and all K paints and $i -$ is counted average arithmetical values

$$g_i = \frac{1}{2K+1}(f_{i-K} + f_{i-K+1} + \dots + f_i + \dots + f_{i+K})$$

That is:

$$g_i = \frac{1}{2K+1}\sum_{j=-K}^{K} f_{i+j}. \quad (1)$$

Each point of the digital signal from the given formulas can be used in the same way. However, the closer it is to the point i the more important it is, the less important it will be as long as it is longer. For example mathematical expression, it is often possible to express the weight of each point by means of which it can be expressed. Then the formula 1 is expressed as:

$$g_i = \frac{1}{2K+1}\sum_{j=-K}^{K} \omega_i f_{i+j}. \quad (2)$$

The following is a condition that should not be disrupted from the original signal when the middle value function is used:

$$\sum_{j=-K}^{K} \omega_i = 1.$$

2-in formula ω_i – is function that gives points weight. For example, Gaus function is can be used as a weight function. [1].

Another commonly used method for smoothing digital signals is the median. In this method to calculate smoothed i from given N points $\{f_1, f_2, f_3, \dots f_N\}$ i points' left and right side K points are will be taken. Taken K points selected in terms of increasing and decreasing condition. That is, selected K points $\{a_1, a_2, a_3, \dots a_K\}$ and this collection central element j=$\frac{K-1}{2}+1$ is taken as a smoothing result $g_i = a_j$. If K is pair , j=$\frac{K}{2}$ two values' average arithmetic are taken in the centre [1,3]:

$$g_i = \frac{a_j + a_{j+1}}{2}$$

As shown figure 1, the higher the number of K points, the smoothing effect will be much that. That is, small, very large values. That is, K=3 is small, and K=39 is very large. In fact, if K=39, trend will be useless which is certainly natural. Because, smoothing signals is filtering high frequency. So if we remove the high pendulum oscillations in the signal, we get so far from the informative marks. Therefore, K=39 trend that has been increasing will not give us the necessary information.

The question arises, how the value of K should be chosen not to losing informative information? How K value is chosen according to what foundations? In fact, if K is smaller the signal will be not smoothed adequately. If K is the

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biggest, main features of trend will be disappeared. Hence, right choice of K is important to select informative information.

The number of points will be chosen according to K solving issues meaning in smoothing signals. Specifically, the topic we have chosen is to identify sounds not to related to orator's sounds and smoothing firstly speech signals to gain sufficient result. By this, we can get rid of noises that interrupt and will achieve to take trend. Smoothing signals is depend on a calculation of certain equilibrium value points previous and next points of *i*. Smoothed signals' *i* points have how many number points, K's value will be equal those points number. However, the effect of *i* points on smoothing point K may vary. For example, if it the closer to *i* it will be so stronger.

Delivering the time and frequency of the main tone in speech processing is an important subject. The main tone frequency is the frequency of vibrations of the sound curves when these vocal sounds are pronounced. If the problem is to determine the trend of the main tone, then the tone of the speech will be considered at the smallest tone of the main tone. That is, the value of the tone of the speech tone will be equal to the number of points at one time of the majority tone. But since the problem is to find the main tone, it is not the time and frequency of the main tone. Different orators' tone is between 80-400H [2]. For the protection of the main informative elements, it is often desirable to obtain a maximum upper limit of the human speech. 400 Hz. For example, discretion frequency of analogy recording voice is f_d =16 000Гц бўлсин. Calculating maximum value of K is like that:

$$K \leq \frac{f_d}{f}. \quad (3)$$

We can give interpretation to 3- formula: if discretion period frequency of trend which can swing in 8000Hz is f_d =16 000Гц will swing in two points. The smallest period of 4000Hz trend that swing 4000Hz will swing in f_d =16 000Гц diskert frequency signal in four point and the smallest phase of the oscillating trend at the frequency f oscillates at the same point in the signal at which the frequency of discretion f_d. Specifically, the gradient coefficient $K \leq 40$ can be obtained to determine the trend of the main tone of the signal in the above figures according to formula 3.

Another example, if the phoneme's trend is needed, what value can K be equal? trend. On this question, we will take the following calculations. In Analogy voice recording, discretion frequency will be f_d =16 000Hz. Human speech pendulum oscillation is between 300-4000Hz [2]. But, in simple conversation pendulum oscillation may be between 300-2400Hz.

That is, we will take high limit of pendulum oscillation phoneme f = 2400Гц. According to formula 3, in order to identify phoneme trend smoothing coefficient $K \leq 7$.

It should be noted here that when smoothing coefficient K is known, it is possible to set specific weight values for the K element of the signal, respectively. Of course, the weighting values are determined based on the essence of the chosen issues. However, in many cases, the value of the signal K has less than the point *i* which is just as low as the weight gain[4].

In summary, the arguments and example show that intuitive selection of value does not produced good results [4]. Certainly, based on the above mentioned ideas, smoothing of digital signals ensures better results. Although the issue of smoothing signals in the article is perceived in sound data, it is possible to achieve effective results in the study of signals investigated in other areas, such as economic, natural and technical fields.

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