APPLICATION FOR THE TIMBER ANALYSIS OF HARMONIC MUSICAL INSTRUMENTS

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ABSTRACT.

Timbre is one of the most important perceptual properties of sound, along with pitch and loudness. However, unlike pitch and loudness, which are primarily related to a single physical parameter, timbre is a multidimensional attribute that results from complex acoustic properties. Our ability to perceive such properties of sounds allows us to distinguish which instrument produced the sound in a piece of music. In this work, we present a Matlab application for timbre analysis of harmonic musical instruments that considers both steady-state and transient properties for educational purposes. The algorithm identifies the principal components of the spectrum and includes corrections to ensure the robustness of the estimation of other sound parameters. It also includes representations and useful tools for the timbre analysis of musical sounds. The possibilities of its use in teaching a subject in the field of musical acoustics and, in particular, in the practice of the Master's Degree in Acoustical Engineering of the Universitat Politècnica de València, Gandia Campus, are highlighted.

RESUMEN.

El timbre es un atributo perceptivo principal del sonido, junto con el tono y el volumen. Sin embargo, a diferencia del tono y la sonoridad que se relacionan principalmente con un solo parámetro físico (es decir, la frecuencia y la intensidad del sonido), el timbre es un atributo multidimensional que surge de propiedades acústicas complejas. Nuestra capacidad de percibir tales cualidades de los sonidos nos permite discriminar en una pieza musical qué instrumento ha emitido el sonido. En este trabajo se presenta una aplicación de uso docente implementada en Matlab para el análisis tímbrico de instrumentos musicales armónicos, tanto las propiedades estacionarias como las transitorias. El algoritmo identifica las componentes principales del espectro e incluye correcciones para la asegurar la robustez en la estimación de parámetros tímbricos. Asimismo, incorpora representaciones y herramientas útiles para el análisis tímbrico de sonidos musicales. Se destacan las posibilidades de su uso docente en asignatura en el campo de la acústica musical y, en particular, para las prácticas en la asignatura del Máster de Ingeniería Acústica de la Universidad Politécnica de Valencia en el Campus de Gandia.

1. INTRODUCCIÓN

The measurement of acoustic and vibration phenomena associated with musical instruments is complex. Unlike electroacoustic or other electrical devices where the source can be connected to the power grid, musical instruments must be played by humans or automatic devices [joel, artificial mouth, string] to analyze their physical behavior. This makes it difficult to achieve the repeatability and reproducibility necessary to ensure the trustworthiness of the experiments and analyzes. This is a particular challenge in an educational setting where multiple laboratory workstations are typically set up. Laboratory experiments create a lively learning atmosphere and are an interesting element of acoustics instruction for many students.

Acoustic experiments can now be carried out more conveniently thanks to recent technology advancements, lower pricing for acoustic sensors, and the wide availability of audio apps. In the last decade, numerous works have been developed on the use of smartphones for acoustics [1-2], ultrasound measurements [3], and visualization of acoustic phenomena with applications based on numerical methods [4], the Psychoacoustics toolbox [5] for testing auditory thresholds and an application to visualize vibration in a violin [6].

There are many acoustical parameters for rooms. Their purpose is to describe subjective sound impressions in an objective manner. Similarly, parameters related to timbre characterize the specific sound of a musical instrument. Same as most of the room acoustics parameters can be determined from the room impulse response, the parameters related to timbre rely on the signal and their analysis involves signal processing. Like this, timbre-related characteristics define the distinctive sound of an instrument. The parameters associated with timbre rely on the signal, and their study entails signal processing, just as most room acoustics parameters may be found from the room impulse response. Students of musical acoustics frequently come from a wide range of backgrounds, including musical studies, audio, physics, mathematics, or various engineering fields. Not all of them have sufficient knowledge of signal processing for the analysis of musical sounds. However, the course includes laboratory skills and concepts that call for comprehension of crucial musical signal processing phenomena.

The timbre is a complex feature of sound, and it is generally described as a multidimensional property [7]. Therefore, it is crucial to have a teaching tool for sound signal analysis available to students in the musical acoustics lab. The Timbre Toolbox is a useful tool for the timbre analysis [8]. It has a large number of audio descriptors derived from various representations to obtain the temporal, spectral and spectro-temporal characteristics of sound events. It is a multidimensional tool for measuring the acoustic structure of complex sound signals, which is very useful for perceptual and music research. However, it is too complex for its use in an educational context. The user interface provides too many options for the potential student user, and a high level of post-processing and analysis of musical sounds leaves no room for the student to learn. This work presents a computer application for the tonal analysis of harmonic musical instruments. This is a tool, Timbrapp, to be used in the context of subjects related to musical acoustics and should be used by teachers and students in engineering, science, or music studies. It may also be useful for musical instrument makers and luthiers, as some of the specific tools may complement their expertise.

2. TIMBRAPP

2.1. User interface

The user interface has been developed using Matlab's APP designer. It allows presenting the application in a friendly, intuitive, and interactive environment. It includes the graphic representation of the analyzed sound, both in its temporal waveform and its frequency spectrum, in the same way that it represents the time-frequency variation in two ways: through a visualization of the time-frequency transformation of the complete signal, and by means of the representation of the temporal spectrum calculated in short spaces of time. This is useful, for example, to know the instantaneous frequency information while the sound is reproduced. All these representations are adjustable by the user, being able to configure all the parameters that intervene in the computation for the calculation and visualization of the information that can be represented through the configuration option. The user can choose between linear or logarithmic representation in both axes in the frequency representations.

The main window shows contextual information of the analyzed sound that may be relevant for its interpretation, such as its fundamental frequency, the musical note (and its MIDI code) with which it corresponds or the deviation from the nominal frequency of the sound. It is also possible to represent the time envelope of the signal, entering the configuration options, opening the system explorer to select an audio file or calculate and represent the frequency response of an auto-regressive (AR) model that fits the spectral envelope of the signal. Finally, there are two advanced level tools, such as exporting the harmonic and ASDR model to a similar interface that allows the user to synthesize the sound and generate all the results of calculations for the descriptors of the audio signal that we select as active from the configuration options.

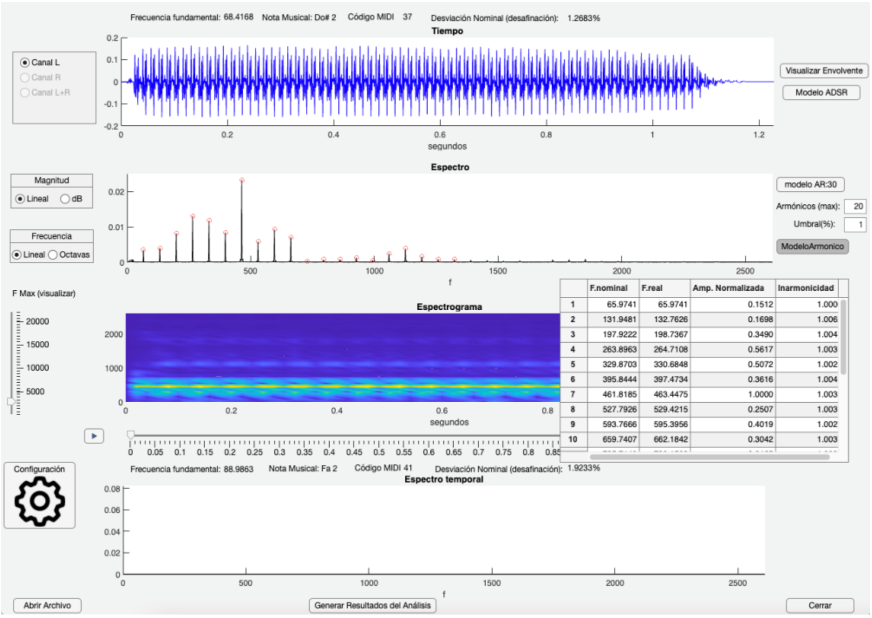

Figura 1 – Interface of the TimbrApp including the time signal, the spectrum, the spectrogram and the output of the Harmonic Model.

3. MODEL

The tool calculates and plots the temporal waveforms of the signal being studied, along with its Fourier transform and other helpful charts, before loading a sound. These data serve as a basis for obtaining other parameters and representations of the signal, such as the harmonic model, the vibrato model, the spectral envelope and the transient features of sound.

3.1. Harmonic model

It is obtained by jointly processing the two previously calculated representations of the signal. It is calculated in several steps:

1. An estimate of the pitch or fundamental frequency of the signal is obtained. For this purpose, the PEFAC algorithm [9] is used, considering a valid frequency range for the pitch between 30 Hz and 4 kHz (covering the typical bandwidth of common musical instruments). A temporal windowing of the waveform of at most 80 ms without overlap is used, and the pitch estimate is obtained in each temporal window. The initial pitch estimate is taken as a statistical mode of the values of all frames, considering only the

frames whose energy is greater than a certain threshold. It is checked to see how many values are close to the modal value, f_{0m}, its half value, or its double value, and the most likely pitch, f_0, is chosen from these three candidates. This is done because the modal value f_{0m} does not necessarily have to be the pitch, because the initial estimated values can be unique, and because these estimated pitch values could correspond to twice or half the frequency due to the inadequacy of the method used.

2. The calculation of the pitch value is refined by searching for local maxima (peaks) in the information of the previously calculated Fourier transform. The Matlab function "findpeaks" is used for this, along with additional parameters for the minimum distance between peaks and the minimum value for a maximum that qualifies as a peak. By choosing the peak closest to f_0, the peak with the highest energy, the peak that most closely resembles the distance between the peaks, the peak that repeats the most, and the peak that most closely resembles the middle of the distance between the peaks, we can improve the choice of pitch. The candidate that repeats the most is picked.

3. Once a reliable value for the pitch is obtained, the theoretical values of the different frequencies of this pitch (theoretical harmonics) are calculated and the values of the peaks closest to these theoretical frequencies and their deviation from these values are obtained. In this way, the harmonic model is obtained.

3.2. Vibrato model

A straightforward vibrato model is also derived in addition to the harmonic model, taking into account that vibrato can be characterized as frequency modulation. The frequencies of other peaks around each harmonic are then examined. In order to estimate the frequency of variation of vibrato, a fit with Bessel functions is utilized until the one with the least error is reached.

3.3. Spectral envelope

Another important spectral information is the spectral envelope. Although there are many methods to obtain this parameter, one of the most used is to create an autoregressive (AR) filter model of a particular order. This model characterizes the sound source, the instrument emitting the sound. The tool allows to obtain a model of order 30 AR of the analyzed signal, although the order can be changed by the user. For this purpose, the linear prediction filter of the desired order is obtained, and the inverse filter is proposed as AR model.

3.4. Attack-Decay-Sustain-Realese (ADSR) model

As for the temporal parameters, this tool automates the calculation of the ADSR model, although it also allows manual modification or definition by the user based on the visualization of the waveform of the signal to be analyzed. To automate this calculation, we start from the estimation of the temporal envelope of the waveform of the analyzed sound. The envelope can be estimated by different methods, such as moving average filtering (FIR filtering), exponential average filtering (IIR filtering), use of Hilbert transform or interpolation between peaks. The user can choose the method and configure it accordingly. We start with a silence threshold to obtain the beginning of the attack period. That is, the instant time when the envelope crosses the threshold. We get the instant time when the maximum of the first transient is found. This is the end of the attack period and beginning of the decay period. Although in most cases this coincides with the global maximum, for certain musical sounds (e.g. string instruments) the local maximum may occur well after the end of the attack time. The end of the decay period is the beginning of the sustain period. It is estimated as the time at which the energy of the envelope stops decaying or decays at a lower rate than in previous frames. This energy value is used as a reference value for determining the end of the sustain period. Finally, the end of the sustain period (and the beginning of the release) is estimated as the time when the energy at the beginning of the stationary period has dropped by 3 dB compared to the energy calculated at the beginning of the stationary period. The end of the recession period is the time when the signal falls below the silence threshold again.

4. STUDENTS TESTING THE APPLICATION

In an initial session, Timbrapp was introduced to two groups of students from the subject Musical Acoustics in the Master of Engineering Acoustics program at the Universitat Politècnica de València. During the course, 3 laboratory sessions are developed using the application. The objectives of each session are presented in table 1. Each practical laboratory session includes a report with a theoretical introduction and the activities to be performed by the students. The development of the exercise includes an assessment of prior knowledge, a brief explanation by the instructor, the performance of measurements and their analysis. In the following, a summary of the contents of each session is presented:

4.1. Session 1. Tuning systems. Measure in cents.

The student is introduced to the application during the first session. It is suggested that this be accomplished by obtaining the fundamental frequency and harmonics of pre-recorded sounds. It is also suggested that the student note the frequencies of four notes in two fundamental tuning systems (Pythagorean and equal temperament) and compare them to the theoretically computed values to determine how far off tune each note is. The knowledge of fundamental musical acoustics principles, such as the usage of acoustic intervals, logarithms, and the cent unit, is also required for this exercise. Finally, the student is then instructed to determine the inharmonicity of the various harmonics in the notes and to express it in various units.

4.2. Session 2. Accuracy and precision of musical acoustic measurements.

Due to emission and recording conditions, acoustic measurements using musical instruments exhibit significant fluctuation. As a result, it's crucial to focus on the principles of measurement repeatability and reproducibility in the lab. In this lesson, it is suggested that the student repeatedly record the amplitudes and frequencies of several sounds' first harmonics. The student calculates the brightness from the mean value and its accompanying error using error propagation. It must assess how the error spreads in a mathematical expression including measured variables. It also focuses on how to express the measured magnitude and the error correctly.

4.3. Session 3. Spectral and transient parameters of timbre.

This session concludes the investigation, measurements, processing, and analysis of the timbre magnitudes. A thorough examination of the timbre of various musical instruments must be conducted by the student while applying the practical information they have gained from prior laboratory sessions. The analysis of timbre evolution over time (tristimulus) and comparison of the outcomes with the use of the Timbre Toolbox [Peeters2011] are suggested as more advanced exercises.

Tabla 1 – Objectives of the Laboratory sessions

Laboratory session	Objetives
Session 1 Tuning systems. Measure in cents.	- Distinguish the spectral components of a sound and measure the frequencies of the fundamental frequency and harmonics. - Know and identify the measurement systems of the main scales in the standard

	nomenclature. - Know and understand Pythagorean tuning and the equal temperament. - Be able to calculate acoustic intervals in cents and convert them to frequencies.
Session 2 Accuracy and precision of musical acoustic measurements	- Measure tones with the experimental instrument and calculate the intensity and Natural frequencies of the harmonics and fundamental. - Know the quantities and unities to characterize the accuracy of an experimental measurement. - Understand the importance of normalized quantities and harmonic content of a note. - Know how to estimate the error of physical quantities and properly an experimental result.
Session 3 Spectral and transient parameters of timbre	- Knowledge of the most important parameters for characterizing of the timbre in the steady-state regime. - Compare the timbral characteristics of two different instruments. - Analyze the timbre variation of an instrument in different registers and dynamics

3. CONCLUSIONES

Timbre is a complex property of great relevance for the description of the sound characteristics of musical instruments. An application for educational use is proposed here that includes the implementation of models for the processing of harmonic musical sounds. Through its use, students have the tools to obtain and analyze timbral parameters of musical sounds. Through three work sessions on various basic concepts in musical acoustics in the acoustics laboratory, students acquire the skills in the matter.

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