Objective evaluation of chronic dysphonia laryngeal origin and follow-up of their treatments by the implementation of three characteristic parameters of acoustics vocal signal in patients with tumor or inflammatory chronic dysphonia S. Abdelouahed¹, M. Benabdallah², S. A. Aounallah³, F. Hadj Allal⁴

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Abstract: In this paper we develop a system dedicated to the objective characterization of chronic dysphonia laryngeal origin. The purpose of this system is threefold: diagnosis, treatment and monitoring of patient. For that we design an experimental protocol which consists of the recording and the archiving of the acoustic voice signal by means of the software environment Audacity. Our contribution consisted of the implementation, under the Visual Basic environment, of an algorithm that allows performing the analysis of a spectro-temporal acoustic voiced speech signal in this case the vowel "a" sustained for three seconds. We applied this algorithm to healthy subjects and pathological subjects. The results obtained show a variability of spectro-temporal characteristics between healthy and pathological subjects and prone pathological between them according to the nature of the lesion in particular with regard to spectral content evaluated by Short-Time discrete Fourier Transform STD.FT, Disturbances in the medium term STD and the fundamental frequency F0 averaged over several frames of the voiced signal.

Keywords: chronic dysphonia, fundamental frequency, STD, ST.DFT, Telemedicine, voiced sound.

INTRODUCTION

The voice is a spectacular indicator of physical and mental health of a person. The technology of voice pathology has been a marked increase over the past two decades, the voice processing is now a fundamental component of engineering [14] [1].

The special importance of voice processing in the more general framework is due to the privileged position of the speech as a vehicle of information in our human society. [11] The voice is indeed produced by the vocal tract, continuously monitored by the motor cortex [9] [4].

Among the voice treatment applications we distinguish [5] [13]:

1) Temporal spatio-spectro analysis of the vocal signal observing the objective characterization of dysphonia of laryngeal origins [8][10].

2) Quantitative estimation of characteristics parameters of the vocal signal during its acoustical representation including the Fundamental Frequency and the STD of voiced sounds [7] [12].

MATERIEL AND METHODES

The slide that we have implemented is composed of an interface for acquiring the acoustic speech signal consists of a dynamic microphone to reproduce sound in analog form and sound card for digitization and an environment software to archive the signal in Wave format in order to perform the calculation of different indices and we had to implement an algorithm for converting Wave format to decimal format in Visual Basic environment. The experimental protocol includes the following steps:

- Pronunciation of a voiced sound in this case 'a' sustained for three seconds.
- The division of the signal into 6 frames each 0.5 seconds
- The calculation of the three indices (spectral content, fundamental frequency and STD) averaged over six frames.
- The correlation between the indices themselves and the balance sheet and para-clinic of patients.
- The implementation of an interactive database of physiological and pathological acoustic voice signals for a clinical and epidemiological study and better therapeutic management.

A. STD.FT algorithm



It comprises the following steps [6][2]:

Sampling

The sampling in the time domain is to multiply the signal by a Dirac comb:

$$\hat{f}(t) = f(t).\,\omega_{t,T_e} \tag{1}$$

$$\hat{f}(t) = f(t) \sum_{k=1}^{N} \delta(t - kT_S)$$
(2)

Where:

$$\widehat{F}_{a}(f) = F_{e} \sum_{n \in \mathbb{Z}} F_{a} (f - n. F_{S}) \cdot \sum_{l=-\infty}^{+\infty} F_{0} \cdot \delta(f - lF_{0})$$

The number of samples contained in the time sequence t is: $N=T_0/T_sTs$.

The number of samples contained in the spectral sequence is: $N = \frac{1}{T_s} / \frac{1}{T_0} = \frac{1}{T_s} T_0 = T_0 / T_s$

$$\hat{f}(t) = \sum_{k \in \mathbb{Z}} f(k, T_s) . \, \delta(t - kT_s)$$
(3)

This amounts to a convolution frequency by another Dirac comb:

$$\widehat{F}(f) = F(f) * \Omega_{f,F_s}$$
(4)

• Limiting the duration of the signal

Limiting the duration of the time sequence, generally dictated by the memory of the computers, obtained by multiplying the latter by a rectangular door, height and length unit T_0 , sufficiently large for its spectrum can be treated as such is approximately a Dirac impulse. This second operation provides us with a sequence containing truncated N samples:

$$\hat{f}_{T_0}(t) = \sum_{k=0}^{N-1} f(k, T_s) . \, \delta(t - kT_s)$$
(5)

On the spectral plane is approximated the FT of the window truncation by a Dirac truncation, which gives:

$$\begin{aligned} \widehat{F}_{a}(f) &= \delta_{f} * \mathcal{F}\{\widehat{f}_{t0}(t)\} \\ &= \mathcal{F}\{\widehat{f}_{t0}(t)\} \\ &= \widehat{F}_{a}(f) \end{aligned}$$
$$\begin{aligned} \widehat{F}_{a}(f) &= F_{s} \sum_{n \in \mathbb{Z}} F_{a} \left(f - n.F_{s}\right) \end{aligned}$$
(6)

• Periodization of the signal

Periodization of:

$$\hat{f}_{T_0}(t) = \sum_{k=0}^{N-1} f(k, T_s) \delta(t - kT_s)$$

Is obtained by convolution with a Dirac comb of period T0 which gives:

$$\hat{f}_{T_{0p}}(t) = \left[\sum_{k=0}^{N-1} f(k, T_s) \cdot \delta(t - kT_s)\right] * \left[\sum_{l=-\infty}^{+\infty} T_0 \cdot \delta(t - lT_0)\right]$$
(7)
$$N = T_0 / T_s$$

The number of samples contained in each of the time periods is equal to the number of samples contained in each of spectral periods.

• ST.DFT definition

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The discrete Fourier transform of the infinite sequence of samples is:

$$\mathcal{F}\left\{\hat{f}_{T_0}(t)\right\} = \mathcal{F}\left\{\sum_{k \in \mathbb{Z}} f(k, T_s), \delta(t - kT_s)\right\}$$

$$\mathcal{F}\left\{\hat{f}_{T_0}(t)\right\} = \sum_{k \in \mathbb{Z}} f(k, T_s). e^{-j2\pi f k T_s}$$
(8)

Where the spectrum obtained after periodization is discreet and its samples are separated by the interval F0, the frequency takes the values:

$$\mathcal{F}\{\hat{f}_{T_0}(t)\} = \sum_{k=0}^{N-1} f(k) \cdot e^{-j2\pi f k T_s}$$
(9)

 $f = f_0 \cdot n = \frac{n}{T_0}$ With p=0, 1, 2, (N-1). It follows

with
$$n=0, 1, 2....$$
 (N-1). It follows:

$$\mathcal{F}\{\hat{f}_{T_0}(t)\} = F\left(\frac{n}{T_0}\right) = \sum_{k=0}^{N-1} f(k) \cdot e^{-j2\pi f k T_s}$$

n=0, 1, 2.... (N-1).

$$F\left(\frac{n}{T_0}\right) = \sum_{k=0}^{N-1} f(k) \cdot e^{-j2\pi n k/N}$$
(10)

It usually arises to simplify calculations $T_0 = 1$, where the normal forms of the discrete Fourier transform:

$$F(n) = \sum_{k=0}^{N-1} f(k) \cdot e^{-j2\pi \frac{n}{N}k}$$
(11)
$$N = T_0/T_s = T_0$$

B. Global algorithm of the application



The fundamental frequency is established by gliding average method [3] [12] and STD are given by the following equations:

$$F_{s} = \frac{1}{N} \sum_{i=1}^{N} F_{s}^{(i)}$$

STD= $\sqrt{\frac{1}{N} \sum_{i=1}^{N} (F_{s} - F_{s}^{(i)})^{2}}$

Summary of Results

Table 1: The STD and the average fundamental frequency of healthy subjects of corpus

Healthy subjects										
	1 st	2 nd	3 rd	4 th	5 th	6 th	7 th	8 th	9 th	10 th
	subject	subjec	subject	subject	subjec	subjec	subjec	subject	subject	subject
Fundamental		t			t	t	t			
frequencies Fs (HZ)	Male sexe								Female sexe	
1 st selection	227.05	200.65	190.09	184.81	216.49	205.93	184.81	211.21	200.65	190.09
2 nd selection	184.81	211.2	184.8	211.21	184.81	179.53	205.93	190.09	211.2	184.8
3 rd selection	184.81	184.8	184.8	179.53	200.65	221.77	227.05	195.37	184.8	184.8
4 th selection	227.05	184.8	216.49	184.81	221.77	205.93	211.21	216.49	184.8	216.49
5 th selection	195.12	211.4	184.81	211.21	200.65	184.81	195.37	179.53	184.81	184.81
6 th selection	184.8	195.37	211.21	184.8	184.81	205.93	195.37	195.37	195.37	211.21
The average	200.6	198.03	196.68	192.72	201.53	200.65	203.29	198.01	193.59	195.36
fundamental										
frequencies (HZ)										
STD (sec)	13.94	13.58	13.29	13.2	12.93	13.49	13.17	13.17	12.44	13.28

Table 2: The STD and the average fundamental frequency of sick subjects of corpus

sick subjects											
	1 st	2^{nd}	3 rd	4 th	5 th	6 th	7 th	8 th	9 th	10 th	
	subject	subject	subject	subject	subjec	subjec	subjec	subject	subject	subject	
Fundamental					t	t	t				
frequencies Fs (HZ)	Pathologie tumorale						Pathologie inflammatoire				
1 st selection	73.92	58.08	68.64	47.52	47.52	50.52	105.6	121.44	105.6	174.25	
2 nd selection	73.92	47.52	47.52	52.8	63.36	67.36	147.85	105.6	137.29	163.69	
3 rd selection	52.8	47.5	47.52	52.8	47.52	50.52	126.73	110.88	126.73	142.57	
4 th selection	73.92	42.24	68.64	47.58	58.08	58.08	142.57	100.32	110.88	174.81	
5 th selection	63.36	42.24	68.64	58.08	63.36	68.64	105.6	137.29	121.44	163.69	
6 th selection	73.92	47.52	73.92	58.08	68.64	47.52	110.88	137.29	105.6	142.57	
The average	68.64	47.51	62.48	52.81	58.08	57.10	123.20	118.8	117.92	160.17	
frequencies (HZ)											
STD (sec)	4.97	5.28	4.5	4.29	4.31	4.78	8.97	9.97	8.13	10.31	

Discussion of Results

In healthy subjects, the fundamental frequency was situated at around 200 Hz. corresponding to the physiological frequency of vowel "a". STD extend to 13 Hz.

In patients with the cancer of vocal cords, the fundamental frequency was significantly reduced by around to 60 Hz, while the STD was greatly decreased to 4Hz.

In patients with inflammatory disease (chronic laryngitis), the fundamental frequency was reduced to 120Hz, and STD was decreased slightly to 8 Hz.

In the inflammatory polyp of vocal cords, fundamental frequency slightly decreased to 160 Hz, while STD was slightly decreased to 10 Hz. The spectral range was significantly diminished in cancer patients due to the total absence of vibration of vocal cords. This limitation of frequency range was also present but truncated in the case of chronic inflammatory disease of the larynx.

Conclusion

Characterization and objective assessment of chronic dysphonia were studied using three parameters. The fundamental frequency was around 120 Hz and the STD was around 9Hz in cases of inflammatory disease, while the frequency was low at around 60 Hz and the STD increased to 4 Hz in larynx cancer, where the STD was three times smaller than healthy subjects.

The clinical and para-clinical examinations, notably the pathological diagnosis, were in perfect agreement with the evolution of their indices. Clinical validation of the results is still subject to much larger samples supported by a rigorous statistical support.

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