

Dstvrgrgy some vehicle operators by using Wavelet the sound driverAref Aghayari¹¹. Department of Mechanic, Takestan branch, Islamic Azad University, Takestan, Iran

Abstract: With advances in technology, and mechanical key into the electronic transformation of the many operators to control multiple devices as well as emerging issues relating to the collection and processing of data retrieval is possible and storage. The explanations for some of the functions that are not critical vehicle parameters intended for passengers comfort, sound can be used for the by the drivers for the control of not need to be operator mechano. In this paper, wavelet methods to do this with the car in vitro were investigated. Data processing and access to favorable results of the wavelet transform as suitable for signal analysis in time domain - frequency was used. Using the wavelet coefficients for each of the files and feature extraction of wavelet and principal component analysis method was used to select features. The optimal Bayes classifier from a statistical technique used to separate data by using a decision function for each of the sounds and audio files were found. Finally by using ISA method, Programs written in MATLAB program was sent to the output port on the computer. By the various wire harness connection between driver and car audio according to the performance of each of these mechanisms was tested.

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Introduction:

One of the areas of speech processing, in particular telecommunication engineering, electrical engineering and digital signal processing is the aim of the talks. The processing tasks such as sampling, compression, speech synthesis, etc. is entered. When working in other branches of science and engineering topics speech processing occurs such that it can be acoustic, digital signal processing, linear algebra, physiology, linguistics, computer, etc. The. All surrounding areas, although this may not be necessary, but partial knowledge of the above topics can be useful in a better understanding of speech processing. Another remarkable thing is that most of the progress achieved in the field of signal processing and speech processing is a must. For example, the oldest Voice Applications in speech processing is proposed by landline or mobile. Mtmrkr three major areas of speech processing in general is:

- 1- speech signal to a digital display and parameters are also important in terms of its
- 2- Implement processing techniques (a computer program).
- 3 - employing algorithms and implementation techniques for speech processing

Sampling theorem (sampling theorem) imply, as a number of samples, each sample is usually used in computer, an integer is to be expressed. The implementation of signal processing algorithms implemented with this sometimes changes in

algorithm parameters and threshold limits, according to the speech signal is discussed. These algorithms on speech files (containing the input speech samples) was applied and the desired processing results are recorded. For processing the speech audio files into a series of numbers with a feature rich software that makes audio files of the work is done.

Speech Recognition:

The time it is entered. In order for a text to speech conversion, there are typically two steps:

Level of education:

The system should have already taken a series of words on them, learn the characteristics of different parts of speech. A series of models or patterns of speech must be obtained from the archive.

Test phase:

At this stage, the main stage of the speech signal as a set of features that can be extracted from it. The next step, features are compared with previous models. In general, the level of phonemes (units of sound), then the use of linguistic information, syntactic and semantic errors will be detected and corrected.

Fourier analysis:

One of the mathematical tools for research into signal processing is an area to which signals from other areas and thereby transmit the hidden information, they will reveal a. Change the fact that the windows are visible signals of different angles and some more

useful features of the show. A transformation can be written in general form:

$$X(a, b) = \int_{-\infty}^{+\infty} x(t) \cdot \Psi_{a,b(t)} dt \quad (1)$$

Where $x(t)$ is the signal is processed.

$$\Psi_{a,b(t)} = e^{-j2\pi ft} \quad (2)$$

Is frequency: f

$$\Psi_{a,b(t)} = e^{-j2\pi ft} \Psi(t-b) \quad (3)$$

$$a = f$$

$$\Psi_{a,b(t)} = \frac{1}{\sqrt{|a|}} \Psi\left(\frac{t-b}{a}\right) \quad (4)$$

$$X(t) = a_0 + \sum_{n=1}^{\infty} a_n \cos(n\omega_0 t) + \sum_{n=1}^{\infty} b_n \sin(n\omega_0 t) = \sum_{k=-\infty}^{\infty} a_k e^{j(k\omega_0 t)} \quad (5)$$

$$a_k = \frac{1}{T_0} \int_{T_0} e^{-j(k\omega_0 t)} x(t) dt \quad (6)$$

Non-periodic signals can be obtained from (7) of the form (8) wrote.

$$X(\omega) = \int_{-\infty}^{+\infty} x(t) e^{-j\omega t} dt = \langle e^{j\omega t}, x(t) \rangle > 0 \quad (7)$$

$$X(t) = \frac{1}{2\pi} \int_{-\infty}^{+\infty} X(\omega) e^{j\beta\omega t} d\omega \quad (8)$$

As in equation (7) was shown to be the Fourier coefficients of the inner product of the signal with sinusoidal basis functions on an infinite interval are calculated and the frequency components of the signal amplitude into the place of regardless of their the available shows. So the focus is no place or time Fourier transform, and it can not help the signal parameters can be determined along the time axis or space. Subject to periodic signals that are stationary or in other words, is not a problem, but for non-stationary signals, which are changes along the time axis, can not work, they reveal the characteristics. A solution to this problem is limited to the length of the signal through the windows. Assuming that the signal is stationary over the window, rather, they are treated as static signals can be overcome to some extent. This method was originally proposed by Gabor transformation term and short-term strategy to become a name known which can be expressed as:

$$STFT(t_0, \omega_0) = \int x(t) g(t-t_0) e^{-j\omega_0 t} dt \quad (9)$$

follows.

$$= \int_{-\infty}^{+\infty} x(t) g_{(\omega_0, t_0)}^*(t) dt$$

$$g_{(\omega_0, t_0)}^* = e^{j\omega_0 t} g(t-t_0) \quad (10)$$

$$\sigma_t^2 = \int_{-\infty}^{+\infty} t^2 |g(t)|^2 dt$$

$$\sigma_\omega^2 = \int_{-\infty}^{+\infty} \omega^2 |g(\omega)|^2 d\omega \quad (12)$$

Are SD of $g_{\omega_0, t_0}(t)$ σ_t, σ_ω

Fourier transform:

$$STFT(t_0, \omega_0) = \int x(t) \cdot g_{t_0, \omega_0}^*(t) dt \quad (13)$$

$$= \int x(\omega) \cdot g_{t_0, \omega_0}^*(\omega) d\omega$$

$$[\omega_0 - \sigma_\omega, \omega_0 + \sigma_\omega], [t_0 - \sigma_t, t_0 + \sigma_t]$$

calculation between

Given a fixed length window for all frequencies will be fixed resolution of the converter. With regard to the issue of uncertainty (Heisenberg unequal) we have:

$$\sigma_t^2 \cdot \sigma_\omega^2 \geq \frac{\pi}{2} \quad (14)$$

$$\Delta f^2 = \frac{\int f^2 |G(f)|^2 df}{\int |G(f)|^2 df} \quad (15)$$

$$\Delta t^2 = \frac{\int t^2 |g(t)|^2 dt}{\int |g(t)|^2 dt} \quad (16)$$

$$\Delta f \cdot \Delta t \geq \frac{1}{4\pi} \quad (17)$$

According to the above equation can not be arbitrary small in size but there is a balance between time and frequency resolution of the two images together and act. Gaussian window was establishes equality of states for the highest resolution in this case is, in practice most applications. In order to know the appropriate time and frequency resolution with the width of the window when we used to. Thus the frequency of small, high window width and window width is small at high frequencies. The concept of variable window, the first step was the development of wavelet transform. This method is similar to a microscope Rami mathematics can be assumed that it

will be able to set up different portions of a signal we studied.

Wavelet analysis:

After an introduction to Fourier transform and short-time Fourier transform was mentioned in the previous section, the two become fully meet the needs of non-stationary signal analysis are not particularly. The wavelet analysis window size is variable and accurate information about low frequencies with large windows and high frequencies with precise information about the small window is available. Figure 5 shows this.

Recently wavelet theory is a powerful tool in analyzing and understanding the phenomena is used due to its nonstationary nature of the Fourier transform is not available. Data preprocessing, analysis, compression, noise removal and feature extraction of the signal from the converter are used.

$$\Psi_{a,b(t)} = \frac{1}{\sqrt{|a|}} \Psi\left(\frac{t-b}{a}\right) \quad (18)$$

$$\begin{cases} C_\omega = \int_{-\infty}^{+\infty} \frac{|\hat{\Psi}(\omega)|^2}{|\omega|} d\omega < \infty \\ \int_{-\infty}^{+\infty} \Psi(t) dt = 0 \\ \|\Psi(t)\| = 1 \end{cases} \quad (19)$$

$\hat{\Psi}(\omega)$ is the Fourier transform of $\Psi(t)$.

Principal components analysis and processing:

One of the problems in pattern recognition and classification issues in the determination of the quantities to be extracted is the choice of. The purpose of measured physical properties into mathematical properties. A sampling of current in an electrical circuit that converts mathematical model to quantify the flow. The feature extraction technique for the production of new features such as conversion or a combination of the main features. The choice of features, a subset of features is extracted from the Minimum Data species that disappear. In fact, the chosen feature vector of reduced dimension and sometimes it is called the reduced dimensions. Several methods have been introduced to reduce the dimension of feature vectors, which are mentioned below.

Independent Component Analysis and Processing:

Independent component analysis to separate signal sources or variable data is used. In this method, the characteristics of the components are calculated according to them and they are made from them.

$$(25) \begin{cases} x_1 = k_{11}(IC_1) + k_{12}(IC_2) + \dots + k_{1m}(IC_m) \\ x_2 = k_{21}(IC_1) + k_{22}(IC_2) + \dots + k_{2m}(IC_m) \\ \cdot \\ x_n = k_{n1}(IC_1) + k_{n2}(IC_2) + \dots + k_{nm}(IC_m) \end{cases}$$

Analysis and linear separation process:

The analysis and processing of linear separation of class labels can be used for dimension reduction. LDA, as a method for supervised dimension reduction which is aimed at increasing the distance between vectors in different classes of vectors in a class is reduced.

Random illustration:

Illustration of random matrix methods that are randomly selected for dimension reduction of image vectors and vectors from one space to another space actually used. It has been shown that if the space of random vectors with dimensions large enough image, the similarity between data remained constant.

Principal components analysis and processing:

Principal components analysis and processing method, one of the most practical methods for feature dimension reduction in practical applications. This method of finding the number of feature vector with orthogonal linear combination with the largest variance in the data is reduced.

$$(26) \begin{cases} PC_1 = k_{11}x_1 + k_{12}x_2 + \dots + k_{1n}x_n \\ PC_2 = k_{21}x_1 + k_{22}x_2 + \dots + k_{2n}x_n \\ \cdot \\ PC_m = k_{m1}x_1 + k_{m2}x_2 + \dots + k_{mn}x_n \end{cases}$$

Data collection :

One of the important topics in pattern recognition step is to gather information , identify or distinguish the correct model depends on having access to the information is correct and true . In this project, using MATLAB software to various audio files that were recorded by different individuals were used as initial data . The noise is due to the beginning of each audio signal was detected. converter or a combination of the main features In this paper, a class of audio signals with sampling frequency of 11025Hz and the wavelet transform as suitable for non- stationary signal analysis has been used . Different coefficients for different levels of signal analysis has been considered as the main feature . Then the function of

different factors combined to means necessary characteristics to be extracted.

Determining patterns of classes and each class :

Since the aim of this project is to identify the sound driver and ten of the ten different classes are used. Thus , each of the two audio files at one time is said to have been placed in this class .

How to extract characteristics :

is used. Any audio files according to instructions given by the driver that is contained 3 -second sound signal data sampling frequency is 11025Hz . Selection of mother wavelets for signal analysis using wavelet transform is one of the steps. Coefficients were calculated from levels 1 to 5 for each of the signals from each of the coefficients by using different functions , different features are extracted . Thus, any pattern or feature vector consists of 5 digits or characters for each of the audio files.

Feature selection:

To reduce the dimension of feature vector access patterns used for classification of principal component analysis (PCA) was used . Thus, the covariance matrix and the diagonal form of the transformation matrix A , respectively. The least important characteristics with low energy were excluded from the feature vector and the feature vector is reduced to that extent.

Bayes classifier:

After selecting the best model should be reduced feature vector of each class to be classified . To test , put half of each class were selected for testing and the other half were used for training . Using 10 patterns per class (patterns Train) mean vector and covariance matrix of each class , and the decision was made for each class . Then the test data , and the use of classifiers as a criterion for diagnosis was confirmed by this method .

Voice recognition:

The voice recognition by placing each audio file read by different people, different classes of decision functions are calculated. The intended function of the class that is more decision-making. It should be noted here that the DSP is one of the ways it can be used in order to analyze the signal processing performed on the signal. The relevance of the ISA is written in MATLAB using the computer and the outside world . The ISA can be ordered with different bits that are classified according to the type and function, the MATLAB program and to establish the connection. The hardware used to communicate with the car adapter, before converting the design of the

mechanism that is supposed to be done by voice, the driver of the vehicle, the same parameter is passenger comfort , the exact mechanism by which it has become familiar and Maps they were electric .

Accuracy of the simulation results:

In this work, the feasibility of the relationship between car and driver were via voice command. The first stage of processing complex speech topic was discussed.was used. With the application of these techniques in the world to communicate with the computer's sound driver program was written in MATLAB . The digital signal processing (DSP) was administered by a strong application . This microphone is the sound drivers for the same sensor as the analog signal into a computer and analyzed by a program written in the voice of the person (driver) and type the command was recognized . After identifying the command, this command will turn connections to the outside world applications and software programs .ISA MATLAB command feature rich program used to transfer commands are identified with a unique code (the binary code) and ISA hardware outside the computer user provides. After removal number , address and command type is specified. Then an adapter to connect the performance of each of the numbers used. To communicate properly and perfect the mechanism of each of the members were to be guided by sound driver and all drawings related to electrical converters have been investigated . After some pieces of identification mechanisms such as coolers , heated glass rear window and the top of the driver, the vehicle wire harness components that are connected by also were identified . Wiring harness adapter associated with each of the components . Figure 13 connector for connecting to the vehicle wiring harness to connect to the show. The relationship between each of the wiring harness instructions for the final part of this work was carried out successfully and to communicate with the driver of the vehicle was provided. Figure 14 vehicles communicate with a computer and a microphone mounted on the car to get the driver out of the show. This work is part of the emerging science and technology this technique is difficult to reach any of the horizon to the feasibility of this research were provided .

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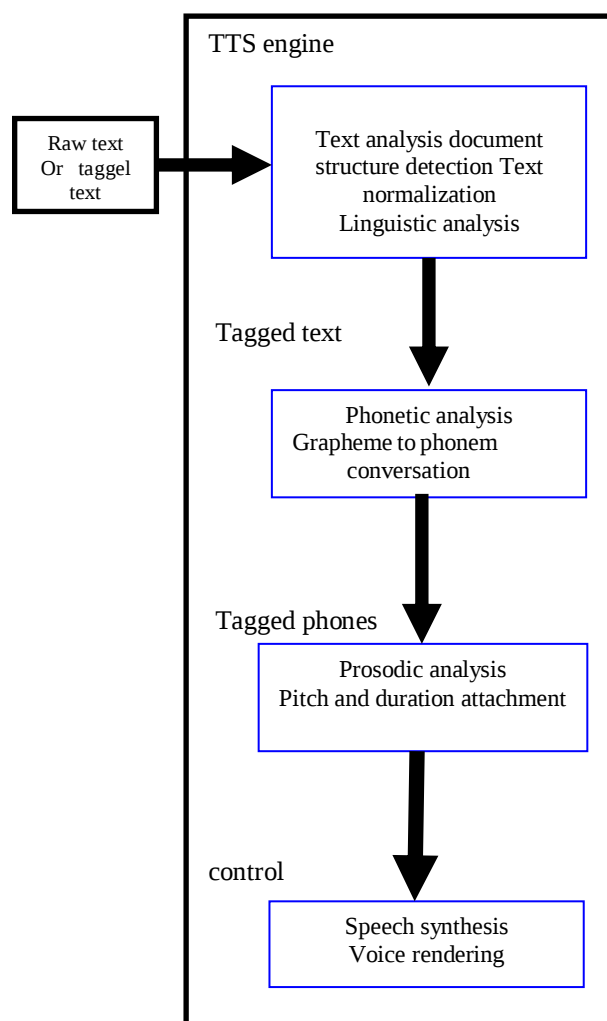
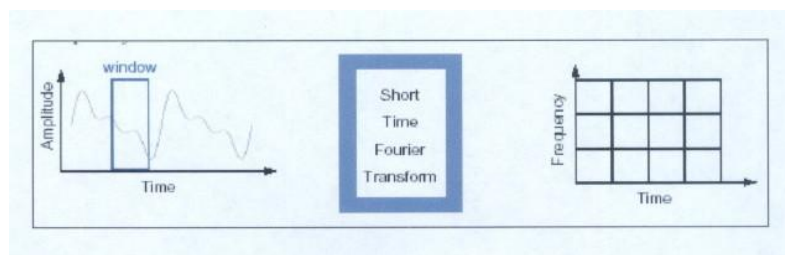


Fig1

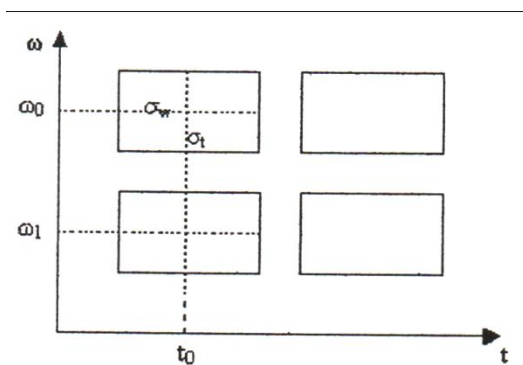
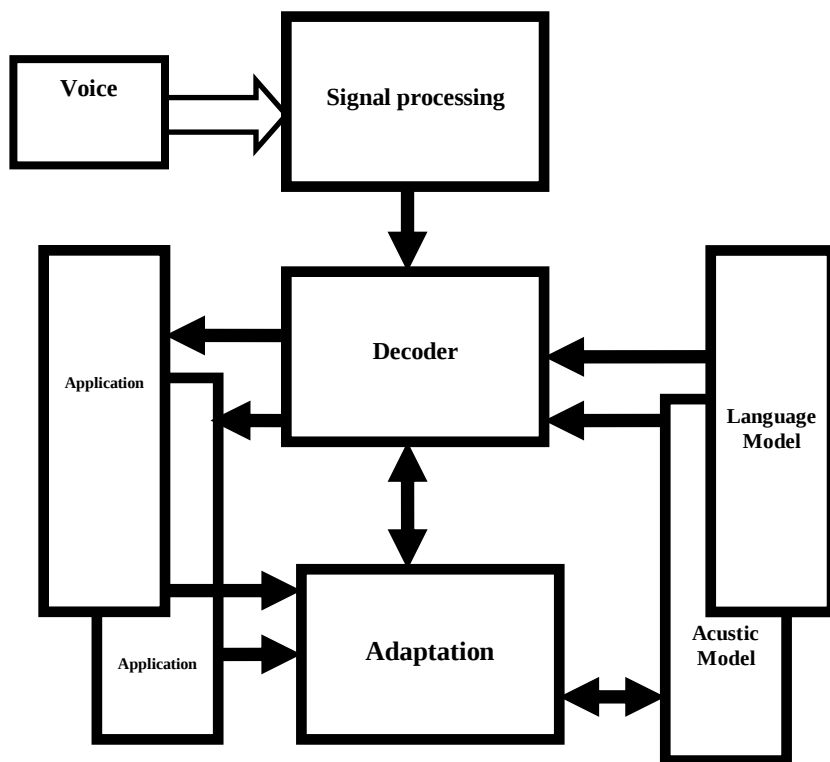


Fig3

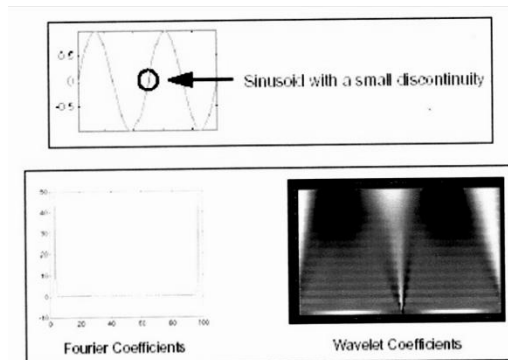


Fig5

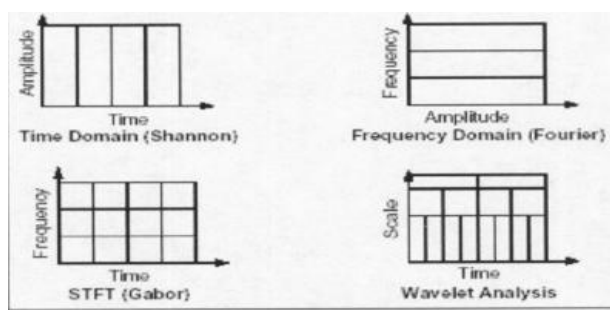


Fig4

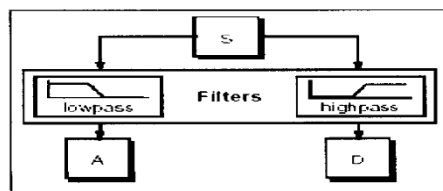


Fig6

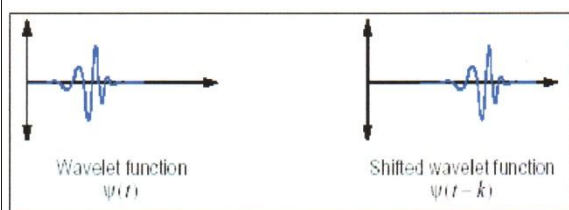


Fig7

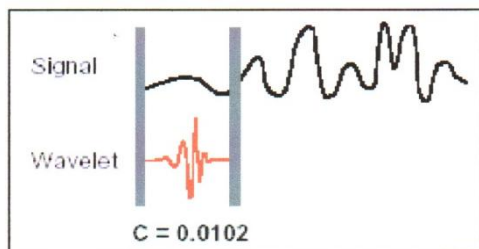


Fig8

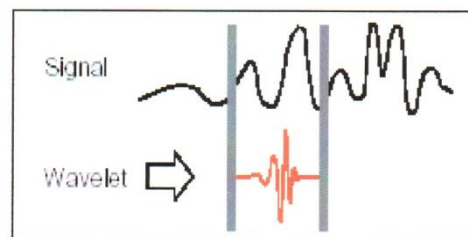


Fig9

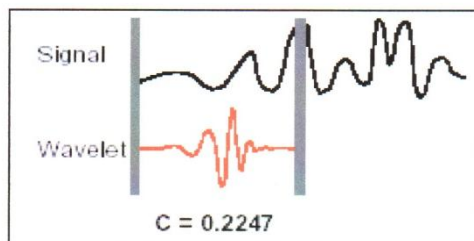


Fig10

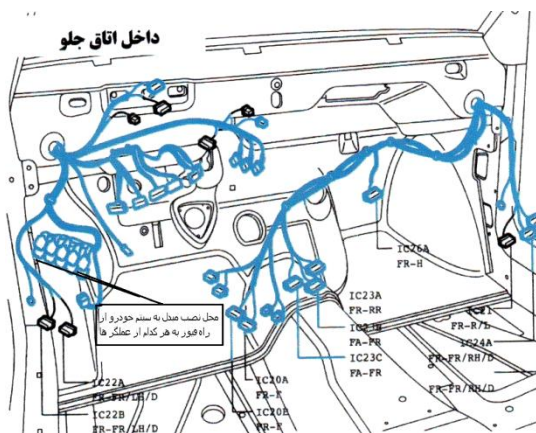


Fig11



Fig12

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