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AcMus: Computational Tools for Measurement, Analysis and Simulation of Room Acoustics

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RESUMO: Em 2002 formamos o AcMus, um grupo de estudos voltado para questões de acústica de salas. Atualmente desenvolvemos um projeto centralizado na criação de um *software* para cálculo, medição, análise e simulação de salas de pequeno e médio porte destinadas à execução musical. Neste artigo descrevemos brevemente os objetivos do projeto e os resultados obtidos até o momento. Inicialmente, apresentamos os sistemas de medição baseados em MLS e varredura logarítimica desenvolvidos. Em seguida apresentamos algumas ferramentas computacionais de otimização que podem ser utilizadas no projeto de salas ou na correção do comportamento acústico das mesmas. Finalmente, apresentamos a última etapa a ser desenvolvida no projeto que é a de simulação acústica de ambientes. Todos os módulos do programa serão integrados de modo a permitir o fácil intercâmbio entre diferentes tipos de dados. Durante a implementação do projeto tivemos uma grande preocupação com a portabilidade e acessibilidade do *software*. Por esse motivo, o mesmo será desenvolvido com código aberto, permitindo o livre acesso na Web e a colaboração de outros pesquisadores interessados, tanto como usuários, quanto como desenvolvedores do programa. Buscamos também que o sistema seja multiplataforma, o que deverá ser alcançado por meio de sua implementação em Java.

ABSTRACT: The AcMus research group was created in 2002. This group is devoted to issues related to room acoustics. Currently, we develop a project whose general goal is the development of a software for estimation, measurement, analysis and simulation of small and medium sized rooms specially designed for musical performance. In this article, we present the specific goals of this project and the results obtained thus far. First, we introduce the measurement systems based on sine sweep and MLS under development. Second, we present some computational optimization tools that can be used for room design or for the correction of the acoustic behavior of existing rooms. Finally, we describe the last stage of our project, namely the development of computational tools for the acoustic simulation of rooms. The system under development is organized in modules. All modules shall be integrated in a way as to allow seamless flow of data, regardless of the variety of data types and structures involved in the system. Portability and accessibility of the software developed has been a major concern in the implementation of the system, and for this reason the system has been developed as an open source project, so that users and system designers can collaborate with its further improvement. The system core is being implemented in Java, with great care to make it platform neutral.

1. INTRODUCTION

In the last few years the research in the field of acoustics has experienced an expressive growth in Brazil. However, the establishment of research groups remains irregular and mainly concerned with issues related to environmental comfort and vibration and noise control. The application of acoustic research into musical matters is not very common among academic circles and its realization is usually due to individual efforts. Nevertheless there is an increasing concern about the acoustic qualities offered by spaces designed for music production and diffusion such as recording studios, movie theaters and lecture halls. Many companies and professionals are becoming specialized in room acoustics, but usually their activity is based on the use of standard solutions which are not necessarily well adapted to our demands.

In 2001 we created a research group in acoustics at the University of São Paulo with the aim of gathering researchers interested in the connection between music and acoustics. One year later we decide to formalize a project on room acoustics called AcMus in collaboration with individuals from different fields, such as music, architecture, engineering, physics and computer science. Our main purpose is to start a regular work on musical acoustics and to join the efforts of many individuals with similar interests.

The AcMus project is guided towards the analysis, simulation, and optimization of acoustic parameters for small and medium sized music rooms such as recording studios, small concert halls, home theaters and so on. The core of the project is the implementation of a software based on modular functions related to the analysis of the acoustic parameters [1].

The development of the software has been guided by some specific concerns. First, since we intend to promote the exchange of the results of our research with other groups, the software is based on open source codes. In the same direction we are trying to use, whenever possible, source codes and algorithms that are already made available by other research groups. Second, the system must not be limited to a single platform. For this reason, the software has been developed mostly in Java so that it can run easily on Linux, MacOS and Windows platforms. If the need arises, specific computationally-intensive functions can be implemented in C++ for performance reasons.

2. THE ACMUS SOFTWARE

The core of the AcMus project is the development of a modular software which allows the integration of different functions for acoustic simulation, measurement and optimization. The software consists of three modules integrated by the same interface: 1) Measurement Module, 2) Utilities Module; 3) Simulation and Optimization Module. Together, they will help in the design, analysis, and treatment of rooms dedicated to music production and diffusion.

2.1 Mesurement Module

A large number of parameters related to the room acoustic behavior can be obtained from its impulse response (IR), that is, the room's response to a very short impulse whose energy is distributed throughout the frequency spectrum (ideally an impulse described by a Dirac's delta function) [2, 3]. Once the IR is obtained, a series of parameters can be calculated by using well known formulas and expressions, such as:

- **Reverberation Time (RT60, RT30 e RT20):** obtainable from the impulse response, it consists in one of the most important factors for determining the acoustic quality of a room. It is related to the time that a specific sound takes to decrease 60dB after the sound source stops [2, 4, 5]. RT 30 and RT20 are used for situations in which the low signal-to-noise ratio disturbs the direct evaluation of the RT60. The results can also be filtered to indicate the reverberation time for different frequency bands.
- **Early Decay Time (EDT):** represents the decay time for the first 10dB of the reverberation curve;
- **Sound Strength (G):** the logarithmic ratio between the sound pressure of the measured IR to that of a response measured at a distance of 10 meters from the same source in a free field;
- **Clarity (C_t):** the logarithmic ratio between the energy level at the first 50ms (for speech) or 80ms (for music) and the rest of the IR. This parameter is related to the balance between reverberation and the clarity of the received sounds;
- **Definition (D_t):** the ratio between the energy received during the first 50ms and the whole energy of the IR.
- **Center Time (Ts):** corresponds to the gravitational center of the IR (in ms). A small Ts means that energy is concentrated at the beginning of the response which helps clarity while a large Ts means that the energy is distributed over time giving the impression of a more reverberant space.
- **Interaural Crosscorelation Coefficient (IACC):** maximum value for the correlation between the signals arriving at the left and right ears; expressed in values ranging from -1 (arriving signals equal in magnitude but exactly out of phase) to 0 (arriving signals have no similarity) to +1 (identical arriving signals, i.e. same amplitude & phase).
- **Initial Time-Gap Delay (ITGD):** time difference between the direct sound and the first reflection. The feeling of *intimacy* is quantitatively measured by the ITGD.

Initially we have implemented two methods for acquiring IR. One of them is the MLS (Maximum-Length Sequence) method [6, 14], which uses a signal based on a periodic sequence of randomly distributed positive- and negative-going impulses with a flat energy distribution in the frequency domain. The MLS is largely used in many acoustic measurement systems. The other method, which we call LSF (Log Sweep FFT) [7, 8], is a more recent development and can be understood as an evolution of the TDS method [9] and uses a logarithmic sweep as a excitation signal. These methods offer a much better signal-to-noise ratio in comparison to other methods that use mechanically produced impulses. Besides, numerical methods like MLS, TDS and LSF allow for the repeatability of tests since they work with an stable and previously known signal.
Although the MLS has been implemented in a number of systems, our tests indicate that LSF has some advantages, specially in terms of signal-to-noise ratio. Since our intention is to build a low cost system which may be subject to non-linear responses and critical signal-to-noise ratio, the use of LSF seems to offer better results.

The main advantage of LSF in relation to the TDS method is that the former uses a logarithmic sine sweep while the later uses a linear one. The spectrum of a linear sweep is almost white which usually leads to a low signal-to-noise ratio in the lower frequencies since the signal remains most of the time in the high-frequency region. On the other hand, the logarithmic sweep has an almost pink spectrum showing the same energy per octave due to a better energy distribution.
The system we used so far for the preliminary tests with the software consists of a PC computer (Pentium 4, 1.6 GHz, 120 MB RAM), an M-Audio Duo audio interface, a Berhinger ominidirectional microphone and self-powered Electro-Voice SxA100 loudspeakers. In comparison to the measurements performed by other measuring systems our software has given very consistent and stable results.

2.2 Utilities Module

The Utilities Module is an acoustic tool box which offers functions that can be useful for the design of rooms, acoustic measurement and audio processing. Some of these tools have been already implemented while other are in process of implementation or improvement. For example, this module calculates the reverberation time and modal frequencies from the description of room's dimensions and surfaces (Figs. 1 and 2).

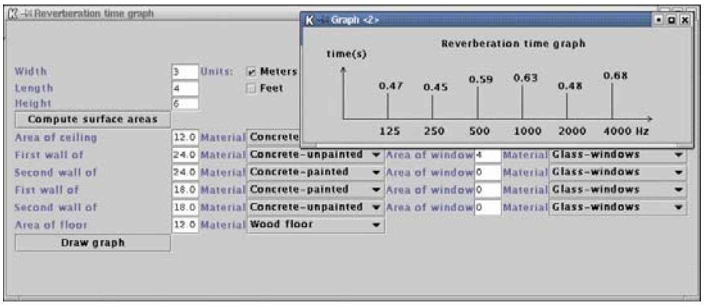

Figure 1 – *Reverberation Time - RT60*

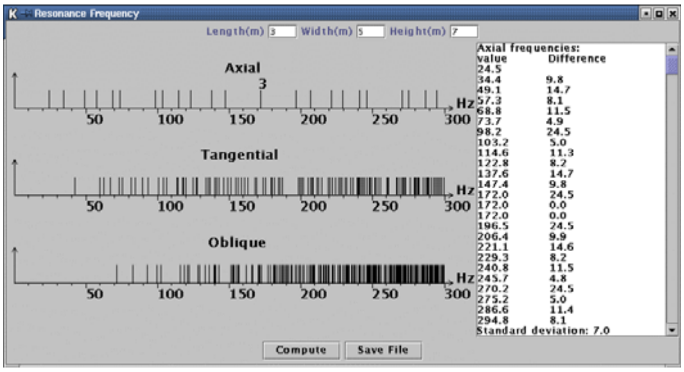

Figure 2 - *Modal Response*

Other functions being implemented include the generation of different types of audio signal, filtering and spectral analysis. Finally, this module will offer function to help in the design of Schröder diffusors [10, 11, 12, 13] and low frequency absorbers.

2.3 Simulation and Optimazation Module

The acoustic design of a listening room involves a large number of variables and constraints besides several alternative criteria with respect to acoustical design parameters; optimization techniques are an interesting tool since they allow the automatic search of solutions which at the same time satisfy all constraints and can be shown to be the best ones available for a set of given criteria. In this part of the work we consider acoustic models that rely on computer simulations based either on wave equation solving or geometrical methods.

To simplify discussion, a cuboid room is assumed, though much of the material may be adapted to polyhedral rooms. The main design variables considered here are the room dimensions and reflection coefficients of the surfaces, which must obey a given set of constraints. The main goal is to obtain a nearly flat frequency response over the range of possible source and listener positions. Other design criteria may be expressed analogously and are mentioned at the end of the section.

Let $r(s,l,d)$ be the approximate response factor as function of the source (s) and listener (l) positions and a description (d) of the room (comprising room dimensions and reflection coefficients). This function might be represented by a vector indexed by a finite set of frequencies $\Im$. Let $\bar{r}(s,l,d)$ be the least-squares approximation of $r(s,l,d)$, i.e. the result of the linear regression problem applied to the values of the vector $r(s,l,d)$. Let $e(s,l,d)$ be an error measurement function for the deviation of the response $r(s,l,d)$ from the affine response $\bar{r}(s,l,d)$. Natural choices for this function are $e_1(s,l,d) = {}_{f \in \Im} |r_f(s,l,d) - \bar{r}_f(s,l,d)|$, the quadratic residue $e_2(s,l,d) = {}_{f \in \Im} (r_f(s,l,d) - \bar{r}_f(s,l,d))^2$ and $e_\infty(s,l,d) = \max_{f \in \Im} |r_f(s,l,d) - \bar{r}_f(s,l,d)|$.

Suppose the range of possible source positions is described by a set S (stage) and the range of possible listener positions is described by a set A (audience). Let D describe the constraints on the description of the room. This description will typically involve box constraints (i.e. maximal and minimal values) on the dimensions, and a finite set of possible values for reflection coefficients (as related to available materials). Each possible pair of source and listener positions determine a distinct frequency response. One approach for the optimization model is to ask that the frequency response be as flat as possible in the average case. This corresponds to

$$\begin{cases} \text{min} & \int_{s \in S} \int_{l \in A} e(s,l,d) \\ \text{subject to} & d \in D. \end{cases}$$

The integrals on the problem above might be computed approximately, i.e. as sums over a finite number of pairs (s,l). Another possibility is trying to guarantee that the worst possible combination of source and listener positions will have the best possible frequency response with respect to the chosen measure of deviation from flatness. This corresponds to

$$\begin{cases} \text{min} & \sup_{s \in S, l \in A} e(s,l,d) \\ \text{subject to} & d \in D. \end{cases}$$

These problems are very difficult ones due to several reasons: lack of a closed formula for the objective function (it depends on simulations); no convexity-like properties in the objective function; mixing of continuous and discrete variables leading to a combinatorial behavior. The solution to this problem will involve the derivation and implementation of a global optimization heuristic method tailored to the structure of the problem, with techniques like those in [15]. The next steps in this direction correspond to the study of the mathematical structure of the functions $r(s,l,d)$ and $e(s,l,d)$ and their relations to the Stage and Audience sets, in order to minimize the computer time involved in the calculations of the objective function, as well as efficient ways of improving the design variable d.
To obtain a frequency response as flat as possible is one of many possible design criteria; for instance, one could ask for a room in which frequency responses approximate as closely as possible a given ideal frequency response, not necessarily flat; instead, one could ask that reverberation times, expressed as a function of a finite number of frequency bands, match as closely as possible a previously established reverberation curve. All these criteria correspond to minimizing the deviation of a function obtained by simulation from a given ideal function, thus defining analogous optimization problems [16].

3. STAGE OF DEVELOPMENT

The AcMus Project began in 2002 and is supposed to be completed by the end of 2005. The software is been developed as the research work advances. Our intention is to use Java as much as possible to develop the software making it runnable on multiple platforms. The use of Java brings a few challenges regarding the computer's performance for functions which demand a high processing power, but we are confident that we will come up with satisfactory solutions.
The Measurement Module has been implemented as a Matlab prototype and has been successfully tested during acoustic measurement of different rooms at University of São Paulo. Currently we are working on its implementation in Java and on the creation of a user-friendly interface. This stage should be completed by the end of 2004.
Many tools that are part of the Utilities Module have been implemented as independent Java applications and are already available. The next step is to group these Java classes in a more general application so that they can share data.
The Simulation and Optimization Module is the most complex and is in its initial stage of development. Two well known acoustic simulation methods are being implemented: ray tracing and image source, both based on geometrical acoustics. These two methods will be used in association. The software will provide the impulse response for the simulated rooms as well as the acoustic parameters calculated from this response. Also, we intend to build a module that convolves binaural impulse responses with anechoic audio for auralization.

4. CONCLUSIONS AND FURTHER RESEARCH

AcMus is an ongoing project whose main goal is the consolidation of a research group interested in the relation between acoustics and music. Currently the work is guided toward the creation of a computer software able to measure, analyze and simulate the acoustic behavior of rooms. Part of the software has already been implemented as a prototype. The current version is been tested under different situations and we are obtaining good results in comparison to acoustic measurements performed with other systems.

Since one of the major concerns of our research relies upon the use and development of open source algorithms, the partial results of the AcMus project can be accessed on the Web at the following address: http://gsd.ime.usp.br/acmus

Other related works are under way at our research group. One of them is an investigation on how to correlate objective and subjective acoustic parameters [4, 17] by analyzing the acoustics of some concert halls in São Paulo in relation to a survey applied among professional musicians who are used to play in those halls. Another one is the re-design of a room at the University of São Paulo based on collected data and analysis via the AcMus software.

5. AKNOWLEDGEMENTS

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