

# 1 Why Things Sound the Way They Do

We are very fortunate to have ears. Our auditory system provides us with an incredibly rich and nuanced source of information about the world around us. Listening is not just a very useful, but also often a very enjoyable activity. If your ears, and your auditory brain, work as they should, you will be able to distinguish thousands of sounds effortlessly—running water, slamming doors, howling wind, falling rain, bouncing balls, rustling paper, breaking glass, or footsteps (in fact, countless different types of footsteps: the crunching of leather soles on gravel, the tic-toc-tic of stiletto heels on a marble floor, the cheerful splashing of a toddler stomping through a puddle, or the rhythmic drumming of galloping horses or marching armies). The modern world brings modern sounds. You probably have a pretty good idea of what the engine of your car sounds like. You may even have a rather different idea of what the engine of your car *ought to* sound like, and be concerned about that difference. Sound and hearing are also enormously important to us because of the pivotal role they play in human communication. You have probably never thought about it this way, but every time you talk to someone, you are effectively engaging in something that can only be described as a telepathic activity, as you are effectively “beaming your thoughts into the other person’s head,” using as your medium a form of “invisible vibrations.” Hearing, in other words, is the telepathic sense that we take for granted (until we lose it) and the sounds in our environment are highly informative, very rich, and not rarely enjoyable.

If you have read other introductory texts on hearing, they will probably have told you, most likely right at the outset, that “sound is a pressure wave, which propagates through the air.” That is, of course, entirely correct, but it is also somewhat missing the point. Imagine you hear, for example, the din of a drawer full of cutlery crashing down onto the kitchen floor. In that situation, lots of minuscule ripples of air pressure will be radiating out from a number of mechanically excited metal objects, and will spread outwards in concentric spheres at the speed of sound, a breathless 340 m/s (about 1,224 km/h or 760 mph), only to bounce back from the kitchen walls and

ceiling, filling, within only a few milliseconds, all the air in the kitchen with a complex pattern of tiny, ever changing ripples of air pressure. Fascinating as it may be to try to visualize all these wave patterns, these sound waves certainly do not describe what we “hear” in the subjective sense.

The mental image your sense of hearing creates will not be one of delicate pressure ripples dancing through the air, but rather the somewhat more alarming one of several pounds of sharp knives and forks, which have apparently just made violent and unexpected contact with the kitchen floor tiles. Long before your mind has had a chance to ponder any of this, your auditory system will already have analyzed the sound pressure wave pattern to extract the following useful pieces of information: that the fallen objects are indeed made of metal, not wood or plastic; that there is quite a large number of them, certainly more than one or two; that the fallen metal objects do not weigh more than a hundred grams or so each (i.e., the rampaging klutz in our kitchen has indeed spilled the cutlery drawer, not knocked over the cast iron casserole dish); as well as that their impact occurred in our kitchen, not more than 10 meters away, slightly to the left, and not in the kitchen of our next door neighbors or in a flat overhead.

That our auditory brains can extract so much information effortlessly from just a few “pressure waves” is really quite remarkable. In fact, it is more than remarkable, it is astonishing. To appreciate the wonder of this, let us do a little thought experiment and imagine that the klutz in our kitchen is in fact a “compulsive serial klutz,” and he spills the cutlery drawer not once, but a hundred times, or a thousand. Each time our auditory system would immediately recognize the resulting cacophony of sound: “Here goes the cutlery drawer again.” But if you were to record the sounds each time with a microphone and then look at them on an oscilloscope or computer screen, you would notice that the sound waves would actually look quite different on each and every occasion.

There are infinitely many different sound waves that are all recognizable as the sound of cutlery bouncing on the kitchen floor, and we can recognize them even though we hear each particular cutlery-on-the-floor sound only once in our lives. Furthermore, our prior experience of hearing cutlery crashing to the floor is likely to be quite limited (cutlery obsessed serial klutzes are, thankfully, a very rare breed). But even so, most of us have no difficulty imagining what cutlery crashing to floor would sound like. We can even imagine how different the sound would be depending on whether the floor was made of wood, or covered in linoleum, or carpet, or ceramic tiles.

This little thought experiment illustrates an important point that is often overlooked in introductory texts on hearing. Sound and hearing are so useful because *things make sounds, and different things make different sounds*. Sound waves carry valuable clues about the physical properties of the objects or events that created them,

and when we listen we do not seek to sense vibrating air for the sake of it, but rather we hope to learn something about the sound *sources*, that is, the objects and events surrounding us. For a proper understanding of hearing, we should therefore start off by learning at least a little bit about how sound waves are created in the first place, and how the physical properties of sound sources shape the sounds they make.

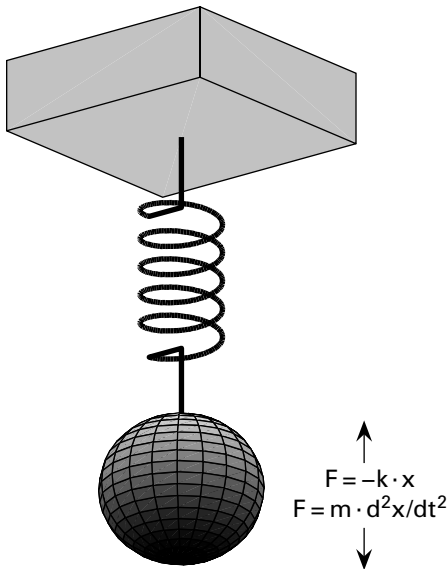
### 1.1 Simple Harmonic Motion—Or, Why Bells Go “Bing” When You Strike Them

Real-world sounds, like those we just described, are immensely rich and complex. But in sharp contrast to these “natural” sounds, the sounds most commonly used in the laboratory to study hearing are by and large staggeringly dull. The most common laboratory sound by far is the sine wave pure tone, a sound that most nonscientists would describe, entirely accurately, as a “beep”—but not just any beep, and most certainly not an interesting one. To be a “pure” tone, the beep must be shorn of any “contaminating” feature, be completely steady in its amplitude, contain no “amplitude or frequency modulations” (properties known as *vibrato* to the music lover) nor any harmonics (overtones) or other embellishing features. A pure tone is, indeed, so bare as to be almost “unnatural”: pure tones are hardly ever found in everyday soundscapes, be they manmade or natural.

You may find this puzzling. If pure tones are really quite boring and very rare in nature (and they are undeniably both), and if hearing is about perceiving the real world, then why are pure tones so widely used in auditory research? Why would anyone think it a good idea to test the auditory system mostly with sounds that are neither common nor interesting? There are, as it turns out, a number of reasons for this, some good ones (or at least they seemed good at the time) and some decidedly less good ones. And clarifying the relationship between sinusoidal pure tones and “real” sounds is in fact a useful, perhaps an essential, first step toward achieving a proper understanding of the science of hearing. To take this step we will need, at times, a mere smidgen of mathematics. Not that we will expect you, dear reader, to do any math yourself, but we will encourage you to bluff your way along, and in doing so we hope you will gain an intuitive understanding of some key concepts and techniques. Bluffing one’s way through a little math is, in fact, a very useful skill to cultivate for any sincere student of hearing. Just pretend that you kind of know this and that you only need a little “reminding” of the key points. With that in mind, let us confidently remind ourselves of a useful piece of applied mathematics that goes by the pleasingly simple and harmonious name of *simple harmonic motion*. To develop an intuition for this, let us begin with a simple, stylized object, a mass-spring system, which consists of a lump of some material (any material you like, as long as it’s not weightless and is reasonably solid) suspended from an elastic spring, as shown in figure



1.1. (See the book’s Web site for an animated version of this figure.)



**Figure 1.1**  
A mass-spring system.

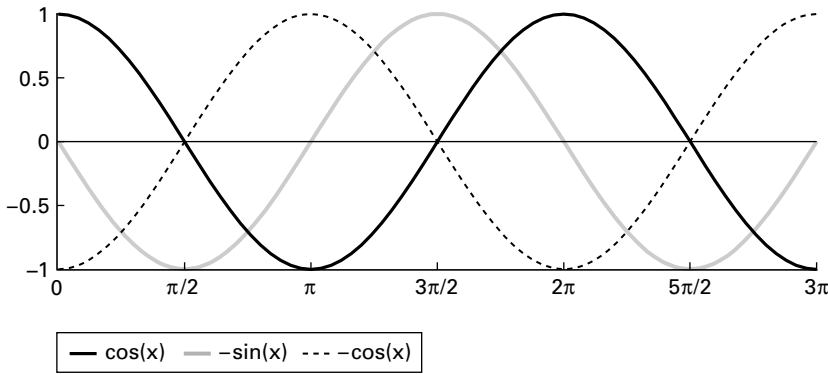
Let us also imagine that this little suspended mass has recently been pushed, so that it now travels in a downward direction. Let us call this the  $x$  direction. Now is an excellent time to start pretending that we were once quite good at school math and physics, and, suitably reminded, we now recall that masses on the move are inert, that is, they have a tendency to keep on moving in the same direction at the same speed until something forces them to slow down or speed up or change direction. The force required to do that is given by Newton's second law of motion, which states that force equals mass times acceleration, or  $F = m \cdot a$ .

Acceleration, we further recall, is the rate of change of velocity ( $a = dv/dt$ ) and velocity is the rate of change of position ( $v = dx/dt$ ). So we can apply Newton's second law to our little mass as follows: It will continue to travel with constant velocity  $dx/dt$  in the  $x$  direction until it experiences a force that changes its velocity; the rate of change in velocity is given by  $F = m \cdot d^2x/dt^2$ . (By the way, if this is getting a bit heavy going, you may skip ahead to the paragraph beginning "In other words...." We won't tell anyone you skipped ahead, but note that bluffing your way in math takes a little practice, so persist if you can.) Now, as the mass travels in the  $x$  direction, it will soon start to stretch the spring, and the spring will start to pull against this stretch with a force given by Hooke's law, which states the pull of the spring is proportional to how far it is stretched, and it acts in the opposite direction of the stretch

(i.e.,  $F = -k \cdot x$ , where  $k$  is the spring constant, a proportionality factor that is large for strong, stiff springs and small for soft, bendy ones; the minus sign reminds us that the force is in a direction that opposes further stretching).

So now we see that, as the mass moves inertly in the  $x$  direction, there soon arises a little tug of war, where the spring will start to pull against the mass' inertia to slow it down. The elastic force of the spring and the inertial force of the mass are then, in accordance with Newton's third law of motion, equal in strength and opposite in direction, that is,  $-k \cdot x = m \cdot d^2x/dt^2$ . We can rearrange this equation using elementary algebra to  $d^2x/dt^2 = -k/m \cdot x$  to obtain something that many students of psychology or biology would rather avoid, as it goes by the intimidating name of *second-order differential equation*. But we shall not be so easily intimidated. Just note that this equation only expresses, in mathematical hieroglyphics, something every child playing with a slingshot quickly appreciates intuitively, namely, that the harder one pulls the mass in the slingshot against the elastic, the harder the elastic will try to accelerate the mass in the opposite direction. If that rings true, then deciphering the hieroglyphs is not difficult. The acceleration  $d^2x/dt^2$  is large if the slingshot has been stretched a long way ( $-x$  is large), if the slingshot elastic is stiff ( $k$  is large), and if the mass that needs accelerating is small (again, no surprise: You may remember from childhood that large masses, like your neighbor's garden gnomes, are harder to catapult at speed than smaller masses, like little pebbles).

But what does all of this have to do with sound? This will become clear when we quickly "remind" ourselves how one solves the differential equation  $d^2x/dt^2 = -k/m \cdot x$ . Basically, here's how mathematicians do this: they look up the solution in a book, or get a computer program for symbolic calculation to look it up for them, or, if they are very experienced, they make an intelligent guess and then check if it's true. Clearly, the solution must be a function that is proportional to minus its own second derivative (i.e., "the rate of change of the rate of change" of the function must be proportional to minus its value at each point). It just so happens that sine and cosine functions, pretty much uniquely, possess this property. Look at the graph of the cosine function that we have drawn for you in figure 1.2, and note that, at zero, the cosine has a value of 1, but is flat because it has reached its peak, and has therefore a slope of 0. Note also that the  $-\text{sine}$  function, which is also plotted in gray, has a value of 0 at zero, so here the slope of the cosine happens to be equal to the value of  $-\text{sine}$ . This is no coincidence. The same is also true for  $\cos(\pi/2)$ , which happens to have a value of 0 but is falling steeply with a slope of  $-1$ , while the value of  $-\sin(\pi/2)$  is also  $-1$ . It is, in fact, true everywhere. The slope of the cosine is minus the sine, and the slope of minus sine is minus the cosine. Sine waves, uniquely, describe the behavior of mass-spring systems, because they are everywhere proportional to minus their own second derivative [ $d^2 \cos(t)/dt^2 = -\cos(t)$ ] and they therefore satisfy the differential equation that describes the forces in a mass-spring system.



**Figure 1.2**

The cosine and its derivatives.

In other words, and this is the important bit, *the natural behavior for any mass-spring system is to vibrate in a sinusoidal fashion*. And given that many objects, including (to get back to our earlier example) items of cutlery, have both mass and a certain amount of springiness, it is perfectly natural for them, or parts of them, to enter into “simple harmonic motion,” that is, to vibrate sinusoidally. Guitar or piano strings, bicycle bells, tuning forks, or xylophone bars are further familiar examples of everyday mass-spring systems with obvious relevance to sound and hearing. You may object that sinusoidal vibration can’t really be the “natural way to behave” for all mass-spring systems, because, most of the time, things like your forks and knives do not vibrate, but sit motionless and quiet in their drawer, to which we would reply that a sinusoidal vibration of zero amplitude is a perfectly fine solution to our differential equation, and no motion at all still qualifies as a valid and natural form of simple harmonic motion.

So the natural behavior (the “solution”) of a mass-spring system is a sinusoidal vibration, and written out in full, the solution is given by the formula  $x(t) = x_0 \cdot \cos(t \cdot \sqrt{k/m} + \phi_0)$ , where  $x_0$  is the “initial amplitude” (i.e., how far the little mass had been pushed downward at the beginning of this little thought experiment), and  $\phi_0$  is its “initial phase” (i.e., where it was in the cycle at time zero). If you remember how to differentiate functions like this, you can quickly confirm that this solution indeed satisfies our original differential equation. Alternatively, you can take our word for it. But we would not be showing you this equation, or have asked you to work so hard to get to it, if there weren’t still quite a few very worthwhile insights to gain from it. Consider the  $\cos(t \cdot \sqrt{k/m})$  part. You may remember that the cosine function goes through one full cycle over an angle of  $360^\circ$ , or  $2\pi$  radians. So the mass-spring system has swung through one full cycle when  $t \cdot \sqrt{k/m}$  equals  $2\pi$ . It follows that the period

(i.e., the time taken for one cycle of vibration) is  $T = 2\pi/\sqrt{k/m}$ . And you may recall that the frequency (i.e., the number of cycles per unit time) is equal to 1 over the period, so  $f = \sqrt{k/m}/2\pi$ .

Translated into plain English, this tells us that our mass-spring system has a preferred or natural frequency at which it wants to oscillate or vibrate. This is known as the system's resonance (or resonant) frequency, and it is inversely proportional to the square root of its mass and proportional to the square root of its stiffness. If this frequency lies within the human audible frequency range (about 20–20,000 cycles/s, or Hz), then we may hear these vibrations as sound. Although you may, so far, have been unaware of the underlying physics, you have probably exploited these facts intuitively on many occasions. So when, to return to our earlier example, the sounds coming from our kitchen tell us that a box full of cutlery is currently bouncing on the kitchen floor, we know that it is the cutlery and not the saucepans because the saucepans, being much heavier, would be playing much lower notes. And when we increase the tension on a string while tuning a guitar, we are, in a manner of speaking, increasing its “stiffness,” the springlike force with which the string resists being pushed sideways. And by increasing this tension, we increase the string's resonance frequency.

Hopefully, this makes intuitive sense to you. Many objects in the world around us are or contain mass-spring systems of some type, and their resonant frequencies tell us something about the objects' physical properties. We mentioned guitar strings and metallic objects, but another important, and perhaps less obvious example, is the resonant cavity. Everyday examples of resonant cavities might include empty (or rather air-filled) bottles or tubes, or organ pipes. You may know from experience that when you very rapidly pull a cork out of a bottle, it tends to make a “plop” sound, and you may also have noticed that the pitch of that sound depends on how full the bottle is. If the bottle is almost empty (of liquid, and therefore contains quite a lot of air), then the plop is much deeper than when the bottle is still quite full, and therefore contains very little air. You may also have amused yourself as a kid by blowing over the top of a bottle to make the bottle “whistle” (or you may have tried to play a pan flute, which is much the same thing), and noticed that the larger the air-filled volume of the bottle, the lower the sound.

Resonant cavities like this are just another version of mass-spring systems, only here both the mass and the spring are made of air. The air sitting in the neck of the bottle provides the mass, and the air in the belly of the bottle provides the “spring.” As you pull out the cork, you pull the air in the neck just below the cork out with it. This decreases the air pressure in the belly of the bottle, and the reduced pressure provides a spring force that tries to suck the air back in. In this case, the mass of the air in the bottle neck and the spring force created by the change in air pressure in the bottle interior are both very small, but that does not matter. As long as they are balanced to give a resonant frequency in the audible range, we can still produce a clearly

audible sound. How large the masses and spring forces of a resonator are depends a lot on its geometry, and the details can become very complex; but in the simplest case, the resonant frequency of a cavity is inversely proportional to the square root of its volume, which is why small organ pipes or drums play higher notes than large ones.

Again, these are facts that many people exploit intuitively, even if they are usually unaware of the underlying physics. Thus, we might knock on an object made of wood or metal to test whether it is solid or hollow, listening for telltale low resonant frequencies that would betray a large air-filled resonant cavity inside the object. Thus, the resonant frequencies of objects give us valuable clues to the physical properties, such as their size, mass, stiffness, and volume. Consequently, it makes sense to assume that a “frequency analysis” is a sensible thing for an auditory system to perform.

We hope that you found it insightful to consider mass-spring systems, and “solve” them to derive their resonant frequency. But this can also be misleading. You may recall that we told you at the beginning of this chapter that pure sine wave sounds hardly ever occur in nature. Yet we also said that mass-spring systems, which are plentiful in nature, should behave according to the equation  $x(t) = x_0 \cdot \cos(t \cdot \sqrt{k/m} + \varphi_0)$ ; in other words, they should vibrate sinusoidally at their single preferred resonance frequency,  $f = \sqrt{k/m}/2\pi$ , essentially forever after they have been knocked or pushed or otherwise mechanically excited. If this is indeed the case, then pure tone-emitting objects should be everywhere. Yet they are not. Why not?

## 1.2 Modes of Vibration and Damping—Or Why a “Bing” Is Not a Pure Tone

When you pluck a string on a guitar, that string can be understood as a mass-spring system. It certainly isn’t weightless, and it is under tension, which gives it a springlike stiffness. When you let go of it, it will vibrate at its resonant frequency, as we would expect, but that is not the only thing it does. To see why, ask yourself this: How can you be sure that your guitar string is indeed just one continuous string, rather than two half strings, each half as long as the original one, but seamlessly joined. You may think that this is a silly question, something dreamt up by a Zen master to tease a student. After all, each whole can be thought of as made of two halves, and if the two halves are joined seamlessly, then the two halves make a whole, so how could this possibly matter? Well, it matters because each of these half-strings weighs half as much and is twice as stiff as the whole string, and therefore each half-string will have a resonance frequency that is twice as high as that of the whole string.

When you pluck your guitar string, you make it vibrate and play its note, and the string must decide whether it is to vibrate as one whole or as two halves; if it chooses the latter option, the frequency at which it vibrates, and the sound frequency it emits, will double! And the problem doesn’t end there. If we can think of a string as two

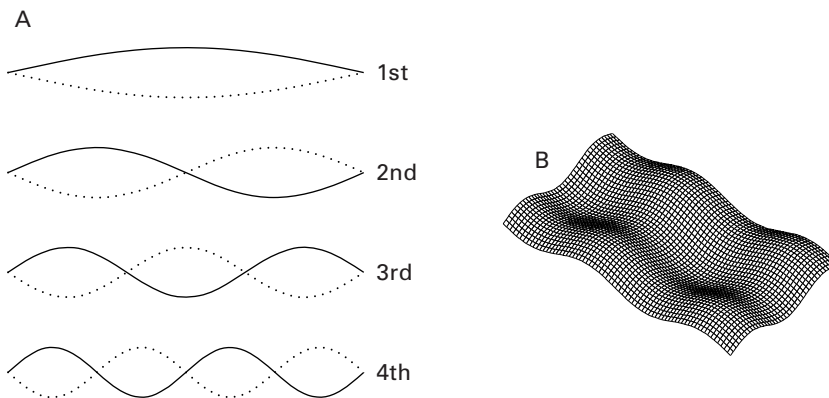


half-strings, then we can just as easily think of it as three thirds, or four quarters, and so forth. How does the string decide whether to vibrate as one single whole or to exhibit this sort of “split personality,” and vibrate as a collection of its parts? Well, it doesn’t. When faced with multiple possibilities, strings will frequently go for them all, all at the same time, vibrating simultaneously as a single mass-spring system, as well as two half mass systems, and as three thirds, and as four quarters, and so on. This behavior is known as “modes of vibration” of the string, and it is illustrated schematically in figure 1.3, as well as in an animation that can be found on the book’s



Web site.

Due to these modes, a plucked guitar string will emit not simply a pure tone corresponding to the resonant frequency of the whole string, but a mixture that also contains overtones of twice, three, four, or  $n$  times that resonant frequency. It will, in other words, emit a complex tone—“complex” not in the sense that it is complicated, but that it is made up of a number of frequency components, a mixture of harmonically related tones that are layered on top of each other. The lowest frequency component, the resonant frequency of the string as a whole, is known as the fundamental frequency, whereas the frequency components corresponding to the resonance of the half, third, fourth strings (the second, third, and fourth modes of vibration) are called the “higher harmonics.” The nomenclature of harmonics is a little confusing, in that some authors will refer to the fundamental frequency as the “zeroth harmonic,” or  $F_0$ , and the first harmonic would therefore equal twice the fundamental frequency, the second harmonic would be three times  $F_0$ , and the  $n$ th harmonic would be  $n + 1$  times  $F_0$ . Other authors number harmonics differently, and consider the fundamental




**Figure 1.3**

(A) The first four modes of vibration of a string. (B) A rectangular plate vibrating in the fourth mode along its length and in the third mode along its width.

to be also the first harmonic, so that the  $n$ th harmonic has a frequency of  $n$  times  $F_0$ . We shall adhere to the second of these conventions, not because it is more sensible, but because it seems a little more commonly used.

Although many physical objects such as the strings on an instrument, bells, or xylophone bars typically emit complex tones made up of many harmonics, these harmonics are not necessarily present in equal amounts. How strongly a particular harmonic is represented in the mix of a complex tone depends on several factors. One of these factors is the so-called initial condition. In the case of a guitar string, the initial condition refers to how, and where, the string is plucked. If you pull a guitar string exactly in the middle before you let it go, the fundamental first mode of vibration is strongly excited, because we have delivered a large initial deflection just at the first mode's "belly." However, vibrations in the second mode vibrate around the center. The center of the string is said to be a node in this mode of vibration, and vibrations on either side of the node are "out of phase" (in opposite direction); that is, as the left side swings down the right side swings up.

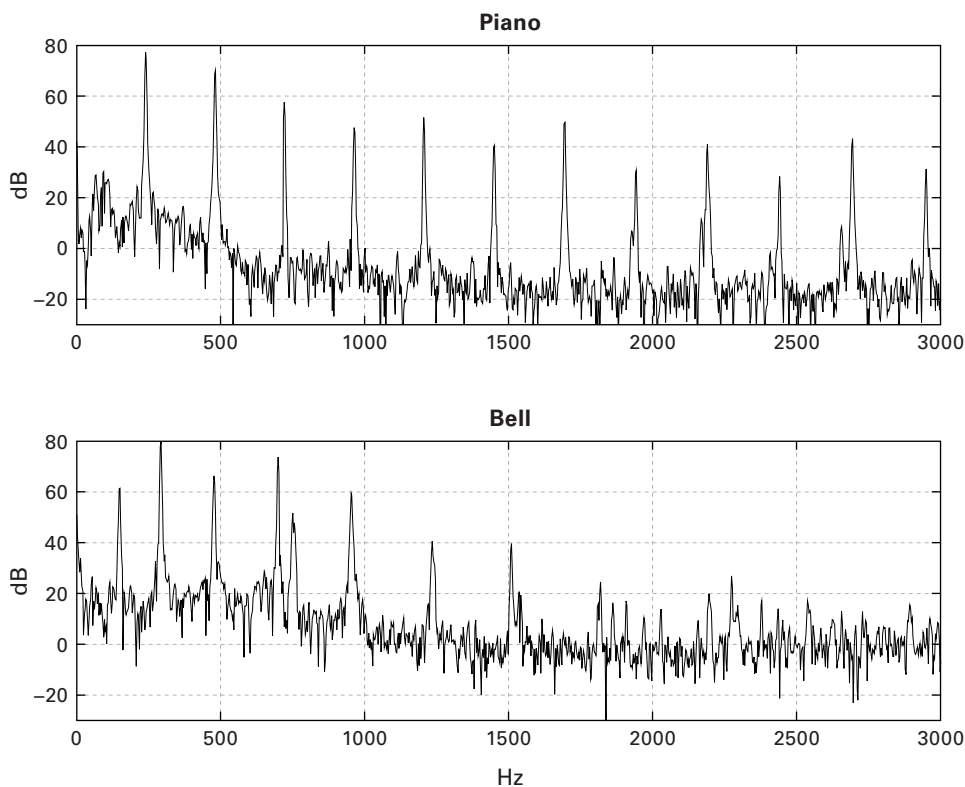
To excite the second mode we need to deflect the string asymmetrically relative to the midpoint. The initial condition of plucking the string exactly in the middle does not meet this requirement, as either side of the midpoint is pulled and then released in synchrony, so the second mode will not be excited. The fourth and sixth modes, or any other even modes will not be excited either, for the same reason. In fact, plucking the string exactly in the middle excites only the odd modes of vibration, and it excites the first mode more strongly than progressively higher odd modes. Consequently, a guitar string plucked in the middle will emit a sound with lots of energy at the fundamental, decreasing amounts of energy at the third, fifth, seventh ... harmonics, and no energy at all at the second, fourth, sixth... harmonics. If, however, a guitar string is plucked somewhere near one of the ends, then even modes may be excited, and higher harmonics become more pronounced relative to the fundamental. In this way, a skilled guitarist can change the timbre of the sound and make it sound "brighter" or "sharper."

Another factor affecting the modes of vibration of an object is its geometry. The geometry of a string is very straightforward; strings are, for all intents and purposes, one-dimensional. But many objects that emit sounds can have quite complex two- and three-dimensional shapes. Let us briefly consider a rectangular metal plate, which is struck. In principle, a plate can vibrate widthwise just as easily as it can vibrate along its length. It could, for example, vibrate in the third mode along its width and in the  fourth mode along its length, as is schematically illustrated in figure 1.3B. Also, in a metal plate, the stiffness stems not from an externally supplied tension, as in the guitar string, but from the internal tensile strength of the material.

Factors like these mean that three-dimensional objects can have many more modes of vibration than an ideal string, and not all of these modes are necessarily harmoni-

cally related. Thus, a metal plate of an “awkward” shape might make a rather dissonant, unmelodious “clink” when struck. Furthermore, whether certain modes are possible can depend on which points of the plate are fixed, and which are struck. The situation becomes very complicated very quickly, even for relatively simple structures such as flat, rectangular plates. For more complicated three-dimensional structures, like church bells, for example, understanding which modes are likely to be pronounced and how the interplay of possible modes will affect the overall sound quality, or timbre, is as much an art as a science.

To illustrate these points, figure 1.4 shows the frequency spectra of a piano note and of the chime of a small church bell. What exactly a frequency spectrum is is explained in greater detail later, but at this point it will suffice to say that a frequency spectrum tells us how much of a particular sinusoid is present in a complex sound. Frequency spectra are commonly shown using units of decibel (dB). Decibels are a



**Figure 1.4**

Frequency spectra of a piano and a bell, each playing the note  $B_3$  (247 Hz).

logarithmic unit, and they can be a little confusing, which is why we will soon say more about them.

To read figure 1.4, all you need to know is that when the value of the spectrum at one frequency is 20dB greater than that at another, the amplitude at that frequency is ten times larger, but if the difference is 40dB, then the amplitude is one hundred times greater, and if it is 60dB, then it is a whopping one thousand times larger. The piano and the bell shown in figure 1.4 both play the musical note  $B_3$  (more about musical notes in chapter 3). This note is associated with a fundamental frequency of 247Hz, and after our discussion of modes of vibration, you will not be surprised that the piano note does indeed contain a lot of 247-Hz vibration, as well as frequencies that are integer multiples of 247 (namely, 494, 741, 988, 1,235, etc.). In fact, frequencies that are not multiples of 247Hz (all the messy bits below 0dB in figure 1.4) are typically 60dB, that is, one thousand times smaller in the piano note than the string's resonant frequencies. The spectrum of the bell, however, is more complicated. Again, we see that a relatively small number of frequencies dominate the spectrum, and each of these frequency components corresponds to one of the modes of vibration of the bell. But because the bell has a complex three-dimensional shape, these modes are not all exact multiples of the 247-Hz fundamental.

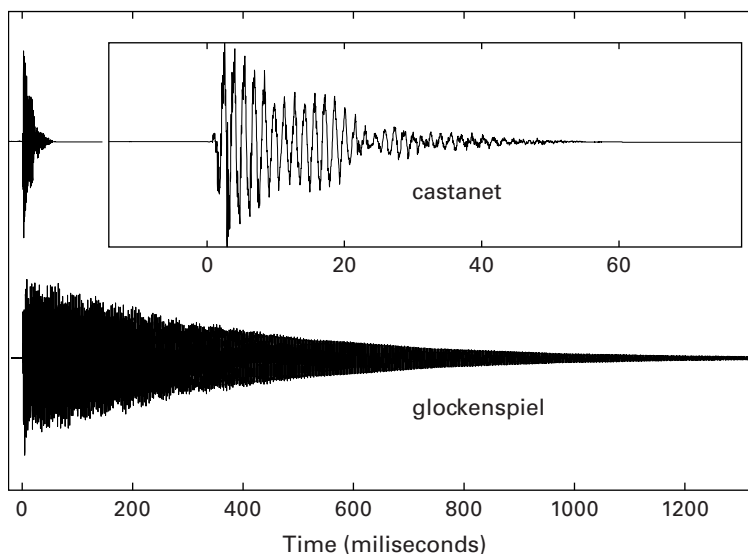
Thus, real objects do not behave like an idealized mass-spring system, in that they vibrate in numerous modes and at numerous frequencies, but they also differ from the idealized model in another important respect. An idealized mass-spring system should, once set in motion, carry on oscillating forever. Luckily, real objects settle down and stop vibrating after a while. (Imagine the constant din around us if they didn't!) Some objects, like guitar strings or bells made of metal or glass, may continue ringing for several seconds, but vibrations in many other objects, like pieces of wood or many types of plastic, tend to die down much quicker, within just a fraction of a second. The reason for this is perhaps obvious. The movement of the oscillating mass represents a form of kinetic energy, which is lost to friction of some kind or another, dissipates to heat, or is radiated off as sound. For objects made of highly springy materials, like steel bells, almost all the kinetic energy is gradually emitted as sound, and as a consequence the sound decays relatively slowly and in an exponential fashion. The reason for this exponential decay is as follows.

The air resistance experienced by a vibrating piece of metal is proportional to the average velocity of the vibrating mass. (Anyone who has ever ridden a motorcycle appreciates that air resistance may appear negligible at low speeds but will become considerable at higher speeds.) Now, for a vibrating object, the average speed of motion is proportional to the amplitude of the vibration. If the amplitude declines by half, but the frequency remains constant, then the vibrating mass has to travel only half as far on each cycle, but the available time period has remained the same, so it need move only half as fast. And as the mean velocity declines, so does the air resis-

tance that provides the breaking force for a further reduction in velocity. Consequently, some small but constant fraction of the vibration amplitude is lost on each cycle—the classic conditions for exponential decay. Vibrating bodies made of less elastic materials may experience a fair amount of internal friction in addition to the air resistance, and this creates internal “damping” forces, which are not necessarily proportional to the amplitude of the vibration. Sounds emitted by such objects therefore decay much faster, and their decay does not necessarily have an exponential time course.

By way of example, look at figure 1.5, which shows sound waves from two musical instruments: one from a wooden castanet, the other from a metal glockenspiel bar. Note that the time axes for the two sounds do not cover the same range. The wooden castanet is highly damped, and has a decay constant of just under 30ms (i.e., the sound takes 30ms to decay to  $1/e$  or about 37% of its maximum amplitude). The metal glockenspiel, in contrast, is hardly damped at all, and the decay constant of its vibrations is just under 400ms, roughly twenty times longer than that of the castanet.

Thus, the speed and manner with which a sound decays gives another useful cue to the properties of the material an object is made of, and few people would have any



**Figure 1.5**

A rapidly decaying (castanet) and a slowly decaying (glockenspiel) sound. The castanet sound is plotted twice, once on the same time axis as the glockenspiel, and again, in the inset, with a time axis that zooms in on the first 70ms.

difficulty using it to distinguish the sound of a copper bar from that of bar made of silver, even though some research suggests that our ability to use these cues is not as good as it perhaps ought to be (Lutfi & Liu, 2007).

We hope the examples in this section illustrate that objects in our environment vibrate in complex ways, but these vibrations nevertheless tell us much about various physical properties of the object, its weight, its material, its size, and its shape. A vibrating object pushes and pulls on the air around it, causing vibrations in the air which propagate as sound (we will look at the propagation of sound in a little more detail later). The resulting sound is hardly ever a pure tone, but in many cases it will be made up of a limited number of frequencies, and these are often harmonically related. The correspondence between emitted frequencies and physical properties of the sound source is at times ambiguous. Low frequencies, for example, could be either a sign of high mass or of low tension. Frequency spectra are therefore not always easy to interpret, and they are not quite as individual as fingerprints; but they nevertheless convey a lot of information about the sound source, and it stands to reason that one of the chief tasks of the auditory system is to unlock this information to help us judge and recognize objects in our environment. Frequency analysis of an emitted sound is the first step in this process, and we will return to the idea of the auditory system as a frequency analyzer in a number of places throughout this book.

### 1.3 Fourier Analysis and Spectra

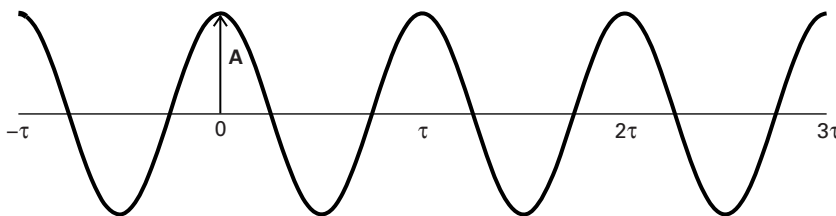
In 1822, the French mathematician Jean Baptiste Joseph Fourier posited that any function whatsoever can be thought of as consisting of a mixture of sine waves,<sup>1</sup> and to this day we refer to the set of sine wave components necessary to make up some signal as the signal's Fourier spectrum. It is perhaps surprising that, when Fourier came up with his idea, he was not studying sounds at all. Instead, he was trying to calculate the rate at which heat would spread through a cold metal ring when one end of it was placed by a fire. It may be hard to imagine that a problem as arcane and prosaic as heat flow around a metal ring would be sufficiently riveting to command the attention of a personality as big as Fourier's, a man who twenty-four years earlier had assisted Napoleon Bonaparte in his conquests, and had, for a while, been governor of lower Egypt. But Fourier was an engineer at heart, and at the time, the problem of heat flow around a ring was regarded as difficult, so he had a crack at it. His reasoning must have gone something like this: "I have no idea what the solution is, but I have a hunch that, regardless of what form the solution takes, it must be possible to express it as a sum of sines and cosines, and once I know that I can calculate it." Reportedly, when he first presented this approach to his colleagues at the French Academy of Sciences, his presentation was met by polite silence and incomprehension. After all, positing a sum of sinusoids as a solution was neither obviously correct, nor

obviously helpful. But no one dared challenge him as he was too powerful a figure—a classic case of “proof by intimidation.”

In the context of sounds, however, which, as we have learned, are often the result of sinusoidal, simple harmonic motion of mass-spring oscillators, Fourier’s approach has a great deal more immediate appeal. We have seen that there are reasons in physics why we would expect many sounds to be quite well described as a sum of sinusoidal frequency components, namely, the various harmonics. Fourier’s bold assertion that it ought to be possible to describe *any* function, and by implication also any variation in air pressure as a function of time (i.e., *any* sound), seems to offer a nice, unifying approach. The influence of Fourier’s method on the study of sound and hearing has consequently been enormous, and the invention of digital computers and efficient algorithms like the fast Fourier transform have made it part of the standard toolkit for the analysis of sound. Some authors have even gone so far as to call the ear itself a “biological Fourier analyzer.” This analogy between Fourier’s mathematics and the workings of the ear must not be taken too literally though. In fact, the workings of the ear only vaguely resemble the calculation of a Fourier spectrum, and later we will introduce better engineering analogies for the function of the ear. Perhaps this is just as well, because Fourier analysis, albeit mathematically very elegant, is in many respects also quite unnatural, if not to say downright weird. And, given how influential Fourier analysis remains to this day, it is instructive to pause for a moment to point out some aspects of this weirdness.

The mathematical formula of a pure tone is that of a sinusoid (figure 1.6). To be precise, it is  $A \cdot \cos(2\pi \cdot f \cdot t + \phi)$ . The tone oscillates sinusoidally with amplitude  $A$ , and goes through one full cycle of  $2\pi$  radians  $f$  times in each unit of time  $t$ . The period of the pure tone is the time taken for a single cycle, usually denoted by either a capital  $T$  or the Greek letter  $\tau$  (tau), and is equal to the inverse of the frequency  $1/f$ .

The tone may have had its maximal amplitude  $A$  at time 0, or it may not, so we allow a “starting phase parameter,”  $\phi$ , which we can use to shift the peaks of our sinusoid along the time axis as required. According to Fourier, we can describe any sound we like by taking a lot of sine wave equations like this, each with a different



**Figure 1.6**

A pure tone “in cosine phase” (remember that  $\cos(0) = 1$ ).

frequency  $f$ , and if we pick for each  $f$  exactly the right amplitude  $A$  and the right phase  $\phi$ , then the sum of these carefully picked sine waves will add up exactly to our arbitrary sound.

The sets of values of  $A$  and  $\phi$  required to achieve this are known as the sound's amplitude spectrum and phase spectrum, respectively. (Amplitude spectra we have encountered previously in figure 1.4, which effectively plotted the values of  $A$  for each of the numerous pure tone components making up the sound of a piano or a bell. Phase spectra, that is, the values of  $\phi$  for each frequency component, are, as we will see, often difficult to interpret and are therefore usually not shown.) Note, however, that, expressed in this mathematically rigorous way, each sine wave component is defined for all times  $t$ . Time 0 is just an arbitrary reference on the time axis; it is not in any real sense the time when the sound starts. The Fourier sine wave components making up the sound have no beginning and no end. They must be thought of as having started at the beginning of time and continuing, unchanging, with constant amplitude and total regularity, until the end of time. In that important respect, these mathematically abstract sine waves could not be more unlike "real" sounds. Most real sounds have clearly defined onsets, which occur when a sound source becomes mechanically excited, perhaps because it is struck or rubbed. And real sounds end when the oscillations of the sound source decay away. When exactly sounds occur, when they end, and how they change over time are perceptually very important to us, as these times give rise to perceptual qualities like rhythm, and let us react quickly to events signaled by particular sounds. Yet when we express sounds mathematically in terms of a Fourier transform, we have to express sounds that start and end in terms of sine waves that are going on forever, which can be rather awkward.

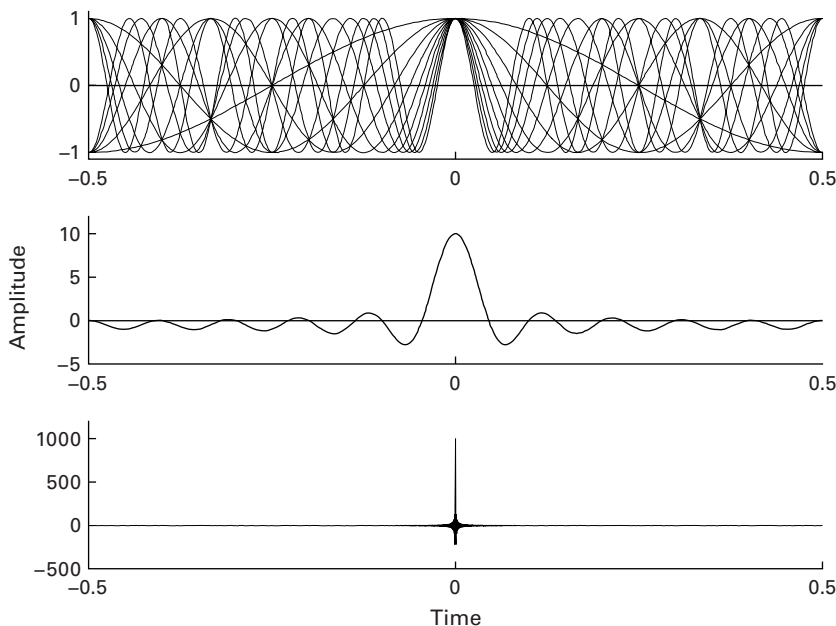
To see how this is done, let us consider a class of sounds that decay so quickly as to be almost instantaneous. Examples of this important class of "ultra-short" sounds include the sound of a pebble bouncing off a rock, or that of a dry twig snapping. These sound sources are so heavily damped that the oscillations stop before they ever really get started. Sounds like this are commonly known as "clicks." The mathematical idealization of a click, a deflection that lasts only for one single, infinitesimally short time step, is known as an impulse (or sometimes as a delta-function). Impulses come in two varieties, positive-going "compression" clicks (i.e., a very brief upward deflection or increase in sound pressure) or negative-going "rarefactions" (a transient downward deflection or pressure decrease).

Because impulses are so short, they are, in many ways, a totally different type of sound from the complex tones that we have considered so far. For example, impulses are not suitable for carrying a melody, as they have no clear musical pitch. Also, thinking of impulses in terms of sums of sine waves may seem unnatural. After all, a click is too short to go through numerous oscillations. One could almost say that the defining characteristic of a click is the predominant absence of sound: A click is a click only



because there is silence both before and immediately after it. How are we to produce this silence by adding together a number of “always on” sine waves of the form  $A \cdot \cos(2\pi \cdot f \cdot t + \phi)$ ? The way to do this is not exactly intuitively obvious: We have to take an awful lot of such sine waves (infinitely many, strictly speaking), all of different frequency  $f$ , and get them to cancel each other out almost everywhere. To see how this works, consider the top panel of figure 1.7, which shows ten sine waves of frequencies 1 to 10Hz superimposed. All have amplitude 1 and a starting phase of 0.

What would happen if we were to add all these sine waves together? Well at time  $t = 0$ , each has amplitude 1 and they are all in phase, so we would expect their sum to have amplitude 10 at that point. At times away from zero, it is harder to guess what the value of the sum would be, as the waves go out of phase and we therefore have to expect cancellation due to destructive interference. But for most values of  $t$ , there appear to be as many lines above the x-axis as below, so we might expect a lot of cancellation, which would make the signal small. The middle panel in figure 1.7 shows what the sum of the ten sine waves plotted in the top panel actually looks like, and it confirms our expectations. The amplitudes “pile up” at  $t = 0$  much more than elsewhere. But we still have some way to go to get something resembling a real impulse. What if we keep going, and keep adding higher and higher frequencies? The bottom



**Figure 1.7**

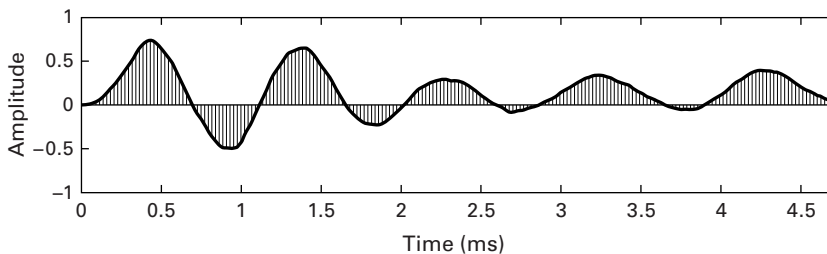
Making an impulse from the superposition of a large number of sine waves.

panel shows what we get if we sum cosines of frequencies 1 to 1,000. The result is a great deal more clicklike, and you may begin to suspect that if we just kept going and added infinitely many cosines of ever-increasing frequency, we would eventually, “in the limit” as mathematicians like to say, get a true impulse.

Of course, you may have noticed that, if we approximate a click by summing  $n$  cosines, its amplitude is  $n$ , so that, in the limit, we would end up with an infinitely large but infinitely short impulse, unless, of course, we scaled each of the infinitely many cosines we are summing to be infinitely small so that their amplitudes at time 0 could still add up to something finite. Is this starting to sound a little crazy? It probably is. “The limit” is a place that evokes great curiosity and wonder in the born mathematician, but most students who approach sound and hearing from a biological or psychological perspective may find it a slightly confusing and disconcerting place. Luckily, we don’t really need to go there.

Real-world clicks are very short, but not infinitely short. In fact, the digital audio revolution that we have witnessed over the last few decades was made possible only by the realization that one can be highly pragmatic and think of time as “quantized”; in other words, we posit that, for all practical purposes, there is a “shortest time interval” of interest, known as the “sample interval.” The shortest click or impulse, then, lasts for exactly one such interval. The advantage of this approach is that it is possible to think of any sound as consisting of a series of very many such clicks—some large, some small, some positive, some negative, one following immediately on another. For human audio applications, it turns out that if we set this sample period to be less than about 1/40,000 of a second, a sound that is “sampled” in this manner is, to the human ear, indistinguishable from the original, continuous-time sound wave.<sup>2</sup> Figure 1.8 shows an example of a sound that is “digitized” in this fashion.

Each constituent impulse of the sound can still be thought of as a sum of sine waves (as we have seen in figure 1.7). And if any sound can be thought of as composed of many impulses, and each impulse in turn can be composed of many sine waves, then it follows that any sound can be made up of many sine waves. This is not exactly a



**Figure 1.8**

One period of the vowel /a/ digitized at 44.1 kHz.

formal proof of Fourier's theorem, but it will do for our purposes. Of course, to make up some arbitrary sound digitally by fusing together a very large number of scaled impulses, we must be able to position each of the constituent impulses very precisely in time. But if one of these impulses is now to occur at precisely some time  $t = x$ , rather than at time 0 as in figure 1.7, then the sine waves making up that particular impulse must now all cancel at time 0, and pile up at time  $x$ . To achieve this, we have to adjust the "phase term"  $\phi$  in the equation of each sine wave component of that impulse. In other words, at time  $t = x$ , we need  $A \cdot \cos(2\pi \cdot f \cdot t + \phi)$  to evaluate to  $A$ , which means  $\cos(2\pi \cdot f \cdot x + \phi)$  must equal 1, and therefore  $(2\pi \cdot f \cdot x + \phi)$  must equal 0. To achieve that, we must set the phase  $\phi$  of each sine component to be exactly equal to  $-2\pi \cdot f \cdot x$ .

Does this sound a little complicated and awkward? Well, it is, and it illustrates one of the main shortcomings of representing sounds in the frequency domain (i.e., as Fourier spectra): Temporal features of a sound become encoded rather awkwardly in the "phase spectrum." A sound with a very simple time domain description like "click at time  $t = 0.3$ ," can have a rather complicated frequency domain description such as: "sine wave of frequency  $f = 1$  with phase  $\phi = -1.88496$  plus sine wave of frequency  $f = 2$  with phase  $\phi = -3.76991$  plus sine wave of frequency  $f = 3$  with phase  $\phi = -5.654867$  plus ..." and so on. Perhaps the most interesting and important aspect of the click, namely, that it occurred at time  $t = 0.3$ , is not immediately obvious in the click's frequency domain description, and can only be inferred indirectly from the phases. And if we were to consider a more complex natural sound, say the rhythm of hoof beats of a galloping horse, then telling which hoof beat happens when just from looking at the phases of the Fourier spectrum would become exceedingly difficult. Of course, our ears have no such difficulty, probably because the frequency analysis they perform differs in important ways from calculating a Fourier spectrum.

Both natural and artificial sound analysis systems get around the fact that time disappears in the Fourier spectrum by working out short-term spectra. The idea here is to divide time into a series of "time windows" before calculating the spectra. This way, we can at least say in which time windows a particular acoustic event occurred, even if it remains difficult to determine the timing of events inside any one time window. As we shall see in chapter 2, our ears achieve something that vaguely resembles such a short-term Fourier analysis through a mechanical tuned filter bank. But to understand the ear's operation properly, we first must spend a little time discussing time windows, filters, tuning, and impulse responses.

## 1.4 Windowing and Spectrograms

As we have just seen, the Fourier transform represents a signal (i.e., a sound in the cases that interest us here) in terms of potentially infinitely many sine waves that last,

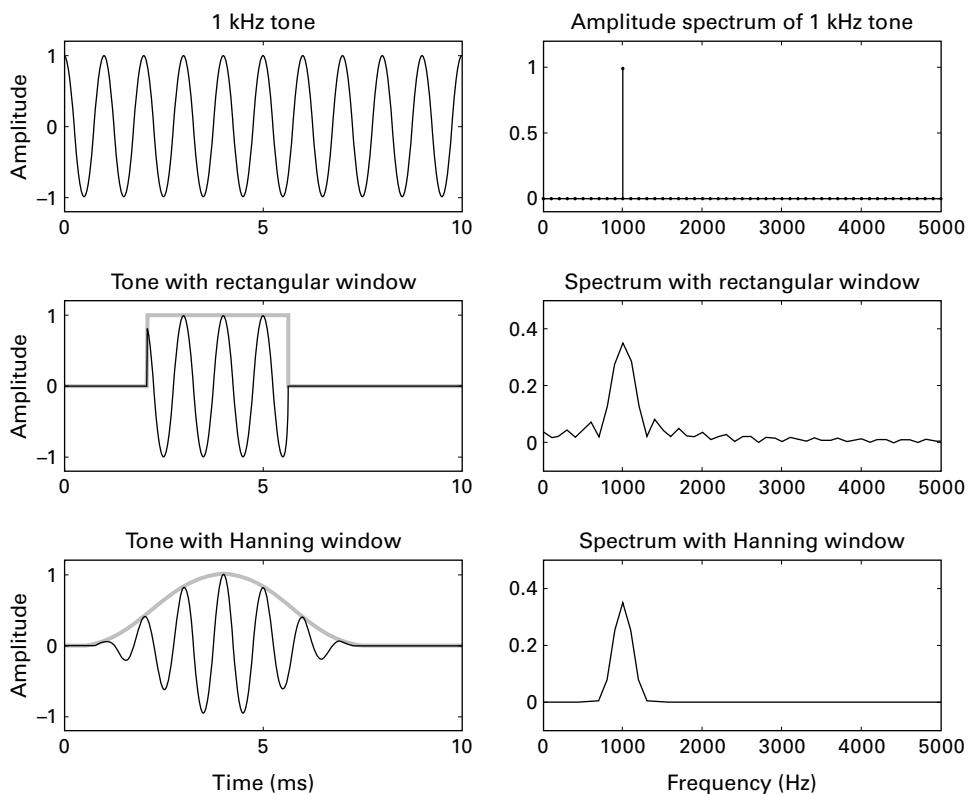
in principle, an infinitely long time. But infinitely long is inconveniently long for most practical purposes. An important special case arises if the sound we are interested in is periodic, that is, the sound consists of a pattern that repeats itself over and over. Periodic sounds are, in fact, a hugely important class of acoustic stimuli, so much so that chapter 3 is almost entirely devoted to them. We have already seen that sounds that are periodic, at least to a good approximation, are relatively common in nature. Remember the case of the string vibrating at its fundamental frequency, plus higher harmonics, which correspond to the various modes of vibration. The higher harmonics are all multiples of the fundamental frequency, and while the fundamental goes through exactly one cycle, the harmonics will go through exactly two, three, four, ... cycles. The waveform of such periodic sounds is a recurring pattern, and, we can therefore imagine that for periodic sounds time “goes around in circles,” because the same thing happens over and over again. To describe such periodic sounds, instead of a full-fledged Fourier transform with infinitely many frequencies, we only need a “Fourier series” containing a finite number of frequencies, namely, the fundamental plus all its harmonics up to the highest audible frequency. Imagine we record the sound of an instrument playing a very clean 100-Hz note; the spectrum of any one 10-ms period of that sound would then be the same as that of the preceding and the following period, as these are identical, and with modern computers it is easy to calculate this spectrum using the discrete Fourier transform.

But what is to stop us from taking *any* sound, periodic or not, cutting it into small, say 10-ms wide, “strips” (technically known as time windows), and then calculating the spectrum for each? Surely, in this manner, we would arrive at a simple representation of how the distribution of sound frequencies changes over time. Within any one short time window, our spectral analysis still poorly represents temporal features, but we can easily see when spectra change substantially from one window to the next, making it a straightforward process to localize features in time to within the resolution afforded by a single time window. In principle, there is no reason why this cannot be done, and such windowing and short-term Fourier analysis methods are used routinely to calculate a sound’s *spectrogram*. In practice, however, one needs to be aware of a few pitfalls.

One difficulty arises from the fact that we cannot simply cut a sound into pieces any old way and expect that this will not affect the spectrum. This is illustrated in figure 1.9. The top panel of the figure shows a 10-ms snippet of a 1-kHz tone, and its amplitude spectrum. A 1-kHz tone has a period of 1 ms, and therefore ten cycles of the tone fit exactly into the whole 10-ms-wide time window. A Fourier transform considers this 1-kHz tone as the tenth harmonic of a 100-Hz tone—100 Hz because the total time window is 10 ms long, and this duration determines the period of the fundamental frequency assumed in the transform. The Fourier amplitude spectrum of the 1-kHz tone is therefore as simple as we might expect of a pure tone snippet: It contains only a single frequency component. So where is the problem?

Well, the problem arises as soon as we choose a different time window, one in which the window duration is no longer a multiple of the period of the frequencies we wish to analyze. An example is shown in the second row of figure 1.9. We are still dealing with the same pure tone snippet, but we have now cut a segment out of it by imposing a rectangular window on it. The window function is shown in light gray. It is simply equal to 0 at all the time points we don't want, and equal to 1 at all the time points we do want. This rectangular window function is the mathematical description of an on/off switch. If we multiply the window function with the sound at each time point then, we get 0 times sound equals 0 during the off period, and 1 times sound equals sound in the on period. You might think that if you have a 1-kHz pure tone, simply switching it on and off to select a small segment for frequency analysis, should not alter its frequency content. You would be wrong.

Cast your mind back to figure 1.7, which illustrated the Fourier transform of a click, and in which we had needed an unseemly large number of sine waves just to cancel



**Figure 1.9**

The effect of windowing on the spectrum.

the sound off where we didn't want it. Something similar happens when we calculate the Fourier transform of a sine wave snippet where the period of the sine wave is not a multiple of the entire time window entered into the Fourier analysis. The abrupt onset and the offset create discontinuities, that is, "sudden sharp bends" in the waveform, and from the point of view of a Fourier analysis, discontinuities are broadband signals, made up of countless frequencies. You might be forgiven for thinking that this is just a bit of mathematical sophistry, which has little to do with the way real hearing works, but that is not so. Imagine a nerve cell in your auditory system, which is highly selective to a particular sound frequency, say a high frequency of 4,000 Hz or so. Such an auditory neuron should not normally respond to a 1,000-Hz tone, *unless* the 1-kHz tone is switched on or off very suddenly. As shown in the middle panel of figure 1.9, the onset and offset discontinuities are manifest as "spectral splatter," which can extend a long way up or down in frequency, and are therefore "audible" to our hypothetical 4-kHz cell.

This spectral splatter, which occurs if we cut a sound wave into arbitrary chunks, can also plague any attempt at spectrographic analysis. Imagine we want to analyze a so-called frequency-modulated sound. The whining of a siren that starts low and rises in pitch might be a good example. At any one moment in time this sound is a type of complex tone, but the fundamental shifts upward. To estimate the frequency content at any one time, cutting the sound into short pieces and calculating the spectrum for each may sound like a good idea, but if we are not careful, the cutting itself is likely to introduce discontinuities that will make the sound appear a lot more broadband than it really is. These cutting artifacts are hard to avoid completely, but some relatively simple tricks help alleviate them considerably. The most widely used trick is to avoid sharp cutoffs at the onset and offset of each window. Instead of rectangular windows, one uses ramped windows, which gently fade the sound on and off. The engineering mathematics literature contains numerous articles discussing the advantages and disadvantages of ramps with various shapes.

The bottom panels of figure 1.9 illustrate one popular type, the Hanning window, named after the mathematician who first proposed it. Comparing the spectra obtained with the rectangular window and the Hanning window, we see that the latter has managed to reduce the spectral splatter considerably. The peak around 1 kHz is perhaps still broader than we would like, given that in this example we started off with a pure 1-kHz sine, but at least we got rid of the ripples that extended for several kilohertz up the frequency axis. Appropriate "windowing" is clearly important if we want to develop techniques to estimate the frequency content of a sound. But, as we mentioned, the Hanning window shown here is only one of numerous choices. We could have chosen a Kaiser window, or a Hamming window, or simply a linear ramp with a relatively gentle slope. In each case, we would have got slightly different results, but, and this is the important bit, any of these would have been a considerable improve-

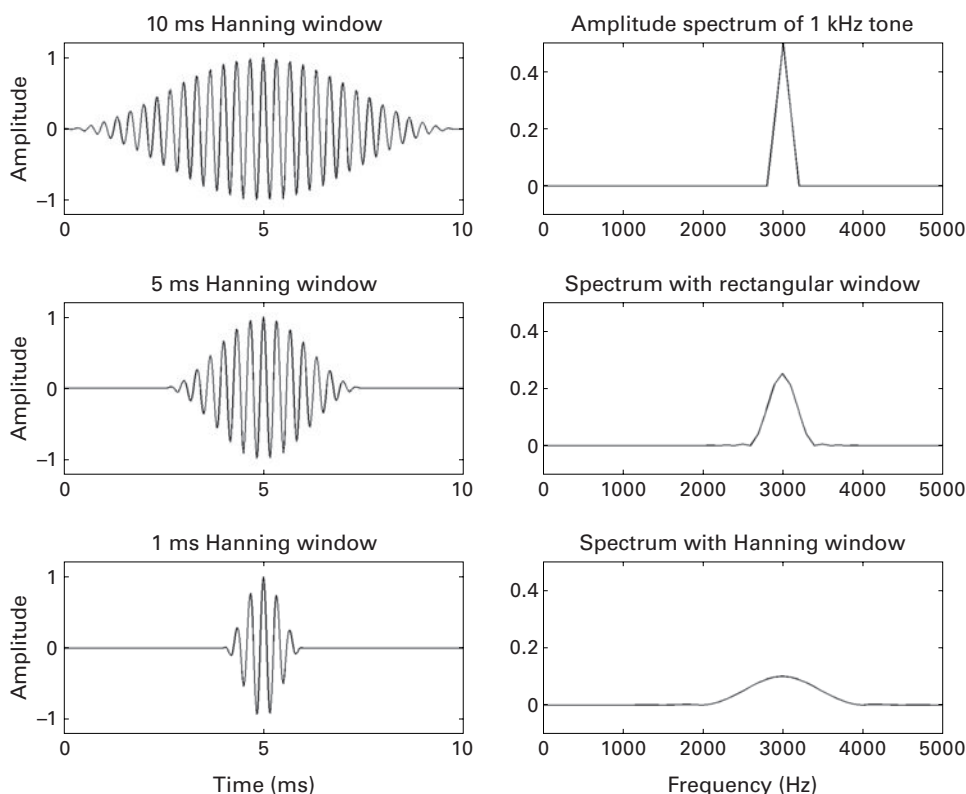
ment over the sharp onset of the rectangular window. The precise choice of window function is often a relatively minor detail, as long as the edges of the window consist of relatively gentle slopes rather than sharp edges.

The really important take-home message, the one thing you should remember from this section even if you forget everything else, is this: If we want to be able to resolve individual frequencies accurately, then we must avoid sharp onset and offset discontinuities. In other words, we must have sounds (or sound snippets created with a suitably chosen analysis window) that fade on and off gently—or go on forever, but that is rarely a practical alternative. Sharp, accurate frequency resolution requires gentle fade-in and fade-out, which in turn means that the time windows cannot be very short. This constraint relates to a more general problem, the so-called time-frequency trade-off.

Let us assume you want to know exactly when in some ongoing soundscape a frequency component of precisely 500Hz occurs. Taking on board what we have just said, you record the sound and cut it into (possibly overlapping) time windows for Fourier analysis, taking care to ramp each time window on and off gently. If you want the frequency resolution to be very high, allowing great spectral precision, then the windows have to be long, and that limits your *temporal* precision. Your frequency analysis might be able to tell you that a frequency very close to 500Hz occurred somewhere within one of your windows, but because each window has to be fairly long, and must have “fuzzy,” fading edges in time, determining exactly when the 500-Hz frequency component started has become difficult. You could, of course, make your window shorter, giving you greater temporal resolution. But that would reduce your frequency resolution. This time-frequency trade-off is illustrated in figure 1.10, which shows a 3-kHz tone windowed with a 10-ms, 5-ms, or 1-ms-wide Hanning window, along with the corresponding amplitude spectrum.

Clearly, the narrower the window gets in time, the greater the precision with which we can claim what we are looking at in figure 1.10 happens at time  $t = 5$  ms, and not before or after; but the spectral analysis produces a broader and broader peak, so it is increasingly less accurate to describe the signal as a 3-kHz pure tone rather than a mixture of frequencies around 3 kHz.

This time-frequency trade-off has practical consequences when we try to analyze sounds using spectrograms. Spectrograms, as mentioned earlier, slide a suitable window across a sound wave and calculate the Fourier spectrum in each window to estimate how the frequency content changes over time. Figure 1.11 shows this for the castanet sound we had already seen in figure 1.5 (p. 13). The spectrogram on the left was calculated with a very short, 2.5-ms-wide sliding Hanning window, that on the right with a much longer, 21-ms-wide window. The left spectrogram shows clearly that the sound started more or less exactly at time  $t = 0$ , but it gives limited frequency resolution. The right spectrogram, in contrast, shows the resonant frequencies of the



**Figure 1.10**

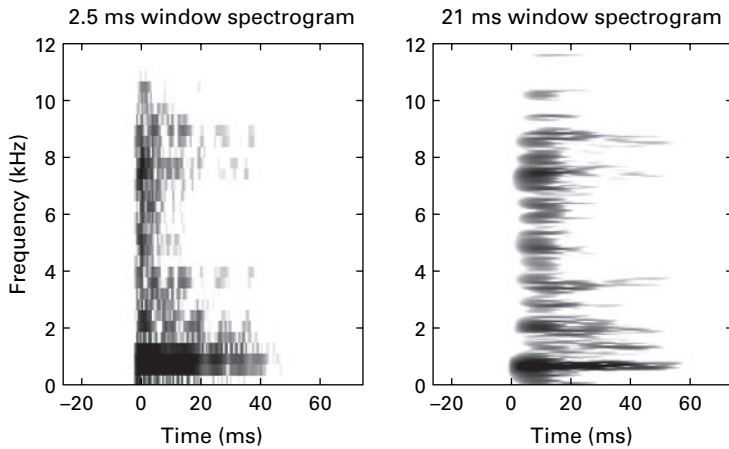
The time-frequency trade-off: Short temporal analysis windows give high temporal precision but poor spectral resolution.

castanet in considerably greater detail, but is much fuzzier about when exactly the sound started.

The trade-off between time and frequency resolution is not just a problem for artificial sound analysis systems. Your ears, too, would ideally like to have both very high temporal resolution, telling you exactly when a sound occurred, and very high frequency resolution, giving you a precise spectral fingerprint, which would help you identify the sound source. Your ears do, however, have one little advantage over artificial sound analysis systems based on windowed Fourier analysis spectrograms. They can perform what some signal engineers have come to call multiresolution analysis.

To get an intuitive understanding of what this means, let us put aside for the moment the definition of frequency as being synonymous with sine wave component,



**Figure 1.11**

Spectrograms of the castanet sound plotted in figure 1.5, calculated with either a short (left) or long (right) sliding Hanning window.

and instead return to the “commonsense” notion of a frequency as a measure of how often something happens during a given time period. Let us assume we chose a time period (our window) which is 1 s long, and in that period we counted ten events of interest (these could be crests of a sound wave, or sand grains falling through an hour glass, or whatever). We would be justified to argue that, since we observed ten events per second—not nine, and not eleven—the events happen with a frequency of 10 Hz. However, if we measure frequencies in this way, we would probably be unable to distinguish 10-Hz frequencies from 10.5-Hz, or even 10.9-Hz frequencies, as we cannot count half events or other event fractions. The 1-s analysis window gives us a frequency resolution of about 10% if we wanted to count events occurring at frequencies of around 10 Hz. But if the events of interest occurred at a higher rate, say 100 Hz, then the inaccuracy due to our inability to count fractional events would be only 1%. The precision with which we can estimate frequencies in a given, fixed time window is greater if the frequency we are trying to estimate is greater.

We could, of course, do more sophisticated things than merely count the number of events, perhaps measuring average time intervals between events for greater accuracy, but that would not change the fundamental fact that *for accurate frequency estimation, the analysis windows must be large compared to the period of the signal whose frequency we want to analyze*. Consequently, if we want to achieve a certain level of accuracy in our frequency analysis, we need very long time windows if the frequencies are likely to be very low, but we can get away with much shorter time windows if we

can expect the frequencies to be high. In standard spectrogram (short-term Fourier) analysis, we would have to choose a single time window, which is long enough to be suitable for the lowest frequencies of interest. But our ears work like a mechanical filter bank, and, as we shall see, they can operate using much shorter time windows when analyzing higher frequency sounds than when analyzing low ones. To get a proper understanding of how this works, we need to brush up our knowledge of filters and impulse responses.

## 1.5 Impulse Responses and Linear Filters

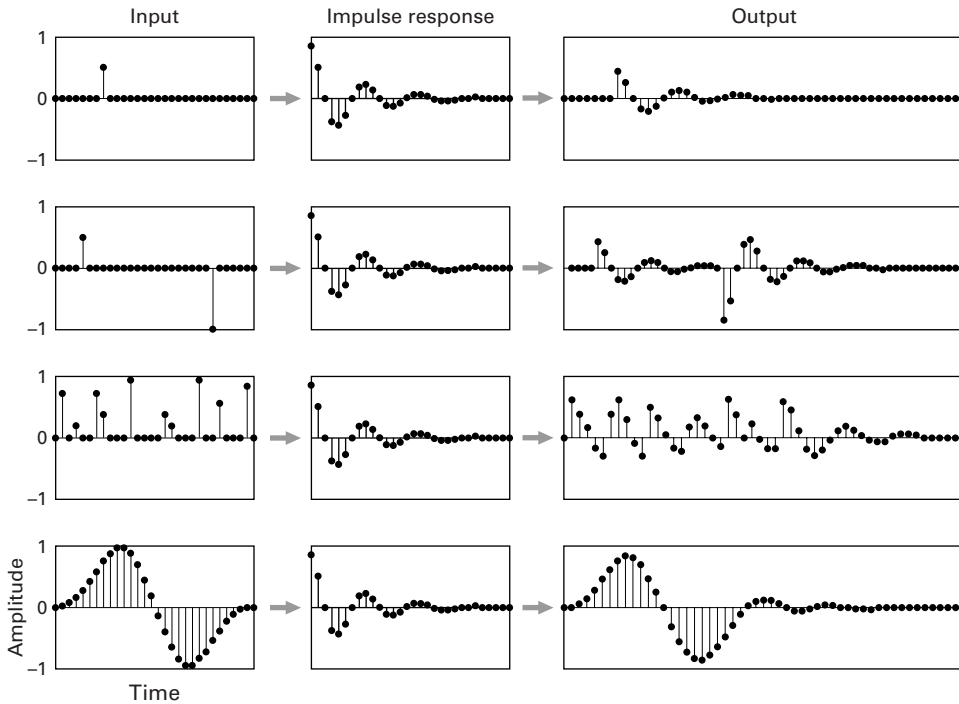
To think of an impulse as being made up of countless pure-tone frequency components, each of identical amplitude, as we have seen in figure 1.7, is somewhat strange, but it can be useful. Let us get back to the idea of a simple, solid object, like a bell, a piece of cutlery, or a piece of wood, being struck to produce a sound. In striking the object, we deliver an impulse: At the moment of impact, there is a very brief pulse of force. And, as we have seen in sections 1.1 and 1.2, the object responds to this force pulse by entering into vibrations. In a manner of speaking, when we strike the object we deliver to it all possible vibration frequencies simultaneously, in one go; the object responds to this by taking up some of these vibration frequencies, but it does not vibrate at all frequencies equally. Instead, it vibrates strongly only at its own resonance frequencies. Consequently, we can think of a struck bell or tuning fork as a sort of *mechanical frequency filter*. The input may contain all frequencies in equal measure, but only resonant frequencies come out. Frequencies that don't fit the object's mechanical tuning properties do not pass.

We have seen, in figure 1.5, that tuning forks, bells, and other similar objects are damped. If you strike them to make them vibrate, their impulse response is an exponentially decaying oscillation. The amplitude of the oscillation declines more or less rapidly (depending on the damping time constant), but in theory it should never decay all the way to zero. In practice, of course, the amplitude of the oscillations will soon become so small as to be effectively zero, perhaps no larger than random thermal motion and in any case too small to detect with any conceivable piece of equipment. Consequently, the physical behavior of these objects can be modeled with great accuracy by so-called *finite impulse response filters* (FIRs).<sup>3</sup> Their impulse responses are said to be finite because their ringing does not carry on for ever. FIRs are *linear systems*. Much scientific discussion has focused on whether, or to what extent, the ear and the auditory system themselves might usefully be thought of as a set of either mechanical or neural linear filters. Linearity and nonlinearity are therefore important notions that recur in later chapters, so we should spend a moment familiarizing ourselves with these ideas. The defining feature of a linear system is a *proportionality relationship* between input and output.

Let us return to our example of a guitar string: If you pluck a string twice as hard, it will respond by vibrating with twice the amplitude, and the sound it emits will be correspondingly louder, but it will otherwise sound much the same. Because of this proportionality between input and output, if you were to plot a graph of the force  $F$  with which you pluck the string against the amplitude  $A$  of the evoked vibration, you would get a *straight line graph*, hence the term “linear.” The graph would be described by the equation  $A = F \cdot p$ , where  $p$  is the proportionality factor which our linear system uses as it converts force input into vibration amplitude output. As you can hopefully appreciate from this, the math for linear systems is particularly nice, simple, and familiar, involving nothing more than elementary-school arithmetic. The corner shop where you bought sweets after school as a kid was a linear system. If you put twice as many pennies in, you got twice as many sweets out. Scientists, like most ordinary people, like to avoid complicated mathematics if they can, and therefore tend to like linear systems, and are grateful that Mother Nature arranged for so many natural laws to follow linear proportionality relationships. The elastic force exerted by a stretched spring is proportional to how far the spring is stretched, the current flowing through an ohmic resistor is proportional to the voltage, the rate at which liquid leaks out of a hole at the bottom of a barrel is proportional to the size of the hole, the amplitude of the sound pressure in a sound wave is proportional to the amplitude of the vibration of the sound source, and so on.

In the case of FIR filters, we can think of the entire impulse response as a sort of extension of the notion of proportional scaling in a way that is worth considering a little further. If we measure the impulse response (i.e., the ping) that a glass of water makes when it is tapped lightly with a spoon, we can predict quite easily and accurately what sound it will make if it is hit again, only 30% harder. It will produce very much the same impulse response, only scaled up by 30%. There is an important caveat, however: *Most things in nature are only approximately linear, over a limited range of inputs.* Strike the same water glass very hard with a hammer, and instead of getting a greatly scaled up but otherwise identical version of the previous ping impulse response, you are likely to get a rather different, crunch and shatter sort of sound, possibly with a bit of a splashing mixed in if the glass wasn’t empty. Nevertheless, over a reasonably wide range of inputs, and to a pretty good precision, we can think of a water glass in front of us as a linear system. Therefore, to a good first-order approximation, if we know the glass’ impulse response, we know all there is to know about the glass, at least as far as our ears are concerned.

The impulse response will allow us to predict what the glass will sound like in many different situations, not just if it is struck with a spoon, but also, for example, if it was rubbed with the bow of a violin, or hit by hail. To see how that works, look at figure 1.12, which schematically illustrates impulses and impulse responses. The middle panels show a “typical” impulse response of a resonant object, that is, an



**Figure 1.12**

Outputs (right panels) that result when a finite impulse response filter (middle panels) is excited with a number of different inputs (left panels).

exponentially decaying oscillation. To keep the figure easy to read, we chose a very short impulse response (of a heavily damped object) and show the inputs, impulse response, and the outputs as discrete, digitized, or sampled signals, just as in figure 1.8.

In the top row of figure 1.12, we see what happens when we deliver a small, slightly delayed impulse to that filter (say a gentle tap with a spoon on the side of a glass). After our recent discussion of impulse responses and linearity, you should not find it surprising that the “output,” the evoked vibration pattern, is simply a scaled, delayed copy of the impulse response function. In the second row, we deliver two impulses to the same object (we strike it twice in succession), but the second impulse, which happens after a slightly larger delay, is delivered from the other side, that is, the force is delivered in a negative direction. The output is simply a *superposition* of two impulse responses, each scaled and delayed appropriately, the second being “turned upside-down” because it is scaled by a negative number (representing a force acting in the opposite direction). This should be fairly intuitive: Hit a little bell twice in quick

succession, and you get two pings, the second following, and possibly overlapping with, the first.

In the second row example, the delay between the two pulses is long enough for the first impulse response to die down to almost nothing before the second response starts, and it is easy to recognize the output as a superposition of scaled and delayed-impulse responses. But impulses may, of course, come thick and fast, causing impulse responses to overlap substantially in the resulting superposition, as in the example in the third row. This shows what might happen if the filter is excited by a form of “shot noise,” like a series of hailstones of different weights, raining down on a bell at rapid but random intervals. Although it is no longer immediately obvious, the output is still simply a superposition of lots of copies of the impulse response, each scaled and delayed appropriately according to each impulse in the input. You may notice that, in this example, the output looks fairly periodic, with positive and negative values alternating every eight samples or so, even though the input is rather random (noisy) and has no obvious periodicity at eight samples. The periodicity of the output does, of course, reflect the fact that the impulse response, a damped sine wave with a period of eight, is itself strongly periodic. Put differently, since the period of the output is eight samples long, this FIR filter has a resonant frequency which is eight times slower than (i.e., one-eighth of) the sample frequency. If we were to excite such an FIR filter with two impulses spaced four samples apart, then the output, the superposition of two copies of the impulse response starting four samples apart, would be subject to destructive interference, the peaks in the second impulse response are canceled to some extent by the troughs in the first, which reduces the overall output.

If, however, the input contains two impulses exactly eight samples apart, then the output would benefit from constructive interference as the two copies of the impulse response are superimposed with peak aligned with peak. If the input contains impulses at various, random intervals, as in the third row of figure 1.12, then the constructive interference will act to amplify the effect of impulses that happen to be eight samples apart, while destructive interference will cancel out the effect of features in the input that are four samples apart; in this manner, the filter selects out intervals that match its own resonant frequency. Thus, hailstones raining down on a concrete floor (which lacks a clear resonance frequency) will sound like noise, whereas the same hailstones raining down on a bell will produce a ringing sound at the bell’s resonant frequency. The bell selectively amplifies (filters) its own resonant frequencies out of the frequency mixture present in the hailstorm input.

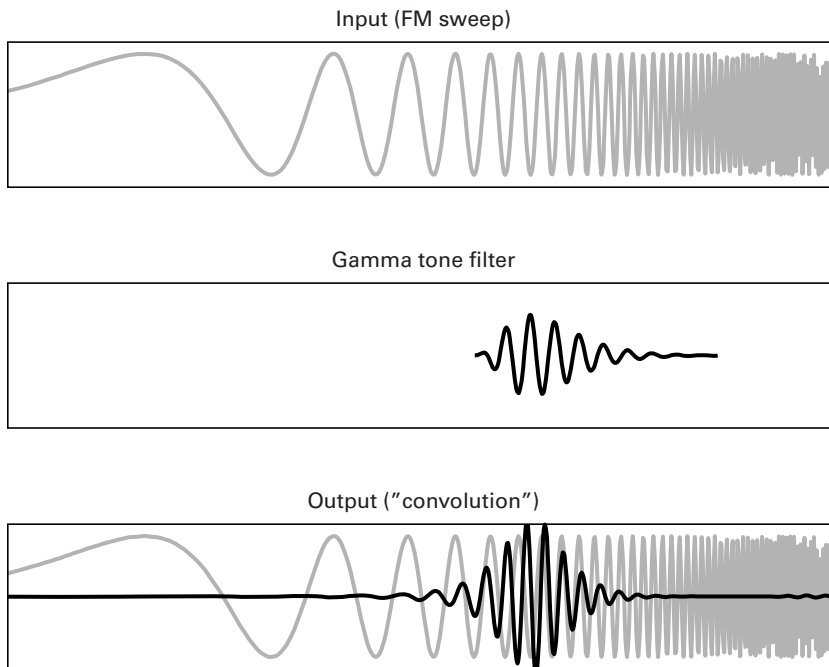
In the fourth row of figure 1.12, we consider one final important case. Here, instead of delivering a series of isolated impulses to the filter, we give it a sine wave cycle as input. This is a bit like, instead of striking a water glass with a spoon, we were to push a vibrating tuning fork against it. The onset and offset of the sine wave were smoothed off with a Hanning window. The frequency of the sine wave of the input

(one-twentieth of the sample rate) is quite a bit slower than the resonant frequency of the filter (one-eighth of the sample rate). What is hopefully quite obvious is that the output is again a sine whose frequency closely matches that of the input.

This illustrates a very important property of linear systems. Linear filters cannot introduce new frequency components; they can only scale each frequency component of the input up or down, and change its phase by introducing delays. Linear systems are therefore said to be sine wave in—sine wave out. Conversely, if we observe that a particular system responds to sine inputs with sine outputs that match the input frequency, then we would take that as an indication that the system is linear. And this holds true even if the input is a mixture of several sine waves. In that case, the output of the linear filter will also be a frequency mixture, and the relative amplitudes of the various frequency components may have changed dramatically, but there will be no components at frequencies that were not present in the input. (Nonlinear filters, in contrast, will quite commonly introduce frequencies into the output signal that weren't there in the input!)

Because we had ramped the sine wave input on and off gently with a Hanning window, the frequency content of this sine cycle is narrow, and contains very little energy at frequencies besides one-twentieth of the sample rate. This input therefore cannot excite the resonant frequency of the linear filter, and, unlike in the hailstones example in the third row, oscillations with a period of eight samples are not apparent in the output. The impulse response of the filter itself, however, has a very sharp onset, and this sharp onset makes its frequency response somewhat broadband. The input frequency of one-twentieth of the sample rate can therefore pass through the filter, but it does lose some of its amplitude since it poorly matches the filter's resonant frequency. If the impulse response of the filter had a gentler onset, its frequency response might well be narrower, and it would attenuate (i.e., reduce the amplitude of) frequencies that are further from its own resonant frequency more strongly.

In fact, filters with suitably chosen impulse responses can become quite highly selective for particular frequencies. We illustrate this in figure 1.13, which shows what happens when a frequency-modulated signal, a so-called FM sweep, is filtered through a so-called gamma-tone filter. A gamma-tone is simply a sinusoid that is windowed (i.e., ramped on and off) with a gamma function. The only thing we need to know about gamma functions for the purposes of this book is that they can take the shape of a type of skewed bell, with a fairly gentle rise and an even gentler decay. Gamma-tone filters are of some interest to hearing researchers because, as we will see in chapter 2, suitably chosen gamma-tone filters may provide a quite reasonable first-order approximation to the mechanical filtering with which the cochlea of your inner ear analyzes sound. So, if we filter a signal like an FM sweep with a gamma-tone filter, in a manner of speaking, we see the sound through the eyes of a point on the basilar

**Figure 1.13**

An FM sweep filtered by a gamma-tone filter.

membrane of your inner ear (more about that in the next chapter, and apologies for the mixed metaphor).

When we say the input to the filter is frequency modulated, we simply mean that its frequency changes over time. In the example shown in the top panel of figure 1.13, the frequency starts off low but then increases. We made the signal in this example by computer, but you might encounter such frequency modulation in real-world sounds, for example, if you plucked the string on a guitar and then, while the string is still vibrating, either run your finger down the string along the fret board, making the vibrating part of the string effectively shorter, or if you wind up the tension on the string, increasing its effective stiffness.

When we compare the gamma-tone filter in figure 1.13 with the wave-form of the FM sweep, it should be reasonably obvious that the resonant frequency of the gamma-tone filter matches the frequency of the FM sweep in some places, but not in others. In fact, the match between these frequencies starts off and ends up very poor (the frequency of the FM sweep is initially far too low and eventually far too high), but somewhere just over halfway through the frequency of the FM sweep matches that of

the filter rather well. And because the gamma-tone is gently ramped on and off, we expect its frequency bandwidth to be quite narrow, so as the FM sweep is filtered through the gamma-tone filter, we expect to see very little in the output at times where the frequencies do not match. These expectations are fully borne out in the third panel of figure 1.13, which shows the output of the gamma-tone filter, plotted as a thick black line superimposed on the original FM sweep input plotted in gray. Frequencies other than those that match the filter's resonance characteristics are strongly suppressed.

It is not difficult to imagine that, if we had a whole series of such gamma-tone filters, each tuned to a slightly different frequency and arranged in order, we could use the resulting gamma-tone filter bank to carry out a detailed frequency analysis of incoming sounds and calculate something very much like a spectrogram on the fly simply by passing the sounds through all the filters in parallel as they come in. As it happens, we are all equipped with such filter banks. We call them “ears,” and we will look at their functional organization in greater detail in the next chapter. But first we need to add a few brief technical notes to conclude this section, and we shall see how what we have just learned about filters and impulse responses can help us understand voices. We also need to say a few things about the propagation of sound waves through air.


First, the technical notes. Calculating the output of an FIR filter by superimposing a series of scaled and delayed copies of the impulse response is often referred to as calculating the *convolution* of the input and the impulse responses. Computing convolutions is sometimes also referred to as *convolving*. If you look up “convolution,” you will most likely be offered a simple mathematical formula by way of definition, something along the lines of “the convolution  $(f * g)(t)$  of input  $f$  with impulse response  $g$  equals  $\sum(g(t - \tau) \cdot f(\tau))$  over all  $\tau$ .” This is really nothing but mathematical shorthand for the process we have described graphically in figure 1.12. The  $g(t - \tau)$  bit simply means we take copies of the impulse response, each delayed by a different delay  $\tau$  (tau), and we then scale each of these delayed copies by the value of the input that corresponds to that delay [that's the “ $\cdot f(\tau)$ ” bit], and then superimpose all these scaled and delayed copies on top of one another, that is, we sum them all up (that's what the “ $\sum$  over all  $\tau$ ” means).

One thing that might be worth mentioning in passing is that convolutions are commutative, meaning that if we convolve a waveform  $f$  with another waveform  $g$ , it does not matter which is the input and which is the impulse response. They are interchangeable and swapping them around would give us the same result:  $(f * g)(t) = (g * f)(t)$ .

Another thing worth mentioning is that making computers calculate convolutions is quite straightforward, and given that so many real-world phenomena can be quite adequately approximated by a linear system and therefore described by impulse




responses, convolution by computer allows us to simulate all sorts of phenomena or situations. Suppose, for example, you wanted to produce a radio crime drama, and it so happens that, according to the scriptwriter, the story line absolutely must culminate in a satanic mass that quickly degenerates into a violent shootout, all taking place right around the altar of the highly reverberant acoustic environment of Oxford's Christ Church cathedral. To ensure that it sounds authentic, you asked the Dean of Christ Church for permission to record the final scene inside the cathedral, but somehow he fails to be convinced of the artistic merit of your production, and declines to give you permission. But recorded in a conventional studio, the scene sounds flat. So what do you do?

Well, acoustically speaking, Christ Church cathedral is just another linear system, with reverberations, echoes, and resonances that can easily be captured entirely by its impulse response. All you need to do is make one decent "impulse" inside the cathedral. Visit the cathedral at a time when few visitors are around, clap your hands together hard and, using a portable recorder with a decent microphone, record the sound that is produced, with all its reverberation. Then use a computer to convolve the studio recorded drama with the canned Christ Church impulse response, and presto, the entire scene will sound as if it was recorded inside the cathedral. Well, almost. The cathedral's impulse response will vary depending on the location of both the sound receiver and the source—the clapping hands—so if you want to create a completely accurate simulation of acoustic scenes that take place in, or are heard from, a variety of positions around the cathedral, you may need to record separate impulse responses for each combination of listener and sound source position. But passable simulations of reverberant acoustic environments are possible even with entirely artificial impulse responses derived on computer models. (See the book's Web site,  [auditoryneuroscience.com](http://auditoryneuroscience.com), for a demonstration.)

Or imagine you wanted to make an electronic instrument, a keyboard that can simulate sounds of other instruments including the violin. You could, of course, simply record all sorts of notes played on the violin and, when the musician presses the key on the keyboard, the keyboard retrieves the corresponding note from memory and plays it. The problem with that is that you do not know in advance for how long the musician might want to hold the key. If you rub the bow of a violin across a string it pulls the string along a little, then the string jumps, then it gets pulled along some more, then it jumps again a little, and so on, producing a very rapid and somewhat irregular saw-tooth force input pattern. Such a saw-tooth pattern would not be difficult to create by computer on the fly, and it can then be convolved with the strings' impulse responses, that is, sounds of the strings when they were plucked, to simulate the sound of bowed strings. Of course, in this way you could also simulate what it would sound like if objects were "bowed" that one cannot normally bow, but for which one can either record or simulate impulse responses: church bells, plastic

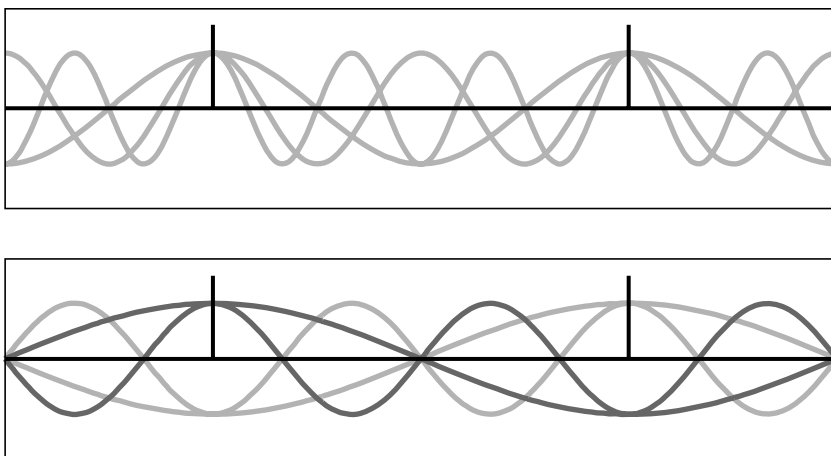
bottles, pieces of furniture, and so on. Convolution is thus a surprisingly versatile little tool for the computer literate who enjoy being scientifically or artistically creative.

## 1.6 Voices

One very widespread class of natural sounds, which is much easier to understand now that we have discussed impulse responses and convolutions, is voices. Given their central role in spoken words, song, and animal communication, voices are a particularly important class of sounds, and like many other sounds that we have discussed already, voices too are a kind of pulse-resonance sound. When humans or other mammals vocalize, they tighten a pair of tissue flaps known as vocal folds across their airways (the larynx), and then exhale to push air through the closed vocal folds. The vocal folds respond by snapping open and shut repeatedly in quick succession, producing a series of rapid clicks known as “glottal pulses.” These glottal pulses then “ring” through a series of resonant cavities, the vocal tract, which includes the throat, the mouth, and the nasal sinuses. (Look for a video showing human vocal folds in  action on the book’s Web site.) When we speak, we thus effectively convolve a glottal pulse train with the resonant filter provided by our vocal tract, and we can change our voice, in rather different and interesting ways, either by changing the glottal pulse train or by changing the vocal tract.

First, let us consider the glottal pulse train, and let us assume for simplicity that we can approximate this as a series of “proper” impulses at very regular intervals. Recall from figure 1.7 that any impulse can be thought of as a superposition of infinitely many sine waves, all of the same amplitude but different frequency, and with their phases arranged in such a way that they all come into phase at the time of the impulse but at no other time. What happens with all these sine waves if we deal not with one click, but with a series of clicks, spaced at regular intervals?

Well, each click is effectively its own manifestation of infinitely many sine waves, but if we have more than one click, the sine waves of the individual clicks in the click train will start to interfere, and that interference can be constructive or destructive. In fact, the sine components of the two clicks will have the same phase if the click interval happens to be an integer (whole number) multiple of the sine wave period, in other words if exactly one or two or three or  $n$  periods of the sine wave fit between the clicks. The top panel of figure 1.14 may help you appreciate that fact. Since these sine waves are present, and in phase, in all clicks of a regular click train of a fixed interval, they will interfere constructively and be prominent in the spectrum of the click train. However, if the click interval is  $1/2$ , or  $3/2$ , or  $5/2$ , ... of the sine period, then the sine components from each click will be exactly out of phase, as is shown in the bottom panel of figure 1.14, and the sine components cancel out.



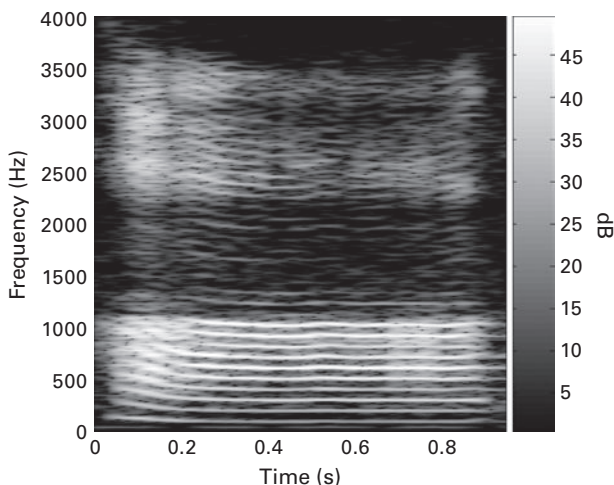
**Figure 1.14**

Sine components of click trains interfere constructively when the sine period is an integer multiple of the click interval, but not otherwise.

The upshot of this is that a regular, periodic click train is effectively a type of sound that we have already encountered in section 1.2, namely, a type of complex tone, made up of a fundamental frequency and an infinite number of higher harmonics, all of equal amplitude. The periods of the harmonics are equal to  $1/1$ ,  $1/2$ ,  $1/3$ , ...  $1/n$  of the click interval, and their frequencies are accordingly multiples of the fundamental. Consequently, the closer the clicks are spaced in time, the wider apart the harmonics are in frequency.

All of this applies to glottal pulse trains, and we should therefore not be surprised that the spectrogram of voiced speech sounds exhibits many pronounced harmonics, reaching up to rather high frequencies. But these glottal pulse trains then travel through the resonators of the vocal tract. Because the resonant cavities of the throat, mouth, and nose are linear filters, they will not introduce any new frequencies, but they will raise the amplitude of some of the harmonics and suppress others.

With this in mind, consider the spectrogram of the vowel /a/, spoken by a male adult, shown in figure 1.15. The spectrogram was generated with a long time window, to resolve the harmonics created by the glottal pulse train, and the harmonics are clearly visible as spectral lines, every 120Hz or so. However, some of the harmonics are much more pronounced than others. The harmonics near 700 to 1,200Hz are particularly intense, while those between 1,400 and 2,000Hz are markedly weaker, and then we see another peak at around 2,500Hz, and another at around 3,500Hz. To a first approximation, we can think of each of the resonators of the vocal tract as a band pass filter with a single resonant frequency. These resonant frequencies are

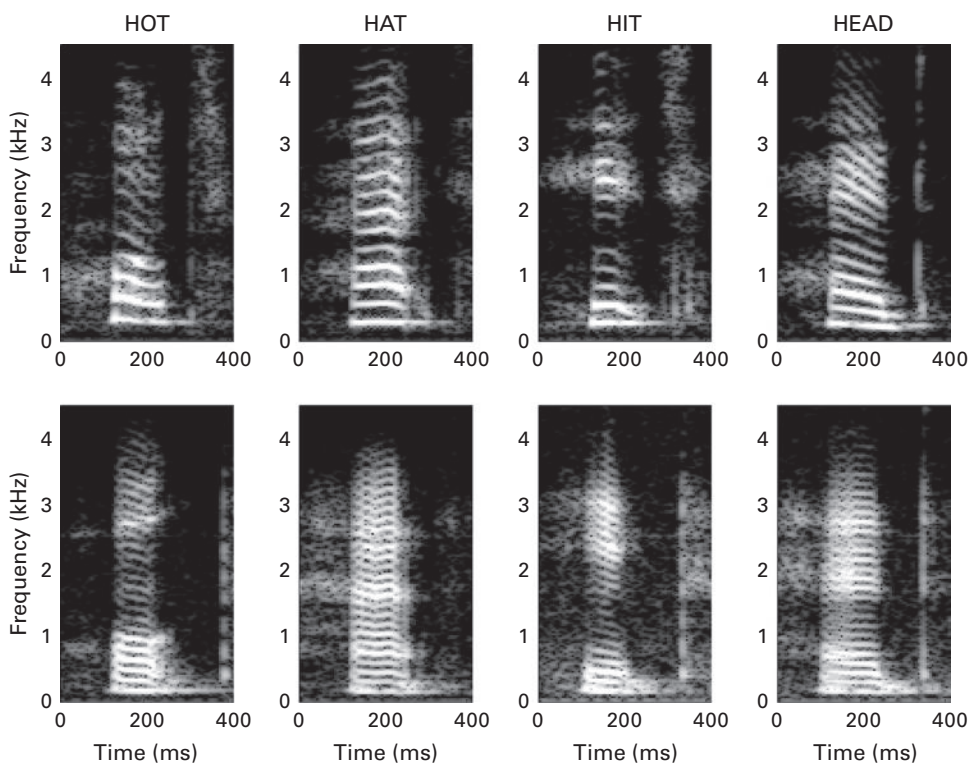


**Figure 1.15**  
Spectrogram of the vowel “a.”

known as “formants,” and those harmonics that lie close to the formant frequencies will be scaled up relative to the others, which leads to the peaks we have just observed.

One important feature of our voice is that it gives us independent control of the harmonic and the formant frequencies. We change the harmonic frequencies by putting more or less tension on our vocal folds. The higher the tension, the faster the glottal pulse train, which leads to a higher fundamental frequency and more widely spaced harmonics, and the voice is perceived as higher pitched. We control the formant frequencies by moving various parts of our vocal tract, which are commonly referred to as “articulators,” and include the lips, jaws, tongue, and soft palate. Moving the articulators changes the size and shape of the resonance cavities in the vocal tract, which in turn changes their resonant frequencies, that is, the formants.

Altering the formants does not affect the pitch of the speech sound, but its timbre, and therefore its “type” may change quite dramatically; for example, we switch between /o/- and /a/-like vowels simply by widening the opening of our lips and jaw. Thus, we control which vowel we produce by changing the formant frequencies, and we control the pitch at which we speak or sing a vowel by changing the harmonic frequencies. This can be seen quite clearly in figure 1.16, which shows spectrograms of the words “hot,” “hat,” “hit,” and “head,” spoken by different native speakers of British English, one with a high-pitched, childlike voice (top row) and then again in a lower-pitched voice of an adult female (bottom row). (A color version of that figure, along with the corresponding sound recordings, can be found on the “vocalizations and speech” section of the book’s Web site.)



**Figure 1.16**

Spectrograms of the words “hot,” “hat,” “hit,” and “head,” spoken in a high-pitched (top row) and a low-pitched (bottom row) voice.

The vowels in the spoken words are readily apparent in the “harmonic stacks” that mark the arrival of the glottal pulse train, and the spacing of the harmonics is clearly much wider in the high-pitched than in the low-pitched vowels. The fact that, in some of the sounds, the harmonics aren’t exactly horizontal tells us that the pitch of these vowels was not perfectly steady (this is particularly noticeable in the high-pitched “hot” and “head”). Where exactly the formants are in these vowels is perhaps not quite so readily apparent to the unaided eye. (Formants are, in fact, easier to see in spectrograms that have short time windows, and hence a more blurred frequency resolution, but then the harmonics become hard to appreciate.) But it is nevertheless quite clear that, for example, in the /i/ sounds the harmonics around 500Hz are very prominent, but there is little sound energy between 800 and 2000Hz, until another peak is reached at about 2,300Hz, and another near 3,000Hz. In contrast, in the /a/ vowels, the energy is much more evenly distributed across the frequency range, with peaks

perhaps around 800, 1,800, and 2,500Hz, while the /o/ sound has a lot of energy at a few hundred hertz, up to about 1,100Hz, then much less until one reaches another smaller peak at 2,500Hz.

Of course, there is more to the words “hot,” “hat,” “hit,” and “head” than merely their vowels. There are also the consonants “h” at the beginning and “t” or “d” at the end. Consonants can be subdivided into different classes depending on a number of criteria. For example, consonants can be “voiced” or “unvoiced”. If they are voiced, the vocal folds are moving and we would expect to see harmonics in their spectrogram. The “h” and the “t” in “hot,” “hat,” “hit,” and “heat” are unvoiced, the vocal chords are still. So what generates the sounds of these unvoiced consonants? Most commonly, unvoiced consonants are generated when air is squeezed through a narrow opening in the airways, producing a highly turbulent flow, which sets up random and therefore noisy vibration patterns in the air.

In speaking the consonant “h,” the air “rubs” as it squeezes through a narrow opening in the throat. Students of phonetics, the science of speech sounds, would therefore describe this sound as a glottal (i.e., throat produced) fricative (i.e., rubbing sound), in this manner describing the place and the mode of articulation. The “t” is produced by the tongue, but unlike in pronouncing the “h,” the airway is at first blocked and then the air is released suddenly, producing a sharp sound onset characteristic of a “plosive” stop consonant. Plosives can be, for example, “labial,” that is, produced by the lips as in “p” or “b”; or they can be “laminal-dental,” that is, produced when the tip of the tongue, the lamina, obstructs and then releases the airflow at the level of the teeth, as in “t” or “d”; or they can be velar, that is, produced at the back of the mouth when the back of the tongue pushes against the soft palate, or velum, as in “k.”



The mouth and throat area contains numerous highly mobile parts that can be reconfigured in countless different ways, so the number of possible speech sounds is rather large. (See the “vocalization and speech” section of the book’s Web site for links to x-ray videos showing the human articulators in action.) Cataloguing all the different speech sounds is a science in itself, known as articulatory phonetics. We shall not dwell on this any further here, except perhaps to point out a little detail in figure 1.16, which you may have already noticed. The “h” fricative at the beginning of each word is, as we have just described, the result of turbulent airflow, hence noisy, hence broad-band; that is, it should contain a very wide range of frequencies. But the frequencies of this consonant are subjected to resonances in the vocal tract just as much as the harmonics in the vowel. And, indeed, if you look carefully at the place occupied by the “h” sounds in the spectrograms of figure 1.16 (the region just preceding the vowel), you can see that the “h” sound clearly exhibits formants, but these aren’t so much the formants of the consonant “h” as the formants of the vowel that is about to follow! When we pronounce “hot” or “hat,” our vocal tract already assumes the configuration

of the upcoming vowel during the “h,” imparting the formants of the following vowel onto the preceding consonant. Consequently, there is, strictly speaking, really no such thing as the sound of the consonant “h,” because the “h” followed by an “o” has really quite a different spectrum from that of an “h” that is followed by an “a.”

This sort of influence of the following speech sound onto the preceding sound is sometimes referred to as “coarticulation” or “assimilation” in the phonetic sciences, and it is much more common than you might think. In your personal experience, an “h” is an “h”; you know how to make it, you know what it sounds like. Recognizing an “h” is trivially easy for your auditory system, despite the fact that, in reality, there are many different “h” sounds. This just serves to remind us that there is a lot more to hearing than merely accurately estimating frequency content, and for people trying to build artificial speech recognizers, such phenomena as coarticulation can be a bit of a headache.

## 1.7 Sound Propagation

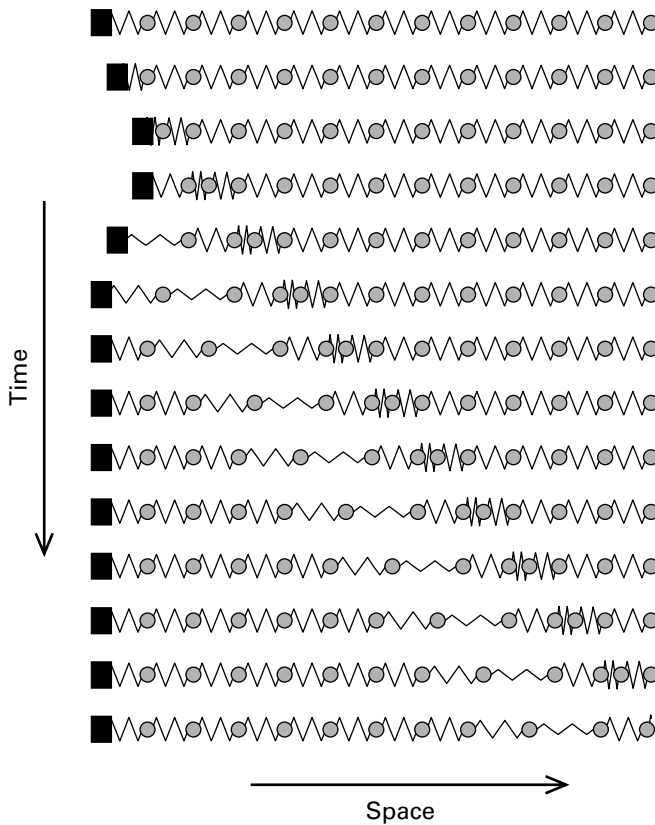
So far, we have looked almost exclusively at vibration patterns in a variety of sound sources. Developing some insights into how sounds come about in the first place is an essential, and often neglected, aspect of the auditory sciences. But, of course, this is only the beginning. We can hear the sound sources in our environment only if their vibrations are somehow physically coupled to the vibration-sensitive parts of our ears. Most commonly, this coupling occurs through the air.

Air is capable of transmitting sound because it has two essential properties: inert mass and stiffness or elasticity. You may not think of air as either particularly massive or particularly elastic, but you probably do know that air does weigh something (about 1.2g/L at standard atmospheric pressure), and its elasticity you can easily verify if you block off the end of a bicycle pump and then push down the piston. As the air becomes compressed, it will start to push back against the piston, just as if it were a spring. We can imagine the air that surrounds us as being made up of small “air masses,” each linked to its neighboring masses through spring forces that are related to air pressure. This is a useful way of conceptualizing air, because it can help us develop a clear image of how sound waves propagate. A medium made of small masses linked by springs will allow disturbances (e.g., small displacements at one of the edges) to travel through the medium from one mass to the next in a longitudinal wave pattern. How this works is illustrated in figure 1.17, as well as in a little computer animation which you can



find on the book’s Web site.

Figure 1.17 shows the output of a computer simulation of a sound source, represented by the black rectangle to the left, which is in contact with an air column, represented by a row of air masses (gray circles) linked by springs (zigzag lines). At first (top row), the source and the air masses are at rest, but then the sound source



**Figure 1.17**

A longitudinal wave propagating through a medium of inert masses coupled via elastic springs. See the book's Web site for an animated version.

briefly lurches a bit to the right, and then returns to its initial position. The rightward movement compresses the spring that links it to the neighboring air mass. That, in turn, causes this air mass to be accelerated rightwards, but because the mass has a small inertia, its movement lags somewhat behind that of the sound source. As the mass immediately to the right of the sound source starts moving rightward, it compresses the spring linking it to the next mass along, which in turn accelerates that mass rightward, and so on and so on. As the sound source returns to its original position, it will start to stretch the spring linking it to the neighboring mass, which in time will pull that mass back to its original position. This stretch also propagates along the air column; thus, while each air mass is first pushed one way it is then pulled the other, so that it ends up where it started. There are thus essentially four phases to the propagating sound wave: first, a compression, followed by a forward displacement, followed by a rarefaction (or stretch, in our spring analogy), followed by a backward



(return) displacement. Sound waves therefore consist of displacement waves as well as pressure waves, but the displacement wave lags behind the pressure wave by a quarter of a cycle, and the final net displacement of the air is zero. (Sound is not a wind).

To generate figure 1.17, we got a computer to apply Hooke's law to work out all the spring forces, and then to use Newton's second law to calculate the acceleration of each inert air mass in turn and update the positions of the masses accordingly at each time step. Figure 1.17 illustrates quite nicely how the simple interaction of Hooke's law and Newton's second law allows for a movement of a sound source to be translated into a "disturbance" in the local air pressure (the spring forces), which then propagates through a column of air away from the source. It is essentially that mechanism which links the surfaces of all the sound sources in your environment to your eardrum. And in doing so, it replicates the vibration patterns of the sound sources quite accurately. The forward movement of the source gave rise to a compression of corresponding size, while the return to the original position created a corresponding rarefaction. This simple correspondence is handy, because it means that we can, to a pretty good first approximation, assume that everything we have learned in previous sections about the vibration patterns of different types of sound sources will be faithfully transmitted to the eardrum.

Of course, this phenomenon of longitudinal propagation of waves is not limited to air. Any material that possesses inert mass and at least some springlike elasticity is, in principle, capable of propagating sound waves. Consequently, dolphins can use their lower jaws to pick up sound waves propagated through water, while certain species of toads, or even blind mole rats, appear to be able to collect sound waves conducted through the ground. But different substrates for sound propagation may be more or less stiff, or more or less heavy, than air, and these differences will affect the speed at which sound waves travel through that substrate. We mentioned earlier that the displacement of the first air mass in figure 1.17 lags somewhat behind the displacement of the sound source itself. The movement of the sound source first has to compress the spring, and the spring force then has to overcome the inertia of the first mass. Clearly, if the spring forces are quite large and the masses only small, then the time lag will not be very great.

Materials with such comparatively large internal spring forces are said to have a high acoustic impedance. The acoustic impedance is defined as the amount of pressure (i.e., spring force) required to achieve a given particle velocity. The speed of sound is high in materials that have a high acoustic impedance but a low density (essentially high spring forces and low masses). For air at ambient pressures and temperature, the speed of sound works out to about 343 m/s (equivalent to 1,235 km/h or 767 mph). The precise value depends on factors such as humidity, atmospheric pressure, and temperature, because these may affect the mass density and the elastic modulus (i.e., the springiness) of the air. For water, which is heavier but also much, much

stiffer than air, the speed of sound is more than four times larger, at about 1,480 m/s (5,329 km/h or 3,320 mph).

In a number of ways, the longitudinal wave propagation shown in figure 1.17 is, of course, a somewhat oversimplified model of real sound waves. For example, in our idealized model, sound waves propagate without any loss of energy over infinite distances. But you know from experience that sounds are quieter if the sound source is further away. Two factors contribute to this. The first, and often the less important reason, is that in real physical media, sound waves tend to dissipate, which simply means that the coordinated motion of air masses will start to become disorganized and messy, gradually looking less and less like a coherent sound wave and increasingly like random motion; that is, the sound will slowly turn into heat. Interestingly, high-frequency sounds tend to dissipate more easily than low-frequency ones, so that thunder heard from a great distance will sound like a low rumble, even though the same sound closer to the source would have been more of a sharp “crack,” with plenty of high-frequency energy. Similarly, ultrasonic communication and echolocation sounds such as those used by mice or bats tend to have quite limited ranges. A mouse would find it difficult to call out to a mate some twenty yards away, something that most humans can do with ease.

The second, and usually major factor contributing to the attenuation (weakening) of sounds with distance stems from the fact that most sound sources are not linked to just a single column of spring-linked masses, as in the example in figure 1.17, but are instead surrounded by air. We really should think more in terms of a three-dimensional lattice of masses with spring forces acting up and down, forward and backward, as well as left and right. And as we mentioned earlier, the springs in our model represent air pressure gradients, and you may recall that pressure will push from the point where the pressure is greater *in all directions* where pressure is lower. Consequently, in a three-dimensional substrate, a sound wave that starts at some point source will propagate outward in a circular fashion in all directions. You want to



imagine a wavefront a bit like that in figure 1.17, but forming a spherical shell that moves outward from the source.

Now, this spherical shell of the propagating sound wave does, of course, get larger and larger as the sound propagates, just like the skin of a balloon gets larger as we continue to inflate it. However, all the mechanical energy present in the sound wave was imparted on it at the beginning, when the sound source accelerated the air masses in its immediate neighborhood. As the sound wave propagates outward in a sphere, that initial amount of mechanical energy gets stretched out over a larger and larger area, much like the skin of a balloon becomes thinner as we inflate it. If the area becomes larger as we move further away from the sound source but the energy is constant, then the amount of energy per unit area must decrease. And since the surface area of a sphere is proportional to the square of its radius (surface =  $4 \cdot \pi \cdot r^2$ ), the

sound energy per unit surface area declines at a rate that is inversely proportional to the square of the distance from the source. This fact is often referred to as the *inverse square law*, and it is responsible for the fact that sound sources normally are less loud as we move further away from them.

Of course, the inverse square law, strictly speaking, only holds in the “free field,” that is, in places where the sound wave really can propagate out in a sphere in all directions. If you send sound waves down a narrow tube with rigid walls, then you end up with a situation much more like that in figure 1.17, where you are dealing with just a column of air in which the sound amplitude should stay constant and not decline with distance. You may think that this is a severe limitation, because in the real world there are almost always some surfaces that may be an obstacle to sound propagation, for example, the floor! However, if you placed a sound source on a rigid floor, then the sound waves would propagate in hemispheres outwards (i.e., sideways and upward), and the surface area of a hemisphere is still proportional to the square of its radius (the surface is simply half of  $4\pi r^2$ , hence still proportional to  $r^2$ ), so the inverse square law would still apply.

If, however, we lived in a world with four spatial dimensions, then sound energy would drop off as a function of  $1/r^3$ —which would be known as the inverse cube law—and because sound waves would have more directions in which they must spread out, their amplitudes would decrease much faster with distance, so communicating with sound over appreciable distances would require very powerful sound sources. Of course, we cannot go into a world of four spatial dimensions to do the experiment, but we can send sound down “effectively one-dimensional” (i.e., long but thin) tubes. If we do this, then the sound amplitude should decrease as a function of  $1/r^0 = 1$ , that is, not at all, and indeed people in the 1800s were able to use long tubes with funnels on either end to transmit voices, for example, from the command bridge to the lower decks of a large ship.

Thus, the inverse square law does not apply for sounds traveling in confined spaces, and substantial deviations from the inverse square law are to be expected also in many modern indoor environments. Although three-dimensional, such environments nevertheless contain many sound-reflecting surfaces, including walls and ceilings, which will cause the sound waves to bounce back and forth, creating a complex pattern of overlapping echoes known as reverberations. In such reverberant environments, the sound that travels directly from the source to the receiver will obey the inverse square law, but reflected sound waves will soon add to this original sound.

## 1.8 Sound Intensity

From our previous discussion of propagating sound waves, you probably appreciate the dual nature of sound: A small displacement of air causes a local change in pressure,

which in turn causes a displacement, and so on. But if we wanted to measure how large a sound is, should we concern ourselves with the amplitude of the displacement, or its velocity, or the amplitude of the pressure change, or all three? Well, the displacement and the pressure are linked by linear laws of motion, so if we know one, we ought to be able to work out the other. The microphones used to measure sounds typically translate pressure into voltage, making it possible to read the change of pressure over time directly off an oscilloscope screen. Consequently, acoustical measures of sound amplitude usually concern themselves with the sound pressure only. And you might think that, accordingly, the appropriate thing to do would be to report sound amplitudes simply in units of pressure, that is, force per unit area. While this is essentially correct, matters are, unfortunately, a little more complicated.

The first complication we have to deal with is that it is in the very nature of sounds that the sound pressure always changes, which is very inconvenient if we want to come up with a single number to describe the intensity of a sound. We could, of course, simply use the largest (peak) pressure in our sound wave and report that. Peak measurements are sometimes used, and are perfectly fine if the sound we are trying to characterize is a pure sinusoid, for example, because we can infer all the other amplitudes if we know the peak amplitude. But for other types of sound, peak amplitude measures can be quite inappropriate. Consider, for example, two click trains, each made up of identical brief clicks; but in the first click train the interval between clicks is long, and in the second it is much shorter. Because the clicks are the same in each train, the peak amplitudes for the two trains are identical, but the click train with the longer inter-click intervals contains longer silent periods, and it would not be unreasonable to think it therefore has, in a manner of speaking, less sound in it than the more rapid click train.

As this example illustrates, it is often more appropriate to use measures that somehow average the sound pressure over time. But because sounds are normally made up of alternating compressions and rarefactions (positive and negative pressures), our averaging operation must avoid canceling positive against negative pressure values. The way this is most commonly done is to calculate the root-mean-square (RMS) pressure of the sound wave.

As the name implies, RMS values are calculated by first squaring the pressure at each moment in time, then averaging the squared values, and finally taking the square root. Because the square of a negative value is positive, rarefactions of the air are not canceled against compressions when RMS values are calculated. For sine waves, the RMS value should work out as  $1/\sqrt{2}$  (70.71%) of the peak value, but that is true only if the averaging for the RMS value is done over a time period that contains a whole number of half-cycles of the sine wave. In general, the values obtained with any averaging procedure will be, to some extent, sensitive to the choice of time window over

which the averaging is done, and the choice of the most appropriate time window will depend on the particular situation and may not always be obvious.

Another factor that can cause great confusion among newcomers to the study of sound (and sometimes even among experts) is that even RMS sound pressure values are almost never reported in units of pressure, like pascals (newtons per square meter) or in bars, but are instead normally reported in bels, or more commonly in tenths of bels, known as decibels (dB). Belts are in many ways a very different beast from the units of measurement that we are most familiar with, like the meter, or the kilogram, or the second. For starters, unlike these other units, a bel is a *logarithmic* unit. What is that supposed to mean?

Well, if we give, for example, the length of a corridor as 7.5 m, then we effectively say that it is 7.5 *times* as long as a well-defined standard reference length known as a meter. If, however, we report an RMS sound pressure as 4 bels (or 40 dB), we're saying that it is 4 *orders of magnitude* (i.e., 4 powers of 10 =  $10^4 = 10,000$  times) larger than some standard reference pressure. Many newcomers to acoustics will find this orders of magnitude thinking unfamiliar and at times a little inconvenient. For example, if we add a 20-kg weight to a 40-kg weight, we get a total weight of 60 kg. But if we add, in phase, a 1-kHz pure tone with an RMS sound pressure amplitude of 20 dB to another 1-kHz pure tone with an amplitude of 40 dB, then we do not end up with a tone with a 60-dB amplitude. Instead, the resulting sound pressure would be  $\log_{10}(10^4 + 10^2) = 4.00432$  bels, that is, 40.0432 dB. The weaker of the two sounds, having an amplitude 2 orders of magnitude (i.e., 100 times) smaller than the larger one, has added, in terms of orders of magnitude, almost nothing to the larger sound.

Because decibels, unlike weights or lengths or money, work on a logarithmic scale, adding decibels is a lot more like multiplying sound amplitudes than adding to them, and that takes some getting used to. But the logarithmic nature of decibels is not their only source of potential confusion. Another stems from the fact that decibels are used to express all sorts of logarithmic ratios, not just ratios of RMS sound pressure amplitudes. Unlike meters, which can only be used to measure lengths, and always compare these lengths to a uniquely and unambiguously defined standard length, for decibels there is no uniquely defined type of measurement, nor is there a unique, universally accepted standard reference value. Decibel values simply make an order of magnitude comparison between any two quantities, and it is important to be clear about what is being compared to what.

Decibel values need not relate to sound at all. To say that the sun is about 25.8 dB (i.e.,  $10^{2.58}$ , or roughly 380 times) further away from the earth than the moon is would perhaps be a little unusual, but entirely correct. But surely, at least in the context of sound, can't we safely assume that any decibels we encounter will refer to sound pressure? Well, no. Sometimes they will refer to the RMS pressure, sometimes to peak pressure, but more commonly, acoustical measurements in decibels will refer to the

*intensity* or the *level* of a sound. In a context of acoustics, the words “level” and “intensity” can be used interchangeably, and an intensity or level stated in decibels is used, in effect, to compare the *power* (i.e., the energy per unit time) per unit area delivered by each of the two sound waves.

Fortunately, there is a rather simple relationship between the energy of a sound and its RMS sound pressure amplitude, so everything we learned so far about pressure amplitudes will still be useful. You may remember from high school physics that the kinetic energy of a moving heavy object is given by  $E = mv^2$ , that is, the kinetic energy is proportional to the square of the velocity. This also holds for the kinetic energy of our notional lumps of air, which we had encountered in figure 1.17 and which take part in a longitudinal wave motion to propagate sound. Now, the average velocity of these lumps of air is proportional to the RMS sound amplitude, so the energy levels of the sound are proportional to the *square* of the amplitude. Consequently, if we wanted to work out the intensity  $y$  of a particular sound with RMS pressure  $x$  in decibels relative to that of another known reference sound whose RMS pressure is  $x_{ref}$ , then we would do this using the formula

$$y \text{ (dB)} = 10 \cdot \log_{10}(x^2/x_{ref}^2) = 20 \cdot \log_{10}(x/x_{ref})$$

(The factor of 10 arises because there are 10dB in a bel. And because  $\log(a^2) = 2 \cdot \log(a)$ , we can bring the squares in the fraction forward and turn the factor 10 into a factor 20).

You might be forgiven for wondering whether this is not all a bit overcomplicated. If we can use a microphone to measure sound pressure directly, then why should we go to the trouble of first expressing the observed pressure amplitudes as multiples of some reference, then calculating the log to base 10 of that fraction, then finally multiply by 20 to work out a decibel sound intensity value. Would it not be much easier and potentially less confusing to simply state the observed sound pressures amplitudes directly in pascals? After all, our familiar, linear units like the meter and the newton and the gram, for which  $4 + 2 = 6$ , and not 4.00432, have much to commend themselves. So why have the seemingly more awkward and confusing logarithmic measures in bels and decibels ever caught on?

Well, it turns out that, for the purposes of studying auditory perception, the orders of magnitude thinking that comes with logarithmic units is actually rather appropriate for a number of reasons. For starters, the range of sound pressure levels that our ears can respond to is quite simply enormous. Hundreds of millions of years of evolution during which you get eaten if you can't hear the hungry predators trying to creep up on you have equipped us with ears that are simply staggeringly sensitive. The faintest sound wave that a normal, young healthy human can just about hear has an RMS sound pressure of roughly 20 micropascal ( $\mu\text{Pa}$ ), that is, 20 millionth of a newton per square meter. That is approximately 10 million times less than the pressure of a

penny resting on your fingertip! Because an amplitude of  $20\mu\text{Pa}$  is close to the absolute threshold of human hearing, it is commonly used as a reference for sound intensity calculations. Sound levels expressed in decibels relative to  $20\mu\text{Pa}$  are usually abbreviated as *dB SPL*, short for sound pressure level.

The sound level of the quietest audible sounds would therefore be around 0 dB SPL. (These almost inaudibly quiet sounds have an amplitude approximately equal to that of the reference of  $20\mu\text{Pa}$ , so the fraction  $x/x_{\text{ref}}$  in the formula above would work out to about 1, and the log of 1 is 0.) But if you listen, for example, to very loud rock music, then you might expose your ears to sound levels as high as 120 dB SPL. In other words, the sound energy levels in these very loud sounds are twelve orders of magnitude—1,000,000,000,000-fold (1,000 *billion* times!) larger than those in the quietest audible sounds. Thousand billion-fold increases are well beyond the common experience of most of us, and are therefore not easy to imagine.

Let us try to put this in perspective. A large rice grain may weigh in the order of 0.04 g. If you were to increase that tiny weight a thousand billion-fold, you would end up with a mountain of rice weighing 50,000 tons. That is roughly the weight of ten thousand fully grown African elephants, or one enormous, city-block-sized ocean cruise liner like the *Titanic*. Listening to very loud music is therefore quite a lot like taking a delicate set of scales designed to weigh individual rice grains, and piling one hundred fully loaded jumbo jet airliners onto it. It may sound like fun, but it is not a good idea, as exposure to very loud music, or other very intense sounds for that matter, can easily lead to serious and permanent damage to the supremely delicate mechanics in our inner ears. Just a single, brief exposure to 120 dB SPL sounds may be enough to cause irreparable damage.

Of course, when we liken the acoustic energy at a rock concert to the mass of one hundred jumbo jets, we don't want to give the impression that the amounts of energy entering your eardrums are large. Power amplifiers for rock concerts or public address systems do radiate a lot of power, but only a minuscule fraction of that energy enters the ears of the audience. Even painfully loud sound levels of 120 dB SPL carry really only quite modest amounts of acoustic power, about one-tenth of a milliwatt (mW) per square centimeter. A human eardrum happens to be roughly half a square centimeter in cross section, so deafeningly loud sounds will impart 0.05 mW to it. How much is 0.05 mW? Imagine a small garden snail, weighing about 3 g, climbing vertically up a flower stalk. If that snail can put 0.05 mW of power into its ascent, then it will be able to climb at the, even for a snail, fairly moderate pace of roughly 1.5 mm every second. "Snail-power," when delivered as sound directly to our eardrums, is therefore amply sufficient to produce sounds that we would perceive as deafeningly, painfully loud. If the snail in our example could only propel itself with the power equivalent to that delivered to your eardrum by the very weakest audible sounds, a power 12 orders of magnitude smaller, then it would take the snail over two



thousand years to climb just a single millimeter! To be able to respond to such unimaginably small quantities of kinetic energy, our ears indeed have to be almost unbelievably sensitive.

When we are dealing with such potentially very large differences in acoustic energy levels, working with orders of magnitude, in a logarithmic decibel scale, rather than with linear units of sound pressure or intensity, keeps the numbers manageably small. But working with decibels brings further, perhaps more important advantages, as it also more directly reflects the way we subjectively perceive sound intensities or loudness. Our ability to detect changes in the amplitude of a sound is governed by Weber's law—at least to a good approximation.<sup>4</sup> Weber's law states that we can detect changes in a particular quantity, like the intensity of a sound, or the weight of a bag or the length of a pencil, only if that quantity changed by more than a given, fixed percentage, the so-called Weber fraction. For broadband noises of an intensity greater than 30dB SPL, the Weber fraction is about 10%, that is, the intensity has to increase by at least 10% for us to be able to notice the difference (Miller, 1947).

If the increase required to be able to perceive the change is a fixed proportion of the value we already have, then we might expect that the perceived magnitude might be linked to physical size in an exponential manner. This exponential relationship between physical intensity of a sound and perceived loudness has indeed been confirmed experimentally, and is known as Stevens's law (Stevens, 1972). Perceived loudness is, of course, a subjective measure, and varies to some extent from one individual to another, and for reasons that will become clearer when we consider mechanisms of sound capture and transduction by the ear in the next chapter, the relationship between perceived loudness and the physical intensity of a sound will also depend on its frequency content. Nevertheless, for typical listeners exposed to typical sounds, a growth in sound intensity by 10dB corresponds approximately to a doubling in perceived loudness. Describing sound intensity in terms of decibels, thus, appears to relate better or more directly to how we subjectively perceive a sound than describing it in terms of sound pressure amplitude. But it is important to note that the link between the physical intensity of a sound and its perceptual qualities, like its perceived loudness, is not always straightforward, and there are a number of complicating factors. To deal with at least a few of them, and to arrive at decibel measures that are more directly related to human auditory performance, several other decibel measures were introduced in addition to dB SPL, which we have already encountered. Some of these, which are quite widely used in the literature, are dBA and dB HL, where HL stands for hearing level.

Let us first look at dBA. As you are probably well aware, the human ear is more sensitive to some frequencies than others. Some sounds, commonly referred to as ultrasounds, with a frequency content well above 20kHz, we cannot hear at all, although other species of animals, for example, dogs, bats, or dolphins, may be able



to hear them quite well. Similarly, certain very low (infrasound) frequencies, below 20 Hz, are also imperceptible to us. We tend to be most sensitive to sounds with frequencies between roughly 1 and 4 kHz. For frequencies much below 1 kHz or well above 4 kHz, our sensitivity declines, and when we reach frequencies either below 20 Hz or above 20 kHz, our sensitivity effectively shrinks to nothing. The function that maps out our sensitivity is known as an audiogram, which is effectively a U-shaped curve with maximal sensitivity (lowest detection thresholds) at frequencies between 1 and 4 kHz. The reason that our ears are more sensitive to some frequencies than others seems to stem from mechanical limitations of the outer and middle ear structures whose job it is to transmit sounds from the outside world to the inner ear. We will look at this in more detail in the next chapter.


One consequence of this U-shaped sensitivity curve is that it introduces a massive frequency dependence in the relationship between the acoustic intensity of a sound and its perceived loudness. A 120-dB SPL pure tone of 1 kHz would be painfully loud, and pose a serious threat of permanent damage to your hearing, while a 120-dB SPL pure tone of 30 kHz would be completely inaudible to you, and would also be much safer. When we try to use physical measures of the intensity of ambient sounds to decide how likely a sound is to cause a nuisance or even a health hazard, we need to take this frequency dependence into account. Noise measurements are therefore usually performed with an “A-weighting-filter,” a band-pass filter with a transfer function that approximates the shape of the human audiogram and suppresses high and low frequencies at a rate proportional to the decline in human sensitivity for those frequencies. Determining sound intensity in dBA is therefore equivalent to determining dB SPL, except that the sound is first passed through an A-filter.

The link between physical energy, perceived loudness, and potential to cause noise damage is not straightforward. The use of A-weighting filters is only one of many possible approaches to this problem, and not necessarily the best. Other filter functions have been proposed, which go, perhaps unsurprisingly, by names such as B, C, and D, but also, less predictably, by names like ITU-R 468. Each of these possible weighting functions has its own rationale, and may be more appropriate for some purposes than for others; those who need to measure noise professionally may wish to consult the recent Industrial Standards Organization document ISO 226:2003 for further details. Although A-weighting may not always be the most appropriate method, it remains very commonly used, probably because it has been around for the longest time, and almost all commercially available noise level meters will have A-weighting filters built in.

Like dBA, dB HL also tries to take typical human frequency sensitivity into account, but unlike dBA, it is a clinical, not a physical measure: dB HL measurements are not used to describe sounds, but to describe people. When it is suspected that a patient may have a hearing problem, he or she is commonly sent to have a clinical audiogram

test performed. During this test, the patient is seated in a soundproof booth and asked to detect weak pure tones of varying frequencies delivered over headphones.

The measured perceptual thresholds are then expressed as sensitivity relative to the threshold expected in normal, young healthy humans. Thus, a patient with normal hearing will have a sensitivity of 0 dB HL. A result of 10 dB HL at a particular frequency means that the patient requires a sound 10 dB more intense than the average young, healthy listener to detect the sound reliably. The patient's detection threshold is elevated by 10 dB relative to what is "normal." In contrast, patients with exceptionally acute hearing may achieve negative dB HL values. A value of +10 dB HL, though slightly elevated, would still be considered within the normal range. In fact, only threshold increases greater than 20 dB would be classified as hearing loss. Since the normal sound sensitivity range covers 12 orders of magnitude, and because sounds with intensities of 20 dBA or less are terribly quiet, losing sensitivity to the bottom 20 dB seems to make little difference to most people's ability to function in the modern world.

Generally, values between 20 and 40 dB HL are considered diagnostic of mild hearing loss, while 40 to 60 dB HL would indicate moderate, 60 to 90 dB HL severe, and more than 90 dB HL profound hearing loss. Note that hearing levels are usually measured at a number of pure tone frequencies, common clinical practice is to proceed  in "octave steps" (i.e., successive frequency doublings) from 125 or 250 Hz to about 8 kHz, and patients may show quite different sensitivities at different frequencies.

Conductive hearing loss, that is, a loss of sensitivity due to a mechanical blockage in the outer or middle ear, tends to present as a mild to moderate loss across the whole frequency range. In contrast, sensorineural hearing loss is most commonly caused by damage to the delicate sensory hair cells in the inner ear, which we discuss in the next chapter, and it is not uncommon for such sensorineural losses to affect the sensitivity to high frequencies much more than to low frequencies. Thus, elderly patients often have mild to moderate losses at 8 kHz but normal sensitivity at frequencies below a few kilohertz, and patients who have suffered noise damage frequently present with focal losses of sensitivity to frequencies around 4 kHz. Why some frequency ranges are more easily damaged than others may become clearer when we study the workings of the inner ear, which is the subject of the next chapter.