trical forces on each other. When the two objects have like charges they repel each other; when they carry opposite charges they attract each other. The deflection of an electron stream in a TV set or a laboratory oscilloscope provides a practical example of electrical forces in action.

In addition to electric forces between charges at rest there are magnetic forces between moving charges. We will not define magnetism but will state some qualitative features about magnetic forces. When an electric current exists in a straight wire or in a loop of wire, a magnetic field is produced. A magnetic field is capable of exerting a magnetic force on any moving charge in its vicinity. (A permanent magnet may be thought of as due to many tiny atomic current loops that have all been oriented in the same direction to produce an effective "internal" current loop in the magnet.)

When a wire coil carrying a current is placed in a magnetic field it may be caused to move because of the magnetic force on its moving electrons; the greater the current the larger the force. Conversely, when a wire coil in a magnetic field is moved, current may be produced because of the magnetic force on the coil's electrons; the larger the motion, the greater the current. Electric motors, dynamic loudspeakers, and dynamic microphones are based on this principle.

When a wire loop is placed in a changing magnetic field a current is induced in the coil. This is the principle on which an electric generator is based.

3.6 Work

A detailed application of the concepts discussed earlier in this chapter would be too laborious to be useful at this point in your studies. The concepts of work, energy, and power may provide alternative descriptions for most things we find of interest in the physical aspects of music, speech, and audio. These new quantities can be defined in terms of the previously defined quantities, but in many ways they are handier to use.

In our everyday life we often evaluate the difficulty of a task in terms of the work involved. If a friend lifts a heavy object for you, he may complain that he is working too hard. If you push with all your strength against an immovable wall, you might claim that you are working equally hard. An observing scientist, although sympathetic to the effort you are exerting, would suggest that your friend is working, and you are not. The scientific definition of work takes into account the force exerted and the distance an object moves when that force is applied. The force must cause displacement of the object or the work done is zero. The **work** done by a force is the product of the force times the displacement in the direction of the force. Symbolically,

$$\textit{work} = \textit{force} \times \textit{displacement (in direction of force)} \qquad (3.8)$$

Because you exerted a force on a wall which did not move, no work was performed. When your friend lifted the heavy object, however, the force exerted caused the object to be displaced upward and work was done. If we double the load that your friend lifts, we double the work done. Likewise, if the load is lifted twice as high, the work is doubled. The unit of measurement in which work is expressed is the product of distance and force units. In the metric system, the unit of work is the **joule** (J), defined as a newton-meter. One joule (rhymes with pool) of work results when a force of one newton displaces an object a distance of one meter in the direction of the force.

3.7 Mechanical Energy

"Energy" is perhaps the most fundamental unifying concept in all scientific disciplines. Yet, despite the prevalence of the term, it is an abstract concept which cannot be simply defined. You use "human energy" to turn the ignition key when you start a car. The ignition key engages the battery which converts chemical energy to electrical energy. The electrical energy from the battery is used in the starter to produce the mechanical energy which turns the flywheel which in turn starts the engine. The engine then converts chemical energy from fuel into heat energy. The heat energy turns into mechanical energy which propels the car. If it is dark you turn on the car's headlights, which convert electrical energy from the battery into light energy. If an animal runs in front of your car you honk the horn (converting electrical energy into sound energy) and step on the brakes (converting mechanical energy into heat energy). The basic points to remember in this hypothetical exercise are that (1) energy in some form is involved in all our activities, (2) energy can appear in many different forms, and (3) energy can be changed from one form to another. We will begin our discussion by considering mechanical energy, one of the most recognizable forms of energy.

Consider a simple frictionless pendulum (or swing) as shown in Figure 3.1. If we do work by applying a force to the pendulum we can move it from its rest position B to position A. If the pendulum is now released from position A it will gain speed until it reaches B, then lose speed until it reaches C, and then return through B to A. If the pendulum is truly frictionless the motion will repeat itself indefinitely, reaching the same height each time at A and C and having the same speed at B. There is an implication here that something is conserved, but that "something" can be neither height nor speed because both change throughout the motion.

We define the conserved quantity as **total mechanical energy,** energy because of position and energy because

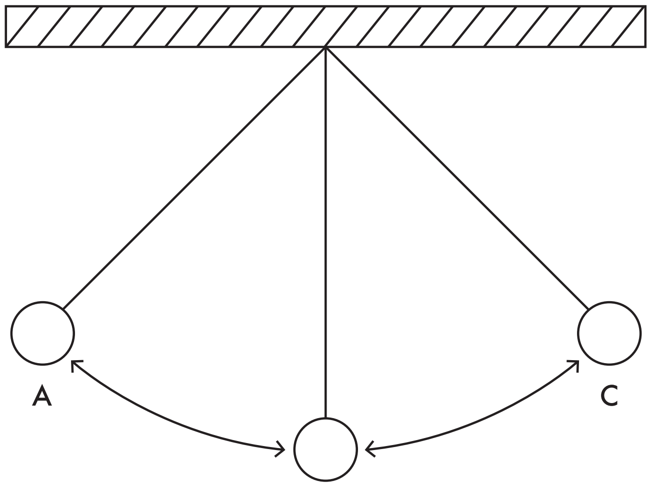

Figure 3.1 *Vibrating Pendulum*

of motion. The energy from position is termed **gravitational potential energy** expressed as

$$GPE = mgh = wh \tag{3.9}$$

where m is the mass of the pendulum, g is the gravitational constant, and h is the pendulum's height above its rest position. The GPE of the pendulum is just equal to the work done by a force (equal to the pendulum's weight, w, which is the product mg) in lifting the pendulum to height h above its rest position. The energy from motion is termed **kinetic energy**, expressed as

$$KE = \frac{mv^2}{2} \tag{3.10}$$

where m is the pendulum mass and v is its speed (magnitude of velocity) at any instant. The total mechanical energy of the pendulum is equal to the sum of KE and GPE and is conserved. In other words it is constant. As KE increases GPE decreases (and vice versa) so that their sum is always the same. As the pendulum goes from position A through position B to position C and back again, there is a continual transfer of energy from all potential (at A and C because the pendulum is not moving there) to all kinetic (at B where the speed is greatest). At the in-between points the energy is a combination of potential and kinetic, but at any point in the motion the sum of the potential and kinetic energy is exactly the same as at any other point.

Potential energy may also be explained with a stretched rubber band, a compressed spring, or a stretched drumhead. Potential energy associated with stretching or compressing objects is more relevant to our studies than is gravitational potential energy. For example, the **potential energy** of a spring is defined as

$$PE = \frac{sd^2}{2} \tag{3.11}$$

where s is a "stiffness" associated with the spring and d is the displacement magnitude of the spring from some rest position. In the preceding example of the pendulum the force required to lift the pendulum is constant (and equal to the weight of the pendulum). The force required to compress (or stretch) the spring, however, increases as the spring is compressed. The work done in compressing the spring is proportional to the displacement and to the force which in turn is proportional to the displacement. Hence, the work done in compressing the spring is proportional to the square of the displacement. Because the potential energy of the spring is just equal to the work done in compressing it, the potential energy of the spring is proportional to the square of the displacement. A mass attached to the spring will be accelerated and decelerated as the spring pushes and pulls on it. The kinetic energy of the mass at any instant will be

$$KE = \frac{mv^2}{2} \tag{3.12}$$

just as in the case of the pendulum.

The greater the amount of work done to raise a pendulum or to compress a spring, the greater its potential energy becomes. We have seen in the pendulum and spring examples that the potential energy in each case is just equal to the work done in raising the pendulum and in compressing the spring, respectively. We can then surmise that energy—potential, kinetic, or any other form—must be expressed in the same units as work, namely the joule.

Whenever a given physical quantity remains constant in a changing situation we generalize the result as a law. In this case the law states that the total mechanical energy (the sum of kinetic and potential energies) of a system remains constant when no frictional forces are present. In any real system there will always be some friction present and so this law is only an approximation of reality. However, it is still useful for analyzing vibrating systems, so long as its limitations are not forgotten.

3.8 Other Forms of Energy

What becomes of mechanical energy when friction is present? Consider the kinetic energy produced by a Girl Scout rubbing two sticks together. Where does this energy go? It turns into a different form of energy, one which we observe as heat in the sticks. Whenever mechanical energy disappears because of friction, **heat energy** appears. If we accurately measure the total heat energy produced and the total mechanical energy that disappears in a given situation, we would discover that the two amounts are equal; the mechanical energy lost equals the heat energy gained.

Exercises

3.1 Compute the average velocity in each of the following cases. An auto moves 60 m west in 100 s. A bicycle moves 1 cm in 1 s. A person walking moves -0.1 cm in 10^{-3} s.

3.2 Compute the average acceleration for each of the following cases. The speed of an auto changes 6000 cm/ s in 10 s. The speed of a bicycle changes -1.0 cm/s in 10^{-2} s.

3.3 Instantaneous velocity can be determined by taking a very small change in displacement (represented as Δd) and dividing it by the very small elapsed time (Δt) to give $v = d/\Delta t$. Assume that a vibrating string moves a distance of 0.001 cm in 0.000 1 s and calculate the instantaneous velocity. (The Δ notation is used to indicate a small change of a quantity such as time, distance, etc.)

3.4 The relationship among instantaneous velocity, displacement, and time can be written $v = \Delta d/\Delta t$. If Δd = 0.50 cm and Δt = 0.01 s, find the instantaneous velocity. If v = 100 cm/s and Δt = 0.05 s, find Δd. If v = 100 cm/s and Δd = 0.20 cm, find Δt.

3.5 The instantaneous acceleration of an object can be written: $a = \Delta v/\Delta t$. When Δv = 0.10 cm/s and Δt = 0.05 s, find a. When a = 6.0 cm/s^2 and Δt = 0.05 s, find Δv. When a = 5.0 cm/s^2 and Δv = 0.10 cm/s, find Δt.

3.6 Imagine that you are driving your car on a perfectly straight highway. Calculate your acceleration for each of the following situations. You increase your speed from 20 km/hr to 40 km/hr in 10 s. You decrease your speed from 40 km/hr to 20 km/hr in 5 s. You remain at a constant speed of 30 km/hr for 20 s.

3.7 Explain why spiked heels will puncture a hard floor while a normal heel will not. As an example, compare the pressure exerted by a 50 kg woman wearing 1 cm^2 heels to that of a 100 kg man wearing 50 cm^2 heels

3.8 The blowing pressure in a clarinet player's mouth is 2000 N/m^2. What force is exerted on a clarinet reed area of 1 cm^2 by the blowing pressure. (The reed area should be converted to m^2 before calculating the force.)

3.9 Air near sea level has a mass density of 1.3 kg/m^3 and a corresponding weight of 12.7 N/m^3. Although the density of the atmosphere decreases with altitude, the total amount of air is equivalent to a column of "sea level air" about 7900 meters high. What is the pressure of this air column (expressed in N/m^2 and Pa)? This pressure is referred to as atmospheric pressure, which as you know changes with altitude and with changing weather patterns.

3.10 How high a column of water would be required to produce a pressure equivalent to that of the atmosphere? (Water has a density of approximately 1000 kg/m^3.)

3.11 Calculate the potential energy of a spring with a spring constant of s = 500 N/m stretched 5 cm from its rest position.

3.12 If the spring in Exercise 3.11 is stretched in 0.5 s, what power is required?

3.13 How much work do you do in pushing your bicycle 100 m up a hill if a constant force of 10 N is required?

3.14 If the bicycle of Exercise 3.13 is pushed up the hill in 10 s, what power is required? How much power is required if 1000 s is taken?

3.15 If a clarinet reed is moved 1 mm under a force of 0.2 N, how much work is done? If the reed vibrates 500 times per second and the work is done during one-quarter of a vibration, what power is required?

3.16 Total energy of 500 J is received during a time of 10 s. What is the power? If the receiving surface has an area of 10 m^2, what is the intensity?

3.17 A microphone diaphragm has a diameter of 10 mm. What power (in watts) does it receive from a sound having an intensity of 10^{-5} W/m^2? How much energy does it receive in 25 s?

Activities

3.1 Demonstration of atmospheric pressure: Watch a can get crushed by the atmosphere as it is evacuated. Try to pull apart the two halves of an evacuated spherical shell.

3.2 Demonstration of energy conservation: Make a pendulum by suspending a large mass on a light string. Set it into motion. Observe the time required for the motion to cease. What becomes of the energy?

3.3 Repeat Demonstration 3.2 for a mass on a spring.

3.4 Demonstration of Bernoulli's law: Place a Ping-Pong ball inside a funnel through which air is streaming upward through the narrow end. The ball will not be blown up and out of the funnel. Now invert the funnel and the ball will be held in position rather than blown away by the air. The airflow around the ball is constricted and the pressure is consequently less than the atmospheric pressure below the ball. The ball is actually pushed into the region of lower pressure above it by the greater pressure below it, so that it remains supported in midair.

3.5 The Bernoulli force can be demonstrated by placing a small card (3 × 5) on a table. Place a thread spool on end and centered on the card. Use a tack pushed through the center of the card to keep it centered on the spool. Blow through the spool and lift it away from the table. The card will also come.

3.6 Get identical spherical balloons. Blow one up to twice the radius of the other. Paste stickers on the balloons to just cover their surfaces. How many times as many stickers are required to cover the large balloon? Total balloon material can be thought of as total sound power. The larger a balloon becomes as it is blown up the less balloon material there is per unit surface area—power per unit area or intensity.

mass-spring system as it varies in time (horizontal axis) is shown in Figure 4.4. The displacement (vertical axis) of a wave as it varies along a string (horizontal axis) is shown in Figure 6.6. This is just a “snapshot” at an instant as the wave travels long the string. If one were to observe a single point on the string, its displacement would vary in time and could be represented by a sinusoid similar to that of Figure 4.4.

6.6 *Energy in Waves*

The energy in waves comes in two forms with which we are familiar (see Chapter 3): KE associated with particle speeds and PE associated with deformations in the medium. When a force produces a transverse wave in a medium such as a string (or a membrane or a bar) the force does work on the medium which results in an increase of energy in the medium. The disturbance travels along the string, deforming and changing the speed of local sections of the string as it passes. The KE and PE associated with the disturbance travel along with the disturbance.

When a vibrating source such as a loudspeaker diaphragm pushes and pulls on the molecules in the surrounding air it does work on the molecules and so increases their energy. In some regions the velocity of the air molecules increases, and the KE density is also increased. (Refer to Chapter 3 for a discussion of energy density as energy per unit volume.) In other regions the air molecules are compressed into a smaller space resulting in an increased pressure with its associated increase in PE density. When a sound wave travels through the air any local region will experience changes in both KE density and PE density as the wave passes. The total energy density is just equal to the sum of the KE and PE densities and to the maximum of either. (This is analogous to the total energy of a simple vibrator being equal to the sum of its KE and PE or to the maximum of either.) Pressure is more easily measured than speed, so the measurement of energy density is most often in terms of PE density.

6.7 *Summary*

The oscillations of a vibrating object travel outward as waves through the surrounding medium. The properties of the medium, particularly the intermolecular forces, determine what wave types can propagate. Both transverse and longitudinal waves can travel in solids. Fluids can only carry longitudinal waves because the interactions between molecules in fluids are weaker than in solids. Both transverse and longitudinal waves can be represented by sinusoids, with the positive value of the sinusoid indicating a positive displacement or a positive pressure. The wave speed on a string is proportional to the square root of the tension-to-density ratio. The wave speed in gases is proportional to the square root of the pressure-to-density ratio. Wave speed is equal to the product of frequency and wavelength. The wave speed in a medium may be considered constant unless properties of the medium change. Waves carry potential and kinetic energy through a medium in which they travel.

References and Further Reading

French, A. P. (1971). *Vibrations and Waves, MIT Introductory Physics Series* (Norton). Chapter 7 provides a technical presentation of waves and pulses in matter.

Hall, D. E. (2002). *Musical Acoustics, 3rd ed* (Brooks/Cole). Chapter 1 provides a brief description of surface waves on water.

Rossing, T. D. and D. A. Russell (1990). "Laboratory Observation of Elastic Waves in Solids," *Am. J. Phys. 58*, 1153–1162.

Questions

6.1 Describe how an impulse travels through a solid and how the "internal" forces of a solid help to transmit the impulse. What keeps the solid from coming apart?

6.2 Describe how an impulse travels through a gas and how the “internal” forces of a gas help to transmit the impulse. What keeps the gas from coming apart?

6.3 What are several disturbances that travel in a gas? In a liquid? In a solid?

6.4 What materials (solid, liquid, gas) will transmit longitudinal waves? Why does a gas not transmit both kinds of waves?

6.5 If a transverse wave cannot be propagated through liquids, how do you explain water waves? What is the medium for these waves?

6.6 What are displacement, velocity, and acceleration for waves on a string?

6.7 What are displacement, velocity, and acceleration for waves in a gas?

6.8 When a clarinet is played, is the clarinet reed free or driven? Is the wave transverse or longitudinal? Does the wave travel in a solid or a gas?

6.9 Repeat Question 6.8 for the vocal folds and also for the vocal tract.

6.10 Repeat Question 6.8 for a piano string and for a bowed violin string.

6.11 Repeat Question 6.8 for an oboe.

6.12 Repeat Question 6.8 for a drumhead and for a chime.

Exercises

6.1 Equations 6.1 and 6.2 give wave speeds for waves in a string and waves in a gas, respectively. Explain how both give the same units of m/s or cm/s for wave speed.

6.2 A wave has a frequency of 500 Hz and a wavelength of 0.01 m. What is the speed of the wave?

6.3 Compute the wavelength of a wave with a frequency of 100 Hz and a speed of 1.0 m/s.

6.4 Given a speed of 10.0 m/s and a wavelength of 0.10 m, find the frequency.

6.5 The tension in a string is 0.1 N, and the string has a mass of 10^{-5} kg and a length of 1.0 m. What is the speed of waves in the string?

6.6 The ambient pressure in air is 10^5 Pa and the density of air is 1.3 kg/m^3. Calculate the speed of sound in these conditions.

6.7 Take the speed of sound in air to be 340 m/s. What is the wavelength in air if f = 340 Hz? What is the frequency when the wavelength is 0.10 m? What is the wave speed in helium if f = 1000 Hz and the wavelength is 0.97 m?

6.8 The velocity of sound in air is about 340 m/s. If the space between the Earth and the Moon were filled with air, how long would it take sound to travel from the Moon to the Earth (a distance of 4×10^6m)?

6.9 Calculate the speed of sound at the following temperatures: (a) 70'C, (b) 32'C, (c) 12'C, (d) 0° C, and (e) 20° C.

6.10 Players bring their trombones in from the cold (0° C) outdoors and sound a note without warming up. What is the wave speed in their instruments under this condition? After warming up, the air in their instruments has a temperature of 35° C. What is the wave speed in the warmed-up condition? If they sounded a frequency of 110 Hz while cold, what frequency will they sound after warming up, assuming the wavelength is the same in both temperature conditions?

6.11 An adult male vocal tract from larynx to mouth is approximately 17 cm in length . How long does it take a sound wave to travel from larynx to mouth if the temperature is 35° C?

6.12 How long does it take a sound wave to travel from the stage to the back of a small auditorium (a distance of 20 m) if the air temperature is 25° C?

6.13 Musical instrument wavelengths remain almost the same but their frequencies change with wave speed. Frequency changes of 0.5% may be important for musical purposes. What temperature change will produce a frequency change of this amount?

6.14 A woofer in a loudspeaker system has a diameter of 40 cm. What frequency corresponds to a wavelength equal to the loudspeaker diameter? Answer the same question for a midrange diaphragm diameter of 8 cm.

6.15 Determine the range of wavelengths for audible sound if the range of frequencies is 20–20,000 Hz.

6.16 A tuning fork sounds a frequency of 440 Hz. What is the wavelength of the resulting sound in air if the wave speed is 35,000 cm/s?

6.17 A trombone produces a sound with a wavelength of 100 cm. What frequency is it sounding if the air temperature is 25° C?

6.18 A symphony broadcast originating in Boston travels west at the speed of light (300,000 km/s) to a listener 4000 km distant. How long does it take the symphony sound to reach the listener? How long does it take the symphony sound within the concert hall to reach a listener 25 m away from the orchestra?

Activities

6.1 Observe longitudinal and transverse waves in a horizontally suspended slinky. Waves can also be seen in a slinky lying on a table.

CHAPTER 10

Standing Waves

As noted in the previous chapter, large standing waves can occur at any frequency on a "long" string where the relationships between incident and reflected waves need be considered at only one end of the string. In this chapter we will consider the conditions that must be satisfied to produce standing waves on "short" strings and in "short" tubes where relationships between incident and reflected waves must be considered at both ends of the system. In particular, the boundary conditions at each end of a string or tube determine the natural modes and natural frequencies for standing waves. The same concepts are extended to standing waves in two- and three-dimensional systems.

10.1 Traveling Waves and Standing Waves

It is important to understand the differences between traveling and standing waves and their respective representations. A traveling wave on a string is illustrated in Figure 10.1 where the solid curve represents the displacement of the string at some initial time and the dashed curve represents the displacement at a later time. The maximum displacement shown at point A at the initial time moves to point B at the later time. At any point along the string (A for example) the displacement changes from large positive, to zero, to large negative, to zero, and back to large positive as the wave travels along the string.

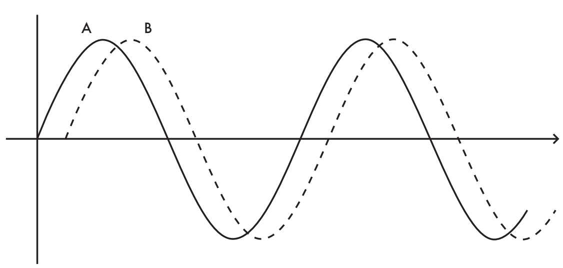

Figure 10.1 *Traveling wave on a string. The solid curve shows the string's displacement at some initial time and the dashed curve shows the string's displacement at a later time. The arrow shows the direction of wave travel.*

A standing wave on a string is illustrated in Figure 10.2 where the solid curve represents the displacement along the string at some initial time and the dashed curve represents the displacement at a later time. At antinodal point A the displacement changes from large positive, to zero, to large negative, to zero, and back to large positive as time progresses. However, at nodal point B the displacement remains zero at all times.

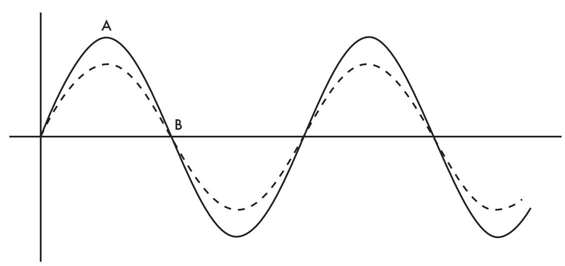

Figure 10.2 *Standing wave on a string. The solid curve shows the string's displacement at some initial time and the dashed curve shows the string's displacement at a later time.*

10.2 Standing Waves on Strings

Strings used on musical instruments such as guitars, violins, or pianos are fixed at both ends. Any disturbance produced on such a string will travel to the two ends of the string where it will be reflected in accordance with the rules discussed in Chapter 9. The various waves traveling back and forth along the string will interfere with each other. Nodes always exist at the two fixed ends of the string but may or may not exist at other points on the string. If we vibrate the string at just the right frequencies, we can produce standing waves where some parts of the string are stationary and where other parts move with maximum amplitude. The standing waves are examples of the natural modes of multimass systems discussed in Chapter 5. We must now explore these natural modes in more detail.

The simplest standing wave pattern which can be produced on a string fixed at both ends is that shown in Figure 10.3A. It consists of a node at each end of the string and an antinode (point of maximum vibration) in the center. The length of the string in this case is one-half wavelength. The frequency corresponding to this wavelength is called the fundamental frequency of the vibrating string, so-called because it is the frequency associated with the first natural mode which can form a standing wave pattern on the string. If the string is vibrated more rapidly (thus increasing the frequency) another standing wave pattern eventually results, as shown in Figure 10.3B. This pattern consists of a node at each end and one in the center. In this case the length of the string is equal to one wavelength. The wavelength is half as long and the frequency twice as great as in the preceding case because the wave speed is the same for all frequencies. The next standing wave pattern which can be produced is shown in Figure 10.3C; the frequency in this case is three times the fundamental frequency. These frequencies are the first three natural mode frequencies of the string and the wave pattern associated with each frequency is called a natural mode of vibration.

The natural frequencies can also be calculated from the relation

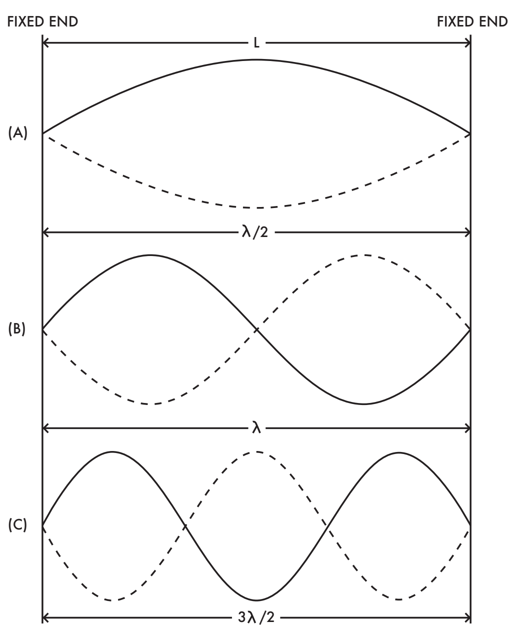

Figure 10.3 *Natural modes of string fixed at both ends: (A) first or fundamental mode with λ = 2L, (B) second mode with λ = L, and (C) third mode with λ = 2L/3. The solid curves show the string with maximum displacement and the dashed curves show the string one-half period later with maximum displacement.*

$$v = f\lambda \qquad (10.1)$$

where the wave speed is determined from the string's density and tension. The wavelength can be expressed in terms of the length of the string. For the first mode half a wavelength is equal to the length of the string as can be seen in Figure 10.3A. The wavelength of the first mode is then twice the length of the string or $\lambda = 2L$. This results in

$$f_1 = v/\lambda_1 = v/2L \qquad (10.2)$$

for the fundamental frequency of a string fixed at both ends. Similarly, for the second mode the wavelength is equal to the string length or $\lambda = L$. This results in a second mode frequency of

$$f_2 = v/\lambda_2 = v/L \qquad (10.3)$$

which is twice the frequency of the first mode. The third mode has a wavelength one-third that of the first mode and a frequency three times that of the first mode, and so on for the higher modes.

From the foregoing we see that the frequency of each mode is the mode number multiplied by the fundamental frequency for a string fixed at both ends. If the fundamental frequency is 100 Hz, the second mode has a frequency of 200 Hz, the third 300 Hz, and so on. We can summarize by saying that the natural frequencies of a "fixed-fixed" string, that is, a string fixed at both ends, are given by

$$f_n = n \times f_1 \qquad (10.4)$$

where n = 1, 2, 3, ... The natural frequencies of the fixed-fixed string are termed **harmonics** because they are integer (whole number) multiples of the fundamental frequency. More will be said in Chapter 16 about harmonics and their role in musical harmony.

10.3 Standing Waves in Tubes

While musically useful strings are usually fixed at both ends, musically useful air-filled tubes can be open at both ends, as with the flute, or closed at one end and open at the other, as with the clarinet. We can use the same basic techniques to explore natural modes for air columns as we used for strings.

In a tube whose diameter is small compared to the wavelength of waves traveling in it, the pressure waves in the air in the tube exhibit many of the same features as the standing waves in a string, except that pressure is approximately zero at the open end of a tube. That is, a pressure node exists at an open end. This is because the air outside the tube is at atmospheric pressure which is the reference or zero pressure. Hence, at an open end the incident and reflected waves add together destructively, thus producing a pressure node. At the closed end of a tube, a pressure wave reflects with the same phase as the incident wave. The incident and reflected waves then add together constructively, producing a pressure antinode at the closed end of a tube.

In many ways, pressure waves in an open-open tube are analogous to displacement waves on a fixed-fixed string. The simplest standing wave pattern that can be produced in a tube open at both ends is shown in Figure 10.4. It consists of a node at each end of the tube and an antinode (point of maximum pressure fluctuation) in the center. The length of the tube in this case is one-half wavelength. The frequency corresponding to this wavelength is the fundamental frequency of the air column, so-called because it is the frequency associated with the first (or fundamental) natural mode that can form a standing wave pattern in the tube. There are actually two representations of the wave (similar to those in Figure 6.4) shown in Figure 10.4 so that we can be reminded of the correspondence between the graphical and "molecular" representations.

If the tube is driven at a higher frequency another standing wave eventually results as shown in Figure 10.5. This pattern consists of a pressure node at each end and one in the center. In this case the length of the tube is equal to one wavelength. The wavelength is half as long and the frequency twice as great as in the preceding case because the wave speed is the same for all frequencies. The third mode would have three antinodes, a frequency three times that of the first mode, and a wavelength one-third that of the first mode.

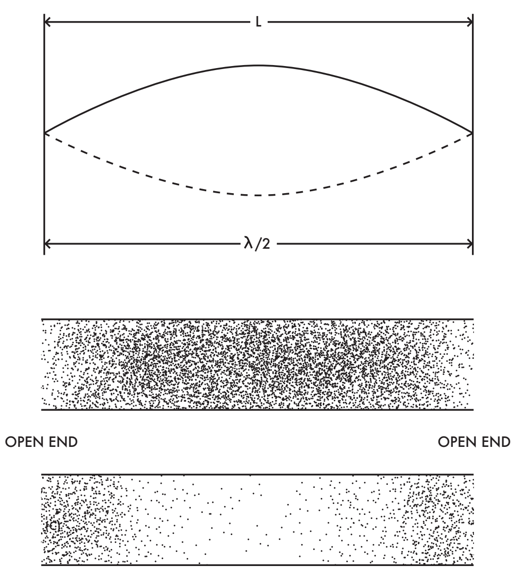

Figure 10.4 *First (or fundamental) natural mode of an air-filled tube open at both ends with* $\lambda = 2L$*. The solid curve shows the tube when there is maximum positive pressure at the antinode. The dashed curve shows the tube one-half period later when there is maximum negative pressure at the antinode. The upper molecule-filled tube corresponds to the solid curve and the lower tube corresponds to the dashed curve.*

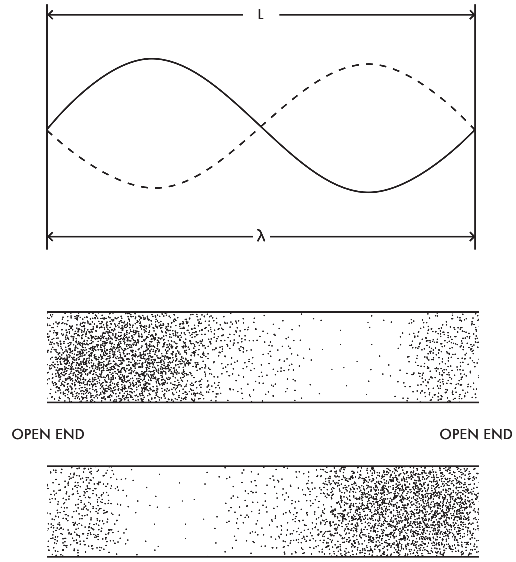

Figure 10.5 *Second natural mode of an air-filled tube open at both ends with* $\lambda = L$*. The solid curve shows the tube when there is maximum positive pressure at one antinode and maximum negative pressure at the other antinode. The dashed curve shows the tube one-half period later. The upper molecule-filled tube corresponds to the solid curve and the lower tube corresponds to the dashed curve.*

The natural frequencies can be calculated from Equations (10.1)–(10.4) that were used for the fixed-fixed string. We can summarize by saying that the natural frequencies of a tube open at both ends are given by Equation (10.4) just as for the fixed-fixed string. The natural frequencies of the open-open tube are termed harmonics because they are integral multiples of the fundamental frequency.

We consider now the simplest standing wave pattern that can be produced in a tube closed at one end and open at the other as shown in Figure 10.6. There is a pressure node at the open end of the tube as before. However, there is a pressure antinode at the closed end of the tube because the pressure can become higher or lower than atmospheric pressure. The standing wave shown in Figure 10.6 is the smallest part of a sine wave that can "fit" in the tube and still satisfy the endpoint conditions. Clearly this is one-quarter of a wavelength, so the fundamental wavelength is four times the length of the tube. The next smallest fraction of wavelength that "fits" in the tube is shown in Figure 10.7. For this second natural mode three-fourths of a wavelength fits in the tube. This mode has a wavelength equal to one-third that of the fundamental and a frequency three times that of the fundamental.

The modal frequencies of a closed-open tube can be calculated in a manner similar to that used for the fixed-fixed string. Referring to Figure 10.6, we see that the wavelength of the lowest mode is four times the length of the tube or $\lambda = 4L$. This results in

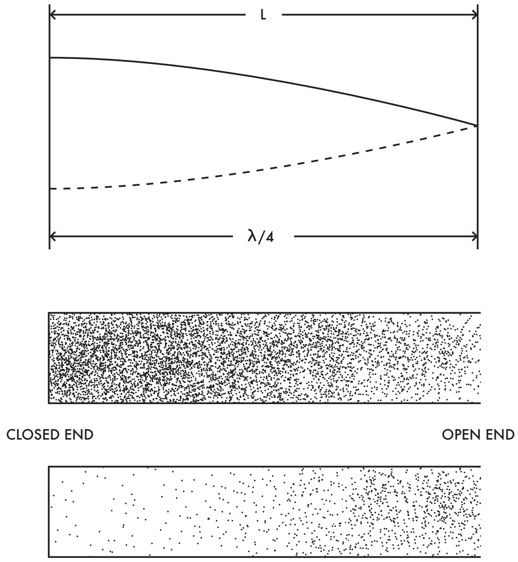

Figure 10.6 *First or fundamental natural mode of an air-filled tube closed at one end and open at the other with λ = 4L. The solid curve shows the tube when there is a maximum positive pressure at the antinode and the dashed curve shows the tube one-half period later when there is maximum negative pressure at the antinode. The upper molecule-filled tube corresponds to the solid curve and the lower tube corresponds to the dashed curve.*

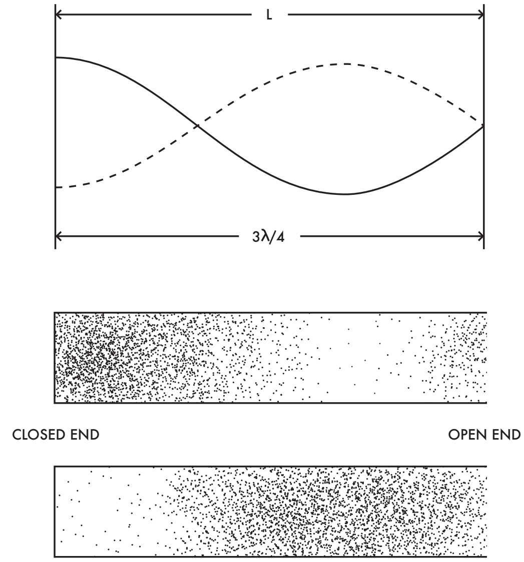

Figure 10.7 *Second natural mode of an air-filled tube closed at one end and open at the other end with λ = 4L/3. The solid curve shows the tube when there is maximum positive pressure at one antinode and maximum negative pressure at the other antinode. The dashed curve shows the tube one-half period later. The upper molecule-filled tube corresponds to the solid curve and the lower tube corresponds to the dashed curve.*

$$f_1 = v/\lambda_1 = v/4L \qquad (10.5)$$

as the frequency of the fundamental mode. For the second natural mode (Figure 10.7), three-fourths of a wavelength is equal to the tube length or $\lambda = 4L/3$ which results in

$$f_2 = v/\lambda_2 = 3v/4L \qquad (10.6)$$

for the frequency of the second mode, which is three times the frequency of the first mode. By studying higher modes we discover that the higher natural mode frequencies are odd integer multiples of the fundamental frequency. We can summarize by saying that the natural frequencies of a closed-open tube are given by

$$f_n = (2n - 1) \times f_1 \qquad (10.7)$$

where n = 1, 2,3 … .

Two interesting features become apparent when we compare an open-open tube with a closed-open tube. First, for open-open and closed-open tubes of equal length, L, and with the same wave speed, v, the fundamental frequency of the open-open tube ($f_1 = v/2L$) is twice that of the fundamental frequency of the closed-open tube ($f_1 = v/4L$). Secondly, the open-open tube has natural frequencies that are integer multiples of its fundamental frequency (see Equation 10.4), whereas the closed-open tube has natural frequencies that are odd integer multiples of its fundamental frequency (see Equation 10.7).

10.4 Losses and Impedance

Two important things happen when the initial and reflected waves travel back and forth in a tube. The traveling waves produce large standing waves at the natural frequencies. These standing waves have antinodes at some points in the tube produced by constructive interference and nodes at other points produced by destructive interference. Traveling waves in the tube lose part of their energy to frictional forces at the walls of the tube. These losses limit the amplitudes of the standing waves. Because the losses are generally greater at higher frequencies the higher frequency modes will generally have smaller amplitudes.

The following method can be used to determine experimentally the natural frequencies of a tube. In order to discover the frequencies at which the tube resonates and thus produces the largest standing waves, a variable frequency sine wave oscillator is used to drive a small loudspeaker that is placed in the closed end of a tube as shown in Figure 10.8. A microphone is also inserted in the closed end to measure pressure, and the corresponding microphone voltage output is displayed on an oscilloscope. A quantity termed **impedance** (a kind of "resistance" to flow) can be then be defined as the ratio of pressure (measured by the microphone) to the volume of airflow (produced by the quencies of membranes are often not integer multiples of the fundamental. Rooms have natural modes which can be composed of one, two, or three-dimensional standing waves. Their natural frequencies are often not integer multiples of the fundamental.

References and Further Reading

Backus, J. (1977). *The Acoustical Foundations of Music, 2nd ed* (Norton). Chapter 4 contains material on standing waves in one- and two-dimensions.

Benade, A. H. (1976). *Fundamentals of Musical Acoustics* (Oxford Univ. Press; Dover edition 1991). Chapters 7 and 9 contain material on standing waves in one- and two-dimensional structures.

Copeland, P. L. (1939). "The Simple Harmonic Vibrations of a Stretched Rectangular Membrane," *Am. J. Phys. 7*, 233–237.

Munley, F. (1988). "Phase and Displacement in Sound Waves," *Am. J. Phys. 56*, 1144.

Questions

10.1 Why is pressure used to describe waves in tubes while displacement is used to describe waves in strings? (Hint: Which is easier to measure in air-filled tubes?)

10.2 Do displacement antinodes occur at open or closed ends of air-filled tubes? Displacement nodes?

10.3 What gives rise to "dead spots" in rooms?

10.4 Describe how treating walls with absorbing materials and placing objects in a room help to alleviate "dead spots" in the room.

10.5 By what fraction of a wavelength are a node and its nearest antinode separated? Nearest neighbor antinodes? Nearest neighbor nodes?

Exercises

10.1 Sketch the third, fourth, and fifth modes for a fixed-fixed string. Express the frequency of each mode in terms of the fundamental frequency, f.

10.2 Sketch the third, fourth, and fifth modes for an open-open tube. Express the frequency of each mode in terms of the fundamental frequency, f.

10.3 Sketch the third and fourth modes for pressure waves in a molecule-filled tube for an open-open tube.

10.4 Sketch the third and fourth modes for pressure waves in a closed-open tube.

10.5 List frequencies of the next three modes of waves on a fixed-fixed string if its first or fundamental mode has a frequency of 200 Hz. What is the relationship of the higher mode frequencies to that of the fundamental?

10.6 List frequencies of the next three modes of waves in an open-open tube if its first or fundamental mode has a frequency of 200 Hz. What is the relationship of the higher mode frequencies to that of the fundamental?

10.7 List frequencies of the next three modes of waves in a closed-open tube if its first or fundamental mode has a frequency of 200 Hz. What is the relationship of the higher mode mode frequencies to that of the fundamental?

10.8 Calculate the frequencies for (1,0), (1,2), (2,1), and (2,2) modes of a square membrane with a side length 0.5 m if the wave speed is 30 m/s. What special relationship exists between the (1,2) and (2,1) modes that does not hold for rectangular membranes in general?

10.9 Determine the first, second, and third natural frequencies for an open-open pipe 240 cm in length.

10.10 Determine the first, second, and third natural frequencies for a closed-open pipe 240 cm in length. Compare your results with those of Exercise 10.9.

10.11 Determine the first natural frequency for an open-open pipe 240 cm in length when filled with carbon dioxide ($v = 260$ m/s). Do the same thing for a case in which the pipe is filled with helium ($v = 970$ m/s). Compare your results with the first natural frequency of the pipe in Exercise 10.9.

10.12 Find the first three natural frequencies of a 50 cm length of nylon string having a density of 0.01 gm/cm and stretched with a force of 1 N.

10.13 A certain pipe of length 36 cm sounds A4 = 440 Hz. What length might be expected for a similar pipe that sounds A3 = 220 Hz?

10.14 You are given a string of length 55 cm. What are the wavelengths of its first four natural modes?

10.15 Sketch the displacement at successive intervals of time during one cycle for a string vibrating in its third mode. Also indicate its direction of motion. (Hint: Use the lower part of Figure 10.3 as a basis, but scale its amplitude appropriately for each time interval.)

10.16 A cylindrical-tube musical instrument is open at one end and closed at the other. If its second natural frequency is 300 Hz, what are the frequencies of its first, third, fourth, and fifth modes?

10.17 Determine the wavelength (in cm) of the second mode for a tube closed at one end and open at the other if the length of the tube is 80 cm.

10.18 Determine the frequency (in Hz) of the first mode of an open-open tube if its length is 34 cm and the wave speed is 34,000 cm/s.

10.19 The wave speed is 22,000 cm/s on a string of length 44 cm. What is its first natural frequency? What is its fourth natural frequency?

10.20 By how much would the tension on a string have to be increased to double its frequency?

10.21 A guitar string sounding flat is found to have a frequency 6% too low. By how much must the string's tension be increased to make it sound in tune?

10.22 An organ is to be designed to sound the lowest note possible but the pipe length is limited by a floor-to-ceiling distance of 6 m. What is the lowest frequency possible if an open-open pipe is used? If a closed-open pipe is used?

Activities

10.1 "Standing waves in strings"—The apparatus shown in Figure 10.11 is set up to produce standing waves in a wire. A 60-Hz driver is attached to the left end. The density of the wire is D = 0.000642 kg/m. The length of the wire is 1.0 m. The tension applied to the wire can be controlled by adding mass. When a mass of 0.3 kg is suspended from the wire, the fourth mode is produced as shown. What is the wave speed for this mode? What is the wavelength? What mass must be suspended to produce the second mode? The first mode?

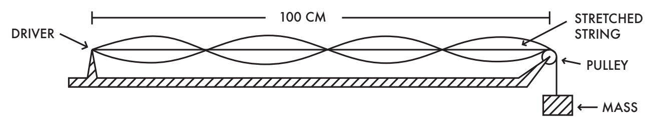

Figure 10.11 *Apparatus to demonstrate standing waves in a wire.*

10.2 A loudspeaker is set in the corner of a hard-walled rectangular room with dimensions of $3.0 \times 4.0 \times 5.0$ m. A sine wave generator is used to drive it. A sound level meter is placed in a diagonally opposite comer to measure sound pressure levels. A maximum sound pressure level is observed to occur at a frequency of 78 Hz. What room mode is being excited? What is the approximate sound speed in the room? For what mode and at what next higher frequency would another sound pressure level maximum occur?

10.3 Observe the effect of resonance while you are singing in a shower or bathtub. (R. D. Edge, 1985, "Physics in the Bathtub or, Why Does a Bass Sound Better While Bathing?," *Phys. Tchr. 23,* 440–444).

10.4 Listen to standing waves in a tube. (J. M. Reynolds, 1973, "Sound in a Tube," *Phys. Tchr. 11*, 31).

in Figures 11.2 and 11.3 might be those of a struck membrane or struck string, respectively. On the other hand, systems driven with a periodic driving force will have a periodic response. Such is at least approximately true for bowed strings and sustained sounds of speech.

11.2 Analysis of Complex Waves

As noted above, each ingredient of a complex wave is a sinusoid having a different frequency. The spectrum for a particular wave tells us the frequency and amount of each ingredient present. The analysis of a complex wave therefore means determining the spectrum of the wave. Several common methods are used to analyze complex waves in order to determine spectra, including electronic filtering and digital Fourier analysis. Because the several spectral analysis methods produce similar results, the conceptually simpler filter method will be discussed.

A **filter** is a device which allows certain frequencies to pass through unchanged, while other frequencies are eliminated. Think of a set of frequency filters as functioning like a set of screens, each with a different mesh size. The screens sort a conglomerate of gravel into coarse, medium, and fine components. Likewise, a set of three filters (each having a different set of characteristics) could be used to separate a complex wave into low-, medium-, and high-frequency components. Figure 11.4 shows how a complex wave could be resolved into its components by passing it through a set of filters. The high-resolution spectrum analyzer described in Chapter 7 may be thought of as being composed of some 200 to 400 filters. The greater the number of filters, the more finely the frequency components can be resolved during a spectrum analysis; that is, with more filters smaller frequency differences may be detected.

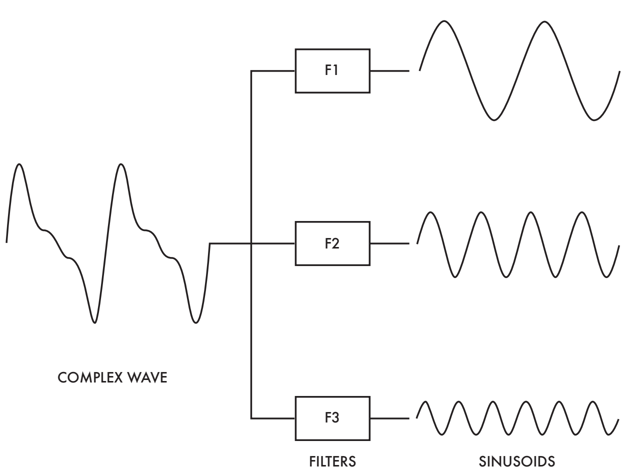

Figure 11.4 *The analysis of a complex wave with a set of filters tuned to different frequencies. (The sinusoids shown occur after the initial transients have decayed.)*

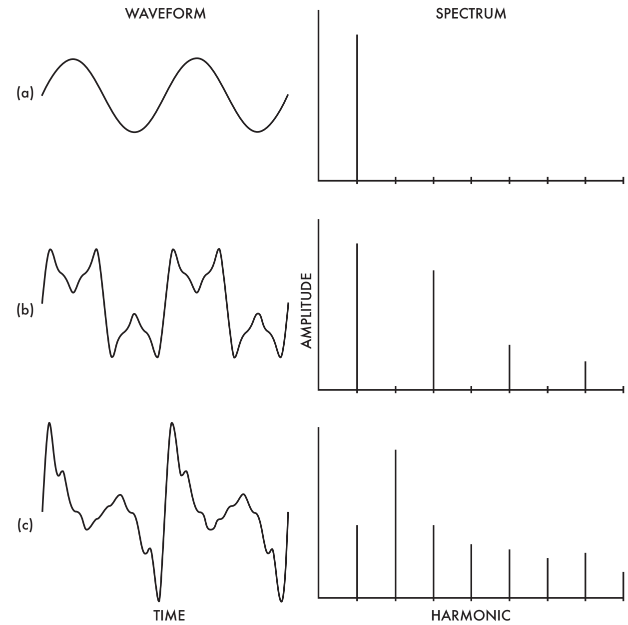

Figure 11.5 *Idealized waveforms and spectra of several "musical instruments": (a) tuning fork, (b) clarinet, and (c) trumpet.*

As an example of the spectral analysis of complex waves, consider the three waveforms shown in the left column of Figure 11.5. The waveforms shown are the pictures obtained when three different "stylized" instruments are sounded: (a) a tuning fork, (b) a clarinet, and (c) a trumpet. Because of the complex nature of these waveforms, little information can be gleaned by observing them. If the waveforms are analyzed with a spectrum analyzer to obtain their spectra, as shown in the right column of Figure 11.5, the pertinent information is more apparent. We see, for instance, that the tuning fork waveform consists of only one component, the fundamental, while the clarinet displays predominantly odd harmonics. The trumpet is seen to have all harmonics, but the second harmonic has the greatest amplitude.

11.3 Synthesis of Complex Waves

In Chapter 9 we used the superposition principle to add waves having identical or only slightly different frequencies. The same principle can be used when waves of quite different frequencies are added together. The synthesis of a complex wave implies that the resultant wave is constructed by adding together simple sinusoids, "piece by piece." A repetitive waveform of any shape can be constructed by this method if enough components are added together. As a simple example of a graphical method used to add sinusoids, a mass is attached to a spring which is mounted on a frame in a boat. When the mass is set into vibration it repeats its motion every second when observed

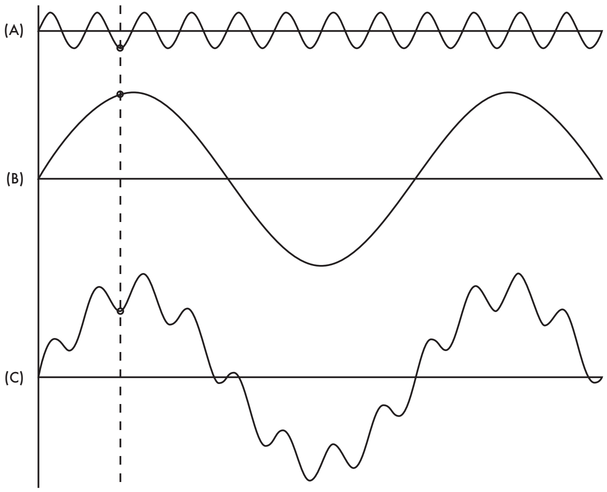

Figure 11.6 *(A) Vibrating mass as seen from boat. (B) Motion of boat as seen from dock. (C) Motion of mass as seen from dock.*

from inside the boat. The boat rocks up and down once every eight seconds due to passing waves. When the motion of the mass is observed from the dock, we see a resultant motion that is the sum of two motions. If we plot the motion of the mass as seen from the boat we obtain Figure 11.6A. Figure 11.6B is a plot of the motion of the boat as seen from the dock. Using a synthesis approach and adding these two yields Figure 11.6C, which is the motion of the mass as seen from the dock.

The procedure used to add two waves graphically is illustrated in Figure 11.6. A point along the time axis of the two waves (Figures 11.6A and 11.6B) is marked by the dashed vertical line. A ruler is used to measure the displacement of each wave at this point in time along the vertical line. The two measurements are added, keeping in mind that displacements above the axis are positive numbers while those below the axis are negative. A mark is made in Figure 11.6C representing the result of the addition. The procedure is repeated for the remaining points on the waveforms. Connecting the marks provides a smooth curve.

An equation representing pressure waves can be written as

$$p = A \sin (360\, ft + \phi) \qquad (11.2)$$

where p is the quantity you would plot on a graph of pressure versus time, A is the pressure amplitude of the sinusoid, f is the frequency, t is time, and ϕ is the phase in degrees. As an example, we can obtain the relative amplitudes for the first two nonzero partials of a square wave as 1.0 and 0.33 from Chapter 7. We do not get information on the relative phases which some experimentation would show should be 0 in both cases. If the frequencies for these partials are 1 Hz and 3 Hz, we can write an expression for our square wave as

$$p = p_1 + p_2$$

where

$$p_1 = 1.0 \sin (360t)$$

and

$$p_2 = 0.33 \sin (1080t)$$

We now construct Tables 11.1 and 11.2 which give values for p_1 and p_2 at different instants of time. Adding p_1 and p_2 for each instant of time gives p.

Table 11.1 *Steps in calculation of $p_1 = \sin(360t + 0)$. Time t is in seconds and arguments for sine function are in degrees.*

t	360t	(360t + 0)	sin (360t + 0)	p_1
0.0	0	0	0.00	0.00
0.1	36	36	0.59	0.59
0.2	72	72	0.95	0.95
0.3	108	108	0.95	0.95
0.4	144	144	0.59	0.59
0.5	180	180	0.00	0.00
0.6	216	216	-0.59	-0.59
0.7	252	252	-0.95	-0.95
0.8	288	288	-0.95	-0.95
0.9	324	324	-0.59	-0.59
1.0	360	360	0.00	0.00

Table 11.2 *Steps in calculation of $p_2=0.33 \sin(1080t+0)$. The starting phase is 0°. (Whenever the sine argument exceeds 360°, it is adjusted to lie between 0° and 360° by subtracting 360°.)*

t	1080t	(1080t + 0)	sin (1080t + 0)	p_2
0.0	0	0	0.00	0.00
0.1	108	108	0.95	0.31
0.2	216	216	-0.59	-0.19
0.3	324	324	-0.59	-0.19
0.4	72	72	0.95	0.31
0.5	180	180	0.00	0.00
0.6	288	288	-0.95	-0.31
0.7	36	36	0.59	0.19
0.8	144	144	0.59	0.19
0.9	252	252	-0.95	-0.31
1.0	360	360	0.00	0.00

To produce smoother curves the graphs showing p_1, p_2, and p in Figure 11.7 were plotted at much shorter time intervals than those given in Tables 11.1 and 11.2. The data points from the tables are also shown in the figure. The lower graph is a sum of the upper two. The total time interval from 0 to 1.0 s was chosen to cover one cycle of vibration.

Now consider a different relative phase for p_2 so that

$$p_2 = 0.33 \sin (1080t + 180)$$

11.4 Outline a method by which a musical tone may be artificially created.

11.5 Explain why identical notes plucked on a guitar and a banjo have distinctly different sounds.

11.6 In what physical characteristics do loud, high violin tones differ from soft, low flute tones?

11.7 Is a steadily blown trumpet a free or a driven system? Will its pressure waveforms be periodic or nonperiodic? Will its spectrum have harmonic or inharmonic partials?

11.8 Is a bowed cello string with vibrato a free or a driven system? Will its displacement waveforms be periodic or nonperiodic? Will its spectrum have harmonic or inharmonic partials?

11.9 Is a plucked guitar string a free or a driven system? Will its displacement waveforms be periodic or nonperiodic? Will its spectrum have harmonic or inharmonic partials?

11.10 Is a struck piano string a free or a driven system? Will its displacement waveforms be periodic or nonperiodic? Will its spectrum have harmonic or inharmonic partials?

11.11 Is a struck bass drumhead a free or a driven system? Will its displacement waveforms be periodic or nonperiodic? Will its spectrum have harmonic or inharmonic partials.

11.12 When a steady vowel sound is produced, is the system free or driven? Will the resulting pressure waves be periodic or nonperiodic? Will its spectrum have harmonic or inharmonic partials.

11.13 When a plosive burst such as the /p/ in "plosive" is produced, is the system free or driven? Will the resulting pressure waves be periodic or nonperiodic? Will the spectrum have harmonic or inharmonic partials?

11.14 Are the partials in a nonperiodic complex wave harmonic or inharmonic?

11.15 Do inharmonic partials produce a periodic or a nonperiodic wave?

Exercises

11.1 Consider two sinusoids given by sin (360ft) and sin (360ft + ϕ), where ϕ is the phase. Plot these two sinusoids for f = 2 Hz and ϕ = 90° over an interval of 1 second.

11.2 What is the amplitude of each wave in Exercise 11.1? The frequency? The phase?

11.3 The conventional spectra in Table 11.5 represent six waves, some simple and some complex. The number preceding each "sin" is the amplitude or the amount of that particular sinusoid present. What is the fundamental frequency of each wave? What are the frequencies and amplitudes of the higher partials for each wave? Are the higher partials harmonic or inharmonic relative to the fundamental?

11.4 Wave 4 in Table 11.5 is the sum of two sinusoids differing in phase by 180°. What is the net result?

11.5 Wave 5 in Table 11.5 is an approximately triangular wave. Plot one cycle of it on graph paper by plotting each component separately and then graphically adding the three components to form the complex wave.

11.6 Waves 5 and 6 in Table 11.5 have the same spectra. However, they differ in phase. If both of them are plotted and the wave shapes compared, they look different but they will sound the same. Plot the two waves. Then plot their spectra as bar graphs.

11.7 The relative amplitudes for the first few partials of a sawtooth wave are 1, 1/2, 1/3, 1/4, 1/5, and 1/6. The relative phases for all partials are zero. Plot one cycle of the sinusoid representing the first partial of the sawtooth wave on a sheet of graph paper. Next plot two cycles of the second partial with the proper relative amplitude. Repeat this

Table 11.5 *Various waves and their components.*

Wave 1: 1.0 sin $(360f_1t)$	f_1 = 100 Hz
Wave 2: 1.0 sin $(360f_1t)$ + $(360f_2t)$	f_1 = 100 Hz, f_2 = 200 Hz
Wave 3: 1.0 sin $(360f_1t)$ + $(360f_2t)$	f_1 = 100 Hz, f_2 = 205 Hz
Wave 4: 1.0 sin $(360f_1t)$ + $(360f_1t + 180)$	f_1 = 100 Hz
Wave 5: 1.0 sin $(360f_1t)$ + 0.11 sin $(360f_3t +180)$ + 0.04 sin $(360f_5t)$	f_1 = 100 Hz, f_3 = 300 Hz, f_5 = 500 Hz
Wave 6: 1.0 sin $(360f_1t)$ + 0.11 sin $(360f_3t)$ + 0.04 sin $(360f_5t)$	f_1 = 100 Hz, f_3 = 300 Hz, f_5 = 500 Hz

for the remaining partials. Now graphically add all of these sinusoids together to produce a fairly good approximation to a sawtooth wave.

11.8 Repeat Exercise 11.7 but with arbitrary phases. What happens to wave shape?

11.9 A soprano sings a steady A4 =440 Hz with no vibrato. What are the frequencies of the partials in the sound? Are they harmonic or inharmonic?

11.10 A male voice produces a steady vowel sound. The resulting complex waveform is seen to repeat every 8 ms. What are the frequencies of its partials? Are they harmonic or inharmonic?

11.11 A complex tone is composed of four sinusoids having frequencies of 100, 201, 302, and 403 Hz. Would the partials be best described as harmonic, slightly inharmonic, or very inharmonic?

11.12 A complex tone is composed of four sinusoids having frequencies of 100, 200, 300, and 400 Hz. What is the frequency of the second partial? Of the fundamental or first partial?

Activities

11.1 Complex waves: analysis. A function generator is connected to a band-pass filter and its output is observed on an oscilloscope. When the function generator is set to produce sinusoids at a frequency of 200 Hz, the voltages in the "sine" row of Table 11.6 are measured on the oscilloscope at the frequencies shown. Similar measurements are made when the function generator produces square waves, triangular waves, and sawtooth waves, as shown in the table. What can you say about the spectrum for each of the four wave types? Why were some voltages only slightly different from zero measured?

Table 11.6 *Voltages measured at three different frequencies for four different functions.*

Function	200 Hz	400 Hz	600 Hz
Sine	10	0.01	0.02
Square	10	0.03	3.3
Triangular	10	0.02	1.1
Sawtooth	10	5	3.3

11.2 Analyze various complex waves by running them into an oscilloscope to see the waveform and a spectrum analyzer to see the spectrum. Use a function generator, tape recordings, and microphone signals as inputs.

11.3 Synthesize various complex periodic waves with a Fourier synthesizer. Run the output to an oscilloscope, a spectrum analyzer, and a loudspeaker. Does the "output" of the spectrum analyzer agree with the "input" of the synthesizer? Change relative phases and observe the waveform and the sound. Which change with phase? Why?

11.4 Listen to sound "through" a seashell, a papertowel tube, and so on. Describe what you hear and explain why.

CHAPTER 16

Musical Scales and Harmony

In discussion of JNDs we discovered that at a constant intensity level the ear can discriminate several thousand frequencies. However, this represents a far too many frequencies to be useful in music, where a few hundred frequencies are found to be sufficient. The actual set of frequencies used to create music has evolved over many centuries. Melody (homophony), or tones played in sequence, probably exerted some influence on the notes (or frequencies) chosen. Multiple concurrent melodies (polyphony) may have exerted further influences on the tones selected. However, the appearance of harmony, in which two or more tones are sounded concurrently to form chords or other harmonic musical structures, has probably had the most significant influence on the selection of notes used in the scales of Western music. In parts of the world where harmony play a less important role in music, the tones selected differ from those in Western music. In this chapter, after introducing some basic definitions, we will consider some of the systems of tuning that have been given serious consideration over the years. We will then briefly consider a physical basis for the evolution of Western harmony.

16.1 Scales and Intervals

No one knows when humans created the first musical melody, but we can be reasonably certain that the human voice was the first musical instrument. The word *melody* itself derives from two Greek words, *melos* (song) and *aoidos* (singer), indicating the important role of our vocal apparatus in producing melodies. Although we usually associate the word "melody" with a pleasing succession of tones, it is almost impossible to obtain agreement as to which succession of tones is pleasing and which is not. We have a natural bias toward the succession of tones which our particular culture has dictated as "pleasant," while the tones used in other cultures may seem strange, or "out of tune." To avoid cultural bias, we define **melody** as a succession of tones arranged in a particular order. Melodies in classical music of the Western world are arranged as a series of definite pitches. Melodies of the epic singing of Eastern European countries may slip and slide in a seemingly random way around a tonal center. Between these two extremes there are other systems, such as the music of India and American "soul" music which combine a set of basic pitches with fluctuating tones (called microtones) having a much smaller frequency variation.

The particular set of basic tones used to construct melodies, when arranged in order of ascending and descending pitches, is called a **scale** (from the Latin, *scala*, "a ladder"). An individual frequency or pitch of a scale is called a **note**, or tone, of the scale. (The word "note" is also used to connote a symbol on a sheet of music paper or the name of a musical tone.) Different scales are characterized by the number of notes per octave; a chromatic scale has 12 notes per octave, a diatonic scale has seven, and a pentatonic scale has five.

The "pitch spacing" between two notes is called an **interval** and may be given special names such as octave, fifth, third, etc. Intervals may be expressed in terms of frequency ratios such as 2/1, 3/2, 5/4, etc. An **octave** is an interval between two tones so that the ratio of their fundamental frequencies is 2/1. Probably because there is a near identity of a tone with its octave, the octave appears in virtually every system. We will limit our discussion of scales to a description of the way in which the notes within an octave are defined. The notation used to specify which octave is being referred to will be that of the USA Standards Association. In this notation, the octave numbering starts on C, with C4 being middle C. A4 is the A above C4, C5 is the C an octave above middle C, C3 is an octave below middle C, and so on.

Since the terminology of interval names has reference to a piano keyboard, we digress briefly to consider the evolution of the piano. The piano keyboard was borrowed from a much older instrument, the organ. In 1361 in the Saxon city of Halberstadt an organ builder, Nicholas Faber, completed a three-manual instrument which was destined to exert a substantial influence on all future organs. The upper two manuals of the Halberstadt organ had a series of nine front keys and five raised rear keys in groups of two and three, as shown in Figure 16.1. Even though these keys were made to be struck by the fists, this was the prototype of the now well-known seven white and five black keys per octave. ("Prototype" does not imply color, as black and white keys were not used until 1475, and even at that time the convention was reversed, a practice that prevailed for the next 300 years.)

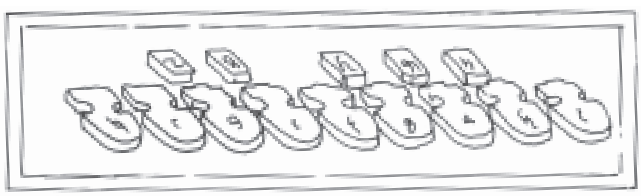

Figure 16.1 *Third manual of the Halberstadt organ. (After Praetorius' Syntagma Musicum, 1619.)*

The modern piano keyboard is shown in Figure 16.2 along with an example of musical notation. If we number the white keys starting with C as one, we find that F is the fourth white key or "fourth," and G is the "fifth." The octave ends on the eighth white key, which is C again. Consecutive white keys, however, actually contain two different types of interval: whole tones and the semitones. The interval between any two consecutive keys, white or black, is a semitone, regardless of whether we go from white to black, white to white, or black to white. A whole tone is arrived at by passing through two semitones. The interval from C to D is a whole tone, while the interval from E to F is a semitone. Musical intervals can be defined from any starting note. Figure 16.2 shows that the interval from C to G (a fifth) consists of 7 semitones, while a third (C to E) consists of 4 semitones. The third of G is thus seen to be B, while the fifth of G is the D above. But what is the fifth of B? If we go up by 7 semitones we end on a black key, F#, which is the fifth of B.

We have seen that intervals can be expressed as frequency ratios. An octave is a frequency ratio of 2/1, a fifth is a ratio of 3/2, a fourth is a ratio of 4/3, and so on. When two intervals are combined, the frequency ratio of the combined interval is given by the product of the original ratios. For example, combining a fifth and a fourth gives an octave whose frequency ratio is

$$(3/2) \times (4/3) = (2/1)$$

The interval-combining procedure can be simplified by expressing the intervals in such a way that they can be

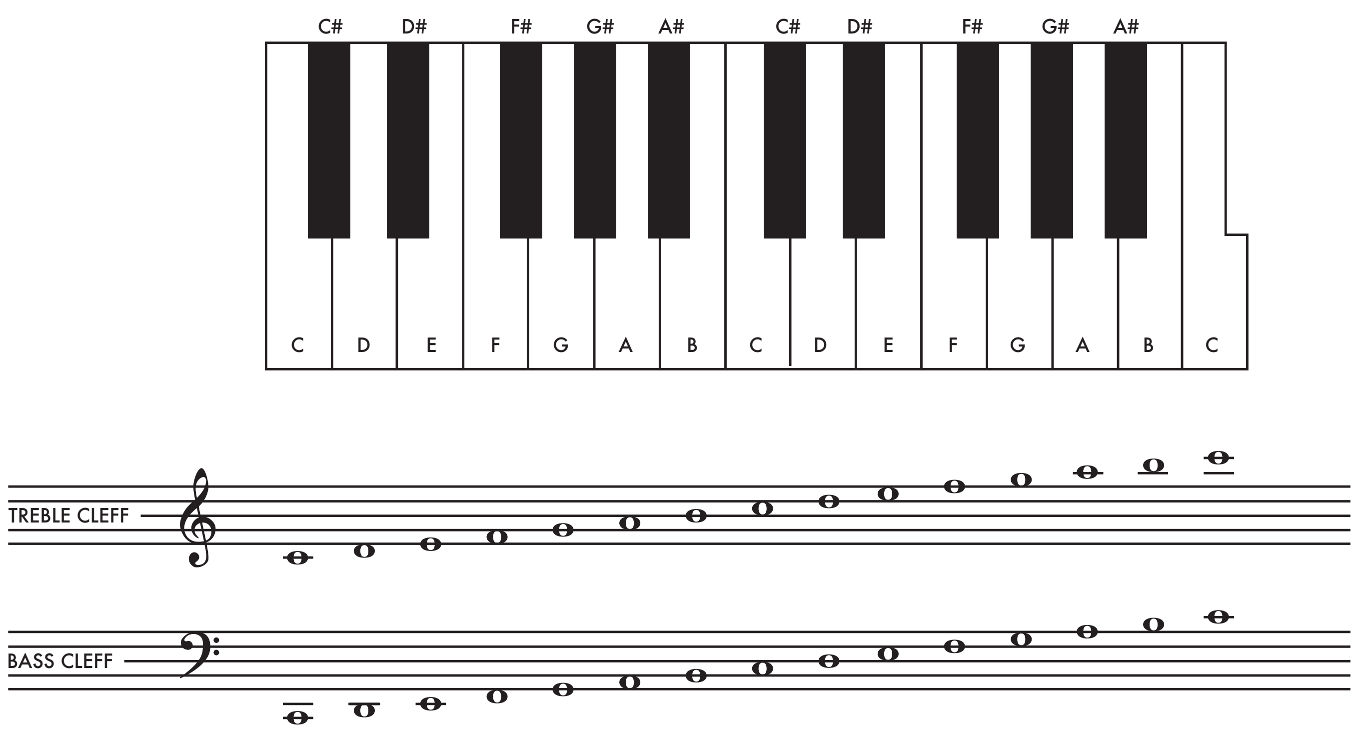

Figure 16.2 *Section of a modern piano keyboard with note names (upper) and corresponding treble cleff and bass cleff musical notation for the white keys (lower).*

16.3 Calculate the fundamental frequencies for an equal-tempered diatonic scale from C4 to B4 when A4 is tuned to 440 Hz.

16.4 Calculate the fundamental frequencies for an equal-tempered diatonic scale from C5 to B5 when A4 is tuned to 440 Hz.

16.5 Consider the discussion of the A-major triad given in the section entitled "Harmony and Its Evolution." Assume that each note of the chord has the first six harmonics present. Calculate the frequency of every difference tone that will be present.

16.6 The cent can be defined as some scaling constant times the logarithm of the frequency ratio of one one-hundredth of a semitone. Let SC be the scaling constant and we can write 1 cent = SC × log(1.00057779). Use a calculator to determine the value of SC. How does it correspond to the scaling constant in the text?

16.7. Any musical interval expressed as a frequency ratio can be expressed in cents by Equation (16.2). What is the interval in cents for a semitone whose frequency ratio is 1.059463?

16.8 What is the interval in cents for an octave whose frequency ratio is 2?

16.9 What is the interval of a musical fifth expressed in cents? Of a musical fourth? Does the tuning make a difference?

16.10 The frequency ratio for a semitone is defined as the one-twelfth root of two so that the product of 12 of these ratios gives the octave ratio as $(2^{1/12})^{12} = 2$. How does it correspond to the semitone frequency ratio in the text?

16.11 Repeat Exercise 16.10 for an interval of one cent.

16.12 How many semitones are there in an octave (such as C4 to C5)?

16.13 How many semitones are there in an interval of a fifth (such as C4 to G4)? What note is a fifth above G4?

16.14 How many semitones are there in an interval of a third (such as C4 to E4)? What note is a third above E4?

16.15 Determine the frequency of A4 in a diatonic scale with Pythagorean tuning if C4 is tuned to 260 Hz. (Refer to Table 16.3.)

16.16 If A4 were tuned to 440 Hz in Exercise 16.15, what would the frequency of C4 be?

16.1. Determine the frequency of A4 in a diatonic scale with just tuning if C4 is tuned to 260 Hz. (Refer to Table 16.3.)

16.18 Determine the frequency of A4 in a diatonic scale with equal-tempered tuning if C4 is tuned to 260 Hz. (Refer to Table 16.3.)

16.19 Name the intervals formed by the following note pairs: C4/C♯4; C4/D4; C4/E4; C4/F4; C4/G4; G4/ A4; and G4/C5. (Refer to Table 16.3.)

16.20 Express the intervals of the note pairs in Exercise 16.19 in semitones.

16.21 Express the intervals of the note pairs in Exercise 16.19 in frequency ratios assuming equal-tempered tuning.

16.22 What three intervals (by name, semitones, and ratio) occur in the C-major triad C5/E5/G5? What three intervals occur in the C-minor triad C5/D♯5/G5?

16.23 We will see later (in Chapter 34) that a clarinet can be approximated as a closed-open tube having natural mode frequencies related by the odd integers. The lowest note on a clarinet is D3. By opening a register hole a note with three times the frequency is possible. This interval is referred to as a twelfth. What is it in semitones? What interval can be combined with an octave to give a twelfth?

16.24 In addition to our semitone scale with 12 equal intervals in the octave, scales with 19, 24, and 53 equal intervals in the octave have been proposed. How close could the "best" notes in a 19-tone scale come to those in a justly tuned diatonic scale?

16.25 How close could the "best" notes in a 24-tone scale come to those in a justly tuned diatonic scale?

16.26 How close could the "best" notes in a 53-tone scale come to those in a justly tuned diatonic scale?

16.27 It is stated in the text that sinusoids differing in frequency by one-quarter of a critical band produce maximum roughness. This suggests that musical intervals will have different amounts of roughness depending on the frequency range in which they are sounded. Imagine that the tone pair C4/C♯4 is sounded and then successive chromatic tone pairs C4/D4 ... C4/C5. What predictions can you make about the relative roughness of each pair? Would things change if C1/C♯1 were the starting pair? If C7/C♯7 were the starting pair? Does the number of partials in each tone in a pair play any role?

16.28 Do the predictions made in Exercise 16.27 depend in any way on the number of harmonics present in the tones in a pair? Consider, for example, sinusoids, square waves, sawtooth waves, piano tones, bowed string tones, and various wind instrument tones.

16.29 Do the considerations in Exercise 16.28 suggest which tones might be better for tuning? (Hint: Greater potential roughness implies greater ease of tuning.)

16.30 What musical intervals can be combined to form an octave? Verify your predictions by multiplying the frequency ratios of the intervals to see if you get a result of two. Verify your predictions by adding the intervals expressed in cents to see if you get a result of 1200.

16.31 Assume that the frequency difference producing maximum roughness when two sinusoids are sounded together is approximately equal to one-quarter of a critical band of their average frequency. What frequencies (both lower and higher) will produce maximum roughness when sounded with a frequency of 250 Hz? 500 Hz? 1000 Hz? 2000 Hz?

16.32 Tuning forks designed for use in science labs were tuned to C3 = 128 Hz, C4 = 256 Hz, and C5 = 512 Hz. Consider corresponding Cs tuned to A4 = 440 Hz and determine how many cents sharp they are relative to the lab tuning fork Cs. Why is the result the same in all three cases?

Activities

16.1 Use a monochord to recreate some of the Pythagorean consonance experiments.

16.2 Set up two function generators to operate into a mixer, scope, and speaker system. Check perceived roughness as a function of waveform and frequency difference.

16.3 Explore many two- and three-note combinations on the piano in the low, mid, and high octaves.

16.4 Sound the tone pairs suggested in Exercise 16.27 and rate each in terms of roughness. Compare your rating with the predictions of the exercise.

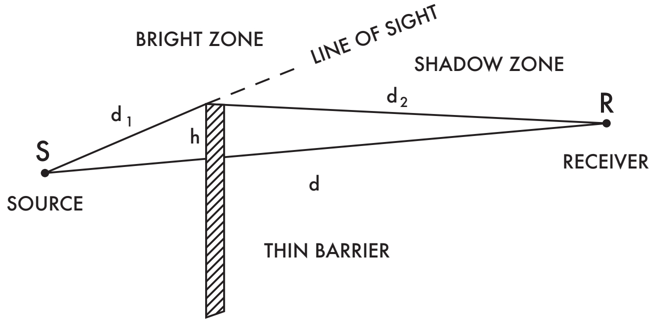

Figure 20.2 *Source-to-receiver relationships across a thin barrier.*

wall must be massive and without openings so that sound will not be transmitted through the wall. The only sound then able to reach the listener is diffracted over the top of the wall. The reduction in sound level is given approximately as

$$R = 10 \times \log(3+10N) \qquad (20.1)$$

where N = (2/λ)x(d1+d2-d) is the Fresnel number.

Note that lower frequencies with longer wavelengths have smaller N values because they diffract more sound over the wall. Also note that making the wall higher will result in higher N values because d1+d2-d will be larger. As an example, suppose in Figure 20.1 h=3m, d1=d2= 10m, so d=19m. At f=100 Hz lambda=3.4m, N=0.59, and R=9.5 dB. At f=1000 Hz lambda=0.34m, N=5.9, and R=18 dB. Thick walls produce double diffraction and may provide several decibels of additional attenuation.

20.4 Enclosures as Barriers

Even under the best circumstances, with control of noise emission, zoning, and external barriers, significant amounts of unwanted noise may still remain. Some sources of external noise and their transmission paths are illustrated in Figure 20.3. These can be prevented from entering a living/listening environment by enclosing the environment. The enclosure serves as a barrier to sounds arriving from external sources.

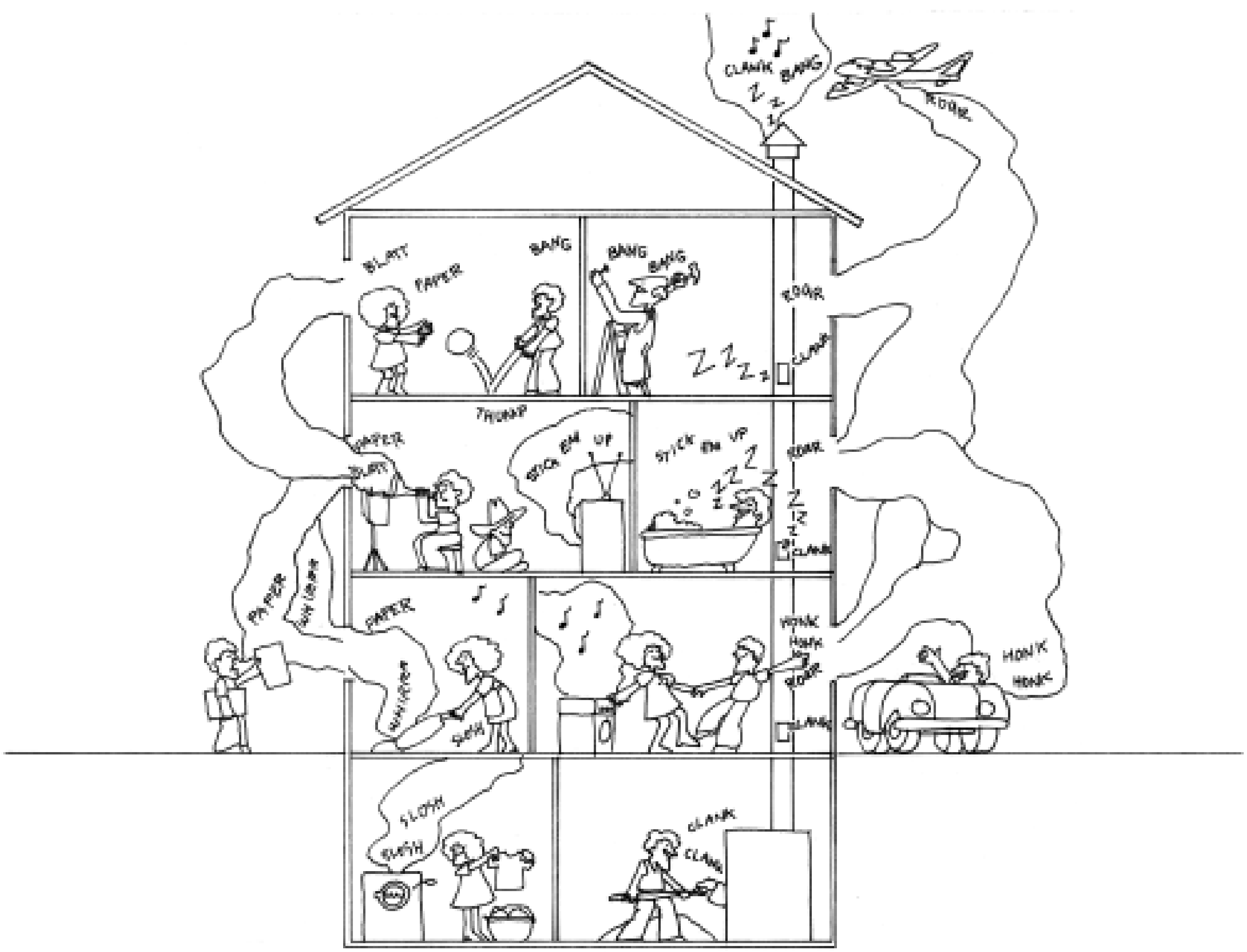

Figure 20.3 *Some external and internal sources of sound and some of the paths through which they are transmitted.*

As an example of a structure where the external environment was a particular problem, consider the J.F. Kennedy Center for the Performing Arts in Washington, D.C. The Center is near the Potomac River and close to the National Airport. Aircraft often fly as low as 100 meters above the roof, and low-flying helicopters are often seen in the immediate vicinity of the Center. In addition, the structure is surrounded by the usual noises of automotive traffic. The design used to suppress external noise in the Center is the "box-within-a-box" concept. The Center's three auditoriums are completely enclosed within an exterior shell. The exterior shell is a double-walled construction with enclosed air spaces. The columns supporting each auditorium have been designed to isolate both airborne noise and mechanical vibrations from the interior surfaces. At all outside entrances special doors are used, and there is a "soundlock" region between the foyer and the interior of each auditorium. Because of this special construction it is possible for patrons to enjoy performances free of interference from external noise.

20.5 Sound in Enclosures

Once a location has been selected for a living or listening structure, typical noise conditions at the site should be measured so that appropriate construction materials can be used to provide acceptable internal background noise levels. The "acceptability" of noise levels depends on the nature of the sound, as discussed in Chapter 18. However, for noises without distinct tone colors it is possible to specify a meaningful noise level in terms of a single criterion. The **noise criteria** (NC) is the number used to rate noise levels, given approximately by NC = 1.24 (dBA - 13), where dBA is the A-weighted sound level. Typical acceptable NC values are shown in Table 20.1 for various environments. The external sound level at the site, minus the acceptable background level, gives the required transmission loss . Building materials can then be selected to achieve this transmission loss.

Table 20.1 *Typical acceptable noise levels in enclosures.*

Enclosure	NC Value
Studio	15–20
Concert Hall	15–20
Theater	20–25
Auditorium	25–30
Bedroom	25–30
Living Room	30–35
Business Offices	30–35
Restaurant	35–45

20.6 Barrier Materials

Barrier materials should satisfy two basic conditions: they should be massive, and they should be airtight. Sound waves are transmitted effectively from one medium to another when the two media have similar densities and sound speeds. However, when sound waves in air strike a massive barrier, they are mostly reflected because of the mismatch in sound speed and density between the air and the barrier. Even though a barrier is massive, it will not be effective unless it is made fairly airtight. For example, a poorly fitting window may have cracks around the edge equal in area to 1 % of the window area. Of the total sound energy striking the window, about 1–4% (depending on frequency) will leak into the building. This limits the attenuation produced by the window to 14 dB for 4% leakage or 20 dB for 1% leakage, while an airtight window may provide a loss of 30 dB. Poor-quality construction can negate the value of an otherwise good sound barrier.

When selecting materials to be used in sound barriers, one must consider their sound transmission-loss characteristics. Typical transmission-loss characteristics are shown in Table 20.2 for various materials and structures. Generally,

Table 20.2 *Typical sound transmission losses for various acoustical barriers. The losses expressed in dB are for the frequencies shown and assume no leakage around the barrier.*

	Frequency (Hz)		
Barrier	125	500	2000
Solid wall			
Density of 0.25 kg/m^2	-	23	Average
Density of 1.0 kg/m^2	-	29	for
Density of 5.0 kg/m^2	-	38	100 Hz
Density of 25 kg/m^2	-	50	to 3200 Hz
Double wall with air core – increase in loss over solid wall with same mass.			
4 cm air core	1	2	8
15 cm air core	4	11	18
Double wall with filled core			
17 cm foam core	28	51	61
6 cm foam core and 1 cm soundboard	27	45	58
10 cm mineral wool and 1 cm soundboard	28	46	60
Hollow core door (14 kg)	11	16	22
Solid door (27 kg)	15	14	25
Solid core door (42 kg)	20	14	26
Solid glass window (3mm)	12	20	25
Insulating glass window (10 mm)	17	19	27
Insulating window with storm sash	16	27	35

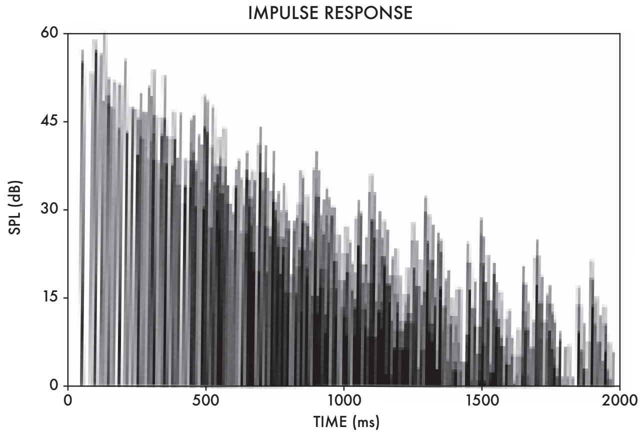

Figure 21.9 *Simulated microphone response for sound in a two-dimensional room such as shown in Figure 21.8. (Courtesy of S. N. Li.)*

have a spectrum not too different from that of the direct sound, and it must not be too much louder than the direct sound.

The arrival time of early sound is controlled by the distance of reflecting surfaces from source and listener. The first reflection from the nearest wall (shown at top of Figure 21.8) will be the first early sound to arrive at the listener. First reflections from more distant walls, along with multiple reflections, will arrive later, but still soon enough to be part of the early sound.

The earliest reflections arriving within 20–30 ms of the direct sound play a special role in determining the "intimacy" of a concert hall. This early sound should come from the walls rather than the ceiling because people prefer sound lying in the plane of the ears and the source. Halls designed to provide early lateral reflections should be rectangular in shape and should not be too wide. In fan-shaped halls, the earliest reflections to reach the listener come from the ceiling and not from the walls.

Reverberant sound is all reflected sound arriving later than the early sound. The purpose for which an enclosure is designed largely determines how the early and reverberant sounds should be managed. Rooms designed for speech generally require a reduction in the reverberant sound so that spoken sounds at a given instant are not blended with and masked by previously spoken sounds. Speech intelligibility requires a large amount of early sound compared to reverberant sound. Increasingly more reverberant sound is desirable for small musical ensembles, opera, orchestra, and organ (in that order). Reverberant sound plays an important role in determining the "liveness" and the sense of being immersed in sound. The longer the reverberant sound lingers the greater the sense of immersion. The reverberant sound is controlled by the absorptive properties of surfaces and objects in a room.

21.5 Reverberation Time

A hypothetical one-dimensional room is a useful conceptual device used to introduce the concept of "reverberation time," the time required for a sound in an enclosure to "die out." We take as our one-dimensional room a hard-walled tube 34 m in length, as shown in Figure 21.10. We imagine that a pulse of sound is somehow introduced into the tube so that it bounces back and forth between opposite ends of the tube. We assume that no sound energy is lost to the air in the tube or to the sides of the tube, and that all losses are at the ends of the tube. (In many cases, such as with a wind instrument, this assumption does not hold true. As a wave travels in a tube, part of the sound energy is absorbed by the walls of the tube so that the pulse weakens as it travels.) If the ends of the tube were completely nonabsorbing, the pulse would bounce back and forth forever. If the ends of the tube were moderately absorbing, the pulse would bounce back and forth for a long time, but would eventually die out. If the ends of the tube were highly absorbing, the pulse would die out quickly.

We can place a microphone at the midpoint of the tube (see Figure 21.10) and measure how quickly the pulse dies out. The microphone will detect the pulse once every trip along the tube, or, because $t = L/v$, once every $t = 0.1$ s. For purposes of illustration we consider two cases, one with small absorption and another with large absorption. Case 1 assumes that 27% of the energy incident on a tube end is absorbed and 63% is reflected. (Since pressure amplitude varies as the square root of energy, the pressure amplitude of the reflected pulse will be $\sqrt{0.63}$ = 79% of that of the incident wave.) The sound level decrease (in dB) after each reflection can be determined by taking ten times the logarithm of the fraction of energy reflected. The sound level decrease for this case is just $10 \times \log(0.63) = -2$ dB. Case 2 assumes that 75% of the incident energy is absorbed and 25% is reflected, resulting in a decrease of 6 dB per reflection. Sound levels of the reflected pulse are shown at one-tenth-second intervals for the two cases in Figure 21.11.

On examining Figure 21.11 we see that the pulse "dies out" in an amount of time that depends on how often it encounters an absorbing surface, and what fraction of its energy is lost on each encounter. The **reverberation time**

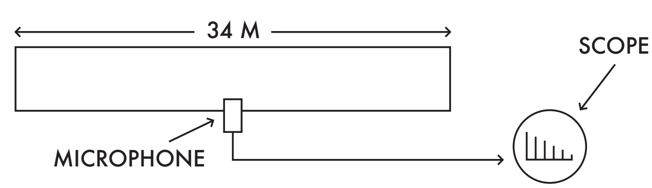

Figure 21.10 *Hard-walled tube in which a stimulated pulse of sound bounces back and forth between the partially absorbing ends. The "display" on the scope is the output of the microphone.*

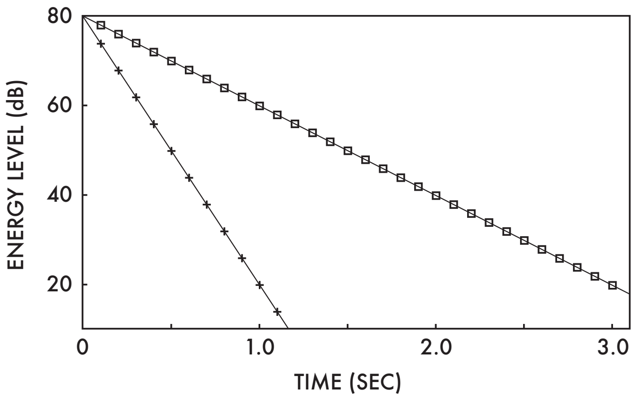

Figure 21.11 *Energy levels of successively reflected sound pulses in the 34 m tube of Figure 21.10. No energy is lost to the walls. Two conditions at the ends of the tube are considered: one in which 63% of the energy is reflected (□) and the other in which 25% of the energy is reflected (+).*

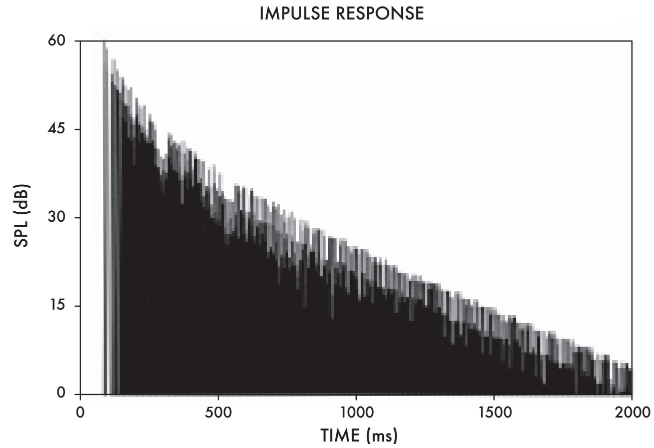

Figure 21.12 *Simulated microphone response for a sound pulse in a large (30 m × 23 m × 18 m) concert hall. (Courtesy of S. N. Li.)*

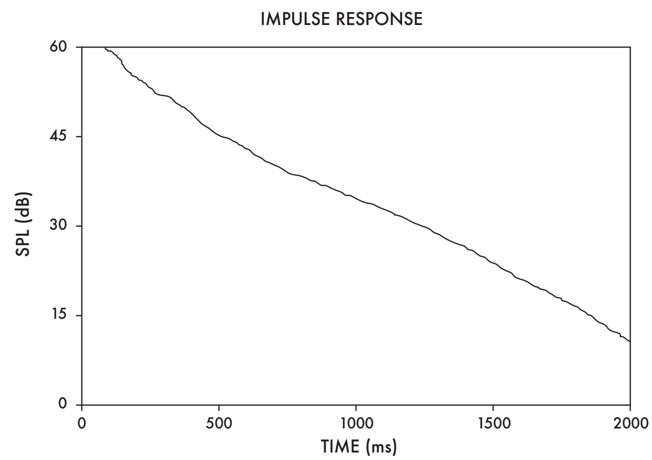

Figure 21.13 *Smoothed microphone response for the simulated microphone response of Figure 21.12. (Courtesy of S. N. Li.)*

(RT) is specifically defined as the time required for the sound level to decrease by 60 dB. We note that for case 1 the sound level decreases by 2 dB each tenth of a second so that its RT is three seconds. For case 2 the sound level decreases by 6 dB each tenth of a second so that it has decreased 60 dB in one second and has a RT of one second. The larger the amount of absorption the shorter the RT, and RT may be expressed as being inversely proportional to absorption. For the sound pulse in a tube the RT also depends on the length of the tube and the speed of sound in the tube. If the pulse moved faster or if the tube were shorter, the RT would be shorter because the pulse would strike the ends of the tube more times per second.

21.6 Reverberation Time in Rooms

The behavior of sound in three-dimensional enclosures such as auditoriums and concert halls is of primary interest in building design. Although the three-dimensional nature of rooms makes reflected sounds harder to visualize, the microphone response (see Figure 21.12) is similar to that found in two-dimensional rooms. The direct sound and the first few reflected sounds are quite distinct, followed by a nearly continuous, diffuse sequence of reflected sounds.

A smoothed version of the microphone response in a room is usually used to estimate the RT of the room. Such a smoothed response is shown in Figure 21.13 for the microphone response of Figure 21.12.

The RT in three-dimensional structures is of significant interest in building design. As might be expected on the basis of the one-dimensional example, RT varies inversely with absorption, but it depends on the volume of the enclosure (rather than on length) as in the one-dimensional example. An expression for reverberation time, called the Sabine equation, is

$$RT = \frac{0.16\,V}{TA} \qquad (21.1)$$

where V is the room volume in cubic meters and TA is total absorption associated with the interior surfaces of the room. If all the surfaces of the room were completely absorbing, TA would be equal to the surface area of the room in square meters. (The only perfect absorber, however, is an opening to outside free space.) If, instead of a perfect absorber, we have twice the area of some material which absorbs half of the incident sound energy, the resulting TA will be the same. The **absorption coefficient** (AC) of any material is defined as the fraction of energy absorbed on each reflection of a sound wave. Thus, we can see that for any surface area, S,

$$TA = S \times AC \qquad (21.2)$$

When many different surfaces are present the total absorption is given by

$$TA = S_1 \times AC_1 + S_2 \times AC_2 + S_3 \times AC_3 \qquad (21.3)$$

where the subscripts represent different surfaces. It will be noted that TA has the dimensions of an area and that it is smaller in magnitude than the total surface area of the room.

21.7 Example of Reverberation Time

As one example, consider a room 8 × 10 × 6 m with heavy carpet on the floor, acoustical plaster on the ceiling, and 0.30 cm plywood paneling on the walls. Assume that 50 people are present in the room. Calculation of RT requires absorption coefficients, typical values of which are shown in Table 21.1. (Note that the values vary with frequency for any given material and that most materials are more highly absorbing at higher frequencies.)

The RT at 500 Hz is calculated in the following manner. The room volume is given by V = 8 × 10 × 6 = 480 m^3. The total absorption is given by

TA = 8 × 10 × 0.60 (floor) + 8 × 10 × 0.50 (ceiling) + 2 × 8 × 6 × 0.10 (end walls) + 2 × 10 × 6 × 0.10 (side walls) + 50 × 0.45 (people) = 132.1 m^2.

We then obtain the RT from the Sabine equation as

$$RT = 0.16 \times 480/132.1 = 0.58\ s.$$

From Table 21.1, we see that absorption coefficients for different materials vary with frequency, which means that RT will be different at different frequencies. Some contrasting examples of RT as a function of frequency are shown in Figure 21.14. The anechoic room is seen to have extremely short RTs because of its highly absorbing walls (see Figure 21.3). The racquetball court is seen to have very long RTs because of its hard reflecting walls. The two classrooms have RTs somewhere between the anechoic room and the racquetball court.

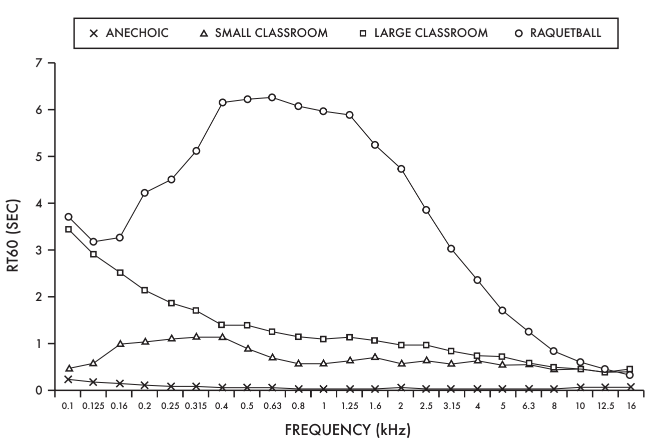

Figure 21.14 *Measured RTs in one-third-octave frequency bands for four different rooms. (Courtesy of R. H. Brey and R. W. Harris.)*

Table 21.1 *Typical absorption coefficients for some building materials. Actual values depend on backing and mounting of material. TA values for adult person and upholstered chair are expressed in square meters.*

	Frequency (Hz)		
Material	125	500	2000
Acoustical plaster	0.15	0.50	0.70
Acoustical tile	0.20	0.65	0.65
Brick	0.02	0.03	0.05
Carpeted floor			
heavy, on heavy pad	0.10	0.60	0.65
light, without pad	0.08	0.20	0.60
Concrete	0.01	0.01	0.02
Draperies			
heavy	0.15	0.55	0.70
light	0.03	0.15	0.40
Fiberglass blanket			
2.5 cm thick	0.30	0.70	0.80
7.5 cm thick	0.60	0.95	0.80
Glazed tile	0.01	0.01	0.02
Paneling – plywood supported at 1m intervals and backed with 5-cm air space			
0.15 cm thick	0.10	0.20	0.06
0.30 cm thick	0.30	0.10	0.08
Plaster	0.04	0.05	0.05
Vinyl floor on concrete	0.02	0.03	0.04
Wood floor	0.06	0.06	0.06
Adult person (TA)	0.30	0.45	0.55

21.8 Ambient Sound Levels

So far we have discussed the relationship of the RT of a room to the room volume and the room absorption. The buildup time of the sound and the final sound level produced by a constant-power sound source in a room are two related aspects of interest. If a sound source having constant power output is placed in a room, the sound level in the room will increase until it reaches some final value, at which time the sound power being absorbed is just equal to the sound power being supplied by the source.

The relationship between absorption, power input, RT, buildup time, and final sound level can be illustrated using a bucket with small holes (representing absorption) punched in its side from bottom to top. Water from a garden hose provides a steady flow rate (source of constant power). Water from the hose pours into the bucket and the time required (buildup time) for the water level to reach its highest possible point (final sound level) is observed. The hose is removed and the time required for the water to drain from the bucket (RT) is observed. The buildup time and the RT both tend to be of about the same length.

If the holes in the bucket are small (small absorption), both times are relatively long and the final water level (sound level) is high. For large holes both times are relatively short and the final level is low.

Now consider the large concert hall of Figures 21.12 and 21.13 with a volume of 12,420 m^3 and a total absorption of 994 m^2, so that its RT is 2.0 s. A 10 watt sound source is placed at one position in the hall and a microphone is placed at a position some distance from the source. When the source is turned on, the sound level (measured at the microphone) initially increases as illustrated by the curve labeled RT = 2.0 in Figure 21.15. After the sound level reaches its final value, it remains constant as long as the sound source is on. When the source is turned off the sound level decreases until some background level is reached. A concert hall having the same volume but four times the total absorption would have the sound level curve labeled RT = 0.5 in Figure 21.15. Comparing the two curves we see that the greater the absorption, the shorter the buildup time and RT, and the lower the final sound level.

A formula that gives the final sound level in a room in terms of the sound power of the source and the total absorption of the room is

$$SL = 10 \log(P/10\text{-}12) + 10 \log(4/TA) \qquad (21.4)$$

where P is the source power in watts and TA is the total absorption in square meters. Referring to the first example in Figure 21.15, we calculate the final sound level as

$$SL = 10 \log(10/10\text{-}12) + 10 \log(4/994) = 106 \text{ dB}$$

A similar calculation for the second example in Figure 21.15 is

$$SL = 10 \log(10/10\text{-}12) + 10 \log(4/3976) = 100 \text{ dB}$$

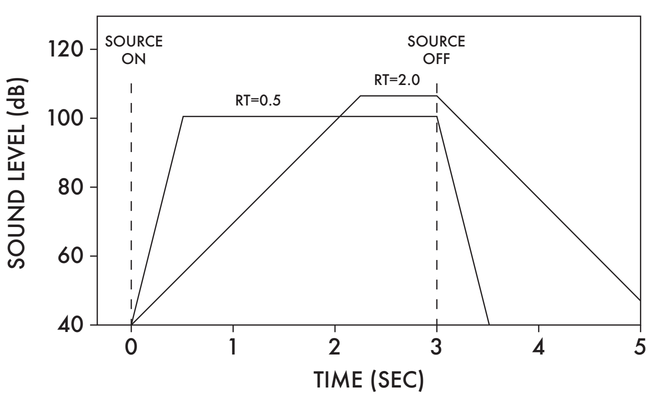

Figure 21.15 *Curves showing sound-level increase, final sound level, and sound-level decrease for two rooms. Both rooms have the same volume but differ in acoustical absorption and RT.*

21.9 Nonuniform Reverberation

An additional aspect of reverberation not obvious from the foregoing discussion is that reverberant sound should decay uniformly. A nearly ideal reverberation curve is shown in Figure 21.13, in which the reverberant sound level dies away uniformly with time in almost a straight-line. This kind of uniform reverberation response is highly desirable because the sound is getting "soft" at an even rate. Figure 21.16 shows a highly irregular response in which the sound level decreases, then increases, then decreases again in a kind of erratic warble. Generally, the more diffuse the sound field, the less likely the chance of finding this type of irregular curve. Finally, Figure 21.17 shows another situation which is very undesirable, even though the reverberation curve is smooth. In this illustration, the reverberant sound falls off rapidly during the first half-second, thus telling the ear that the RT will be short. In actuality, RT is 2.0 s as

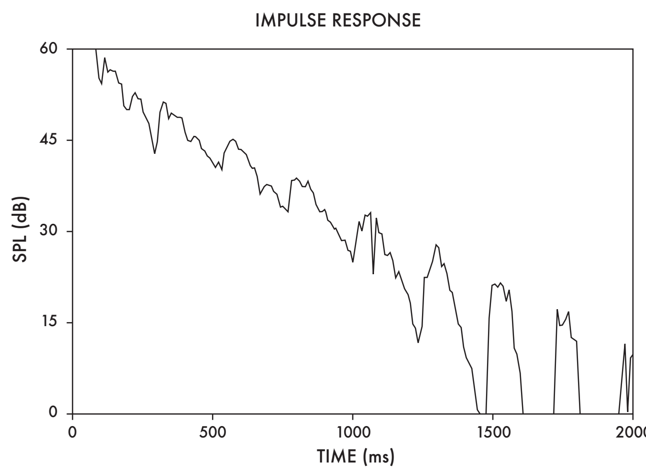

Figure 21.16 *Smoothed microphone response curve showing irregular reverberant-energy decay. (Courtesy of S. N. Li.)*

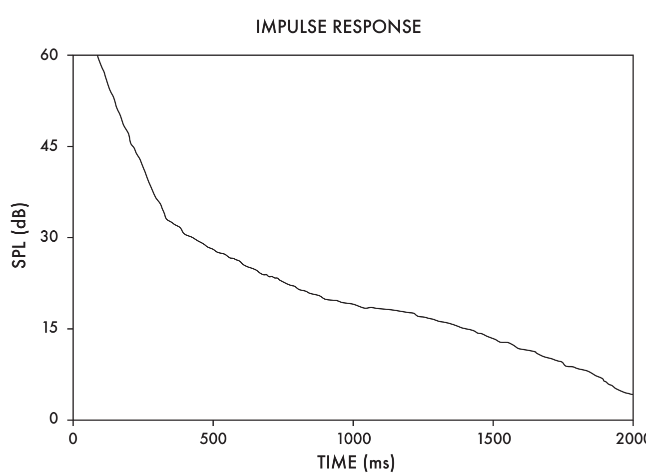

Figure 21.17 *Smoothed microphone response curve showing double-sloped reverberant-energy decay. (Courtesy of S. N. Li)*

common building materials. Generalizing his empirical data, he obtained the well-known formula for reverberation time which enables one to calculate RT as a function of volume and absorption. He could then make his contribution to the acoustical design of the symphony hall.

21.14 Summary

Sound in an enclosure can travel directly from source to receiver, it can undergo specular and diffuse reflections, it can be diffracted, it can be transmitted through walls, and it can be absorbed. The acoustical properties of an enclosure are determined by the interplay of these behaviors. An anechoic room and a reverberation room lie at the extremes of room acoustical properties. Defects to avoid in rooms are: echoes, sound focusing, sound shadows, flutter, distortion, room resonances, and acoustical glare. Three types of sound in rooms are direct, early (within 80 ms of direct), and reverberant (all later sound). Reverberation time is the time required for a sound level to decrease by 60 dB. Reverberation time depends on room volume and—inversely—on total room absorption. Ambient sound level in a room depends on source power and total room absorption. Irregular and/or double-sloped room decay is undesirable. Optimum reverberation time depends on room size and program material. Various methods are used to measure the reverberation time of a room.

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Questions

21.1 The level of reverberant sound necessary for optimum results varies between speech and music. Why should the optimal level of reverberant sound generally be less for speech than for music?

21.2 What type of music requires the greatest reverberant sound for optimum results? What type of music requires the least reverberant sound for optimum results?

21.3 Describe qualitatively the absorption characteristics and reverberation time you might expect to find in each of the following rooms: concrete-walled room, living room, bathroom, anechoic room, concert hall, recording studio, and wood-paneled room.

21.4 Discuss the acoustical reason for each of the modifications made in the Frostburg State University lecture hall.

21.5 Explain why miles of wire stretched across the ceiling of a reverberant room absorb only a negligible amount of sound energy.

21.6 The large lecture hall whose RT is shown in Figure 21.4 had poor speech intelligibility. Why?

Exercises

21.1 Suppose people in the front row of a room receive strong reflections from the back wall 20 m away. Will the reflections be perceived as an echo?

21.2 Suppose a time delay of 200 ms produces maximum interference for a lecturer. How far away from the lecturer is a rear wall that will produce reflected waves giving rise to maximum interference? How does this distance compare with the distance between the stage and the back wall in typical theaters and concert halls?

21.3 What is the repetition rate of flutter echoes in a hall having parallel side walls 20 m apart? Does the repetition rate depend on where a listener stands? How can these flutter echoes be reduced?

21.4 A concert hall has dimensions of $30 \times 23 \times 18$ m. A sound source is located at one end of the hall. Are there any places in the hall where strong reflections from the ceiling or walls will produce echoes? (Make some sketches of listener positions in the hall to get path lengths and travel times of reflections.)

21.5 Consider an auditorium 30 × 20 × 15 m high. Assume that the floor is concrete, with an average of one upholstered chair every square meter. Assume that the ceiling and walls are plastered. Calculate the RT at 500 Hz. What is the sound level in the hall if the orchestra produces a sound power of 2 W?

21.6 What would the RT be for the auditorium for Exercise 21.5 if the seats were made of wood? If half of the wood seats were filled with people?

21.7 A listener is seated 5 m from the rear wall, 7 m from one sidewall, and 13 m from the other sidewall in the auditorium of Exercise 21.5. Does the first reflected sound come from a wall or the ceiling? Determine where the first several reflections come from and their time delays relative to the direct sound. Will the precedence effect play an important role in this auditorium?

21.8 Compare the absorption coefficients for concrete, plaster, and plywood at 500 Hz. What TA value would 100 m^2 of each material provide?

21.9 What are reasonable values for RT in a 300 m^3 hall at frequencies of 125, 250, 500, 1000, and 2000 Hz if the hall is to be used for speech? For opera? For orchestral music?

21.10 Calculate RT for a 3000 m^3 hall with all concrete surfaces. Would it be satisfactory for speech? For organ music?

21.11 Suppose that a halving of loudness from one syllable to the next is required for good speech intelligibility. If a SL decrease of 10 dB cuts the loudness in half, will a RT of 2 seconds be adequate for fast speech spoken at a rate of 8 syllables per second? What RT would be required for speech spoken at this rate? What speaking rate should be used with the original RT of 2 seconds? (Remember that RT is the time required for the SL to decrease by 60 dB.)

21.12 What RT would be optimum for a Brahms symphony in a hall of 3000 m^3 volume? In a hall of 12,000 m^3 volume?

21.13 A small church often used for concerts has no heating or cooling system so patrons wear light clothing in summer and heavy clothing in winter when attending concerts. The hall has a floor area of 300 m^2 with an average AC of 0.4 when empty, 0.6 when occupied in summer, and 0.8 when occupied in winter. What are the RT values for the three conditions if the average AC is 0.2 for the other surfaces?

21.14 A band rehearsal room of 3000 m_3 volume has concrete surfaces everywhere. What is its RT with a 100 member band present? Is this likely to be satisfactory? How much TA must be added to reduce the RT to 1 second? Would this be feasible to accomplish at 500 Hz with light drapery? With acoustical tile?

21.15 A small ensemble can generate 1 watt of acoustic power. What SL can it produce in a small (300 m^3) practice room with an RT of 1 second? What SL can it produce in a small (3000 m^3) auditorium with an RT of 1.2 second?

21.16 Some modern auditoriums use reversible panels to adjust the RT. What is the shortest RT that can be achieved in a 20 × 20 × 20 m enclosure if 200 m^2 of panel are perfectly absorbing and the rest of the enclosure has an average AC of 0.2? What is the longest RT possible if the panels are perfectly reflecting?

21.17 What is the RT at 500 Hz in a 6 × 8 × 3 m living room with heavy carpet on the floor and plastered walls and ceiling? How adequate would this be for normal conversation? For playing the piano?

21.18 If the wall and ceiling surfaces of the room in Exercise 21.17 were covered with materials having an average AC of 0.3 what would the RT be? Would the room be improved for conversation? For playing the piano?

21.19 Use equation 21.1 to eliminate TA from equation 21.4 Does the resulting equation justify the claim that sound level can be kept constant if RT is increased with volume?

Activities

21.1 Make a sharp clap or pop a balloon to provide an impulsive sound source in various rooms, large and small. Comment on the RT you experience.

21.2 Choose an auditorium known to have good acoustics. Perform various qualitative tests to estimate its RT and other sound properties.

21.3 Choose a poorly designed auditorium and perform various tests to estimate its RT and other sound properties.

21.4 Experiment using cans or pipes with various-sized holes punched in them (as described in the text) to demonstrate the effect of total absorption and source power on final sound level and RT.

22.2 Will broadening of the modes in Figure 22.4 to widths of 5 Hz result in an even room response? Will broadening to 10 Hz?

Exercises

22.1 Change the length value of the 9x6x4 m room to 10 m. What width value is required to keep the volume constant? Calculate frequencies similar to those for the modified room dimensions in the text. Are these in any way better?

22.2 Calculate the length-height tangential mode frequencies that lie within the 105–115 Hz range for the concert hall in Section 22.2.

22.3 Calculate the width-height tangential mode frequencies that lie within the 105–115 Hz range for the concert hall in Section 22.2.

22.4 Calculate ten oblique mode frequencies that lie within the 105–115 Hz frequency range for the concert hall of Section 22.2.

22.5 Scale the dimensions of the studio in Section 22.4 to accommodate 10–15 musicians. Calculate the crossover frequency and the lowest mode frequency. Calculate axial mode frequencies within this range.

22.6 A listening room has dimensions of 6.6 x 5.0 x 3.0 m. What frequencies lying below 200 Hz will be emphasized? What frequencies will be under emphasized? Assume bandwidths of 6 Hz.

22.7 A large family room (9.6m × 4.8m × 3.6m) serving as a listening room was found to have a very strong response to an organ tonal mass at a particular frequency. What was the most likely frequency?

22.8 The first reflection in a studio is received 15 ms after the direct sound. The control room is to be designed so that it's first reflected sound is received by the mixing engineer 20 ms after the direct sound from the monitor loudspeakers. If the engineer is positioned midway between the front and back walls of the control room, what should be the length of the control room if the first reflection comes from the back wall?

Activities

22.1 Place a sinusoidally driven loudspeaker in the corner of a hard walled room with a microphone placed in a diagonally opposite corner. Record the room response as the frequency is varied. Beyond what frequency do you get a fairly even response?

22.2 Place a sinusoidally driven loudspeaker in the corner of a living room with a microphone placed in a diagonally opposite corner. Record the room response as the frequency is varied. Beyond what frequency do you get a fairly even response?

22.3 Design a listening room limited by a height of 3m and a volume of approximately 200 m^3. Determine frequencies that are most likely to give coloration. Specify how these frequencies are to be handled.

What sound level would be produced at 200 m under the same conditions?

24.3 Estimate the surface area over which the sound will be spread at a distance of 100 m from the loudspeaker clusters in Figure 24.10. If the sound level is 100 dB at this distance, what total acoustical power passes over the surface? Estimate the electrical signal power supplied to the loudspeakers if they are 10% efficient in converting electrical power into acoustical power.

24.4 The RT in the Marriott Center is 4 s. What is the average absorption coefficient of its surfaces if one assumes its shape is square with a height of 14 m?

24.5 By how many dB does the one-third octave at 1600 Hz exceed the preferred response in Figure 24.15?

24.6 About 3900 W of electrical power are required to produce 23 W of acoustical power in the Marriott Center. What is the efficiency of the sound system?

24.7 How do the assisted RTs in Figure 24.17 compare with optimum RTs discussed in Chapter 21 for a concert hall with a volume of 25,000 m3?

24.8 Suppose two loudspeaker clusters are mounted next to the walls in a church with floor dimensions of 40 m by 20 m wide. Will a person sitting next to a wall and 10 m back from one cluster hear an "echo" if both clusters radiate the same signal?

24.9 If distributed loudspeakers are mounted 7 m apart from front to rear in a hall, what should the signal delay time be between neighboring loudspeakers?

24.10 Suppose a digital delay line is used to provide the delay of Exercise 24.9. In the digital delay the signal is sampled and the samples are stored in memory. There must be enough memory to store a number of samples equivalent to the time delay. How much memory is required if the sampling rate is 20,000 samples per second?

24.11 If a tape loop system (now obsolete) is used to provide the time delay of Exercise 24.9, what must the tape speed be if the record and playback heads are 1 cm apart?

24.12 A listener is 100 m away from a sound source where the critical distance is 25 m. How many dB below the reverberant sound will the direct sound be at the listener's position?

24.13 A room has a volume of 2600 m3 and total absorption of 400 m^2. Approximately what acoustical power is necessary to achieve a sound level of 80 dB? (Refer to Chapter 21.) What electrical input power is required if the speakers are 10% efficient?

24.14 How far from the source are the direct and reverberant sound levels equal in a room of $6 \times 5 \times 3$ m with RT = 0.5 s? In a room of $30 \times 23 \times 15$ m with RT = 2.0 s? Is it true that reverberant sound dominates in most of a room?

24.15 What is the largest room in which a 0.0002 watt voice source can produce a sound level of 70 dB without sound reinforcement? Do factors other than room volume play a role? If so, make plausible assumptions in answering the question.

24.16 An important criterion for flat frequency response of a microphone is that the diaphragm diameter be no larger than half a wavelength. What is the maximum diaphragm diameter for speech with frequencies up to 10,000 Hz? For the upper end of the audio range at 20,000 Hz?

24.17 What is the efficiency of a loudspeaker producing a sound level of 95 dB at a distance of 10 m when the electrical power input is 5 W? Calculate values for radiation patterns assumed to be spherical and hemispherical.

24.18 At what lowest frequency would self-enhancing feedback occur in a sound-reinforcement system where the microphone is 6 m in front of the loudspeaker? What higher frequencies would result in self-enhancing feedback?

Activities

24.1 Set up in a room a sound-reinforcement system consisting of microphone, amplifier, and loudspeaker. Experiment with amplifier gain to see how it affects feedback resulting in squeal.

24.2 Move the microphone and loudspeaker of Activity 24.1 to different locations to see how feedback depends on location.

24.3 Gain access to halls in which sound systems have been installed to meet different needs. Experiment with different adjustments of the sound system if possible to see how they affect the sound. Explore the effects of signal delay and equalization if possible.

to airflow changes rather drastically as the glottal area changes. In practice, doubling the area more than doubles the airflow (especially for small areas) if all other conditions are constant. (3) The pressure across the glottis is not constant. Even if we assume the pressure in the trachea to be constant, the pressure in the larynx will be fluctuating because of standing waves in the vocal tract. The fluctuating pressure across the glottis gives rise to fluctuations in the airflow.

Standing waves in the vocal tract will affect the pressure across the glottis. This in turn affects the glottal airflow. The glottal airflow affects the strength of the Bernoulli force which drives the vocal folds. Standing waves that typically occur in the vocal tract do not have much influence on the vocal folds themselves. However, when vocal tract resonances occur at low frequencies, they may influence vocal-fold vibration frequency. Later, when we come to consider the lips as a similar vibrating system for wind instruments, we will find that the tube of the instrument exerts a very substantial influence on the lips.

An idealized spectrum typical of the nominal glottal airflow of Figure 25.8 is shown in Figure 25.9. In this example the airflow is shut off for part of a cycle. The resulting flow is pulse-like. The flow begins rather gradually after being shut off, but then is shut off rather abruptly after reaching some largest value. The spectrum shows a rich harmonic content because of the pulse-like nature of the wave.

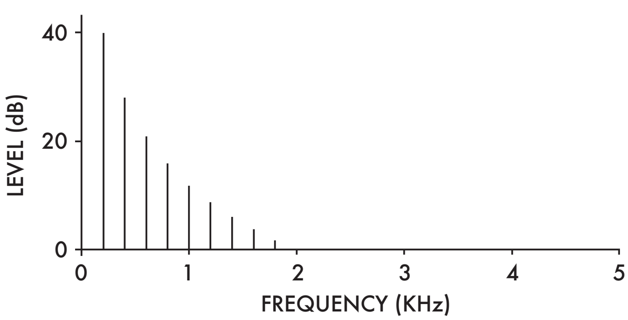

Figure 25.9 *Idealized spectrum of glottal airflow. The largest harmonic is arbitrarily assigned a level of 40 dB.*

25.5 Vocal-Fold Oscillation

Muscular adjustments establish an equilibrium position for the vocal folds in preparation for voice production. Air pressure from below and between the folds pushes them apart. The muscular tension in the folds opposes this motion and tends to pull them back together. Potential energy is stored in the folds as they are pushed apart, but is partly recovered when the folds return to their equilibrium position. Because there is resistance, the vocal fold motion will die out unless energy is supplied to them. The air "driving" them from below and rushing between them supplies the needed energy. A simplified version of the interaction among glottal area, vocal fold velocity, glottal pressure, and power input to the vocal folds is shown in Figure 25.10. An idealized (though reasonably realistic) glottal opening waveform is shown in the upper part of the figure. The vocal-fold velocity (upper middle) is positive as the glottal opening increases and then becomes negative as it decreases. When the glottis is very small on opening, the glottal pressure (lower middle) tends to force the folds apart, but as the glottis gets larger, the rushing air creates a reduction in pressure between the folds (from the Bernoulli effect described in Chapter 3). In some instances (as in this example) the glottal pressure may become negative as the glottal opening decreases, tending to assist the fold tension in bringing the folds back together.

The bottom part of the figure shows the power input to the vocal folds which maintains their oscillation. During the opening of the glottis, the glottal pressure is positive. It acts in the same direction as the motion of the folds, thus providing a power input to the folds. As the glottal opening becomes larger, the flow increases and the glottal pressure decreases because of the Bernoulli effect. During closing of the glottis the glottal pressure and the negative fold velocity act in opposition and extract power from the vocal folds. Toward the end of glottal closing, the glottal pressure becomes negative (in this example) and acts in the same direction as the motion of the folds. This again provides power to the folds. (The energy gained is equal to the power times the time it acts, which is the "positive" area minus the "negative" area of the power curve.) When the vocal

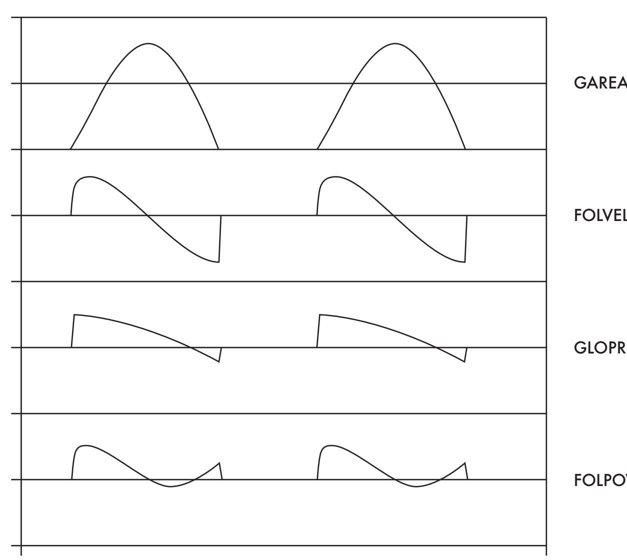

Figure 25.10 *Simplified illustration of the relationships among the glottal opening (upper), vocal-fold velocity (upper middle), glottal pressure (lower middle), and power input to (+) and taken from (-) the vocal folds via the interaction of vocal-fold velocity and glottal pressure (lower).*

system is adjusted for phonation, the vocal folds gain energy from their interaction with the glottal pressure during each vibratory cycle. The energy gained by the vocal folds can maintain their oscillation by offsetting energy losses from other causes.

25.6 Other Sound Sources

The voicing source is located at the vocal folds, but noise sources may occur at any one of several different locations in the vocal tract. Any time air is forced through a small or irregular constriction, the smooth flow of the airstream is disrupted and the flow becomes irregular and turbulent. The resulting noise source serves for the production of unvoiced fricative sounds, such as the /s/ in sue. Typical points of constriction in the tract are teeth to teeth, teeth to tongue, and teeth to lips. When the vocal tract is constricted and the vocal folds are also caused to vibrate, a mixture of both voice and noise energy results. The voicing causes alternate puffs of air to pass through the constriction, giving rise to repetitive bursts of noise. The resulting mixed source serves for the production of voiced fricatives, such as the /z/ in zoo.

It is possible to block off the vocal tract completely while trying to force air from the lungs, increasing the pressure on the lung side of the constriction. If the constriction is suddenly removed, the sudden release produces a burst of energy which is characteristic of many of the unvoiced plosive sounds, such as the /p/ in pat. A more gentle release of a smaller amount of pressure, typical of the voiced plosive sounds, such as the /b/ in bat, usually results in a negligible burst of energy.

You can discover for yourself what energy types are used in the production of the various speech sounds by applying the following techniques. Test for the presence of voicing by placing the fingertips gently on the Adam's apple; a slight vibration will be felt when voice energy is present. Test for the presence of noise energy simply by listening for a "hissy" character in the sounds produced. If both of these conditions are satisfied, we say the result is a mixture of voice and noise energy. Look for burst energy by noting whether a large amount of pressure is built up in a constricted tract and, if so, whether or not it is suddenly released.

25.7 Classification of Speech Sounds

A convenient concept in dealing with speech is that of the **phoneme**, defined as a "distinguishable speech sound." The number of phonemes to be used depends to some extent on how finely one wishes to divide the world of speech sounds. The human vocal mechanism is capable of producing an almost infinite variety of different sounds. However, most of these are not readily distinguishable from one another. We will use the phonemes listed in Table 25.1 Grouping of speech sounds in the table is done partly on the basis of energy type. In later chapters other ways of grouping and classifying phonemes will be discussed.

Table 25.1 *Phoneme classification with corresponding symbols in the text and International Phonetic Alphabet (IPA) symbols. An example of each phoneme used in a word is given in the last column.*

	IPA symbol	Example
Vowels	i	beet
	I	hit
	ɛ	bed
	e	ate
	ae	had
	a	father
	ɔ	awl
	U	put
	u	cool
	Λ	fun
Diphthongs	eI	may
	aI	sigh
	ɔI	oil
	au	shout
	ou	tone
Nasals	m	me
	n	no
	η	sing
Liquids	l	law
	r	red
Semivowels	w	we
	j	you
Unvoiced fricatives	ʍ	when
	h	he
	f	fin
	θ	thin
	s	sin
	ʃ	shin
	tʃ	chin
Voiced fricatives	v	view
	δ	then
	z	zoo
	ʒ	mirage
	ʤ	judge
Unvoiced plosives	p	pea
	t	tea
	k	key
Voiced plosives	b	bee
	d	down
	g	go

25.8 Summary

The speech production system consists of many parts. A person can produce three types of energy when speaking: voiced energy, noise energy, and burst energy. Voiced energy is created when airflow is interrupted periodically by the vibrating vocal folds. Observation of the vocal folds has been achieved in various ways. Three models can be used in representing different aspects of vocal fold action: the simple vibrator model, the vocal cord model, and the multi-mass model. There are three reasons the glottal airflow is not sinusoidal: the glottal area does not vary sinusoidally, the glottal airflow is not directly proportional to the glottal area, and the pressure across the glottis is not constant. The pulse-like nature of the glottal airflow has a spectrum with many harmonic components. Noise energy is created when air is forced through a small or irregular constriction in the vocal tract. Burst energy is produced by the sudden release of pressure from the vocal tract. Combinations of the three forms of energy are used in the production of the various sounds of speech.

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Questions

25.1 What are the basic parts of the human vocal mechanism, and what is the function of each?

25.2 What is the primary force that causes the vocal folds to vibrate?

25.3 What controls the fundamental vibration rate of the vocal folds?

25.4 Why would you expect the vibration frequencies of adult female vocal folds to be higher than those of an adult male? Frame your description in terms of a simple vibrator model of the vocal folds.

25.5 What ranges of fundamental voicing frequency are associated with the speech of adult males, adult females, and children? Are these ranges comparable to the respective singing ranges?

25.6 What effect does a higher blowing pressure have on the behavior of the vocal folds? How does this affect the spectrum of the vocal fold waveform?

25.7 What features of vocal fold action are well represented by the single mass model? What features are poorly represented, if at all?

25.8 When voiced speech sounds are produced, the airflow through the glottal opening is approximately periodic. What does this imply about the spectra being harmonic or nonharmonic?

Exercises

25.1 What is the mass of an adult male vocal fold if its dimensions are 10 mm × 5 mm × 5 mm? (Assume the tissue density is 1 gm/cm^3.) What stiffness (in dynes/cm) is needed to produce a fundamental frequency of 125 Hz?

25.2 What is the mass of an adult male vocal fold if its dimensions are 16 mm × 3 mm × 3 mm? What stiffness (in dynes/cm) is needed to produce a fundamental frequency of 250 Hz?

25.3 Assuming the vocal cord model is valid, what is the cord length of a child's vocal fold in comparison to that of an adult male?

25.4 The product of pressure (in Pa) and flow (in m^3/s) gives power (in W). Suppose that in normal breathing the pressure is 200 Pa and the flow is 100 cm^3/s. What power is generated?

25.5 Suppose that in vowel production at a conversational level the pressure is 1000 Pa and the flow is 120 cm^3/s. What power is generated?

25.6 Suppose the vowel sound of Exercise 25.5 produces a sound level of 70 dB at a distance of 1 m. What is the intensity? Calculate the total acoustic power at this distance by assuming spherical radiation. What fraction of the "breathing power" of Exercise 25.5 is converted into acoustic power?

25.7 Suppose a singer's lung volume allows the singer to exhale 3000 cm^3 of air in voice production. How long can a softly sung vowel be sustained if the flow rate is the same as that claimed for speech in Exercise 25.5?

Activities

25.1 Convince yourself that the act of exhaling in and of itself is not sufficient to produce disturbances that are useful in the speech communication process. To do this, exhale without using the vocal folds and without constricting the vocal tract.

25.2 Bernoulli force: Place one small sheet of paper on top of another small sheet. Tape them together at one end, leaving a small opening in the middle. Insert one end of a soda straw into the opening and tape around it to prevent air leakage. Blow into the straw so that air flows between the two sheets of paper. What happens? Why?

25.3 Produce each of the phonemes listed in Table 23.1 and determine its energy type (voice, noise, mixture, burst). Apply the tests described in the text.

speech sounds. Spectra of vowels and other voiced speech sounds are harmonic and show the different formant frequencies (Figures 26.12 and 26.13). Airflow through constrictions in the vocal tract produces fricative sounds with inharmonic spectra (Figure 26.14). Pressure built up by closing the mouth and suddenly releasing it produces the plosive sounds. In addition to changing vocal tract shape, formant frequencies can be changed by changing vocal tract length and/or by filling the tract with gases other than air.

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Questions

26.1 What are the main components of the vocal tract?

26.2 What is the function of each component of the vocal tract?

26.3 What determines the spectrum of the output speech waveform?

26.4 How might the vowel spectra of Figure 26.12 change for softer voicing? For louder voicing? Would the formant frequencies shift or would their relative amplitudes change?

26.5 If helium is breathed into the vocal tract the resulting speech sounds have a "Donald Duck" quality. Is this because of changes in formant frequencies or fundamental frequency?

26.6 Suppose the vocal fold tension is increased and the vocal tract is lengthened. What happens to fundamental frequency and formant frequencies?

Exercises

26.1 Suppose a "neutral female tract" is a cylindrical tube 14 cm long. At what frequencies will the first three formants occur?

26.2 A "neutral child's tract" is 11 cm long. What are the first three formant frequencies?

26.3 Suppose a neutral tract 17 cm in length is filled with pure helium (v = 970 m/s). At what frequencies will the formants occur?

26.4 What would the formant frequencies be if a neutral tract 17 cm in length were filled with krypton (v = 200 m/s)?

26.5 Use Figures 26.6 and 26.8 to determine the spectrum for /i/. How does your result compare with Figure 26.12?

26.6 Use Figures 26.6 and 26.9 to determine the spectrum for /a/. How does your result compare with Figure 26.12?

26.7 How do the ratios of F2 to F1 and F3 to F1 from Table 26.1 compare for male and female speech? Does this provide any clues as to how we are able to perceive the same vowel for male and female speech even though the actual formant frequencies are different?

26.8 Estimate the length ratio of an average male tract to an average female tract on the basis of ratios of third formant frequencies for different vowels. Do you get the same result for all vowels?

26.9 Use the spectra in Figure 26.12 to determine the approximate frequencies of the first three formants for the phonemes /Λ/, /a/, /æ/, /u/, and /i/ by looking for the broad peaks in the spectrum. Record the results. How do your results compare with those for men in Table 26.1?

26.10 Estimate relative female and male vocal tract lengths on the basis of relative heights of females and males. How does your estimate compare with the text?

26.11 An alternative means can be used to determine which frequencies the neutral vocal tract will emphasize.

Consider, for example, the effect of the neutral tract upon air pulses. The time required for a single pulse to travel the length of a 17 cm tract is equal to 0.5 ms. At the end of 0.5 ms the first pulse will be at the open end of the tract and will be reflected back toward the closed end as a negative pulse. After a total elapsed time of 1 ms (0.5 ms to return) the negative pulse will be back at the closed end of the tract. At the closed end of the tract the negative pulse is reflected toward the open end. After another 0.5 ms the pulse arrives at the open end of the tube, where it is reflected as a positive pulse to return to the closed end. If a new positive pulse is produced just as the positive pulse arrives back at the closed end, the two positive pulses add together to produce a larger positive pulse. From this example we deduce that the tube serves to enhance the pulses if they are produced at a rate of one every 2 ms. To what frequency does the pulse period of 2 ms correspond? How does this relate to the first formant frequency?

26.12 Repeat Exercise 26.11 for pulses produced at the rate of one every 1 ms. Are these pulses enhanced or diminished?

26.13 If a neutral vocal tract is shortened by 10%, what is the approximate change in formant frequencies? What changes would be produced by a 10% lengthening?

26.14 If an average female vocal tract is about 17% shorter than an average male vocal tract what might be expected for the ratio of average female-to-male formant frequencies? Take a few values of formant frequencies from Table 26.1 and calculate the ratios between female and male and compare with the preceding result. How do you account for discrepancies?

26.15 Draw output spectra for a multitube /i/ sound by combining the "source" spectrum of Figure 26.4 with the "filter" spectrum of Figure 26.8.

26.16 Draw output spectra for a multitube /a/ sound by combining the "source" spectrum of Figure 26.4 with the "filter" spectrum of Figure 26.9.

26.17 Make an F2 versus F1 plot for the male and female vowels in Table 26.1. Plot F2 on the vertical and Fl on the horizontal axes, respectively. Do the overall vowel patterns have the same shape and differ mostly by a scale factor?

26.18 Calculate the amplitudes corresponding to the transmission, source, and radiated pressure levels at 1000 Hz in Figures 26.3–26.5. (Assume the levels were obtained from 20 times the logarithm of the amplitudes.) Does the product of the transmission and source amplitudes give the radiated pressure amplitude? Does the sum of 20 times the logarithms of the transmission and source amplitudes give the same result as 20 times the logarithm of the radiated pressure amplitude?

Activities

26.1 Attach a sinusoidally driven loudspeaker to one end of a model vocal tract. Use a microphone hooked to an oscilloscope at the other end to pick up the pressure wave. Measure the response as the frequency is varied.

26.2 Take short sections of tube and place them in contact with your lips so as to extend your lips while producing vowel sounds. Does the vowel color change? Why? Does the pitch change? Why? If possible, perform spectral analyses.

26.3 Perform spectral analyses of voiced and whispered vowels. Are the formant frequencies the same or different? Why? Do harmonic lines appear in the spectra of both? Why?

26.4 Estimate relative female and male vocal tract lengths by having a marshmallow stuffing contest. Determine the

sources of speech spectra. Chapter 10 discusses coarticulation effects.

Strong, W. J. (1967). "Machine-aided Formant Determination for Speech Synthesis," *J. Acoust. Soc. Am. 41*, 1434–1442.

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Questions

27.1 Which of the phonemes in Table 25.1 is "steady"? Which of the phonemes is "transitory"? For each phoneme, indicate the place of articulation (where appropriate) as one of the following: labial, labiodental, alveolar, palatal, velar, glottal. For each phoneme, indicate the manner of articulation as one of the following: vowel, semivowel, liquid, nasal, voiced fricative, unvoiced fricative, voiced plosive, unvoiced plosive.

25.2 What are the distinguishing features typically seen in a spectrogram for the following categories of sounds: nasals, voiced fricatives, unvoiced fricatives, diphthongs, voiced plosives, unvoiced plosives? (For example, vowels are typically characterized by three or more fairly well-defined energy bands or formants.) Describe what the vocal folds and the vocal tract are doing to produce the observed spectrographic features

27.3 Compare the sound-pressure wave in Figure 27.1 with the spectrograms in Figures 27.4 and 27.5. All were obtained from the same utterance of "Joe took father's shoe bench out." What are the corresponding features of the three?

Exercises

27.1 Fundamental frequency can be determined by measuring the frequency of some higher harmonic (such as the 10th) and then dividing by the harmonic number (10 in this case). Use this technique to determine the fundamental frequency at various points in the narrow-band spectrogram of Figure 27.5.

27.2 Sketch spectrographic features in the form of stylized spectrograms for the various categories of sound in Question 27.2.

27.3 Compare formant frequencies obtained from approximately steady portions of the spectrograms in Figures 27.4 and 27.6 with values obtained from the spectra of Figures 26.12 and 26.13. How well do they agree? List some variables that might account for the discrepancies.

27.4 Compare the spectrogram in Figure 27.6 to the "formant spectrogram" in Figure 27.7. Both are for the same utterance, "Robby will like you, daddy-oh." Describe similarities and differences between the two spectrograms.

27.5 Show the approximate time duration of each phoneme in Figures 27.4 and 27.6 by bounding each with vertical lines. Note transition regions and the influences of one phoneme on another.

27.6 Imagine you have a sheet of spectrogram paper. Draw a stylized spectrogram for the diphthong /a/I/. First draw a spectrogram at the left side for /a/. Then draw a spectrogram at the right for /I/. Connect the first, second, and third formants of the /a/ to those for the /I/. Compare your stylized spectrogram with the long "i" in "like" of Figure 27.6.

27.7 Suppose a two-formant speech synthesizer is to be controlled by means of a lapboard. A lapboard (shown in Figure 27.13) is used by placing a pointer on the board in a position appropriate for the first two formant frequencies of the desired sound. The first formant frequency is controlled by the horizontal position of the pointer as it increases from left to right, and the second formant frequency is controlled by the vertical position of the pointer as it increases from bottom to top. Show on the diagram of the lapboard where the pointer should be placed to produce the following vowel sounds: /i/, /I/, /ɛ/, /æ/, /a/, /U/, and /u/. Also show approximate paths that might be traced out by the pointer to produce the diphthongs /aI/ and /au/, and to produce the syllables /d/a/, /g/a/, /b/i/, /d/i/, and /g/i/. Use information from Figure 27.12.

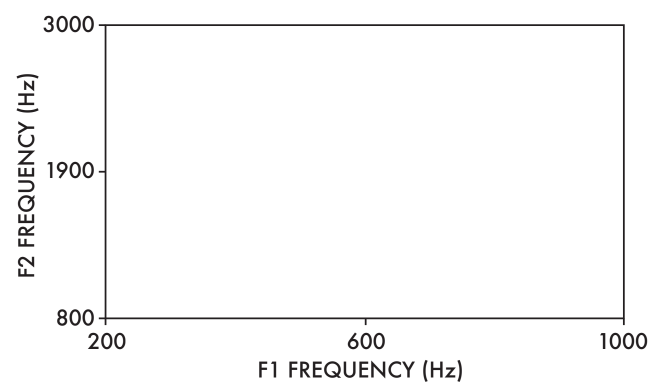

Figure 27.13 *Lapboard for controlling first and second formant frequencies of a two-formant synthesizer.*

27.8 Estimate the fundamental frequency for different voiced parts of the narrow-band spectrogram in Figure 27.5. Do this by determining the frequency difference between two adjacent harmonics indicated by the horizontal striations.

27.9 Estimate the fundamental frequency for different voiced parts of the narrow-band spectrogram in Figure 27.5. Do this by determining the frequency of a strong harmonic and dividing by the harmonic number.

27.10 Estimate the change in F1 and F2 of "dy" in "daddy-oh" from Figures 27.6 and 27.7. How long do the transitions take?

Activities

27.1 Perform an analysis of energy types in speech. Identify the portions of the waveform in Figure 27.1 associated with the various phonemes in "Joe took father's shoe bench out." Determine the degree of periodicity in each part of the wave (much, some, none) for each of the 12 phonemes. Compare these "periodicity" results with the "energy type results" of Activity 25.3. What correlation is there between periodicity and voicing or noise?

27.2 Use a sound spectrograph to produce a sound spectrogram for your own voice while speaking a sentence that is about 2 seconds in length. Interpret the sound spectrogram by labeling different parts of the spectrogram with appropriate phoneme symbols. Also indicate points of high and low pitch.

27.3 Use a speech synthesizer to synthesize some steady vowel sounds and some consonant-vowel syllables.

27.4 Listen to synthetic speech produced by a "Speak & Spell" or other commercial synthesizer. Comment on the intelligibility and naturalness of the synthetic speech. Suggest what the deficiencies might be.

50 clock pulses and -1.0 after 100 clock pulses. The procedure could then be repeated to give an appropriate number of cycles. (The same disclaimers as for the square wave apply here.)

The representation of a sine wave generator is more complex because it cannot be done with simple changes of sign as with the square wave or by adding fixed increments as for the sawtooth. A method commonly used for sine wave generation is table lookup in which the values of the sine function are stored in a table. The clock pulse count effectively produces an argument for the sine function and so specifies where in the table to obtain the sine value. (Interpolation between adjacent table values can be used to obtain a more precise sine value.) Another method of sine wave generation is left to the exercises.

A simple low-pass filter can be represented by

$$OC = (l - ED) \times IC + ED \times OP$$

and a simple high-pass filter can be represented by

$$OC = IC - IP + ED \times OP$$

where OC is the current output of the filter, OP is the previous output, IC is the current input to the filter, IP is the previous input and ED = exp(-6.28fT) with f the filter cutoff frequency and T the clock period. Bandpass filters can be represented in a similar, though more complex, way.

The preceding examples are intended to illustrate how some of the digital components might be implemented. One additional component important for many electronic organs is a frequency divider, where it is important to provide frequencies at octave intervals. Let us suppose that we have the frequencies (or actually periods) specified for the 12 notes in some upper octave. We can think of a pulse generated for each period of the fundamental of each note. For a note an octave lower we want period pulses generated only half as often and a frequency divider is used to accomplish this. The frequency divider receives period pulses from the higher octave note and then produces output pulses only for every other pulse it receives. By successive application of frequency division, frequencies can be obtained for all lower octave notes.

42.6 Electronic Organ Controls

A player has access to two basic sets of controls on an electronic organ: the keyboard (and pedalboard) and the stops. The stops can be used to select particular voices (which determine tone color and, sometimes, attack and decay characteristics) available on the organ. Waveforms produced by different voices on a given organ are determined by the manufacturer's choice of a synthesis method and its implementation. (Synthesis methods will be discussed in Chapter 43.) Stops can also be used to determine expressive features such as tremolo and vibrato. Once the stops have been selected, the player depresses the keys and pedals to produce the appropriate note frequencies and durations. When a key is depressed on the organ, a step voltage is produced as shown in the upper part of Figure 42.9; when the key is released the voltage returns to its previous value. In order to illustrate some additional components and their functions, suppose that a voice has been selected that generates the waveform shown at the lower left in the figure. The voltage from the key goes to an envelope generator (ENV) whose function is to smooth abrupt initial and final voltage changes. A low-pass filter described above can be used to provide the ENV function. The output of the ENV is a smoothed initial transient, followed by a level voltage for as long as the key remains depressed, and terminated with a gradual decay. The output from the ENV is used to control a voltage-controlled amplifier (VCA) whose function is to modulate the amplitude of the input voice waveform. The output of the VCA is the waveform shown at the lower right in the figure and is just the voice waveform as amplitude modulated by the ENV control signal. (A VCA is implemented digitally in terms of a point by point multiplication of the two waveforms involved.)

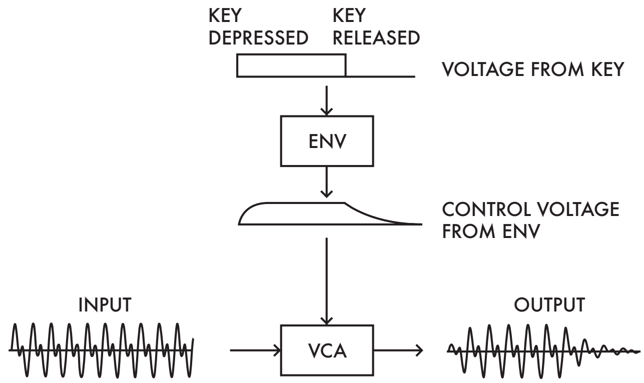

Figure 42.9 *Amplitude modulation of a waveform with a time envelope. Signal from key goes to the ENV which sends a signal to the VCA.*

We now repeat the single organ tone example, but with the addition of amplitude tremolo selected with a stop. The amplitude tremolo is achieved by using a voltage-controlled oscillator to provide an amplitude modulation signal as shown in Figure 42.10. A voltage-controlled oscillator (VCO) generates waveforms with the frequency controlled by the voltage sent to it—increasing the voltage raises the frequency and vice versa. The control voltage can come from a keyboard, from another VCO, or from any control source. (A VCO can be implemented digitally in terms of the sine wave, sawtooth, or square wave generators described above.) Note that the outputs of the ENV and VCO are combined

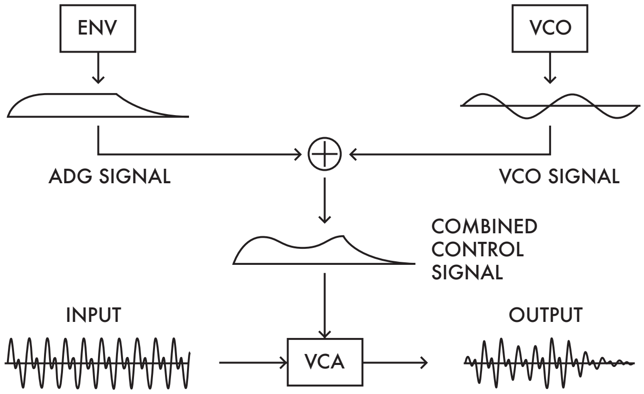

Figure 42.10 *Amplitude modulation of a waveform with a time envelope. Signals from the ENV and VCO are added to provide a control signal to the VCA.*

(with a point by point digital addition) to form a composite signal to control the VCA. The resulting waveform is amplitude modulated in such a way as to have an attack, a tremolo, and a decay as shown in Figure 42.10.

42.7 Electronic Organs

Electronic organs come in many sizes and styles, including small portable models, spinet models, church models, and concert models. The smaller ones may have one or two abbreviated keyboards and a few foot pedals, while the largest ones may have three keyboards, a full pedal board, and numerous stops and controls. A typical church model organ is illustrated in Figure 42.11. Most organs use the method of subtractive synthesis for tone generation. Basically, it involves starting with a complex waveform such as a sawtooth and then filtering out (subtracting) some of the harmonic energy. Subtractive synthesis is used in organs such as the Baldwin, Conn, Rodgers, and so on. Sawtooth, square wave, and sine wave generators are used to produce periodic waveforms with rich harmonic content; noise generators are used to produce waveforms with random consecutive values, but rich in frequency content. (Refer to Chapter 7 for details on waveforms and spectra.) Low-pass, high-pass, and bandpass filters are used to shape the spectra. As an example, suppose that an oboe-like sound is to be produced on an electronic organ. This might be done by taking a sawtooth wave (having all harmonics) and passing it through two bandpass filters set to match the peaks in the oboe spectral envelope of Figure 34.19.

Figure 42.11 *Quantum© model Q385 Digital Organ (Courtesy of Allen Organ Co.)*

There are other synthesis methods including additive and waveform sampling, which are being used more and more frequently in electronic organs. The older Hammond organs using tone wheels (see Figure 42.5) were the best-known examples of an instrument using additive synthesis. With this instrument the performer had control over the relative amplitudes of nine independent harmonics, each having different frequencies but related to the fundamental. The various amounts of the nine harmonics were added together to give steady-state spectra matching those of various musical instruments, including the pipe organ. The fact that the Hammond was unable to even approach a reproduction of such tones indicates the importance of factors other than steady-state spectra. Factors such as attack transients, decay time, and choral effects must all be considered if a realistic tone is desired.

Although considerable work has been done toward achieving realistic attack and decay times, the complex attack and decay transients of organ pipes cannot be duplicated inexpensively in electronic organs. Even though the steady state can be duplicated more effectively, it is not as important as the attack transient. One of the major deficiencies of electronic organs is the lack of an ensemble effect, as many models use only 12 tone generators for the entire instrument. A set of 12 master oscillators is used to generate the notes in a high-frequency octave, which are then frequency divided to produce notes in the lower octaves. This results in all oscillators being mathematically in tune, with only small amounts of beating occurring (see Chapter 33). In comparing this to even a small pipe organ having hundreds of individual tonal sources, it is no wonder that the imitation is a rather poor one. Some organs use many individually tuned oscillators which provide a more realistic ensemble effect, but the basic deficiencies of tone remain. Some of the more expensive electronic organs use several loudspeakers to produce a source with spatial properties more like those of pipe organs. The Allen organ uses a digital computer in a version of waveform sampling in an attempt to produce pipe organ sounds. When the

Exercises

42.1 Sketch an electronically assisted flute. Where might the transducer be placed?

42.2 Figure 42.7 is a diagram of an electronically assisted clarinet. How could the frequency divider be used to change the sound? How could the filters be used to change the sound?

42.3 The following describes an alternative method for sine wave generation. Suppose we assume some previous value for the sine function as SP and a corresponding value for the cosine function as CP. The current value for the sine function can be calculated as

$$SC = SP \times CD + CP \times SD$$

and the current value for the cosine can be calculated as

$$CC = CP \times CD - SP \times SD$$

where CD = cos(360fT), SD = sin(360fT), and T is the clock period. Take f = 500 Hz and T = 20 μs to calculate CD = 0.9980 and SD = 0.0628. Now suppose that the previous value of the sine function was 0 with a corresponding value of the cosine function of 1.0. Now calculate the current value of the sine function as 0.0628 and that of the cosine function as 0.9980. The values for the sine function are the output of our sine wave generator, while those of the cosine function provide the output for a cosine wave generator. The current values for sine and cosine become the previous values at the next step in the calculation and so waveforms of arbitrary duration can be generated. Generate additional sine values and compare them to those from a calculator.

42.4 Repeat Exercise 42.3 with f = 440 Hz and T = 100 μs.

42.5 Imagine an electronically assisted trumpet setup similar to that shown in Figure 42.7 for a clarinet. How realistic will the sound picked up in the trumpet mouthpiece be relative to that directly radiated by the trumpet? Which will have relatively more high-frequency energy?

42.6 The electronic part of the electronic trumpet in Exercise 42.5 takes the trumpet mouthpiece pressure and produces a bassoonlike tone an octave lower. What components are needed to achieve this tone modification?

42.7 If tone wheels having 100 and 106 teeth are rotated at the same speed, what musical interval results? Does the interval depend on the speed of rotation? At what speed would they need to rotate for the 100 tooth wheel to sound A3 = 220 Hz? A4 = 440 Hz?

42.8 Sometimes an attempt is made to create more realism in electronic organ sounds by producing frequency modulation with a moving loudspeaker. If a loudspeaker moves in a circular path of 15 cm radius at 0.33 revolution per second, what is its speed? What maximum frequency shift is produced? Is it the same for all listening positions?

Activity

42.1 Visit several music stores and observe the tonal characteristics of the various electronic organs.

Receiver, the Human Auditory System," *J. Audio Eng. Soc. (39)*, 115–126.

Questions

49.1 What percent of your stereo budget would you allocate for purchase of the loudspeakers, the amplifier, and the cassette deck in a basic stereo system?

49.2 What percent of your stereo budget would you allocate for each of the components in a 5.1 system?

49.3 If you have a limited budget, would it be wiser to invest the money in a high-quality, four-component system or a lesser-quality, six-component system? Explain the pros and cons of each approach.

49.4 List several reasons why the largest budget allocation should be for speakers.

49.5 Summarize the recommended procedure for purchasing loudspeakers.

49.6 Describe how speaker placement in various locations within a room can produce different amounts of bass boost.

49.7 Describe the "dispersion test." Why is a speaker with wide-angle dispersion characteristics desirable?

49.8 Describe how you might expect the perceptual auditory experience to differ between binaural listening with earphones and stereo listening with loudspeakers.

Exercises

49.1 What acoustic output power from your speakers is required to achieve 95 dB (in an average living room)? 105 dB?

49.2 For a speaker that is 10% efficient, compute the required input power for both cases in Exercise 49.1.

49.3 For a speaker that is 1% efficient, make the same calculations as in Exercise 49.2.

49.4 Do the calculations of Exercise 49.2 for a speaker that is 0.5% efficient in a highly absorptive room.

49.5 Which of the following speakers is most efficient. Speaker A: 200 watts in, 2 watts out; Speaker B: 20 watts in, 1.6 watts out; or Speaker C: 2 watts in, 0.02 watt out.

49.6 A speaker with an efficiency of 5% receives 120 watts of electrical power. How much of the input power is lost as heat? How much power is radiated as sound?

49.7 A room has dimensions of 3×4.8 m. Determine where the speakers should be placed in order to get the largest area of good listening. Should the speakers be placed on the shorter or the longer wall?

49.8 How far apart should stereo loudspeakers be placed for the best reproduction of the original sound field? What happens if the speakers are too close? Too far apart?

49.9 A large cathedral-ceilinged room has a volume of 300 m^3 and a RT of 1 s. What acoustic power is required to produce a sound level of 95 dB? What amplifier power is required if the loudspeaker is 5% efficient?

49.10 Calculate the first four natural frequencies of a room with dimensions $10 \text{ m} \times 8 \text{ m} \times 3$ m. (Refer to Chapter 22.) Sketch the room and indicate where pressure maxima will occur. Will the pressure maxima in the room enhance any audio frequencies? Will loudspeaker placement have any effect on the enhancement?

Activities

49.1 Visit some audio stores and obtain specifications and prices of various stereo systems. Listen to the systems and decide which system would be your choice if you were in the market.

49.2 What exactly should you be listening for when you audition a loudspeaker, and how should you listen for it? First, think about how and where the speakers will be used, and try to duplicate this placement in the store. Next, be certain that the speakers you are comparing are placed according to the manufacturers' recommendation, and that neither set has an unfair advantage, because of placement, over the other. Finally, concentrate on listening for tonal balance, bass extension, and stereo imaging.

Appendices

A1. Review of Elementary Math

Definitions

(1) Numerical constant: a quantity that always retains the same value. Example: speed of sound in air at T = 20^0 C.
(2) Arbitrary constants: numerical values assigned to a particular situation. Example: Assuming a constant speed of 20 mph, a man travels ten miles in one-half hour.
(3) Independent variable: variable for which the numerical value is chosen arbitrarily. (A variable is a quantity to which an unlimited number of values can be assigned.)
(4) Dependent variable: variable for which the numerical value depends on the value chosen for the independent variable. Example: Given that a car is traveling at a constant speed of 20 mph, two variables are still involved, time and distance. Taking time as the independent variable, choose any arbitrary time interval, such as two hours. Then the dependent variable (distance) is determined to be 40 miles.
(5) Ratio: a quotient or indicated division, often expressed as a common fraction.

Review of Symbolic Relations and Equations

The math used in physics can be thought of as a set of symbolic relations used to show how "real things" relate to or depend on one another. For example, the distance traveled by an auto can be expressed with word symbols as the distance traveled is equal to the speed of travel multiplied by the time traveled. Alternatively, this relation can be expressed with mixed symbols as length = (speed) (time) or with other symbols, as L = (v)(t).

We often deal with equations, which show some set of symbols equal to some other set. In many cases all symbols but one have known values. Then it becomes our task to calculate a value for the unknown symbol. To solve algebraic equations, a simple rule is to treat each side of the equation as one of a pair of identical twins. Each time something is done to one side of the equation (or twin) the same thing must be done to the other side of the equation (or twin). The object is to get the unknown quantity on one side of the equation and the known quantities on the other. Consider the following examples:

(1) f = 1/T T = 0.1 f = ?

putting the value of T in the equation gives f=1/0.1=10

(2) L = (v)(t) L = 1000 t = 5 v = ?

putting values of L and t in the equation gives 1000 = 5v dividing both sides of the equation by 5 gives 1000/5 = 200 = v so we find v = 200

Review of Square Roots and Logarithms

In addition to equations there are some special symbols that represent operations or special actions. √ is the square root symbol. It means to find a number which when multiplied by itself gives the number under the square root symbol. Study the following examples:

(1) $\sqrt{4} = 2$ $2 \times 2 = 4$.
(2) $\sqrt{5} \cong 2.24$ $2.24 \times 2.24 \cong 5$

The logarithm (or "log") of a number is the power to which 10 must be raised to produce that number. The log of the product of two numbers is the sum of the logs of the two numbers. Study the following examples:

(1) log10=1 $10^1 = 10$

(2) log 1 = 0 $10^0 = 1$

(3) log 100 = 2 $10^2 = 100$

(4) log 0.1 = -1 $10^{-1} = 0.1$

(5) log 100 = log (10)(10) = log 10 + log 10 = 2

(6) log 20 = log (2)(10) = log 2 + log 10 = 1.3

The Sine Function

The "sine" of an angle (where the angle is expressed in fractions of a cycle or degrees) is simply a number. The number can represent "real things," such as displacement, force, etc. The reason for using the "sine function" is that it can conveniently represent different kinds of waves. The sine for different angles can be generated by taking a stick L units in length, pinning one end at the point where the horizontal and vertical axes cross each other, rotating the stick through an angle θ, and measuring the displacement D of the other end of the stick from the horizontal axis, as shown in Figure Al. 1. Note that the sine values will be the same for all rotations after the first so the table of sine values needs to go only from 0 to 360 degrees. The sine of the angular rotation θ is defined as the displacement D divided by L or sine θ = D/L. Once a table of sine values has been put together we can look in the table for the sine value we want, rather than generating it with our rotating stick.

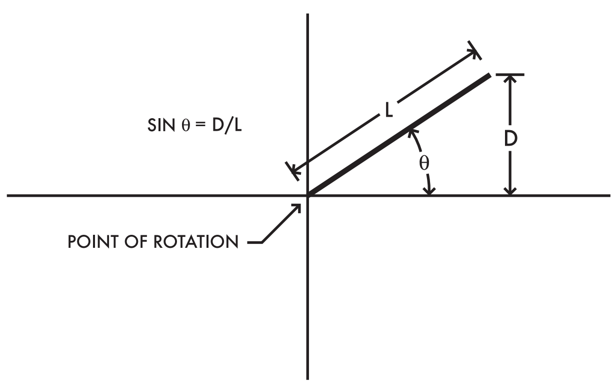

Figure A1.1 *Rotating stick method for generating values of the sine function.*

Exercises

A1.1 If v = f λ ,and you are given λ = 3 and v = 21, find f.

A1.2 If f = 6 and λ = 3, find v.

A 1.3 If f = 8 and v = 12, find λ.

A 1.4 If f = 1/T, and you are given T = 0.2, find f.

A1.5 If f = 30, find T.

A1.6 If d = vt, and you are given d = 30 and v = 10, find t.

A1.7 If v = 2 and t = 5, find d.

A1.8 If t = 100 and d = 2, find v.

A1.9 If f = 0.16 √s/m and you are given s = 90 and m = 10, find f.

A1.10 If s = 360 and m = 10, find f.

A1.11 Use Table A6.3 to determine the following: sin 180°, sin 270°, and sin 45°.

A1.12 Use Table A6.3 to determine the following: sin 0°, sin 90°, sin 162°, and sin 360°.

A1.13 Why is the sine table only given for 0 to 360 degrees?

A1.14 Set up the "sine generator" of Figure A1.1 and compare values generated by it with those given in Table A6.3. Where do negative values come from?

A1.15 Solve for the following by finding the number in the first column of Table A6.2 and its corresponding logarithm in the second column: log 100, log 1000, log 2, log 4, log 10, and log 1.

A1.16 Solve for the following by finding the exponent in the second column of Table A6.2 and its corresponding power of ten in the first column: 10^0, 10^1, 10^6, 10^3, and 10^2.

A 1.17 Solve for the following by factoring the number into two or more smaller numbers, finding the logarithms of the smaller numbers in Table A6.2, and adding the logarithms: log 200, log 40, log 60,000, log 4000, and log - 9000.

A1.18 The logarithm of a number specified by a single nonzero digit followed by a string of zeros can be obtained by counting the zeros and adding to this the logarithm of the digit. Why does this work? Apply this procedure to Exercise A1.17.

A2. Graphs and Graphical Analysis

It is frequently useful to observe how one part of a physical system changes with respect to some other part; that is, when one variable (the independent variable) is changed, how does another variable (the dependent variable) change? If a quantitative relationship exists between the variables, the relationship may be expressed (1) in an equation, (2) in a table, or (3) in a graph. As examples, we analyze some typical relationships found in the physical world. The relationships between the variables may be linear, nonlinear, or oscillatory. In each example the information is expressed in equation form, in tabular form, and in graphical form. An example of graphical addition is also given.

Linear Relationship

The distance a car travels on a straight road when moving at a constant speed is given by d = vt, where d is distance, v is speed, and t is time. The distance traveled (in km) is tabulated in Table A2.1 at six different times (in hr) for a car moving with a speed of 40 km/hr. The values in the table are plotted in Figure A2. 1. The plotted values are connected with a straight line to show the linear relationship between distance and time. The dependent variable (d in this example) changes by equal amounts when the independent variable (t in this example) changes by equal amounts in a linear relationship.

Table A2.1 Distance traveled at a constant speed of 40 km/hr.

Time (hr)	Distance (km)
0	0
1	40
2	80
3	120
4	160
5	200

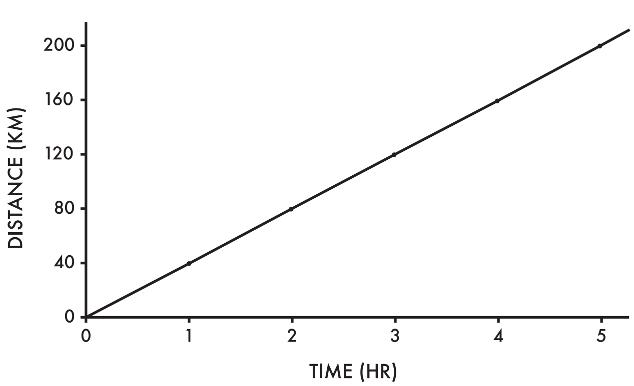

Figure A2.1 *Plot of the linear relation d = 40t.*

Nonlinear Relationship

The height of a rock above ground level when dropped from a 100 m high cliff is given by h = 100 – 4.9 t^2, where h is height (in m) and t is time (in s). The height is tabulated in Table A2.2 at different times. The values in the table are plotted in Figure A2.2. The plotted values are connected with a curved line to show the nonlinear relationship between height and time. The dependent variable (h in this example) changes by unequal amounts when the independent variable (t in this example) changes by equal amounts in a nonlinear relationship.

Table A2.2 *Height of a rock dropped from a cliff.*

Time (s)	Height (m)
0	100.0
0.5	98.8
1.0	95.1
1.5	89.0
2.0	80.4
2.5	69.4
3.0	55.9
3.5	40.0
4.0	21.6

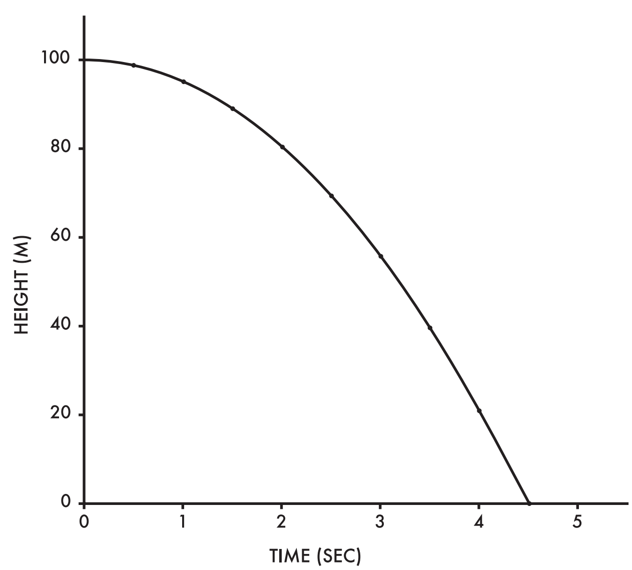

Figure A2.2 *Plot of the nonlinear relation d = 100— 4.9t2.*

Table A2.3 Swing displacement, oscillator displacement, and their combined displacement.

	Displacement (m)		
Time (s)	Swing	Oscillator	Combined
0	0	0	0
0.2	1.18	0.31	1.49
0.4	1.90	0.59	2.49
0.6	1.90	0.81	2.71
0.8	1.18	0.95	2.13
1.0	0	1.00	1.00
1.2	-1.18	0.95	-0.23
1.4	-1.90	0.81	-1.09
1.6	-1.90	0.59	-1.31
1.8	-1.18	0.31	-0.87
2.0	0	0	0
2.2	1.18	-0.31	0.87
2.4	1.90	-0.59	1.31
2.6	1.90	-0.81	1.09
2.8	1.18	-0.95	0.23
3.0	0	-1.00	-1.00
3.2	-1.18	-0.95	-2.13
3.4	-1.90	-0.81	-2.71
3.6	-1.90	-0.59	-2.49
3.8	-1.18	-0.31	-1.49
4.0	0	0	0

Oscillatory Relationship

The displacement of a child in a swing is given by d = 2.0 sin (180t), where d is displacement (in m) from the swing's resting position and t is time (in s). The displacement is tabulated in the second column of Table A2.3 at different times. (The argument of the sine function is assumed to be given in degrees.) The values from the table are plotted in the upper part of Figure A2.3. The plotted values are connected with a sinusoidal curve to show the oscillatory relationship between displacement and time. The dependent variable (d in this example) oscillates between positive and negative values as the independent variable (t in this example) increases.

Graphical Addition

It is often useful to add two simple oscillations to form a complex oscillation because this situation occurs in many everyday phenomena. Suppose we have an oscillator whose displacement, given by d = 1.0 sin (90t), is tabulated in the third column of Table A2.3 and plotted in the middle part of Figure A2.3. This oscillation can be added to the swing oscillation described in the previous section by adding the values in the second and third columns of Table A2.3 to give the values in the fourth column.

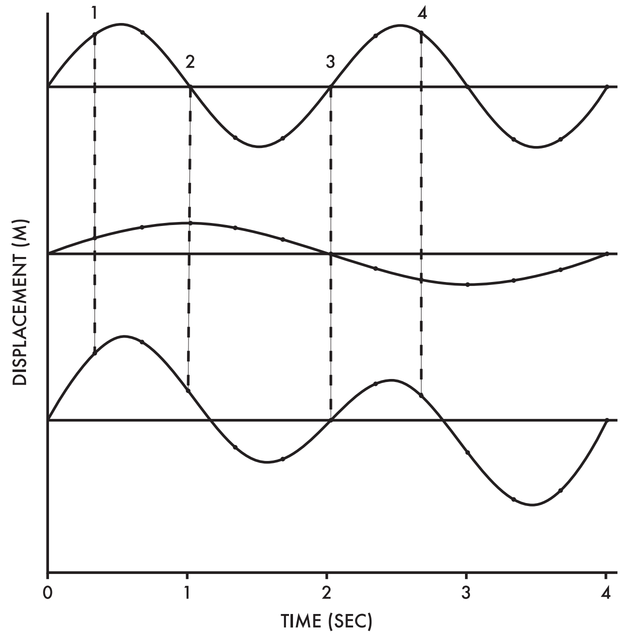

Figure A2.3 *Plot of d = 2 sin (180t) in the upper curve, d = sin (90t) in the middle curve, and their sum in the lower curve.*

These two sinusoidal oscillations (upper and middle parts of Figure A2.3) can also be added graphically to give a complex oscillation (lower part of Figure A2.3). We will consider the addition of the numbered points to illustrate various features of graphical addition. Upper and middle points 1 are both positive; when they are added they produce the larger positive value plotted as point 1 in the lower graph.

Upper point 2 added to middle point 2 gives lower point 2; lower point 2 is equal to middle point 2 because upper point 2 is zero and adds nothing.

Upper point 3 added to middle point 3 gives a lower point 3 of zero because both the upper and middle points have values of zero. Adding middle point 4 to upper point 4 gives a value smaller than upper point 4 because middle point 4 is negative. The result appears as lower point 4. Again, the points of the complex waveform (lower curve) are connected with curved lines to show the nature of the combined waves.

Exercises

A2.1 Tabulate and graph values of v for the relation v = f λ as f is varied from 1 to 25 if λ = 4.

A2.2 Tabulate and graph values of F for the relation F = ma as a is varied from 1 to 10 if m = 10.

A2.3 Tabulate and graph values of f for the relation f = 1/T as T is varied from 0.1 to 10.

A2.4 Tabulate and graph values of f for the relation $f = 0.16\sqrt{(s/m)}$ as (s/m) is varied from 0 to 100.

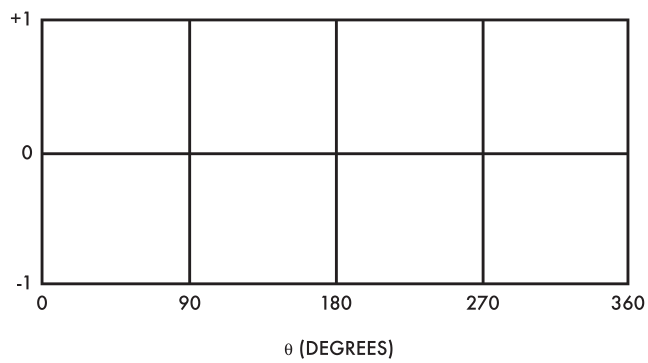

Figure A2.4 *Grid to be used for plot of Exercise A2.6.*

A2.5 Tabulate and graph values of dB for the relation dB = 20 log (p/20) as p is varied from 20 to 20,000.

A2.6 Plot sin θ in Figure A2.4 for values of θ between 0 and 360°.

A2.7 Graphically add the upper and middle curves in Figure A2.5 and plot the result in the lower part of the figure. (A ruler can be used to carry out the point-by-point additions.)

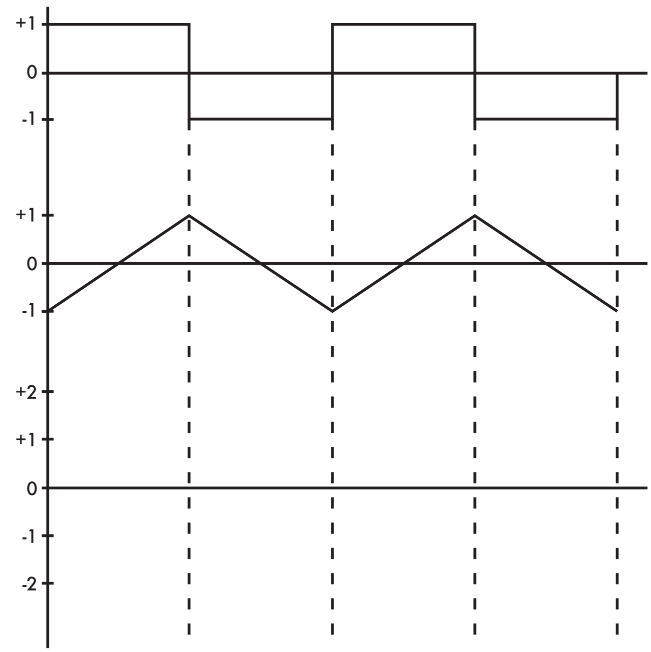

Figure A2.5 *Upper and middle curves to be graphically added and plotted in the lower part of the figure as Exercise A2.7.*

A3. Symbols, Quantities, and Units

In the hypotheses and abstractions of science it is necessary to express relationships between natural phenomena. Special symbols are typically used as a shorthand way of representing various physical quantities. Relationships expressed symbolically permit a great deal of economy as compared to writing everything out in words. Special symbols are not peculiar to science; they are used extensively in music, speech, and other facets of our lives. It is very helpful to use standardized symbols so that new symbols do not have to be learned every time a different person's writings are studied. However, although much standardization exists in science, uniformity of symbol usage is not complete.

When carrying out the measurement step in a scientific method it is necessary to quantify the results of the various measurements. Standard units of measure should be employed so that one set of measurement results can be easily communicated to and interpreted by others. Combining the prefixes of Appendix 4 with standard unit symbols makes a whole range of standard units available. For example, ms (millisecond) is the symbol for one thousandth of a second and cm (centimeter) is the symbol for one-hundredth of a meter. Scientists making similar observations can easily compare their experimental results with one another when standard units of measure are employed. (Think for a moment of the confusion that might develop in comparing various measurements of length of a particular object if each of ten different observers used as a unit of length the length of his shoe or index finger. In this case, you can see that the same object would be described in terms of ten length measures, each having a different numerical value.)

The definition of standard symbols and standard units of measure is quite arbitrary in many respects. Once the definitions have been made, however, a consistent use of the standards makes their utility value vary significant.

Table A3.1 *Symbols, quantities, and units.*

Symbol	Quantity	Common Units
a	acceleration	m/s^2
d	displacement	m (meter)
f	frequency	Hz (cycles per second)
i	electric current	A (ampere)
L	length	m (meter)
m	mass	kg (kilogram)
n	integer	—
p	pressure	Pa (pascal)
q	electric charge	C (coulomb)
s	stiffness	N/m
t	time	s (second)
v	velocity or speed	m/s
w	weight	N (newton)
x	any unknown quantity	
A	amplitude	various
D	mass density	kg/m^3, kg/m
E	energy or work	J (joule)
F	force or tension	N (newton)
K	constant	—
I	intensity	W/m^2
P	power	W (watt)
R	electric resistance	Ω (ohm)
S	surface area	m^2
T	period	s (second)
V	volume	m^3
Δ	indicates small change	
λ	wavelength	m (meter)
ϕ	phase	degree
θ	angle	degree

Table A3.2 *Symbols and quantities.*

Symbol	Quantity
AC	absorption coefficient
ADC	analog-to-digital conversion
AFC	automatic frequency control
CB	critical band
DAC	digital-to-analog conversion
DRT	diagnostic rhyme test
GPE	gravitational potential energy
IMD	intermodulation distortion
JND	just noticeable difference
KE	kinetic energy
LL	loudness level
NC	noise criteria
PCM	pulse code modulation
PE	potential energy
PTS	permanent threshold shift
RMS	root-mean-square
RT	reverberation time
SHM	simple harmonic motion
SIL	sound intensity level
SL	sound level
S/N	signal-to-noise ratio
SPL	sound pressure level
STC	sound transmission class
TA	total absorption
THD	total harmonic distortion
TTS	temporary threshold shift
VCA	voltage-controlled amplifier
VCF	voltage-controlled filter
VCO	voltage-controlled oscillator

Continuous equivalent level: (L_{eq} dBA) A scale for non-steady noises; the dBA level averaged over time to yield an equivalent dBA level of steady sound that would provide the same total sound energy in the same time.

Continuous wave: A disturbance of continuing duration.

Contour tones: Tones in which contrasts are made among changing pitch contours.

Coulomb: (C) Unit of measure of electric charge—the total charge of 6.24×10^{18} electrons.

Critical band: (CB) A frequency range within which two sinusoids interact significantly. The frequency range of a critical band depends on the "stimulation width" on the basilar membrane.

Crossover network: A system that divides an output signal into different ranges and routes them to the appropriate driver units.

Crosstalk: The leakage of information between two adjacent tracks on recording tape.

Crystal microphone: A microphone that employs the piezoelectric effect.

Cutoff frequency: The frequency at which waves begin to travel into an open-hole section of tubing, rather than being reflected.

Cutoff frequency: The lowest frequency that can be efficiently transmitted by a horn speaker.

Damped oscillation: Oscillation in which the amplitude of an oscillator decreases with time.

Damping factor: A description of an amplifier's ability to "dampen" or control unwanted residual movements of the speaker cone, resulting from inertia, after a signal is terminated.

Damping time: The time required for an amplitude to decrease to one-half its initial value.

Day-night average level: A measure of sound level that adjusts for greater sensitivity to noise at night.

Deductive reasoning: Beginning with a generalization and searching for evidence of it in scientific observations.

Demi-syllables: Acoustic segments employed in concatenation systems consisting of either the first half or the last half of a syllable and containing transitions from consonants to vowels and vowels to consonants.

Density: (D) The mass occupied by a standard volume of a material; mass divided by volume.

Destructive interference: The out-of-phase addition of waves to create a smaller wave.

Diapasons: A rank of cylindrical open pipes that go "through all" the notes of an organ keyboard (meaning there is a pipe for each key) and produce a tone which is the most characteristic of the organ.

Diaphragm: A muscular structure at the bottom of the chest cavity that controls pressure in the chest cavity surrounding the lungs.

Diffraction: The bending of waves around obstacles or through openings.

Diffuse reflection: The reflection in many different directions occurring when a wave encounters a rough surface.

Diffusion: An approximately equal distribution of sound energy throughout a room; usually produced by irregularly shaped objects which scatter the sound.

Digital-to-analog conversion: The conversion of numbers into discrete voltages which are then smoothed into continuous voltages.

Digitizing: Converting an analog signal into a string of discrete numbers.

Diphones: Acoustic segments employed in concatenation systems consisting of the last half of one phoneme and the first half of the next phoneme, including the inter-phoneme transition boundaries.

Direct sound: The sound going directly from the sound source to the microphone (or listener) with no reflections.

Discrimination testing: A test used to measure whether the difference between two speech signals can be detected.

Dispersion: The "spread" of a sound wave emanating from a loud-speaker.

Displacement amplitude: (A)The maximum displacement in either direction from the rest position of an oscillator.

Displacement: The change in position of an object, as measured by distance and direction.

Distortion: In auditorium acoustics, any undesirable change in the quality of a musical sound because of the uneven or excessive absorption of sound at certain frequencies.

Dolby Pro-Logic: A surround sound encoding/decoding system where the center (C) and surround (S) channels are extracted from the left (L) and right (R) channels. The LRCS information is encoded into two channels that go to the recorder. On playback the LRCS channels are decoded from the two channels to drive five speakers.

Dolby Surround: A surround sound encoding system that creates a rear speaker surround channel as L-R with Dolby-B noise reduction, but delayed by 15 or 20 msec.

Doppler effect: The change in the apparent frequency of a sound due to a relative motion between the sound source and the listener.

Driven vibrator: Any vibrator to which energy is continuously supplied by an external source.

Dynamic headroom: The number of dB of power above the rated continuous power an amplifier can deliver for short time periods without exceeding the rated THD.

Dynamic loudness: An assessment relative to what might be anticipated between a pianissimo and a fortissimo, corresponding to the physical measure of strength.

Dynamic microphone: A microphone where the transducing element is a coil of wire attached to the diaphragm and is free to move between the poles of a permanent magnet.

Dynamic pickup: A pickup that couples mechanical motion to a coil in a magnetic field.

Dynamic range: The difference in dB level between the softest perceivable signal that is not noise, and the loudest signal within the given limits of distortion (THD and IMD).

Dynamic range: The range of amplitudes over which a tranducer's output response is nearly linear.

Early Decay Time: (EDT) Defined in terms of the first 10 dB of sound decay. It is multiplied by a factor of 6 to correspond to RT, which accounts for 60 dB of decay.

Early sound: The direct sound and any reflected sound arriving within approximately 35-80 ms of the direct sound.

Echo: Loud reflected sounds arriving more than 50 ms later than the direct sound.

Echoes: When a reflected or delayed sound (having the same frequency content as the direct sound) is heard by a listener as separate and distinct from the direct sound.

Effective perceived noise level: (EPNdB) A noise level scale that takes into account maximum loudness and duration.

Electret condenser microphone: A condenser microphone which eliminates the need for a high-voltage bias supply by using a permanently-charged material.

Electric charge: (q) A measure of the unbalanced electricity in a body of matter as determined by the number of positively charged particles versus the number of negatively charged particles.

Electric current: (i) The amount of charge passing a given point per unit time.

Electrical potential difference: (V) Difference between two points which will cause current to flow in a closed circuit; measured in volts.

Electromagnetic effect: When a current flows through a coil in a magnetic field, the electrons and the coil containing them experience a magnetic force.

Electronic hearing aid: Consists of a microphone which converts sound into electrical energy, an amplifier, and an earphone (receiver) to convert the electrical signal back into sound.

End correction: Any adjustment made in the measure of the actual length of a flute's main tubing to determine its effective length. End corrections can result from how the air interacts at the open end of the tube or the geometry of the tube.

End correction: The distance from the end (or mouth) of the pipe to the node of the fundamental pressure wave.

Ensemble: A performer attribute of a hall that enables performers to hear each other, and thus to play in time, in tune, in balance, and with a blending of their sounds.

Envelopment: The degree to which reverberant sound appears to come from all directions.

Epiglottis: A cartilage flap that can cover the tracheal opening.

Equal-tempered tuning: Division of the octave into 12 equal intervals, or semitones, any two consecutive notes having the same frequency ratios. In an equal-tempered octave, a whole tone is equal to exactly two semitones.

Equilibrium position: "Rest" position of an object where the sum of all forces acting on it is zero.

Esophagus: A muscular tube behind the trachea that transports materials from the pharynx to the stomach.

Experimentation: The observation of changes in one or more dependent variables as a result of manipulating one or more independent variables.

Facts: The experimental observations or data measurements made by scientists.

Falsetto register: Vocal register used for the production of mid to high fundamental frequencies. The following features are characteristic of the falsetto register: (1) The vocal folds are longer (up to 50%) and thinner, with smaller vibrating mass. (2) Longitudinal tension in the ligament is comparatively high. (3) Vibration amplitude of the folds is small. (4) The vocal folds tend to lack complete closure during any part of the vibratory cycle. This results in increased airflow and a more breathy quality. (5) The signal produced by glottal airflow tends to be poorer in higher harmonics. (6) The conversion of "lung energy" into