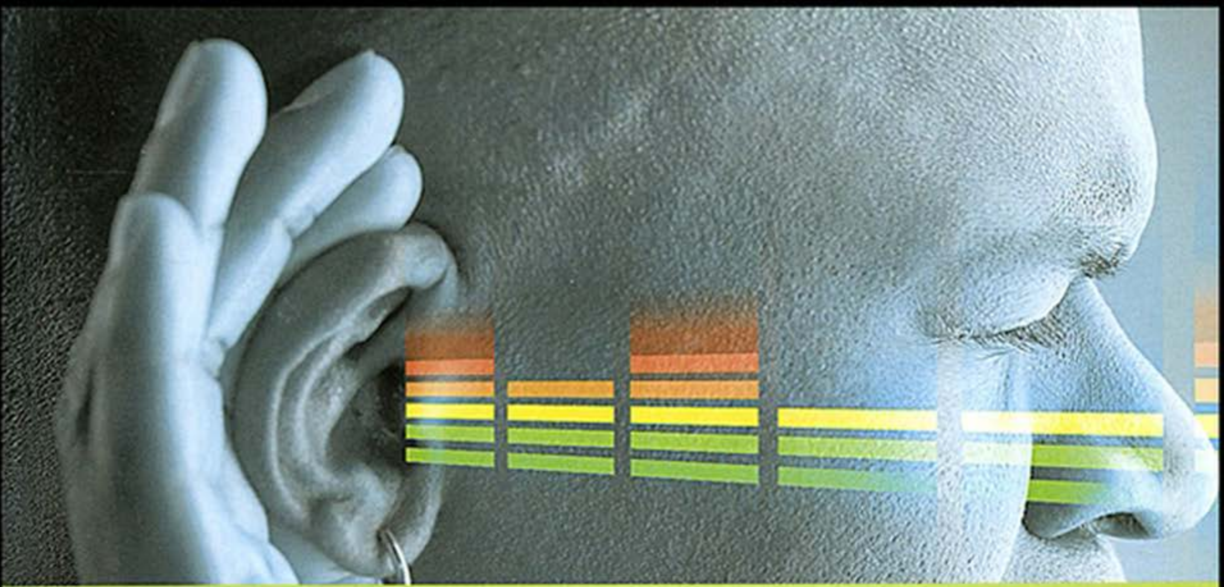

Music and Acoustics

From Instrument to Computer

Philippe Guillaume

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To Nicole, Cécile and Olivier

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Foreword

How does a tuner achieve such a precise tuning of a piano or an organ using nothing but his ears? Why does the clarinet, though equal in length to the C flute, play one octave lower? What difference is there between the Pythagorean scale and the tempered scale? How can a series of notes seem to rise indefinitely even though it always repeats the same notes? What are the possibilities offered by digital sound? What are its limitations? How can a compression technology such as MP3 achieve a tenfold reduction of a sound file's size without significantly altering it? What is the very simple principle underlying audio synthesis in Yamaha's famous keyboard, the DX7? These are a few examples of the questions we will try to answer.

The goal of this book is to use these questions to give the reader an overview of the nature of musical sound, from its production by traditional musical instruments to sounds obtained by audio synthesis, without trying to be exhaustive however: this book is not meant as a catalogue, but instead, I hope, as a first step that will enable the reader to move on to more specific areas in this field. Musical sound is addressed from a scientific standpoint, and the succession of causes that lead to a specific type of sound are, as much as possible, described in a simplified but precise manner. The fact, for example, that a particular sound is composed of harmonics (strings, pipes, etc.) or of partials (bells, timpani, etc.) finds its causes in the physical laws that govern the behavior of materials, laws that induce mathematical equations, the nature of which leads to a certain characteristic of the produced sound.

This book is intended for any reader interested in sound and music, and with a basic scientific background: students, teachers, researchers, people who work in a scientific or technical field. It describes and relies on concepts of acoustics, mathematics, psychoacoustics, computer science and signal processing, but only to the extent that this is useful in describing the subject. In order to broaden its reach, it was written in such a way that the reader may understand sound phenomena with simple analytical tools and the smallest possible amount of required knowledge. Those who teach this material will find diverse and motivating study problems, and students will find ideas for different kinds of ‘projects’ they may encounter in their undergraduate and graduate studies. In the end, my greatest wish would be to succeed in sharing with the reader the pleasure I find in understanding the basic mechanisms underlying the manifestation and the perception of the sound and music phenomenon.

After an introduction to acoustics, a bit of music theory, and a study of sounds and their representation in Chapter 1, we will discuss vibrational modes and the timbre of a few typical instruments in Chapter 2, and in Chapter 3, we will relate this with the question of scales and tuning systems. After wandering off into psychoacoustics in Chapter 4, and using the opportunity to discover a beautiful acoustic illusion, we will discuss several aspects of digital sound in Chapters 5 and 6: sampling, compression technology based on the properties of hearing (such as the widely known MP3 format), sound effects (vibrato, reverberation, the Leslie effect) and synthesized sounds, such as for example those produced using the Chowning technique, made popular by DX7 synthesizers.

For further development, each chapter ends with the following:

– study problems, to explore certain themes, or to study them further in depth. For the reader’s information, the difficulty and the amount of work required are indicated with stars: (*) means easy, (**) is average and (***) is difficult;

– practical applications meant to be carried out on a computer, where the reader will create different kinds of sounds and play them on a crude synthesizer, experimenting on the phenomena described in the book, as well as put his or her hearing to the test, and practice his or her scales! Practical instructions relevant to these applications are given at the end of the first chapter.

Website. A website is available to illustrate the book. It contains many examples of sounds, as well as the programs used to generate them. It also contains the programs and sound files necessary to perform the practical applications, along with the answers. The address of the website is:

`www-gmm.insa-toulouse.fr/~guillaum/AM/`

Throughout the book, it will be referred to simply as the `AM` website.

Reading advice. The chapters were written in a particular, logical order, and the concept and methods developed in a given chapter are assumed to be understood in the chapters that follow. For example, the approach used to go from the wave equation to the Helmholtz equation, which is detailed in Chapter 1, will not be explained again when studying the vibrations of sonorous bodies in Chapter 2. However, you can also browse through it in any other order, referring if necessary to the previous chapters, and using the cross-references and the index to easily find where a given concept was discussed. Finally, because some phenomena are easier heard than explained, listening to the website's audio examples should shed light on any areas that may still be unclear!

Philippe GUILLAUME

Chapter 1

Sounds

Sound and air are closely related: it is common knowledge that the Moonians (the inhabitants of the Moon) have no ears! This means we will begin our study of sound with the physics of its traveling medium: air. Sounds that propagate through our atmosphere consist of a variation of the air's pressure $p(x, y, z, t)$ according to position in space x, y and z and to the time t. It is these variations in pressure that our ears can perceive. In this chapter, we will first study how these sounds propagate as waves. We will then describe a few different types of sounds and various ways of representing them. Finally, we will explain the concept of filtering, which allows certain frequencies to be singled-out.

1.1. Sound propagation

The propagation of a sound wave can occur in any direction, and depends on the obstacles in its path. We will essentially be focusing on *plane waves*, that is to say waves that only depend on one direction of space. We will assume that this direction is the x-axis, and therefore that the pressure $p(x, y, z, t)$ is independent of y and z. Hence it can simply be denoted by $p(x, t)$. This type of function represents a plane wave propagating through space, but also a sound wave inside a tube (see Figure 1.1), such as for example the one propagating through an organ pipe.

1.1.1. *A look at the physical models*

The propagation of sound through air is governed by the wave equation (see page 21), an equation we will come across several times since it also determines the movement of sound waves in the vibrating parts (strings, membranes, tubes...)

of many instruments. In the following paragraphs, we will see that, in the case of air, this equation is inferred from three fundamental equations of continuum mechanics.

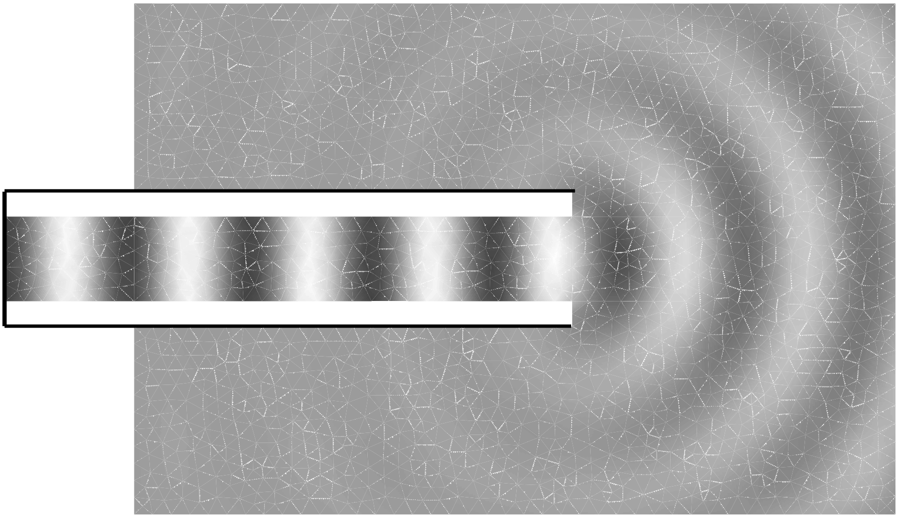

Figure 1.1. *Pressure waves in a tube open at its right end, with pressure imposed at the other end*

Along with the pressure $p(x,t)$, we rely on two other variables to describe the state of air: its density $\rho(x,t)$, and the *average* speed $v(x,t)$ of the air molecules set in motion by the sound wave, which is not to be confused with the norm of the *individual* speed of each molecule due to thermal agitation, the magnitude of which is close to that of the speed of sound, denoted by c. In the case of the plane wave that we are studying, the air moves in a direction parallel to the Ox-axis, and both the speed v, and the pressure are independent of y and z. In the absence of an atmospheric perturbation, v varies around the average value 0, and p and ρ vary around their average values p_0 and ρ_0 (see section 1.1.2), that is to say, their values in the equilibrium state: silence.

1.1.1.1. *Mass conservation*

In a *fixed* section of space, bounded by a cylinder with its axis parallel to the Ox axis and the two surfaces S_a and S_b, with respective x-coordinates a and b and areas S (see Figure 1.2), the variation of the air mass $m(t)$ is due to the amount of air going through the two surfaces. Nothing goes through the other interfaces, because the speed is parallel to the Ox axis. The air mass located inside the section is

$$m(t) = S \int_a^b \rho(x,t)\, dx,$$

and the variation of the air mass per unit of time is the derivative of $m(t)$, denoted by $m'(t)$. The incoming *flux* through S_a, that is to say, the amount of air entering the section per unit of time, is equal to $S\rho(a,t)v(a,t)$. As for the incoming flux through S_b, it is equal to $-S\rho(b,t)v(b,t)$, the change of sign being due to the fact that we are calculating the balance of what is *entering* the section (and not of what is going from left to right). The total flux is therefore

$$\Phi(t) = S[\rho(a,t)v(a,t) - \rho(b,t)v(b,t)].$$

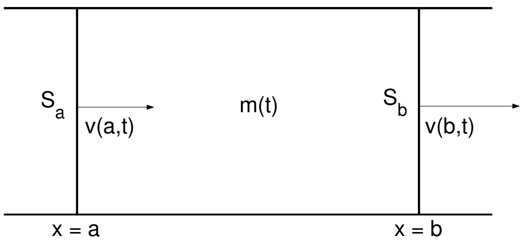

Figure 1.2. *Mass balance in the air section: there is no disappearance or creation of air!*

The fact that the total flux $\Phi(t)$ is the derivative of the mass $m(t)$,

$$\Phi(t) = m'(t),$$

can be expressed, if ∂_t denotes the partial derivative with respect to t, by

$$S[\rho(a,t)v(a,t) - \rho(b,t)v(b,t)] = S\int_a^b \partial_t\rho(x,t)\,dx.$$

If we divide by $b-a$ and if $b-a$ tends to 0 (calculation of the derivative with respect to the first argument), then after dividing both sides of the equation by x (who was on parole, confined between a and b):

$$-\partial_x(\rho(x,t)v(x,t)) = \partial_t\rho(x,t). \tag{1.1}$$

The linear acoustics hypothesis consists of assuming that the variations with respect to the equilibrium state are small, hence the use of the parameter ε, assumed to be ‘small’:

$$v(x,t) = \varepsilon v_1(x,t), \quad \rho(x,t) = \rho_0 + \varepsilon\rho_1(x,t).$$

If we substitute these two expressions in (1.1), and if we neglect ε^2, we get the conservation of mass equation, also called *continuity equation*:

$$\partial_t\rho_1(x,t) + \rho_0\partial_x v_1(x,t) = 0. \tag{1.2}$$