DOI: http:/dx.doi.org/10.18180/tecciencia.2014.17.5

TECCIENCIA

Speech pattern recognition for forensic acoustic purposes

Reconocimiento de patrones de voz para fines acústicos forenses

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Received: 10/Aug/2014 *Accepted: 12/Sep/2014* *Published: 30/Dec/2014*

Abstract

The present paper describes the development of a software for analysis of acoustic voice parameters (APAVOIX), which can be used for forensic acoustic purposes, based on the speaker recognition and identification. This software enables to observe in a clear manner, the parameters which are sufficient and necessary when performing a comparison between two voice signals, the suspicious and the original one. These parameters are used according to the classic method, generally used by state entities when performing voice comparisons

Keywords: Acoustic parameters Software, Digital Signal Processing, Speaker Identification, Voice.

Resumen

El presente artículo describe el desarrollo de un software para el análisis de los parámetros acústicos de la voz (APAVOIX), que puede ser usado para los propósitos acústicos forenses, basado en el reconocimiento y la identificación del hablante. Este software permite observar de una manera clara, los parámetros que son suficientes y necesarios cuando se realiza una comparación entre dos señales de voz, el sospechoso y la original. Estos parámetros se utilizan de acuerdo con el método clásico, utilizado generalmente por las entidades estatales cuando se realizan comparaciones de voz

Palabras clave: parámetros acústicos de software, procesamiento de señales digitales, la identificación del hablante, de voz.

1. Introduction

Forensic acoustics is an unknown branch the field of Sound Engineering, nevertheless it is a necessary branch, widely used for performing comparisons between two or more voice signals, with the aim to prove guilty or innocence of a suspect involved in a justice process. For the development of this work, the responsible person judges, require a tool, which enables them to make a correct comparison of certain parameters, which enable to give an specific characterization to the voice of any speaker, in the same manner that a fingerprint would do it; that means that two voices cannot be equal, due to physiological characteristics of each speaker. Each of these comparisons are performed through the classic combined method, which is the result of years of research in the field of forensic acoustics, in order to have an exact reference of the elements to analyze and the results obtained in voice comparison.

This method is analyzed in a qualitative manner, without quantitative values, which enable to give more weight to the voice recordings. Nevertheless, it is possible to give a value to some of these variables, in order to be more accurate in a judgment, but the judges of this process do not have such a wide and specific knowledge in Digital Signal Processing, and other parameters. Therefore it is necessary to rely on a professional person with this type of knowledge, in order to have more processes with justice, and that involved persons could prove their guilty or innocence in a more exact and reliable manner.

The main goal of the present work is to implement an algorithm which enables to analyze the sufficient and

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How to cite: Herrera Martinez, M. et al., ***Speech pattern recognition for forensic acoustic purposes,*** TECCIENCIA, Vol. 9 No. 17., 37-44**,** 2014, **DOI: http:/dx.doi.org/10.18180/tecciencia.2014.17.5**

necessary acoustic characteristics of the human voice used in speaker identification.

Therefore, there is the need to identify the fundamental frequency and the intensity, the energy spectrum, LPC (Linear Predicting Code), the formants, signal-to-noise ratio (S/N), the signal spectrogram, time percentage of the presence of the voice inside the recording and the gender, required in the analysis of the behavior of the voice, which give the proper characteristics of each speaker.

Afterwards, the software is developed, in order to analyze the characteristic parameters of the voice in junction with the graphic interface.

Tests are performed in order to prove the adequate performance of the software, through a professional software that would be developed for voice analysis.

Finally, the influence of external agents in the signal capture or processing, such as background noise, the space in which capture is made, sampling frequency and the compression format, are also analyzed.

2. Theoretical background

In order to develop a software for the analysis of voice acoustic parameters, it is necessary to know about the phenomena in speech generation, as well as the parameters that may be used for the performance of voice comparison in the field of forensic acoustics.

2.1. Voice General Concepts

Voice is produced by the phonation organs. The voice apparatus consists on:

The respiratory apparatus is the motor which gives the intensity, the force, power and the sustain. It consists on the lungs, the bellows, and the air cavity. The respiratory apparatus is divided on two main parts:

Superior respiratory pathways: They consist on the nassal cavities, nasal pharinge, and the accessory cavities; they constitute the first part of the air trajectory, penetrating through the nose.

The inferior respiratory pathways are constituted on the larinx, trachea, bronchial tube and lungs. The lung is the essential organ of respiration. The air is stored in the lung alveoli.

The vibrating vocal organ is the sound generating mechanism; pitch is given by the vibrations of the vocal chords. It is composed by the larinx, vocal chords and the ventriculus.

The larinx is the organ where the sound is generated and it is covered by a mucosae membrane, provided with segregation glands. In the middle part of the larynx, there is a region called glottis, constituted by the vocal chords; they are two mobile bands, jointed in its anterior part, leaving a triangular space for the glottis. In order to determine the opening or closing of the glotis, there exist the tension muscles and constrictors, respectively. The vocal chord muscle tenses the vocal chords, also called aritenoides.

The resonance system gives the timber, color and harmonic enrichment. It is the sound reinforcement. It enables voice localization and range. It is compossed by the resonators and the resonance cavities. It may be divided in:

- **The fixed hard parts**:
These are the bone parts, the superior maxilar, the nasal bones, the cavities and the bóveda palatina ósea, teeth. These parts are rigid, fixed and hard. In order to favor the resonance, it is necessary for them to be flat. If there are tiny obstacles inside the nasopharinge, polips inside the nostrils, liquid or pus inside the cavities, gross mucosa, the voice will be blind, and the resonance will be hardly produced.

- **The soft mobile parts:**
These are membrane-muscle of the pharingea: the veil of the soft palate, the tongue, the cheeks, and the lips. But there exists a mobile bone, the inferior maxilar. These parts must be sane, free, and with good mobility. If there is a volume tonsil, then the tongue movements, or movements from the palate veil would be impaired, and moreover, these masses would constitute an obstacle at the exit of the sounds. The voice location will be not adequate, resonance will be diminished and the range will be less [1].

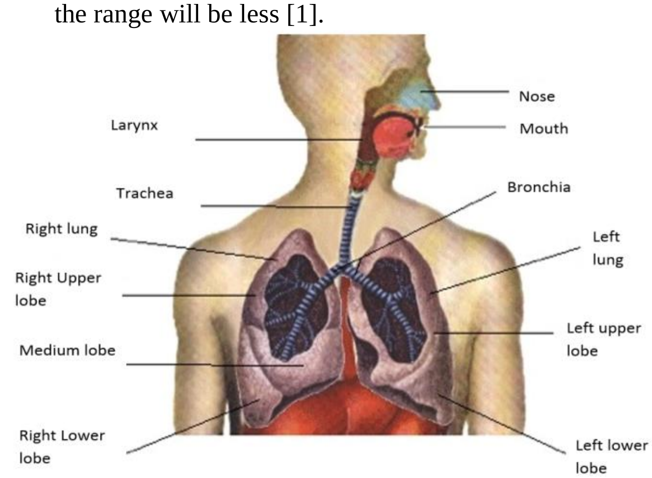


Figure 1 Phonate Apparatus [2]

2.2. Formants and Spectrogram

Due to the fact that the vocal tract evolves in time in order to produce different sounds, spectral characterization of the voice signal will be also variant in time. This temporal evolution may be represented with a voice signal spectrogram or a sonogram. This is a bi-dimensional representation that shows the temporal evolution of the spectral characterization.

The formants appear as horizontal frames; instead of that, amplitude values in function of frequency are represented in a gray-scale in vertical sense. There exist two types of sonograms: wideband, and narrow-band.

In the case of the wideband spectrograms, a good temporal resolution will be obtained. Instead of that, in the case of narrowband spectrograms, a good frequency resolution will be obtained, because the filter enables to obtain more precise spectral estimations [1].

2.3. Localized Analysis in The Frequency Domain

A conventional analysis in the frequency domain for voice signal gives little information, due to the special characteristics that this signal has. In the voice-signal spectrum, two convolutions components appear.

One comes from the fundamental frequency and its harmonics, and the other from the formants of the vocal tract. Therefore, the only information that may be achieved is a visual one, through the spectrogram. Depending on the type of the spectrogram (wideband, narrowband), one of the two components may be achieved (formants or the refined structure).

2.3.1. The Fast Fourier Transform

The Fast Fourier Transform (FFT) is based on the reorganization of the signal. In the same way as the TDF, in the FFT, the transformed signal into the frequency domain, must be decomposed on a series of sines and cosines, represented by complex numbers. A 16-point signal must be decomposed in 4, then in 2, and in this way successfully, until the signal is in point-to-point way. This operation is a reorganization in order to diminish the number of operations and to improve its velocity. After finding the spectrum frequency, these may be reordered in order to find the operation in the time-domain [3]

2.4. Analysis in the Cepstral Domain

The spectrum or cepstral coefficient c (τ), sis defined as the inverse Fourier Transform of the logarithm of the spectral module | X (ω) |

$$c(\tau) = IDFT[Log|X(\omega)|] \tag{1}$$

The term "cepstrum” is derived from the english Word “spectrum”, the independent variable in the spectral domain is “quefrency”.

Due to the fact that the cepstrum represents the inverse transform of the frequency domain, the “frecuency” is a variable on a pseudotemporal domain.

The essential characteristic of the cepstrum is that it enables to separate the two contributions of the production mechanism: fine structure and spectral envelope [4].

2.5. Analysis by LPC (Linear Time Prediction)

Linear prediction is a good tool for the analysis of voice signals. Linear prediction models the human vocal tract as an Infinite Impulse Response IIR system, which produces the voice signal. For vocal sounds and other voice regions, which have a resonant structure and a high grade of similarity through temporal changes, this model predicts an efficient representation of the sound.

In order to use LPC, one must get into account:

- Vocals are easier to recognize, using LPC.
- Error may be calculated, as a^T Ra, where R is the matrix of autocovariance or autocorrelation of a segment and “a” is the prediction coefficient vector of a standard segment.
- A pre-emphasis filter, before LPC, may improve the performance.
- Tone period is different in men and women.
- For voice segments, (r_ss [T])/(r_ss [0])≃0.25 where T is the tone period [5].

2.6. Pitch

The prosodic information, that is, the tonation velocity, is strongly influenced by the fundamental frequency of the vibration of the vocal chords f0, which inverse is known as the fundamental period T0. In a general way, periodicity is given in intervals of infinite analysis; nevertheless, its estimation is realized over finite intervals, in such a way that many pitch periods are covered or by the difference between two consecutive time instants of the glotic closure [6].

2.7. Signal-to-Noise Ratio

S/N relationship gives a quality measure of a signal on a determined system, and it depends, on the received signal as well as on the total noise; that is the sum of the noise that comes from external sources and the inherent noise of the system. In system design, it is desired that signal-to-noise ratio to have a high value. Nevertheless, depending on the application context, this value may change, because, high S/n values leads to huge costs. An adequate value of this relationship is that when the received signal may be considered without defects or with minimal defects [7].

3. Software design and implementation

For the SW design, it was necessary to perform a previous exploration of the necessary parameters in speaker identification, with emphasis in forensic acoustics.

For this purpose, software Computerized Speech LabCSL4500de Kaypentax® was taken as a reference, with the proper characteristics fpr the userin order to develop an original SW APAVOIX (Análisis de Parámetros Acústicos de la Voz).

At the development stage of this software, the graphic interface was also realized, using the GUI (Graphic unit Interface) in Matlab®.

For the spectrogram design, "VOICEBOX" [8] Matlab toolbox was used, which contains many features for voice parameter evaluation. Function spgrambw, from this toolbox, enables to plot a spectrogram in MATLAB, using a more precise algorithm, than using specgram from MATLAB.

In order to use this tool, the program requires the audio for the analysis, which in this case, it is a selected fragment by the cursors, the sampling frequency, 11025 Hz, because for voice-signal-processing, it is not necessary to use a high sampling frequency. Besides this, the function enables to modify the spectrogram colors, in order to clear the visualtion, to modify the scale of visualization (Hertz, Logaritmo, Erb, Mel, or Bark).

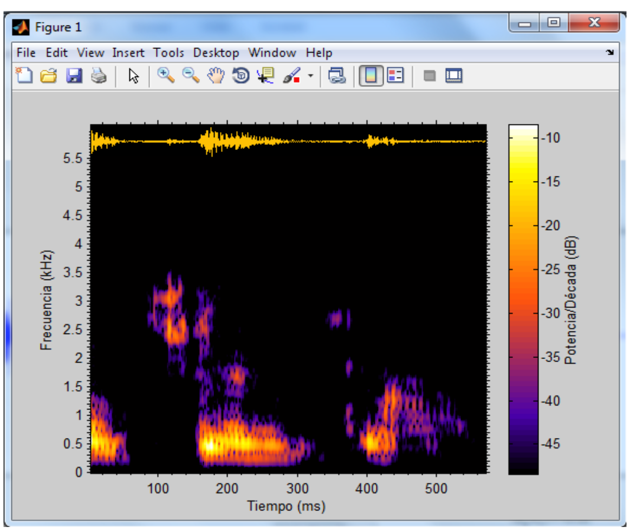


Figure 2. Spectrogram

Fort the Fast-Fourier Transform, the audio segment is selected; it must be small, as for example a vocal, due to the fact that this analysis requires to be performed in a small portion of the audio, but with relevant information, as the signal formants present in the vocals.

Afterwards, a pre-emphasis filter is applied, which is optional, and it may be modified by the SW parameters, in a range 0 a 1.5; afterwards, fft function is used in order to obtain the components of the Fast-Fourier-Transform. These components may be depicted applying the signal-windowing (Hamming, Hanning, Blackman, rectangular or triangular), besides of using the simple size, 8192 points, but it may be configured by the established quantity of points in the SW, and after this frequency limitation is performed.

This parameter is modifyble as well. Afterwards, minimum and maximum values of the frequency are taken and depicted, with the dBu values of the energy in each frequency value. Besides this, in the upper part, a graphic with the analyzed waveform will appear.

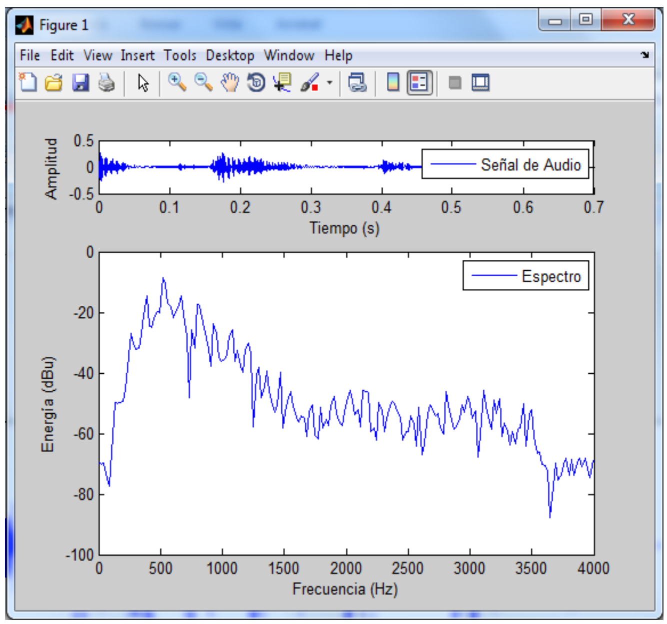


Figure 3. FFT

LPC is implemented as well as FFT, taking a vocal signal fragment, and applying a pre-emphasis filter. Afterwards, spLpc function is applied, from the sptoolbox [9] which enables to obtain the LPC coefficients with the selected signal, 11025 Hz, sampling frequency, and with the filter order, varying by multiples of 2, from 2 till 36. Afterwards, window type may be modified, and to select the sample size for the analysis.

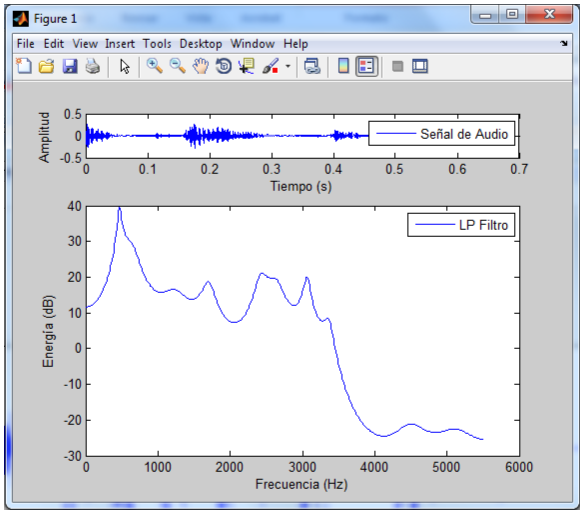


Figure 4. LPC

Fort the Cepstrum calculation, a signal fragment is taken, pre-emphasis is performed, FFT, and windowing were performed. Logarithm is applied, and finally the inverse Fourier Transform; in this way, cepstrum values are obtained. These values are called "quefrencies"; pseudotemporal values in ms. From the "quefrency" values and the cepstrum, the graphic is posible.

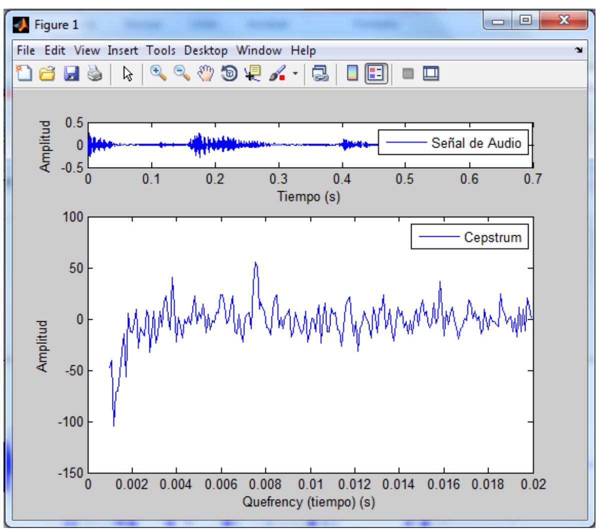


Figure 5. Cepstrum

For the pitch calculation, spPitchTrackCepstrum function is used from sptoolbox, which requires an audio track, sampling frequency (11025 Hz), frame length in ms, frame superposition and the window type used for the tone calculation. This function enables to perform tone calculation, which prevails in the signal, through the cepstrum, detecting the main frequency, related to the "quefrency" for each frame and afterwards it is depicted.

Cepstrum therefore calculates the fundamental frequency of the voice. If a complete word is taken, tone variation is observed, related to the consonant produced, and its nature of pronunciation (oclusive, fricative, nasal, lateral or vibrant), and the toney mantains itself constant on a determined vocal.

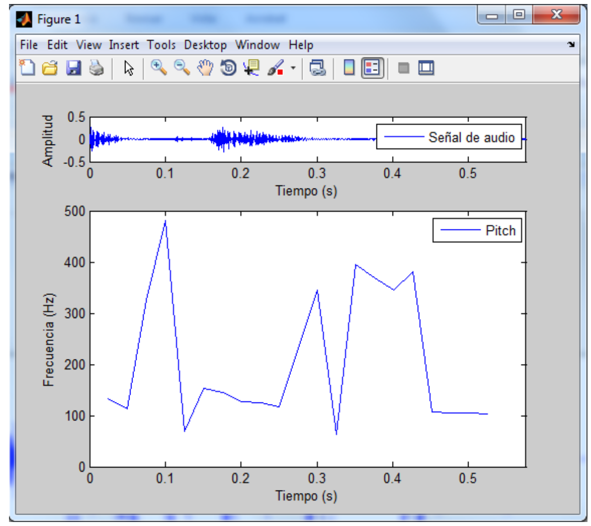


Figure 6. Pitch by Cepstrum

For the pitch calculation (by autocorrelation), the spPitchTrackCorr function from the sptoolbox is used. This method enables to have the information of the repetitiveness of the signal with respect to itself; because of this, the pitch value, while repeating, not with the same values for the vocals, but at least close to each other generates a higher value in the correlation. The highest values are taken and the relative distances between values are calculated and then the maximal common divisor of these distances corresponds with the value of the fundamental period. This also has periods of great variations, because of the consonants, and the characteristics of its emission.

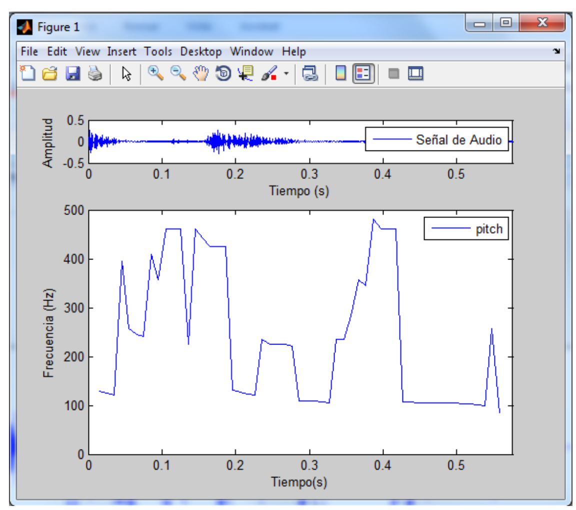


Figure 7. Pitch by Auto-Correlation

In order to calculate the robust pitch, a robust tone estimation filter has been designed for high levels of noise

(PEFAC), and the calculation has been performed, through the fxpefac function, from the voicebox toolbox. This type of estimation is present in the parts where vocals are found, and consonant behavior is not shown. Therefore tone behavior may be observed.

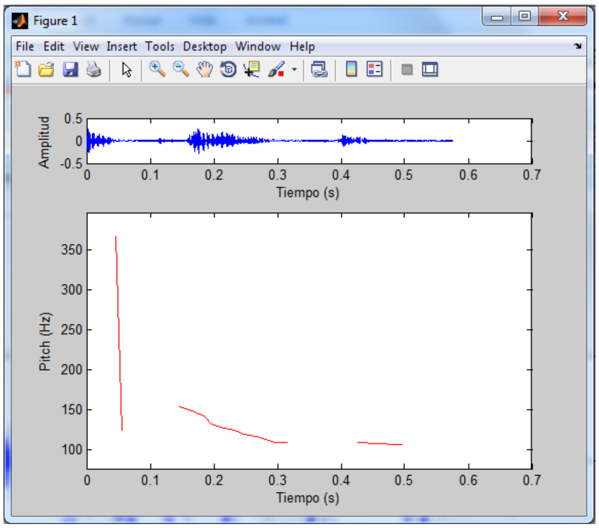


Figure 8. Robust pitch

To depict the formants, spFormantsTrackLpc function was used, from the sptoolbox. This function performs the LPC method. The signal fragment to analyze is required, sampling frequency (11025 kHz.), the filter order, frame size, the superposition and the window type. Initially, formant values are obtained, using the LPC; afterwards the frequency range is divided in 5 parts, calculating in this way, for each formant separately, and enabling to modify points, in order to observe the different colors by formant. Similar to the pitch estimation by correlation and cepstrum, in the case of the formants, the ones which are present in the consonant space, because of their irregular form and undefinition, are not always taken into account; therefore vocal formant are often necessary and sufficient for the analysis.

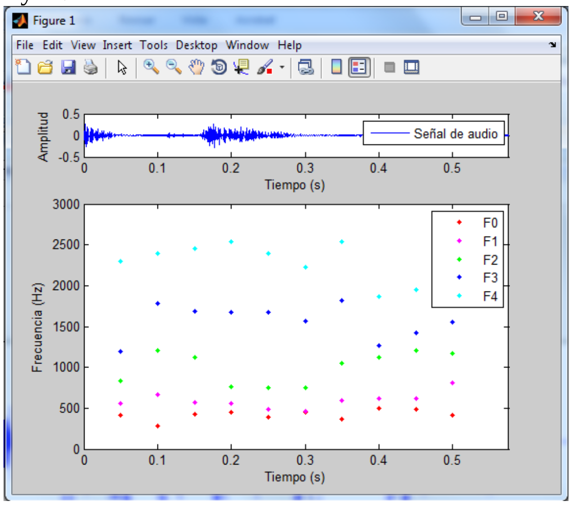
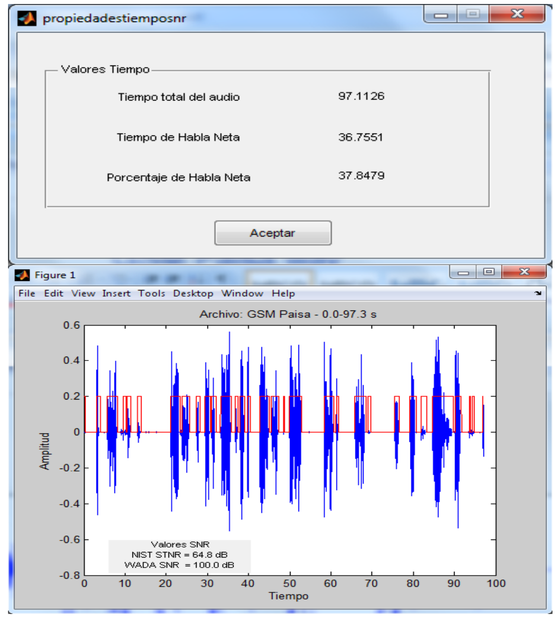


Figure 9. Format graphics

In order to obtain the signal-to-noise ratio, snreval [10] function is used. It belongs to a collection of functions, which measure some objective metrics for speech quality. This function enables to observe in a graphic manner the whole signal; points where speech is present, and points with no relevant information. Besides this, it gets the signal-to-noise ratio values, Wada and Stnr. File name and file address are the only requirements for this function to calculate. Modifications were performed to this function in order for APAVOIX to correctly work.



(a)

(b)

Figure 10. S/N Results (a) Speech-time properties (b) Voice activity

At the moment of energy extraction, the selected voice fragment is taken, and afterwards it is processed by the Hamming window. Afterwards, the summatory of the absolute value of the signal is performed and squared. This is the energy calculation and from here, graphic of the behavior of the audio-signal energy is obtained, with the analyzed signal.

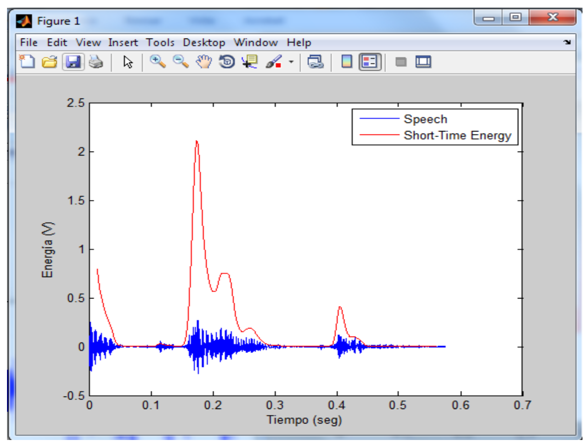


Figure 11.. Signal energy of the signal (Voltage)

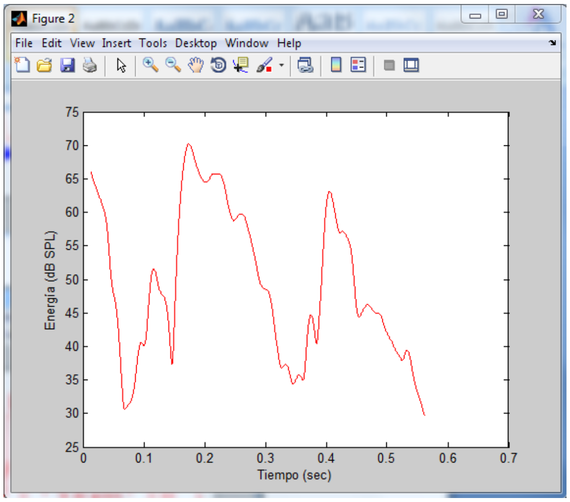


Figure 12. Signal Energy of the signal (SPL)

In order to obtain the gender, the autocorrelation is the used method, through which, a signal fragment is taken, maximal and minimal frequencies are calculated, between 50 and 500 Hz.

Signal autocorrelation is performed; region containing maximum points is analyzed and the main frequency is found. If the obtained value lies in the range from 80 to 190 Hz, the detected gender will be masculine; from 190 till 255 Hz, femenine gender will be detected; if the value does not lie in none of the two parameters, it will show that recognition is not possible, and other part of the audio signal should be selected.

APAVOIX counts with a graphic interface, which enables to handle easily the required funtions in speech analysis. Besides this, it enables to load the desired file, record it and stored it; it has cursors to select the signal part to analyze. This cursors were obtained by the function dualcursor [11], which can be actualized, and evenmore their positions may be stored as .mat files, in order to be loaded and used for further analysis.
A toolbar is provided, for zoom, signal shifts, and reproduction.

4. Conclusions

The designed software (APAVOIX) enables to understand the voice signal digital processing.
As stated at the beginning of the project, the software is capable of showing the sufficient and the necessary characteristics when working with voice-comparisons:

- The fundamental frequency, which varies according to the vocal emission under evaluation: consonant or vocal pronunciation.
- Signal intensity.
- Spectrum and signal energy spectral density, through periodogram or the Welch method.
- LPC.
- Formants, which, vary in a similar way to the fundamental frequency; that is, according to the emitted sound and they keep present during the whole time of the analyzed signal.
- Signal-to-noise ratio using two different algorithms for an optimal result, taking into account that WADA is part from the initial NIST algorithm. It enables to visualize the time where the voice signal is present inside the recording.
- Time-percentage that the voice is present inside the recording.
- The gender, evaluated through the correlation method.

The software enables to observe the energy per frequency, through the FFT (the Fast Fourier Transform), and the cepstrum of the signal which enables to identify the fundamental frequency in terms of “quefrencies”.
Besides this fact, the software counts with different characteristics, which may be modified according to the analysis parameter, in order to obtain the most precise and easy results, in order to be visualized for the persons that perform voice comparisons.

APAVOIX has tolos that enable to work easily with the signal, as cursors, play buttons, zoom and panning. It also counts with a filemenu that enables to load and store data from cursors. Moreover signal recording and storage is available.

External agents influence the signal capture, as background noise, or the space for this realization. It is also necessary that an expert evaluate the voice conditions; that is, noise conditions, voice under communication channels such as GSM.

The software is designed to work under a sampling frequency of 11025 Hz. Only WAV formats are supported, and 16 bits of resolution, mono-channel.

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