# Pitch Tracking of Acoustic Signals based on Average Squared Mean Difference Function

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## **Abstract**

In this paper, a method of pitch tracking based on variance minimization of locally periodic subsamples of an acoustic signal is presented. Replicates along the length of the periodically sampled data of the signal vector are taken and locally averaged sample variances are minimized to estimate the fundamental frequency. Using this method, pitch tracking of any text independent voice signal is possible for different speakers over any database.

#### 1. INTRODUCTION

Extraction or determination of fundamental frequency (or pitch) of a speech signal is a fundamental problem in both speech processing and speaker recognition. The typical pitch range for a male human being is 80-200 Hz, and for female 150-350 Hz. Many methods to extract the pitch of speech signals have been proposed. Improvements in accuracy of performance, robustness against noise of these methods are still desired. Unfortunately, we do not have very reliable and accurate method for pitch extraction in noisy environments. Also measuring the period of a speech waveform, varying in and with the detailed structure of the waveform, can be quite difficult. Another problem is automatic selection of the window of the voiced speech segments.

Autocorrelation based methods [7] and [13] are known to be comparatively robust against noise. Also the average magnitude difference function (AMDF) method [12] is classified into this category.

Based on these methods there are some other techniques like auditory modeling [2], probabilistic AMDF modeling [4], fourier transform modeling [6], real-time digital hardware pitch detector [9], semiautomatic pitch detector (SAPD) [10], automatic formant analysis [11], modified autocorrelation and AMDF [14], projection measure technique [15], pseudo-pitch synchronous analysis [16] and many more [8]. Ideas on pitch extraction have also been discussed in some tutorials [3] and [5].

In this paper, we have proposed a new method for extraction of fundamental frequency of speech signal in clean environment using simple statistical techniques. Statistical characteristics of sample variances calculated over periodic subsamples have been used to extract fundamental frequency from speech signals. Here these sample variances have been averaged and then minimized to use the unbiased estimator property. This way we estimate pitch of clean signals better than any other standard methods.

The remainder of this paper is organized as follows. Section 2 describes the motivation and the principle of the proposed method. In Section 3, we show the results of preliminary test for the proposed method and comparing with some standard pitch detection methods we confirm the effectiveness of our method. In Section 4, we gave an error analysis of the method. In Section 5 we conclude this paper giving views regarding further development that can be done. In the appendix, given as in Section 6, important calculations, which show the link between the proposed method and the autocorrelation function, are explained.

## 2. PROPOSED METHOD

Given a discrete time signal x(n), the autocorrelation function is defined as

$$r_x(m) = \frac{1}{n} \sum_{i=0}^{n-1} x(i)x(i+m),$$

defined for all n and lag m.

A variation of autocorrelation analysis for measuring the periodicity of voiced speech uses the average magnitude difference function (AMDF), defined by the relation

$$D_{S}(t) = \frac{1}{L} \sum_{j=1}^{L} |S_{j} - S_{j-t}|,$$

where  $S_j$  are the samples of input speech  $S_j = (S_1, S_2, ..., S_L)$  and  $S_{j-t}$ , are the samples time shifted t milliseconds backwards.

In case of AMDF,  $D_S(t)$  has been approximated by a scalar multiple of  $\frac{1}{L} \left\{ \sum_{j=1}^{L} (S_j - S_{j-t})^2 \right\}^{1/2}$  which

again has been approximated by the scalar multiple of  $\left[2\left\{r_S(0)-r_S(t)\right\}\right]^{1/2}$ , where  $r_S(t)$  is the autocorrelation defined as above. These approximations may suppress calculations important for pitch extraction.

Here we propose a new function which has been linked to the autocorrelation and refined in the Appendix.

Consider a voiced segment  $y = (y_1, y_2, ..., y_n)$  in a digital speech signal. Since most speech signals can be viewed as a quasi-periodic sequence the fundamental frequency may not be uniquely defined mathematically. In our approach we estimate the fundamental frequency by statistically enhancing the most significant harmonics present in y.

For a given time period k consider a periodically subsampled data  $y_{b,w} = \{y_b, y_{b+1}, ..., y_{b+w}\}$ , where b

is an index varying from 1 to  $w\left(\left[\frac{n}{w}\right]-1\right)$  and w is taken to be a constant. We define the following parameters as

$$m = \left[\frac{w+1}{2}\right] - 1,$$

$$S_{ik} = \{i, i+k, ..., i+pk : p \in Z^+\},$$

where k = 1, 2, ..., m; i = 1, 2, ..., k; p is the greatest integer such that,  $i + pk \le b + w$ .

Now we define our Average Squared Mean Difference Function (ASMDF) as

where

$$\overline{y}_{b,w} = \frac{1}{p+1} \sum_{i \in S_{-}} y_{b,w}(j)$$
.

Let  $f_0$  be the sample rate of the original speech signal and  $f(k) = \frac{f_0}{2k}$ ,  $k = 1, 2, ..., \lceil \frac{w+1}{2} \rceil - 1$ . In

view of (1) g can be thought of as a function of f. Also g can be thought of as a mean squared mutual difference function which can be approximated with the standard autocorrelation function. Let i be the

index of the minimum of the components of g(k). Then f(i) is referred as the estimated fundamental frequency of the speech signal of y.

#### 3. IMPLEMENTATION

# A. Experimental Details

Clean speech signals were obtained from IViE Corpus [1]. Speech samples were uttered by one female speaker (F1) and one male speaker (M1). Each of such speech signals consisted of a maximum of five English words, which were sampled by different rate.

## B. Preliminary Test

Taking w = 1000 we found data sets of f and the following graphs:

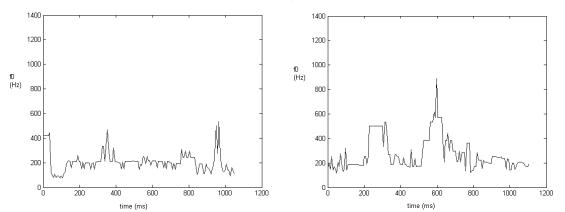


Figure A: Graph for fundamental frequency of speaker F1 Figure B: Graph for fundamental frequency of speaker M1

## C. Comparison with Some Standard Methods

To investigate the accuracy of the ASMDF, we have conducted experiments which compare it with two conventional methods. The conventional methods are the methods of Autocorrelation and AMDF.

Autocorrelation method does the correlation analysis frame-by-frame to the estimated average pitch period of the speaker. Given a discrete time signal x(n), the autocorrelation function is defined as

$$r_x(m) = \frac{1}{n} \sum_{i=0}^{n-1} x(i)x(i+m)$$
,

defined for all n and lag m. The property of this function is that  $r_x(m)$  is large when x(n) has similar value with x(n+m). If x(n) has a pitch period  $\rho$ , then  $r_x(m)$  has peaks at the integral multiples of  $\rho$ . Obviously  $r_x(0)$  is maximum among these values, the second largest being  $r_x(\rho)$ . Other maximals usually decrease as m increases. Therefore using this method we can estimate  $\rho$  from the location of the peak at  $m=\rho$ .

A variation of autocorrelation analysis for measuring the periodicity of voiced speech uses the AMDF, defined by the relation

$$D_{S}(t) = \frac{1}{L} \sum_{j=1}^{L} |S_{j} - S_{j-t}|,$$

where  $S_j$  are the samples of input speech  $S_j = (S_I, S_2, ..., S_L)$  and  $S_{j-t}$ , are the samples time shifted t milliseconds backwards. The separation of the nulls that appear in calculating  $D_S(t)$  is a direct measure of the pitch period.

In [12] a detailed comparative performance study has been discussed of these two methods. The readings suggest that the proposed method is competitive with the other two methods. Although in practical situations we face noisy speech in most of the cases, robustness of ASMDF doesn't take much care of the fact whether the environment is noisy or not. The graphs are as follows:

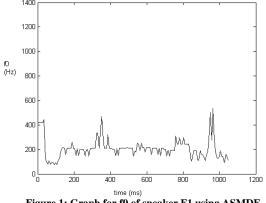


Figure 1: Graph for f0 of speaker F1 using ASMDF

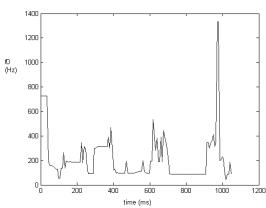


Figure 2: Graph for f0 of speaker F1 using AMDF

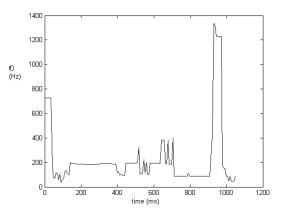


Figure 3: Graph for f0 of speaker F1 using Autocorrelation

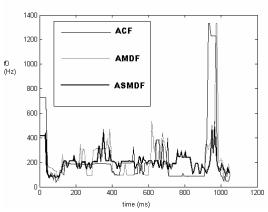


Figure 4: Overlapped graph for f0 of speaker F1 using ASMDF, AMDF and Autocorrelation

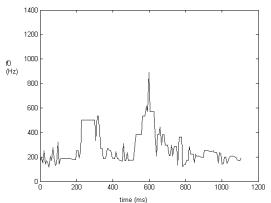


Figure 5: Graph for f0 of speaker M1 using ASMDF

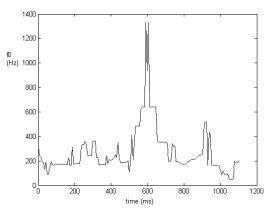
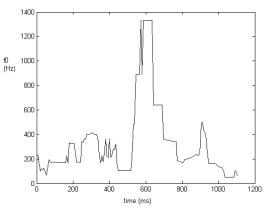


Figure 6: Graph for f0 of speaker M1 using AMDF



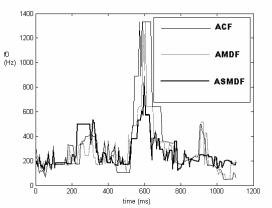


Figure 7: Graph for f0 of speaker M1 using Autocorrelation

Figure 8: Overlapped graph for f0 of speaker M1 using ASMDF, AMDF and Autocorrelation

## D. Observations

#### F1:

Here major fluctuations around 40th ms for the method ACF and AMDF and from 600th to 700th ms and from 900th to 1000th ms for the methods ACF and AMDF are observed, whereas no major fluctuations but minor ones are there in the segment for the method ASMDF.

#### M1:

Here the method ASMDF is much consistent with the methods ACF and AMDF. All major fluctuations (e.g. from 200th and 300th ms, from 500th to 700th ms) are observed through the three methods. Around 900th ms major fluctuations are there for the methods ACF and AMDF but not for ASMDF.

#### 4. ERROR ANALYSIS

Let  $P_c$ ,  $P_m$  and  $P_s$  be the pitch contour found using autocorrelation, AMDF and ASMDF respectively calculated over the same window for the above two speakers. Let us denote

$$e_{sm} = P_s - P_m$$
,  $e_{cm} = P_c - P_m$  and  $e_{cs} = P_c - P_s$ 

as the gross pitch error. Let us define the standard deviation of the gross pitch error as

$$\sigma_e = \sqrt{\frac{1}{L_e} \sum_{i=1}^{L_e} e^2(i) - \overline{e}^2} ,$$

where  $e(i) = e_{sm}(i), e_{cm}(i), e_{cs}(i)$ ;  $L_e$  being the length of each of the gross pitch error and

 $\overline{e} = \frac{1}{L_e} \sum_{i=1}^{L_e} e(i)$  is the mean pitch error. The experimental values of  $\sigma_e$  and  $\overline{e}$  is given in the following table:

	$e_{sm}$	$e_{_{Cm}}$	$e_{cs}$
$\sigma$ .	176.4591(F1)	190.5388(F1)	231.3099(F1)
- e	140.8554(M1)	169.8732(M1)	230.5979(M1)
$ \bar{e} $	8.7260(F1)	12.6060(F1)	21.3320(F1)
' '	1.0787(M1)	56.9400(M1)	55.8613(M1)

## 5. CONCLUSION

Based on the experimental results and error calculations it has been shown that ASMDF is useful in clean environment. It also makes a refinement of the autocorreltion-based methods to a great extent. The idea behind this method also leads to methods of extraction of other features of speech signals. Also there is a scope of using smoothing technique(s) for evaluation of pitch. The limitation of this method is the procedure to obtain the minimum values from the data set which are often used to fall behind the known limits viz. 80-200 Hz for males and 150-350 Hz for females.

## 6. APPENDIX

Let

$$\sigma_{i,k}^2 = \frac{1}{p+1} \sum_{j \in S_{i,k}} \{ y_{b,w}(j) - \overline{y}_{b,w} \}^2$$

which leads to a chi-square test if the signal is too noisy, otherwise it is normally distributed for clean signal. Now,

$$\sum_{j,j' \in S_{i,k}} \{y_{b,w}(j) - y_{b,w}(j')\}^2 = 2(p+1) \sum_{j \in S_{i,k}} \{y_{b,w}(j) - \bar{y}_{b,w}\}^2 = 2(p+1)^2 \sigma_{i,k}^2.$$

And hence

$$\sigma_{i,k}^{2} = \frac{1}{2(p+1)^{2}} \sum_{j,j' \in S_{i,k}} \{y_{b,w}(j) - y_{b,w}(j')\}^{2} = \frac{1}{p+1} \left\{ \sum_{j \in S_{i,k}} y_{b,w}^{2}(j) - \frac{1}{p+1} \sum_{j,j' \in S_{i,k}} y_{b,w}(j) y_{b,w}(j') \right\}$$

where the second sum in the braces is the autocorrelation function (ACF) with lag |j-j'| and the first sum is again the ACF with lag 0, i.e.  $r_v(|j-j'|)$  and  $r_v(0)$  respectively. Subsequently,

$$g(k) = \frac{1}{2k(p+1)^2} \sum_{i=1}^{k} \sum_{j,j' \in S_{i,k}} \{y_{b,w}(j) - y_{b,w}(j')\}^2 = \frac{1}{(p+1)} \{(p+1)r_y(0) - r_y(|j-j'|)\}.$$

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