Appendix A: Composers

Renaissance

Giovanni da Palestrina 1525(?)–1594
Josquin des Pres c 1450–1521
Thomas Tallis 1505–1585
Michael Praetorius 1571–1621
Claudio Monteverdi 1567–1643

Baroque

Dietrich Buxtehude 1637–1707
Johann Pachelbel 1653–1706
Arcangelo Corelli 1653–1713
Henry Purcell 1659–1695
Antonio Vivaldi 1678–1741
Georg Philipp Telemann 1681–1767
Jean-Philipe Rameau 1683–1764
Johann Sebastian Bach 1685–1750
Georg Frederick Handel 1685–1759
Giovani Pergolesi 1710–1736
Christoph W. Gluck 1714–1787

Rococo

Francois Couperin 1668–1733
Domenico Scarlatti 1685–1757
Wilhelm Friedman Bach 1710–1784
Carl Phillip Emanuel Bach 1714–1788
PDQ Bach 1807–1742 (?)

Classic

Josef Haydn 1732–1809
Johann Christian Bach 1735–1782
Luigim Boccherini 1743–1805
Muzio Clementi 1752–1832
Wolfgang Amadeus Mozart 1756–1791
Antonio Salieri 1750–1825

Romantic

Ludwig van Beethoven 1770–1827
Gioacchino Rossini 1792–1868
Franz Schubert 1797–1828
Hector Berlioz 1803–1869
Felix Mendelssohn 1809–1847
Robert Schumann 1810–1856
Richard Wagner 1813–1883
Johannes Brahms 1833–1897
Cesar Franck 1822–1890
Anton Bruckner 1824–1896
Camille Saint-Saens 1835–1921
Pablo de Sarasate 1884–1908

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Virtuoso

Niccolo Paganini 1782–1840
Frederic Chopin 1810–1849
Franz Liszt 1811–1886

Russian Movement

Alexander Borodin 1834–1887
Modest Musorgsky 1839–1881
Petr I. Tchaikovsky 1840–1893
Nicholai Rimsky-Korsakov 1844–1908
Alexander Scriabin 1872–1915
Serge Prokofiev 1891–1952

Nationalist

Bedrich Smetana 1824–1884
Antonin Dvorak 1841–1904
Edvard Grieg 1843–1907
Carl Nielsen 1865–1931
Jean Sibelius 1865–1957

Impressionists

Claude Debussy 1862–1918
Fredrick Delius 1862–1934
Eric Satie 1866–1925
Maurice Ravel 1875–1937
Ottorino Respighi 1879–1936
Charles T. Griffes 1845–1920

Post Romantics

Edward Elgar 1857–1934
Gustav Mahler 1860–1911
Richard Strauss 1864–1949
Ralph Vaughn Williams 1872–1958
Sergei V. Rachmaninoff 1873–1943
Gustav Holst 1874–1934

Expressionists–Serialists

Arnold Schoenberg 1874–1951
Anton von Webern 1883–1945
Alban Berg 1885–1935

Neo Classic

Charles Ives 1874–1954
Manuel de Falla 1876–1946
Darius Milhaud 1892–1974
Walter Piston 1894–1976
Carl Orff 1895–1982
Howard Hansen 1896
Roger Sessions 1896–1985
Francis Poulenc 1899–1963
Joaquin Rodrigo 1902–1999
Aaron Copland 1906–1990
Dmitri Shostakovich 1906–1975
Samuel Barber 1910–1981
William Schuman 1910–1992
Benjamin Britten 1913–1976

Modern

Bela Bartok 1881–1945
Igor Stravinsky 1882–1971
Edgar Varese 1885–1965
Paul Hindemith 1895–1963
Harry Partch 1901–1974
Olivier Messiaen 1908–1992
John Cage 1912–1992
Pierre Boulez 1925–
Luciano Berio 1925–2003
Karlheinz Stockhausen 1928–2007

Appendix B: Trigonometric Functions

Definitions

The trigonometric functions are defined in terms of a right triangle, shown in Fig. B1.

Figure B1 shows angle θ in the first quadrant of the x–y plane. The adjacent and opposite sides are both positive numbers in this quadrant. Further, the hypotenuse is always a positive number. Therefore, both sine and cosine functions are positive in this quadrant. Figure B2 shows angle θ in other quadrants and gives the signs for sine and cosine functions.

By definition, the hypotenuse (with length hyp) is the side opposite to the right angle (90°) and it is the longest side. The angle of interest (θ) determines the definition of the "opposite side" (op) and the "adjacent side" (adj). The basic trigonometric functions are the ratios of the lengths of these sides. Because they are ratios, the trigonometric functions depend only upon angle θ; they do not depend upon the size of the triangle:

$$\sin\theta = (\text{op})/(\text{hyp}) \qquad \text{The sine function.} \tag{B1}$$

$$\cos\theta = (\text{adj})/(\text{hyp}) \qquad \text{The cosine function.} \tag{B2}$$

$$\tan\theta = (\text{op})/(\text{adj}) \qquad \text{The tangent function.} \tag{B3}$$

Using Fig. B1 and the definitions of the sine, cosine, and tangent, we see that the sine and cosine functions can only have values from +1 to -1. The tangent function, however, can have any value between plus and minus infinity.

The definitions immediately relate the tangent function to the sine and cosine functions:

$$\tan\theta = \frac{\sin\theta}{\cos\theta} \qquad \text{for all } \theta. \tag{B4}$$

The Pythagorean relation for a right triangle relates the lengths of the sides:

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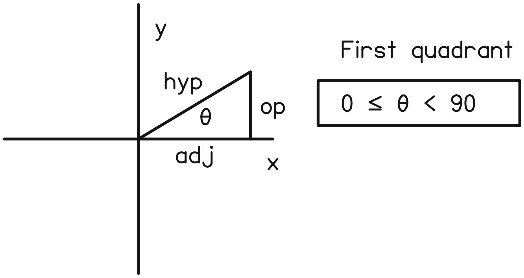

Fig. B1 An angle in the first quadrant. The opposite side and the adjacent side are both positive

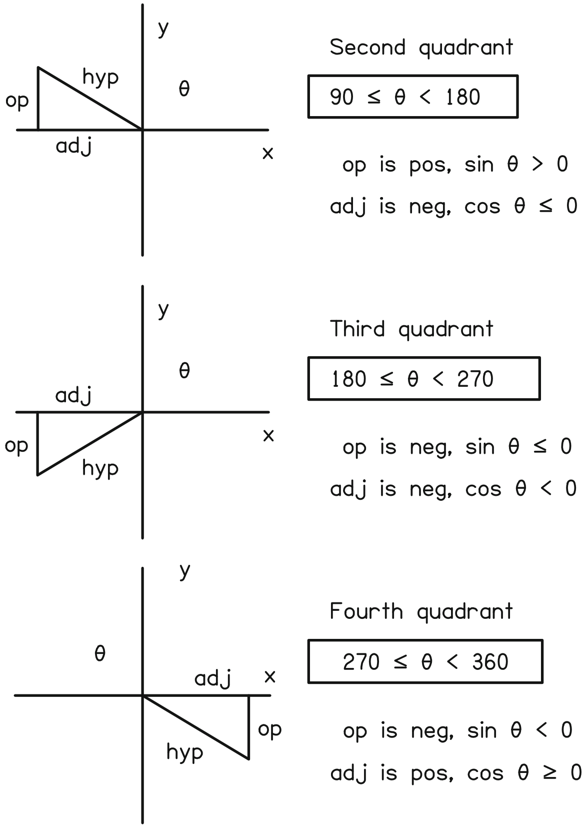

Fig. B2 Angles in the other three quadrants

$$\mathrm{adj}^2 + \mathrm{op}^2 = \mathrm{hyp}^2$$

or

$$\cos^2\theta + \sin^2\theta = 1. \tag{B5a}$$

♠

Appendix C: Traditional Musical Instruments

This appendix indicates the fundamental frequencies in units of Hertz (Hz) of significant notes on some important instruments of the orchestra.

For most instruments the bottom note (lowest frequency) is rather firmly fixed by the construction of the instrument, though non-traditional playing techniques may produce lower frequencies. The top notes are much less well defined. For instance, the top note on the trumpet is given as 1,244 Hz, but some jazz players can hit the "double high C" at $B\flat_6 = 1,863$ Hz. Normally the top notes given below are those that a musician may expect to see written in musical scores.

It is important to remember that the entries below only correspond to the *fundamental* frequencies of instruments. Each of the tones has harmonics that lead to energy at much higher frequencies. For instance, the bottom note of the trumpet is listed as 165 Hz, but there is lots of energy in the 10th harmonic of this note, and that frequency is 1,650 Hz (higher than the fundamental frequency of the top note).

All frequencies have been rounded off to the nearest Hz. Careful tuning tries to be more precise than this.

Strings

There are four or six strings. Their note names are given first in order of ascending frequency. Then the corresponding fundamental frequencies are given.

Violin: G_3, D_4, A_4, E_5, 196, 294, 440, 659, top at $E_7 = 2637$
Viola: C_3, G_3, D_4, A_4, 131, 196, 294, 440, top at $A_6 = 1760$
Cello: C_2, G_2, D_3, A_3, 65, 98, 147, 220, top at $E_5 = 659$
Bass: E_1, A_1, D_2, G_2, 41, 55, 73, 98, top at $D_4 = 587$
Guitar: E_2, A_2, D_3, G_3, B_3, E_4, 82, 110, 147, 196, 247, 330,
top at $B_5 = 938$

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Woodwinds

Flute: bottom at $C_4 = 262$, top at $D_7 = 2349$
Piccolo: bottom at $C_5 = 523$, top at $C_8 = 4186$
Clarinet*: bottom at $D_3 = 147$, top at $B\flat_6 = 1865$
Alto Saxophone*: bottom at $D\flat_3 = 139$, top at $A\flat_5 = 831$
Tenor Saxophone*: bottom at $A\flat_2 = 104$, top at $E\flat_5 = 622$
Oboe: bottom at $B\flat_3 = 233$, top at $G_6 = 1568$
Bassoon: bottom at $B\flat_1 = 58$, top at $E\flat_5 = 622$

Brass

Trumpet*: bottom at $E_3 = 165$, top at $E\flat_6 = 1244$
Trombone*: bottom at $E_2 = 82$, top at $D_5 = 587$
Tuba*: bottom at $E\flat_1 = 39$, top at $B\flat_3 = 233$
French horn*: bottom at $B_1 = 62$, top at $F_5 = 698$

Other

Tenor voice: top (not falsetto) at approximately $C_5 = 523$
Harp: bottom at $B_0 = 31$, top at $A_7 = 3520$
Piano: bottom at $A_0 = 28$, top at $C_8 = 4186$
Organ: bottom at $C_0 = 16$, top at $C_9 = 8372$

*This table gives frequencies and note names that are, so-called, "concert" notes, established by the notes on the piano or the notes recognized by a violin player. An instrument marked with an asterisk is a "transposing" instrument, meaning that the note name recognized by the player is different from the "concert" note name. For instance, the trumpet is a $B\flat$ instrument. Therefore, when the trumpet player plays what he reads on the score as "middle C," and what he calls "middle C," the fundamental frequency of the note that sounds is not C-262 Hz. Instead it is $B\flat$-233 Hz. The note $B\flat$ is a whole step lower than C. The transposition is consistent in that the notes written for a trumpet always sound a whole step lower than concert pitch.

♠

Appendix D: Keyboard Tuning

Keyboard instruments like pianos, organs, and synthesizers give a musician the opportunity to play many notes at once. In exchange for this benefit, the keyboard player sacrifices control of intonation—fine control of the frequency during playing. The piano player cannot bend notes up or down like the player of wind instruments or string instruments. The individual notes of a keyboard instrument must be tuned before the instrument is played.

Figure D1 shows a standard keyboard with notes named C, $C\sharp$, D, etc. It also shows musical intervals, unison, minor second, Major second, etc., using C as a reference. The figure shows a little more than an octave. You will note that the pattern of the keys is the same for each octave. An octave corresponds to a factor of two—*always*, whatever the tuning. For instance, if the note called C on the far left has a frequency of 261.6 Hz then the note called C to the right (the "Octave") has a frequency of $261.2 \times 2 = 522.4$ Hz.

Table D1 shows frequency ratios corresponding to those intervals for three different tunings. The table gives the frequency ratios for musical intervals within the octave in three tunings, Just, Pythagorean, and Equal Tempered, per ANSI standard S1.1-1994. There are 12 intervals besides the unison.

Just Tuning The tuning called "Just" chooses the frequency ratios to be the ratios of small integers like 5/4 for the interval called the "Major third." For instance, if we tune a keyboard in the key of C and make the frequency of middle C equal to 261.6 Hz, then the frequency of the note called E (Major third) will be $261.6 \times 5/4 =$ 327.0 Hz. The Just tuning leads to very smooth sounding chords.

Pythagorean Tuning The tuning called "Pythagorean" is obtained by using the ratios 3/2 and 4/3 systematically. For example the weird ratio 1024/729 is equal to 4/3 raised to the sixth power $(4/3)^6 = 4096/729$, which is then brought into the first octave by twice dividing by 2 (i.e., by 4). For instance, if we tune a keyboard in the key of C and make the frequency of middle C equal to 261.6 Hz, then the frequency of the note called E (Major third) will be $261.6 \times 81/64 = 331.1$ Hz.

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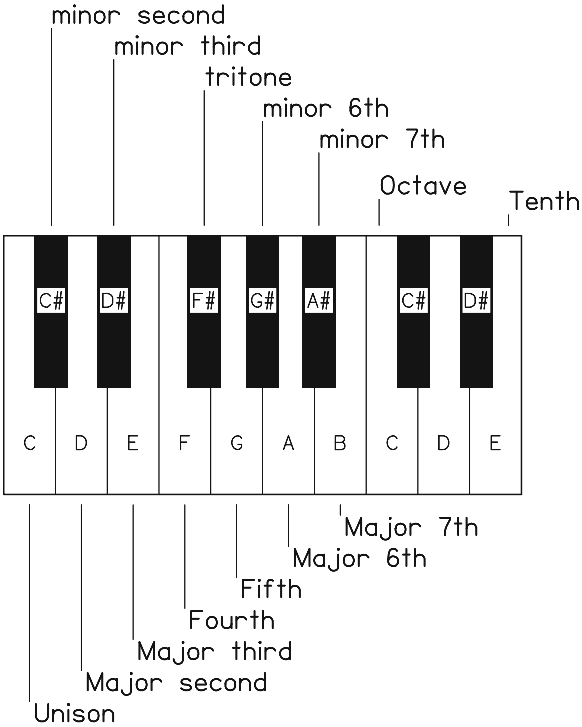

Fig. D1 A section of the piano keyboard with note names and interval names. The assignment of interval names assumes that the low C is the reference

The Pythagorean tuning was the starting point for many different tunings of the baroque era known as meantone tunings.

Equal Temperament The tuning called "Equal temperament" chooses the frequency ratios to be powers of $2^{1/12}$, the ratio of the semitone. Therefore, any two adjacent notes on the keyboard have a frequency ratio of $2^{1/12}$, regardless of the key. For instance, whatever key the music may be, if we make the frequency of middle C equal to 261.6 Hz, then the frequency of the note called E (Major third) will be $261.6 \times 2^{4/12} = 329.6$ Hz. Equal temperament is used almost universally in our own day. It has all but replaced the other tunings.

Table D1 Notes of the scale in three temperaments

Interval	Just tuning	Pythagorean tuning	Equal temperament
Unison	1.	1.	$2^0 = 1.$
minor Second	$16/15 = 1.067$	$256/243 = 1.053$	$2^{1/12} = 1.059$
Major Second	$10/9 = 1.111$	$9/8 = 1.125$	$2^{2/12} = 1.122$
	or $9/8 = 1.125$		
minor Third	$6/5 = 1.200$	$32/27 = 1.185$	$2^{3/12} = 1.189$
Major Third	$5/4 = 1.250$	$81/64 = 1.266$	$2^{4/12} = 1.260$
Fourth	$4/3 = 1.333$	$4/3 = 1.333$	$2^{5/12} = 1.335$
Tritone	$45/32 = 1.406$	$1024/729 = 1.405$	$2^{6/12} = 1.414$
	or $64/45 = 1.422$	or $729/512 = 1.424$	
Fifth	$3/2 = 1.500$	$3/2 = 1.500$	$2^{7/12} = 1.498$
minor Sixth	$8/5 = 1.600$	$128/81 = 1.580$	$2^{8/12} = 1.587$
Major Sixth	$5/3 = 1.667$	$27/16 = 1.688$	$2^{9/12} = 1.682$
minor Seventh	$7/4 = 1.750$	$16/9 = 1.778$	$2^{10/12} = 1.782$
	or $16/9 = 1.778$		
	or $9/5 = 1.800$		
Major Seventh	$15/8 = 1.875$	$243/128 = 1.898$	$2^{11/12} = 1.888$
Octave	$2/1 = 2.000$	$2/1 = 2.000$	$2^{12/12} = 2.000$

♠

Appendix E: Standard Musical Frequencies

This appendix gives the frequencies (in Hertz) of the notes of the equal-tempered musical scale assuming that concert A is tuned to 440 Hz, per the international standard.

C_0	16.35	C_2	65.41	C_4	261.6
$C\sharp_0$ or $D\flat_0$	17.32	$C\sharp_2$ or $D\flat_2$	69.30	$C\sharp_4$ or $D\flat_4$	277.2
D_0	18.35	D_2	73.42	D_4	293.7
$D\sharp_0$ or $E\flat_0$	19.45	$D\sharp_2$ or $E\flat_2$	77.78	$D\sharp_4$ or $E\flat_4$	311.1
E_0	20.60	E_2	82.41	E_4	329.6
F_0	21.83	F_2	87.31	F_4	349.2
$F\sharp_0$ or $G\flat_0$	23.13	$F\sharp_2$ or $G\flat_2$	92.50	$F\sharp_4$ or $G\flat_4$	370.0
G_0	24.50	G_2	98.00	G_4	392.0
$G\sharp_0$ or $A\flat_0$	25.96	$G\sharp_2$ or $A\flat_2$	103.83	$G\sharp_4$ or $A\flat_4$	415.3
A_0	27.50	A_2	110.00	A_4	440.0
$A\sharp_0$ or $B\flat_0$	29.14	$A\sharp_2$ or $B\flat_2$	116.54	$A\sharp_4$ or $B\flat_4$	466.2
B_0	30.87	B_2	123.47	B_4	493.9

C_1	32.70	C_3	130.8	C_5	523.3
$C\sharp_1$ or $D\flat_1$	34.65	$C\sharp_3$ or $D\flat_3$	138.6	$C\sharp_5$ or $D\flat_5$	554.4
D_1	36.71	D_3	146.8	D_5	587.3
$D\sharp_1$ or $E\flat_1$	38.89	$D\sharp_3$ or $E\flat_3$	155.6	$D\sharp_5$ or $E\flat_5$	622.3
E_1	41.20	E_3	164.8	E_5	659.3
F_1	43.65	F_3	174.6	F_5	698.5
$F\sharp_1$ or $G\flat_1$	46.25	$F\sharp_3$ or $G\flat_3$	185.0	$F\sharp_5$ or $G\flat_5$	740.0
G_1	49.00	G_3	196.0	G_5	784.0
$G\sharp_1$ or $A\flat_1$	51.91	$G\sharp_3$ or $A\flat_3$	207.7	$G\sharp_5$ or $A\flat_5$	830.6
A_1	55.00	A_3	220.0	A_5	880.0
$A\sharp_1$ or $B\flat_1$	58.27	$A\sharp_3$ or $B\flat_3$	233.1	$A\sharp_5$ or $B\flat_5$	932.3
B_1	61.74	B_3	246.9	B_5	937.8

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C_6	1046.5	C_7	2093.	C_8	4186.
$C\sharp_6$ or $D\flat_6$	1108.7	$C\sharp_7$ or $D\flat_7$	2218.	$C\sharp_8$ or $D\flat_8$	4435.
D_6	1174.7	D_7	2349.	D_8	4699.
$D\sharp_6$ or $E\flat_6$	1244.5	$D\sharp_7$ or $E\flat_7$	2489.	$D\sharp_8$ or $E\flat_8$	4978.
E_6	1318.5	E_7	2637.	E_8	5274.
F_6	1396.9	F_7	2794.	F_8	5588.
$F\sharp_6$ or $G\flat_6$	1480.0	$F\sharp_7$ or $G\flat_7$	2960.	$F\sharp_8$ or $G\flat_8$	5920.
G_6	1568.0	G_7	3136.	G_8	6272.
$G\sharp_6$ or $A\flat_6$	1661.2	$G\sharp_7$ or $A\flat_7$	3322.	$G\sharp_8$ or $A\flat_8$	6645.
A_6	1760.0	A_7	3520.	A_8	7040.
$A\sharp_6$ or $B\flat_6$	1864.6	$A\sharp_7$ or $B\flat_7$	3729.	$A\sharp_8$ or $B\flat_8$	7459.
B_6	1975.5	B_7	3951.	B_8	7902.

Appendix F: Power Law Dependences

Linear (First Power) Dependence If one pencil costs 10 cents, then two pencils costs 20 cents, and three pencils costs 30 cents, etc. The total cost of pencils depends linearly on the number of pencils purchased. We say that the cost varies as the *first* power of the number of pencils. Double the number of pencils and you double the cost.

Square Law (Second Power) Dependence If the side of a square has a length of 5 in., then the square has an area of 25 in.^{2}. If the side of a square has a length of 10 in., then the square has an area of 100 in.^{2}. The area depends on the *square* of the length of a side. We say that the area varies as the *second* power of the length. Double the length of a side and you multiply the area by four.

The linear law and the square law are examples of power laws. When there is a power law, it is possible to make statements about how many times larger or smaller a dependent quantity becomes when another quantity is varied.

Here is another power law, based on Eq. (3.1). That equation says that the frequency of a simple harmonic vibration varies as the square root of the stiffness, quantity s. To gain insight into how the equation works, we ask what happens if the stiffness is doubled. We will answer this question in two ways, one quick, the other slow and systematic. We reproduce Eq. (3.1) for convenient reference

$$f = \frac{1}{2\pi}\sqrt{\frac{s}{m}} \tag{3.1}$$

Quick Thinking: Square-Root Law (One-Half Power) Dependence Because the frequency varies as the square root of the stiffness, if we double the stiffness we multiply the frequency by the square root of 2. For instance, if the frequency is initially 100 Hz then after the stiffness is doubled, the frequency becomes $100 \times \sqrt{2}$, or 141 Hz. That is the quick way.

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Slow and Systematic We identify two conditions:

OLD CONDITION: Stiffness s gives frequency of 100 Hz.

NEW CONDITION: Stiffness $2s$ gives a new, unknown frequency.

Our task is to find the new frequency.

We write Eq. (3.1) for both conditions

$$f_{OLD} = \frac{1}{2\pi}\sqrt{s_{OLD}/m_{OLD}}$$

$$f_{NEW} = \frac{1}{2\pi}\sqrt{s_{NEW}/m_{NEW}}$$

We can divide the new equation by the old equation:

$$\frac{f_{NEW}}{f_{OLD}} = \frac{\frac{1}{2\pi}\sqrt{s_{NEW}/m_{NEW}}}{\frac{1}{2\pi}\sqrt{s_{OLD}/m_{OLD}}}.$$

The factors of 2π appear in both the numerator and the denominator, and they cancel. Likewise, the old mass and the new mass are the same and they too cancel. Therefore,

$$\frac{f_{NEW}}{f_{OLD}} = \frac{\sqrt{s_{NEW}}}{\sqrt{s_{OLD}}}.$$

Because the ratio of square roots is equivalent to the square root of the ratios,

$$\frac{f_{NEW}}{f_{OLD}} = \sqrt{\frac{s_{NEW}}{s_{OLD}}}.$$

We have succeeded in reducing this equation down to something we know, because we know that the new stiffness is twice the old stiffness. Therefore,

$$\frac{f_{NEW}}{f_{OLD}} = \sqrt{2}$$

or

$$f_{NEW} = f_{OLD}\sqrt{2}.$$

Now if f_{OLD} is 100 Hz then f_{NEW} is $100\sqrt{2} = 141$ Hz.

That is the end of the slow and systematic way, and it gives the same answer as the quick way. In the quick way we just look at the dependence and apply it directly. For instance, if the stiffness is increased by a factor of three (becomes three times larger), then the frequency is increased by the square root of 3. If it was originally 100 Hz, then after increase the frequency f_{NEW} is $100\sqrt{3} = 173$ Hz.

Inverse Square-Root Law (Minus One-Half Power) Dependence Equation (3.1) shows that the frequency depends inversely on the square root of the mass. "Inverse" means reciprocal. For instance, if the mass is doubled then the frequency becomes smaller. It becomes smaller by a factor of the reciprocal of the square root of two. That means that $f_{NEW} = f_{OLD}/\sqrt{2}$. If the frequency was originally 100 Hz, then after the increase in mass the frequency f_{NEW} is $100/\sqrt{2} = 71$ Hz.

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Appendix G: Telephone Tones

The telephone uses pairs of sine tones to encode the numbers in touch-tone dialing. The sine tones are added together with approximately equal amplitudes. The sine tone frequencies (in Hz) are given in the table below

Frequencies (Hz)	1209	1336	1477	1633
697	1	2	3	(not
770	4	5	6	yet
852	7	8	9	used)
941	*	0	#	

For example pressing the button called "4" sends a signal with components at 770 and 1,209 Hz.

The dial tone of a telephone is also a sum of sine tones, with frequencies 350 and 440 Hz.

The dial tone is different from the tones from the numeric keypad because it is consonant. The ratio 440/350 equals 1.257, which is an interval of a Major third. For comparison note that Appendix D gives the Major third as 1.25 in Just tuning, 1.266 in Pythagorean, and 1.260 in Equal temperament. Historically, there have been so many different meantone tunings with different ratios for the Major third, that the ratio 1.257 must have appeared in somebody's favorite tuning.

The telephone bandwidth for speech is 300 Hz to 3.3 kHz. The dynamic range is about 40 dB.

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Appendix H: Greek Alphabet

Some of the equations in this book use letters of the Greek alphabet. Here are all 24 of them, upper case, lower case, and their names.

A	α	Alpha
B	β	Beta
Γ	γ	Gamma
Δ	δ	Delta
E	ϵ	Epsilon
Z	ζ	Zeta
H	η	Eta
Θ	θ	Theta
I	ι	Iota
K	κ	Kappa
Λ	λ	Lambda
M	μ	Mu
N	ν	Nu
Ξ	ξ	Xi
O	o	Omicron
Π	π	Pi
P	ρ	Rho
Σ	σ	Sigma
T	τ	Tau
Υ	υ	Upsilon
Φ	ϕ	Phi
X	χ	Chi
Ψ	ψ	Psi
Ω	ω	Omega

♠

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Answers to Exercises

Chapter 1

1.1: A reflecting wall is part of the transmission path. A musical instrument is a source. A microphone is a receiver of acoustical waves, but in turn, it transmits electrical waves. So it could be considered either a receiver or part of the transmission path.

1.2: Fish perceive sound vibrations in the water, whales communicate, and dolphins and submarines echolocate, all with sound waves. Sounds travel very well in solids. It is believed that elephants hear waves that are transmitted through the ground. Putting your ear on a railroad track can give the first evidence that a train is coming. Mostly, however, the sounds that are transmitted through solids, such as a solid wall, are an annoyance.

1.3: If a sound source is far away it may be too weak to hear. Intense sounds can *mask* quieter sounds, such as the sounds of a conversation that you want to hear. Excess reverberation in a room can cause speech to be garbled. Listening to a single talker in a context with many other talkers is often possible, but stressful.

1.5: Ocean waves and X-rays would be examples.

1.6: The difference is $137 - 120 = 17$ kg. The ratio is $137/120 = 1.14$. The ratio has no units. It would be wrong to say that the ratio is 1.14 kg. The percentage change is $100 \times (137/120 - 1) = 14\,\%$.

1.7: The simple way to do this problem is to use the idea of Eq. (1.2) to realize that decreasing a number by 10 % is equivalent multiplying it by 0.9, and that increasing a number by 10 % is equivalent to multiplying it by 1.1. Thus decreasing and then increasing by 10 % multiplies the original number by $0.9 \times 1.1 = 0.99$ or 99 %. Hence the change is a 1 % loss. The order of decreasing and increasing events does not matter because the order of multiplying two numbers together does not matter.

W.M. Hartmann, *Principles of Musical Acoustics*, Undergraduate Lecture Notes in Physics, DOI 10.1007/978-1-4614-6786-1,

The long way to do the problem is to take a particular case. Start with \$100. Watch it decrease by 10 % to \$90 and then watch it increase by 10 % to \$99. Then do the reverse. Start with \$100. Watch it increase by 10 % to \$110 and then watch it decrease by 10 % to \$99.

1.8: (a) 10^5, (b) 10, (c) 10, (d) 10^5, (e) 1100.

1.9: (a) \$ 2.0×10^{10}, (b) \$ 1.0×10^{-1}, (c) 2.31×10^5, (d) 3.4×10^{-4}.

1.10: (a), (b), and (c) One million for all three.

Chapter 2

2.1: 1/20 s = 0.05 s = 50 ms. 1/20,000 s = 0.05 ms or 50 μs.

2.2: 1/60 = 0.01667 Hz.

2.3: (a) 2 s = 2,000 ms; (b) 30 μs = 0.03 ms.

2.4: (a) 16,384 Hz = 16.384 kHz. (b) 10 kHz = 10,000 Hz.

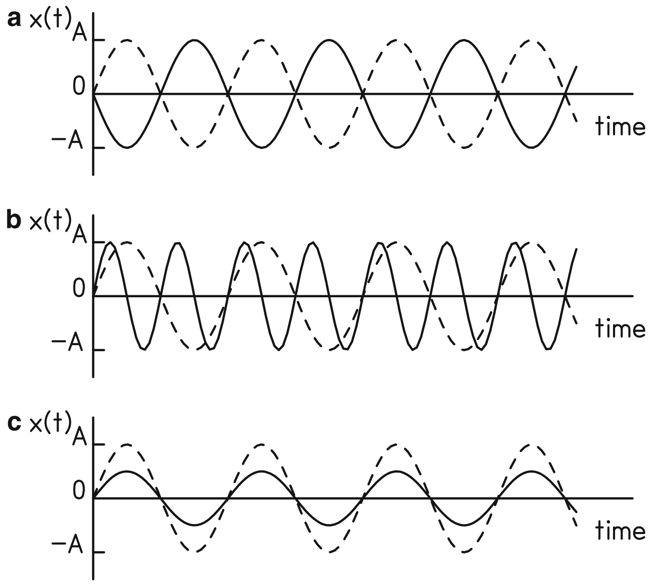
2.5: (a) $f = 1/T$ = 1/0.001 = 1,000 Hz = 1 kHz
(b) and (c) $T = 1/f$ = (1/10,000) s = 0.1 ms = 100 μs.

2.6: In Fig. 2.4 the dashed line is a reference, and its phase ϕ is zero. Compared to this reference another wave is lagging if its phase angle is between 0 and −180°.

2.7: "Infrared" light has a frequency that is too low to see. The term "infrared" means light with a frequency lower than red light—the lowest frequency of light we can see.

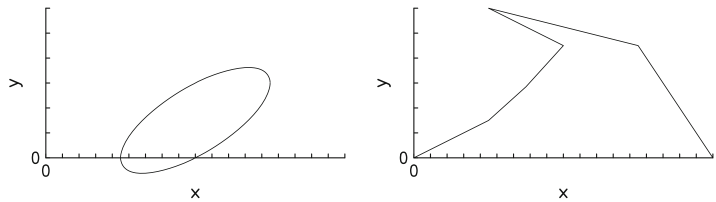
2.8: The band from 300 to 3,300 Hz comprises three octaves. You can prove that by starting at 300 and progressing by octaves: up one octave is 600, up two octaves is 1,200, up three octaves is 2,400. To go up a fourth octave leads to 4,800 Hz, but that is outside the band. Therefore, the telephone bandwidth is between three and four octaves.

2.9:



For Exercise 2.9 (a–c). The *dashed line* is the reference sine

2.9 (d): There are many possible answers to this question. Almost any scribble that a child would do is likely to be a function that is not single-valued. Figure below shows two simple functions. You can see that in each case knowing the value of x could leave you uncertain about the value of y.



For Exercise 2.9 (d). Two functions that are not single-valued

Chapter 3

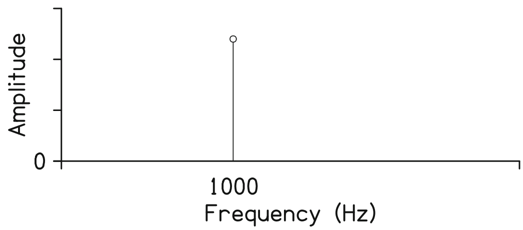
3.1: The square root of (100/0.25) is 20. Divide by 2π to get 3.18 Hz.

3.2: The mass is near the ends of the tines, and the stiffness comes from the shoulders. File the tines and the frequency increases because the mass gets smaller. File the shoulders and the frequency decreases because with less material there is less stiffness.

3.3: In engineering, one measures the lifetime as the time it takes for the amplitude to fall below its initial value by a factor of $1/e$, where e is the natural logarithm base, $e \approx 2.72$. That corresponds to 37 % of the initial value. In Fig. 3.1, the amplitude has dropped to 37 % after about four cycles. Each cycle has a period of 1/200 s. Therefore, the lifetime is 4 times (1/200) or 1/50, which is 0.02 s.

In acoustics, one measures the lifetime as the time it takes for the amplitude to fall below one one-thousandth of the initial value (0.001). By the that standard, the lifetime in Fig. 3.1 is at least as long as the entire time axis with about 18 cycles. Therefore, we'll say the waveform has disappeared after 18 times (1/200) or 18/200, which is 0.09 s.

3.4: The recipe for salt is as follows: "Add salt." That is the end of the recipe. The spectrum of a 1,000-Hz sine tone consists of a 1,000-Hz sine component. That is the end of the spectrum. The problem does not state an amplitude so we are free to pick any amplitude we like.

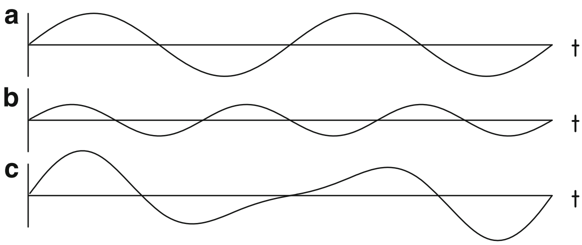


For Exercise 3.4. Spectrum of a 1,000-Hz sine tone

3.6: There are two people (systems) and a suggestion made by one of them is gladly followed by the other because the two people (systems) are in agreement on the matter in question.

3.7: (a) A child on a swing is like a swinging pendulum. The pendulum has one mode of vibration and if you drive the pendulum system at its natural frequency the amplitude of the oscillation becomes huge. Thus pushing a child in a swing is just like adding energy to a pendulum to increase the amplitude and make the child swing higher.
(b) The playground swing is not an exact example of resonance. The driving force is not a sine wave force. Instead, it is impulsive. If you push the child on every other cycle of motion, or every third, the amplitude increases, even though the driving frequency is one-half or one-third of the natural frequency of vibration. Strictly speaking, resonance is not like that. The driving frequency and the mode frequency have to be exactly the same for resonance.

3.8: (a) The frequency stays the same. (b) The frequency gets smaller by a factor of the square root of two. That means that if the frequency is originally f_1 then after doubling the mass the frequency becomes $f_1/\sqrt{2}$ or $f_1/1.414$.



For Exercise 3.9. Adding (a) and (b) point-by-point gives you (c)

3.9: (a)
(b) For every cycle of *a* there are 1.5 cycles of *b*. Therefore, the frequency of *b* is 1.5 times the frequency of *a*. If the frequency of *a* is 100 Hz the frequency of *b* is 150 Hz.

3.10: None of these is an example of resonance because the driving force is not a sine function and so the driving system does not have any particular frequency. The essential condition for resonance is that the frequency of the driving must equal a natural frequency of the driven system.

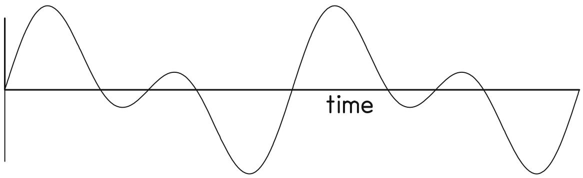
3.11: The top pattern shows the mode with the highest frequency and the bottom pattern shows the lowest. Complicated vibration patterns generally have high frequencies.

3.12: A larger volume leads to lower frequencies for the modes of vibration of air in the box. Lower frequency modes will resonate with tines of lower pitch.

Chapter 4

4.1: There are two transducers, a microphone and a miniature loudspeaker.

4.2: Many of different periodic waveforms might have two positive-going zero crossings per cycle. Here is one of them



For Exercise 4.2. A waveform to fool the frequency counter

The frequency counter adds a count to the total whenever the signal input circuit detects a positive-going zero crossing. With two of these per cycle, the total count, after any span of time, will be twice the number of cycles. Therefore, the counter will read twice the true fundamental frequency.

4.3: If the frequency really is 262.3 Hz then in 10 s there will be 2,623 positive-going zero crossings. If the gate is set to have a duration of 10 s, then the counter will register 2,623 counts. One only has to divide by ten to get an average of 262.3 counts per second. The down side to this technique is that the measurement takes a full 10 s. During that time the signal must be completely stable.

4.4: Guitar pickups, headphones, tape heads, compact disc read and write heads are optical transducers. A light bulb is also a transducer—it converts electrical energy into heat and light energy. The list goes on and on.

4.5: For a 500-Hz wave the period is 2 ms. To display two periods in a single sweep requires that the sweep be 4 ms long. The sweep period should be 4 ms. Therefore, the sawtooth wave that drives the horizontal deflection mechanism should have a rate of 1/(4 ms) or 250 Hz.

4.6: When it comes to comparisons, two is a special number. If you can compare two things then you can compare an indefinitely large number of things by working serially, two at a time. The dual-trace oscilloscope allows you to compare waveforms A and B. If you then compare waveform A with waveform C, you can infer the relationship between C and B. You can continue by comparing A with D and so on, without end. By contrast, a single-trace scope does not allow simultaneous comparisons.

4.7: You want the electron attracted to the left plate and be repelled from the right plate, so that it hits the left side of the oscilloscope screen. Thus since the electron has a negative charge the left plate should be positively charged, and the right plate should be negatively charged.

4.8: The answer is 800 Hz. Here is how you get it. 10 divisions $\times$ 0.5 ms/division = 5 ms for the entire screen width. 4 (cycles/screen)/5 (ms/screen) = 4 cycles/(5 ms) or 0.8 cycles/ms or 800 cycles/s, and so 800 Hz.

4.9: Two divisions would correspond to 2 $\times$ 0.5 ms or 1 ms. The period is slightly longer than 1 ms. Therefore, the frequency is slightly less than 1,000 Hz.

4.11: 0.5 v/cm vertically and 1 ms/cm (or 0.001 s/cm) horizontally.

Chapter 5

5.1: It is not the constant part of the pressure that makes a sound painfully loud. In fact, the first 0.2 ms of the sound in the figure have a pressure of 101,325 Pa, and there is no sound at all. It is the variation in pressure that causes a sound, and a large variation can create a sound that is so loud it is painful.

5.2: Remember that 1 m^2 is 10,000 cm^2. Therefore, to find the pressure in Newtons per square centimeter you divide by 10,000.

It is useful to apply the concepts of unit analysis to this exercise. The exercise requires that we convert from units of Newtons per square meter to units of Newtons per square centimeter. To do this operation we need to multiply by something that is equivalent to 1.0 that gets rid of the square meters and replaces it with square centimeters. The kind of thing that is likely to work is a fraction with square meters in the numerator and square centimeters in the denominator.

Algebraically it looks like this:

$$100{,}000\ \frac{\text{Newtons}}{\text{meter}^2} \times \left(\frac{1}{100}\frac{\text{meter}}{\text{cm}}\right)^2 = 10\,\frac{\text{Newtons}}{\text{cm}^2}$$

The factor equivalent to 1.0 is $1/100$ m/cm. This factor needs to be squared (it's still 1.0) to get the units right and the number $1/100$ gets squared right along with the units. Therefore, the number 100,000 gets divided by 10,000, not by 100.

5.3: $14.7\,\text{pounds/in.}^2 \times 4.45\,\text{N/pound} \times [1./0.000645]\,\text{in.}^2/\text{m}^2 = 101,419\,\text{N/m}^2$

5.4: According to Eq. (5.1) the speed of sound at −18 °C becomes

$$v = 331.7 + 0.6(-18) = 331.7 - 10.8 = 320.9\,\text{m/s},$$

and at 40 °C

$$v = 331.7 + 0.6(40) = 331.7 + 24 = 355.7\,\text{m/s}.$$

5.5 (a): Start with Eq. (5.1) and substitute Eq. (5.6) for T_C, the centigrade temperature.

$$v = 331.7 + 0.6[\frac{5}{9}(T_F - 32)] = 331.7 + \frac{1}{3}T_F - 0.6 \times \frac{5}{9} \times (32) = 312 + \frac{1}{3}T_F$$

and this is Eq. (5.7).

5.5 (b): Equation (5.7) has two terms. To convert to feet per second, they both need to be multiplied by the number of feet per meter. That number is

$$39.37\,\text{in./m} \times \text{ft}/12\,\text{in} = 3.28\text{ft/m}.$$

i.e. multiply by 3.28 to get Eq. (5.8).

5.6: (a) time = distance/speed = 10 m/ (344 m/s)
time = 0.029 s or 29 ms.

(b) With races decided by hundredths of a second (tens of milliseconds) a delay of 29 ms is significant. That is why a starter pistol is no longer used to start sprints. Instead there is a signaling device in each lane behind the runner.

5.7: The light can be considered to travel instantaneously. The delay of 2 s is all due to the relatively slow speed of sound. Therefore, distance $=$ speed $\times$ time = 344 m$/$s $\times$ 2 s, or distance = 688 m away. That's 0.688 km, i.e., about 0.7 km which is about 0.4 miles.

5.8: 1128 ft/s $\times$ (1/5280) miles/ft $\times$ (3600) s/h = 769 miles/h. Notice how the units cancel numerator and denominator.

5.9: The difference in audible frequencies is 20,000 – 20, which is about 20,000 Hz. The difference in visible frequencies is $7 \times 10^{14} - 4 \times 10^{14}$ which is about 3×10^{14} Hz. Obviously the range of visible light is enormously wider in terms of differences. But when it comes to the behaviors of physical and biological systems, ratios are more important than differences. In terms of ratios, the range of audible sound frequencies is far greater than the range of visible light frequencies. The human ear needs to be able to deal with a low frequency like 20 Hz and with a high frequency like 20,000 Hz, i.e. 1,000 times greater. The human eye needs to be able to deal with red light, 4×10^{14} Hz, and with violet light, 7×10^{14} Hz, less than a factor of two greater. In other words, the range of visible light is less than an octave, but the range of audible sound is ten octaves, i.e. $2^{10} = 1024 \approx 1000$.

5.10: At room temperature for 20 Hz, wavelength = 344/20 = 17 m. That is more than 55 ft.

At room temperature for 20,000 Hz, wavelength = 344/20,000 = 17 mm. That is less than 3/4 in.

5.11: At room temperature, frequency = 344 (m/s)/1 (m) = 344 Hz.

5.12: The formula $\lambda = v/f$ conforms to the convention that independent variables are on the right side of the equals sign and the dependent variable is on the left. The speed of sound is determined by the environment, and the frequency is independent of the environment. The wavelength is what one gets depending on the other two variables.

5.13: The period is half a second; the fundamental frequency is 2 Hz. Successive solid circles may be the pressure peaks of each successive clap. The distance between the circles is the wavelength, $\lambda = vT$, or $344 \times 1/2 = 172$ m.

5.14: The unit "gpf" stands for "gallons per flush." To determine the water usage in a day you need to know the number N of "flushes per day." You do the calculation:

1.6 gallons/flush $\times$ N flushes/day = 1.6 N gallons/day.

Notice that the units of "flushes" cancels between numerator and denominator, giving you a final answer in units of "gallons per day."

5.15: The low-frequency parts of a complex sound would arrive at you sooner than the high-frequency parts. If the sound source is close to you it might not matter much, but the relative delay would be longer for a distant source. A distant explosion would sound to you like a whooosh with a pitch that rises. A voice in the distance would become unintelligible because the high and low frequencies would not be synchronized.

5.16: Transverse waves cannot propagate in water. To carry a transverse wave, a medium must have some resistance to being bent. Water has a lot of resistance to being compressed (longitudinal waves) but no resistance to being bent.

Chapter 6

6.1: According to Eq. (6.3) the requirement for cancellation is that the difference in distance be an odd number of half wavelengths. For this exercise, the difference in distance is 0.4 m. Therefore, for parts (a) and (b):

A distance of 0.4 m is one half of a wave length if the wavelength is 0.8 m. Then the frequency is 344/0.8 or 430 Hz.

A distance of 0.4 m is three halves of a wave length if the wavelength is 0.2667 m. Then the frequency is 344/0.2667 or 1290. Hz. This frequency is just 3 times 430. Hz.

(c) In theory, there is no limit to the number of magic wavelengths and frequencies. Any odd number of half wavelengths will work: 5/2, 7/2, 9/2, etc.

6.2: (a) There are two ways to do this exercise. Both give the same answer.

First way—use the period. The period is 1 ms and half a period is 0.5 ms. For cancellation the delay ought to be half a period. One gets a delay of 0.5 ms with a distance difference of 0.5×344 or 172 mm which is 0.172 m. Therefore, cancellation occurs if you put the second source $1.000 + 0.172 = 1.172$ m away from you, or else $1.000 - 0.172 = 0.828$ m away from you.

Second way—use the distance. The difference in distance ought to be half a wavelength. For a 1,000 Hz tone the wavelength is $344/1000 = 344$ mm. Half a wavelength is 172 mm which is 0.172 m. Therefore, cancellation occurs if you put the second source $1.000 + 0.172 = 1.172$ m away from you or $1.000 - 0.172 = 0.828$ m away from you.

(b) In order to get complete cancellation the amplitudes of the two tones, as measured independently at the position of the receiver (namely at your position), need to be the same.

6.3: At low frequencies the wavelengths are larger than the dimensions of your room. If they are cancelled by a phasing error they are cancelled everywhere in your room. Getting a good bass response is a challenge for most audio systems. Recorded music is mixed with the left and right channels in phase for low frequencies to help with this situation. One thing that you really don't want to do is to cancel the bass.

When the wavelength is longer than the dimensions of the room, you can think of the loudspeakers as a pump, alternately compressing and expanding the entire volume of air in the room. For effective compression and expansion, the two speakers need to work together and not against one another.

By contrast, high frequencies have wavelengths that are much shorter than the dimensions of your room. High frequency sounds may cancel in one spot, but they reappear at another spot. This is a normal effect for any sound in any room; it is not special to stereophonic reproduction.

6.4: We know that the difference of the two frequencies is 5 Hz. If one of the frequencies is 440 then the other frequency must either be 445 or 435 Hz.

6.5: The beats between two sine waves are strongest if the amplitudes are equal because only then can complete cancellation occur. If the amplitudes are slightly different, there will always be a little bit of the larger wave left over when the phase relationship between the two waves leads to the greatest cancellation.

6.6: $\lambda = c/f$ or $3 \times 10^8/(6 \times 10^{14}) = 0.5 \times 10^{-6}$ = half a micrometer. A mirror needs to be polished so that bumps on its surface are smaller than that.

6.7: The wavelength of a 100-Hz wave is much longer than a few inches, but the wavelength of a 10,000-Hz wave is smaller than a few inches. The stone wall appears to be smooth to the 100-Hz wave, but it appears to be rough to the 10,000 Hz wave. The 10,000-Hz wave is diffusely reflected.

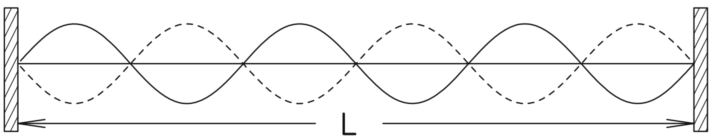
6.8: According to Fig. 6.8, sound waves bend toward the direction of cool air. Therefore, a channel is made by a layer of cool air between layers of warm air.

6.9: We use our ears to warn us of danger to ourselves or others. Apart from that, norml social interaction depends on verbal communication.

6.10: According to Fig. 6.11a, the slope of the input/output line is negative. The more the input pressure is positive, the more the output voltage is negative—and vice versa. The transducers described by Figs. 4.2 and 6.11 are actually the same except that Fig. 6.11 is *inverted.* It could be that the two transducers are identical except that the positive and negative wires coming from the transducer are reversed in Fig. 6.11.

Chapter 7

7.1: For the sixth mode there are six half wavelengths along the string, and five nodes.



For Exercise 7.1. The sixth mode of vibration of a stretched string

7.2: $f = v_s/(2L)$ or 154 (m/s)/(1.4 m) = 110 Hz. The third harmonic has a frequency three times larger, 330 Hz.

7.3: The linear density (μ) is determined by the choice of guitar string. The tension (F) is determined by stretching the string on tuning the instrument. The length (L) is determined by the player who presses the string down on the frets.

7.5: The frequency is inversely proportional to the string length. We set up a proportion for the unknown length L.

$$82/110 = L/70.$$

The solution is $L = 52.2$ cm.

7.6: The top E frequency is four times greater or about 328 Hz. To get this high frequency the speed of sound needs to be four times faster.

7.7: Bend notes by stretching strings, increasing the tension. Some rock guitars have hinged bars (whammy bars) that stretch all the strings at once.

7.8: Double the tension and the speed of sound increases by a factor of the square root of 2. Therefore, the frequency increases by the same factor and becomes 141 Hz.

7.9: Start with F/μ. It is $\frac{\text{kg}\cdot\text{m}/[(\text{s})(\text{s})]}{\text{kg}/\text{m}}$. The units of kg divide out and the denominator just becomes $1/\text{m}$. Thus F/μ has units of m^2/s^2, and the square root of this is m/s as advertised.

7.10: The role of the guitar body is to radiate the sound. The frequencies of the sound are almost entirely determined by the string only. However, because the bridge and nut of the guitar body vibrate a little, the boundary conditions change slightly, and there is a tiny effect of the body on the playing frequencies. In this book, we always neglect that tiny effect.

7.11: In a wind instrument, the air pressure within the body of the instrument vibrates. In a drum, the drumhead (a membrane) vibrates. Both have modes of vibration. These modes are equivalent to standing waves. For instance, these modes have nodes at certain places.

7.12:

$$f_n = \frac{n}{2L}\sqrt{\frac{F}{\mu}}$$

7.13: The easiest solution is to recognize that the flat lines at 4 and 12 ms are separated in time by half a period. Therefore, half a period is $12 - 4 = 8$ ms and a full period is 16 ms. Therefore, the frequency is $1000/6 = 62.5$ Hz.

Chapter 8

8.1: (a) For open-open pipe $f = v/(2L) = 1130/(2) = 565\,\text{Hz}$.

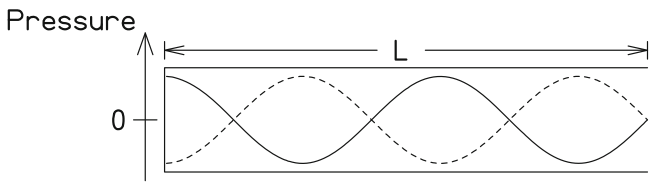
(b) For open-closed pipe $f = v/(4L) = 1130/(4) = 282\,\text{Hz}$.

(c) The open-closed pipe has natural mode frequencies that are odd-integer multiples of the lowest mode frequency. Only modes with these frequencies are allowed to exist in this pipe. All the even numbered harmonics are missed.

8.2: (a) For open-closed pipe number of nodes = mode number −1

(b) For open-open pipe number of nodes = mode number −1,
i.e., the relationships are the same.

8.3 Mode number four for a open-closed pipe has three nodes as shown $f = v/(4L) \times 7$:



For Exercise 8.3. The fourth mode of air vibration in an open-closed pipe

8.4: The drawing should look like Fig. 8.2 except that there will be three nodes instead of one.

8.5: To put this matter in a clear light, we write an equation for the lowest frequency of a pipe that is open at both ends. Putting in the end correction explicitly, we find $f = v/[2(L + 0.61d)]$, where L is the measured length of the pipe and d is the diameter. The neat way to write this is to rearrange the denominator slightly to get.

$$f = v/[2L(1 + 0.61d/L)].$$

Now the formula looks just like our old friend, $f = v/2L$, with a denominator that is multiplied by something that is pretty close to 1.0 if d/L is small. This makes it clear that it is the diameter compared to the measured length, d/L, that causes the end-correction deviations.

8.6: The open-closed pipes need to be slightly longer because only one end correction is applied to them and in order to get a precise octave, the *effective* lengths of the two pipes need to be the same.

8.7: (a) The equation $f = v/(2L)$ can be solved for length L, to give $L = v/(2f)$. Therefore, $L = 344/(2{\cdot}16) = 10.75$ m. (b) One meter is 3.28 ft, and so $L = 35.3$ ft. (c) You can use an organ pipe that is half as long if you close off one end. Then, for an open-closed pipe $L = v/(4f)$. A 17- or 18-ft pipe will fit comfortably in the available space. (d) The solution does not come without cost. Closing off one end eliminates the even-numbered harmonics in a cylindrical pipe.

8.8: If an organ pipe is blocked off anywhere along its length, the pipe becomes an open-closed pipe with an acoustical length equal to the unblocked part. Therefore, if a designer wants a pipe to be 4 m long, but the pipe needs to have an acoustical length of only 1 m, a rigid barrier can be put into pipe at the 1-m mark and the remaining 3 m is just for show.

8.9: The tension and the mass density of a stretched string determine the speed of sound on the string. By contrast, the speed of sound of air in a pipe is determined by the air and its temperature. With air, we normally have to take what we get. To make a pipe more interesting, we could try to control the speed of sound in air. For instance, with refrigeration coils and a some blow torches we might be able to play a few different notes on the same organ pipe. Upon consideration, it does not seem like a very good idea.

Alternatively we might use a gas other than normal air. The speed of sound in helium is about three times the speed of sound in ordinary air. The speed of sound in nitrous oxide N_2O is about 80 % of the speed of sound in air.

8.10: After the initial impact, the air in the pipe continues to vibrate for a little while to make the tonal sound. If the pipe is pressed into your palm it becomes an open–closed pipe, but if the pipe was excited by extracting your finger, it is still an open–open pipe. The latter leads to a pitch that is an octave higher than the former.

8.11: $f = v/(4L) = 34400/(4 \cdot 60) = 143$ Hz, close to $D_3 = 147$ Hz.

8.12: The diameter converted to cm is $1.5 \times 2.54 = 3.81$ cm. The end correction is $0.305 \times 3.81 = 1.16$ cm.

$f = v/4L_{effective}$.

If $f = 65.4$ Hz, then $L_{effective} = 34400/(4 \times 65.4) = 131.50$ cm.

Taking the end correction into account, we must cut a length of pipe 130.34 cm long.

8.13: (a) For the longest possible cylinder (287 mm), expressing the speed of sound in mm/s, $344000/(4 \cdot 287) = 299.7$ Hz. The shortest cylinder is 232 mm shorter, or 55 mm long. Then, $344000/(4 \cdot 55) = 1564$ Hz. (b) The frequency ratio is 1564/299.7 or 5.2. This ratio is greater than the ratio for two octaves, which would be a ratio of 4.

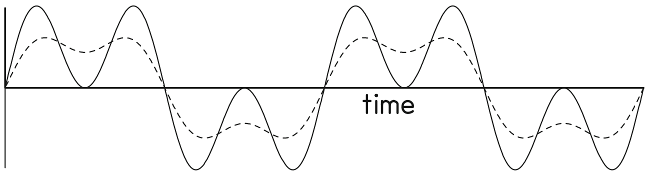
Chapter 9

9.1: The problem only defines the relative amplitudes. It does not specify the absolute amplitude of any harmonic. Therefore, we can arbitrarily choose the amplitude of the fundamental to be 1.0.

Then

$$\text{Signal} = 1.0 \sin(360 \cdot 250\,t + 90) + 0.5 \sin(360 \cdot 500\,t + 90) +$$
$$0.333 \sin(360 \cdot 750\,t + 90) + 0.25 \sin(360 \cdot 1000t + 90).$$

9.2: The easy way to think about the change is to realize that the original complex signal looks like the fundamental except that the third harmonic puts dimples in each positive and negative peak. When the amplitude of the third harmonic grows to be as large as the fundamental, the dimples become large enough to cancel the fundamental entirely and bring the waveform to zero momentarily, as shown in the figure.



For Exercise 9.2. The *dashed line* is the reference. The *solid line* is the answer

9.3: (a) After 1/200 s, the 200-Hz wave has executed one complete cycle and is ready to begin again.

After 1/200 s, the 600-Hz wave has executed three complete cycles and is ready to begin again.

Therefore, after 1/200 s the sum of those two waves is ready to begin again and it has executed one complete cycle. Thus the period is 1/200 s. These facts are unrelated to the relative phases of the two components.

(b) A wave with components at 200 and 500 Hz has a period of 1/100 s.

(c) A wave with components at 200 and 601 Hz has a period of 1 s.

(d) A wave with components at 150, 450, and 750 has a period of 1/150 s. The wave consists of the first three odd-numbered harmonics only.

(e) A wave with components at 220, 330, 550, and 660 Hz has a period of 1/110 s. The fundamental at 110 Hz is missing and the periodicity comes from harmonics 2, 3, 5, and 6.

9.4: There is no perfect way to identify a psychological sensation like pitch with a physical measurement like frequency such that pitch can be expressed in Hz. What is done in practice is to define pitch in Hz as the frequency of a sine tone that leads to the same sense of pitch, i.e., the same height on a musical scale. For instance, we might play a 400-Hz sawtooth to a listener and ask the listener to adjust the frequency of a sine tone so as to match the pitch of the sawtooth. Suppose that the listener adjusts the sine frequency to be 401 Hz. Then we say that for this listener, on this experimental trial, the pitch of the 400-Hz sawtooth is 401 Hz.

9.5: Both the amplitude spectrum and the phase spectrum contribute to the shape of the waveform. Ohm's law of phase insensitivity applies to the *sound* of the waveform, not to its shape on an oscilloscope.

9.6: The next three terms are:

$$\frac{1}{4}\sin(360\cdot 1600\,t+180)+\frac{1}{5}\sin(360\cdot 2000\,t+180)+\frac{1}{6}\sin(360\cdot 2400\,t+180).$$

9.7 Drill answers:

(a) Four harmonics; (b) four lines on a spectrum analyzer; (c) second harmonic has the highest amplitude; (d) 5 units; (e) 60°; (f) Yes, a phase can be negative; (g) Yes, 240° is equivalent to $-120°$ because those two angles differ by 360°; (h) The period is 1/200 s; (i) the quantity ft must have physical units of cycles. Therefore, if f is in Hertz, then t must be in seconds.

Chapter 10

10.1: (a) The square of (1/3) is 1/9.

(b) $10\log(1/9) = -10\log(9) = -10\times(0.954) = -9.54$ dB

10.2(a): From Eqs. (10.1) and (10.2) we know that $I \propto A^2$ and that $I \propto 1/d^2$. Substituting the first proportionality into the second, it follows that $A^2 \propto 1/d^2$, and the square root leads to $A \propto 1/d$, as advertised.

10.3(a): $I = P/(4\pi d^2) = 10/(4\pi 10^6) = 8\times 10^{-7}$ W/m^2
(b): $10\log(8\times 10^{-7}/10^{-12}) = 59$ dB.
(c): An airplane in flight is perhaps the only condition where one finds 1 km of empty air in all directions, and an airplane can make 10 W of acoustical power. Alternatively—maybe a skydiving brass band?

10.4: Intensity dependence on distance (d) from the source for worlds with different dimensionality (D). As dimensionality goes from 3-D to 2-D to 1-D, the wavefront spreading law goes from (inverse second power, $1/d^2$) to (inverse first power, $1/d$) to (inverse zeroth power, 1). Zeroth power means no dependence on distance at all. Therefore, in a one-dimensional world there is no intensity loss due to wavefront spreading. The only losses are due to friction (absorption) in the medium of transmission.

10.5: $10\log(30/1) = 10 \times (1.477) = 14.8$ dB.

10.6: $L_2 - L_1 = 10\log(I_2/I_1) = 10\log(2) = 10 \times 0.301 = 3$ dB.

10.7: (a) The statement of the situation in the exercise is that $20 = 10\log(I_{\text{tmb}}/I_{\text{flute}})$
Therefore, $I_{\text{tmb}}/I_{\text{flute}} = 10^2 = 100$

(b) $I_{\text{tmb}}/I_{\text{flute}} = [A_{\text{tmb}}/A_{\text{flute}}]^2$, and so $[A_{\text{tmb}}/A_{\text{flute}}] = 10$.

10.8: We begin with the statement that

$$L_2 - L_1 = 10\log(I_2/I_1)$$

Now $(I_2/I_1) = (A_2/A_1)^2$, so $L_2 - L_1 = 10\log[(A_2/A_1)^2]$.
But the log of a square is twice the log, and so $L_2 - L_1 = 20\log(A_2/A_1)$

10.9:

$$130 = 10\log(I/10^{-12})$$
$$13 = \log(I/10^{-12})$$
$$I = 10^{13} \times 10^{-12} = 10\,\text{W/m}^2.$$

Chapter 11

11.1: (a) The ear canal is said to be about 2.5 cm long. It resembles an open-closed pipe. Therefore, the resonant frequency is

$$f = v/(4L) = 34400/(4 \times 2.5) = 34400/10 = 3440\,\text{Hz}.$$

(b) The second mode of such a pipe has the frequency

$$f = 3v/(4L) = 3 \times 34400/(4 \times 2.5) = 10,320\,\text{Hz}.$$

11.2: The speed of sound in air is 344,000 mm/s. Five times that speed is 1,720,000 mm/s. Therefore, the time to go twice the distance of 35 or 70 mm in the fluid is

$$\text{time} = 70/1,720,000 = 0.000,041\,\text{s} = 41\,\mu\text{s}$$

11.3: The acoustic reflex is triggered by a whistle blowing next to your ear and other continuous sounds containing frequencies in the range from 1,000 to 4,000 Hz.

11.4: (a) For 440 Hz, about 9 mm from the apex. (b) For 880 Hz, about 13 mm from the apex.

11.5: Expect intervals of 1/440 s, 2/440 s, 3/440 s, and so on.

11.6: The lost hair cells must be those near the oval window.

11.7: There will be more interference when an intense 1,000-Hz tone masks a weaker 1,300-Hz tone because excitation patterns on the basilar membrane are unsymmetrical with greater excitation extending toward the base, i.e., toward the direction of higher frequency places.

11.8: For 5 mm,

$$f_c = 165(10^{(0.06)\cdot(5)} - 1) = 165(1.995 - 1) = 164\,\text{Hz}.$$

For 10 mm,

$$f_c = 165(10^{(0.06)\cdot(10)} - 1) = 165(3.981 - 1) = 492\,\text{Hz}.$$

etc.

11.9 (a): We start with the Greenwood equation,

$$f_C = 165(10^{az} - 1)$$

Our goal is to get z by itself on one side of the equation. We proceed systematically, doing what we can to move in the right direction at each step.

$$f_C/165 = 10^{az} - 1$$
$$1 + f_C/165 = 10^{az}$$
$$\log_{10}(1 + f_C/165) = az$$
$$\tfrac{1}{a}\log_{10}(1 + f_C/165) = z$$

Done!

(b): If $f_C = 440$ then, with $a = 0.06$,

$$\frac{1}{0.06}\log_{10}(1 + 440/165) = z = 9.4\,\text{mm}.$$

11.10: In the normal human auditory system there are half a dozen stages of neural processing between the cochlea and the final destination in the auditory cortex. Every one of these stages is tuned in frequency, with slightly different spatial regions for tones of different frequency. To best simulate a normal system, a cochlear nucleus implant ought to have multiple electrodes that the surgeon can place in appropriate regions of the midbrain. Cochlear nucleus implants are, in fact, made in this way.

11.11: (a) Microphone. (b) Microphone, Speech processor, Transmitter. (c) About one turn.

Chapter 12

12.1: Horizontal axis is frequency in Hz; vertical axis is level in dB.

12.2: About 41 phons.

12.3: The tone is inaudible. The point at 200 Hz and 10 dB is below the threshold of hearing curve.

12.4: The threshold of hearing is 0 phons whatever the frequency. That is the definition of zero phons. A sound level of 0 dB is also related to the threshold of hearing by definition, but the dashed line in Fig. 12.1 makes it clear that the actual decibel level for threshold depends on frequency. So long as the frequency is not too low or high, the threshold (dashed line) is rather close to 0 dB, i.e. close to $10^{-12}\,\mathrm{W/m^2}$.

12.5: From the equal-loudness contours, about 32 dB.

12.6: Choose 6,000 Hz. The equal-loudness contours dip down near 6,000 Hz, suggesting that this is a loud tone by itself. That is not true for 100 Hz. Also, this tone will excite different neurons from the 1,000-Hz tone. That cannot be said for the 1,100-Hz tone. Because neurons are compressive, you would prefer to spend your energy on different neurons.

12.7: (a) The equation says:

$$\psi \propto I^{0.3} \tag{12.1}$$

Therefore,

$$\psi_2/\psi_1 = (I_2/I_1)^{0.3}$$

We are told that the ratio of the intensities is 10 to 1. Therefore,

$$\psi_2/\psi_1 = 10^{0.3} = 1.995 \approx 2.$$

(b)

$$\psi_2/\psi_1 = (I_2/I_1)^{0.3} = 50^{0.3} = 3.2$$

Therefore, 3.2 times louder.

12.8: (a) We know that the intensity ratio raised to the 0.3 power is 4. Therefore, the intensity ratio is 4 raised to the power 1/0.3, which is 102. Therefore, about 100 times. (Because 4 is the square of 2, we might have expected to find that two doublings would require multiplying the intensity by 10 twice.)

(b) $10 \times 10 \times 10 \times 10 = 10,000$. Therefore, 10,000 times.

12.9: The dip in equal loudness contours is caused by the resonance of the ear canal near 3,500 Hz.

12.10: From Eq. (12.7) we calculate that a level difference of 7 dB corresponds to a loudness ratio of 1.6. Thus, the trombone sound is 1.6 times louder.

12.11: In order to understand the effect of road noise on the loudness of a car radio, we need to assume that the auditory system can somehow separate the sound of the radio from the background noise and evaluate the loudness of the radio independent the noise. Just considering the total neural firing rate can never explain the effect. It is very likely that separating the sound of the car radio depends on the synchrony of neurons as they respond to the radio. The road noise suppresses the neural response that is synchronized with tones from the radio. When the road noise is gone the neural response is not suppressed.

12.12: Loudness is defined as a psychological quantity, in contrast to intensity, which is a physical quantity. Therefore, if it sounds louder, it really is louder, regardless of how the intensity might or might not change.

12.13: The noisy fan is 8 times louder than the quiet fan. 4 sones/ 0.5 sones = 8. The purpose of the sone scale is to make comparisons like this really easy.

12:14: In comparison with 1,000 Hz, the contours at 62 Hz are very close together. Therefore, the loudness increases rapidly as the sound level increases. At 62 Hz, as the sound level increases from 50 to 60 dB, the loudness level increases from 20 phons to 40 phons—twice as large an increase.

Chapter 13

13.1: 200 Hz appears at 5 mm from the apex and 2,000 Hz at 17 mm from the apex. The maxima for these two tones are 12 mm apart. That is a huge distance.

13.2: For Part a the histogram shows the following numbers of interspike intervals:
For 1 cycle (1/200 s) 5 intervals
For 2 cycles (2/200 s) 4 intervals
For 3 cycles (3/200 s) 2 intervals
For Part b, the histogram is
For 1 cycle (1/200 s) 2 intervals
For 2 cycles (2/200 s) 1 interval
For 3 cycles (3/200 s) 1 interval

For 4 cycles (2/200 s) 1 interval
For 5 cycles (3/200 s) 1 interval

13.3: These components look like harmonics 5, 6, and 7 of 150 Hz. The integers 5, 6, and 7 are consecutive and not very large. The pitch is very likely to be 150 Hz.

13.4: (a) This tone looks rather like harmonics 2, 3, and 4 of 200 Hz, but the harmonics are mistuned. We get a decent fitting template if we say that the template has a fundamental frequency of 620/3 or 206.7 Hz. The template expects a second harmonic at 2×206.7 or 413 Hz, and this is flat compared to the real component at 420 Hz. The template expects a fourth harmonic at 4×206.7 or 826.6 Hz, and this is sharp with respect to the real component at 820. In the end, the template is too low for the second harmonic, too high for the fourth harmonic, and just right for the third. The brain regards this template as a reasonable compromise and perceives this tone to have a pitch of 206.7 Hz.

(b) This problem is an extension of Exercise 13.4 (a). More mistuned harmonics (five and six) are added to the top end. If we think about a template with a fundamental frequency of 206.7 Hz, the sixth harmonic has a frequency of 1,240 Hz, and this is too far above the real component at 1,220 Hz. We are going to have to decrease the fundamental frequency of the template to get a better fit. We fit the center component (looking like the fourth harmonic) by choosing a fundamental frequency of $820/4 = 205$ Hz.

13.5: An analytic listener recognizes that the only difference between tones "1" and "2" is that 800 Hz component goes down to 750 Hz. This listener says that in the sequence "1" followed by "2" the pitch goes down. Most listeners hear Experiments a and b in this way.

A synthetic listener interprets tone "1" as the 4th and 5th harmonics of 200 Hz. This listener interprets tone "2" as the 3rd and 4th harmonics of 250 Hz. This listener says that in the sequence "1" followed by "2" the pitch goes up. The continuity in Experiment b should cause the synthetic listener hear the sequence differently. It should help him hear analytically if he wants to hear like most people.

The ambiguous sequence in Experiment a appears as demonstration number 25 on a compact disc entitled "Auditory Demonstrations," by A.J.M. Houtsma, T.D. Rossing, and W.W. Wagenaars, and is available from the Acoustical Society of America. It was reviewed by W.M. Hartmann, "Auditory demonstrations on compact disk for large N," J. Acoust. Soc. Am. *Review and Tutorial* **93**, 1–16 (1993).

Chapter 14

14.1: (a) $20 \log_{10}(1.18/1) = 1.44$. Thus, about 1.4 dB.

(b) $20 \log_{10}(0.68/0.5) = 2.67$. Thus, about 2.7 dB.

14.2: (a) The wavelength of a 100-Hz sound is about 3.4 m. This is about 20 times the diameter of the head. The wavelength is so much larger than the head diameter that the sound diffracts around the head and is just as large in the far ear as it is in the near ear. (b) To use the interaural level difference at a low frequency like 100 Hz, it would be necessary to have a head about five times larger than normal.

14.3: From Fig. 14.2 it looks as though there is about 1.8 dB separating the 10° curve from the 45° curve at 2,000 Hz. If the smallest detectable difference is 0.5 dB, then the range from 10° to 45° is divided into about 1.8/0.5 or 3.6 intervals. On average these intervals correspond to $(45 - 10)/3.6$ or 10°. Thus, the ILD is seen to provide only a crude estimation of azimuthal location.

14.4: (a) The left-hand side is time. The right-hand side is a distance divided by a speed, i.e., (meters) divided by (meters/second). The units of meters cancels in numerator and denominator leaving us with 1/(1/s), which is simply seconds.

(b) For 30°, 45°, and 60°, the ITD values from Eq. (14.1) are 382, 540, and 661 μs, respectively.

14.5: Localization is better in the horizontal plane in that it is possible to discriminate between two locations that are closer together in the horizontal plane. A listener can judge differences of about 1.5° in the forward direction in the horizontal plane, but can only judge differences of about 4° in the vertical plane.

14.6: The outer ear acts as a direction-dependent filter. As a result, a sound that comes from the back has a different spectrum, as it reaches the eardrum, compared to a sound that comes from the front.

14.7: The most important frequencies in front–back localization are above 6,000 Hz. Older people tend to lose high-frequency hearing, and many of them do not hear frequencies higher than 6,000 Hz. Because important speech frequencies are lower than 6,000 Hz, you would expect older listeners to lose front–back discrimination while maintaining good understanding of speech. Left–right (azimuthal) localization does not depend on such high frequencies. Low frequencies near 500 Hz tend to dominate.

14.8: According to the precedence effect, the first arriving sound determines the perceived location of the sound, even if later arriving sounds—early reflections and reverberation from a room—seem to point to different locations.

Chapter 15

15.1: Sound is transmitted through the wall by means of the studs. The drywall on the noisy side shakes the studs, and the studs shake the drywall on the other side. Putting fiberglass insulation in the gaps between the studs does not change this sound transmission process at all and is not an effective noise reduction method. What is effective is to use a staggered-stud wall where no stud is in physical contact with both sides of the wall. Once there is no contact between the wall surfaces it makes sense to add the fiberglass absorber.

15.2: Many rooms are acoustically bad because the heating, ventilating, air conditioning (HVAC) systems are poorly installed and make a lot of background noise. Other rooms may suffer from too much reverberation. Many environments have public address systems that are incorrectly adjusted or incorrectly used.

15.3: Compared to a plaster wall, a brick wall is not rough on an acoustical scale. The small bumps and valleys of a brick wall are not comparable to the wavelength of audible sound, and they do not produce diffuse reflections.

15.4: Decibels again—just when you thought it was safe to forget about them. If 99 % of the intensity is absorbed then 1 % of the intensity remains. The attenuation is found from the ratio of what remains to the initial amount. That ratio is 1 % or 0.01.

$$L_{remains} - L_{initial} = 10 \log(0.01) = -20.$$

15.5: Nuclear reactions boil water to make steam to turn a turbine to run a generator to make electricity. Nothing about the final fate of all this energy is fundamentally changed by this generating process.

15.6: $T_{60} = 0.16\frac{V}{A_T} = 0.16\frac{2000}{218} = 1.5\,\text{s}.$

15.7: To calculate the reverberation time using the Sabine equation we need the volume and total absorbing area. The volume is easy: 6.3 by 7.7 by 3.6 or $174.6\,\text{m}^3$.

The total absorbing area requires more work. There are two kinds of surfaces, plaster and vinyl on concrete. According to the table, the absorption coefficients at 500 Hz are 0.05 and 0.03, respectively.

There are two walls 6.3 by 3.6 m, each with an area of $22.7\,\text{m}^2$.

There are two walls 7.7 by 3.6 m, each with an area of $27.7\,\text{m}^2$.

There is a ceiling and a floor, 6.3 by 7.7 m, each with an area of $48.5\,\text{m}^2$.

Therefore,

$$A_T = 2 \times 22.7 \times 0.05 + 2 \times 27.7 \times 0.05 + 48.5 \times 0.05 + 48.5 \times 0.03 = 8.92\,\text{m}^2.$$

Then from the Sabine equation,

$$T_{60} = 0.16(174.6)/(8.92) = 3.1\,\text{s}.$$

That is a long reverberation time. In fact, the description of the room corresponds to a reverberation room used in acoustical testing.

15.8: The purpose of a shower is to provide an environment for singing, and most of the energy in the singing voice is below 500 Hz. We will assume that all surfaces are glazed tile and use an absorption coefficient of 0.01.

(a) Assume that the dimensions are 3 ft by 3 ft by 7 ft high. Assume that the shower has a glass door that also has an absorption coefficient of 0.01. Then the volume is

63 ft^3, and the total absorbing area (A_T) is $4 \times 3 \times 7 \times 0.01 + 2 \times 9 \times 0.01$. The A_T is thus 1.02 ft^2.

Sabine's formula involves $V/(A_T)$, which is 63/1.02 or 61.8 ft. To use the formula we need to convert that dimension to meters, and we find 18.8 m. Thus, $T_{60} = 0.16 \times 18.8 = 3$ s, and that makes a wonderfully long reverb time. Unfortunately, the human singer inside the shower adds absorption and reduces the reverberation time.

(b) Assume that everything is the same as in (a) except that there is no door. The shower is open on one side and that leads to an additional absorbing area of $3 \times 7 \times 1 = 21$ ft^2! Thus, $T_{60} = 0.16 \times 0.87 = 0.13$ s, and that is no fun at all.

15.9: If all the absorption coefficients are 1.0, then 100 % of the sound is absorbed on the walls. Such a room is an idealization and would be called "perfectly anechoic." Alternatively, such a room is really no room at all because it behaves just like empty space. With 100 % of the sound absorbed there is no sound reflected back into the room and the reverberation time is zero. But the Sabine formula says that $T_{60} = 0.16\,V/(A_T)$, where A_T would be the total surface area in the room (multiplied by 1.0). All of the quantities are finite numbers, and the formula predicts a finite reverb time, contrary to the truth about this unusual, ideal room.

15.10: First, it is assumed that volume and total absorbing area are computed in the metric system. Then, the numerator (volume) has dimensions of cubic meters and the denominator (area) has dimensions of square meters, so the fraction has dimensions of meters. In terms of a dimensional equation:

$$\text{meters}^3/\text{meters}^2 = \text{meters}.$$

15.11: The volume of B is 8 times the volume of A. The total absorbing area of B is 4 times that of A. Consequently, the reverberation time of B is twice that of A.

Chapter 16

16.1: Charge is the basic quantity of electricity. Voltage is a force on a charge. The force of voltage causes charge to flow. Current is the rate of charge flow.

16.2: Both the dynamic microphone and the generator use the electromagnetic generator principle to convert motion into an electrical voltage.

16.3: No, not possible. Although a permanent magnet, like the magnet in a loudspeaker driver or a steel compass needle, has no *apparent* electrical current there actually is a microscopic current. On an atomic level the magnetism of a permanent magnet is caused by tiny electron currents and spins in the magnetic material.

16.4: For the loudspeaker to operate, the electrical current comes from the power amplifier and flows through the voice coil. The magnetic field comes from permanent magnets that surround the voice coil. The force is transmitted to the cone or dome of the speaker, having large surface areas, in order to move air creating an acoustical pressure wave.

16.5: The electron beam in the CRT is a flow of electrons, and that is a current. The loudspeaker has to have permanent magnets in order to work at all, and some of the magnetic field can leak out and cause a force on the beam in a nearby CRT. This is the motor principle, even though the current is not actually carried by a wire. The force bends the electron beam, which causes objects to appear in slightly wrong places on the screen, thus distorting the picture. Because of this effect, loudspeakers intended to go next to computer monitors are shielded to reduce the leakage of magnetic field.

16.6: The coil is wrapped around the magnet and there would seem to be no relative motion between the wire of the coil and the magnet. Therefore, it would seem that the generator principle would not apply. However, we know that these pickups *do* work, and so we have to look further. We begin by noting that the generator principle only requires motion between the coil of wire and the magnetic *field.* That's the key. What actually happens is that the motion of the steel guitar string close to the magnet of the pickup causes a small change in the amount of magnetic field that threads its way through that magnet. Thus the magnetic field is changing, and the coil of wire is right there in the presence of this changing field. Thus a voltage is induced in the coil of wire.

16.7: The coil is lighter than the magnet. Attaching the coil to the diaphragm, instead of the magnet, would lead to less inertia and more relative motion between coil and magnetic field.

16.8: (a) Physically, both the motor and the generator consist of a coil of wire in a magnetic field. The coil of wire is attached to a shaft which transmits mechanical rotation. The wire has two ends, and an electrical current can flow in and out. If a current passes through the coil of wire, there is a force on the wire because of the motor principle. That force can turn the shaft. That makes a motor. It can do your laundry or start your car engine. In the reverse operation, an external source of mechanical energy turns the shaft, and a voltage is induced between the two ends of the wire because of the generator principle. (b) Yes, a loudspeaker can be used in reverse to serve as microphone. It's easy to show if you connect the two terminals of a loudspeaker driver (a woofer works very well) to the input to a sensitive oscilloscope. If you then talk into the loudspeaker you will see the generated voltage on the 'scope.

16.9: When the wire of the voice coil melts it no longer carries any current. With no current, the speaker cone feels no force and the loudspeaker driver makes no sound at all. The only practical remedy is to replace the driver, which could be expensive.

Chapter 17

17.1: Both 60 and 50 Hz are low frequencies where the equal loudness contours are rising steeply. For such low frequencies, the lower the frequency the harder it is to hear. Therefore, the answer to this exercise is, Yes, all other things being equal it is harder to hear hum at 50 Hz than 60 Hz. There is more. Hum normally includes second and third harmonics—possibly even stronger than the fundamental. Then, the fact that 100 Hz is harder to hear than 120 Hz, and 150 Hz is harder to hear than 180 Hz also would make the European hum less objectionable.

17.2: In order to avoid distortion, the output of a power amplifier must have the same shape as the input. When an amplifier is overdriven, the amplifier has reached the limit of its output voltage. It cannot produce a voltage that is high enough (or negative enough) to capture the peak (or valley) required by the shape of the input. The resulting change in waveform shape is nonlinear distortion.

17.3: (a) Chords consist of many frequencies—the fundamentals and harmonics of all the notes in the chord. Nonlinear distortion causes difference tones among all these components in the chord. That leads to a mass of spectral components that are the characteristic sound of the distorted rock guitar. (b) What sounds bad depends on the context. In heavy metal music distortion is used as an artistic device.

17.4: You should choose the 500-Hz tone (d). With the four sources available to you, you can only test for harmonic distortion. You will listen for spectral components in the output that do not occur in the input. With the 500-Hz tone you will easily hear the second and third harmonics at 1,000 and 1,500 Hz. The harmonics of the 5,000-Hz tone (c) are too high to hear easily. The complex tone (b) already has harmonics in the input. Therefore, if harmonics are heard in the output you learn nothing. The broadband noise (a) has components at all frequencies and so distorted noise sounds just like undistorted noise.

17.5: The fundamental frequency is 1,000 Hz.

17.6: Harmonic distortion products occur at 2,000, 3,000, 4,000, … Hz and 2,400, 3,600, 4,800, … Hz. Important difference tones are $f_2 - f_1 = 1200 - 1000 = 200$ Hz, $2f_1 - f_2 = 2000 - 1200 = 800$ Hz, and $3f_1 - 2f_2 = 3000 - 2400 = 600$ Hz. Important summation tones can be found by replacing the minus signs above with plus signs. Therefore, they include 2,200, 3,200, and 5,400 Hz. Physically, the summation tones have the same amplitudes as the corresponding difference tones, but the difference tones are much more audible. One reason for the greater prominence of the difference tones is that the difference tones have frequencies that are lower than the input sine tones and so they are not as efficiently masked by the input sine tones.

17.7: When frequency f_2 increases from 1,200 to 1,250 Hz, the difference tone $2f_1 - f_2$ decreases from 800 to 750 Hz. If this difference tone is prominent, it may even change a listener's perception from an upward pitch change to a downward change.

17.8: Assuming that the tolerable distortion is 0.1%, the dynamic range is $90-10 = 80$ dB.

17.9: Fundamentally, there is no difference between the 60-Hz electrical power from the outlet in the wall and a 60-Hz audio signal from a power amplifier. Both are called "alternating current," both have the same waveform (sine wave) and frequency. The 120-V power from the wall has an amplitude of $120\sqrt{2} = 177$ V. That is much higher the voltage used to drive loudspeakers in a home audio system, normally only a few volts. Therefore, a home audio system does not have enough power to run a toaster (about 1,000 W). On the other side, if you were to plug a standard loudspeaker into a wall outlet, you would be driving the loudspeaker at a rate of $(120)^2/8 = 1800$ W ... briefly ... because the loudspeaker coil would burn out very quickly.

17.10: (a) The tall peaks in the spectrum have frequencies that are integer multiples of 350 or 450 Hz. They occur at 700, 1,050 Hz, ... and 900, 1,350 Hz ... These are harmonics which tells you that there are two periodic waveforms here.
(b) It's a good bet that there are two physical horns, each one making a periodic wave.
(c) The frequency ratio is 450/350 or 1.286. Table D1 indicates that the closest musical interval is an equal-tempered major third with a ratio of 1.26.
(d) If we define tall peaks as peaks that are no smaller than 20 dB less than the -60 dB peak (i.e. 20 dB down) at 350 Hz (i.e. 20 dB down), then all the peaks are accounted for by assuming that they are harmonics.
(e) If we decide to include all the peaks higher than the 30-dB down point, then we see distortion components at frequencies (Hz) given by $2f_1 - f_2 = 250$, $f_1 + f_2 = 800$, $4_1 - f_2 = 950$, $3f_1 + 2f_2 = 1950$, $4f_1 + 2f_2 = 2300$, $3f_1 + 3f_2 = 2400$, $6f_1 + f_2 = 2550$. There are probably more difference tones too, but they are masked by low frequency noise—likely wind noise. Interestingly, all the smaller peaks can also be accounted for with combination tone formulas like those above but different integers, sometimes large integers.

Chapter 18

18.1: The amplifier consists of a source of voltage that may be positive or negative. This voltage initially comes from positive and negative power supplies, which are the most massive part of the amplifier. The positive and negative voltages from the supplies represent the largest magnitudes of voltage that the amplifier will ever be able to produce. What the output voltage actually is at any given time is determined by the input signal, which can have a voltage that is much smaller than the output voltage. The process of controlling a large output voltage with a small control signal is called amplification, and it is normally accomplished with transistor amplifiers or integrated circuits that contain transistors.

18.2: (a) $20\log(100) = 40$ dB. (b) $20\log(1000) = 60$ dB.

18.3: For an 11-band, octave-band equalizer, the center frequencies of the bands, starting at 20 Hz, would be 20, 40, 80, 160, 320, 640, 1,280, 2,560, 5,120, 10,240, 20,480 Hz.

18.4: A valve is a device that controls a flow. A water valve turns the water on and off and regulates its flow. It's natural to refer to a vacuum tube as a valve because it can control the flow of electrical current in the output circuit by means of a small input voltage.

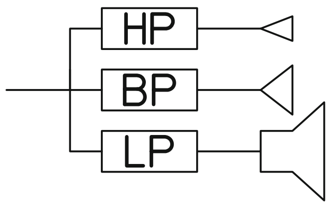
18.5: (a) The car radio has a tuner, preamplifier, tone controls, and power amplifier. It may have a CD drive. (b) The boom box is the same as the car radio but includes loudspeakers too. (c) A desk top computer has a CD drive and a sound card that includes a preamplifier for microphone input. A desk top computer normally is connected to speakers that include their own power amplifier. (d) A lap top computer has everything on the list except for a tuner and tone controls. (e) A camcorder has a microphone, preamplifier, a tiny power amplifier, and a tiny speaker. Digital cameras with audio are the same.

18.6: There is a school of audio recording that advocates *tight mikeing* whereby a large number of microphones are placed all around the orchestra to make close-up recordings of individual instruments or groups of instruments. Dozens of individual recordings feed separate channels of a mixer where they can be adjusted to taste and mixed with the signals from other microphones that have been placed to capture the ambiance of the environment. One can easily use 40 microphones and 40 channels.

Chapter 19

19.2: Choose a resonance frequency for the enclosure that is somewhat less than 60 Hz. By boosting frequencies where the driver is weak, you can extend the bass response of the system to lower frequencies.

19.3:



For Exercise 19.3. Three-way loudspeaker. The crossover has one input and three outputs From top to bottom, the drivers are tweeter, midrange, and woofer

19.4: Efficiency is output power divided by input power. For A: $1.5/20 = 7.5\,\%$. For B: $1/14 = 7.1\,\%$. Thus A is slightly more efficient, but for all practical purposes the two speakers have equivalent efficiency. [$10\log(7.5/7.1) = 0.2$ dB—an inaudible difference.]

19.5: To calculate the output of the speaker in dB SPL we imagine that we put 1 W into the speaker. Then, from the description in Exercise 4, we expect to get 1/14 acoustical watt out of the speaker. To calculate the level of sound produced we imagine that this power, 1/14 W, is distributed over the surface of a sphere that is 1 m in radius. We further assume homogeneous distribution. The area of a sphere with radius R is given by $4\pi R^2$. For a 1-m sphere that is $12.6\,\mathrm{m}^2$. A power of 1/14 W spread over a surface of $12.6\,\mathrm{m}^2$ is an intensity of $5.68\,\mathrm{mW/m}^2$. The threshold of hearing is $10^{-12}\,\mathrm{W/m}^2$ or $10^{-9}\,\mathrm{mW/m}^2$. Therefore, the level is

$$10\log(I/I_0) = 10\log(5.68/10^{-9}) = 97.5 \text{ dB SPL}$$

From a physical point of view this seems like a strange way to quote an efficiency, but it has obvious practical value to the speaker designer who wants to know how loud the speaker is going to sound.

19.6: Near the origin, where signal levels are low, the slope of part (b) appears to be about the same as the slope of part (a). Maybe (a) and (b) are really supposed to be the same device, but (b) is defective in some way.

19.7: At low frequencies the wavelength is long and the ratio of wavelength to driver cone diameter is large. That is the condition for good diffusion, and so you expect the frequency response for 45° to be similar to the response for 0°. Similar response at high frequencies would violate the rule for wave diffraction.

19.8: The plot shows that compared to the response at a mid frequency like 0.5 kHz, the response at the low frequency of 0.05 kHz has rolled off by about 6 dB.

Chapter 20

20.1: Obviously the two statements are equally accurate.

20.2: For a three-bit word, $N = 3$. From the formula we expect there to be eight possible values of the signal. These are:

$$000\ 001\ 010\ 011\ 100\ 101\ 110\ 111.$$

In a system with positive numbers only, these represent the numbers 0, 1, 2, 3, 4, 5, 6, and 7 respectively.

In the audio standard that uses negative numbers, these represent the numbers 0, 1, 2, 3, −4, −3, −2, and −1 respectively.

20.3: If there are 44,100 samples per second per channel and there are two channels in a stereo recording, then stereo requires 88,200 samples per second. If there are 16 bits in a sample, then a stereo recording requires 1,411,200 bits per second. If there are 8 bits in a byte, then you can find the number of bytes by dividing the number of bits by 8 or 176,400 bytes per second.

These values of the data rate apply to the compact disc format, which uses no data compression. Data compression techniques can greatly reduce the data rate and storage requirements, but information is lost in that process. In actuality, the compact disc format requires a higher data rate than we have calculated because this format includes redundant information that enables the playback system to recover from errors in reading the data. The error correction codes need to be added to the minimum set of data represented by 88,2000 samples per second.

20.4: If the sample rate is at least twice the highest frequency, then there must be at least two samples for every cycle.

20.5: The telephone uses only a single channel (no stereo telephone) and the sample rate needs to be twice the maximum frequency or 10,000 samples per second.

20.6: Digital electronic music synthesizers compute the musical waveform from a formula, as controlled by the musician. This music has no existence as an analog waveform until it is converted to an audio signal by a DAC.

20.7: Reasonable people can differ on this question, but an important argument for the analog vinyl recording is that the ET s will quickly recognize what it is and figure out how to play it back. By contrast, information in the compact disc format is highly encoded and is further complicated by error correction codes, redundancy checks, and added data outside the audio domain. Getting the information off a compact disc requires that the reader have knowledge of these arbitrary encoding conventions or somehow figure out what they are. If we want the ET s to be able to read the compact disc we had better hope that they are very clever.

20.8: Because loudspeakers and headphones receive an analog signal from a power amplifier, it is hard to see how the format of the original program material—digital or analog—could matter at all. “Digital ready” would seem inappropriate.

20.9: Unless the sampling of the signal is somehow synchronized with periodicity in the signal itself (an unlikely event) the errors made by quantization are randomly varying like noise. We normally reserve the term “distortion” to describe a systematic, time-independent deformation of a waveform which would be the same on each cycle of a periodic signal.

20.10: (a) $2^8 = 256$. Thus, 256 different values. (b) $20 \log(256) = 48$. Thus the signal to noise ratio is 48 dB. That is not nearly good enough for audio. Older technologies, such as vinyl disks and magnetic tape, have better signal to noise ratios than that $20.11 : 1 + 4 + 8 + 32 + 64 = 109$.

20.12: A byte for the number 85 is 0101 0101. It is $1 + 4 + 16 + 64$.

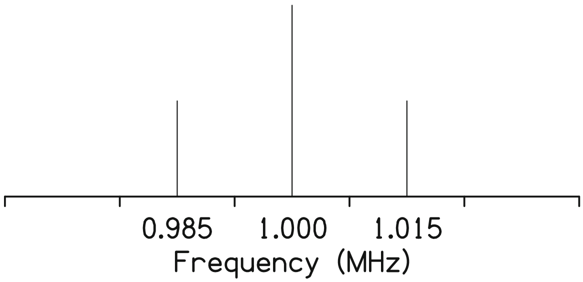
20.13: Commercial analog recordings of music were frequently copied, but each generation had more noise than the previous generation. This loss of quality in reproductions caused consumers to buy original recordings. Digital recordings do not suffer this sequential degradation, and that makes frequent copying more attractive. Nevertheless, sharing files among friends had little effect on the music industry. What really changed the game was the Internet which made it possible for millions of people to share the same digital music file. The battle between consumers and the industry raged throughout the first decade of the twenty-first century.

20.14(c): Digital technology cannot be acausal in the truest sense. However, if we make a continuous recording of a sound and are willing to delay the output of the processing by 10 ms, we can look ahead 10 ms in doing the processing.

Chapter 21

21.1: The wavelength is given by $\lambda = c/f$, where c is the speed of light. Therefore, for 1 MHz, (a) $\lambda = 3 \times 10^8/10^6 = 3 \times 10^2 = 300$ m. For a frequency of 10^{11} Hz, the wavelength is (b) $\lambda = 3 \times 10^8/10^{11} = 3 \times 10^{-3} = 0.003$ m.

21.2:



For Exercise 21.2

21.3: 6,000,000/30,000 = 200 times wider.

21.4(c): An AM receiver is sensitive to the amplitude of the signal and when noise is added to the signal the amplitude includes that noise. A FM receiver is sensitive to the phase of the signal, particularly the rate of change of that phase. It is not sensitive to changes in the amplitude of the signal. When noise is added to the signal it apparently has less effect on the phase than it does on the amplitude.

21.5: *Back row of the opera:* The time required for a sound wave to go 150 ft can be found from the speed of sound, 1.130 ft/ms. The delay to the back row is therefore $150/1.130 = 133$ ms.
Broadcast: Because the circumference of the earth is about 36,000 km, any two points on the surface of the earth are separated by no more than 18,000 km. A radio wave goes that far in $18,000/(3 \times 10^5) = 60$ ms. Adding at most 20 ms for the

acoustical paths from singer to microphone and radio to you leads to no more than 80 ms. Therefore, you hear the singer before the audience in the back row.

21.6: To transmit audio or video information via a radio-frequency carrier, the carrier must be modulated. When the carrier is modulated it acquires sidebands. Although a carrier by itself is only a single frequency, transmitting the sidebands requires a frequency *region* or band. No other transmission can use that frequency region without causing interference. To avoid chaos, the use of frequency bands must be regulated. In the USA, for example, radio bands are assigned by the Federal Communications Commission, part of the Department of Commerce. The greater the rate of transmitting information, the faster the modulation needs to be. High modulation frequencies lead to large bandwidth. Because bandwidth is a scarce commodity, information transmission itself becomes scarce, and economic factors apply. This rule is completely general. It does not matter if one is transmitting audio by AM radio or if one is transmitting digital data by modulating an optical carrier (light beam). The rule is the same: Information transfer requires bandwidth.

21.7: A lightning bolt produces a brief intense burst of electromagnetic radiation. This leads to static in radio reception. Because it is a pulse, static has a broad spectrum. Thus, the power is spread over a wide frequency range. The tuned circuit at the front end of a radio receiver eliminates all of that broad spectrum except for the narrow band that happens to coincide with the desired signal. But since the signal is deliberately put in that band, whereas the static is only accidently in that band, the signal normally dominates the static. But not always—AM broadcasts are frequently interrupted by static.

21.8: Yes, there is agreement. For example, channels 7–13 must fit into a band with a width of $216 - 174 = 42$ MHz. Because there are seven channels, 7, 8, 9, 10, 11, 12, and 13, they will fit if each of them is 42/7 or 6 MHz in width.

Chapter 22

22.2: Lung capacity is 1.32 gallons or 5.28 quarts.

22.3b: B and V are voiced plosives.

22.4: The nasals can be made with the lips tight shut, but a listener may not know which nasal is being pronounced. There is also a nasal aspiration, not on the list of phonemes, as in “hmmmm.” Although these closed-lips phonemes would seem to have minimal content, a lot of information can be transmitted by pitch changes during phonation.

22.5: In theory, all vowels might sound the same if played backward, but speakers of English tend to make diphthongs out of all vowels, more or less depending on regional accents. By their nature, these diphthongs sound different if played backward. For standard American speech (e.g. network TV announcers) the EE and the OO tend to be the same backward and forward.

22.6: The great thing about answering questions like this one is that you don't have to know anything new. Just make the sounds and report what you do. The sequence goes from front to back.

22.8: There is no reason to change. The spoken vowel and whispered vowel both have the same formants as determined by the articulatory structures—mouth, lips, tongue. The difference between speaking aloud and whispering is only in the different excitation source from the lower parts of the vocal tract.

22.9: Low fundamental frequencies and high formants occur for cartoon characters in films and TV ads. For instance, there's the baby who smokes cigars.

22.10: The vowels have the most energy and appear darkest on the spectrogram. They have formants—alternating dark and light bands as a function of frequency. Formant transitions appear as dark bands that slide from a higher frequency range to a lower frequency range or vice versa. Fricatives appear as noise, broadband, or high frequency. They do not show formant structure—there is only one main frequency band. Plosives are like fricatives but they are brief and don't occupy much time on the horizontal axis.

Chapter 23

23.1: The trumpet player calls the notes "C,G,C,E,G,C"—the notes of a bugle call. Their frequencies are: 233, 349, 466, 587, 698, and 932 Hz.

(a) The notes called C correspond to frequencies: 233, 466, and 932. Each one is twice as high (factor of 2) as the previous one. A factor of two in frequency is an octave.

(b) From C_4 to G_4 is a ratio $349/233 = 1.498$. From C_5 to G_5 is a ratio $698/466 = 1.498$. These intervals are close enough to $3/2 = 1.500$ to be good intervals of the fifth.

23.2 (c): For $G4$: Resonances 3 and 6. (Resonance 9 is on the high side.) For $C5$: Resonances 4 and 8. For $E5$: Resonances 5 and 10. For $G5$: Resonance 6. (The second harmonic falls between resonances 11 and 12.) For $C6$: Resonance 8.

23.3: The resonances of a bugle are at successive integer multiples (2, 3, 4, ...) of a (nonexistent) base frequency. A cylindrical pipe that is open at both ends also has resonance frequencies that are successive integer multiples of a base frequency (Chap. 8). In that sense, the two systems are similar. However, the bugle is not a cylindrical pipe. It gets its harmonically related resonances from its bell.

23.4: If we represent a closed valve by the symbol 1 and an open valve by a symbol 0 then the possible positions can be represented by the answer to the first part of Exercise 20.2, namely:

000 001 010 011 100 101 110 111.

The missing configuration is 001, where only the third piston is depressed.

23.5: A tube with a flaring bell and mouthpiece is really quite different from a cylindrical pipe open at one end and closed at the other. The frequencies of the basic cylindrical pipe are substantially altered when the bell is added. Therefore, we cannot answer this question by appealing to the formula $(v/4L) \cdot (2n + 1)$ for the resonances of a cylindrical pipe.

A better answer appeals to the idea of scaling. Suppose that a trombone were absolutely identical to a trumpet except that every dimension of the trumpet is multiplied by 2 to make a trombone. Then every resonance of the trombone would be the same as the resonance of the trumpet except that the trombone resonance would occur for a wavelength that is twice as long. Doubling the wavelength reduces the frequency by an octave, and that would cause the trombone to sound an octave lower. In fact, the trombone is close enough to a scaled trumpet that this scaling argument is valid. It may strike you as surprising that the trombone is a scaled trumpet. The two instruments don't look the same. The trombone has a slide and the trumpet has valves. Acoustically, however, both are cylindrical tubes with flaring bells on the end. Whether extra cylindrical tube is added by extending a slide or by operating a valve is acoustically less important.

23.6: First, refer back to Exercise 23.2. The trumpet player and trombone player use the same resonances of the instrument. The instruments sound different because the resonances of the trombone are an octave lower than the trumpet. By contrast the French horn player uses higher-numbered resonances. Even though the instrument may be as long as a trombone, the playing frequencies are not low like a trombone. Using the higher-numbered resonances leads to notes of higher frequency.

23.7: Dents in a beat up bugle are small and don't change the basic shape of the instrument. They do change the details. The low frequency modes, which are most important in determining the playing frequency, have long wavelengths and they are not much affected by the dents. Dents can mistune the high-frequency modes and cause them not to line up well with harmonics of the tones. When high-frequency modes do not support the upper harmonics of the tone, a brass instrument sounds dull or stuffy.

23.8: Bozo is wrong as usual. The player does not voluntarily move his lips at a rapid rate. He uses his lip muscles to set a favorable lip configuration inside the mouthpiece. After that, his lips vibrate rapidly under the influence of a more or less steady stream of air. Vibration caused by a steady stream is fairly common. Venetian blinds will rattle in a steady breeze. The nozzle of a balloon may vibrate as air comes out of the balloon. You can blow a steady stream of air through your mouth and trill your tongue. The brass instrument player's lips do the same kind of thing, except their motions are organized by the influence of the horn. Of course, setting a configuration that favors lip vibration is not trivial. When a beginner first tries a brass instrument, he is likely to get no sound out of it at all—no lip vibration.

23.9: The resonance curves, such as those in Fig. 23.6, come from the linear response of the instrument itself . . . no musician involved. To measure this response you arrange to have a sinusoidal oscillating flow of air into the instrument so that the amplitude of the flow (measured in cubic centimeters per second) is constant as the frequency of the oscillations changes. Then you measure the response, which is the amplitude of the pressure oscillations in the mouthpiece. That response looks like Fig. 23.6, which can fairly be called a resonance curve because it shows a ratio of output to input as a function of input (driving) frequency. To use a more technical term, you can say that you have measured the magnitude of the "input impedance" of the horn as a function of frequency. Having found this linear response, we can now add the musician. We imagine a musician with lips of steel who can maintain a steady flow of air into the instrument through a fixed opening between his lips. Then the pressure in the mouthpiece would be proportional to the flow. But the real musician does not have lips of steel. Instead, the opening depends on the pressure in the mouthpiece. In the end, the resulting pressure in the mouthpiece depends on the resulting pressure in the mouthpiece, and that is a nonlinear process. The nonlinear process causes the lips to buzz at a frequency and with a waveform determined by the feedback from the horn with its tuned resonances.

Chapter 24

24.1: (a) For a tube like this, the resonances are given by the formula $(v/4L) \cdot (2n + 1)$. (Recall Chap. 8.) The odd-numbered resonances create odd-numbered harmonics in the tone. The clarinet actually differs from a cylinder because it has a bell, and because it has holes bored into its wall. Even if the holes are closed, they represent irregularities in the wall surface. Also the mouthpiece is tapered, and that is another irregularity. These irregularities are responsible for the even-numbered harmonics in the higher register of the clarinet. For the low register the wavelengths of low-numbered harmonics are long enough that the irregularities are hardly noticed. Then the ideal cylindrical pipe, with odd-numbered resonances only, is a better approximation to the real clarinet.

(b) If $v/(4L) = 147$, where $v = 344$ m/s, then $L = 344/(4{\cdot}147)$, and $L = 0.585$ m.

24.2: Consider the pressure waves for the first and second modes of a cylindrical pipe, open at one end and closed at the other, as shown in Chap. 8. One third of the way from the closed end the second mode has a node. If a hole is opened at this point in the pipe, nothing is changed so far as the second mode is concerned. The standing wave pattern for this mode has ordinary atmospheric pressure at that point whether the hole is there or not. Now consider the first mode. The first mode requires considerable pressure variation at the point that is one-third of the distance from the closed end. The first mode cannot survive with a hole at that point. Therefore, the register hole kills off the first mode while not affecting the second mode (frequency of three times the first).

24.3: For an open–open pipe the frequency of the fundamental is $v/2L$. For an open–closed pipe the frequency of the fundamental is $v/4L$. There's the octave.

24.4: From the study of Fourier analysis we know that waveforms that change abruptly in time have high-frequency components. Just imagine all the high frequencies it takes to synthesize a waveform that has a sharp corner. The abrupt closing of a double reed leads to a waveform with sharp corners. The relatively intense harmonics with high-harmonic number lead to a bright sound. Renaissance double reed instruments like the krumhorn make such an impulsive waveform that the high harmonics are very numerous and very intense. As a result such instruments are more than bright—they are buzzy.

24.5: (a) The free reeds and the vocal folds both vibrate at frequencies determined only by themselves, without any feedback from a resonator. The saxophone reed vibrates at a frequency that is largely determined by feedback from the horn. (b) In a double-reed instrument like the oboe, the two reeds vibrate against one another—like the two halves of the vocal folds.

24.6: For a pipe open at both ends the playing frequency should be $f = v/(2L)$. For the flute, $f = 344/1.2 = 287$ Hz. From Appendix E, you find that this frequency is between $C\sharp_4$ and D_4. In fact, the head joint—a tapered extension of the flute—adds some effective length, and the lowest note on the flute is actually C_4 with a frequency of 262 Hz.

24.8: Hints: Use $v = 34,400$ cm/s, and $V = (4/3)\pi r^3$, where r is the radius of the sphere. Use the approximation for the tone hole radius $a = r/2$, and remember that diameter $D$is twice r. Then turn the algebraic crank on Eq. (24.2) for the neckless Helmholtz resonator.

24.9: Remember what you learned in Chap. 13 about the pitch of a periodic tone with a missing fundamental.

24.10: (a) The fundamental appears to be about 262 Hz. (You can get good accuracy by finding the 10th harmonic and dividing its frequency by 10.) Appendix E shows that the closest note is C_4 (261.6 Hz), called "middle C." (b) One can see 19 harmonics out to 5,000 Hz. (c) The spectral components seem to be regularly spaced as expected for a harmonic spectrum, and so one expects the tone to be periodic (period = 1/262 s). (d) The second harmonic seems to be about 6 dB higher than the fundamental. For harmonica tones, the second harmonic is always strong, as expected for a free-reed instrument, but it is not always stronger than the fundamental. (e) The 12th harmonics seems to be 3 or 4 dB higher than the fundamental. That makes the harmonica a highly unusual instrument. For most instruments in their melody-range, harmonics as high as the 12th are much weaker than the fundamental. The strong high harmonics for the harmonica lead to the characteristic reedy sound.

Chapter 25

25.1: From the formula $f = v/(2L)$ we find that the speed of sound is $v = 2Lf = 2 \cdot 346 \cdot 440 = 304{,}480$ mm/s or about 304 m/s.

25.2: The lengths are 346, 326.6, 308.2, 290.9, 274.6, 259.2, 244.6, and 230.9 mm. The successive differences are 19.4, 18.4, 17.3, 16.3, 15.4, 14.6, and 13.7 mm. These differences are similar but not the same.

Note: The frequencies of the scale do not grow linearly—they grow geometrically with a common ratio. The frequencies of the string do not vary linearly with string length—they vary in a reciprocal way. Both these nonlinear functions produce greater frequency differences for greater frequencies.

25.3: For both the violin and the voice, there is a source and a resonator. The source vibration pattern and frequency are very little affected by the resonator. Source and resonator are largely independent in their operations. The role of the resonator is to filter and radiate the sound. The resonator has many peaks and valleys in its response function, which is responsible for the tone color. When a tone is played or sung with vibrato (periodic frequency modulation), the resonances lead to a frequency-modulation-induced amplitude modulation of the harmonics. This relationship between source and resonator is very different from a wind instrument, either brass or woodwind, where the resonator provides feedback that helps determine the vibrating frequency of the source.

25.4: Your graph should show about five or six cycles of a sine wave with an amplitude of about 12 Hz, hence extending from 380 to 404 Hz.

25.5: The acoustic guitar needs to radiate energy so that it can be heard. The bridge and top plate are designed to radiate efficiently. As the energy is radiated the string vibration is damped. The electric guitar does not need to radiate energy. The transducer that produces the electrical signal draws almost no energy from the string. The body of an electric guitar is solid and not designed to vibrate or radiate. Therefore, the string on an electric guitar is not damped by radiation.

25.6: It is true that the playing frequency depends on the timing of the stick–slip process, but the timing of the stick and slip depends on the motion of the kink along the string. The timing of the kink motion depends on the length of the string and the speed of sound on the string, not on the stickiness of the bow.

Chapter 26

26.1: $f_n/f_1 = [(2n+1)/3.011]^2$. Therefore, $f_2/f_1 = 2.76$, $f_3/f_1 = 5.40$, and $f_4/f_1 = 8.93$.

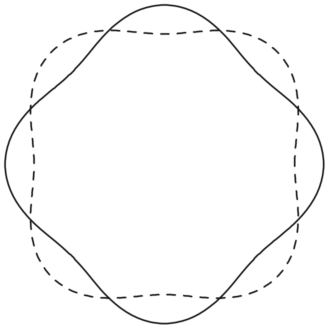
26.2: Doing the squares leads to 81, 121, and 169. Forming the ratios leads to $81/81 = 1$, $121/81 = 1.494$, and $169/81 = 2.086$. The relative ratios are not changed by multiplying by 2 to get 2, 2.988, and 4.172. The listener interprets these spectral components as 2, 3, and 4 times a (nonexistent) fundamental frequency.

26.3: Adding a second drum head leads to a greater number of different frequencies in the radiation, essentially twice as many. The denser spectrum of unrelated components leads to a sound that is less tonal.

26.4: As your mother often told you, dinner plates of all kinds make bad musical instruments. Paper plates are particularly unsuccessful because of internal damping of vibrational waves in the plate material. A paper plate is soft, and vibrations in soft material are quickly converted into heat.

26.5: The modes are: [4,1], [4,1], [5,1], and [6,1].

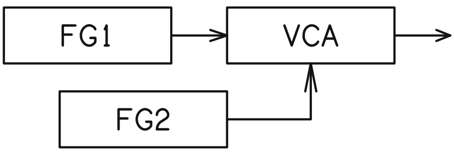
26.6: The figure shows the standing waves on the rim of the bell for a mode with four meridian lines, like the "octave."



For Exercise 26.6

Chapter 27

27.1: A patch to create an amplitude-modulated signal is made from two sine function generators and a voltage controlled amplifier:



For Exercise 27.1

The control voltage to the voltage-controlled amplifier must be positive. Therefore, the output of function generator "2" must be offset so that it does not go negative.

27.2: $100,000,000/44,100 = 2267.6$. Hence, 2,267 operations, even without any parallel processing.

27.4: Pitch and pitch variation are essential elements in music expression. Guitarists bend notes and violinists play with vibrato. It's not surprising that synthesis techniques pay a lot of attention to pitch. (a) The depth of vibrato often increases as a tone is sustained. The modulation controller gives a performer the control to do that. (b) Without bending the pitch of notes, you cannot sing the blues ... or anything else.

27.5: A computer naturally handles data. A computer makes a great sequencer because programs can be written to display control data as musical notes and to edit the data. With a special interface, the computer can put out data at precise times. Computers do not excel in making waveforms. The sound cards inside personal computers are always a compromise.

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