System for Sound Recognition Applied to Musical Instruments Tuning

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Abstract. *More and more, new systems have been developed aiming at providing solutions for any sort of problems. When the subject is music, this is not different. Nowadays, computer programs are frequently used by musicians to assist them in tasks as automatic creation of scores and music composition. In this context, new techniques have been searched to analyse and to process sounds in an efficient way. This work presents a methodology to verify if a musical instrument is in tune. The statistical method of autocorrelation is used to identify the fundamental frequency and consequently the pitch of the sound. Due to some problems presented later, in some cases the relation between frequency and pitch is not completely true. To avoid this problem, special modules were developed, so that specific features of each instrument and the human sensibility could be considered and a better result could be achieved.*

1. Introduction

Musical instruments tuning is one of the greatest problems faced by musicians, especially beginners, since it is a task that demands great sensibility and technical knowledge. Music students frequently feel discouraged to study at home, even when they have the necessary instrument, because they cannot tune it correctly.

Due to relating two different areas, computer science and music, this paper has two different goals. First, for people who study computer science and/or engineering, the objective is to suggest a methodology which can be used to analyse eletronic signals and to identify their frequency in an efficient way, even when noise are present. Second, for musicians and music learners, the goal of this study is to present a computational system to aid people, with some musical notion, to tune their instruments. The diferential of the present study is to consider the distinct characteristics of each instrument, taking also into account the user's knowledge and sensibility.

This paper is organized as follows: introduction; section two, where the autocorrelation method used for analysing the sound is presented; section three, where the model proposed is discussed; and section four, where some considerations about the results are posed.

2. The Process of Musical Instruments Tuning

The process of tuning a musical instrument consists of adjusting its tone pitches on the basis of a tone model. As this property is related to the frequency of the sound wave, in order to tune an instrument, you should adjust the frequency of its sounds according to the frequency defined in some tone model or scale. Thus, the use of an algorithm to identify the frequency of a digital signal is of fundamental importance for any electronic tuning system. Based on this understanding, a statistical method called autocorrelation will be used to achieve the goals proposed to accomplish this paper.

Gujarati, in (GUJARATI 1995), poses that autocorrelation can be defined as the correlation between members of a series of orderly observations in the time or space. According to Heckert in, (HECKERT 2000), the autocorrelation can be used to detect non-randomness in the data and to identify an appropriate time series model if the data are not random.

Taking into consideration that the sounds of most instruments are harmonic, that is, they are waves with defined form and frequency, and based on the second purpose mentioned above, it is possible to determine the number of oscillations the wave completes in one second, and consequently its frequency, by means of the autocorrelation method.

2.1 Calculus of Frequency

In order to calculate the frequency of these sounds, the autocorrelation coefficients of a signal with its delayed version must be computed through the following formula:

$Y(n) = \Sigma^{t}_{k=0} X(k)\, X(k + n).$

This formula adds all the values of each sample multiplied by a delayed version of the signal (AMIR 2000), where X(k) is the amplitude of the signal in the instant k. The variable n represents the amount of time units used to displace the signal and t expresses the size of the sample.

In the first case, the number of displacements (n) is zero, that is, the signal is multiplied by itself resulting on the highest coefficient. In the next ones, the number of displacements (n) is increased and the autocorrelation coefficients are calculated, when they certainly will be smaller than the first one.

As the value of the variable n increases, the signals (original and shifted) become more and more different, resulting on smaller values for the autocorrelation coefficients. This process keeps on going until the displacement (n) is big enough, so that the first cycle of the shifted wave starts to coincide with the second cycle of the original wave (figure 1).

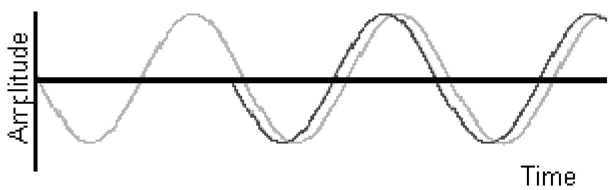

Figure 1 – Shifted wave cycle approaching original wave cycle.

In this case, the values of the autocorrelation coefficients begin to increase, reaching its maximum in the exact instant in which the wave completes one cycle. After calculating a sufficient amount of autocorrelation coefficients, a graph similar to figure 2 is obtained, where the number of displacements made in the signal (n) is shown in the axis x and the autocorrelation coefficients are registered in the axis y.

By measuring the distance between the first point (energy point) and the next peak, it is obtained the exact cycle of the signal. To calculate its frequency, one should divide the file sampling rate (amount of samples per second) by the value of this distance (AMIR, 2000).

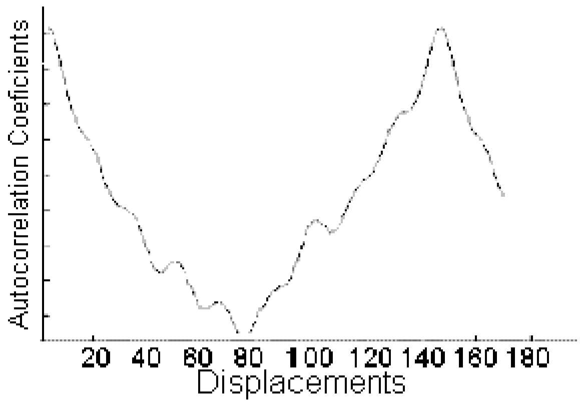

Figure 2 – Autocorrelation graph

3. Proposal of an Electronic Tuner for Musical Instruments

It was already mentioned that tuning an instrument means to adjust its tone pitches and that this property of a sound is closely related to the frequency of the wave. Therefore, generally speaking, the process of tuning a musical instrument consists of adjusting the frequencies of the sounds produced by it. However, some factors make the task of developing an electronic tuner more difficult.

3.1 Problems of an Electronic Tuner for Musical Instruments

The first problem is related to the human psyche. The Greek theory, that pitch and frequency are essentially the same, is attractively simple. However, recent experiments attempting to measure people's sense of pitch have uncovered a number of ways in which this simple idea does not work properly (KIENTZLE 1998). In addition to the factors related to the human mind, the fact that each instrument has its own features makes the development of a generic system more difficult. There are cases, where an instrument sounds a little out of tune, even when it has the frequency of its tones adjusted perfectly to the values of the scale. These small differences, which are easily corrected by humans, may be considered as a great difficulty when this task is done by computers.

Due to these problems, it was decided to create an electronic tuner to analyse the characteristics of each instrument and to allow the user to participate actively of the process, considering his/her knowledge in the search for a better tuning.

3.2 Modelling the System

This system was divided up in two main parts. The first one is the training module, where the user indicates to the system what is the sound of his/her instrument like when it is in tune. By means of this module, the musician or the user of the system, must tune his/her instrument according to his/her preferences, with a specialist's aid if necessary, and record samples of the sound.

The second system module (Tuning) is responsible for recording a sound of an instrument and for indicating to the user whether the instrument is in tune or not. Whenever the user wants, he/she may to indicate which instrument is being tuned and the system is able to use the information previously stored, in the training module, to obtain a better result. In order to help the users who are not able to tune the instrument and to train the system or do not know people who can help them in this task, a default configuration, using the frequencies defined in the chromatic scale was developed. The general structure proposed in this work is organized according to figure 3.

3.3 Describing the Processes

This section will describe each process which is necessary to the system. Initially, the processes that are common for the two modules (Training and Tuning) will be reported.

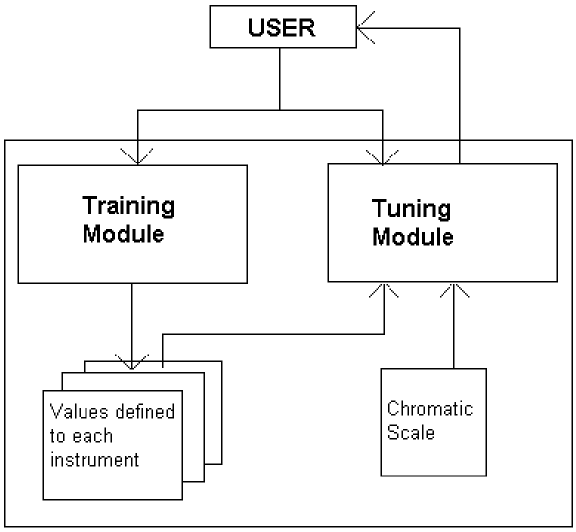

Figure 3 - General architecture of the system

These common processes are: recording and reading of the sound; calculation of the autocorrelation coeficients; noise filtering and searching of the peaks in the autocorrelation graph. After the execution of these processes in the training module, the run process is responsible for saving the values calculated in a file for later comparison in the tuning module. Additionally, in the tunning module, the values calculated are compared to the information coming from a file previously created or from the chromatic scale to verify if the instrument is in tune.

3.3.1 - Sound Recording

The objective of the system is to aid a great number of people who study music, especially the beginner ones. Thus, the possibility of using some specific hardware for recording and/or manipulating the sound was completely discarded . The system just requires a microcomputer with a standard soundcard and a microphone.

3.3.2 Information Reading

Soon after the analogical signals are converted into digital signals and stored in a WAVE file, the process of information reading is initiated. During the recording process, one second of sound is stored, although, just a sample of 1024 bytes (0,023 seconds) is read and analysed.

3.3.3 Implementing the Autocorrelation Method

Through the formula shown in 2.1, a graph is created with the autocorrelation coefficients, aiming at finding the fundamental frequency of the wave. In this application, it was not only used the distance between the first and the second peak, but also the distances between the second, third and fourth peaks, as shown in figure 4.

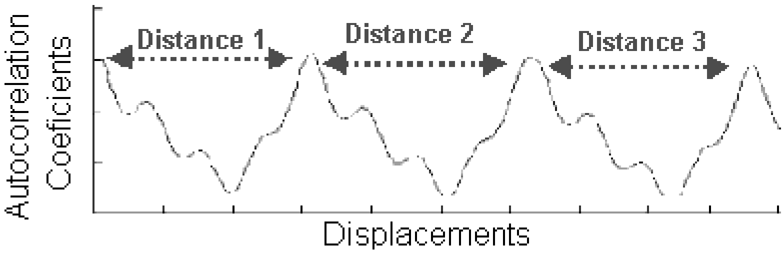

Figure 4 - Measures in an autocorrelation graph with four peaks

3.3.4 Noise Filtering

The option for not using any special equipment or environment for recording the sound may cause a noise interference in the quality of the sound. To avoid this intrusion into the analysis, a method was developed to work with the noise, taking only into account the components coming from the instrument played. This is done after the generation of the autocorrelation graph.

One of the main features of the electronic tuning is that you know where the analysis of the signal should be done, because the system has a previous knowledge about the tone played. Based on this feature, a method was developed to analyse the graph only in places where the signal probably completes a cycle, because even if the sound is below or above the ideal value, the frequency of the wave will be in an interval around its correct value.

3.3.5 Searching for the Results

Figure 4 showed a graph with three peaks where each of them indicates that one cycle was finished by the wave. These peaks are used to conclude if the instrument is in tune. In addition to the three maximum points, three more values are used in the final calculation. They are the neighbours points of the three peaks. Thus, six values are considered to decide if the tone of that instrument is in tune.

3.3.6 Training Module

So far, all the calculations have been done in the same way, independently of the module that was being run. In the training module the values calculated in the previous steps are stored in a file, so that, later on, they can be used by the user to verify if his/her instrument is in tune.

3.3.7 - Tuning Module

In the tuning module the values calculated are compared with other values (base values), to verify whether the instrument is in tune or not. The values (base values) used for the comparison can come from two different places: from a file of values previously created by the user, by means of the training module, or from the chromatic scale.

In order to check if the instrument is in tune or not, the first step is to calculate the difference between each point of the calculated values and its correspondent base values obtaining then, six values that represent the difference between the sound in tune and the sound played. Next, these distances are used in the following formula:

$$V = d1 * □1 + d2 * □2 + d3 * □3 + d4 * □4 + d5 * □5 + d6 * □6$$

This formula multiplies each distance by a coefficient, and based on its result the system verifies if the tone played is in tune. The coefficients n (n = 1..6) represent the weights applied to each of the differences between the sound played and the correct sound, to calculate the value V. The values attributed to these coefficients in the developed prototype were the following:

□1 = 10 □2 = 2 □3 = 4 □4 = 2 □5 = 2 □6 = 1

If the value V varies from an acceptable interval (0 to 5 for the developed application), the played tone is considered in tune. On the other hand, if the value V does not fit the above variation, the system shows to the user how much his/her instrument is out of tune, through a scale. Based on these information, the user can adjust his/her instrument and repeat the process to verify if it is now in tune, or if it still needs some adjustments. Figure 5 presents the result after analysing a tone that is considerably below its correct pitch.

Figure 5 - Result after the execution of a tone below its correct pitch

4. Conclusions

Many kinds of electronic tuners can be easily found in music stores. However, most of them require specialized equipment. The system proposed in this paper works in any computer, by allowing that a great number of people can use it without the necessity of any special hardware

Another important feature of the system proposed in this paper is the personalized configuration, where the characteristics of each instrument and the human sensibility can be considered, aiming at a better result.

The results of this study suggest that its main goal, that was to show that specifc characteristics of each instrument and the human sensibility can be considered to obtain better results when the tuning process is done by computers, was achieved.

5. References

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