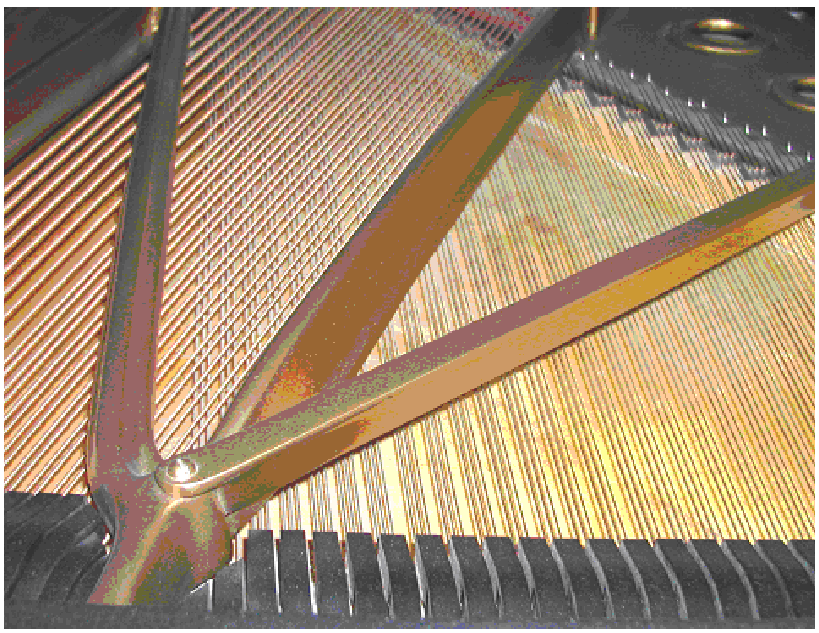
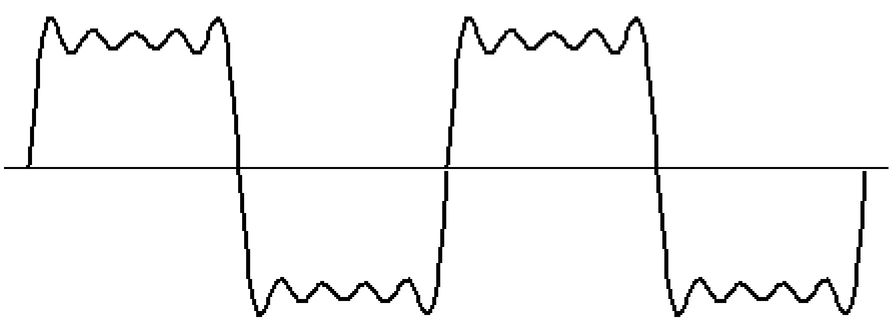
The Physics of Sound and Music

Fourth Edition (2012)

Michael J. Ruiz, Ph.D.

University of North Carolina at Asheville





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Forward

The Physics of Sound and Music is written for a general-education science course at the *University of North Carolina Asheville* (*UNCA*). There are no prerequisites for this course. If you know how to balance your checkbook (most of the time), you know enough mathematics to understand this text.

PHYS 102 *The Physics of Sound and Music* has been taught at *UNCA* since the early 1980s. It was developed by the author for the general student not majoring in science. However, any student may take the course. Originally the course was patterned closely after a similar one developed at the *University of Maryland* by Richard E. Berg in the 1970s. The author worked as a teaching assistant for Prof. Berg at that time. Since then, the course at *UNCA* has evolved into a course of its own.

In the 1970s, music synthesizers were very expensive. A company, *PAIA Electronics*, began making synthesizer kits. *UNCA* acquired one of these around 1980. This old synthesizer was never really suitable for performance; however, it served for many years as a source for excellent demonstrations to illustrate sound.

The course now features applets that illustrate the physics of sound synthesis in remarkable ways. The added advantage here is that you get to play with the "toys" from home!

The *UNCA* lectures that developed around the modular synthesizer have been published in *The Physics Teacher*. The synthesizer takes a central position in our course. Legendary Bob Moog (1934-2005), one of the independent inventors of the synthesizer lived in Asheville for the last quarter century of his life. He had many interactions with Music at UNC Asheville and one year he taught a physics course.

The *Physics of Sound and Music* and its sister course, PHYS 101 *Light and Visual Phenomena*, are two of the most popular science classes at *UNCA*. Sound was written up in the school newspaper in 1987 ("Sounds of Music," *The Blue Banner*, December 10, 1987, p. 3) and Light was featured on WLOS, *Evening News*, in 1989.

Then in Fall 2002, the uses of computer technology in these courses and our astronomy course were featured on *CNN* with anchor Ann Kellan We made top story in technology over the August. 31, 2002 weekend. A 5-minute piece was aired throughout the world on the program *NEXT@CNN*. You can view the video at our website.

You will develop through this text and the supplemental material at our website an understanding of sound and its applications in our daily lives. You will also acquire an appreciation of the beauty of science and an integrative understanding of science across many disciplines.

The author is a theoretical physicist (elementary particles) with a Ph.D. from the *University of Maryland* in 1978. He is also a musician. He studied classical piano under Stewart Gordon and jazz piano under Ron Elliston, at the *University of Maryland* in the 1970s. He has composed three piano concertos, one for each of his children. All three children have performed these concertos with the *Winston-Salem Symphony*.

The author is also a software developer. He and his son Evan designed a unique course web management system for this course. You will need regular access to the internet in order to experience this state-of-the-art learning adventure. All your assignments are done online.

The author received the *1995 UNCA Distinguished Teacher Award*, the *1997 UNCA Distinguished Teacher Award in Natural Sciences*, and the *2004 Board of Governors Teaching Award* He is a former Chair of the *UNCA Department of Physics* (1980-2000).

Revised Summer 2012, Asheville, NC

Chapters

A. What is Sound?

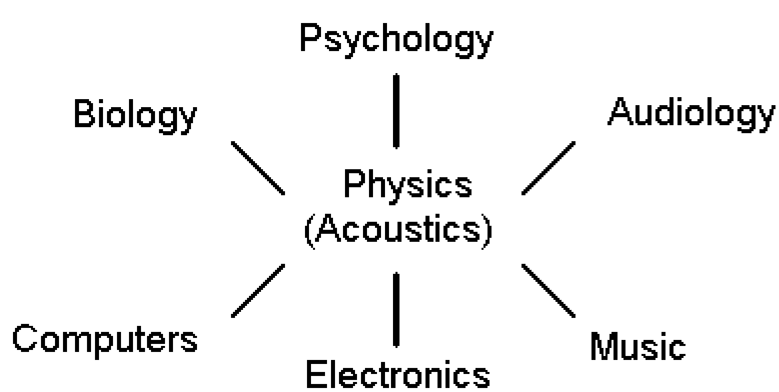
Physics, of course, is considered by many to be inaccessible due to its emphasis on mathematics. This text counteracts by presenting physics within the context of a broad cultural background, using concepts, diagrams, and tables instead of obscure math. We will illustrate this liberal-arts approach with our reflection on the inquiry "What is Sound?" This question draws many responses, depending on your point of view. To a musician, the answer includes pitches, harmony, and beauty. Musical tones serve as a palette for the artist to compose.

A building inspector often looks at sound in terms of its loudness or sound level. Our campus physical plant manager has a meter to check if sound levels are within specifications for office areas, classrooms, and hallways. A biologist studies living organisms and examines the structure of the ear, our detector of sound.

Psychologists investigating perception are interested in how the brain perceives sound. A physicist pays attention to the physical manifestations of sound such as the vibration of air. A philosopher may ponder the interesting question "Is there sound if a tree falls in a forest and no one is present to hear it?"

This text presents sound in both its physical context and its connection to a variety of disciplines. We can place any subject such as music or biology in the center of a diagram and show the connections to other fields. We will place physics in the center since this text uses physics as a foundation in our study of sound and its applications. Physics is also the fundamental discipline that deals with basic physical quantities in our world. It is a good place to start in our study. It will pave the way for our understanding of sound.

Fig. A-1. Major Fields Dealing with Sound.



In Fig. A-1 above, the major fields that involve sound are listed. Physics deals with the fundamental physical world and is concerned with basic aspects such as matter, energy, and their relationships. The subfield of physics that focuses on sound is acoustics. However, acoustics usually means room acoustics so we will stick with physics. The three related areas in the upper part of Fig. A-1, biology, psychology, and audiology, include human detection and perception of sound. Audiology deals with hearing loss and methods to evaluate such loss. Reading clockwise, the three fields in the lower part of the figure are music, electronics, and computers. An important ingredient of these fields is the production of sound. Physics describes sound by looking at physical characteristics. These are features that can be measured and assigned numbers. You should not feel intimidated about the use of numbers. You use numerical values to describe your own physical features. A number describes your height, another your age. We begin our study of sound in the realm of such descriptions.

Since sound interacts with the ear, we will turn to the study of biology at some point. Sound is perceived by the brain, which brings in the psychology of perception. The perception of pleasant harmonies takes us into music. You will learn about consonance, the pleasant combination of sounds, and dissonance, unpleasant sounds. However, these are to some extent subjective. Modern music often breaks away from traditional harmonies and introduces more dissonance.

In recent generations and today, we see the marvelous reproduction of sound with records, tape decks, and CDs. In the 1950s, with the improvement of records, the description of excellent sound reproduction was often called *hifi* (a word coined in 1950), meaning *high fidelity* (1934). One would often refer to one's home sound system as the hifi, and later as *the stereo* for the overall description of a sound system.

The reproduction of sound requires the use of electronics. We will a little basic electronics without detailed math. You will learn how to read simple modular synthesizer block diagrams. Once again, physics will be important to our understanding. There are only a very few fundamental laws of physics, from which all of these great technological innovations come. You will find that symbols and neat diagrams will be very powerful in describing electronic inventions such as the speaker, microphone, record player, and tape deck.

The sociologist Alvin Toffler (1980) wrote in his book *The Third Wave* that there have been three major technological revolutions throughout history. These are the *Agricultural Revolution* (beginning about 10,000 years ago), the *Industrial Revolution* (beginning in the 1700s), and a *Third Revolution* that began in the middle of the 20th century. This third wave of change involves information: the computer, microelectronics, genetic engineering, etc.

A common thread of third-wave technology is the application of microscopic science. We can correlate the three major historical waves of technological change with the development of sound production (Table A-1).

Table A-1. Historical Overview of Sound Production.

1. Voice	Speech, Song
2. Musical Instruments	Strings, Pipes, and Membranes
3. Synthesizers	Electronics and Computers

In agricultural settings we find speech and song. We also find simple instruments. Traditional orchestral instruments reach a peak in the industrial age that began to flourish during the 1800s. Here we have the big orchestras and concert halls.

Musical instruments use strings, pipes, and membranes in their constructions. In the early 1960s, the music synthesizer was invented. One of the key inventors was Bob Moog, who spent most of the last quarter century of his life in Asheville, NC. This is part of the third-wave of musical sound production. Then, in the 1980s, a standard was set that allows for the computer to control music synthesizers. This standard is called *MIDI*, which stands for *Musical Instrument Digital Interface*.

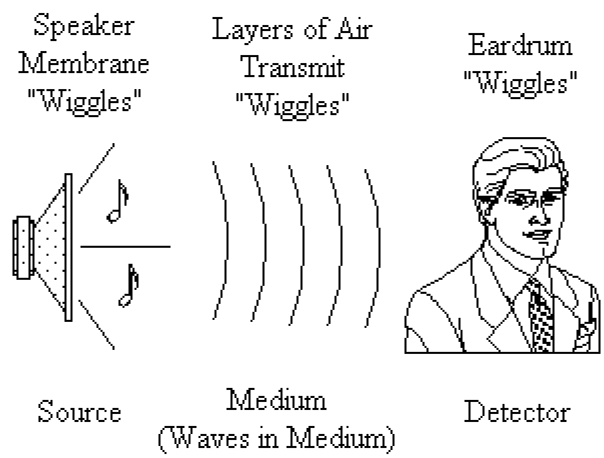
Sound as Vibration

When we hear anything, our eardrums vibrate back and forth very rapidly. It is possible to bypass the eardrum by allowing vibrations to pass through our bones, but these are still vibrations. Thinking of a vibrating eardrum gives us insight into the physical properties of sound. The eardrum responds to vibrations in the air.

The air vibrates in the first place due to a source of sound. Energy at the source is used to make the vibrations, which then travel through the air to reach our ears.

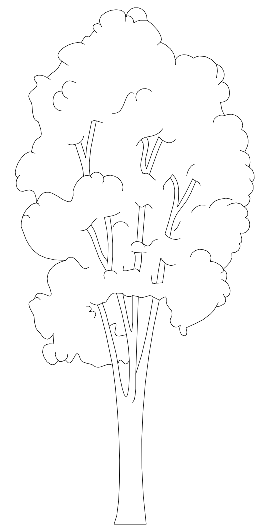
We can summarize this by saying that there is a source or speaker making vibrations, a medium such as air through which the vibrations travel, and a detector or hearer that receives the vibrations. Figure A-2 provides us with this summary.

Fig. A-2. Source, Medium, and Detector.



We can now revisit the question "Is there sound if a tree falls in a forest and no one hears it?" This question came up in the July 17, 1995 issue of *Time* magazine, where the cover story was on the mind and consciousness. The comments of a neuroscientist working at New York University Medical School, Dr. Rodolfo Llinás, are as follows.

"Light is nothing but electromagnetic radiation. Colors clearly don't exist outside our brains, nor does sound. Is there sound if a tree drops in the forest and no one hears it? No. Sound is the relationship between external vibrations and the brain. If there is no brain, there can be no sound." Rodolfo Llinás, MD, New York University Medical School, quoted in "Glimpses of the Mind" by Michael D. Lemonick, *Time*, July 17, 1995, p. 44.



The good doctor's analysis drew the following strong negative response from a reader.

"You quote neuroscientist Dr. Rodolfo Llinás as saying colors and sound don't exist outside our brains, concluding that if a brain doesn't perceive color and sound, then they don't exist. He was using the famous metaphor of a tree falling in the woods with no one around to hear it. I couldn't disagree more with Llinás' conclusion. Light is the energy given off by a heated or excited object in the form of photons.

"Sound is the vibration of molecules in a medium caused by an object. Just because there are no receivers around to pick up the light and sound does not mean they don't exist. When a tree falls in the woods, it hits the ground and vibrates the air and ground. That vibration of the air molecules is, by definition, sound. There is sound present, just no receptors to hear it. If I cannot hear my favorite radio station's broadcast, it doesn't mean that its radio waves don't exist; it just means that my radio is off." Greg Serrano, Lansing, Michigan, Via E-mail. Letters to the Editor, *Time*, August 7, 1995, p. 7.

The way out of this dilemma is to note that if sound is defined as "the relationship between external vibrations and the brain," then you obviously need a brain for the sound to be present. On the other hand, if sound is defined solely as "vibration of the air molecules," then you do not need a brain.

Consider a strictly perceptual definition of sound, i.e., if you hear the sound in your head, sound has been experienced. However, there may have not been an external object making the sound. The

Table A-2. Different Viewpoints on the Existence of Sound.

A Tree Falls in the Forest, But No One is There to Hear It. Is there sound?

Physically	Perceptually	Both
Yes	No	No

A Hearer Hallucinates a Tree Falling, But No Tree is There. Is there sound?

Physically	Perceptually	Both
No	Yes	No

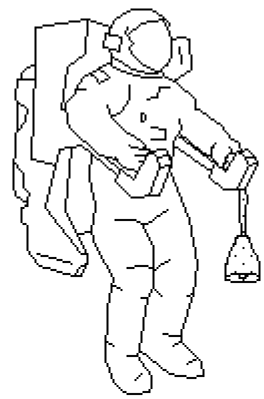
The Medium

According to the viewpoint of physics, we have sound as long as we have a source of sound and a medium to transport these vibrations. We do not need a hearer. However, the medium is important. There would be no sound coming from an astronaut shaking a hand bell in outer space. See Fig. A-3. The structure of the bell would vibrate, but the vibrations would have no way of leaving the bell.

Fig. A-3. Astronaut Shaking a Hand Bell in Outer Space.

strictly physical definition calls for external vibrations only. If the tree falls, there are physical vibrations in the air. So there is sound, whether someone hears it or not.

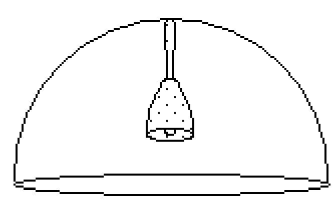
A third definition of sound includes both external vibrations and a perception of these vibrations. Table A-2 lists two questions and answers from these three different viewpoints. Which of the pairs of answers below would our medical doctor agree with? Reread his quote on the previous page if necessary.



No sound vibrations would be able to travel through the emptiness of outer space. However, we would be able to see the astronaut and the bell. Light can travel through the near vacuum of outer space.

The laboratory version of shaking a bell in outer space involves pumping the air out of a container. We do not hear sound from the bell. This demonstration is often referred to as the bell-jar demonstration (see Fig. A-4).

Fig. A-4. Hand Bell in Vacuum Jar.



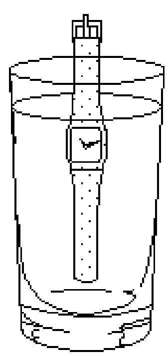
For good results, the bell must hang from the glass in such a way that sound vibrations are not permitted to pass directly from the bell to the glass. Also, a vacuum must be nearly achieved; otherwise, a faint sound will be heard.

The bell inside the glass jar can be seen even when the air is pumped out. We expect this result since light can travel through vacuum. That's how we get light from the sun. It is also how we can see the moon and the stars at night.

In conclusion, sound needs a medium in which to travel. The vibrations from a source of sound vibrate the surrounding medium and the sound travels through this medium. The medium is usually air. However, the medium can be a different gas. The medium can also be a solid. You can usually hear sound from another room by placing your ear up against a wall.

Sound also travels under water. A simple experiment to demonstrate that sound travels in water is shown in Fig. A-5 where a watch alarm is heard through a glass of water.

Fig. A-5. Watch Alarm in Glass of Water.



The Speed of Sound

Sound travels fast, but not so fast that we can't easily get a handle on its speed. If you ever heard an echo, you have experienced the finite speed of sound. Suppose you shout and a short time later hear your echo. The sound has traveled from your mouth, through the air, bounced off some large object, and returned to you. The speed of sound can be determined if you know the distance the sound traveled and the time.

You can determine the speed of a car going to Raleigh if you are given the information that the distance from Asheville to Raleigh is 250 miles and the trip takes 5 hours. The answer is 250 miles per 5 hours, or simply 50 miles per hour. This is an average time because the car does not maintain exactly 50 miles per hour at every moment. That's because you are driving and need to stop occasionally. You also need to vary your speed in traffic. Sound doesn't have to worry about such things. If the air is calm with the same temperature and pressure everywhere, sound travels at a faithful speed.

In science, we do experiments to see how the world works. So far, we have seen an experiment with a bell jar to examine if sound needs a medium to travel. We saw another experiment with a watch in water to determine if sound can travel in water. You may have had a direct experience of hearing sound underwater. If so, this counts as an experimental observation. You should have a distrust of information, even that given in this text, unless it can be supported by an observation, demonstration, or experiment.

This is a healthy scientific attitude. Science employs models that describe the real world. These models are based on observation. The scientific method includes hypotheses (which may or may not be based on the currently accepted model) and experiments to test these hypotheses. Science then consists of two components: theory and experiment. Each must support the other. If not, the theory or model is discarded for a better one. A theory that has withstood countless experiments can be considered a law. However, history has shown us that even the best theories or laws need to be modified from time to time. We now proceed to an experiment in order to measure the speed of sound.

Speed is defined as how far you go per some time interval like an hour or second. For example, 60 miles per hour, which we usually write as 60 mi/h in physics rather than 60 mph, is also 1 mi/min (i.e., 1 mile per minute). Whenever you see the symbol "/," just read it as "per." Abbreviations in science do not have periods, except for inches (in.).

Once we agree on our definition for speed, we can apply it to sound. The "echo technique" to measure the speed of sound requires a source of sound and an obstacle to reflect from. Our scientific instrumentation includes a measuring device to measure the distance and a watch to measure the time.

UNCA Convocation, an event held at the start of the academic year since the mid 1980s presents an excellent opportunity to measure the speed of sound when they hold it outdoors in front of the library.

Convocation is a ceremony to show incoming students that *UNCA* is a serious institution of higher learning. The faculty often dress in academic regalia, there is a procession, and speeches follow. As the decked-out faculty march, students are often dressed in T-shirts and shorts.

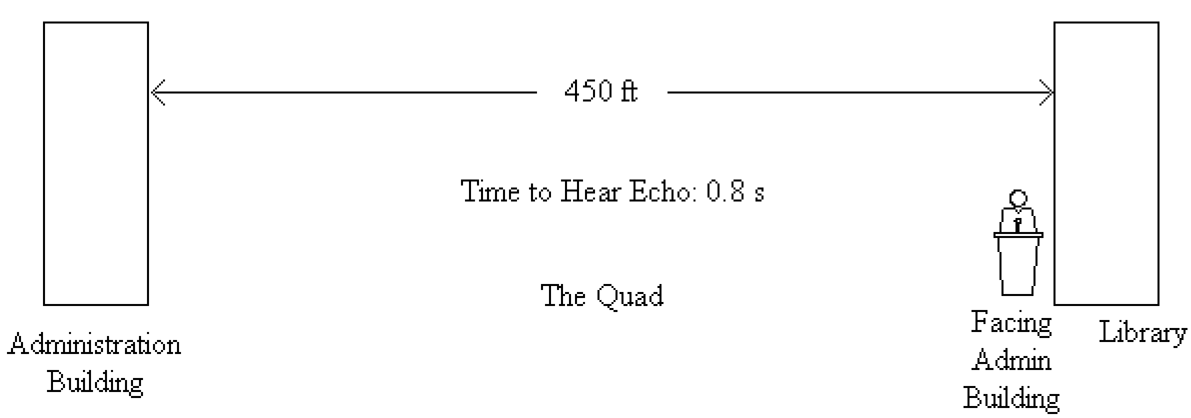
During the first year, some played Frisbee, creating a surrealistic setting. Also, someone in the distance had a radio playing rock music in competition with the serious processional music.

In the 1980s, the faculty often sat up near the library doors, like they do at graduation. If the public address sound system is positioned correctly near the steps of *Ramsey Library*, you get excellent reflections off the *Administration Building* across the *Quad*.

One year in the 1980s, the author was getting bored with the speeches and wanted to amuse himself. He noticed that during the Chancellor's speech, a faint echo could be heard. He used his digital watch to measure the time between the Chancellor's words and the associated echoes. With a little practice, the echo time was measured at 0.8 s, i.e., 4/5 of a second. The author's watch has a chronograph that measures 100ths of a second. However, due to human response time and uncertainties in distances, rounding off to the nearest 10th of a second is appropriate. The author was grateful to get such good data in the 1980s, because now, *Convocation* is usually elsewhere and the current Chancellor is never boring.

If you would like to do a similar experiment, you may be able to find a place where a delayed echo occurs. If not, you can always try it at graduation during the commencement address. Fig. A-6 gives a layout of the experiment. The distance from the *Administration Building* to the *Library* is about 450 ft (i.e., 450 feet).

Fig. A-6. Measuring the Speed of Sound.



Sound must travel from the *Library* to the *Administration Building* and back again before we can hear the echo return to the *Library*. The total distance traveled is therefore 2 x 450 ft = 900 ft. Therefore sound travels 900 ft in 0.8 s. We know immediately that the answer is approximately 900 ft/s (900 feet per second) or about 1000 ft/s since 0.8 s is nearly one second.

The actual result is worked out below using 0.8 s. The trick in dealing with the decimal is just to note that if you go 900 ft in 0.8 s, you go 9000 ft in 8 s (which is 10 times longer). We then divide to get the result per second. We round off at the end because our data is just not that accurate. The conclusion is that the speed of sound is 1100 ft/s.

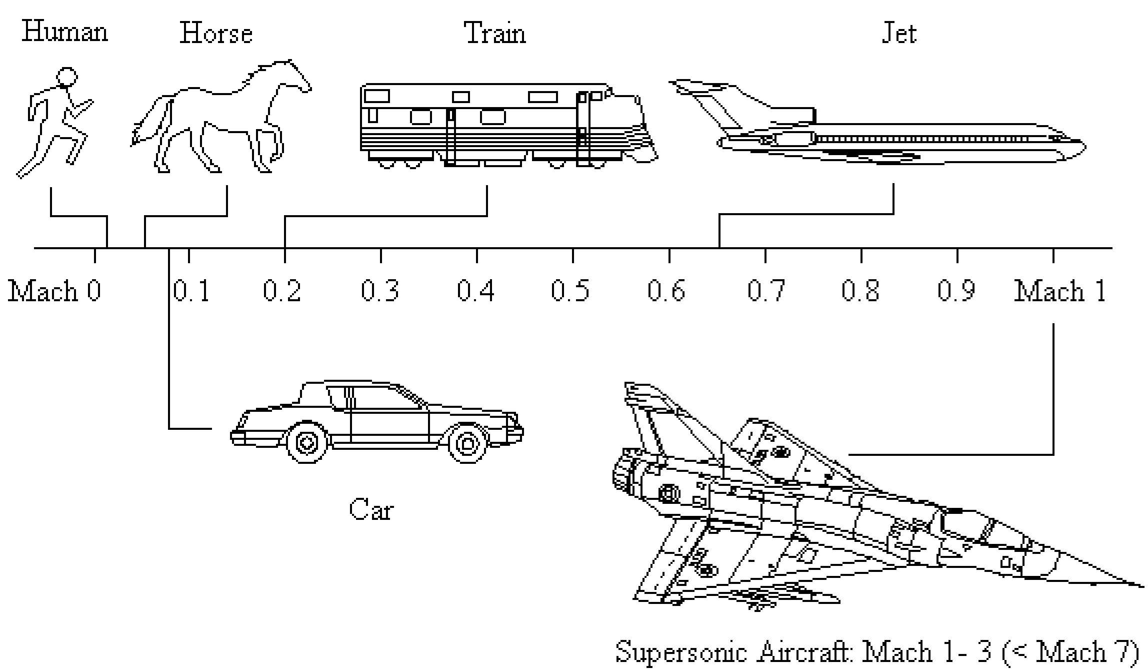
$$\text{Speed of sound} = \frac{\text{distance}}{\text{time}} = \frac{900\text{ ft}}{0.8\text{ s}} = \frac{9000}{8} = 1125\text{ ft/s} = 1100\text{ ft/s} \quad \text{(rounding off).}$$

Or reduce by halving numerator and denominator again and again: 9000/8 = 4500/4 = 2250/2 = 1125. Think money: half of $9000 is $4500 and half of $4500 is $2250. Your mind will work faster.

Mach Speed

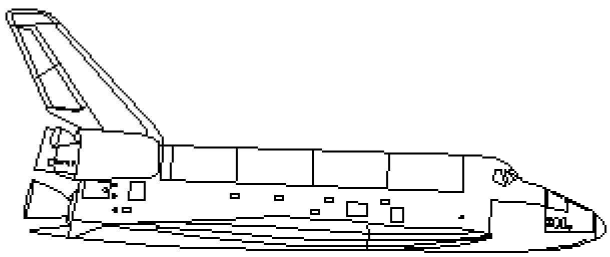
In Fig. A-7 we compare modes of transportation with the speed of sound. The Mach scale is used, where the speed of sound is Mach 1. Mach 0.5 indicates 1/2 sound speed (i.e., 50%).

Fig. A-7. Modes of Transportation and Sound Speed.

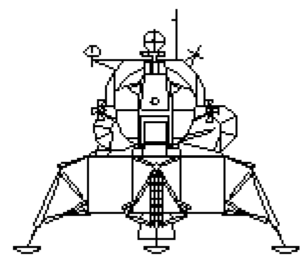


The Mach speed is defined in terms of the speed of sound at a given altitude. The speed of sound depends on the temperature and pressure of the air, which varies with altitude. For example, the speed of sound at freezing temperature is 3% lower than that at room temperature. We can loosely define "Mach 1" as the speed of sound at room temperature and pressure, ignoring this technicality. This is often done today when one refers to the speed of the space shuttle. Strictly speaking, since sound doesn't travel in outer space, Mach speed cannot be applied there. However, if we take "Mach 1" to be the fixed value of 1100 ft/s, then we can assign speeds in outer space a Mach value (refer to Fig. A-8).

Fig. A-8. "Mach" Values Assigned to Speeds in Outer Space.



Space Shuttle: "Mach 20" (17,000 mi/h)



Lunar Visit: "Mach 30" (25,000 mi/h)

Exceeding the Speed of Sound

From the past section we know that the speed of sound can be exceeded. However, in the 1930s and 1940s there were debates as to whether an aircraft could travel faster than the speed of sound. Engineers anticipated instabilities. One began to talk about the sound barrier due to the difficulty in moving faster than the vibrations you generate. We will look at a simple model for this shortly.

Sound speed is fast, 1100 ft/s. This is also about 750 mi/h. In the metric system, sound speed is 340 m/s (340 meters per second). The metric system is the official system of units used by the scientific community throughout the world. Most of your work in this course will involve the metric system.

Think of a meter as being slightly longer than a yard. Sound speed is therefore roughly 340 yard/s, or approximately 3 football fields per second. Imagine how long it takes to run across a football field. Now imagine covering 3 football fields in one second. Or imagine going from the *Library*, across the *Quad* to the *Administration Building*, and back to the Library, in about one second. If you can imagine this, you have a good feeling for the actual speed of sound.

Humans exceeded the speed of sound in the late 1940s. Today, we are accustomed to hearing about supersonic aircraft. The word supersonic refers to speeds faster than the speed of sound. The word entered our vocabulary in the 1920s. Supersonic can also refer to pitches that are too high for us to hear. So the word refers to two different concepts.

The "sound barrier" was broken on October 14, 1947 by Captain Charles E. Yeager of the US Air Force in the X-1. This experimental rocket plane was dropped out of a larger aircraft (a B-29) in flight. The designation X referred to experimental aircraft. During the next decade, breaking the sound barrier became commonplace in the X-series of research aircraft. The last aircraft of this type was the X-15, which was introduced in 1959. It was dropped out of a B-52 Bomber. One X-15 reached an incredible Mach 6.7 during the 1960s.

As early as 1962 Great Britain and France began plans for a supersonic commercial aircraft. The year 1958 saw the first use of commercial jets (US), and in 1970 the wide Boeing 747 jet began flying (US). The US was also developing a supersonic commercial jet, but in 1971, Congress cut off funding. Concerns included noise and air pollution, and danger to the ozone layer. The ozone layer protects us from harmful ultraviolet light.

The Russians developed a supersonic transport (SST) which began carrying passengers in 1975. It was discontinued in a few years. England and France began operation of the Concorde in 1976, which continued until it was retired in 2003.

The Concorde cruised at about Mach 2 at 18 km (about 12 miles or 60,000 ft). At this speed, it crossed the Atlantic in little over 2 hours. The Concorde flew into the Asheville Airport once.

Since more and more people need to fly long distances, in the early 1990s the US once again began working on SST projects. But this was discontinued in 2000 due to the expense involved.

The engineers had been working to lower noise pollution and reduce the risk to our ozone layer. However, there is no way to prevent the "sonic boom." When the sound barrier is broken, a loud boom is generated. Observers on the ground hear it once as the aircraft passes overhead. Echoes of the boom can be heard if mountains are nearby. We turn now to this very interesting phenomenon.

The Sonic Boom

In order to understand the "sonic boom" we look for a model or an analogy. What is similar to an aircraft generating vibrations in a medium? One answer is a boat generating disturbances in water. These disturbances are waves: the water surface moves up and down as waves travel. Although water waves have some marked differences with sound vibrations in air, the analogy can give us a general idea of what is going on. Later we will learn the difference between water waves and sound waves.

Fig. A-9 illustrates a motorboat traveling faster than the waves. The boat drags the waves with it in a "V" formation. This "V" represents a large wave crest since the dragging waves build up. This "V" becomes narrower if the boat goes even faster. Someone in the water will experience a "jolt" as the large wave passes by. This "water jolt" is analogous to our sonic boom. Note that an observer treading water experiences this "jolt" once, as the "V" passes by.

Fig. A-9. Motor Boat Exceeding the Speed of Water Waves, Dragging the Waves Behind It.

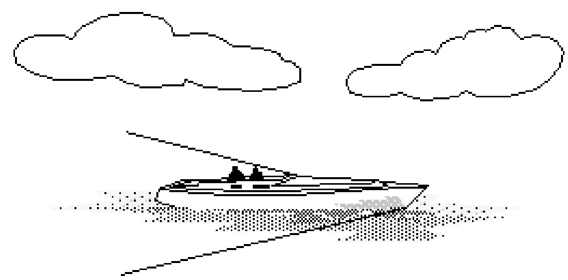
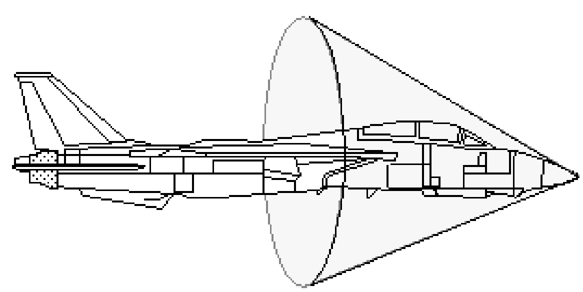


Fig. A-10 illustrates an aircraft exceeding the speed of sound. The supersonic jet drags the sound waves along with it. It does this because it is traveling faster than the sound waves.

Fig. A-10. Supersonic Aircraft Exceeding the Speed of Sound, Dragging the Sound Waves Behind It.



Note that the "V" of the boat is now replaced by a "cone." This is due to the fact that the jet flies through the air while the boat travels on the surface of the water. The case for the boat is a two-dimensional phenomenon, while the case for the jet is a three-dimensional effect.

The "cone" sweeps over observers on the ground. When it does so, the observers experience the large wave as a loud boom, the *sonic boom*. This large wave is also called a shock wave. The boom is only heard once for each observer. After one observer hears it, then another observer down the road hears it as the jet passes by this second observer, and so on.

The space shuttle makes such a sonic boom as it returns to the ground from Earth orbit. Why isn't there a sonic boom when the space shuttle is in orbit, even though it is traveling at "Mach 20"?

Another thing to consider is the whip. The tip of a whip of a cowgirl can exceed the speed of sound. The crack of the whip is actually a baby sonic boom.

There is a *speed limit* in the universe. It is obviously not the sound barrier. It is the light barrier. Einstein's *Theory of Relativity* states that there is a speed limit in the universe as a law of nature. This speed is 300,000 km/s (186,000 mi/h), the speed of light. Light speed is so fast that light appears to instantly arrive whenever it travels. It is beyond the scope of this text to delve into this mystery.

We can use the fact that light travels so fast to estimate how far storms are away from us. The lightning and thunder occur at the same time, but the light travels to us almost instantly. We start counting and wait for the thunder. How many seconds do you have to count for a storm one mile away? Hint: One mile equals 5280 ft, i.e. about 5000 ft. Sound speed is roughly 1000 ft/s.

Condensation Cloud. Dramatic conical shape cloud formations can occur when planes travel near Mach 1 due to complex pressure variations and rapid condensation. This is not a sonic boom. See the F-18 (Hornet) below.

Fig. A-11. F-18 and Condensation Cloud.

Courtesy United States Navy

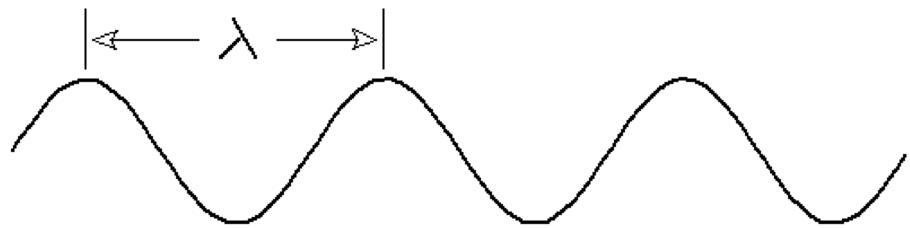
Simple Harmonic Motion

Before we leave this chapter we would like to get a better understanding of sound vibrations or waves. Just what are they? How can we draw a picture of a vibration? Let's consider a simple system for generating sound. You pick up a guitar and pluck the string. The string vibrates rapidly back and forth. The string's vibrations cause the surrounding layers of air to vibrate. These vibrations are in step with the vibrations of the string. The vibrations are transmitted through the air and reach our ears. Our eardrums vibrate in step with the vibrations of the air.

Strings are part of nature and vibrate in a most natural way. But since the string vibrates so fast, we will use another vibrating system to get a handle on natural vibrations. You. Since you are part of nature, and a "natural" person, you should be able to shake in a natural way. Take your hand and wave it up and down. Be natural. You will not hear any sound unless you can shake your hand up and down about 20 times a second. Probably not. But you can see your hand move. If you were to walk across a blackboard moving your hand up and down with a piece of chalk in it, you would trace out something like Fig. A-12 on the blackboard.

This simple or natural wave is called a *sine wave*. The distance from one peak to the next is called the *wavelength*. The wavelength is designated by the Greek letter lambda, written as λ. Note the natural way the wave rises and falls. The natural type of vibrational motion that generates such a wave is called *simple harmonic motion*.

Fig. A-12. Simple or Sine Wave with Wavelength λ.



The number of times you shake your hand back in forth per second is called the *frequency*. There is nothing special about a second. The number of times you shake back and forth per minute is also a frequency. Shaking once per second is equivalent to shaking 60 times per minute. The key idea is to count the number of cycles or times you shake during some designated time interval. Be careful that you count up and down as one cycle and not double count. A complete cycle takes you up and down.

We are now ready to observe an interesting phenomenon. If you shake your hand more rapidly and walk across the blackboard, the wiggles are spaced closer together; i.e., the wavelength is shorter. If you shake your hand less rapidly, the wiggles are spaced farther apart; i.e., the wavelength is longer. Shaking more rapidly means the frequency is greater. We say the frequency is higher.

This is heard as a higher pitch if the frequency is within the range of human hearing. Shaking less rapidly implies a lesser or lower frequency. These observations are summarized in Fig. A-13.

Fig. A-13. Relationship Between Frequency and Wavelength.

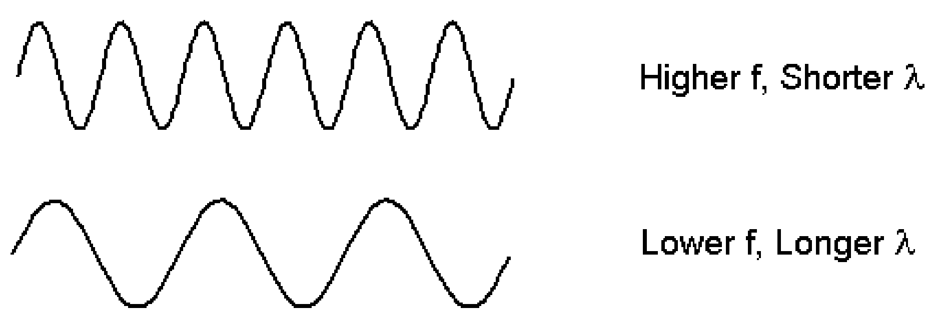
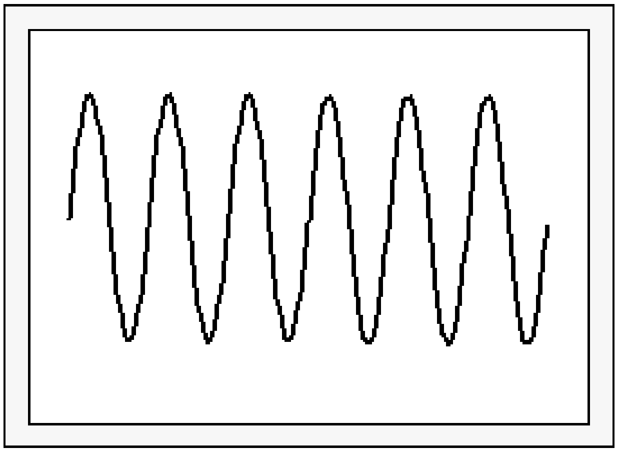


Fig. A-14. Sine Wave on an Oscilloscope.



Oscilloscope (Scope)

The oscilloscope is an electronic device that shows us a picture of a sound wave much like the way the blackboard shows a picture of your hand wave. A microphone converts the air vibrations to electrical vibrations. The oscilloscope (or scope) sweeps out the picture, performing the electrical analog of tracing the wave on the blackboard.

We can use an oscillator in a music synthesizer to generate a wave for us electronically. These can be monitored with an oscilloscope. They can also be sent to a speaker, where the electrical waves become mechanical as the speaker membrane vibrates.

The moving membrane creates sound waves in the air. We can call these acoustical waves, where air is implied. Technically, these are also considered mechanical by engineers. When the waves reach our ears, an interesting sequence occurs, just the opposite as before.

The sound waves enter the ear canal as acoustic waves in air. These waves cause the eardrum to vibrate, converting the oscillations to mechanical vibrations on a membrane. Finally, these mechanical vibrations are converted to electrical signals in the inner ear. The brain receives electrical information. So we begin and end with electrical information. This sequence is listed in Table A-3.

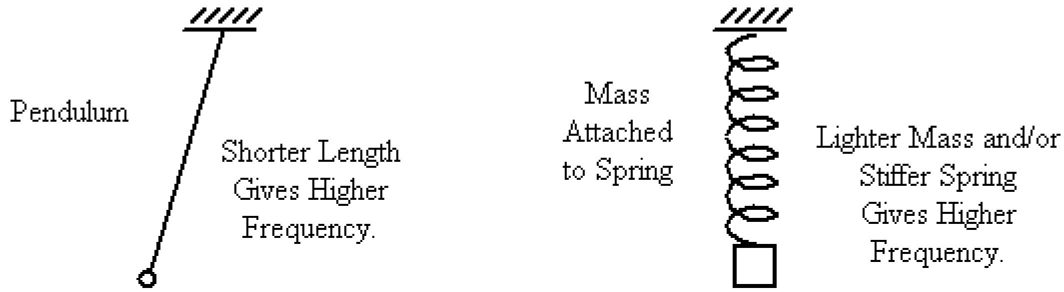
Table A-3. Different Forms of Energy as a Signal Travels from a Synthesizer to Our Ears.

Energy	**Location**
Electrical	Circuit
Mechanical	Speaker
Acoustical	Air
Mechanical	Eardrum
Electrical	Inner Ear/Brain

Two Examples of Simple Harmonic Motion

Finally, consider the motion of a pendulum and a mass attached to a spring (see Fig. A-15 below). These systems exhibit simple harmonic motion. Technically, the swing of the pendulum must not be too great for simple harmonic motion. The frequency of the pendulum and spring system depends on the string or spring length (shorter length for higher frequencies). The pendulum frequency does not depend on the mass of the swinging bob. All masses fall or swing at the same rate! However, the mass attached to the spring does affect the frequency of oscillation (lighter masses for higher frequencies). The stiffness of the spring is also relevant. Stiffer springs provide for higher frequencies of vibration.

Fig. A-15. Two Mechanical Systems that Exhibit Simple Harmonic Motion.



Deciding if Motion is Simple Harmonic

For your homework, you will be challenged to determine whether motion qualifies for simple harmonic motion. For example, is the up and down dribbling of a basketball by Michael Jordan simple harmonic motion?

This question is equivalent to asking "Can the motion be described by a sine wave?" Consider the following when analyzing any motion for simple harmonic motion.

- Is there a middle position where the object "would like" to be at rest?
- Is there motion on either side of the middle position?
- Do the fastest speeds occur at the middle position?

The simplest mechanical configuration for harmonic motion is a system with a linear restoring force. This means that when the distance from equilibrium doubles, the force doubles. A force law of this kind is said to satisfy Hooke's Law. A spring is an excellent example of a linear restoring force. For such systems, periodic motion is harmonic. For more complicated systems, like waving your hand in class, the real test is to graph the motion. If you get a sine wave, the motion is harmonic.

Do not worry if the harmonic motion eventually dies down. This is called damped harmonic motion and we will study it in the next class. You have to keep supplying energy to keep any motion going. Note that motion can be periodic and not be harmonic. We will encounter many waveforms such as the triangle, square, and ramp waves.

A sure-bet way to tell if the motion is simple harmonic is to try to mimic or follow the motion with your hand moving up and down. Now stick a pencil in your hand, move your hand across a sheet of paper, and let your motion across the paper draw a picture of the movement you are investigating. This is called a trace.

--- End of Chapter A ---

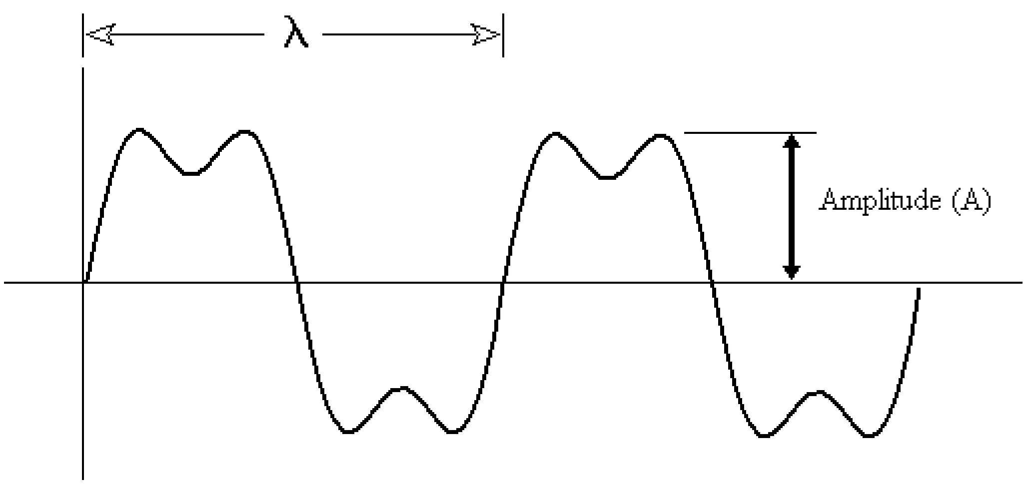
B. Vibrations

In this chapter we will learn about different kinds of periodic waves. We have already seen the sine wave, which results from simple harmonic motion. We will characterize periodic waves in general. The concepts of frequency and wavelength will still apply. We will then compare the physical characteristics of periodic waves with perceptual ones. How do different periodic waves sound? After this, we will see that no wave continues forever. Waves tend to die out; i.e., damp out. Energy must constantly be put into a system to maintain vibrations. We will see that driving a system at a special frequency gives the most efficient results. This is the subject of resonance. Resonance is very important in the design of musical instruments. It also has an analog in electrical circuits. It helps engineers design circuits that can respond to the least amount of energy.

Complex Periodic Waves

A periodic wave is any wave that repeats its pattern. Figs. A-11 and A-12 illustrate examples of periodic waves. There, you see sine waves, the simplest type of periodic wave. A complex periodic wave is any periodic wave that is not a sine wave. So that's everything else, as long as it repeats. An example of a complex periodic wave is given below in Fig. B-1. We include horizontal and vertical reference axes to help us describe the wave more clearly. Another example of a complex periodic wave is found on the cover of this text.

Fig. B-1. Example of a Complex Periodic Wave.



The *wavelength* λ is the distance from any point on the wave to the place where the pattern begins to repeat. It is easy for us to take the beginning point where the wave meets the horizontal axis and slopes upward in Fig. B-1. However, any starting point can be taken. The *frequency* is the number of patterns or cycles made per second.

Scientists like to use the second as the time interval. We will find this convenient since the sound we hear has frequencies easily expressed using this time unit. The *amplitude* is a measure from the horizontal reference (equilibrium) to the maximum point of the wave. The wavelength is measured in meters (m), centimeters (cm), or some other length unit of your choice. The metric system is simple since you only have to think in terms of 10, 100, etc. For example, the centimeter is 1/100 of a meter. This is better than having units like the yard, which breaks into 3rds to get feet, each of which then has 12 equal divisions (inches).

We will introduce metric units as we need them rather than hitting you with many at once. Often texts overwhelm the student with many metric units and then use less than half of them. You should not worry about learning all the metric units. The professional scientist doesn't know all of them either.

Let's consider frequency. Frequency tells us how many cycles or patterns occur per second. As an example, consider a source vibrating 50 times per second. We can write f = 50 "cycles" per second (50 cps), or f = 50 "cycles"/s. We can also write f = 50 "patterns"/s, 50 "vibrations"/s, 50 "oscillations"/s. How about f = 50 "wiggles"/s?

Since we can't get agreement on just what to call the repeating "things," we just write f = 50 1/s. We read this as 50 per second. Call them whatever you like. By convention, 1/s (per second, where cycles, patterns, or vibrations are understood) is named after the physicist that discovered radio waves, Hertz. So now we can write f = 50 hertz or f = 50 Hz for short. The lower case "h" is used when the word is written out as hertz and the upper case "H" is used when the unit hertz is abbreviated as Hz.

The hertz is a metric unit. The metric system consists of special units that scientists agree upon such as the meter, second, and hertz. The metric system also contains a series of prefixes which represents the numbers 10, 100, 1000, etc. and 1/10, 1/100, 1/1000, etc.

The prefix for 1000 is kilo, which can be abbreviated simply as "k." Therefore, one thousand hertz is simply kHz (a kilohertz). The hearing range for humans (rounded off) is from 20 Hz to 20,000 Hz (or 20 kHz). Another example is one hundredth of a meter, 1 cm (one centimeter).

You see, the metric system is easy. Simply attach the appropriate prefix such as centi or kilo to the relevant unit such as meter or second. One thousand seconds is a kilosecond (ks). A hundredth of a second is a centisecond (cs). Some silly authors like to attach metric prefixes to anything they like, which technically is allowed. For example, one thousand lectures is a *kilolecture*. Two thousand mockingbirds is two kilomockingbird, i.e., *To Kill a Mockingbird*.

Another physical parameter is the *period*. The period, designated by T, is the time it takes to complete one pattern or cycle. This depends on the frequency. If the frequency is f = 10 Hz (i.e., 10 1/s), what is the period? In other words, if you do something 10 times per second, how long does it take to do it once? The answer is 1/10 second (0.1 s). If f = 5 Hz (i.e., 5 1/s), the period T = 1/5 s. Note that you simply flip the frequency to get the period. Flipping 10 gives 1/10 and flipping 5 gives 1/5. Also note that the units flip: 10 1/s becomes 1/10 s on the flip. Note that if you flip the period, you get the frequency back and vice versa. We summarize these relationships as follows: **T = 1/f** and **f = 1/T**. The mathematical name for this flip is reciprocal.

It's convenient to learn the metric prefix for a thousandth, which is milli. The period for a 1000-Hz sound wave is 1/1000 s = 0.001 s = 1 millisecond (or 1 ms). Let's do another example.

Consider a 100-Hz sound wave. The period T is 1/100 s by flipping the 100 1/s. Therefore, T = 1/100 s = 0.01 s, just like 1/100 of a dollar is $0.01. This answer is perfectly satisfactory. In order to convert to milliseconds, write the period as T = 0.010 s. This is still one hundredth of a second; however, it is expressed as 10 thousandths of a second. So our period is 10 ms (10 milliseconds). Another way to think of this is to move the decimal point three places to the right and you go from seconds to milliseconds in one sweeping step.

Continue the analogy with money. Consider $10 and proceed slowly. One tenth of a 10-dollar bill is one dollar. One hundredth of $10 is a dime. One thousandth of $10 is a penny, as you need

1000 pennies to make $10. Now, 10 pennies equals one dime; i.e., 10 thousandths of $10 equals one hundredth of $10. Are you confused? If so, this is normal. So what you do is read it over again slowly and perhaps write some things down to help you visualize it.

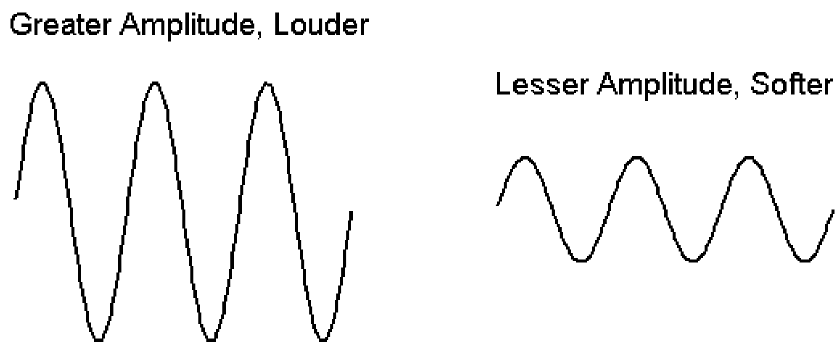
Perception of Periodic Waves

We would like to relate the physical descriptions of the previous section with perceptual characteristics. There are three basic perceptual features of periodic waves to consider.

1. Loudness. The easiest perceptual characteristic to investigate is loudness. The loudness of a wave is determined by the amplitude, as illustrated in Fig. B-2. However, the relationship is not simple. You can't say that if you double the amplitude, the sound is twice as loud. There is a good reason for this. If you want to be able to hear a whisper and a loud thunder crash, the ear has to be stubborn in perceiving loud sounds. It takes more than doubling the amplitude for the ear to be impressed. We will learn more about sound levels and loudness in a later chapter. The only thing we state here is qualitative: the greater the amplitude, the louder the sound. This also applies to non-periodic waves like explosions.

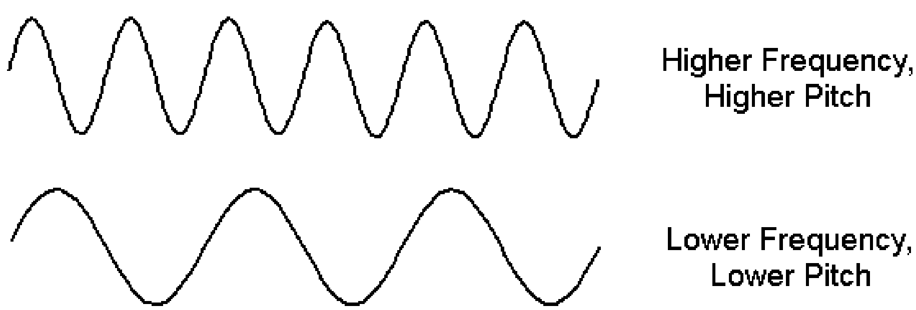
Periodic waves are heard as steady tones. Although amplitude mainly determines loudness, other factors affect how loud sounds appear. Our ears are not uniformly sensitive to all frequencies, so some sounds may sound louder simply due to our sensitivity, especially to high tones.

Fig. B-2. Amplitude and Loudness.



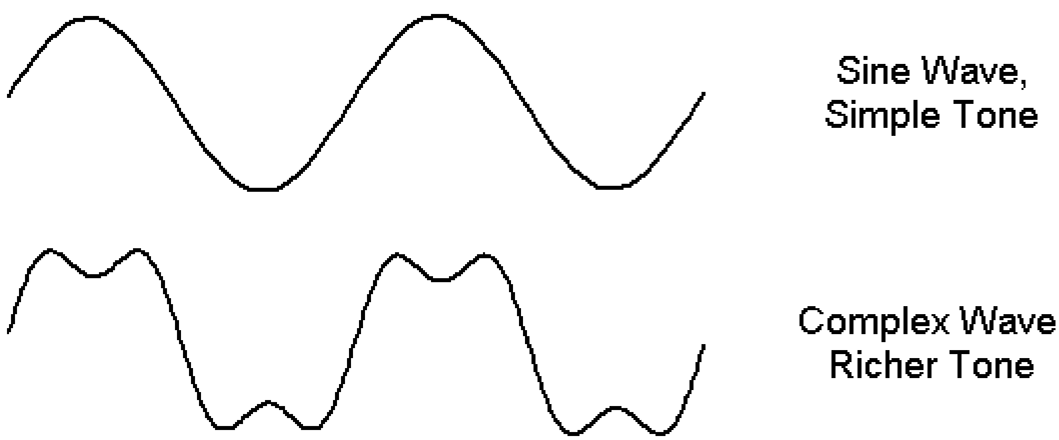
2. *Pitch*. Different frequencies of periodic waves are heard as different pitches or tones. The higher the frequency, the higher the pitch. Notice that our earlier discussion of frequency, wavelength, and period showed that all three of these are related. So we can alternately, say that wavelength and pitch are related, where short wavelengths mean high pitches (see Fig. B-3). Or we can alternately say period and pitch are related, where short periods indicate high pitches.

Fig. B-3. Frequency and Pitch.



3. *Timbre*. The timbre (TAM-ber) of a periodic tone is that quality which enables us to distinguish between the flute and the violin. These instruments may play the same pitch at the same loudness but we still hear a difference. The timbre (also spelled timber and also pronounced TIM-ber if you like) is determined by the shape of the waveform. In fig. B-4 below you find the waveform for a sine wave and one for the complex wave we encountered earlier. Note that the amplitudes and frequencies are essentially the same. The sine wave however will sound pure and innocent while the complex wave will sound richer and harsher.

Fig. B-4. Waveform and Timbre.



The correlation between physical properties and perceptual characteristics helps us understand the connection between physics (acoustics) and psychology (perception). Table B-1 summarizes these results.

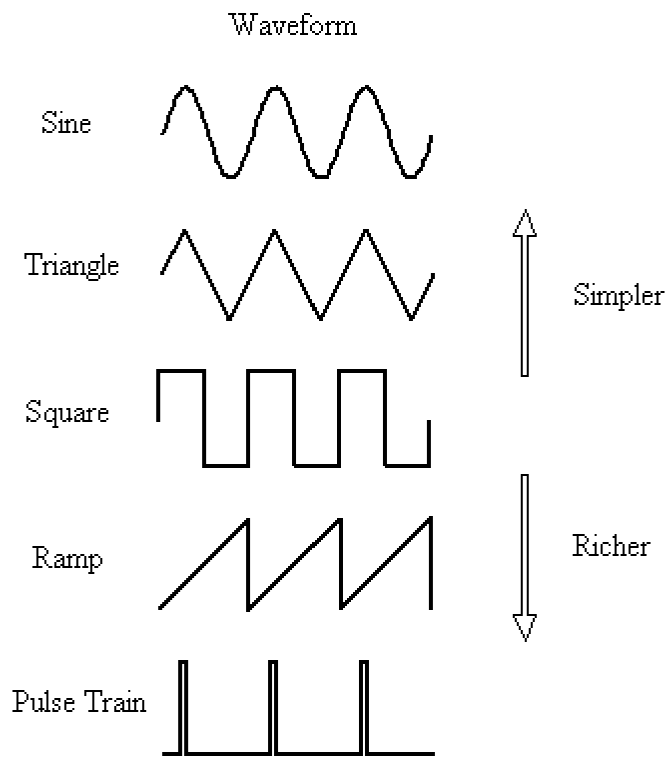
Table B-1. Basic Relationship Between the Physics and Psychology of Sound.

Physical	**Perceptual**
Amplitude	Loudness
Frequency	Pitch
Waveform	Timbre

Later we will learn about the decibel (dB) scale which ranges from 0 dB (a pin drops) to 140 dB (near a jet, which damages your ears). Actually, damage can occur in a machine shop with 90 dB. The range of frequencies we hear is from 20 Hz to 20,000 Hz, as noted earlier. Later we will also analyze complex periodic tones and learn more about timbre.

However, for now, let's look at some different waveforms. We will simply note their pictures. Five waveforms can be found in Fig. B-5 below. These have nice shapes and are easy to synthesize with electronics. The sine wave is the simplest waveform. It sounds the purest in tone. The others are arranged depending on how different they appear compared to the sine wave. Later we will learn precise justification for this order. The pulse train sounds the harshest. Try moving your hand in step with each of these and imagine your eardrum vibrating likewise in step.

Fig. B-5. Five Different Waveforms.

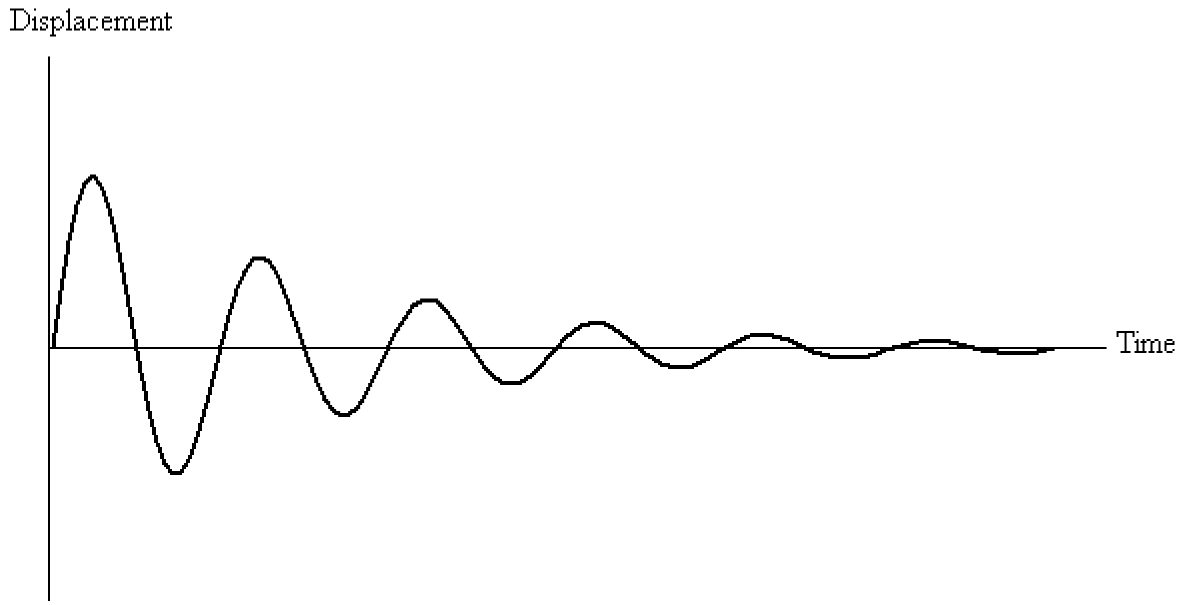


Damped Periodic Waves

Unfortunately, oscillations die down unless there is new input of energy. If you start a pendulum swinging, its amplitude decreases as it swings until it eventually stops. The oscillations are said to be damped. Fig. B-6 below illustrates such motion. It is a plot of displacement (position away from the equilibrium position) versus time. Scientists like to say it's a plot of displacement as a function of time. This may represent a pendulum. You kick it when the clock is set to zero (see beginning of the graph). The pendulum ball or bob starts to swing away from equilibrium (say to the right). It reaches some maximum displacement to the right, then starts to come back. It overshoots the center and proceeds to the left. This is represented in the graph as the curve dips below the horizontal axis. The bob swings back and forth but doesn't reach the larger distances from the center that it originally did. So the graph below indicates a gradually decreasing amplitude. Note that the motion is damped simple harmonic motion since it is represented by a sine wave decreasing in amplitude.

The graph in Fig. B-6 can also represent the vibration of a mass attached to a spring. However, the mass is not pulled back in this case but hit with an object. For a mass pulled back and let go, the graph starts somewhere away from the equilibrium line. The graph can also represent a dying sound wave. There is a law of nature that says there is no such thing as perpetual motion. It is the *second law of thermodynamics*. You may have heard about entropy (disorder) and the law that everything tends toward disorder. This is the same law. The *first law of thermodynamics* states that you can't get anything for nothing (conservation of energy); the second law states that you can't break even. Some energy is lost to friction. The pendulum stops swinging, the mass attached to a spring stops oscillating, the sound we hear dies down.

Fig. B-6. Damped Harmonic Motion



Driven Oscillations

Due to the laws of thermodynamics, we want to be careful when we pump energy into a system. We want to maximize our efforts. Suppose we want to push a child on a swing. We would of course push the youngster at the frequency that the swing wants to go at. This is common sense. In this way, our energy use is maximized. When we stop pushing the child, of course the swing will eventually stop.

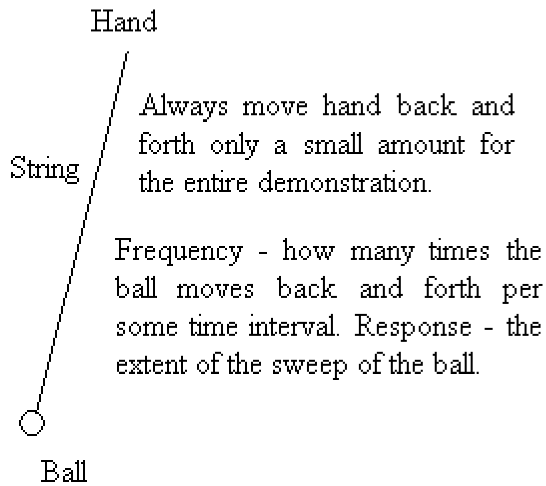
But if we were to push the swing at some crazy frequency, we would be wasting our efforts. This wasted effort could be demonstrated by moving a hand back and forth at some different frequency than the swing. Perhaps, one time when the kid arrives at the hand, the push helps. But most times the push will not be with the swing. Occasionally, we will actually be trying to stop the kid when our hand pushes too soon and collides with the child's back. The second law of thermodynamics is bad enough for us to throw our efforts away. Let's find out when the driven system responds the best to our efforts. This brings us to an analysis of driven oscillations.

Fig. B-7 describes an experimental arrangement to study driven oscillations. Tape a small ball to the end of a string. Grab the end of the string opposite the ball. Shake your hand back and forth, keeping your sweep within a short space (1 or 2 cm). You may have a friend bracket this space with a thumb and forefinger for you.

Can you find the frequency that makes the ball respond most dramatically? Your driving frequency is then called the *resonance frequency*. You are driving the pendulum at the frequency it likes to swing.

Stop, pull back the ball, and let it go so that it swings on its own. Isn't this the same frequency you used earlier? It will be unless you do the experiment in a thick medium like "oil."

Fig. B-7. Hand Driving a Pendulum.

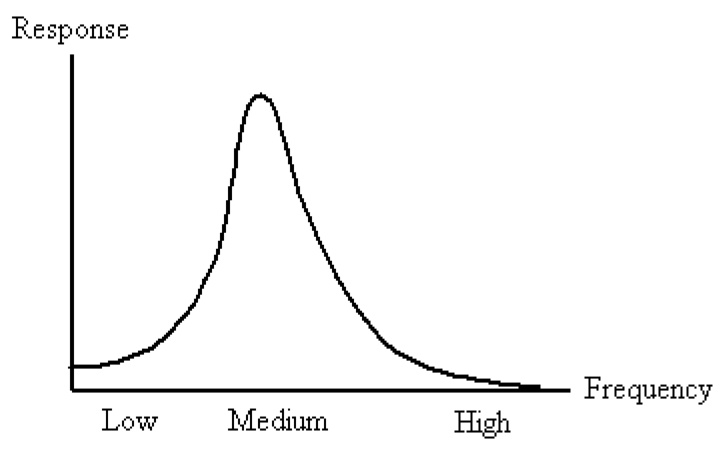


Get a feel for the ball swinging on its own by counting or having someone clap each swing. Now stop the ball and start driving it from rest at this frequency. The ball will gradually increase its swing until it reaches a maximum as you continue to drive the pendulum at the resonance frequency. Resonance occurs when the ball responds with maximum swing. Resonant vibrations are also called *sympathetic vibrations* (the system "is in sympathy" with your vibrations). Compare the responses of the ball (total extent of swing) for different driving frequencies (low, high, medium). Note that the ball moves a little with a low-frequency driving force, and hardly moves at all when driven at high frequency. Somewhere at a medium frequency, the response is greatest (resonance).

Now we are going to make a graphical sketch of our results. In Fig. B-8 below we view a graph of response (vertical axis) versus frequency (horizontal axis). A low response is found at low frequencies. When you shake the pendulum slowly, the ball moves very little. When you shake the string rapidly, the response is even less. You change directions so rapidly that the ball cannot respond quickly enough. So it just sits there. However, at a medium or intermediate frequency, the ball responds the greatest.

Take your pencil and place it at the peak of the graph in Fig. B-8. Now place a ruler vertically and draw a dotted vertical line downward until you hit the horizontal axis. Your dotted line should be parallel to the vertical axis at the left in Fig. B-8. Now make a dark dot where your vertical line touches the horizontal line. This marks the spot for the value of the resonance frequency. If our graph had numbers on the frequency axis instead of the words low, medium, and high, you could read off the value of the resonance frequency.

Fig. B-8. Plot of Response versus Frequency: The Resonance Curve.



Resonance is important in musical instruments. When a violin is played, the energy action of the bow drives the string into resonance. Certain frequencies are further enhanced as the wood and cavity support additional resonances. This gives a richer quality to the tone.

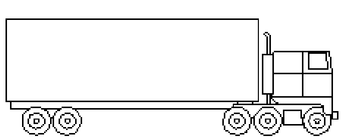
When one softly blows across the opening in a flute, the energy supplied drives the pipe into a resonance, depending on the effective length of the flute (controlled by pressing key pads). Playing a recorder or toy flute, one covers holes. Depending on which hole is uncovered, the pipe acquires a different resonance frequency. Rather than trying to guess this frequency and whistle into the pipe, you gently blow, creating a turbulence of many frequencies.

The amplification due to resonance is so impressive that you will readily generate a tone. The multi-frequency breath sound at the mouthpiece is called noise. But only that frequency component that corresponds to the resonance frequency of the pipe is picked out and amplified as the pipe resonates at that frequency.

Examples of Resonance

Your author has searched extensively for 10 examples of resonance to help you master this important concept. Examples were found in such diverse places as the author's home, *Star Trek*, and the *Bible*. All of these involve vibrations of some kind or another. You should try to identify the source of energy in each case, the system being driven or excited, and the actual resonance effect itself.

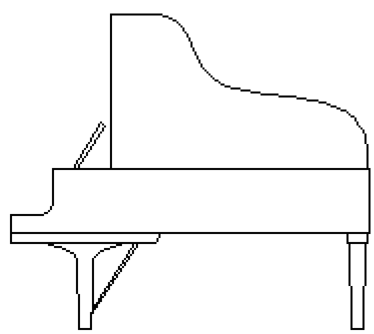
R1. A Passing Truck Shakes Things in a House. Large trucks generate low frequencies.

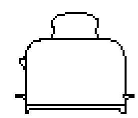


Such low frequencies can cause items in one's home to vibrate. The author was often scared in Philadelphia (at his in-laws' former home) when he felt the entire house shaking from the third-floor bedroom as large city buses passed outside.

R2. Piano Note Rattles Toaster. In the author's former apartment at the *University of Maryland*, hitting a key near the middle of the piano caused a toaster element across the room to rattle. Only one key near the center of the piano did the trick. The notes produced by keys to the left were too low in frequency, while those produced by keys to the right were too high.

It is good to remember that the lowest note on the piano is about 30 Hz and the highest note is about 4000 Hz (or 4 kHz). The key causing the resonance was about 300 Hz. An important middle key on the piano is called middle C and has a frequency of near 260 Hz.

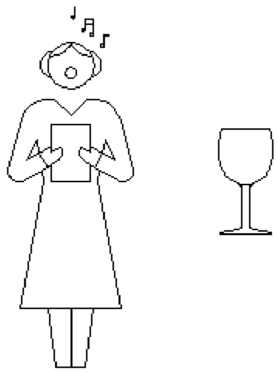




R3. Singer Breaks Wineglass. This is the classic case of a singer who sings a particular note that shatters a glass. In the 1980s, a *Memorex Commercial* on TV featured Ella Fitzgerald singing into a microphone and breaking a glass. They taped her. Then they played the tape and yes, the recording broke the glass also.

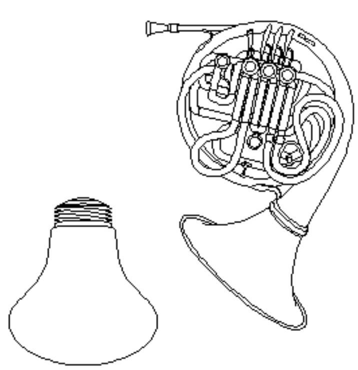
Resonance is used dramatically in the movie *The 4th Tenor* (2002) as our star (Rodney Dangerfield) "shatters" more than a wine glass with his new operatic voice.

The author once found broken pieces of glass from a lamp on the floor in *Lipinsky Hall*. He inquired as to what was going on the night before, suspecting resonance. He discovered that the *UNCA Choir* had been practicing.



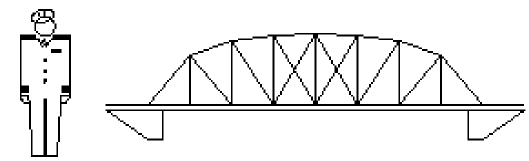
Interestingly, breaking a glass without a microphone was not documented until 2005, when Jamie Vendera did it for *MythBusters*. He hit 105 decibels at 556 Hz and had to try 12 delicate wine glasses before finding one with a proper structural defect.

R4. An Orchestra and Floodlights. The author was at a concert at the *University of Maryland* in the early 1970s. The university orchestra was playing the Brahms *4th Symphony*. The 3rd movement opened with the usual French horns playing the E (330 Hz) above middle C. As the horns held this note, the same tone could be heard coming from a source somewhere to the left. It was an eerie sound.



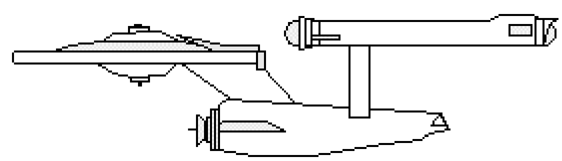
Then there was a pop and a floodlight exploded. In the movie *The Mask* (1994), Jim Carrey uses a surrealistic toy sound maker and blows out the windshield of a car.

R5. Soldiers Marching Across a Bridge. Soldiers do not march across bridges because the uniform steps could induce resonant vibrations in the bridge. They break step to prevent any possible driven oscillations.



R6. The Avalanche. An avalanche can be started by vibrations. You do not want to shout in an environment where an avalanche can occur. This might involve an unstable arrangement of snow or rocks. Vibrations at the proper frequency can shake the snow or rock formation causing an avalanche.

In a very early *Star Trek* episode, *Friday's Child* (December 12, 1967), Captain Kirk and Science Officer Spock are running from bad guys on a planet. The medical Doctor McCoy is in the mountains assisting a lady about to give birth. Kirk and Spock come up to a mountain. They are trapped.



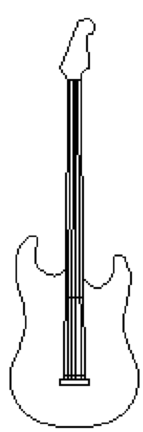
Spock suggests that they try to induce sympathetic vibrations in the mountains, dislodging the rocks. Spock knows his science - that's resonance. Kirk asks for the probability of success, which Spock points out is not real good.

They of course try it anyway. Unfortunately, they turn on their communicators to do it. The communicator sends out radio waves, not sound waves. Well, let's imagine they have a setting for high-frequency acoustic waves we can't hear. They need a high frequency since the stones they wish to excite are very small. At this point, we are trying to save the story from incorrect physics. The rocks do come down and the bad guys get wasted. But there is 25 minutes remaining in the story, so Kirk and Spock get caught by a new wave of bad guys. Kirk uses psychology to get out of the mess at the end of the episode.

An avalanche scene also appears in the Disney movie *Herbie Goes to Monte Carlo* (1977). Here they get the physics right. Don Knots (in the car "Herbie" with his friend driving) yodels in the mountains, impressed with the echoes. They are actually lost and he is calling for help. A small avalanche starts, then stops.

Bad guys show up, as the good guys were tricked into making a wrong turn during a race. But Herbie has the idea by now and begins to yodel (the car's horn). A larger avalanche occurs, where rocks land on the bad guys' car.

R7. Rock Concert. There was a rock concert at Princeton during the 1970s where everyone starting stomping to the music. The gym started shaking. Remember, soldiers know to break step when walking across bridges. However, rock lovers don't know that they shouldn't stomp in unison inside a large auditorium. Police ran in to get the audience to stop. Can you imagine the confusion?

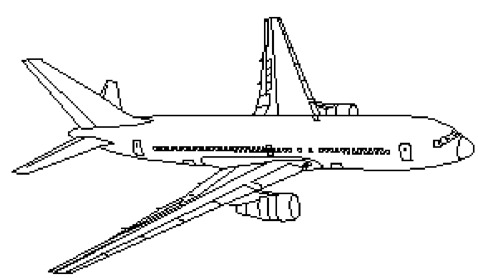


Hippies entranced with the music probably didn't notice the police at first. With all the noise and distraction, some thought the police might be after them. Some probably shouted "Fuzz, Fuzz" to their friends, worrying about the "joint" they were passing around. The police had more urgent concerns - the gym itself. Well, the gym did not collapse.

However, skywalks in a Midwest hotel did collapse in the early 1980s. People were dancing, so resonance was a possibility. But the investigation showed that the floor wasn't bolted correctly. The collapse was due to weight, not oscillations.

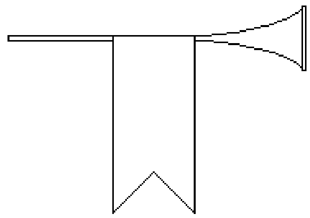
R8. Jet and Construction. A jet (DC 10) on its way to landing at Chicago's *O'Hare Airport* flew over a stadium undergoing construction in 1979. Things shook, some of the structure collapsed, and tragically, five people were killed. The *Asheville-Citizen* (August 14, 1979) reported that a worker described the event by saying "Suddenly, everything began to vibrate. You could hear the roof cracking and then it started falling in."

The paper quoted a spokesman for the *Federal Aviation Administration* (*FAA*) stating that planes were flying "just a couple hundred feet" over the site. These are the clues for resonance - low-flying jets, sympathetic vibrations picked up in the unstable roof. However, the *FAA* spokesmen went on to say that he never heard of airplane turbulence causing any destruction on the ground. He obviously did not know his physics. It's not the air turbulence that caused the destruction, it



was sound resonance!

R9. The Walls of Jericho. Suppose a scientist who was not familiar with the Bible began studying the Bible with the Jericho story. What would our scientist learn about the characters and the events that transpire in the story?



Our scientist would first learn that Joshua is in some sort of trouble. Joshua is leading a group wandering in the desert. Joshua's people need to get into a city, but walls prevent them. So Joshua goes off to a desolate place to get help from God. Joshua hears a voice that responds to his requests and proceeds to tell him what to do.

From what Joshua is told to do, our scientist concludes that God has a very insightful understanding of the laws of physics. Below we give the instructions Joshua receives and alongside include a scientific commentary of these instructions.

Our guide here for the interpretation is what we know about resonance and physics.

Bible	Scientific Commentary
"... On the seventh day march around the city seven times, and have the priests blow the horns. When they give a long blast on the ram's horn and you hear that signal, all the people shall shout aloud. The wall of the city will collapse, and they will be able to make a frontal attack." *Joshua 6:4-5*	1. *March Around* - to spread out. 2. *Blow the Horns* - start the resonance. 3. *Ram's Horn* - resonance frequency. 4. *Long Blast* - so response builds up. 5. *Shout* - noise for reinforcement. 6. *Collapse* - due to resonance.

Everyone needs to *march around* so that the sound can better reach the wall. The driven oscillations begin when *the priests blow the horns*. The *ram's horn* is important as its frequency is near the resonance frequency of the wall. It's necessary to a make a *long blast* so that the wall oscillations can build up. Having the people *shout aloud* adds to the ram's horn. Present in the overwhelming noise is the resonance frequency that reinforces the ram's horn. The *collapse* occurs when the stress limit is reached in the wall due to the resonance vibrations.

Our scientist may wonder why God doesn't explain the physics to Joshua more fully. Probably, Joshua wouldn't understand it. Rather than confuse Joshua, it is more efficient to just tell him what to do. Note that Joshua is told the outcome. In this sense, a scientific principle <u>is</u> explained - s*ound can make a structure collapse* (like our previous roof collapse caused by the low-flying jet).

Our scientist may also wonder why God doesn't just knock out the wall for Joshua. Our scientist surmises that perhaps God interacts only through voice. But wait, God could then perform the "sound" resonance for Joshua. So our scientist concludes that God is willing to assist Joshua, who asks for help, but that Joshua has to do some of the work himself. How does our scientist's interpretation of the Joshua story compare to yours?

R10. The *Tacoma Narrows Bridge*. Physics teachers have discussed the *Tacoma Narrows Bridge* as the "granddaddy" of resonance for several decades. The bridge was completed around 1940. Winds were able to get the bridge oscillating dramatically. The wind in the ravine supplied the energy for the oscillations. If you are somewhat confused as to how the wind caused the oscillations, don't worry, so are many physics teachers.

Today we believe the oscillations were not caused by resonance, but rather by some more complicated interactions.

For an obvious scenario of bridge resonance, consider a monster like *King Kong* giving gentle pushes on an appropriate place of the bridge at the right frequency. In a short time, the oscillations of the bridge build up impressively. This would be resonance.

However, we include the *Tacoma Narrows Bridge* here for its historical association with resonance and its spectacular display of catastrophic behavior.

The bridge's road surface oscillated in a twisting fashion. Such waves are called *torsional waves*. The *Tacoma Narrows Bridge* collapsed one morning after about 40 minutes of twisting. A film of the incredible oscillatory bridge motion is observational proof of the elasticity of solid structures. The bridge vibrated like a large string. There was plenty of time for all the cars except one to vacate the bridge. One car got stuck on the bridge. The driver of this car crawled toward one of the bridge's towers. He heard concrete crackling. He made it off the bridge. However, his main concern afterwards was how to explain to his daughter that he couldn't save her dog, who was riding with him in the car.

In 1995, a commercial advertising a Pioneer Sound System used footage from the *Tacoma Narrows Bridge Collapse*. They sped up the movie. The commercial won 1995 *Grand Clio Award*, the top award for cleverness and creativity in advertising.

Chladni Plates

The German physicist Ernst Chladni (KLAD-knee), regarded by some as the father of acoustics, discovered an ingenious way to visualize resonance vibrations of a plate. You sprinkle some sand or white powder on the plate and then use an oscillator to produce frequencies. When you hit a resonance frequency of the plate, the plate goes wild. Regions where the plate vibrates become free of the powder as it gets shaken off. However, some paths along the membrane do not move and the white powder stays there.

One finds beautiful patterns for the various modes of vibrations. Plates are two dimensional and very complicated compared to the thin strings and pipes will be discussing later. Strings and pipes are more like one-dimensional structures as the long length predominates over the small cross-sectional area.

Fig. B-9. Chladni Pattern

Courtesy www.FreeScienceLectures.com

--- End of Chapter B ---

C. Waves

We have seen that sound consists of vibrations or oscillations. Scientists study traveling vibrations under a general category called waves, a topic we investigate further in this chapter. We are already familiar with several periodic wave characteristics such as amplitude, frequency, and waveform. So we have been studying waves already. Here, we will sharpen our skills by probing deeper into what a wave is. A definition can be given as: **a wave is a traveling disturbance.**

Fig. C-1a below illustrates a very simple traveling disturbance. Our assistant has shaken a hand up and down at the left end of a rope to make a single crest. This "crest disturbance" is a pulse that travels to the brick wall. It is a wave; however, it is not periodic. The hand was shaken once. Notice that the rope moves up and down in the various locations along the rope as the pulse travels. The rope itself doesn't travel but the deformed moving shape does.

The rope is the medium that supports the waves. If there were no waves, the rope would appear at rest in equilibrium everywhere as a horizontal line. Since the rope medium moves up and down, i.e., transverse to the direction of motion as the disturbance passes, we say that the wave is *transverse*. What do you think happens when the pulse reaches the brick wall?

In Fig. C-1b we view a pulse on a slinky. Our assistant has pushed in and out quickly to start a compression wave. Note that the "slinky ringlets" move back and forth along the direction of propagation (parallel) as the pulse travels to the wall. The wave is said to be *longitudinal*.

Fig. C-1a. A Transverse Pulse Wave Traveling Down a Rope.

Fig. C-1b. A Longitudinal Pulse Wave Traveling Down a Slinky.

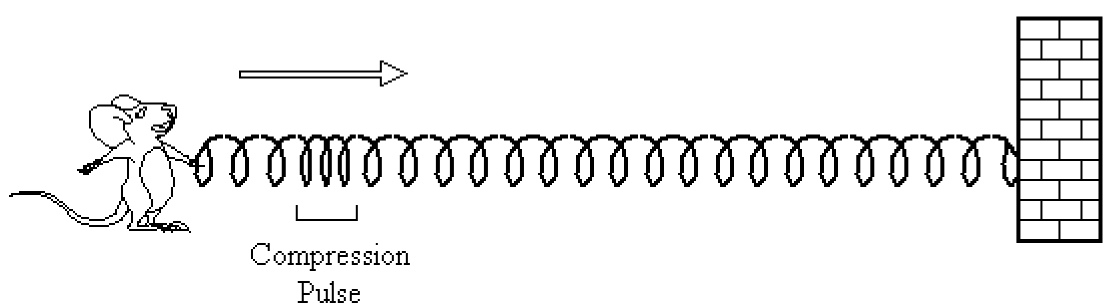
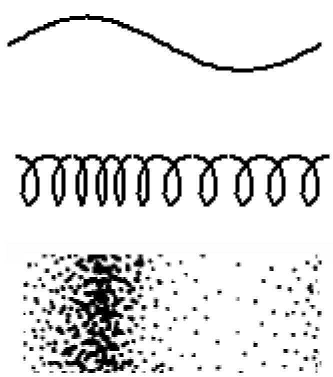


Fig. C-2 illustrates our rope wave, slinky wave, and a representation for a sound wave. The rope wave shows a crest and its counterpart, the trough. This is one complete wavelength for a simple periodic wave. You make a *crest* by pulling your hand up at the end of a rope; you make a *trough* by pulling down below the horizontal. Similarly, the slinky wave shows two sections, a compressed region and its opposite, a stretched region. You make a compressed region by pushing in at the end of a slinky; you make a stretched region by pulling out. A simple longitudinal wave then consists of a compressed region (analogous to the crest of a rope wave) and a stretched region (analogous to the trough of a rope wave). The rope wave is transverse and the slinky wave is longitudinal. The amplitude for the rope wave is the distance measured from the center equilibrium line up to the maximum point. For the slinky wave, we compare the density of "ringlets" (how many per cm for example) with the density for a normal slinky with no waves on it. In each case, the amplitude is a measure relative to the quiescent state of the medium.

Fig. C-2. Three Types of Waves.



Wave	Disturbance	Amplitude
Rope	Crests Troughs	Displacement (centimeters)
Slinky	Compressed Regions Stretched Regions	Density (ringlets per cm)
Sound	Compressions Rarefactions	Pressure (atm)

The third example in Fig. C-2 represents a sound wave. Sound waves are similar to slinky waves. The layers of air are compressed in some regions, where the molecules get closer together. Air layers are less dense in other regions. These less dense regions are called *rarefactions*. The amplitude of a wave is always measured from equilibrium.

The equilibrium density for air is that density when there are no sound waves. We can use this value or alternatively the equilibrium pressure. Then the *compressions* are regions of higher pressure and the *rarefactions* are regions of lower pressure. A unit of pressure is the atmosphere (atm). At sea level under normal conditions, the pressure of still air is 1 atm. We will not need to worry about units for amplitudes of sound waves in this text. Since sound is similar to the slinky case, we state the following important observation: **sound is a longitudinal wave.**

The easiest wave to sketch is the transverse wave (see the rope wave in Fig. C-2). Slinky waves are hard to draw. Layers of air molecules are just too cumbersome to mess with. So we sketch the transverse picture to represent all three types of waves. In Fig. C-2, the three waves are lined up. The crest of the transverse wave coincides with a compressed region in the slinky wave and compression in the sound wave. The trough coincides with the stretched slinky and rarefaction. So the crest in a transverse picture can mean compression (higher pressure) and the trough indicate rarefaction (lower pressure).

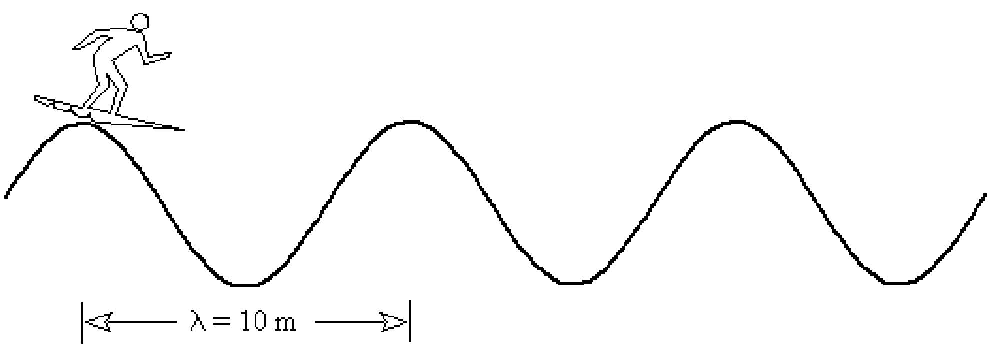
This representation is more meaningful than you may realize. Push your hand in and out like you are driving a slinky. You can imagine your hand to be the membrane of a speaker generating sound waves. In either case, you are making longitudinal waves. Now someone comes along, tapes a piece of chalk to your moving hand, picks you up, orients you properly so that the chalk touches a blackboard, and carries you across the room as your hand writes on the board. You will trace out a transverse picture of your longitudinal wave. This is just what the oscilloscope (mentioned earlier) does for us. So we will use the transverse drawings to represent our longitudinal sound waves.

Before we discuss basic properties of waves in general such as reflection, let's look at the general relationship between frequency and wavelength in more detail. We already know that high frequencies have short wavelengths and vice versa. Now we will learn that the frequency and wavelength are related to the speed of the wave (also called the velocity). Suppose you are riding on a periodic wave (see Fig. C-3 below). For a specific example, say that the wavelength is 10 m and that these pass by a stationary observer on the shore at a frequency f = 5 Hz (5 go by per second). This is some wave! (But not a real water wave.) Then, the rider moves 5 x 10 = 50 m/s. We see that the speed of a wave is equal to the wavelength multiplied by the frequency.

wave speed = wavelength x frequency

$$\boxed{v = \lambda f}$$

Fig. C-3. A 5-Hz Wave with Wavelength 10 m Moving to the Right.



The relationship $v = \lambda f$ incorporates our observation that higher frequencies mean shorter wavelengths. The speed v for the wave depends on the medium. For sound at room temperature, this is 340 m/s. Suppose we have a rope wave where the speed is 100 m/s. Then, a low frequency of 5 Hz means a wavelength of 20 m since 20 x 5 = 100.

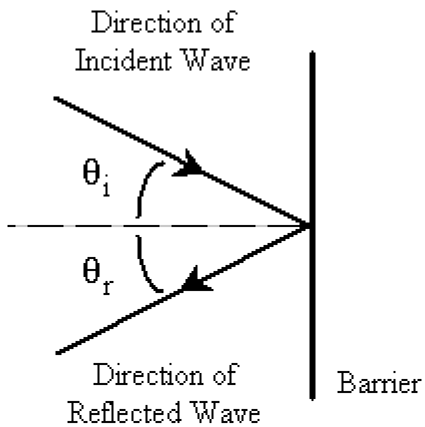
If we increase the frequency to 10 Hz, then the wavelength shortens to 10 m since 10 x 10 = 100. Raising the frequency to 25 Hz, the wavelength becomes even shorter, 4 m (since 4 x 25 = 100). Think of an analogy with money where 100 stands for a dollar. You can get a dollar with 20 nickels or 10 dimes or 4 quarters. Can you think of other combinations? Note that as you increase the value of each coin, the number of coins decreases because the total amount of money is fixed at a dollar. This is analogous to our fixed wave speed. Again, what does this imply in terms of wavelength and frequency?

Reflection

Waves exhibit the property of reflection. A reflection occurs when a wave bounces off an obstacle. The directions of the incident and reflected waves have a simple relationship. Fig. C-4 illustrates the *law of reflection*. We do not show the waves in Fig. C-4, just the direction of the waves. The waves bounce off the barrier so the angles of incidence and reflection are equal. These angles are measured relative to a center reference line called the *normal*. The law of reflection is demonstrated readily with a flashlight, i.e., using light. You can easily achieve a concentrated beam of with your flashlight. You use a mirror for the reflecting barrier.

The law of reflection also describes bouncing a ball off a side of a pool table. Finally, if we neglect gravity or go on the space shuttle, throwing a ball along the direction of incidence in Fig. C-4 will result in the ball leaving along the direction of reflection, after its collision with the wall.

Fig. C-4. Law of Reflection.

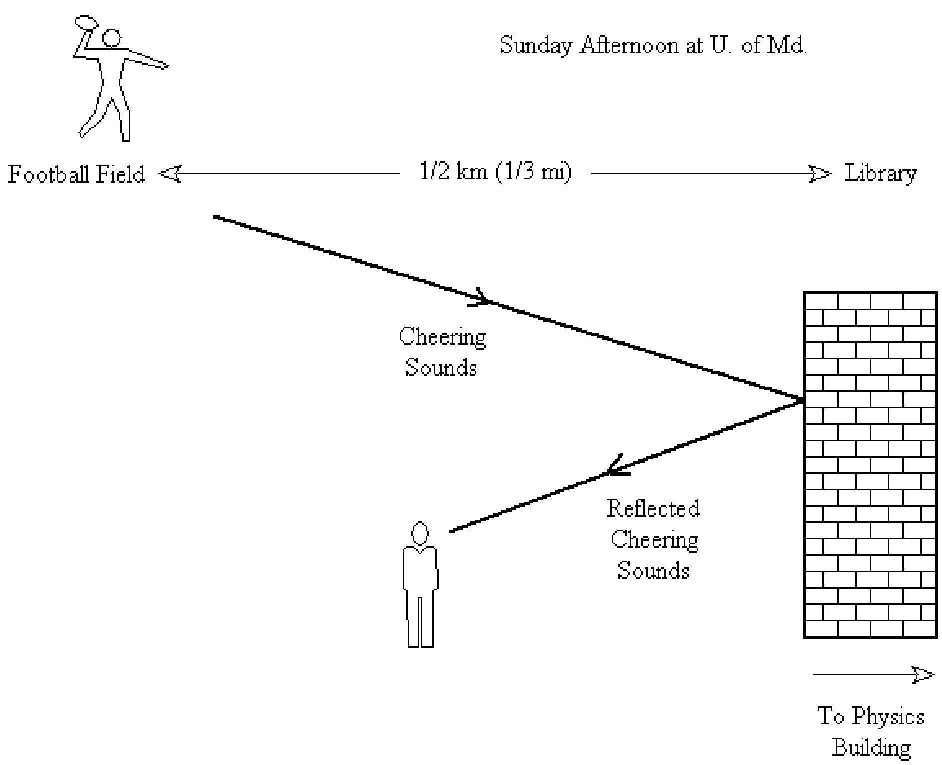


The angle of incidence equals the angle of reflection: $\boldsymbol{\theta_i = \theta_r}$.

Every time you hear an echo, you experience the reflections of sound. One fine Sunday afternoon, the author heard a dramatic reflection outdoors at the *University of Maryland*. He was walking from Graduate Housing to the Physics Building in the early afternoon to do physics. Most people were at the football game. He heard cheers and shouts as he walked through the parking lots. As he approached the *Undergraduate Library*, which is situated across the street from the *Physics Building*, he heard the cheers coming from in front of him. This was an eerie experience as the library was in that direction (see Fig. C-5 below), not a cheering crowd.

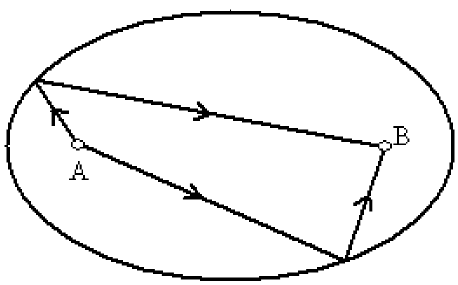
It was especially strange because other buildings between the football field and the observer (not shown below) prevented any direct sounds from the field to reach him. Therefore, the usual echo effect did not occur. One just heard the "reflected beams" and was tricked into interpreting them as the real thing. After a moment's "reflection," the author figured out what was going on.

Fig. C-5. Sound Reflecting From Tall Brick Building.



Another example of reflected sound is found in the *whispering chamber*. This room has an elliptical shape (see Fig. C-6 below). A person at position A can have a conversation with a person at position B even though the distance between these points may be great. In fact, the speaker at A can speak gently or whisper. The sound waves bounce off the walls in such a way that they focus after reflection at point B. Two such rays are shown in Fig. C-6. Imagine numerous rays emanating from A, hitting the wall at various places, yet all reaching B! Other observers throughout the room may get a few rays but the observer at B gets reinforcement from all the reflections of waves starting at A and vice versa.

Fig. C-6. Sound Reflecting In a Whispering Chamber.



There is a *whispering chamber* in the *Capitol Building* in Washington, DC. Other examples can sometimes be found in wall structures outdoors. Can you locate the one on the *UNCA* campus?

Museums sometimes have a demonstration where there are two large dishes. You speak at a designated point in front of one of these and your friend far away at a corresponding point near a second dish can hear you. Dishes for sending waves far to another dish need to be parabolic in shape, a shape similar to the ends (left and right) of the elliptical walls in Fig. C-6.

There is also the story of a church in Europe with the *whispering chamber* effect. By some architectural coincidence, a confessional was placed near one focal point, while the other focal point was somewhere down the aisle. The person at the right spot in line for confession was able to hear the sins being confessed at the moment (so the story goes).

Refraction

Sound travels at different speeds through air with different temperatures. Sound travels faster in warmer air than it does in cooler air. Table C-1 below lists some air temperatures and the corresponding speeds of sound at those temperatures. Can you see a relationship between speed and temperature? The general relationship is that the speed of sound increases as temperature increases.

Analyze the data and see if you can determine by how much the speed increases in m/s (meters per second) for every 10 degrees on the Celsius or centigrade scale. Scientists get excited when they discover such relationships in data. They are fascinated by the recurring order and beauty found in nature. Their delight in such order and beauty is similar to the appreciation of balance and grace in a Rembrandt painting or the experience of harmony and elegance in a Mozart sonata.

The nice rule incorporated in the data of Table C-1 is valid over the temperatures that we usually find in our environment. Things get complicated if the air becomes extremely hot or cold.

Table C-1. Sound Speed and Temperature.

Temperature	Sound Speed (m/s)	Fahrenheit Value
Freezing (0°C)	331	32°F
Cold (10°C)	337	50°F
Room (20°C)	343	68°F
Hot (30°C)	349	86°F

The speed of sound's dependency on air temperature has interesting effects. In order to easily illustrate this, we need to learn how to sketch wave crests. Fig. C-7 shows a sine wave with a sketch of the wave crests below it. Remember that the sine wave can also describe longitudinal waves. The vertical lines can represent the crests of a water wave or the high-pressure regions of a sound wave (the compressions).

Fig. C-7. Sine Wave and Wave Crests.

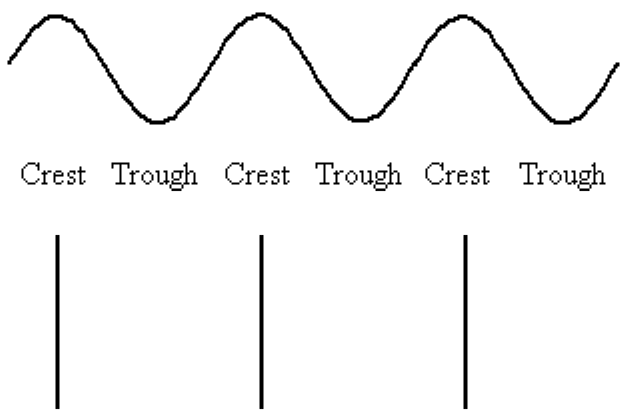


Fig. C-8 below illustrates crests emanating from a point source. Think of a stone dropped into water. Circular waves move outward. Fig. C-8 is a snapshot in time showing the circular crests at a given moment. The center source can also represent a sound source. Then the outgoing waves are spherical in three dimensions. However, you have the basic idea in the two-dimensional picture.

Fig. C-8. Crests From Center Point Source.

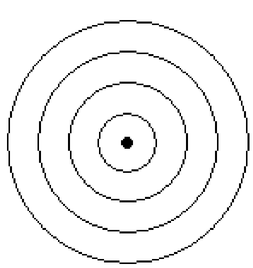
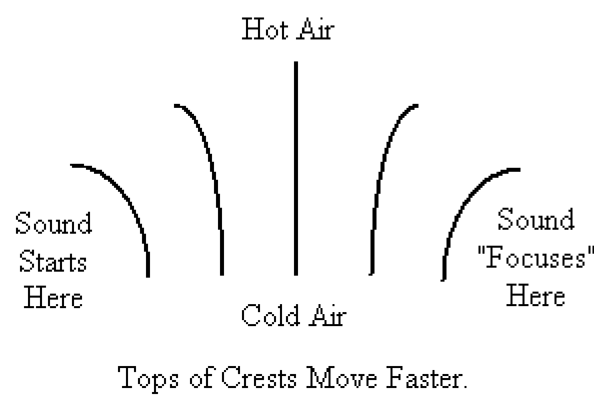


Fig. C-9 depicts a source of sound at the left of the figure. Only part of the outgoing circular (really spherical) wave is shown, the part heading toward the right. These circular crests get distorted since the air temperature is not the same everywhere. In our example, the air is cooler at the ground and warmer above. The crest travels faster in warm air so the top parts of the crests get ahead of the lower parts. This causes the wave crests to straighten out (see center crest) and then curve the other way. The waves in a sense "focus" at a point to the right. This converging is the opposite of spreading out and the sound is reinforced at the far right. The second half is kind of like running a movie backwards. In summary, sound waves first spread out, then recollect. This phenomenon is called *refraction*.

Fig. C-9. Sound Crests and Refraction.



An observer at the far right is in a better position to hear the source than an observer in the middle. This will occur when hot air is above cold air. In the daytime, heat from the sun makes the ground hot. The air near the ground is hotter than the air above. Have you ever walked in your bare feet on an n asphalt driveway at noon in the summer? At night, the ground cools as the hot air rises with no additional sunlight streaming down. This cool ground with warm air above is now the opposite of what we had in the day. It is called a *temperature inversion*. This temperature inversion allows sound to be heard better farther away. It's not that sound "carries" at night; the bending waves do the trick.

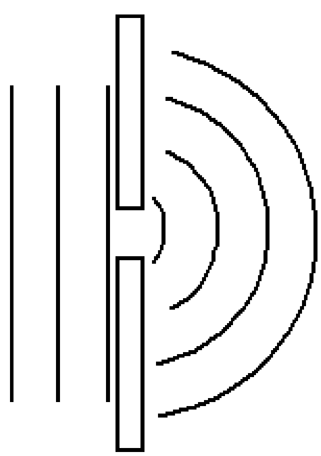
The author often heard music from the *Grove Park Inn* at his former home on Howland Road about half a mile away (little less than 1 km). He thought there was a party around the block until he realized it was refraction of sound due to the temperature inversion at night. UNCA's first chancellor, Chancellor Highsmith, told the author (around 1980) that he heard it also from the Chancellor's Residence on Macon Avenue.

Diffraction

Diffraction refers to the bending of waves near openings, corners, and obstacles. This is another type of bending of waves in addition to refraction.

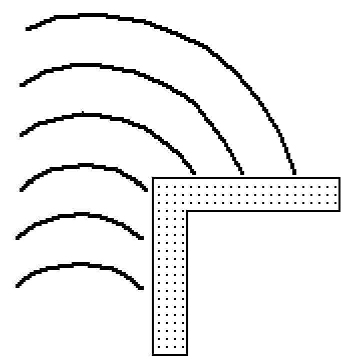
D1. Openings. An easy way to understand diffraction is to think of water waves heading toward an opening between two barges (see Fig. C-10a). The crests of the water waves are linear as they approach the opening. Then circular waves spread out on the other side of the opening. Do sound waves spread out like this through open doorways? Can you hear someone call you from within a classroom if you are down the hall?

Fig. C-10a. Diffraction Through Opening.



D2. Corners. Waves also diffract around corners. Can you call a friend who has just walked around the corner of a building? Does the sound reach your friend? A general guide for diffraction is that waves will diffract, i.e., bend around openings, corners, or obstacles, provided that the encountered structures are comparable in size to the wavelength of the waves (distance between crests).

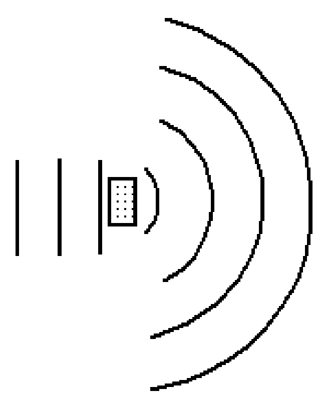
Fig. C-10b. Diffraction Around Corner.



D3. Obstacles. Waves also diffract around obstacles. Have you ever tried talking to someone on the other side of a tree? Can they hear you? Can you turn around and still maintain a conversation with someone behind you at home? Do the sound waves wrap around your head and get to the other person?

In light of what was said in 2 above about the conditions for diffraction, why do you think light doesn't bend around trees?

Fig. C-10c. Diffraction Around Obstacle.



Interference

Two waves combine when they meet. You simply add the amplitudes. Fig. C-11 considers square waves so that addition is easy. In Fig. C-11a the waves reinforce because the crests of the two waves match and likewise the troughs. The waves are *in phase*. The *interference* is *constructive*. In Fig. C-11b the interference is *destructive* as crest meets trough and vice versa. The waves are *out of phase*.

Fig. C-11a. Constructive Interference (Two Identical Waves In Phase).

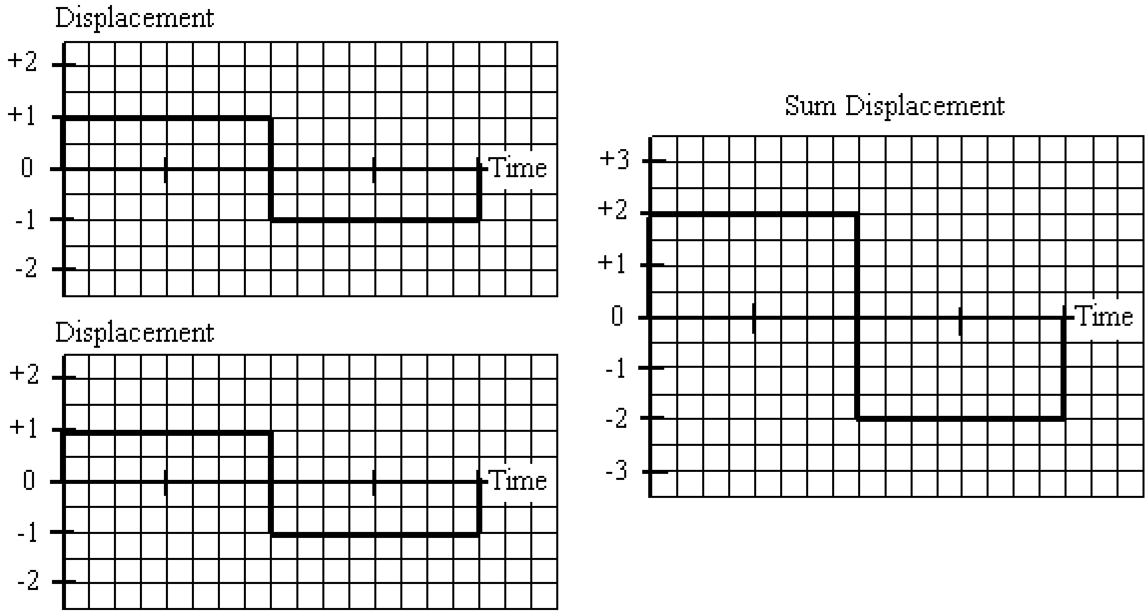
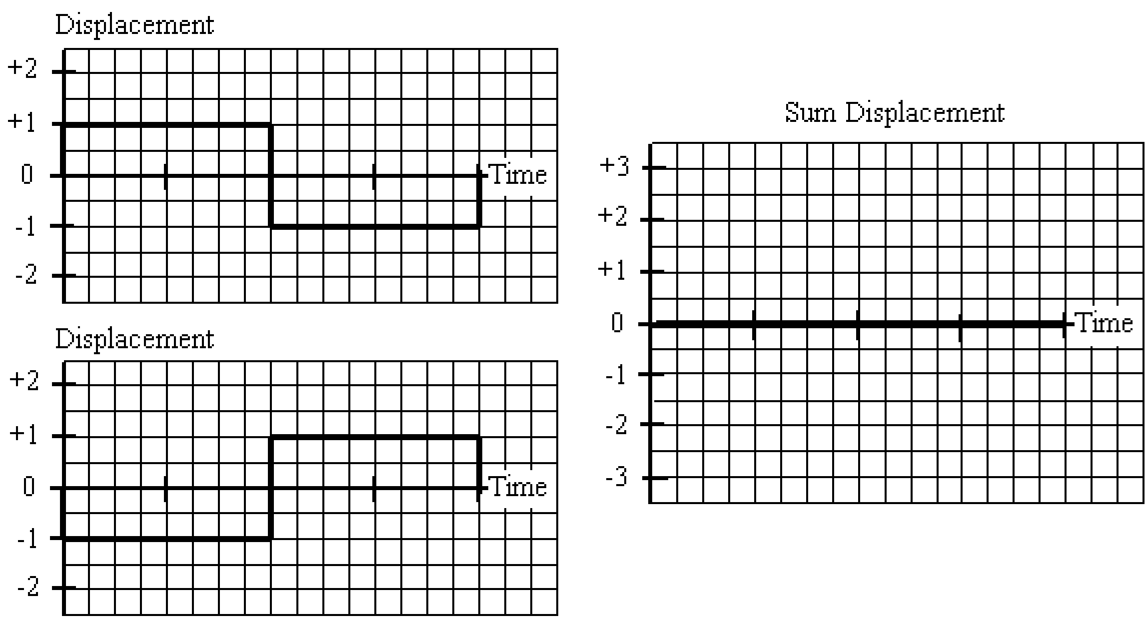


Fig. C-11b. Destructive Interference (Two Identical Waves Out of Phase).

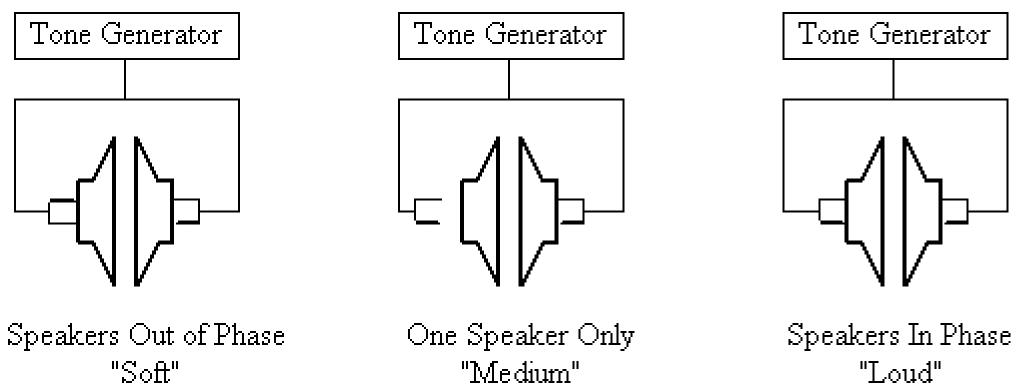


Sketch a sine wave on a piece of paper and see if you can sketch underneath your sine wave another similar wave aligned so that the crests and troughs work together. Now do another sketch for the case where the waves work against each other and cancel.

Fig. C-12 illustrates an arrangement to study interference. We have two speakers. We let them face each other to get them close to each other. We send a simple tone to each speaker to keep the input easy to compare. We do not use stereo but send the same signal to each speaker.

We experiment with switching the polarity of one of the speaker connections. The speaker wire contains two wires, insulated but bundled together in the wire leaving the source. With the polarity reversed for one speaker, one speaker membrane pushes out to produce compression while the other pulls in to make a rarefaction. The waves then work against each other as compression competes with a rarefaction. The sound level drops. In fact, it is softer than one speaker alone (middle case illustrated in Fig. C-12). Of course the loudest sound occurs when the speakers send out compressions together (case at far right of Fig. C-12).

Fig. C-12. Interference with Speakers.



Instructions usually warn you not to cross one of your speaker hook-up wires. These speaker wires attach your amplifier to your speakers. They assume that you sit equidistant from each speaker so that a compression leaving one will arrive at you directly when a rarefaction does so from the other speaker, if wired incorrectly. Here are some questions to think about.

1. If your speakers are wired incorrectly, would you be better off in an open field (assuming electricity is available) or in a house with walls? Think of reflection.

2. If speakers are wired incorrectly, are you better off listening to stereo or an old monaural recording?

3. What happens if you reverse the polarity of both speakers?

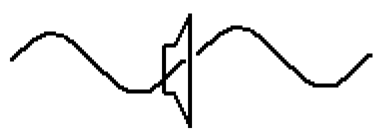
There is an interesting demonstration that illustrates both diffraction and interference. Consider a speaker by itself. Then consider a speaker inserted into a hole of a flat board. The latter speaker is said to have a baffle. You could also insert the speaker into a wall to get a similar effect.

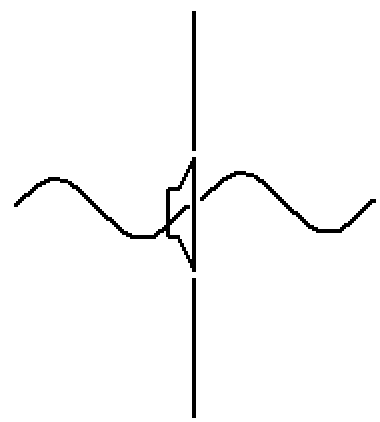
A speaker membrane sends out waves both to the front and to the rear. When the membrane pushes out, compression is sent out toward the front. At the same time, a rarefaction is sent out the rear. The waves leaving the front are out of phase with the waves leaving the rear. Refer to the sine waves in Fig. C-13. Without the baffle, a portion of the front and rear waves diffract. That is, part of the rear wave diffracts and gets to the front and vice versa. Suppose you were talking to two friends, one in front of you and one behind you. Both would hear you. The person in the rear would hear part of the forward wave that diffracts around your head. But since you do not talk out of the back of your head at the same time, you don't need to worry about a second wave interfering.

The problem with the speaker is that unlike you, the speaker produces waves out the front and rear simultaneously due to its simple membrane structure. And these are out of phase! When they diffract and mix, they interfere destructively. The sound is weakened.

The baffle (see right diagram in Fig. C-13) prevents the waves from diffracting. Therefore the waves cannot mix. There is no destructive interference and the sound is louder. The speaker can now produce the sound it is capable of without working against itself. The emitted sound is improved in both the forward and rear directions.

Fig. C-13. Speaker With and Without Baffle.





--- End of Chapter C ---

D. Wave Applications

The wave phenomena we studied in the previous chapter emphasized fundamental properties of waves. In this chapter, we focus more on applications. Some of the applications we encounter here will build on the basic physics of the previous chapter.

SONAR

SONAR stands for SOund Navigation And Ranging. One can determine the depth of a body of water by sending sound waves to the bottom and timing how long it takes the sound to return. However, we must note that sound travels at different speeds in different media. In fact, even in the same medium such as air, the sound varies if the temperature of the air changes.

Table D-1 is a short list of sound speeds in a few media. The three main states of matter are accounted for: gas, liquid, and solid. Sound travels faster in lighter gases such as helium. Helium atoms are light and respond quickly. We commonly encounter a use for helium in party balloons. Helium's lightness causes balloons filled with helium to float upward in air.

Table D-1. Speed of Sound in Various Media.

	Medium	Sound Speed (m/s)
Gas	Air	340
	Helium	1000
Liquid	Water	1500
Solid	Wood (Typical)	4000
	Steel	6000

Sound speed is greater for liquids, where the molecules are closer together and they can transmit their vibrations more readily. The stiffness of solids allows for even more rapid propagation of sound waves.

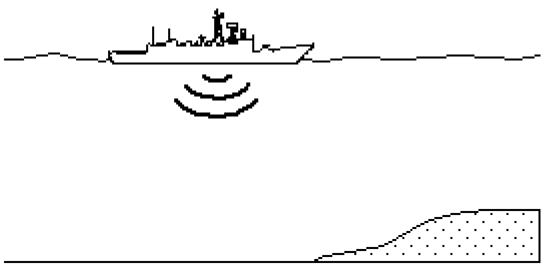
The value for wood given in the table is typical of elm, maple, and oak. Steel is the stiffest of the examples in Table D-1 and therefore has the greatest speed of sound. Sound speed in steel is approximately 6000 m/s = 6 km/s (6 kilometers per second, which is about 4 miles per second).

Question: The circumference of the Earth is 40,000 km (25,000 miles). How long would it take to go around the world if you could travel at the speed of sound in steel?

Answer: How many kilometers per hour do you go if you travel at 6 kilometers per second? Multiply 6 km/s by 3600 to get 21,600 km/h and round this off to 20,000 km/h. Since you have to go 40,000 km, you get an estimate of 2 hours for your travel time!

Let’s measure water depth by sound waves. See Fig. D-1 for a sketch of the general idea of SONAR. The sound is sent to the bottom of the water, bounces off the ground, and returns to the ship. The time it takes to return is then used to determine the distance. A travel time of 1 second indicates 1500 meters traveled since the speed of sound in water is 1500 m/s. For this case, the water is 750 meters deep, one half of 1500 meters since the sound needs to travel down and back. We have an echo effect, a result of the fundamental principle of waves called reflection.

Fig. D-1. SONAR: Sound Waves Used to Measure the Depth of Water.



Recall that we encountered an experiment to measure the speed of sound on the *UNCA Quad*. There we knew the distance and time. We figured out the speed. Here we apply our knowledge of the speed and measure the time to determine the third variable, the distance.

If you are not too familiar with the metric system, remember that a meter is approximately a yard. A meter is actually a little longer than a yard, about 8% longer (8 centimeters or 3 inches longer). One hundred meters is the distance between the goal posts on a football field, a distance a little longer (8%) than the 100-yard playing field. A mile is 5280 feet or 1760 yards. In terms of meters, the value is approximately the same numerical value as that for yards: 1609 meters. Therefore, the speed of sound in water (1500 m/s) is almost one mile per second. Our body of water is 750 m deep or about a half-mile deep.

Ultrasound

Ultrasound refers to frequencies above the limit of human hearing. Rounded off, this limit is 20,000 Hz or 20 kHz. Ultra-high frequencies can be used to make images of internal body structure. The word ultrasonics is also often used to describe such high-frequencies.

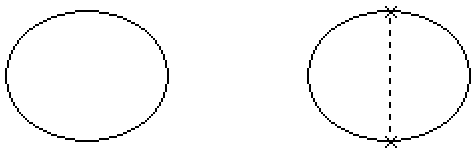
Ultrasonic means "beyond sound," which can refer to passing the speed of sound or to exceeding the frequency of the sound we hear. We use the latter definition in this section. The word supersonic we encountered in an earlier chapter is synonymous with ultrasonic.

Ultrasound penetrates the body and reflects off underlying structure. The waves that return are used to construct an image of the inner body. Such a medical image is called an *ultrasound* or *sonogram*. Ultrasounds are often used to take pictures of the fetus since common medical imaging using x-rays would be very harmful to the fetus. When you are x-rayed, a section of your body is chosen. A heavy cover containing lead may be placed over nearby parts of your body, and the technician may leave the room when the x-rays are emitted.

Sound waves allow the technician to look at parts of the fetus and capture images on a computer. Parts can be examined. Lengths of bones can be measured and compared with normal growth patterns. See Fig. D-2 for dimensions of a fetus from an ultrasound exam done at *Asheville Women's Medical Center* in 1983.

Medical researchers constantly improve imaging techniques and study their effects on the patient. As a general rule, you should always consult with your doctor about the possible side-effects for even supposedly safe treatments. While basic physics can tell us for example that ultrasound is very safe compared to x-rays, statistical medical observation over a long period of time is necessary to assess subtle or long-term biological effects to any diagnostic test or treatment. As with any developing field of research or science, the latest information can always modify previously-held beliefs.

Fig. D-2. Dimensions of Fetus (Author's First Daughter) from Ultrasound Exam.



View of Head   Marked for Measurement

Distance: 61 mm

Single Fetus, 25th Week (7/29/83).

Body Section	Size
Width of Head	61 mm
Humerus (upper arm)	43 mm
Femur (thighbone)	46 mm

The smaller variety of bats, called microbats, use ultrasound to navigate. These bats are about the size of a small bird. They emit an ultrasound which reflects from objects. When they receive an echo, they know something is in front of them.

This is convenient for bats living in dark caves. Or maybe, if a cave is not available, the bats may find a home in an old dark abandoned house - perhaps the one on the hill near the graveyard. Or a bat may settle for your dark attic. The bat's ability to locate objects and prey by an echo is referred to as *echolocation*. Echolocation does not have to use ultrasound. Some birds use sounds in the audible range. Porpoises also use echolocation.

We will use a simple model to estimate the ultrasound frequency of bats as physicists often use simplified ideas to estimate things. Bats typically use pulses of ultrasound rather than a continuous emission. The high frequency has very short wavelength. Recall that waves diffract around obstacles comparable in size to their wavelengths. So if you want to get a good bounce off a very small object, use a very short wavelength. Then, the structure is "large" compared to the wavelength and reflection predominates instead of diffraction. The ability to get a reflected wave from a small object in order to acquire its image is called *resolution*.

Bats use short wavelengths to obtain good resolution. This is especially important if you like to dine on insects. And most bats do. However, three of the hundreds of species of bats prefer blood. These are called vampires!

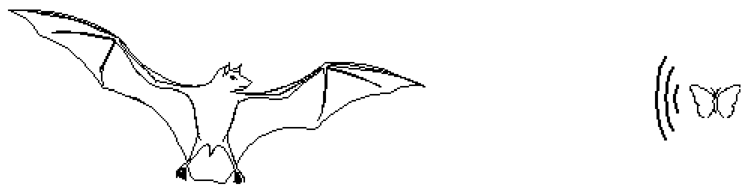
We can use the relation $v = \lambda f$ to find out how high the frequency has to be in order to image an insect. Let's suppose that a bat wants to have a moth for dinner. Estimate the size of a moth. How about 3 cm (three centimeters)? That's a little more than an inch. To get a significantly smaller wavelength than this, let's pick 1/10 of this value. That gives a wavelength of 0.3 cm or 3 mm (3 millimeters). Think of 1 cm as the width of your little finger and 1 mm as the size of lead in a pencil. To use our relation $v = \lambda f$ properly we need to be consistent with our units for distance. If we stick with mm, then we express the speed of sound, 340 m/s, as 340,000 mm/s. Remember that 1000 mm gives one meter.

So if you travel 340 meters, this distance is also 340,000 millimeters. Since our analysis is an estimate, we can round off 340 to 300. Then $v = \lambda f$ becomes 300,000 mm/s = (3 mm) f. What must 3 be multiplied by to get 300,000? The answer is 100,000. Therefore, f = 100,000 Hz (one hundred thousand hertz).

Bats actually use frequencies as high as this. See Fig. D-3 for a wave reflecting from a moth. Bats are not blind, so you shouldn’t use the expression "blind as a bat." But echolocation enables the bat to detect a flying insect from afar.

Challenging Problem: Estimate the frequency for a medical ultrasound head if the resolution is 1 mm and sound speed in tissue is 500 m/s. Answer: f = 5,000,000 Hz = 5 Megahertz = 5 MHz.

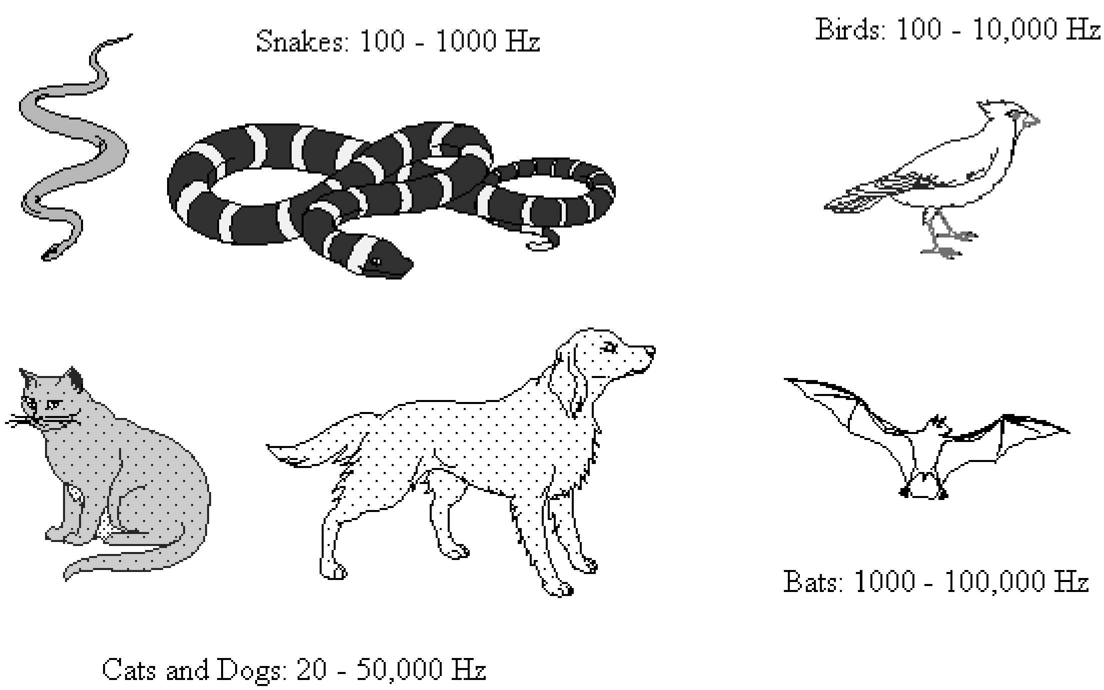
Fig. D-3. Reflected Ultrasound from Flying Insect.



The hearing ranges of some common animals reach well into the ultrasonic region. We are familiar with a dog's ability to easily hear high-pitched sounds. The silent whistle produces a tone of about 30,000 Hz (30 kHz), well beyond human hearing. Yet dogs respond to the whistle. The whistle is small since the wavelength is very short. We will study waves in pipes later. A general feature to remember is that small structures produce short-wavelength sounds, i.e., high pitches. Large vibrating mechanisms produce long-wavelength sounds, i.e., low pitches. That's why during Saturday-morning cartoons, the little creatures speak in high-pitched voices and the large ones sound low and deep.

Fig. D-4 indicates the hearing ranges of some animals. Snakes do not do well. They are restricted to lower frequencies. Humans, with a hearing range of 20 – 20,000 Hz, hear many more frequencies than snakes. Human speech falls mostly between 250 and 4000 Hz. In order to get an idea of what frequencies snakes can hear, have your friends talk to you with rags jammed in their mouths. This will filter out the higher frequencies. The other animals listed in Fig. D-4 surpass humans. Our friend, the bat, ranks the highest.

Fig. D-4. Some Animals and Very Approximate Hearing Ranges.



Elephants have a range from about 16 Hz to 12,000 Hz, but they are very sensitive at the lower end. For the other animals we have generously rounded off at the lower end.

Courtesy www.ghananewsagency.org

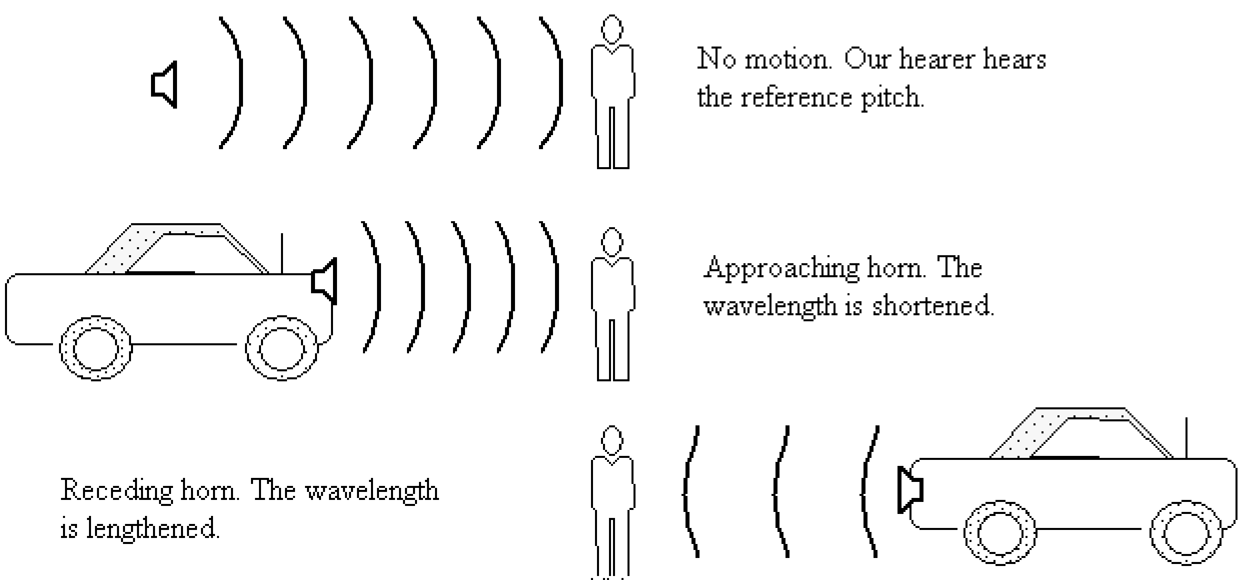
The Doppler Effect

When waves are bounced off moving objects, there is a shift in frequency. This phenomenon is named after physicist-meteorologist, Doppler. Back in 1840 there were no cars driving around every day in order to hear a passing car's horn shift in pitch. We have all heard such shifts. When a car blowing its horn comes toward you, the pitch is higher. When the car is leaving you, the pitch is lower. The effect is most dramatic when a car heading toward you, passes by you with the horn sounding. Measuring the shift in frequency can tell you how fast the car is moving relative to you. Police use the *Doppler effect* with an invisible form of light when they bounce radio waves off your car (RADAR). Bats experience frequency shifts when their high-frequency acoustic waves bounce off moving targets. This can help them determine the speed of flying insects. The delay of the echo gives the bat the distance to the prey.

Doppler had musicians help him with an experiment to verify that frequency changes due to motion. Some musicians played horns on a moving train. They chose a specific note. Other musicians on the ground could tell with their well-trained ears that the tone was higher as the train approached them, to their surprise. In music jargon, a raised pitch is said to be "sharp" or "sharper." A lowered pitch is said to be "flat" or "flatter." The musicians on the ground heard a tone that was too sharp as the source of sound approached them.

Fig. D-5 illustrates the Doppler Effect. Think of the horn as sending out a crest (compression) every so often. When the car approaches, it moves a little toward the last crest it sent out to the observer. Therefore, the distances between crests are shortened and the pitch is higher. When the car is moving away, the opposite takes place and longer wavelength means lower pitch.

Fig. D-5. The Doppler Effect.



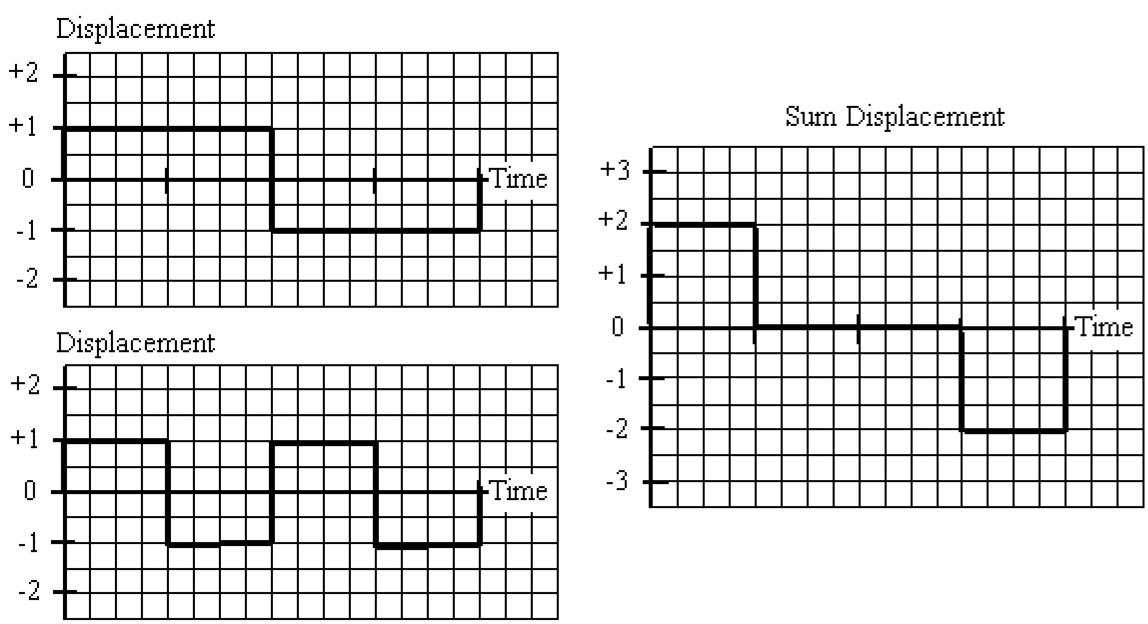
Wave Addition and Beats

We have seen the effects of adding waves in our previous discussion on interference. In our study of interference we considered two identical waves. When the waves were added in phase, we obtained constructive interference; when the waves were added out of phase, we found destructive interference. Now we are going to add different waves. To demonstrate the phenomenon of beats we need two waves that are almost the same. However, we will first take this opportunity to just practice adding different waves. This will help us develop the necessary skills to analyze waves in more depth later. After considering two examples of wave addition, we will proceed to discuss beats.

In Case I below (Fig. D-6) we add two square waves. Note that the second wave (lower diagram) is one half the wavelength of the first (upper diagram). Therefore, the second wave is twice the frequency of the first one. Remember our relation $v = \lambda f$. This tells us that the product of the wavelength and frequency is a constant, the speed of the wave in the particular medium. If you halve the wavelength, you double the frequency. Recalling the analogy with money, if you have 4 quarters, you have a dollar. If you halve the number of coins to 2, you double the amount of each coin. You need two fifty-cent pieces.

To readily find the sum wave, divide the waves into 4 parts or sections. In the first quarter, both top and bottom waveforms have displacement values of 1. Therefore, the sum displacement is 1 + 1 = 2. For the second quarter, we have the top wave at 1 and the bottom one at –1. The plus 1 and minus 1 cancel. We get 1 – 1 = 0. For the third quarter, we have –1 + 1 = 0. Finally, for the last quarter, we have –1 –1 = –2. Think of plus values above 0 as being "in the black," while negative values mean we're "in the red" (in debt).

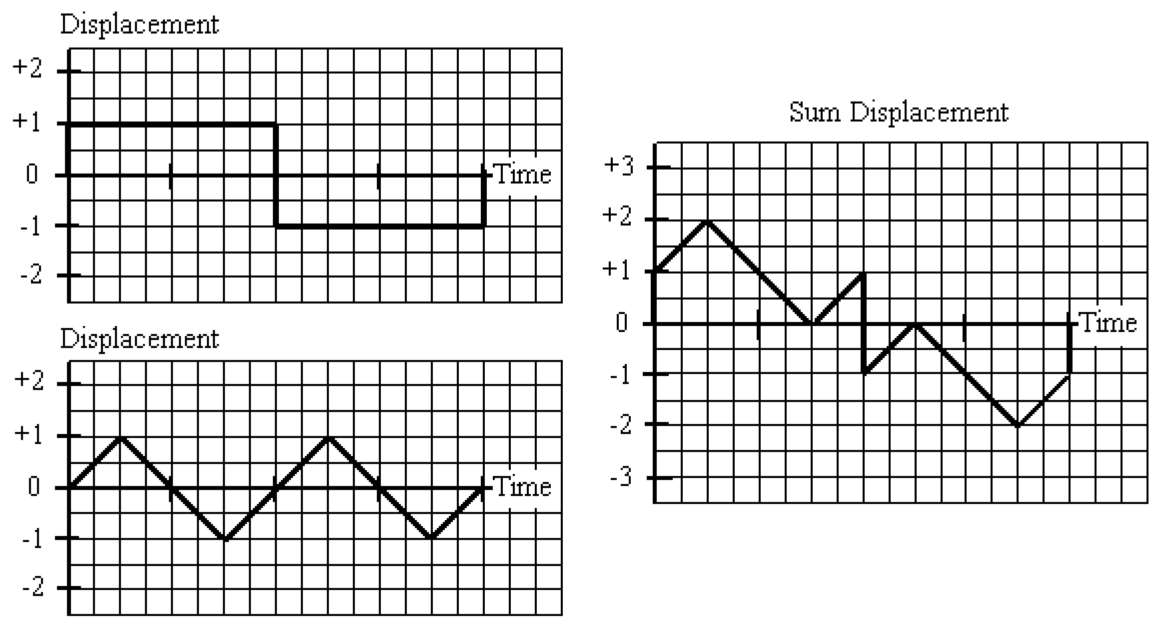
Fig. D-6. Wave Addition: Case I.



Case II is given in Fig. D-7. We are to add a square wave to a triangle wave. Note that the triangle waveform has one half the wavelength as the square wave, which is above it. The triangle wave therefore has twice the frequency. It goes through two complete cycles or patterns in the time it takes the top waveform to go through one cycle of a crest and trough.

The first wave (upper one) is at +1 for the first half and –1 for the latter half. Consider the first half of the triangle wave and add +1 everywhere. This is equivalent to raising the entire first half of the triangle up by 1. The second half of the triangle waves gets lowered by 1 since our square wave is at a minus 1 for the second half of its cycle. Study the sum displacement. Visualize the first half of the sum as a raised triangle. Visualize the second half of the sum displacement as a depressed triangle.

Fig. D-7. Wave Addition: Case II.

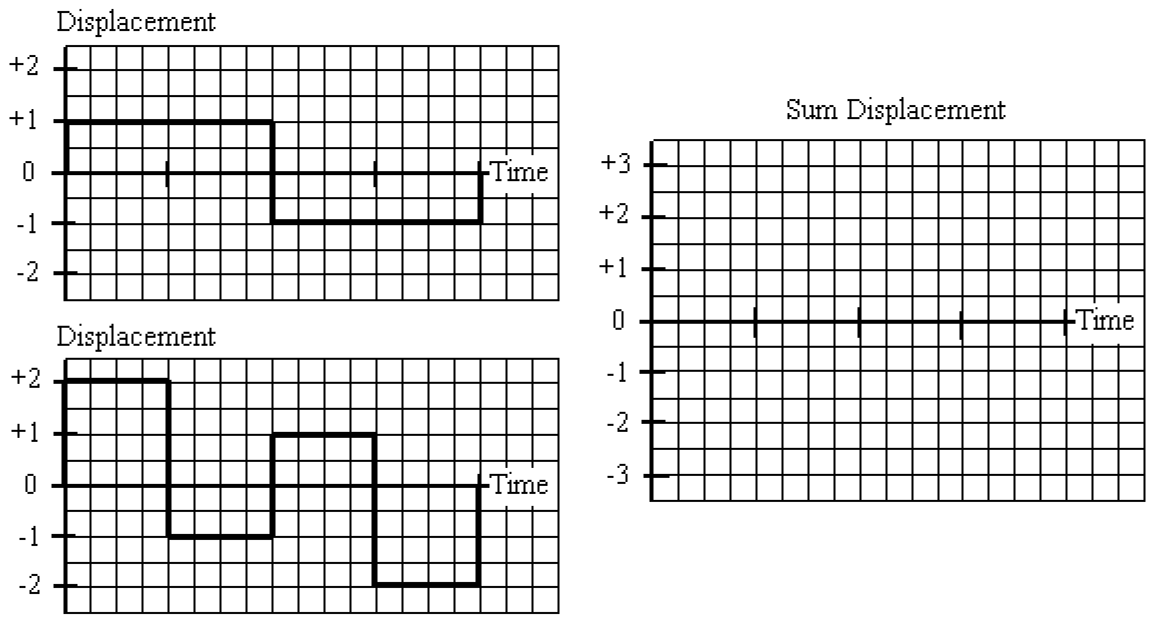


You can add any two waves if you have enough patience. Look at each unit of time above. Along the time axis (horizontal) there are 16 divisions in our graphs, for the waves under consideration. Consider the 3rd division along the horizontal axis. The upper wave has a value of 1 there and the lower wave has a value of 0.5 or 1/2. Therefore, the sum displacement has a value of 1.5 or 3/2 at the 3rd division. Since each vertical division is 1/2, you need to count up 3 divisions (or boxes) to find the 1.5 mark. You need to be a little careful at some places like the beginning. The square wave shoots up from 0 to 1. Well, since the triangle wave is 0 here, the sum wave will go from 0 + 0 = 0 to 1 + 0 = 1.

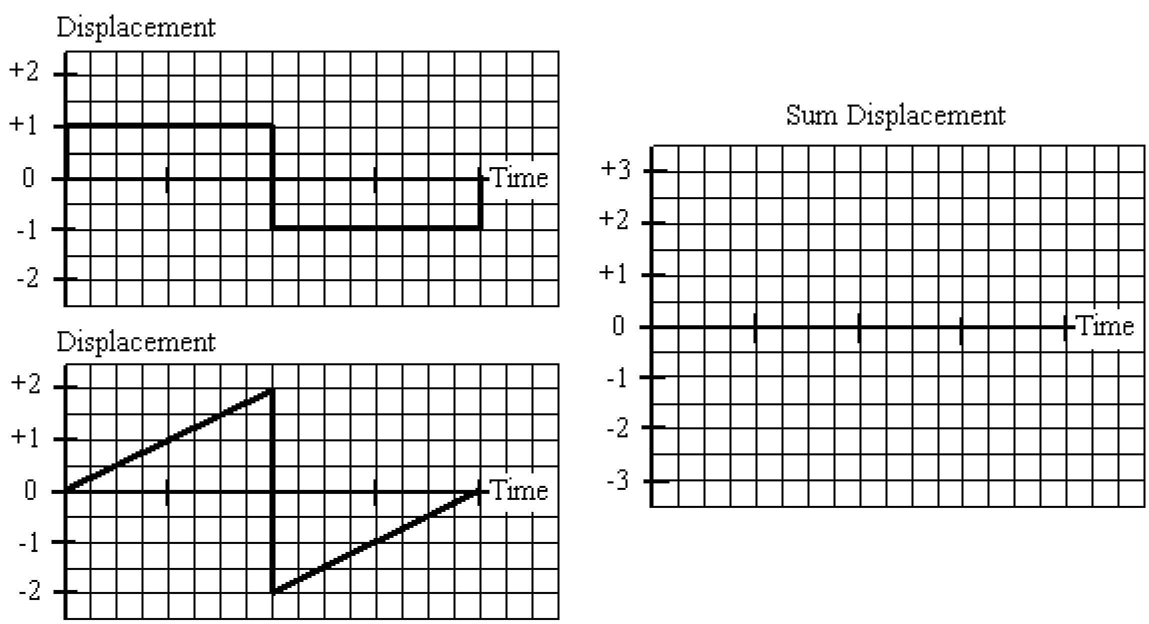
See if you can determine the sum displacements for the cases in Fig. D-8.

Fig. D-8. Practice with Addition. Sketch in the Sum Displacement for Each Case.

Practice Case I.



Practice Case II.



We now turn to the subject of beats. Consider two pure tones very close in frequency. Let the sources of these waves start oscillating in phase. Our sources might be two electronic oscillators sending signals to amplifiers and then speakers. Since the waves do not have exactly the same wavelength, after several cycles, the peaks drift out of alignment. There comes a time when a crest from one is aligned with a trough from the other. The waves, starting with constructive interference, now experience destructive interference. However, after awhile, the waves are once again in phase. This drifting in and out of phase is heard as a pulsation, a pulsating tone. We call these pulsations *beats*.

An analogy of drifting in and out of phase can be found where two people are walking with different step sizes. They can start off walking with each moving right foot forward. Shortly, the right foot of each will be doing something different. After awhile, when one person's right foot steps forward, the other person's left foot will be stepping forward. Then awhile later, we can find an instant when both right feet will step forward at the same time. Try observing this when you walk with a friend sometime.

Playing two sine waves (pure tones) with similar frequencies f_1 and f_2 results in beats. You hear the average frequency pulsating at a frequency given by the difference. You subtract the smaller frequency from the greater one to get this beat frequency.

Table D-2. Playing Two Similar Pure Tones with Frequencies f_1 and f_2.

Frequency of Tone Heard	Average	$\frac{1}{2}(f_1 + f_2)$
Beat Frequency	Difference	$f_2 - f_1$ where $f_2 > f_1$

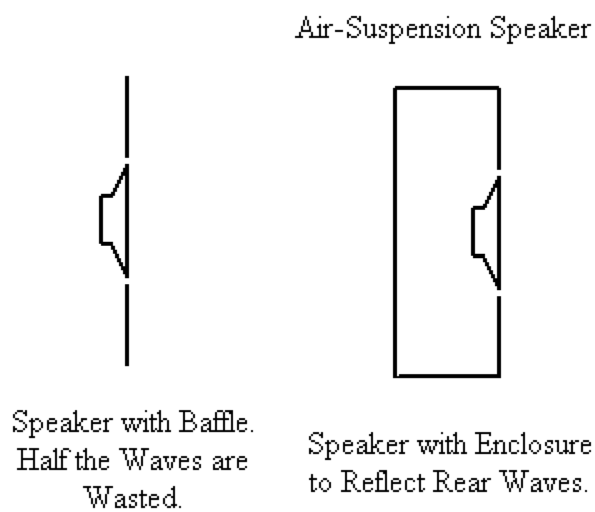
Let's do an example. Let one frequency be 440 Hz (f_1) and the other be 444 Hz (f_2). You hear the average. That would be $\frac{1}{2}(f_1 + f_2) = \frac{1}{2}(440 + 444) = \frac{1}{2}(884) = 442$ Hz. This 442-Hz tone pulsates at a beat frequency of $f_2 - f_1 = 444 - 440 = 4$ Hz, i.e., 4 pulsations per second.

Musicians can use beats in tuning instruments. They know they are close when they hear their instrument "beating" with a reference tone. In fact, you don't even need talent to hear beats. Tuning without some reference, using a sense of *perfect pitch* is rare. Very few musicians have so called perfect pitch, a gift where they can call out the name for any single note they hear and know if its pitch is off a little. On the other hand, matching two tones using beats requires one to adjust the instrument until the beats stop! When the beat frequency decreases, you are getting closer. For example, a beat frequency of 1 Hz (one beat per second) means you are very close. The piano however is sufficiently complex that it takes a skilled piano tuner to tune all the strings properly. We will see why in a later chapter.

Basic Speaker Design

We started this chapter by looking at applications of the law of reflection. Then, the last phenomenon we investigated dealt with interference. Speaker design incorporates the principles of reflection and interference. Diffraction is also relevant. The baffle we saw in the previous chapter is there to prevent waves from diffracting around to the other side where the waves can interfere destructively. Fig. D-9 illustrates how reflection plays a role in speaker design. The speaker is enclosed in a box so that the rear waves can reflect off the back wall. The reflection adds energy to the vibrating membrane.

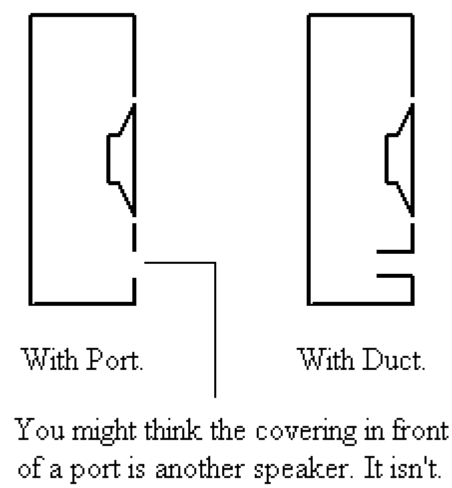
Fig. D-9. Simple Speaker Design.



If a port or duct is added (see Fig. D-10) the mass of air in the enclosure begins to undergo vibrations as a whole, in addition to supporting the sound waves. This is an example of resonance. The mass of air in a large cavity can swish around. It does so at a low frequency since large air masses are actually displaced. This type of system is called a *Helmholtz resonator*. An empty gallon jug of apple cider is another example of a Helmholtz resonator. See Fig. D-11. Blow across the top to hear the low resonance frequency - a nice bass tone.

In the speakers of Fig. D-10, internal reflections (reflex) prevent half the waves from being lost. The other effect is the swishing of the air. Since the resonance frequency for the swishing air is low, you get enhanced bass when speaker frequencies are near resonance. In summary, the cavity reflex enhances the bass.

Fig. D-10. Bass-Reflex Speakers.



Check out the following Helmholtz resonators: one from the late 1800s based on the original 1850 design and a modern one.

Fig. D-11. Helmholtz Resonator

Courtesy acultivatednest.com

--- End of Chapter D --

E. Modulation

Ovenbird, Courtesy Photographer Dr. Dan Sudia

WOXL Van and Oldies

Photo by Prof. Ruiz, December 23, 2002

We now turn to modulation. The idea is simple, yet its use in acoustics gives us an enriching variety of sounds. *Modulation* simply means change. We know from experience that too much of any one thing can get boring. We like variety. Experts warn that if a teacher is lecturing in the same fashion for more than 12 minutes, then the class is gone - bored. You must move around, change pace, or do something different.

Musicians will often change how loudly and softly they play. Musical lines are shaped into phrases much like sentences. They rise and fall with "inflection." Composers change keys in a composition - such changes are actually called modulations. In everyday life, people change the clothes they wear. Most people vary the food they eat from night to night. Can you think of other examples of change?

We have seen that sound has three basic physical characteristics, which also have perceptual counterparts. These are amplitude (loudness), frequency (pitch), and timbre (waveform). This chapter is divided into three sections where we investigate modulation of each of these fundamental wave features.

Amplitude Modulation

The easiest wave characteristic to consider first is the amplitude. This physical feature corresponds essentially to the perception of loudness. Remember that the amplitude is a measure of the wave disturbance relative to equilibrium. If there is no wave, i.e., no sound, then the air is at its normal equilibrium pressure everywhere.

We represent equilibrium by drawing a horizontal line. This straight line indicates zero disturbance. The medium is at equilibrium everywhere. It's like a quiet lake. The water surface is flat. It is still and there are no waves.

A simple sine-wave disturbance about equilibrium consists of crests (above the equilibrium line) and troughs (below the equilibrium line). A sketch of such a wave is given in Fig. E-1.

The definition of amplitude is a measure from equilibrium to the maximum height of a crest. Note that all the crests in Fig. E-1 have the same height. Therefore, the amplitude of the sine wave in Fig. E-1 is constant. The wave does not increase in loudness but stays at a steady volume.

The wavelength (distance between crests) is also constant. Therefore, the frequency is constant too. So we hear the same pitch. Of course, the timbre is the same since from cycle to cycle we have the same repeated waveform. The timbre or waveform for the wave in Fig. E-1 is a sine wave, the simplest of all periodic waves.

Fig. E-1. A Simple Sine Wave Before Modulation.

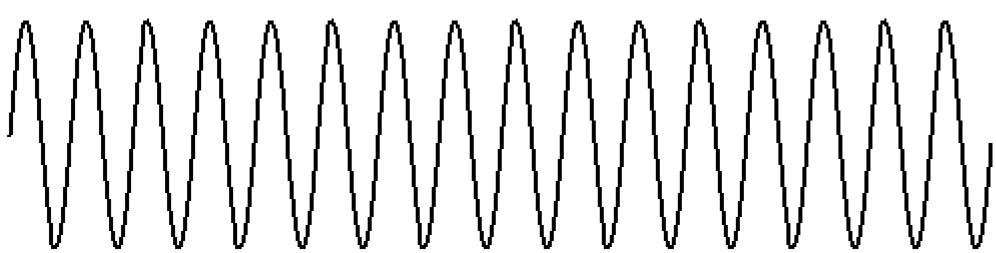


Fig. E-2b is an amplitude-modulated sine wave. It results from taking the sine wave in Fig. E-1 and varying the amplitude according to the sketch in Fig. E-2a. Imagine sliding the sine wave of Fig. E-2a down so that it fits snugly on top of the amplitude-modulated wave of Fig. E-2b. Fig. E-2a tells us how the amplitude is changing.

We note two things here. First, the amplitude of Fig. E-2a changes smoothly, but it is a new sine wave. Second, this sine wave has a frequency, which we call the *modulating frequency,* that is much lower than the frequency of the basic wave (Fig. E-1). Compare the wavelength of Fig. E-2a with the wavelength of Fig. E-1 to confirm this.

Fig. E-2a. Amplitude Change for Amplitude-Modulated Sine Wave in Fig. E-2b.

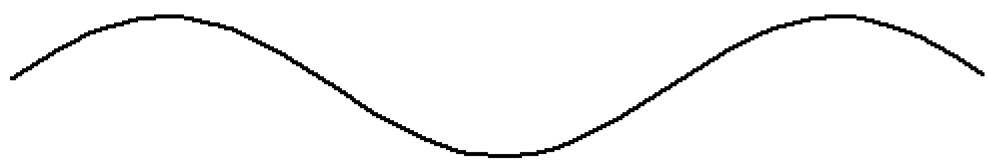
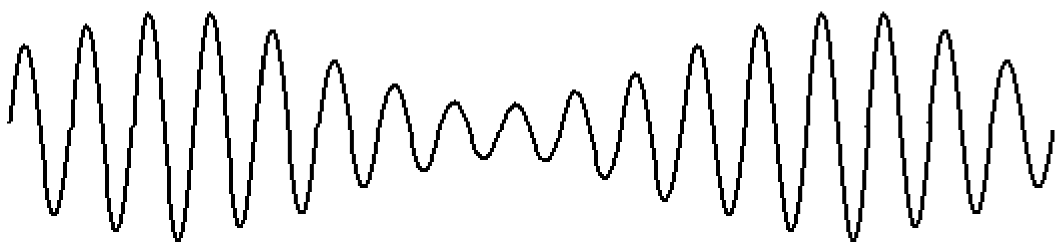


Fig. E-2b. Amplitude-Modulated Sine Wave.



Therefore, the waves of Fig. E-1, Fig. E-2a, and Fig. E-2b are related. The basic wave found in Fig. E-1 is called the carrier wave or simply the *carrier*. It is the wave that experiences a change imposed on it. The wave in Fig. E-2a is called the modulator wave or simply the *modulator*. It describes how the changes are made. When this modulator is applied to the amplitude, the amplitude changes in step with it. The result is the amplitude-modulated wave, Fig. E-2b.

An easy way to keep all of the above straight is to image a simple flute playing a constant tone from your CD player. We can assume that the flute produces a sine wave. You hear the same pitch and loudness as the flute continues with the same note. Fig. E-1 can represent the flute tone. This is the carrier wave. Then you take your hand and turn the volume control up and down in a natural way (sine wave). Your hand motion is described by the sine wave in Fig. E-2a. This is the modulator wave. Note that since your hand can't wiggle back and forth very fast, this modulator wave has a long wavelength and therefore low frequency. The frequency is so low that you never hear sound when you wiggle your hand back and forth. The result you hear is the flute tone with pulsations in loudness. After all, you are changing (modulating) the amplitude. The resulting wave is described by Fig. E-2b. Musicians call this pulsating tone *tremolo*.

In AM Radio broadcasting, a radio wave is sent to your home. A radio wave is not a sound wave; it is a wave involving electric and magnetic interactions. The frequency of these waves is very high, like millions of cycles per second. The frequency of a station's transmitted AM radio waves is obtained from the number the radio announcer constantly reminds you to tune in to. For example, *WISE 1310* broadcasts at 1310 kHz (kilohertz).

The lower-frequency sound information is coded in the amplitude variation of this radio wave. In this analogy, Fig. E-1 represents the radio wave, Fig. E-2a represents a sound frequency such as a whistling tone. As the radio travels through space, there is no sound. The sound information is transmitted as the amplitude variation. It is said that the radio wave carries the sound information. That's why we call the basic wave the carrier.

Radio carrier waves can travel in the vacuum of outer space. They are a very special kind of waves called electromagnetic waves. Electromagnetic waves do not need a medium in which to travel. Light is also an electromagnetic wave.

The sound information is extracted by your AM radio. Your radio gets the audio information from the variations in the amplitude. You have probably guessed by now what AM stands for - *amplitude modulation*.

Think carefully about our two analogies. In the first case, your hand-wave frequency modulates a sound wave. In the second application, a sound-wave frequency modulates a radio wave. The modulator effect on a carrier's amplitude can be understood by multiplying waves.

We have seen how to add waves. We will review wave addition and then take up wave multiplication. Fig. E-3a reviews wave addition. Think of the addition in 4 stages. For the 1st stage or quarter, both the top wave and bottom wave in Fig. E-3a have displacements of 1. Therefore, the total displacement for that region is $1 + 1 = 2$. Then we combine -1 with $+1$ for the 2nd quarter to obtain $-1 + 1 = 0$.

For the 3rd and 4th quarters the lower wave is 0. We can just sketch the top wave for the sum here. Adding nothing leaves the wave alone. Note that the amplitude of the top wave in Fig. E-3a is 1. The displacement of the wave changes as we go through ups and downs (crests and troughs). But the definition of the amplitude calls for a measure from the equilibrium to the highest point. If the amplitude (maximum displacement) changes from cycle to cycle, we then have amplitude change or modulation.

For multiplication, we multiply the appropriate pairs of numbers instead of adding. For the first and second quarters, the upper wave is untouched since the lower wave has a value of 1 in this region. Refer to Fig. E-3b. Multiplying by 1 doesn't change anything.

However, the situation is quite different for the third and fourth quarters where the lower wave has a value of 0. Multiplying by zero gives zero. The upper wave is wiped out in this latter region. The lower wave acts like a gate. It allows the upper wave to survive when the lower wave has a value of 1 (gate is open).

Then, when the lower wave has a value of 0 (gate is closed), there is no surviving wave. This effect is called *gating*. It's important in music synthesizers. When you press a key, you get something; when you release the key, the tone is destroyed.

Fig. E-3a. Wave Addition.

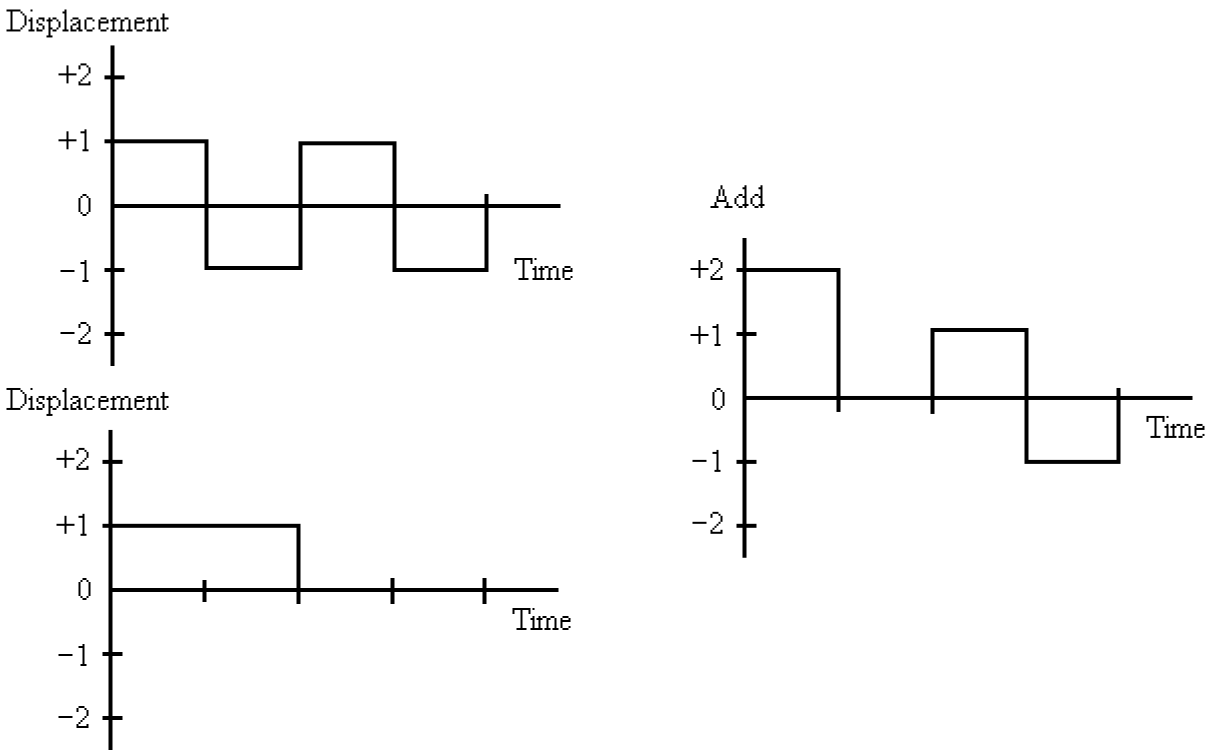
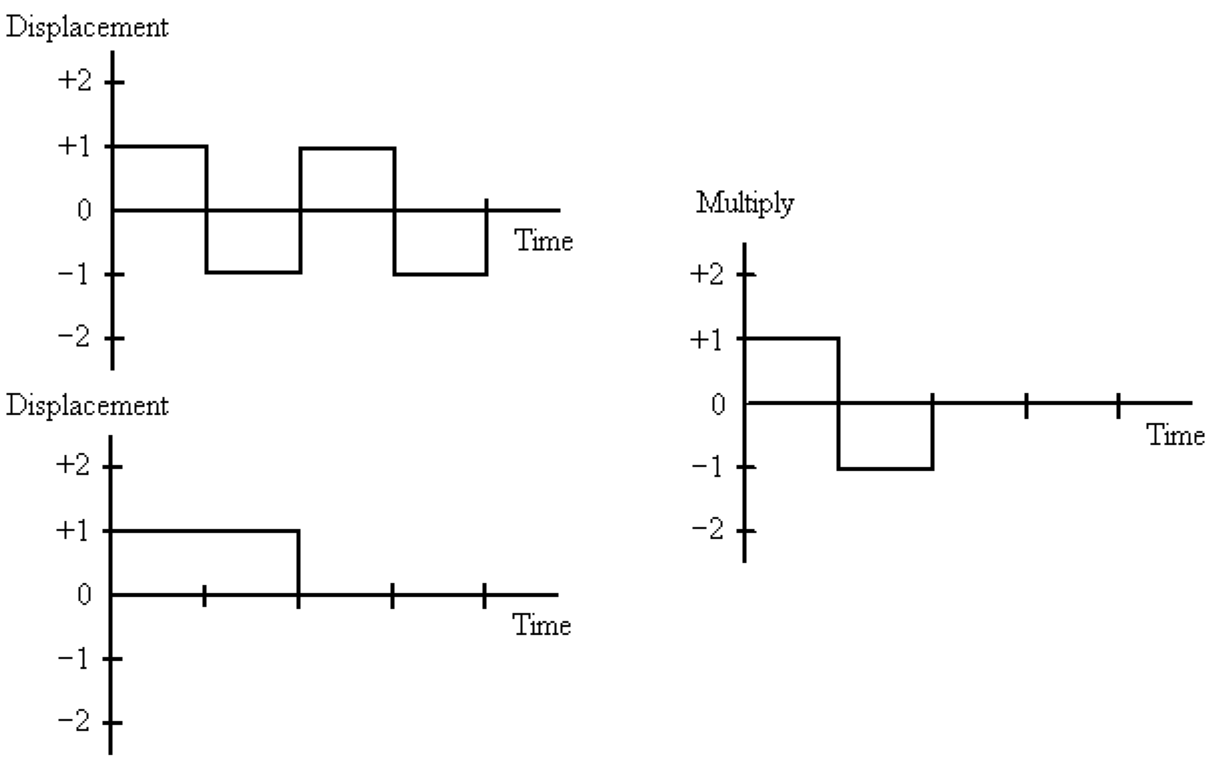


Fig. E-3b. Wave Multiplication.



We will now relate wave multiplication to amplitude modulation. We will use the pulse train wave as the carrier and a triangle wave as the modulator. See Fig. E-4 below. Don't be concerned that the reference line is at the far bottom here. In the analysis of waves, the reference can be freely chosen to simplify the way of looking at the problem. You might say we introduce an offset, i.e., offset the horizontal line at our leisure.

In the electronic production of waves, this is referred to as a voltage offset. Physicists play the same game analyzing falling objects. Sometimes, the ground is called the zero point for height. If a ball falls to a table, then the table is taken to be the zero point.

Fig. E-4a. Pulse-Train Carrier and Triangle Modulator.

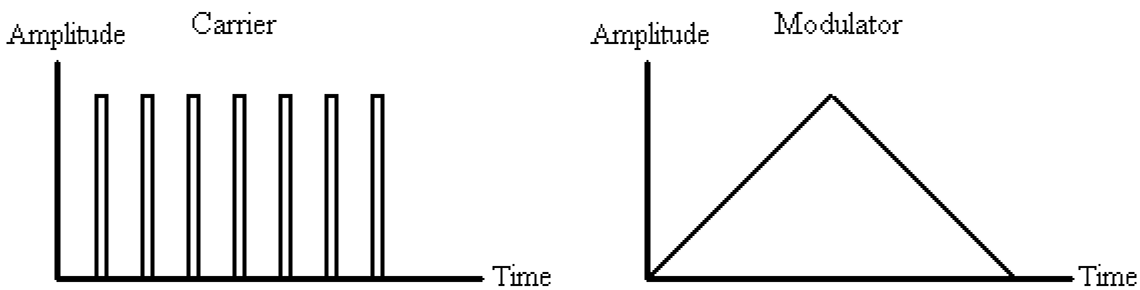
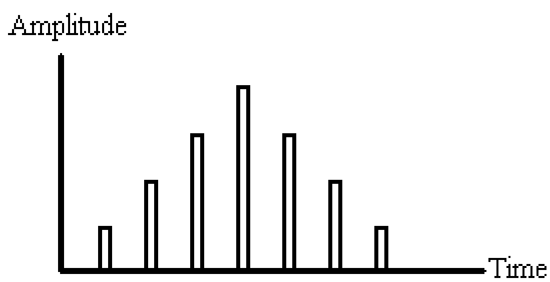


Fig. E-4b. Result When the Modulator Modulates the Amplitude of the Carrier (see above).



It is too tedious to multiply the carrier and modulator in Fig. E-4a point by point to arrive at the amplitude modulation in Fig. E-4b. Rather, let's reason it out. The triangle wave starts out at zero. This means the carrier is knocked out at the beginning since zero times anything is zero. Now look at the middle of the above triangle. The triangle is high. Consider this top point to be 1, 100% full strength. The effect on the carrier is to leave the carrier alone here: one times the carrier equals the carrier. Between the beginning and this high point, the modulator gradually increases. So we sketch the carrier's crests gradually increasing from the beginning to this point (see Fig. E-4b). The reverse happens for the decreasing side.

Consider a "cookie-cutter" analogy. Take the modulator shape in your mind and place it over the carrier. Imagine the carrier to have crests of "dough." Then slice the "dough peaks" with the modulator "cookie cutter." You have modulated the amplitude.

The previous modulation can be demonstrated by playing a tape of a pulse-train wave with your tape deck. Imagine a somewhat harsh tone with a fixed pitch. You then manipulate the volume control, i.e., modify the loudness. You gradually turn the volume up from the zero level to some maximum, then turn the volume off.

However, you are not turning the knob in the most natural wave (the sine wave or sinusoidal motion). Your turning is described by the triangle wave.

Try moving your hand across a sheet of paper with a pencil in it. Begin moving your hand up and down in a "triangular" sort of way. See if you can do it. Remember to keep the sweep speed across the paper a constant rate. Your sweeping hand will trace out the wave the way the oscilloscope draws pictures of waves.

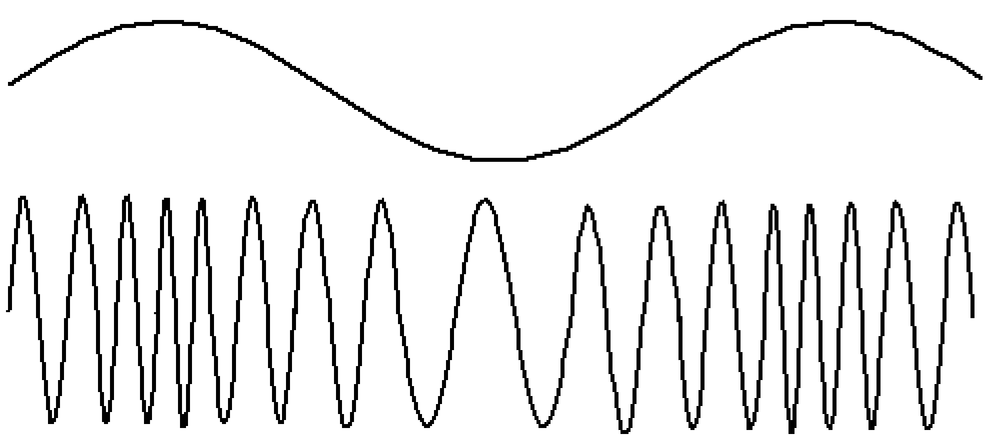
When the oscilloscope gets to the far right in its sweep, it starts over. Its electronics is designed so that it can retrace its steps for a nice periodic tone. The retrace gives us a steady picture on the screen. However, when the input wave is very low in frequency, like a typical modulator frequency, you can often see a bright dot wiggling across the screen. For faster frequencies, the retrace gives a complete picture due to the lit screen glowing for a split second and our persistence of vision. Persistence of vision is our retention of the previous image seen for a fraction of a second. It's what makes movies possible.

Frequency Modulation

We will now apply the same modulator used in Fig. E-2a, but not to change the amplitude of the sine wave in Fig. E-1, but to change its frequency. The resulting frequency modulation is given in Fig. E-5. There isn't a simple way to get this modulated picture by some arithmetic manipulation. We simply sketch the result.

When the modulator wave is high in its value (crest), this means increase the frequency, i.e., shorten the wavelength. Before (with amplitude modulation) we increased the amplitude, now we increase the frequency. When the modulator dips to a trough, we decrease the frequency, i.e., lengthen the wavelength. So we can also call this wavelength modulation. FM radio encodes its information this way. FM stands for *frequency modulation*. With FM radio, the carrier is once again a radio wave.

Fig. E-5. Modulator (Upper Wave) Modulating Frequency to Get FM Wave.



We continue our application in the area of radio broadcasting. The modulator wave in Fig. E-5 can represent the sound wave we are to broadcast. The radio station is playing a tape with a perfect sine wave. It can be the sine wave of a whistling constant tone. Whistlers can produce good sine waves. The information is encoded by varying the frequency of the carrier radio wave.

The carrier waves for frequency modulation are greater than those used for amplitude modulation. For example, our classical station, *WCQS*, broadcasts at 88.1 million hertz. The metric prefix for million is Mega, so we can write this as 88.1 MHz (Megahertz). Our earlier example for an AM station was *WISE* at 1310 kHz, which is 1,310,000 Hz or 1.31 MHz.

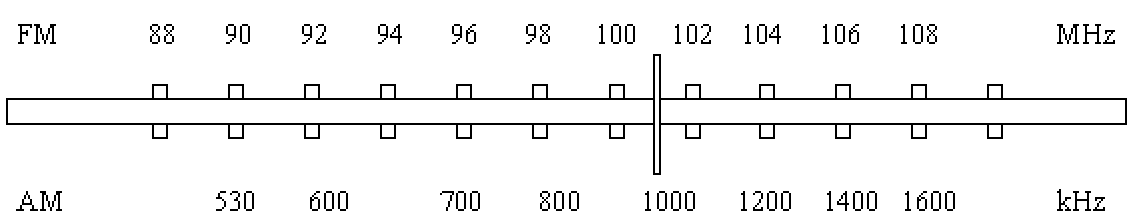
The frequency 88.1 MHz is reserved for *WCQS*. Also, some frequencies a little less and a little greater are also reserved because the 88.1-MHz radio wave will be frequency modulated. This band of frequencies is called that station's *bandwidth*. The center is 88.1 MHz. Another example of an FM station is *Kiss Country* at 99.9 MHz (*WKSF*). If country is not your style, try its neighbor *Mix 96.5* (*WOXL*) at 96.5 MHz on the radio spectrum for a mixture of music.

We stress that sound waves are not broadcasted. Radio waves are sent to your home. The sound information is encoded in the frequency modulation. The station's radio wave undergoes variations in frequency at the rate of the audio frequency it's carrying. For example, a 400-Hz sound wave is encoded by WCQS by varying the frequency of the 88.1 MHz carrier at the rate of 400 Hz. Your FM radio extracts this information using an FM radio circuit.

Why do you think FM radio reception is better than AM? Which part of the carrier is more susceptible to distortion or deterioration - the amplitude or frequency? Remember that the valuable information is encoded in the amplitude for AM broadcast and the frequency for FM broadcast. HINT: When you comb your hair in your car during a dry winter morning, static readily affects the amplitude.

A sketch of the front panel of an AM-FM Radio is given in Fig. E-6. Note that the FM numbers are given in MHz while the AM numbers are in kHz. Also note that the numbers are not equally spaced on the AM scale. That's because the same dial control is used for both and the spacing reflects the numbers dictated by the AM circuit. Did you ever notice this? Then again, you might have a digital radio and never deal with turning a tuning knob.

Fig. E-6. FM and AM Radio.



You can actually do a frequency-modulation yourself. On a synthesizer, you vary the control that alters the frequency. This is the pitch-bend control. First, press a key to play one note and keep the key held down. Then vary the pitch-blend control just a little at some periodic rate. Your frequency will increase slightly and then decrease in a periodic fashion.

If you do not have a synthesizer, you can sing a tone and try varying the frequency with your voice. Musicians call this *vibrato*. It sounds like a quiver.

We will shortly see a variety of examples of frequency modulation from everyday life. We now give the picture for a triangle wave modulating the frequency of a pulse train. In Fig. E-4 we encountered the result when a triangle wave modulates the amplitude of a pulse train. Fig. E-7 provides us with the same carrier and modulator as Fig. E-4. However, now the modulator works on the frequency rather than the amplitude. Employ the same reasoning as we used in understanding Fig. E-5. Higher frequency is evident by shorter wavelength. When the modulator wave has a high value, look for shorter wavelengths in the carrier.

Fig. E-7a. Pulse-Train Carrier and Triangle Modulator.

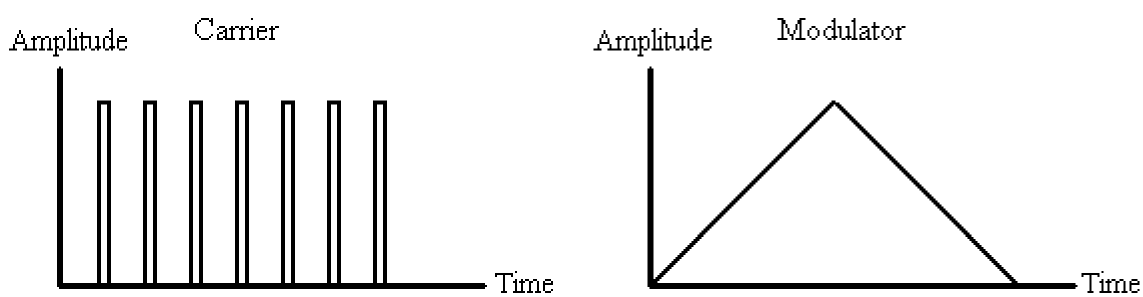


Fig. E-7b. Result When the Modulator Modulates the Frequency of the Carrier (see above).

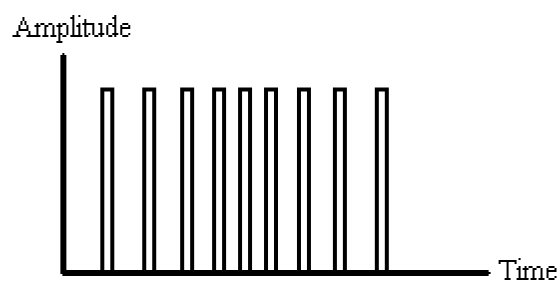
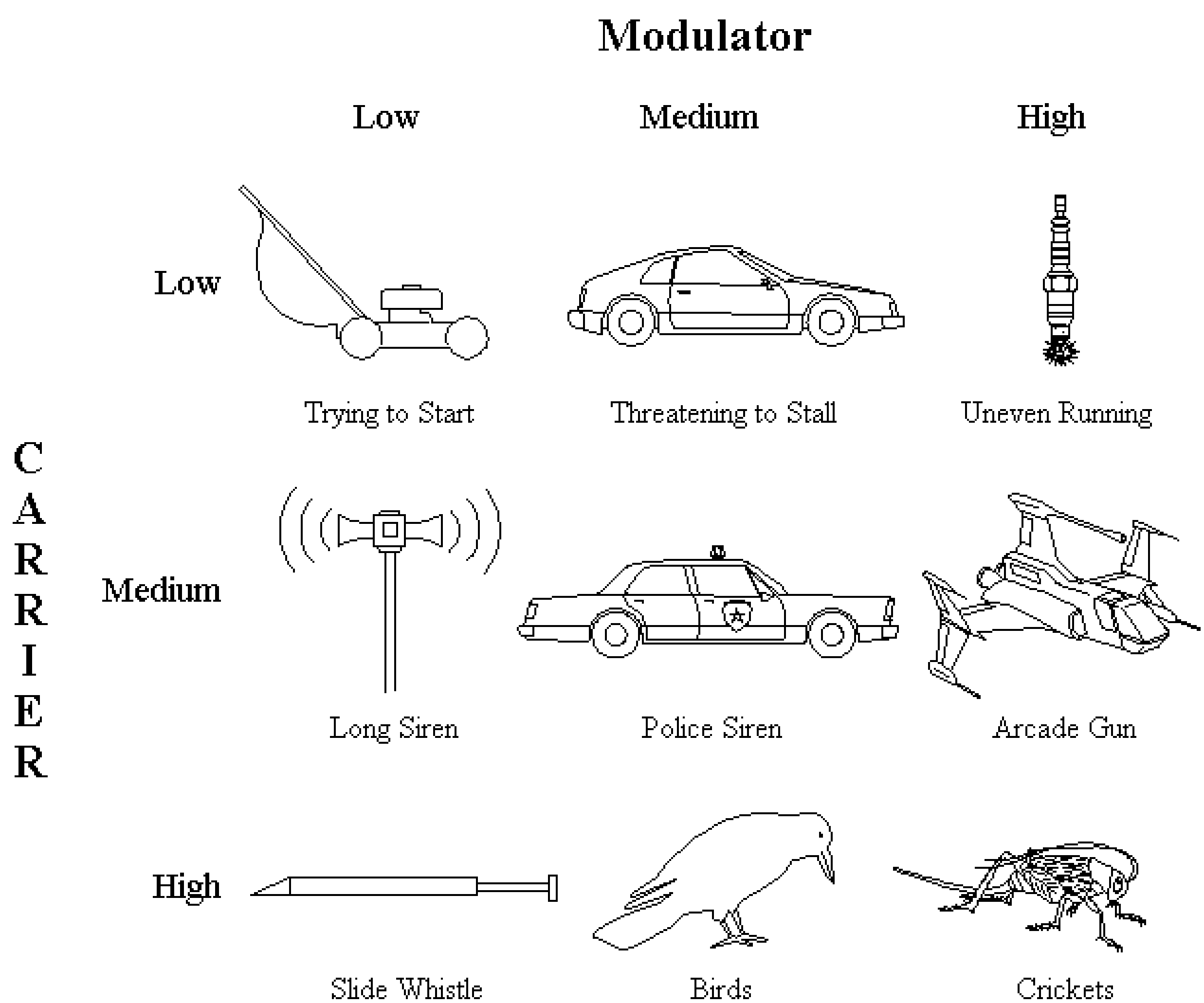


Table E-1 below gives a variety of sounds from everyday life. These sounds can be approximated by frequency modulation. The carrier wave is in the audible range. Low means bass, pitches near the bottom of the piano, medium is around the middle of the piano, and high is near the top. The carrier should be a square wave for the best results overall. It gives the carrier some richness, yet not too much.

Modulator frequencies are very low, below the threshold of hearing. Use your vibrating hand as a guide. Shaking at a low frequency is about 1 Hz (once per second). A medium modulator is a quiver at about 4 Hz. A high modulator is changing at a greater frequency, perhaps as high as 10 Hz or more. This is a rough guide in interpreting Table E-1.

Table E-1. Frequency Modulation in Everyday Life.



Consider a low modulator changing the frequency of a low carrier (Table E-1). The example in the table for these conditions is trying to start a lawn mower. As you pull on the cord, the engine starts to increase its rotational speed. But often it doesn't start and begins to slow down. You keep pulling the cord every second or two (the low modulator) and the engine increases and decreases its turning rate in the slow realm (low carrier frequency) until it starts.

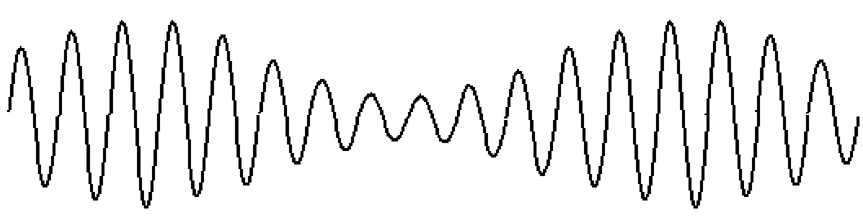
For the case of a medium carrier, we use a running car about to stall. The engine is turning slowly, in the low-carrier audio spectrum, but higher than the lawn mower. Low carrier simply means low audible frequencies. There is variation in the engine speed as it slows down threatening to stall, then speeds up. Imagine this variation (medium modulator) happening at a rate faster than your repeated pulling on the lawn mower cord. But remember that the medium modulator has a frequency of still only about 4 or 5 Hz.

On the other hand, the low carrier frequency is much higher than this. Audible frequencies are higher than the modulator frequencies in Table E-1. Finally, consider the uneven running of an engine (still at a low carrier value of rotations per minute, i.e., rpm), however, with more rapid variations (higher modulating frequency).

The next group of three (going across from left to right) in Table E-1 is the middle group with a medium carrier frequency. The first example is the long siren. It takes awhile (low modulator frequency) to sweep through different frequencies of the siren carrier wave (medium or mid-range carrier frequency). Once the author was freaked out as a child in Camden, NJ when he suddenly heard at night a slowly-varying siren from the desolate Philadelphia shipyard nearby. Increasing the rate of change for our siren brings us to the police siren. The changes are more rapid (medium modulator) and the carrier is once again in a middle frequency range, like our other siren (which makes it easier to hear). Changing a medium carrier very rapidly mimics the sounds in arcade or video games.

The final group of three at the bottom of Table E-1 takes us to high carrier frequencies. These are high pitches. The slide whistle is an example of changing a fairly high-pitched tone slowly. The change occurs as the player slowly moves the sliding part of the whistle in and out. If the variation is more rapid, we obtain sounds that approximate singing birds. We may have to start with a carrier even higher in frequency than the slide whistle. Our table is approximate and it is understood that you may have to search for the best frequencies in each case. If we vary the high carrier really fast (high modulator), the sound resembles crickets.

Some of the sounds in Table E-1 are very effective when demonstrated electronically using a square wave for the carrier. The actual waveform for the modulator is not that critical. However, a triangle wave works well. The triangle modulator makes nice sliding-whistle tones and sirens. A factor that we didn't mention is the extent of the sweep. When we change a 500-Hz carrier wave once a second, do we sweep up to 505 Hz or 600 Hz every second? If we sweep very little, staying close to the original carrier frequency, the resulting sound is more of a quiver (vibrato). If we sweep a fair amount, we have our siren sounds.



Timbral Modulation

The final basic characteristic to change is the timbre. The pulse train is an excellent waveform to work with. A simple way to vary the waveform is to increase the width of the pulses. In other words, change it into a square wave and back, in some periodic fashion. Note that this is only one way to vary the waveform. It is simple conceptually and electronically. This particular type of timbral modulation is called *pulse-width modulation*. See Fig. 8 below.

When the triangular modulator has a maximum value, we now interpret this to mean the widest pulse widths. Think of the pulse train as an array of buildings. A large modulator value means that we have a wide building at that point. Note that all the buildings have the same height (amplitude) and are equally spaced (wavelength, frequency). It's the widths of the buildings that change. The lower values of the modulator indicate narrower buildings or pulse widths. We decide to let a zero modulator value signify a minimum pulse width rather than no pulse width. Our spirit is qualitative; and remember, there is freedom in choosing the zero-reference point.

Fig. E-8a. Pulse-Train Carrier and Triangle Modulator.

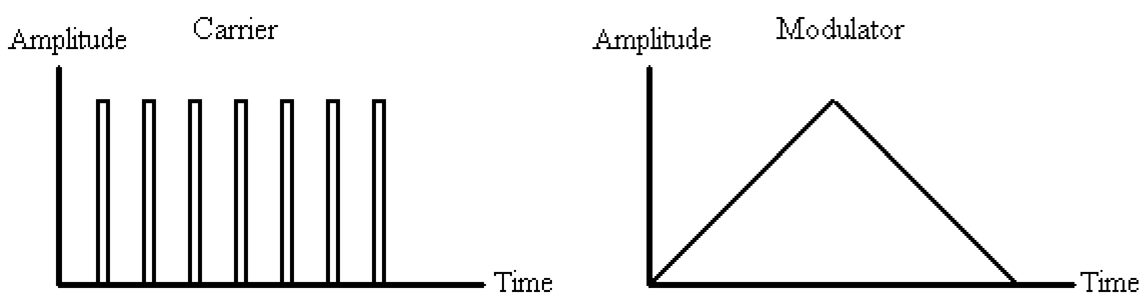
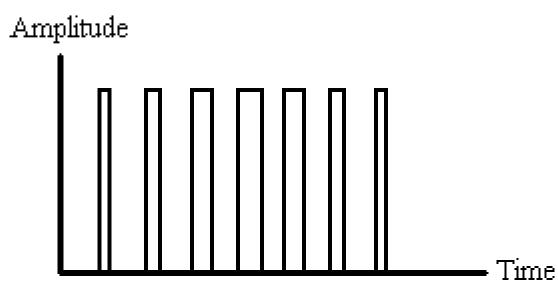


Fig. E-8b. Result When the Modulator Modulates the Pulse Width of the Carrier (see above).



Timbral modulation is heard when the quality of the sound changes in an intrinsic way, not the loudness or pitch, but the very nature of the sound. This occurs dramatically when a trumpet player moves a mute in and out at the bell end while holding a note. Try singing vowel sounds at the same loudness and pitch. As you change each vowel, you change the timbre.

Pulse-width modulation sounds like buzzing bees. We will try to understand why. Similar bees produce similar sounds. We can assume the bees are identical so the waves are identical in amplitude, frequency, and waveform. But different parts of these identical waves reach our ears at any given time. Perhaps a crest (compression) reaches an ear from the 15th bee, while the point 60% of the way up the crest reaches the same ear from the 20th bee, and so on.

This brings us back to the concept of phase. Remember that when two crests overlap, we say the waves are in phase. When a crest meets a trough from another wave, we say the waves are out of phase. The relationship among the waves of our many bees is complicated. There is no simple phase relation. The phases are all mixed. There are all possible cases in between in-phase and out-of-phase. This sound of mixed phases from similar sources has a distinct quality. It's called the *chorus effect*. The bees are singing together like a choir.

The rapid changes of the pulse width in our pulse-width modulation is a crude approximation to superimposing many low-intensity square waves on top of each other. The random shifting phases are approximated to some extent by the shifting width of the pulse wave. Therefore, the pulse-width modulation sounds like a buzzing choir of bees.

The distinct sound of the chorus effect is very different from the sound of a single source. We can pick out a solo instrument such as a violin if it plays something different from the rest of the violins. Therefore, you can usually hear the soloist in a violin concerto against the background of many violins playing in unison. Of course, the conductor keeps the violin section from overpowering the single soloist. Nevertheless, the chorus effect is relevant and likewise plays an important role. The mixed phases of the violins playing in unison are heard as a different sound than a solo melodic line.

The common example of the chorus effect is people singing in unison. They can also recite in unison. A less common example of the chorus effect is the group sounds of cicadas (sih-KAY-duhs).

Fig. E-9. Cicada.

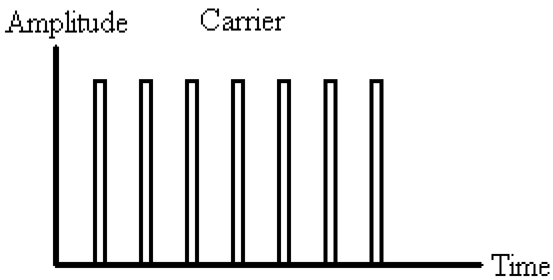
Courtesy www.artsjournal.com

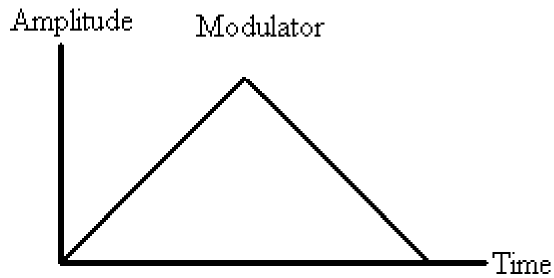
Hordes of cicadas are not around every year. A common type of cicada (see Fig. E-9) returns every 17 years. They live underground sucking nourishment from roots until they are mature. They then come out in the thousands. Males make sounds to attract females. You can turn down a side street and often be surprised at the tremendous cicada chorus effect when you run across hordes of them in trees. This lasts for only a few weeks. The adults die after mating and the new generation waits underground for the next cycle of 17 years.

A group made a riot of sound in Asheville in 1991. The party of the next generation of cicadas was held in 2008. However there is more than one type of cicada. Some return at different time intervals.

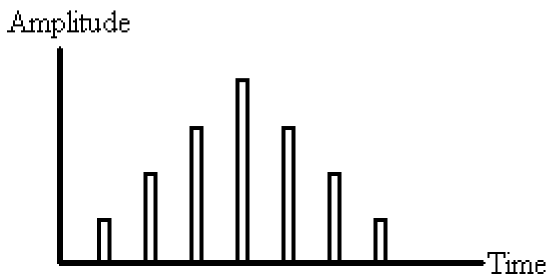
The three basic types of modulation encountered in this chapter are summarized below in Fig. E-10. You will recognize these as figures found earlier in the chapter. They are reproduced here for review and comparison.

Fig. E-10. Three Basic Types of Modulation.

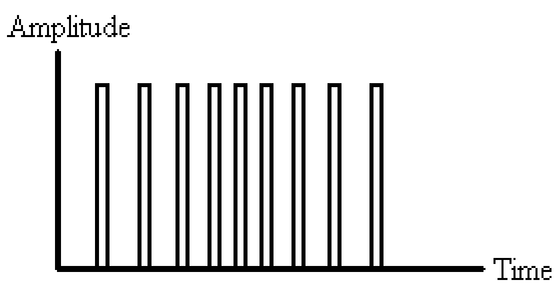




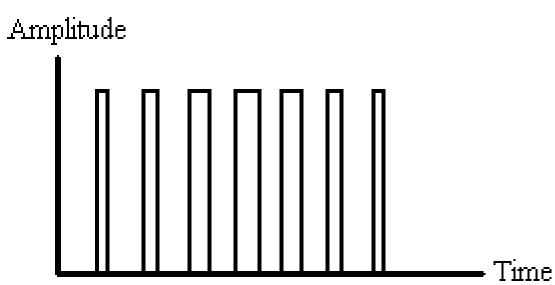
Case 1. Amplitude Modulation



Case 2. Frequency Modulation



Case 3. Timbral Modulation (Pulse-Width Modulation)

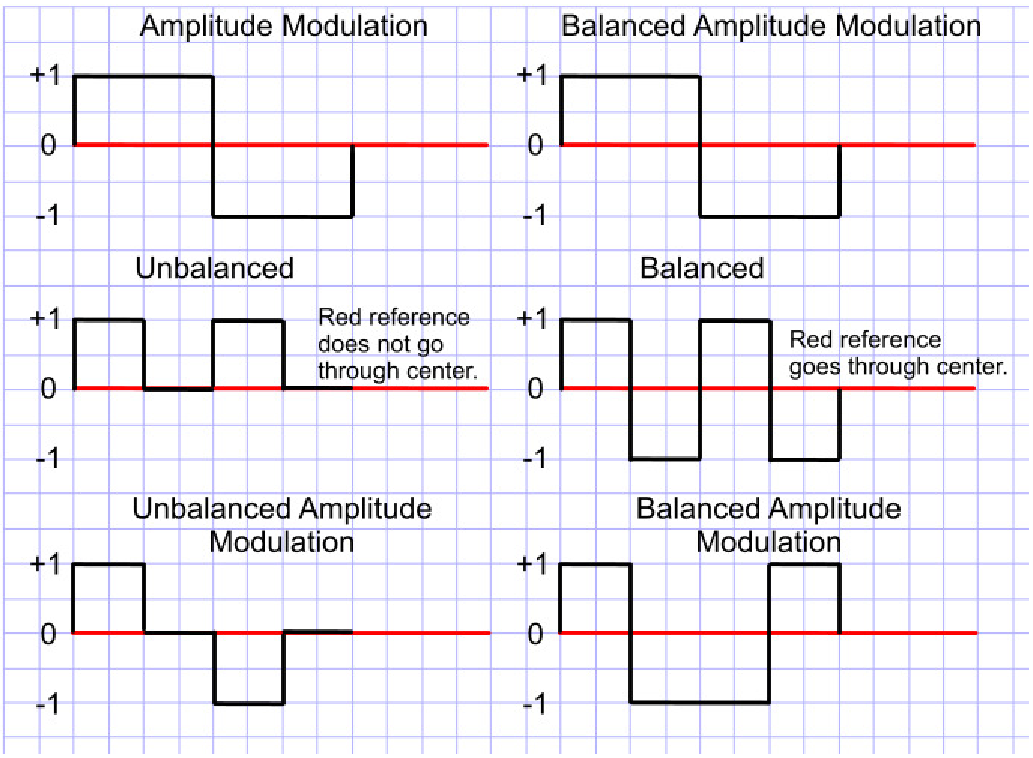


Balanced Modulation

Earlier we mentioned that the vertical position of the wave could be freely chosen. We did that to simply our multiplications so that one wave would be at height 1 or height 0. You still have balanced modulation. However, if we are very careful to have the zero level go through the center of each wave, then we then have balanced amplitude modulation. We will illustrate that here.

Balanced modulation is amplitude modulation where the two waves have the zero-reference line running through the center of each wave. In Fig. E-11 balanced amplitude modulation, or simply balanced modulation, occurs in the right case. At the left, we have amplitude modulation, but the waves are not balanced. To be balanced, the red line at zero must go through the center of each wave.

Fig. E-11. Amplitude Modulation (Multiply in Each Case). Unbalanced (left), Balanced (right).



Here is a summary of the four possibilities for multiplication in balanced modulation. See Table. E-1. We can summarize these multiplications in Table E-2 where HIGH represents crest, a value of +1 and LOW is a trough, a value of -1.

Table. E-1. The Cases of Multiplication.

crest meets crest:	(+1) (+1) = +1
crest meets trough:	(+1) (-1) = -1
trough meets crest:	(-1) (+1) = -1
trough meets trough:	(-1) (-1) = +1

Table E-2. Input A and B with Output.

A. Top Wave	B. Bottom Wave	Output Wave
HIGH	HIGH	HIGH
HIGH	LOW	LOW
LOW	HIGH	LOW
LOW	LOW	HIGH

--- End of Chapter E ---

F. Frequency Ratios

We study in this chapter the fundamental topic of frequency ratios. A frequency ratio is relevant when we consider two tones. Some combinations are pleasant, others are unpleasant. The absolute frequency is not important. It is the relative frequency that is significant. You know this. For example, you can sing a song like *Mary Had a Little Lamb*. Then you can start the song again at a slightly higher pitch. You will unconsciously adjust all the subsequent pitches to their proper relative positions so that the song once again sounds like *Mary Had a Little Lamb*. You have preserved the song. This is called *transposing*.

Pianists are often asked to play the same song starting at different pitches in order to accommodate singers. The pianist is said to *transpose* the song into another key, i.e., another starting point. The word transpose is used because the song is often written in a different key. The musician is not reading the music, not playing where it is written, but transposing to a different key. This takes practice. Of course, if the pianist can play by ear, music is not necessary in the first place.

Over the ages people of different cultures have chosen tones that coordinate well with each other. These groups of tones are called *scales*. We will work with the *western major musical scale* in this chapter. This is the scale you learned many years ago: *Do-Re-Mi-Fa-Sol-La-Ti-Do'*. It's all the music you need to know in order to understand this text. You probably learned it in kindergarten. You see, kindergarten is very important. You learn about scales, among other important activities like relating to others. Someone once said that you learn in kindergarten everything you need to succeed in life.

The musical scale can start on any pitch. You sing *Do* anywhere you like and then proceed to sing the other tones accordingly. Once again, the relative pitch is important for the study of the coordination of these tones. Therefore, we will compare the frequencies by ratios. Ratios are not complicated. Let's use a monetary analogy. You may have $150 and your friend has $300. Well, we only care that your friend has twice as much as you if we are only interested in the ratio. We can say that you have a given amount, which we describe by 1. The "one" simply means the money in your pocket, your "one" pocket of money. Then we say your friend has 2, meaning the equivalent of 2 times what you have. Your friend has "two pockets" of money in a sense. Another person might have 3 or 4 times what you have. We are comparing amounts of money with what you have. The numbers are therefore ratios.

We will proceed first by reviewing the musical scale. We will develop the concept of *musical intervals*. We will develop a trick where we can readily tell which note in the scale is played relative to our reference note *Do*. Secondly, we will develop a technique for measuring frequency ratios. This is the subject of *Lissajous* (LISS-uh-joo) *figures*. Thirdly, we will apply this technique to determine the frequency ratios for the tones in our 8-note musical scale. But we will work backwards at that point. We will first search for tones with simple ratios. Then we will discover that these notes are in our scale. The scale we unfold will be based on perfect ratios. In a later chapter we will learn that, today, our instruments are not tuned perfectly to such a scale. Compromises are made.

Musical Intervals

Let's review a couple of definitions before we investigate musical intervals. Frequency is a measure of how rapidly vibrations occur. The frequency of a sound wave is perceived as pitch. A *musical scale* is built from a set of frequencies called *tones* or *notes*.

Frequency (pitch) - number of vibrations per second (hertz, Hz).

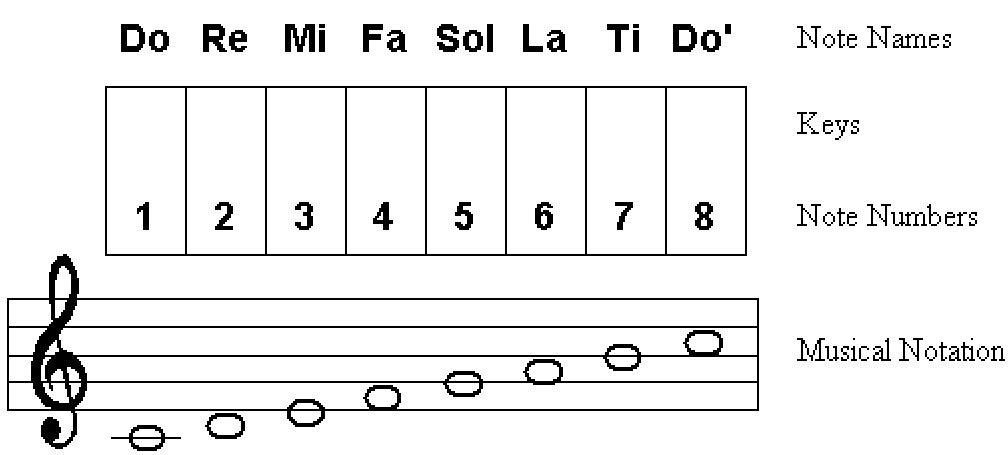
Musical scale - a discrete set of frequencies (tones or notes).

The common major scale is illustrated below in Fig. F-1. There are eight tones in the scale. These are numbered in the figure. It will be convenient for us to refer to a tone by its number from time to time. The first note is *Do*, the second note is *Re*, and so on. We call the last note *Do'* to distinguish it from the first note, *Do*.

You will find musical notation below the keys in Fig. F-1. You do not need to worry if this is the first time you have seen such a thing. Consider it like a thermometer that indicates pitch without using numbers. This "musical thermometer" is called a *staff*. The higher up on the staff, the higher the pitch of the note. In other words, the higher the frequency, the higher the position. But don't think about numbers. The notes, of course, have numbers to define frequencies, but the spacing on the musical staff does not correspond to them in some simple fashion. Remember the AM numbers on our radio. They were spaced unevenly. Such a spacing where equal steps in distance do not correspond to equal steps in frequency is called a *nonlinear scale*.

The beauty of the nonlinear musical staff is that each note of our scale is equally spaced by position, not frequency. The tones alternately fall on lines or spaces. Notice the short line necessary to indicate the first note. The symbol at the left of the staff is called a *clef* symbol. Ours is a *treble clef,* meaning tones in the upper half of the piano.

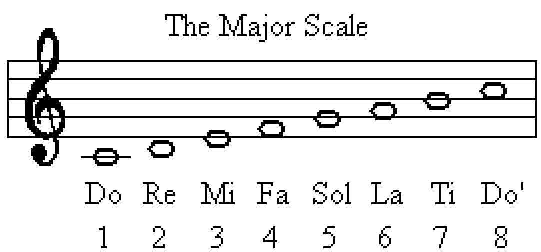
Fig. F-1. The Major Scale.



We take the first note as our reference tone. We will compare the other notes to *Do*. For example we may compare *Sol* to *Do*. Moving from the reference *Do* up to *Sol*, we move up to the 5th note. Musicians say we move up a 5th. If you press these two keys together, you are playing an interval called the 5th. If you choose to play the 4th note (*Fa*) with the reference (*Do*), you are playing a 4th. Going from *Do* to *Fa* defines an interval of a 4th. So an interval is really the "musical distance" between two notes. We will always start on the *Do*.

Since we jump from *Do* to another note, it can be difficult to recognize the upper note if someone plays one for us and asks us to identify it. Music teachers have come up with a technique to help us do this fairly accurately with a little practice. We use the old trick of remembering the start of a song for each of the jumps. Then, when we hear a jump, we scan through the songs in our head until we find a match. The actual practice of doing this is called *ear training*. Table F-1 below lists the 8 intervals and a song to help us recognize each of them.

Table F-1. Intervals.



Begin-End	Interval	Reference Theme
Do-Do	Unison	"America"
Do-Re	Second	"Doe a Deer"
Do-Mi	Third	"Marine's Hymn"
Do-Fa	Fourth	"Here Comes the Bride"
Do-Sol	Fifth	"Twinkle, Twinkle, Little Star"
Do-La	Sixth	"My Bonnie"
Do-Ti	Seventh	"Superman Movie Theme"
Do-Do'	Octave	"Somewhere Over the Rainbow"

As an example, consider the interval of a fourth. This corresponds to playing the first note *Do* as always and then the 4th note *Fa*. These notes come at the beginning of the bridal march by Wagner, played at the start of weddings. These notes may be played simultaneously to perceive how well they blend. Are they pleasing? We will answer such questions in section F-3. The 7th in "Superman" does not come at the very beginning of the theme.

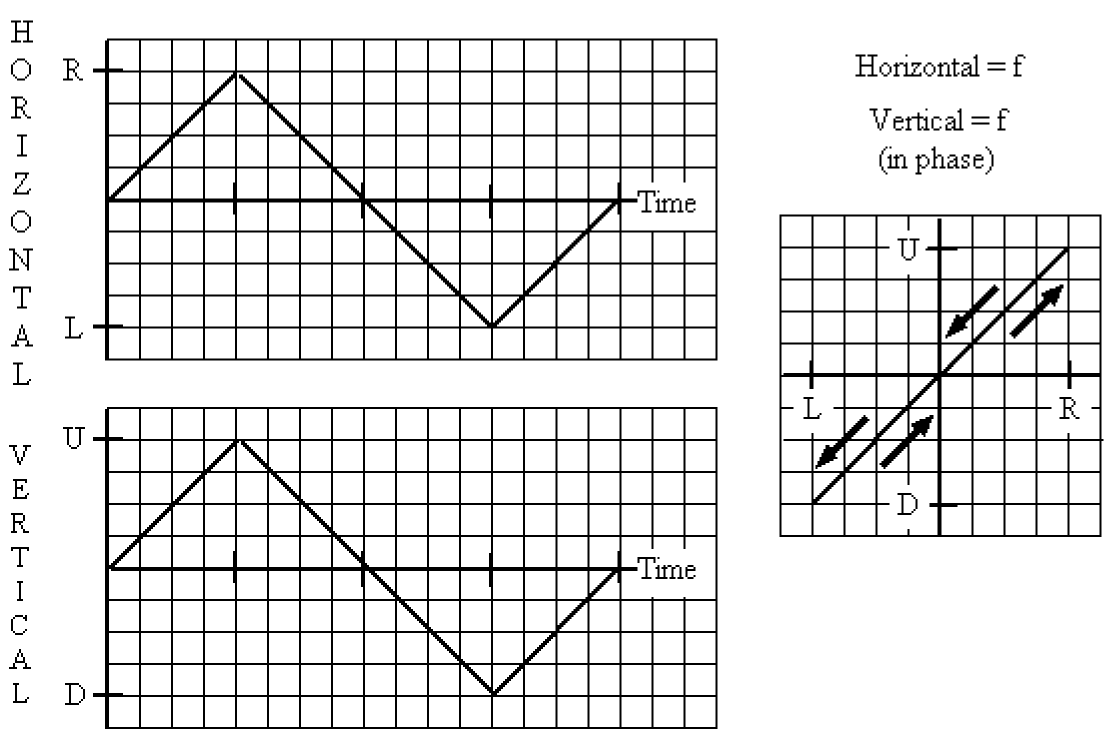
Lissajous Figures

Lissajous (LISS-uh-joo) figures are part of the language of the scientist, i.e., physicist or engineer. We have just discussed components of the language of the musician. In this section we turn to physics. Then in the third section we combine both the musician's and physicist's techniques in a wonderful experiment to determine the frequency ratios of the perfect major scale.

Our analysis will have a blend of mathematics, physics, perception, music, esthetics, and even philosophy. The Greek mathematician and mystic, Pythagoras, was one of the first to study the esthetics of pleasing tone combinations. He showed that tones with the simplest mathematical ratios were the most pleasing. This had profound philosophical implications for him. He discovered that one way to understand the beauty and harmony of nature is through mathematics. The methods we develop here will enable us to pursue the study of harmonious combinations of tones.

We will explain Lissajous figures by example. Consider a party game where you are given two sets of instructions for walking on a floor, the playing field. One set of instructions tells you how to move East or West, which we call right-or-left, while the other tells you how to move North or South, which we simply call up-or-down. You must move simultaneously according to both instructions. Fig. F-2a below shows our first game. The instructions are at the left in graphical form, while the playing field is pictured at the right. R stands for Right, L for Left, U for Up, and D for Down.

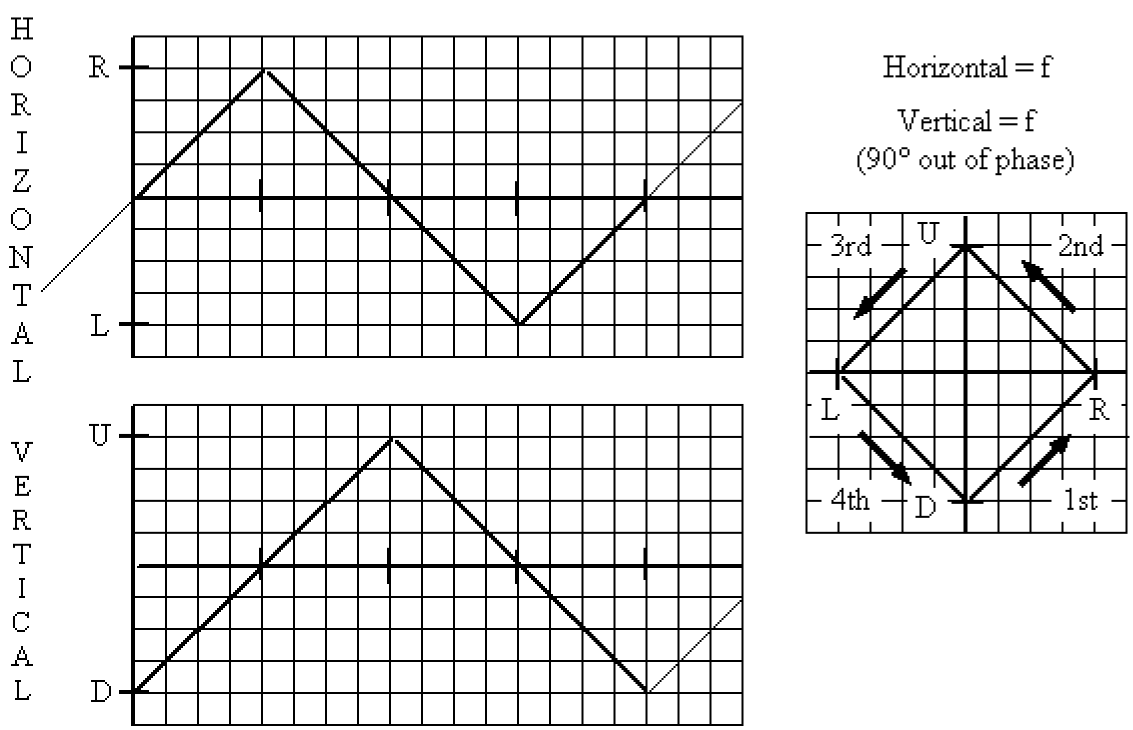
Fig. F-2a. Lissajous Figures: Case I.



The game proceeds in four phases or quarters. The horizontal instructions tell you how much to move left or right, while the vertical instructions tell you how to move up or down. For the 1st quarter, your horizontal instructions tell you to start in a position neither right nor left, then proceed 4 paces (blocks) to the right. Now you must do this while at the same time following the vertical instructions, which tell you to move 4 paces up (from the center) during the 1st quarter. To perform both of these actions simultaneously, you walk along a diagonal to a destination that is to the right and up. The 2nd quarter calls for you to come back to the center position, a position neither right nor left, neither up nor down. The 3rd quarter has you going to the left and down at the same time. The 4th quarter instructs you to come back to the center.

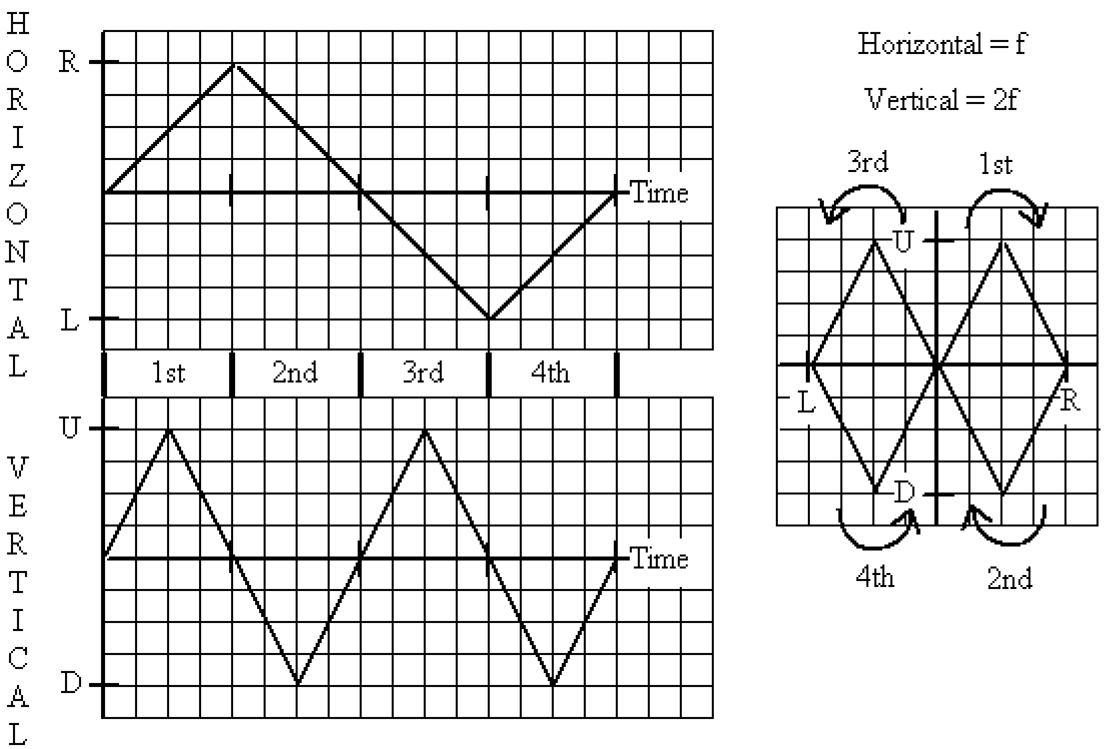
But why are we doing this? Why play the party game? Play one more game before we delve into this. See if you can understand the pattern traced out on the playing field in Fig. F-2b below. Note that the horizontal wave is the same as before. The vertical wave is a similar wave but shifted. It's not completely out of phase (180° out of phase), but halfway there. We say it's 90° out of phase. The traced pattern tells us the frequency ratio of the vertical and horizontal waves. You just count the number of points at the top and the number of points at the right. From Fig. F-2b, the answer is 1 in each case. So the frequencies are the same. In Fig. F-2a, the frequencies are also the same. There is one point at the top and at the right (same point). We plan to send two tones into the oscilloscope which is set in Lissajous mode. Then by counting the points at the top and at the right, we obtain a comparison of the two frequencies!

Fig. F-2b. Lissajous Figures: Case II.



You may be a little confused. So is everybody when they learn a new game for the first time. What should you do? Play another game. By the end of the third game, things will become clear. Refer to the third case in Fig. F-2c below. The instructions indicate that you start in the center. For the 1st quarter, you move to the right and go up and down at the same time. Then you move to the left and go down and up for the 2nd quarter. Refer to the diagram at the right to see these sections of the trip. Can you figure out the paths for the 3rd and 4th quarters?

Fig. F-2c. Lissajous Figures: Case III.



Now for the analysis. We already know the answer for the frequency comparison. The vertical wave has twice the frequency as the horizontal. The vertical wave has two complete cycles in the four quarters of our game time while the horizontal wave has one complete cycle. How can we figure this out from the game board at the right in Fig. F-2c? We compare the number of points at the top to the number of points at the right. The comparison is 2 to 1. We write this as 2:1. As you trace out the combination of vertical and horizontal oscillations, you reach the top two times for every time you reach the far right. That means your vertical frequency is twice the frequency of the horizontal.

We use Lissajous figures electronically to determine the vertical frequency (unknown) relative to our horizontal or reference frequency. However, since it's hard to lock in on the phases, the waves drift. Sometimes we get the baseball diamond for our 1:1 case (Fig. F-2b), shifting into the line of Fig. F-2a as the phases change. The shifting pattern is perceived as a rotation.

The Just Diatonic Scale

We are now ready for our big experiment. We set our oscilloscope to Lissajous mode and connect two tones to it. The reference tone *Do* is connected to the horizontal input. The tone that we will change and measure relative to the reference is connected to the vertical. It is interesting to turn the sound off and just work with the patterns. We scan the vertical until we get nice patterns. These have simple ratios like 2:1 or 3:2. For the ratio 3:2 (read as 3 to 2), we find 3 points at the top and 2 at the right side.

This indicates that the vertical travel is up and down 3 times during the time it takes to go back and forth 2 times along the horizontal. Relating to money, it's the case where your friend has 3 half-dollar coins and you have 2. The ratio is 3 to 2. You have 100 pennies worth of money and your friend has the equivalent of 150 pennies.

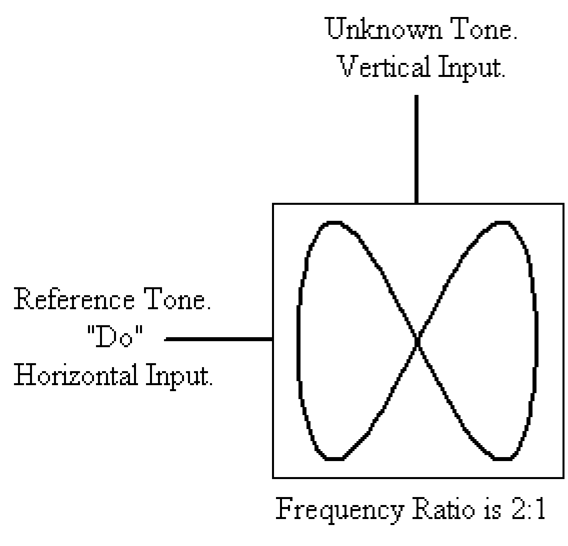
As we continue along these lines we find that simple ratios like 2:1, 3:2, 4:3, 5:4 etc. sound pleasant. We also discover that these tones are in the major scale! For example, suppose we look at the 2:1 case. We play the reference note, then the vertical input, which we know is twice the frequency as the reference. As we listen to the reference (horizontal) and then the tone on the vertical played right after the reference, we recognize the beginning of *Somewhere Over the Rainbow*. So we know we have the octave. The octave higher than *Do* is the note *Do'* and it has double the frequency of the lower note *Do*.

This is one of the discoveries of Pythagoras. However, Pythagoras experimented with strings of different lengths. We are using electronic tone generators and an oscilloscope. Pythagoras would be impressed.

Fig. F-3 illustrates our use of the oscilloscope to obtain a Lissajous figure. We send in our reference tone *Do* into the horizontal input. The unknown tone that we can vary is sent to the vertical input. We turn the knob on the oscillator of our unknown tone until we get a nice pattern.

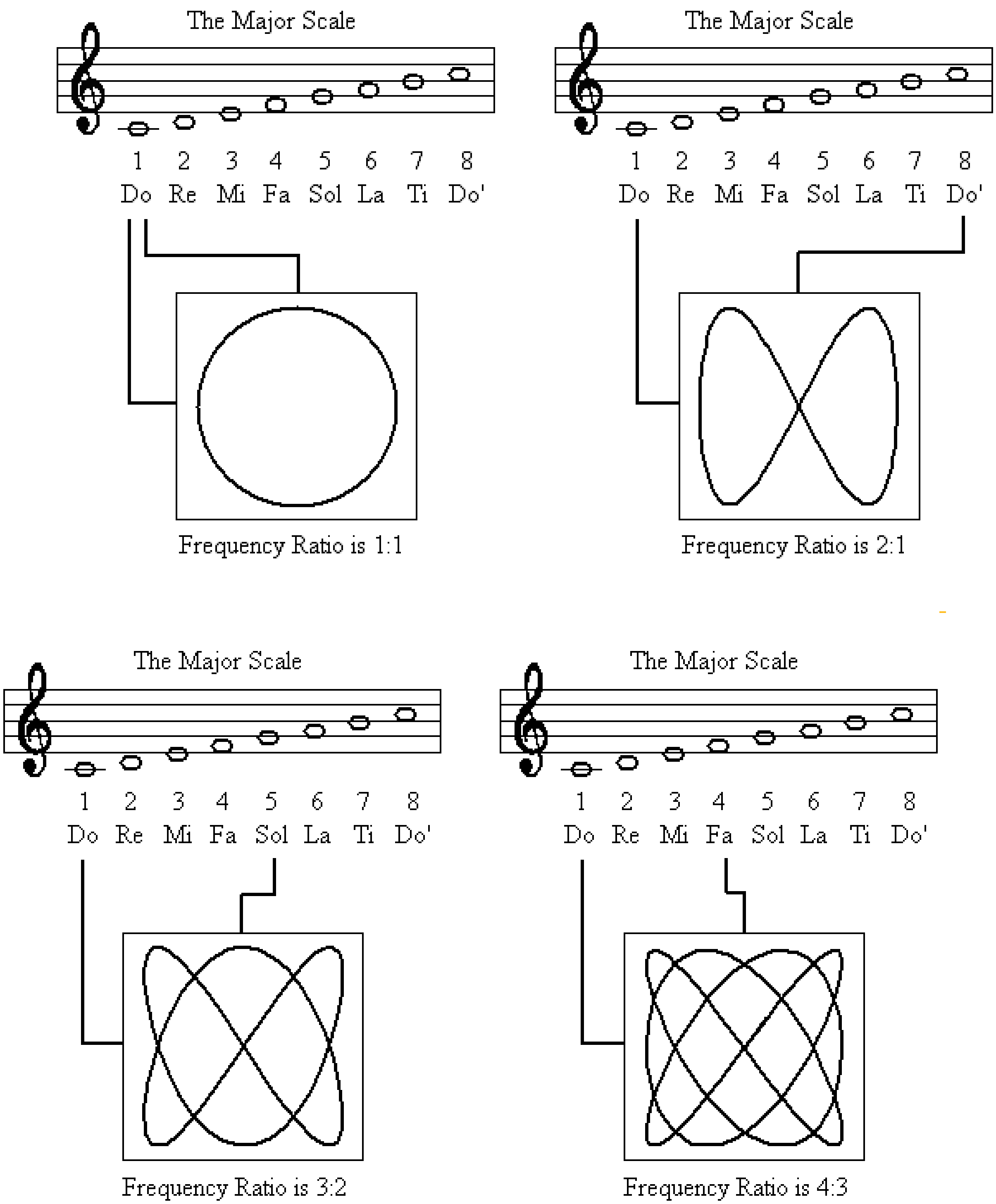
A nice pattern has been found in Fig. F-3. The pattern has rounded edges because we are using sine waves here instead of the triangle waves of Fig. F-2c. However, we can still count the number of times the wave reaches to near the "ceiling" - two times. The number of rounded extremes at the far right is just one. So we conclude that the vertical wave is twice the frequency of the horizontal wave. The ratio is 2 to 1, i.e., 2:1. We play these two tones and we hear the beginning to *Somewhere Over the Rainbow*. The interval defined by Do and the unknown tone is an octave. So the vertical tone is *Do'*.

Fig. F-3. Lissajous Figure with Oscilloscope



The results for several tones are given in Fig. F-4. Note that the ratio is 1:1 when both tones are the same. The unison is 1:1, the octave is 2:1, the fifth 3:2, and the fourth 4:3. These are the most pleasing tone combinations.

Fig. F-4. Lissajous Figures and Frequency Ratios for Some Tones of the Major Scale.



The most pleasant combinations of tones are listed in Table F-2. Tones are said be *consonant* when pleasant, and *dissonant* otherwise. However, there are degrees of consonance and dissonance. The most consonant combination is obviously when both tones are the same. This is the unison. The frequency ratio is 1:1. The next best is the octave. When the 8th tone of the scale is played with the reference tone *Do*, it sounds so pleasing that the 8th tone is also named *Do*. We use the name "Do-prime" (*Do'*) to distinguish this higher *Do* from the lower one. Many composers use octaves in writing for the piano. They are impressive when played quickly by one hand. The flashy 19th-century composer-pianist Franz Liszt often dazzled audiences with rapid octaves.

Next in line for consonance is the fifth, with a ratio of 3:2. The next best consonance is the fourth, with a ratio of 4:3. The amazing feature of Table F-2 is that the listing of consonances from best on down is grounded in mathematics. The simplest numbers are chosen to make these ratios. There is a pattern. In fact, the table suggests that the next to be investigated is the case with a ratio of 5:4. That combination is the result for the interval of a third (*Do-Mi*). You can now appreciate the wonder of Pythagoras as he discovered the mathematical foundation of musical esthetics. The pleasing or harmonious intervals are described by elegant frequency ratios.

Table. F-2. The Most Consonant Tone Combinations (Tuning to Perfect Intervals).

Tones	Interval	Ratio
Do-Do	Unison	1:1
Do-Do'	Octave	2:1
Do-Sol	Fifth	3:2
Do-Fa	Fourth	4:3

Today we do not use perfect intervals for tuning except for the octave. However, the tuning is close to the ratios given in Table F-2. The reason for this will be explained in a later chapter.

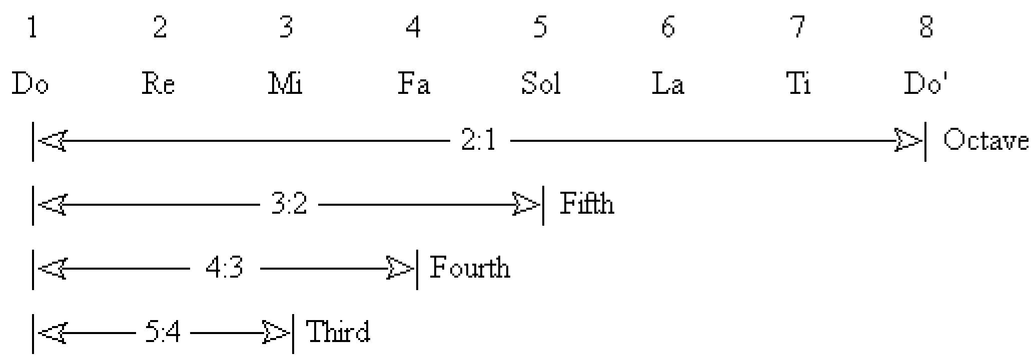
Music theory for composition and harmony incorporates the essence of Table F-2. The following analysis applies to harmony, not the melody line. The most pleasing combination consists of the same notes. However, if you just play the same note, you do not go anywhere. The next best change is to go to the octave. However, this change sounds so close to the reference (*Do*) that there still doesn't appear to be any (significant) change. The next best is movement by a fifth. This is a most pleasing change. Over the years, musicians have dressed up this change by adding related notes to support it and bring it out so to speak. Related notes when played together are called *chords*.

Chord changes by fifths serve as the basis for music theory. The musical palette of fifths is referred to as the c*ycle of fifths*. The author's jazz teacher at the University of Maryland once said (c. 1975) that 80% of popular music consists of chord (harmonic) changes that are fifths.

Frequency ratios for the most consonant intervals are illustrated in Fig. F-5. Once again, note the elegance of the mathematics. If you take the frequency for *Do* to be 100 Hz, then the octave is twice this, i.e., 200 Hz. The fifth is 150 Hz (remember our earlier discussion of the 3:2 ratio with money). The 4:3 ratio gives a frequency of 133 Hz. Can you explain why?

The 5:4 ratio gives a frequency of 125 Hz. The 5 to 4, in terms of our money example, translates to your having 4 quarters and your friend 5. You always have the dollar. You break it into 4 parts, then figure out what 5 of these parts would be. That gives 5 quarters or 125 pennies, the amount of the unknown.

Fig. F-5. Intervals and Frequency Ratios.



We would like to explain shortly, this time in some detail, how to figure out the frequencies from the ratios. We intend to do this for the entire perfect major scale.

Recall that the Greek philosopher-mathematician Pythagoras discovered the mathematics behind the consonant intervals. This was around 550 BC. Pythagoras went on to do work with musical scales, which quickly gets complicated due to playing in different keys.

The Greek astronomer Ptolemy around 150 AD developed a scale with the simplest perfect ratios. The scale presents difficulties for playing in other keys; however, it is excellent if you stick with one key.

The monk Zarlino introduced this scale for Church services in 1558. Table F-3 lists the degrees of the scale with perfect ratios for each degree. The frequencies are compared to the first degree as before. The major scale using these frequencies is called the *just diatonic scale*, *just scale*, or *just intonation*. Since the time of Bach (around 1700) we use the *equal-tempered scale*, which will be discussed in a later chapter.

Table F-3. Scale Degrees and Frequency Ratios Relative to the First Degree (*Just Scale*).

Degree in Scale	1	2	3	4	5	6	7	8
Name of Degree	Do	Re	Mi	Fa	Sol	La	Ti	Do'
Frequency Ratio	1:1	9:8	5:4	4:3	3:2	5:3	15:8	2:1

Table F-4 lists an example of specific frequencies that realizes the *just diatonic scale*. We start with 240 Hz so that the numbers come out easy. The easiest one to determine is the octave or 8th degree, which is double. But we want a systematic way to calculate these. So we will proceed in order.

The second has a frequency ratio of 9:8. The prescription to get the frequency is first to establish your reference frequency. We did that. It is 240. Now take the second number in 9:8, i.e., the 8, and divide 240 into 8 pieces. You get 30 for each of these. We want 9 of these for our tone. So 9 times 30 is 270 and we are finished. The ratio of the second degree to the first degree is 9 to 8; the second tone has 9 parts (30 per part) to the 8 parts of the reference.

For the next case, 5:4, you divide the reference 240 into 4 parts. This gives 60. Now we need 5 of these. That is 300. The third has 5 parts (60 per part) to the 4 parts of the reference. Note that the value of a part here is 60, not the same "piece size" we considered earlier.

Table F-4. Example of Specific Frequencies for the Just Diatonic Scale.

Degree in Scale	1	2	3	4	5	6	7	8
Name of Degree	Do	Re	Mi	Fa	Sol	La	Ti	Do'
Frequency Ratio	1:1	9:8	5:4	4:3	3:2	5:3	15:8	2:1
Example: f (Hz)	240	270	300	320	360	400	450	480

We now summarize these steps to determine frequencies in a compact form. For example, to find the frequency with ratio 5:4 with respect to our reference tone of 240 Hz, you simply write 5:4 as 5/4 and multiply this ratio by 240.

$$\text{Frequency} = \frac{5}{4}\ 240 = 5\ (60) = 300.$$

Use this method to verify all the frequencies in Table F-4. Do not use a calculator. You do not need a calculator to work out the examples in this text. You will understand the material better without a calculator, acquire confidence in scientific calculations, and feel better about yourself as a result.

Some Questions

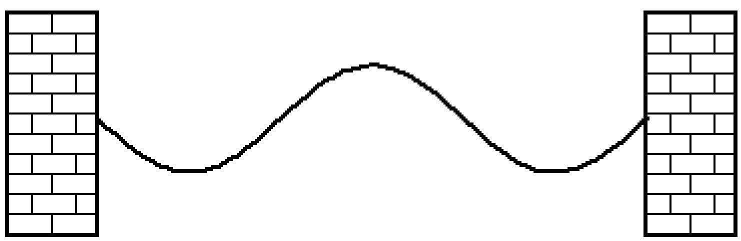
These topics are discussed in class.

Describe how the 1, 5, and 4 are employed in simple songs.

Sketch the “Blues Formula.”

--- End of Chapter F ---

G. Strings



In the previous chapter we started with the musician's scale and found an underlying mathematical simplicity to it. Here we will start with physics and see if a natural scale arises. You might ask if we didn't do this already in the last chapter? Didn't we construct a set of tones based on simple ratios?

Almost. We found pleasant combinations in the ratios 1:1, 2:1, 3:2, 4:3, and 5:4. But what about 6:5, and 7:6? These were not in our scale. The ultimate say was the historical major scale as a given. We were delighted to see such mathematical support for the choices that went into making this scale. However, the choices were made based on historical perceptual esthetics.

Here we are going to discover a set of tones that arise naturally. We will let nature pick all the tones with no interference on our part. Nature has provided us with two very simple structures for tone production - the string and the pipe. These, along with membranes, serve as the basis for the construction of our musical instruments. We will study the strings in this chapter and the pipes in the next.

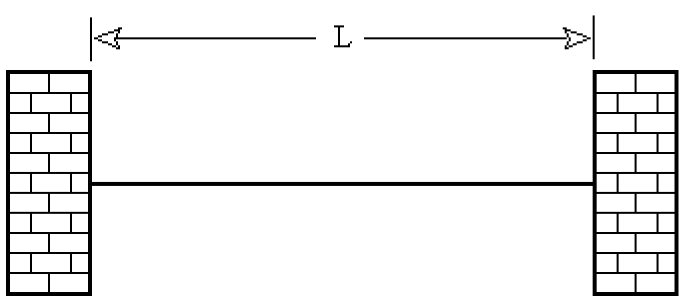
Strings are used in many instruments such as guitars, violins, the harpsichord, piano, and others. We will find that strings produce a natural set of tones. You may have heard of words like harmonics or partials. These refer to such natural tones. Some call these tones the *harmonic series*, the *harmonics*, or the *fundamental* and the *overtone series*. We might call these groups of notes the "physicist's scale" in contrast to the "musician's scale" of the previous chapter.

Just as we discovered that the musician's scale of meaningful tones has mathematical structure, we will find that the reverse is true for the mathematical "physicist's scale" or harmonic series. It has application in music, serving as a basis for musical harmonization and orchestration.

Harmonics

Consider the rope in Fig. G-1 below. Typical rope waves vibrate so slowly that we cannot hear them. However, we can see the patterns of vibration. A single rope can support a series of vibrations. We can measure the frequencies of these. The frequency ratios are most important. They will provide us with the means to establish a new set of tones, the "physicist's scale."

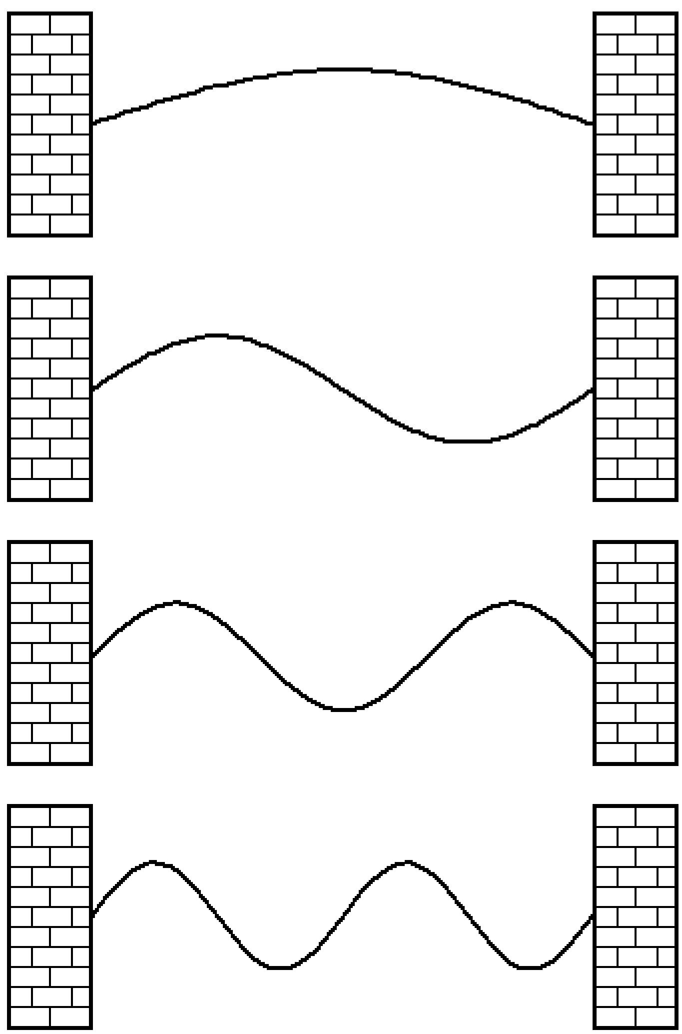
Fig. G-1. Rope or Spring of Length L in Equilibrium (No Waves).



The simplest periodic wave that a rope can support is one that we can start by pulling the middle of the rope up and letting go. The rope will vibrate up and down. We can call this the first mode of vibration for the rope. It is also the first mode of vibration for a string. See this first case in Fig. G-2. Imagine the crest swinging down into a trough and then up again. This mode is also called the *fundamental* or first harmonic.

The next mode can be obtained by pulling the first half of the rope up and the second half down, and then letting go. You will have one crest and one trough (see the second case in Fig. G-2). As the rope vibrates, the left half will go from crest to trough etc. as the right half does the opposite. Fig. G-2 illustrates the first four modes of vibration for a rope or string. These patterns can also be obtained by shaking one end of a rope or spring, while your friend holds the other end.

Fig. G-2. Standing Waves on Ropes (or Strings).



Note that the first mode consists of one half-wave, while the second consists of two half-waves, or one complete wavelength. Therefore, the wavelength of mode 2, i.e., the second harmonic, is shorter. One full wavelength fits between the walls for the second harmonic. The first harmonic has such a long wavelength that just one half the wavelength (crest or trough) fits between the walls.

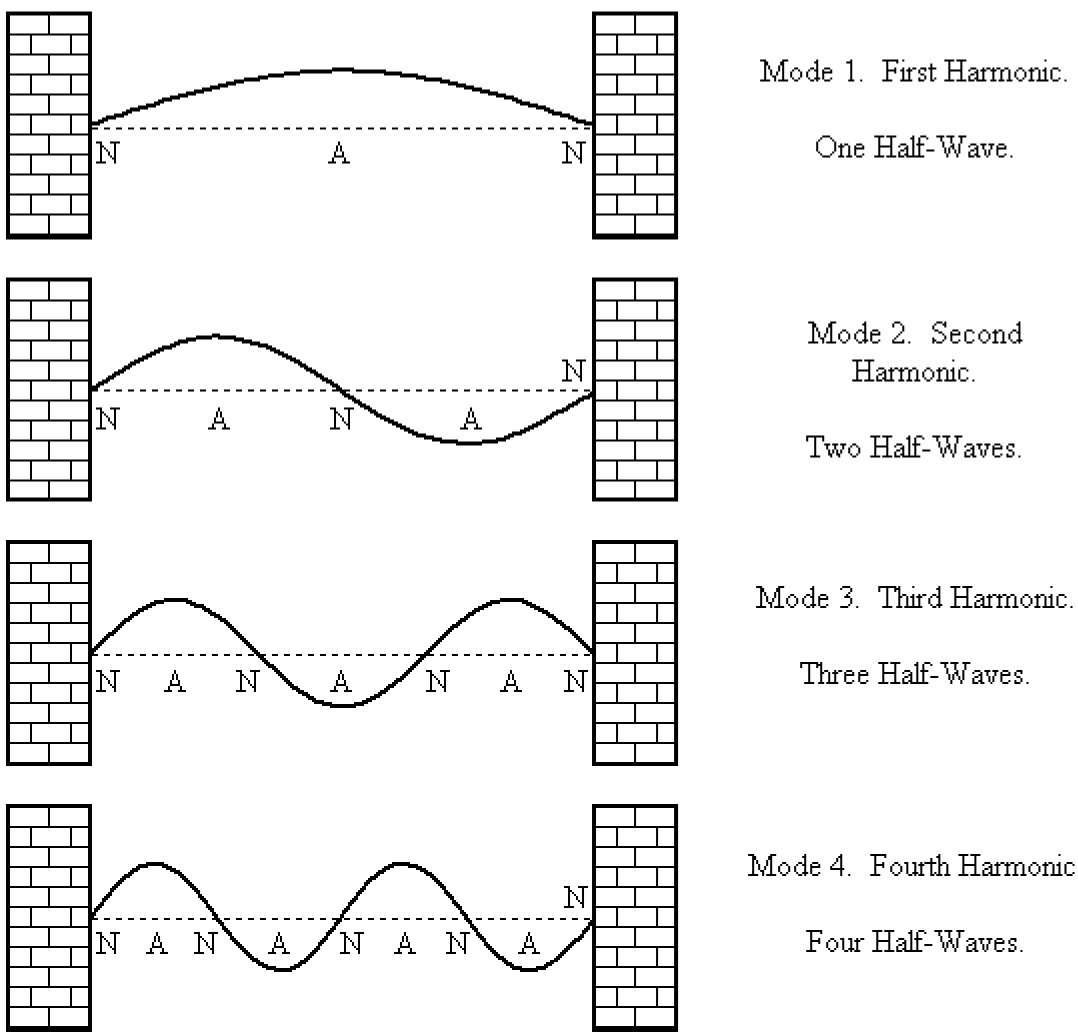
Since the speed of the waves is constant and determined by the rope or medium properties, we know that if you decrease the wavelength, the frequency increases. The second harmonic has half the wavelength of the first harmonic, therefore, the second harmonic has twice the frequency.

It is easy to measure the frequencies of rope or spring waves and directly verify the above. Can you reason in a similar fashion to determine the frequencies for the third and fourth harmonics?

Waves normally want to travel down the rope. But the rope waves hit the fixed ends tied to the brick walls. The waves reflect back and forth constantly. However, for some special frequencies the reflecting waves interfere to establish the patterns such as those in Fig. G-2 (also see Fig. G-3 below). These special waves are called *standing waves*. The patterns are essentially fixed, or "standing." There are points along the rope where the rope does not move. These points are called *nodes*. They are marked by the letter "N" in Fig. G-3. Note that the fixed ends at the walls are always nodes.

There are other points through which the rope swings to extremes in its movement. These points along the horizontal are called *antinodes* and are marked with the letter A. Sketch the fifth harmonic underneath the fourth harmonic in Fig. G-3 and indicate the nodes and antinodes.

Fig. G-3. Nodes (N) and Antinodes (A).

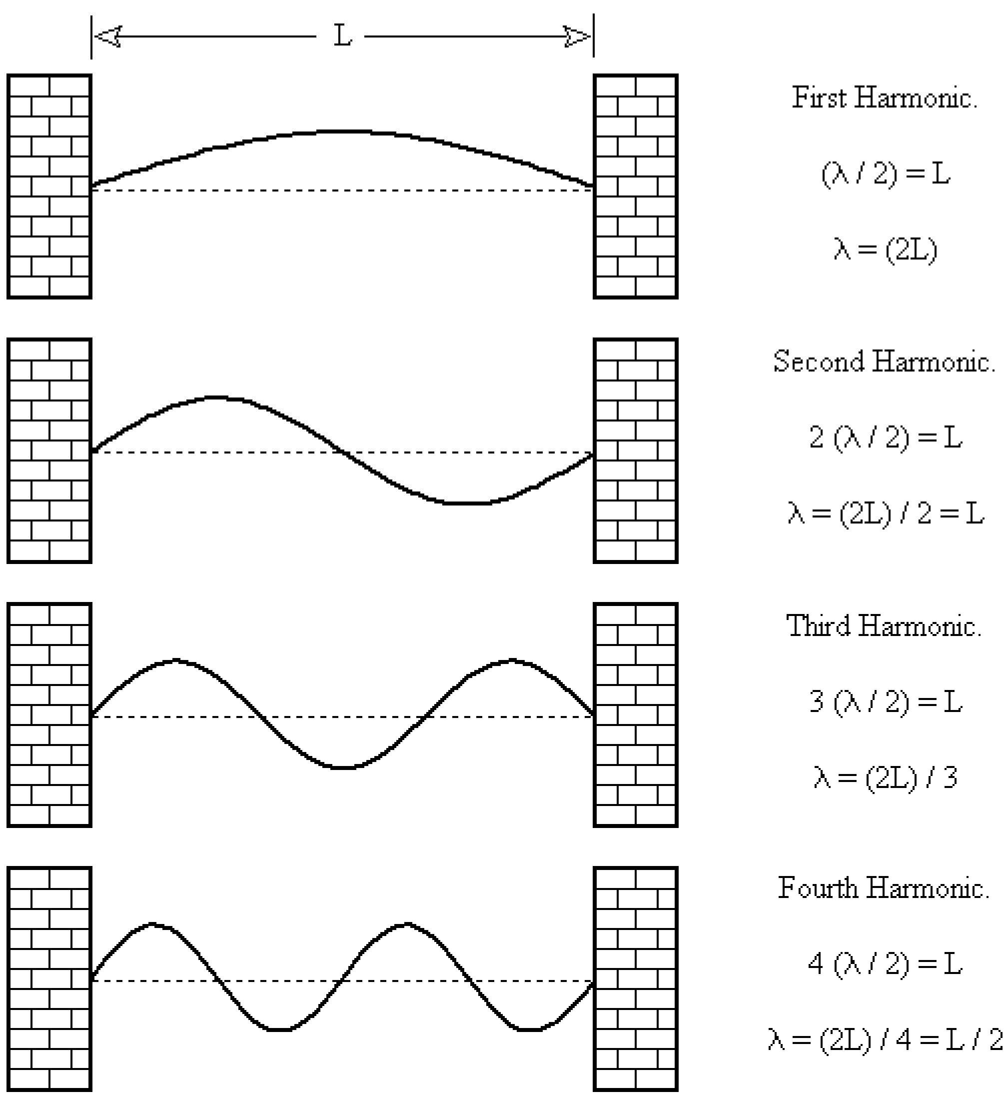


Look at the second harmonic in Fig. G-4. One complete wavelength fits nicely between the two walls. There is one crest and one trough in the snapshot of the second wave depicted in Fig. G-4. Therefore, the wavelength for the second harmonic is L, i.e., $\lambda_2 = L$. Another easy harmonic to look at is the fourth harmonic. Here two complete wavelengths fit between the walls. The wavelength $\lambda_4 = L/2$.

A fast way to understand all of the harmonics is to note that the first harmonic has one half-wave, the second has two half-waves, the third three and so on.

The first harmonic has wavelength 2L. Don't worry that it never has a complete one between the walls. Only the second harmonic has a perfect match of one wavelength between the walls. Then as you squeeze more and more half-waves in, the wavelength must get shorter. If you squeeze in two, the wavelength shortens to 1/2 of what it was before. If you squeeze in 3 instead, the wavelength is 1/3 of the original wavelength for the first harmonic.

Fig. G-4. Wavelengths of the First Four Harmonics.



The Overtone Series

The standing waves for the string or rope are the harmonics. We list in Table G-1 the first eight harmonics. Note that the wavelengths are easily determined. As we squeeze in more and more half-waves, the wavelengths get shorter.

Compare the 5th harmonic with the 1st. The 1st harmonic has one half-wave fitting across the entire length of the string. The 5th harmonic has 5 half-waves in this same distance. Therefore, each of these must be smaller. In fact, if we can fit 5 in where before we had 1, each half-wave must be 1/5 of what we had for the 1st harmonic. Each smaller half-wave for the 5th harmonic is L/5. But since this is a half-wave, we need to multiply by 2 to get a complete wave.

The entry in Table G-1 for the wavelength of the 5th harmonic is 2L/5. To see this one more way, sketch the 5th harmonic. You should have 5 half-waves between the fixed ends. Let each half-wave be 10 cm. Then your length L is 50 cm. A complete wavelength consists of two half-waves, a crest and a trough. This complete wave is then 20 cm, or 2/5 of the length L. This is the (2L)/5 found in the table below.

Table G-1. The First Eight Harmonics

Harmonic Number	Wave-Length	Frequency	Interval from First Harmonic
1	2L	f	Unison
2	2L / 2	2f	Octave
3	2L / 3	3f	Octave + Fifth
4	2L / 4	4f	Two Octaves
5	2L / 5	5f	Two Octaves + Third
6	2L / 6	6f	Two Octaves + Fifth
7	2L / 7	7f	(Two Octaves + Minor Seventh)
8	2L / 8	8f	Three Octaves

The frequencies in Table G-1 are obtained by recalling our wave relation $\lambda f = v$. If you halve the wavelength, the frequency doubles since the product is a constant, the speed. The speed depends on the properties of the string such as its mass and tension. If your wavelength reduces to 1/3 of its original length, the frequency triples and so on. So whatever the denominator is in the wavelength column, that number multiplies our original frequency, which we take to be f.

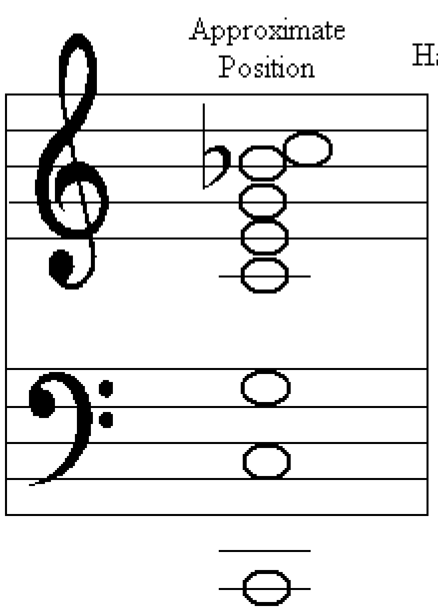
The first harmonic is also called the *fundamental*. The frequency "f" represents the frequency of the fundamental or first harmonic. The first few frequencies can also be measured directly using a spring.

The last column in Table G-1 gives the interval jump from the fundamental. Note that all the harmonics beyond the fundamental have frequencies that keep increasing. These are called *overtones* since their tones are higher or over the fundamental. The approximate positions of these tones in musical notation is given in Fig. G-5, where we have arbitrarily chosen the fundamental. The first overtone is the second harmonic H2, and it is an octave higher. Refer to the previous chapter for a review of intervals and frequency ratios. There we learned that a frequency ratio of 2:1 corresponds to an octave. The frequency ratio of H2 to H1 is 2 since H2 has frequency 2f and H1 has frequency f.

The third harmonic H3 (frequency 3f), compared to the second harmonic H2 (frequency 2f), gives us a frequency ratio of 3:2. This corresponds to a fifth. Therefore, the position of H3 relative to H1 is the interval of one octave (to get to H2) plus an additional interval of a fifth (to get from H2 to H3). To see where we are at H4, simply note that to get there from H1, we double the frequency two times. Double f and you get 2f; double again and you get 4f. We jump two octaves since every time you double the frequency, you go up an octave.

H5 relative to H4 has a frequency ratio of 5:4, which is the interval of a third. H6 compared to H4 has a ratio 6:4, which is also 3:2 by reducing. This is a fifth. So to get to H6, jump 2 octaves to get to H4, then an additional fifth to get to H6. We cannot determine H7 from information in the previous chapter. We list it in parentheses for this reason and also because this interval is very approximate anyway. Finally, to get to H8, you double the frequency of the fundamental 3 times: f to 2f, 2f to 4f, and 4f to 8f. This implies three octaves.

Fig. G-5. Fundamental (H1) and the First Seven Overtones.



Harmonic	Frequency	Interval (from H1)
H8	8f	Three Octaves
H7	7f	(Two Octaves + Minor Seventh)
H6	6f	Two Octaves + Fifth
H5	5f	Two Octaves + Third
H4	4f	Two Octaves
H3	3f	Octave + Fifth
H2	2f	Octave
H1	f	Unison

Notes are approximate since modern tuning does not employ perfect frequency ratios.

Mersenne's Laws

Mersenne listed in the early 1600s the properties of the string that determine the fundamental frequency or pitch. The first property is length. The second two properties, tension and mass, effect the speed of the waves on the string, thereby, influencing the fundamental. See Fig. G-6 below.

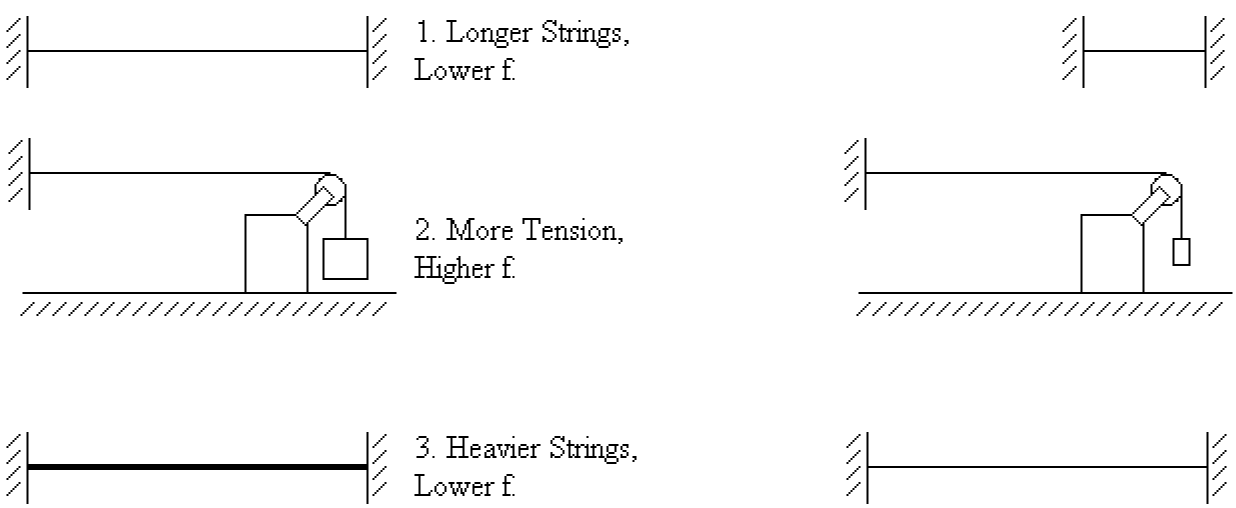
Mersenne's First Law states that the longer the string, the lower the frequency or pitch. We are accustomed to hearing the deep sounds coming from long strings on a bass.

Mersenne's Second Law states that the greater the tension in the string, the higher the frequency or pitch. When a guitar string is tightened, the pitch is raised. This is how strings are tuned. You use your hand to turn the pin that tightens strings on guitars and violins. You need to use a tuning instrument for the piano.

Mersenne's Third Law states that heavier strings result in lower frequencies or pitches. Look inside a piano. The strings down in the bass region are much thicker than the strings at the top end. You will also see that the strings vary in length. Tension is employed to hold the strings in place and give them the correct fundamental frequencies of vibration.

When strings are played, the vibration mainly consists of the fundamental. However, overtones are present in the usual complex vibrations. We will learn in a future chapter that it's the overtones that determine the timbre of a periodic wave.

Fig. G-6. Mersenne's Three Laws.



--- End of Chapter G ---

H. Pipes

We proceed now to the study of standing waves in pipes. The standing waves in the pipe are actually sound waves. We cannot see sound waves in air. However, we can readily hear the tones. The advantage of our earlier experimentation with ropes (or springs) is that we can see the standing waves. The disadvantage is that we cannot hear the rope waves. With pipes, we can hear the waves but not see them. After studying both strings and pipes, you will have an excellent understanding of standing waves.

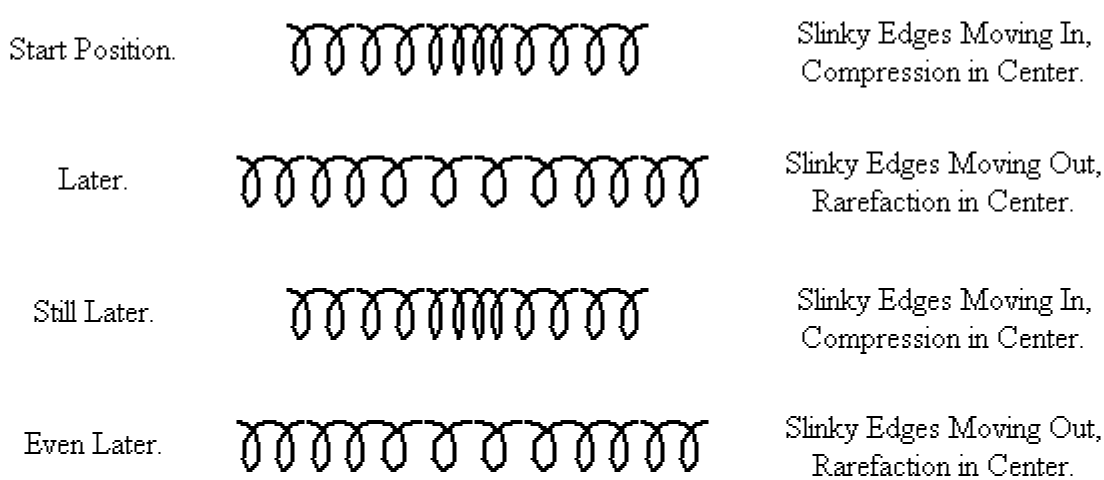
Open Pipes

The first pipes we will consider are open at each end. These are called *open pipes*. You can look through them. They are cylinders. The air inside vibrates as longitudinal standing waves. We can use the slinky as a model to help us visualize what the air does. We replace the open pipe by a slinky. Note that the ends of the slinky are free. This corresponds to the open ends of the pipe where the air can vibrate freely at the ends. If you are worried about gravity, imagine the slinky in outer space inside the space shuttle.

Fig. H-1 illustrates a slinky with the simplest type of oscillation. The edges of the slinky move in and squeeze the center region, then move out and stretch the middle region. This motion repeats. The movement indicated in Fig. H-1 describes the fundamental standing-wave pattern for a longitudinal wave. Try to think of a simpler vibrational mode for the slinky. You will not be able to. Fig. H-1 then depicts the fundamental for the slinky. **We will see shortly that one-half wavelength is depicted in Fig. H-1.**

The layers of air in an open pipe move in a similar fashion. Due to the difficulty in sketching and analyzing longitudinal waves, we will once again draw analogies with transverse waves.

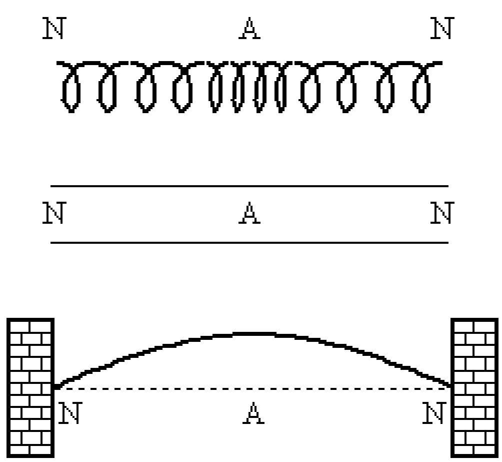
Fig. H-1. Simplest Standing Wave on a Slinky.



The center of the slinky goes through compressions and rarefactions. These are fluctuations in pressure. Imagine being the slinky ringlet in the center. You get squeezed, then stretched. Here, the maximum changes take place. We have designated the points where the greatest changes from equilibrium occur as antinodes.

See Fig. H-2 for the antinode in the slinky. The edges of the slinky are never stretched or compressed. Therefore, they are nodes. The center diagram in Fig. H-2 represents a pipe, with longitudinal waves in air.

Fig. H-2. Nodes for Slinky, Pipe, and Rope.



The air at the center of the pipe goes through high and low pressures as compressions and rarefactions occupy the center. The air at the edges is free to move and stay at equilibrium pressure. The air layers at the edges never get compressed or rarefied. You can remember this by observing that the air near the edges is in direct contact with the ambient air in the room, which air is at equilibrium pressure.

Similarly, in our example with the rope (lowest diagram in Fig. H-2), it is the center that experiences extremes. At one time, the rope is very high in the center (a crest), later it is at the lowest extreme (a trough). The ends of the rope remain at equilibrium, so the ends are nodes.

Note the excellent correspondence among all three systems in Fig. H-2. We will find the similarities between the pipe and the string most meaningful. As the string or rope wave swings from crest to trough, the pipe wave changes in the central region from one of compression to rarefaction.

There is a one-to-one correspondence for the node and antinode regions. We will be able to use this fact to determine the series of standing-wave patterns for the open pipe.

Our plan is to revisit the string and to further develop the analogies found in Fig. H-2. In this way we will be guided by the string. Due to the correspondences between strings and open pipes, we expect that the open pipe will have the same harmonic series as the string. The ends of the pipe replace the ends of the string. We will copy the node-antinode structure from the string vibrations over to the open pipe for each harmonic in order to find the nodes and antinodes along the pipe.

Fig. H-3 below shows the application of string oscillations to determine the standing waves in an open pipe. The analogy indicates that the same harmonics will be supported on the open pipe. If the lengths of the string and open pipe are the same, then the spacing of the nodes and antinodes will be the same. Such spacing determines the wavelength in each case. Therefore, the wavelengths will be the same. The frequencies will differ since the speeds of the waves are not the same on the string and in the pipe.

But we never committed ourselves to a specific fundamental frequency with strings. We simply called the fundamental frequency "f." Therefore, everything applies here.

The waves on the string are string waves. These transverse string waves shake the air surrounding them and produce sound waves in air. On the other hand, the waves in the pipe are already sound waves, waves vibrating in the air inside the open pipe. Note that the distance between a node and antinode is a quarter-wave. Recall that for the first harmonic, we have one half-wave. See the labeling "N-A-N" for this case in Fig. H-3. Note that from "N" to "A" and from "A" to "N" are quarter-waves.

Fig. H-3. Using the String to Determine the Standing Waves for the Open Pipe.

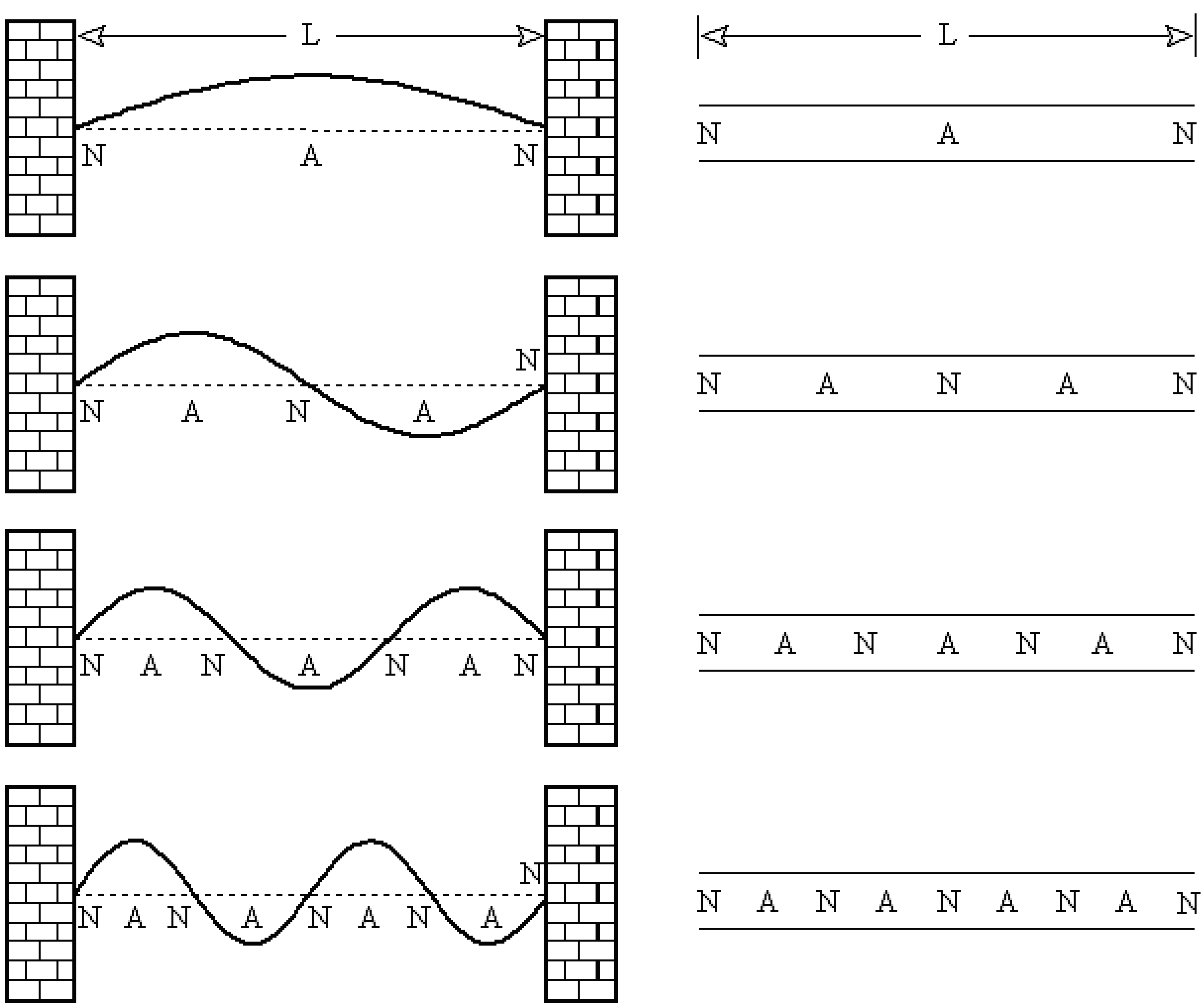


Table H-1 reproduces the first eight harmonics since these also apply to open pipes. The fundamental for the open pipe is determined by the length of the pipe and the speed of sound in air, the medium supporting the standing waves. Whatever this frequency happens to be, we call it "f." Then, the second harmonic (or first overtone) has frequency 2f, the third harmonic (second overtone) has frequency 3f, and so on.

The fundamental for the string is determined by the length of the string (Mersenne's First Law) and the speed of the waves on the string. The wave speed on the string is in turn dependent on the tension in the string (Mersenne's Second Law) and the heaviness (mass) of the string (Mersenne's Third Law).

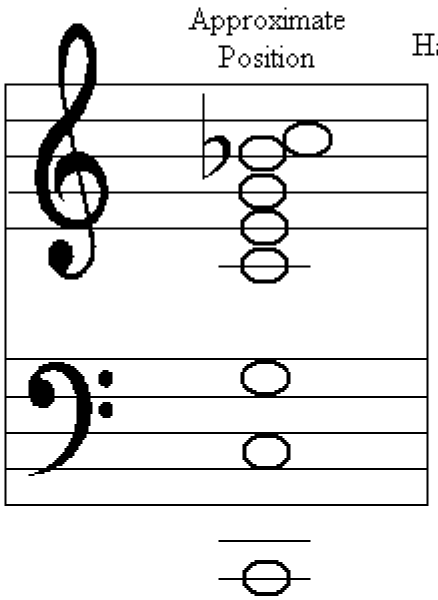
The fundamental for the open pipe is determined by the length of the pipe, as has been noted above. We can say that the speed of sound inside the pipe is likewise determined by the medium inside. We assume it is air, but it can be some other gas. The speed of sound in the gas is in turn dependent on medium properties such as the temperature, pressure, and the density of the gas. However, to some extent, these properties are dependent on each other.

Table H-1. The First Eight Harmonics (Standing Waves on Strings and in Open Pipes).

Harmonic Number	Wave-Length	Frequency	Interval from First Harmonic
1	2L	f	Unison
2	2L / 2	2f	Octave
3	2L / 3	3f	Octave + Fifth
4	2L / 4	4f	Two Octaves
5	2L / 5	5f	Two Octaves + Third
6	2L / 6	6f	Two Octaves + Fifth
7	2L / 7	7f	(Two Octaves + Minor Seventh)
8	2L / 8	8f	Three Octaves

Fig. H-4 lists the first eight harmonics again. We encountered this figure in our study of strings. The significance of harmonics is even more profound now that we find them generated by open pipes as well as strings. The two basic methods of producing sound in nature (strings and pipes) give us the harmonic series. This set of harmonic tones is nature's scale or, what we call in this text, the "physicist's scale."

Fig. H-4. The First Eight Harmonics (Generated by Strings and Open Pipes).

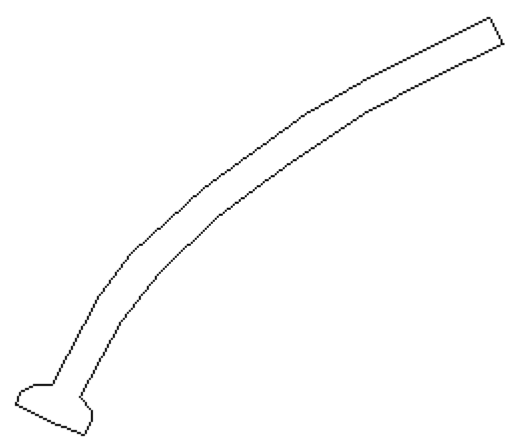


Harmonic	Frequency	Interval (from H1)
H8	8f	Three Octaves
H7	7f	(Two Octaves + Minor Seventh)
H6	6f	Two Octaves + Fifth
H5	5f	Two Octaves + Third
H4	4f	Two Octaves
H3	3f	Octave + Fifth
H2	2f	Octave
H1	f	Unison

Notes are approximate since modern tuning does not employ perfect frequency ratios.

A convenient way to demonstrate overtones is to use an inexpensive toy, the *Twirl-a-Tune* or *Whirl-a-Tune* (see Fig. H-5). The Twirl-A-Tune is a corrugated plastic tube with a handle at one end. It produces overtones when whirled around. Air rushes up the tube when twirled. The rush of air against the ridges and valleys of the tube excites the tube into vibrating at special frequencies - the overtone series for the tube. The faster you twirl, the higher the overtones you obtain. The toy is too excitable to get the fundamental. It readily produces the second harmonic or first overtone.

Fig. H-5. *Twirl-a-Tune*.



Up until this point we have considered nodes and anitnodes for pipes in terms of pressure. A pressure node is a place where the pressure remains at the equilibrium pressure. A pressure antinode occurs when the point goes through periods of increased pressure (*compressions*) and times of lowered pressure (*rarefactions*).

Note that the ends of an open pipe are always pressure nodes since the air is free to move and maintain equilibrium. It is never allowed to compress. This is analogous to the nodes at the ends of a string, where the string remains at equilibrium. But note an important distinction. The rope ends are fixed, not allowed to move, while the air layers at the ends of an open pipe do move. This presents no problem because equilibrium is maintained in each case.

However, we like to define another kind of node-antinode for pipes, one that is defined in terms of displacement (motion) instead of pressure equilibrium. If you think in terms of displacement or movement, then a node is a place that doesn't move.

An antinode is a place that undergoes the maximum motion. The nodes and antinodes on a string are of the displacement type. A string node means the string doesn't move there. But with pipes, both pressure and displacement descriptions apply. So we have both kinds of nodes. However we must be careful.

At a pressure node, like at the ends of a pipe, the air layers are free to move and they do, to maintain equilibrium pressure. Because the end layers of air move, these regions are displacement antinodes. On the other hand, at the places where compressions and rarefactions occur, the central air layer does not move. Recall the slinky ringlet that gets pushed on from both sides and then stretched equally from both sides. It doesn't move as it gets squeezed and stretched. Therefore, this ringlet where a pressure antinode occurs, is the place where we have a displacement node.

Remember it this way. A pressure node is a displacement antinode and a pressure antinode is a displacement node. They are opposite of each other. This feature is illustrated in Fig. H-6 below. Note the shorthand notation for displacement nodes and antinodes. A displacement node is a vertical line, indicating no motion. The dashes indicate motion, displacement antinodes. When we draw these near the ends, we extend them slightly since motion overshoots the edges (think of a slinky).

Fig. H-6. Types of Nodes and Antinodes.

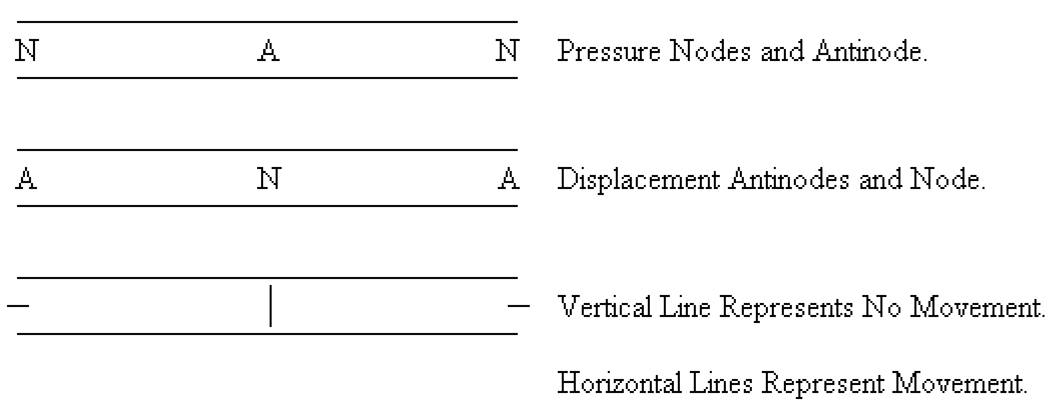


Fig. H-7 illustrates the first four standing waves in an open pipe, where displacement nodes and antinodes are employed instead of pressure nodes and antinodes. The results are opposite the pressure description. Wherever there was a node before, now there's an antinode and vice versa.

The shorthand notation for displacement nodes is included in the diagrams at the right in Fig. H-7. There is a quick way to remember these. Always sketch a dash "-" at each end. You need the number of vertical lines "|" that corresponds to the harmonic number, e.g., the 4th harmonic has 4 vertical lines. Remember you never have two nodes in a row; there is always an antinode in between. So you alternate these.

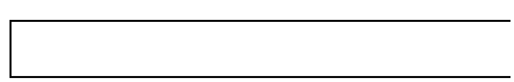
The only problem is that it's a little hard to space them evenly the first time you try it. It helps if you remember one more thing. For odd harmonics, there is a vertical line in the center; for even, there is a dash in the center. The shorthand notation for displacement nodes will be very helpful when we take up the study of closed pipes next.

Fig. H-7. The First Four Standing Waves in an Open Pipe Defined by Displacement.

Closed Pipes

A *closed pipe* is a pipe closed at one end. See Fig. H-8 below. We see immediately that the standing-wave patterns will be different. The closed end forces a displacement node there, a place where there is no motion because "you are up against the wall." The other end is open, free, a displacement antinode. Sketch a little dash at this end. You have the first standing wave!

Fig. H-8. Closed Pipe.



We will use an important observation we made earlier concerning quarter-waves. The fundamental for the closed pipe is compared to that of the open pipe in Fig. H-9 below. For the open pipe we find one half-wave fitted to the pipe length L. The half-wave is spanned by going from an antinode to node and then from a node to an antinode (see open pipe in Fig. H-9). Going from the antinode to node takes you one half the distance across the half-wave. Therefore, an antinode to a node gives us a quarter-wave. Likewise, going from the center node to an antinode is also a quarter-wave. We state these important observations below.

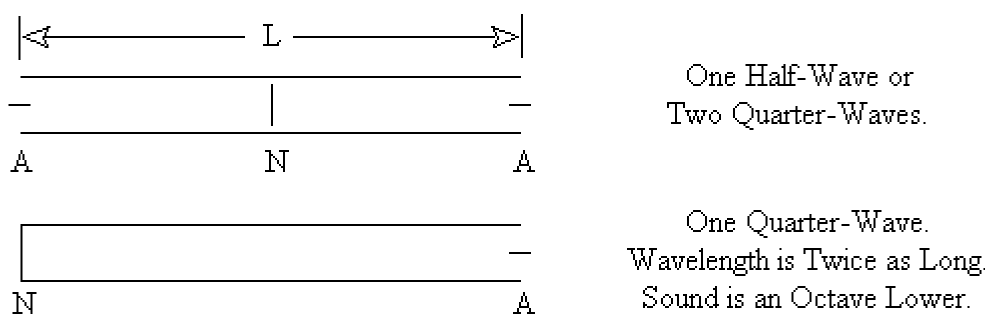
Between a node and antinode is a quarter-wave (quarter of the wavelength).

Between an antinode and node is a quarter-wave (quarter of the wavelength).

Now look at the closed pipe in Fig. H-9. We surely have a displacement node at the closed end. There is no motion at the wall. We also have the usual displacement antinode at the open end. But a node is always followed by an antinode and vice versa. Therefore, the standing wave in Fig. H-9 is the simplest standing wave for a closed pipe.

Imagine a slinky with the left end glued to a wall and the right end free to move. Now pull the right end of the slinky and let go. The right end oscillates in and out at the right end. The left end stays fixed to the wall. You have the slinky version of the fundamental for a closed pipe.

Fig. H-9. Fundamentals for Open and Closed Pipes (Displacement Nodes).



The single quarter-wave in the closed pipe in Fig. H-9 above is twice the length of each of the two smaller quarter-waves found in the open pipe above it. The wavelength of the fundamental for the closed pipe is therefore twice as long as the fundamental wavelength for the same-size open pipe. If we increase the wavelength, we lower the frequency. Since the wavelength is doubled for the closed pipe, the frequency is halved.

The open-pipe fundamental has twice the frequency of the closed-pipe fundamental and is therefore an octave higher. Alternatively, we can say that the fundamental frequency of the closed pipe is an octave lower than that for the open pipe.

There are two secrets in understanding closed-pipe physics. First, if you close one end of an open pipe, you double the wavelength to 4L (lower the fundamental by an octave). Second, as we will see, the closed-pipe harmonics only includes the odd harmonics! These two important characteristics for closed pipes are stated below.

1. Close one end of an open pipe and the wavelength doubles to 4L.

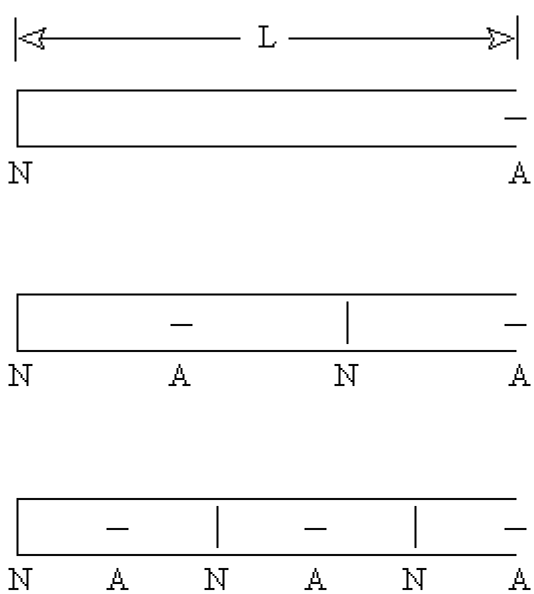
2. The harmonic series for closed pipes includes only the odd harmonics.

The second observation above becomes evident by studying Fig. H-10 below. For the first mode of vibration we have one quarter-wave fitted to L. Therefore, the wavelength is 4L. Do not worry about realizing an entire wavelength within the distance L. This only happens for H2 for the string or H2 for the open pipe.

Mode 1 for the closed pipe has one vertical line (at the wall) and one dash (at the open end). To get the next mode, we squeeze in another pair (the "A" and "N" inside the pipe in Fig. H-10). And herein lies the second secret. There are now three quarter waves. From left to right these are "N" to "A," "A" to "N," and "N" to "A." The squeezed wavelength is now 1/3 of what it was before. The frequency is triple. This is the third harmonic.

We skipped the second harmonic! The next mode adds another antinode-node pair resulting in 5 quarter-waves. Now, the wavelength is 1/5 of what we started with and the frequency is 5 times the fundamental. This gives us the fifth harmonic. We skipped the fourth harmonic. Can you work out the next case?

Fig. H-10. Standing Waves for Closed Pipe (Odd Harmonics Only).



First Mode: $(\lambda / 4) = L$

$\lambda = (4L)$ (First Harmonic, f)

Next Mode: $3(\lambda / 4) = L$

$\lambda = (4L)/3$ (Third Harmonic, 3f)

Next Mode: $5(\lambda / 4) = L$

$\lambda = (4L)/5$ (Fifth Harmonic, 5f)

The results for the first few harmonics for the closed pipe are found in Table H-2 below. Note the two basic secrets we discussed earlier. This enables us to generate Table H-2 from our previous table for the strings and open pipes. First, we note that when we close one end of an open pipe, the wavelength doubles to 4L. So we replace 2L (result for open pipe) by 4L (result for closed pipe) in our table. We always list "f" for the fundamental frequency by definition.

The fundamental is our reference frequency. So even though the closed pipe drops an octave relative to the original open pipe, we still use "f" for the fundamental. You might say we redefine what is meant by "f." This is an important convention. It is always most convenient to call the frequency of the first standing wave "f" and relate all overtones to "f."

Second, we note that the closed pipe only has odd harmonics. So we strike out all the even harmonics from our list and we are finished. We have Table H-2.

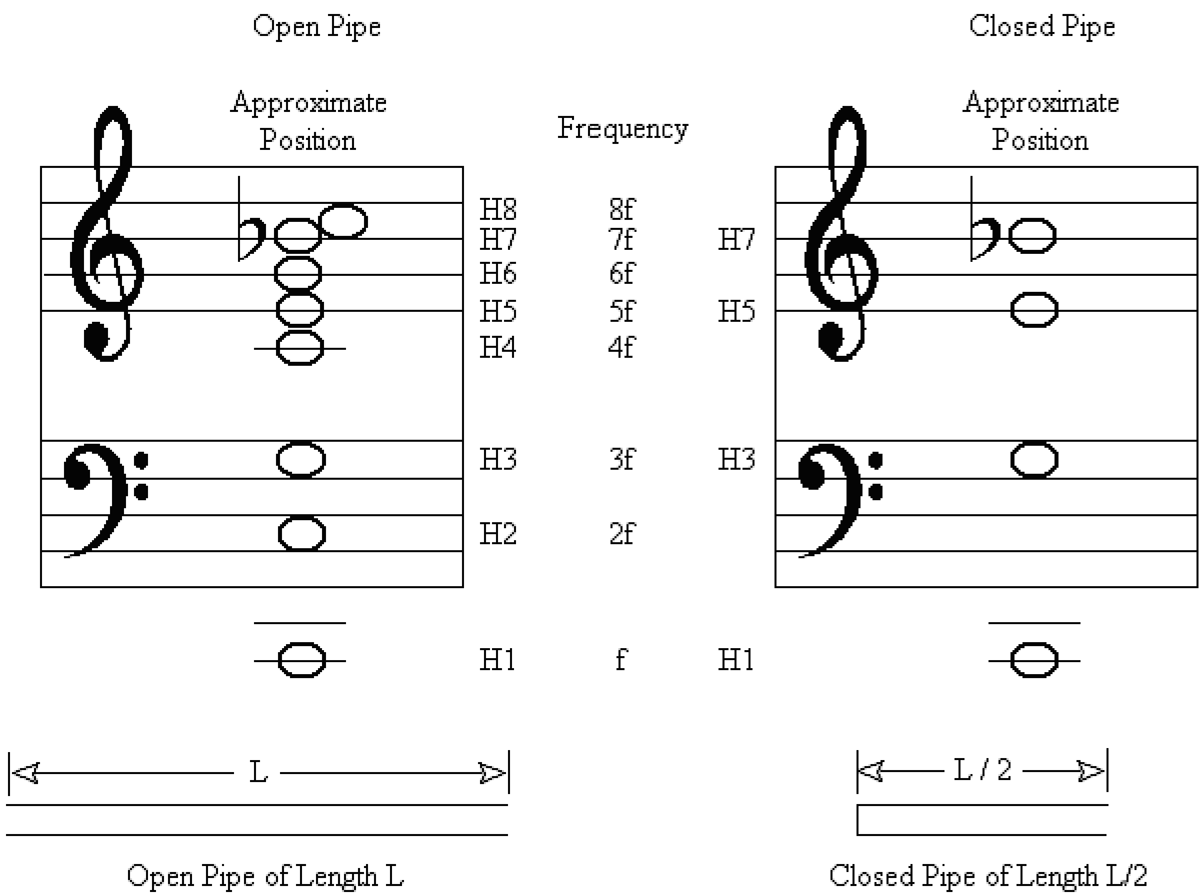
Table H-2. The First Few Harmonics for a Closed Pipe.

Harmonic Number	Wave-Length	Frequency	Interval from First Harmonic
1	4L	f	Unison
2	Not Present.		
3	4L / 3	3f	Octave + Fifth
4	Not Present.		
5	4L / 5	5f	Two Octaves + Third
6	Not Present.		
7	4L / 7	7f	(Two Octaves + Minor Seventh)
8	Not Present.		

We can find an open pipe and a closed pipe of different sizes so that each has the same fundamental. The closed pipe has to be one half as long. This offsets the drop in frequency we get by closing one end of an open pipe. Cutting a pipe in half doubles its frequency because we halve the wavelength by our cut. This compensates for the closed end. In summary, close one end of an open pipe and the frequency drops an octave. Now cut the closed pipe so that the new open end is half as far from the closed end. The closed pipe is one half its original size. This pushes the fundamental back up an octave. Fig, H-11 depicts the first few harmonics of an open pipe and a matched closed pipe.

As before, we point out that the notes on the musical staff are approximate. The scale in use today does not use perfect ratios except for the octave. Today's frequency assignment for the notes is called *equal temperament*. We will explain what this is in a later chapter. We also point out that the seventh harmonic is considerably approximate.

Fig. H-11. Harmonics for Matched Open and Closed Pipes.

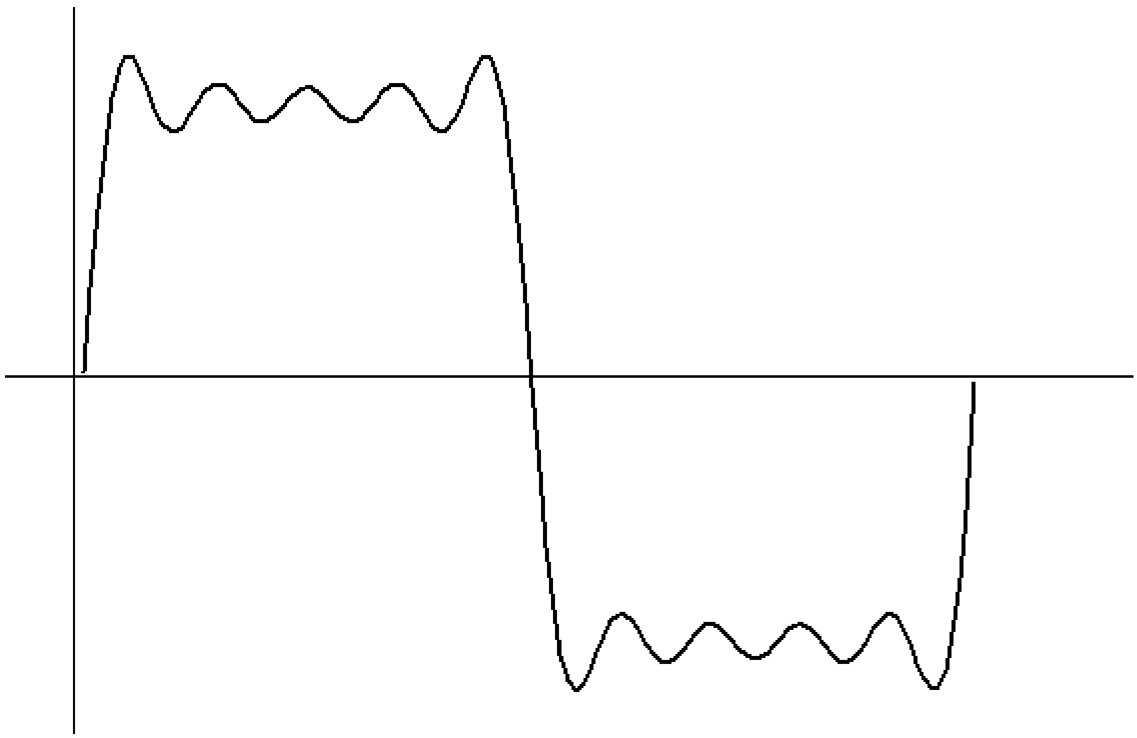


Shortened length of closed pipe keeps the fundamental the same as the open pipe.

The fundamental in Fig. H-11 is about 60 Hz. This answer is based on the position of the fundamental on the musical staff. We will learn later how to determine the length of an open pipe necessary to produce a specific fundamental frequency. For now, we simply state the result. The open pipe needs to be almost 3 m. This is long. But then, this is a very low pitch. The closed pipe needs to be half this length.

--- End of Chapter H ---

I. Fourier Analysis

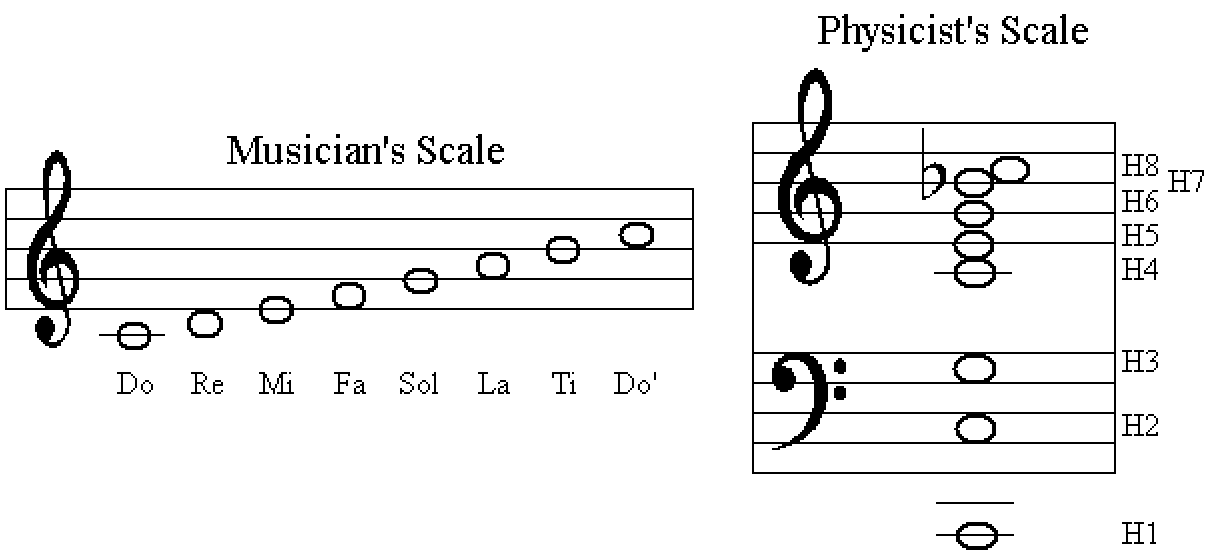


We have seen in previous chapters two sets of notes. These appear in Fig. I-1 below. The musician's scale represented is the *major scale*. What we call the "physicist's scale" comes from the modes of vibration on strings and in pipes.

First, we started with the major scale as a given. We then found that the esthetics of the scale had some underlying mathematical simplicity in terms of frequency ratios. Second, we studied the standing-patterns in strings and pipes. We discovered a very natural grouping of frequencies: f, 2f, 3f, 4f, 5f, etc., the harmonic series (the "physicist scale" in Fig. I-1). Now we will see that this scale of nature has esthetic application in music.

The harmonic series has important applications in harmony. The tones of the harmonic series blend well together. They serve as a guide when a composer wants to choose many notes to sound harmoniously when played together. Think of it this way. Use the musician's scale to pick out a melody. Then use the harmonic series to assist you in finding a group of notes (called a chord) that can be played as background to a note or subset of notes in your melody line.

Fig. I-1. The Scales of the Musician and Physicist.



A melodic line is written on music paper from left to right just like the way we read and write. The notes change in time. Refer to the musical scale at the left in Fig. I-1. The notes appear on the musical staff from left to right. On the other hand, harmonization occurs at the same point in time. Many instruments may participate in playing different notes simultaneously. These supporting notes are arranged vertically underneath the melody note since they are played together. The harmonic series in Fig. I-1 above is written in this vertical fashion. We give an example of harmonization from an orchestral excerpt from the great master Tchaikovsky in Fig. I-2. The use of the harmonic series is apparent. The excerpt is from the second movement of his *Fourth Symphony*, completed in 1878.

The excerpt in Fig. I-2 has been stripped of any musical markings for loudness. It has also been transposed slightly so that the fundamental matches that found in Fig. I-1. It is important for you to refer to Fig. I-1 and check the harmonics listed below. Please do this before continuing. The resulting sound with so many harmonics is very full and satisfying. Note the omission of the jazz-sounding seventh harmonic.

Fig I-2. Excerpt of Tchaikovsky's *Fourth Symphony* Transposed for Harmonic Analysis.

Complex Waves

Fig. I-3 below illustrates three waves, each one getting more complex. The first, Fig. I-3a, is a sine wave. This is the simplest type of wave. It corresponds to a wave of "nature." The standing waves we find on strings and in pipe's are sine waves. The sine wave is sometimes called a pure wave. The sound of a sine wave is innocent, simple, pure in tone.

The second wave, Fig. I-3b, is a *complex periodic wave*. Any periodic waveform that is not simple (not a sine wave, i.e., non-sinusoidal) is a complex periodic wave. These waves have well-defined frequencies (pitches). Different repetitive shapes or waveforms give us the rich variety of timbres we hear.

The third wave, Fig. I-3c, is an *aperiodic complex wave*. Aperiodic complex waves do not repeat. These can be crashes, explosions, or anything else you can think of that cannot hold a steady or definite pitch.

Fig. I-3a. Sine Wave (The Simplest Periodic Wave).

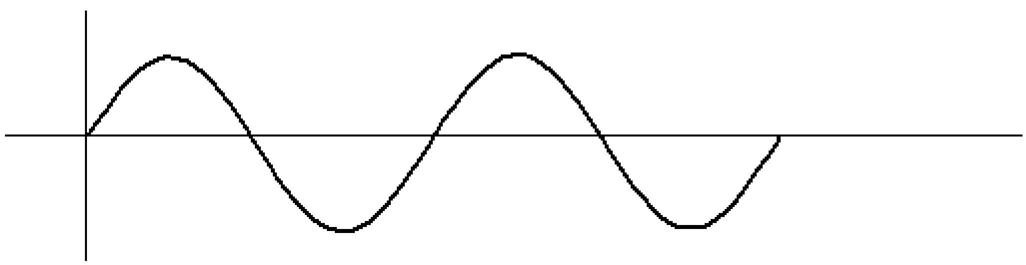


Fig. I-3b. Complex Periodic Wave (Any Periodic Wave that is Not a Sine Wave).

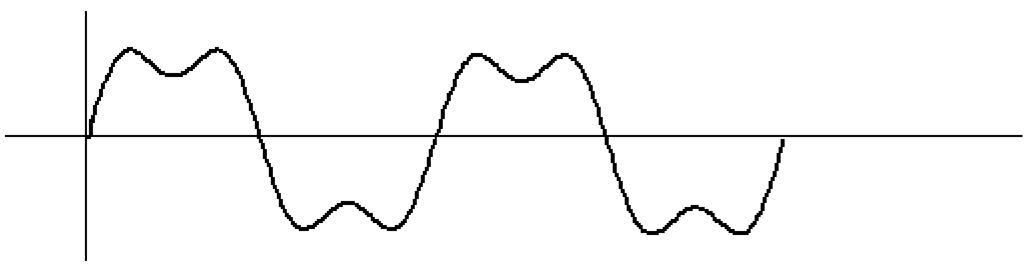
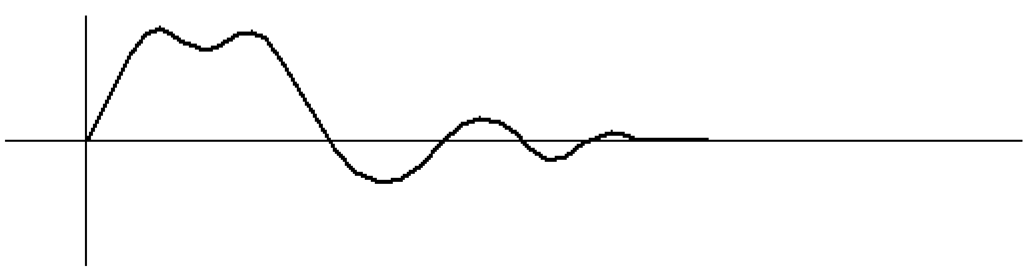


Fig. I-3c. Aperiodic Wave (Any Non-Periodic Wave).



The most interesting waves for us are the periodic waves since these have definite pitches. They include the tones of most musical instruments and the sounds of singers holding a pitch. All of the instruments we encounter in the Tchaikovsky excerpt are playing tones with well-defined frequencies or pitches. These instruments consist of three different woodwinds, the horns, and the strings. The percussive instruments in the orchestra have difficulty producing sustained tones. The piano gets help from the sustain pedal, but technically the sound is aperiodic. It dies down immediately after being produced. Timpani are drums that give a "tuned sound" for an instant. Cymbals are crashes.

There are four main sections in an orchestra: the woodwinds, brass, strings, and percussion. The first three can produce periodic tones. That covers a lot of ground. In fact, these sections of the orchestra are the ones that play most of the time. Strings come first, then woodwinds, and finally brass in order of typical use. Percussion is employed sparingly in a usual orchestral work.

So our focus on periodic waves is justified. And remember, all singers are included because they can sing a pitch and hold it. Even those of us that can't sing usually can hum a note for a few seconds. So most instruments, all singers, and the rest of us produce tones with different timbres.

Even within the same instrument category, timbre varies. The timbre of a *Stradivarius* is different from your common-variety violin. In fact, the timbre is the signature or fingerprint of the instrument. However, timbre also varies somewhat for different ranges of notes on the same instrument! In the next section you will learn how to analyze timbre.

Fourier's Theorem

While the musician may analyze a melodic line in terms of a musical scale, the physicist analyzes timbre in terms of harmonics (which we have called the "physicist's scale"). A mathematical physicist, Baron Jean Baptiste Joseph Fourier, presented an astounding theorem in 1807, which we state below. He found that all periodic waves can be constructed from sine waves in the harmonic series!

Fourier's Theorem

One can construct any periodic
wave having frequency f,
using sine waves with frequencies
f, 2f, 3f, 4f, ...
(the Harmonic Series).

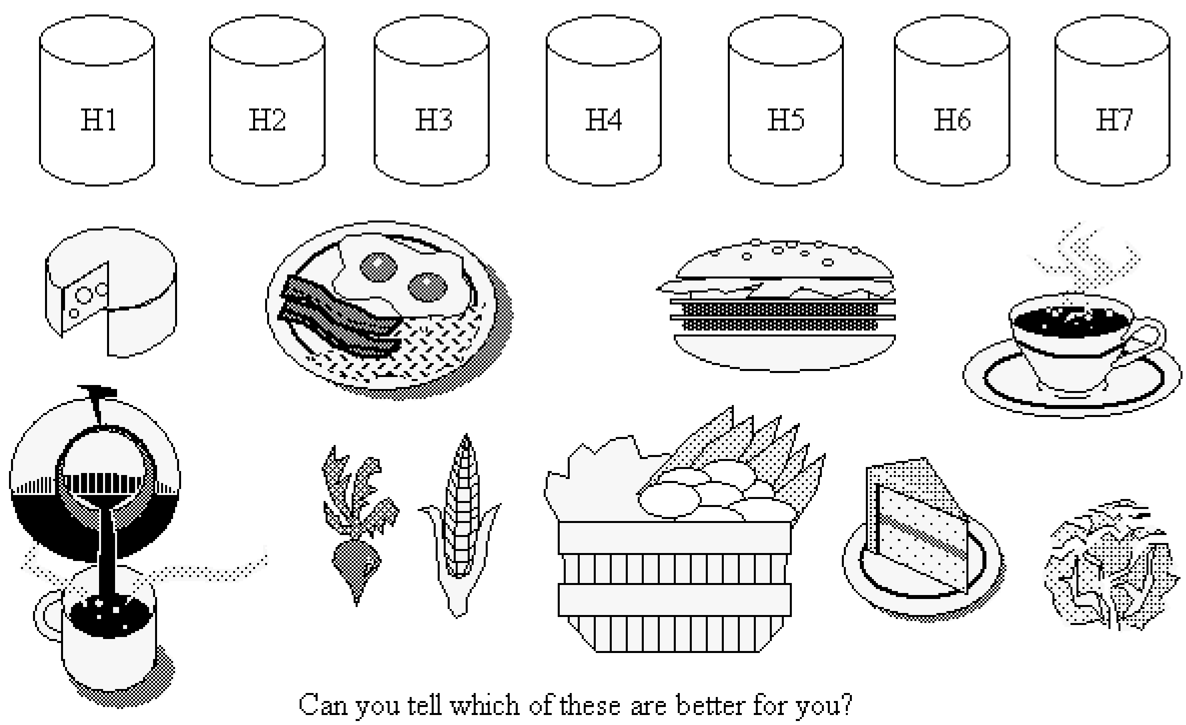
The claim is that you can make any periodic wave from its harmonics. This is a very profound statement, and one which we would like to demonstrate with a specific example. We will take a square wave and challenge the theorem. How can sine waves be combined to get a square? You might be skeptical. That's good. You should be. It makes for a good scientist.

Suppose there was a claim that you could make any food from a set of basic ingredients. You might call the basic series of ingredients the "harmonic series of cooking." Fig. I-4 below shows such a hypothetical series of ingredients, appropriately labeled H1, H2, H3, and so on. Only seven are shown below. Imagine these in your magic cupboard. You accept any challenge of a food to cook. As you go to work, you might not have to use every ingredient in your collection. You would also use specific amounts of those ingredients needed. If you wrote out what you used and how much of each ingredient, you would call this a recipe. It would be the secret for cooking the specific meal you were asked to cook.

Similarly, when presented with a periodic wave, we can write down which ingredients, i.e., harmonics, we use and just how much of each. We mix the ingredients by simply adding harmonics. We know how to add waves; we simply add the displacements. So the recipe will tell us which harmonics to use and just what amplitude to choose for each. We will refer to an amplitude of 1 as one cup's worth. The various amounts we need of each harmonic can be called the Fourier recipe in honor of Fourier.

Actually it's called the *Fourier spectrum*. But there is one more thing a complete recipe needs to tell us. This is how to position the harmonics before adding them. How are they aligned? Are they in phase? So we need the phase relationships. But the spectrum will only supply the amplitudes since these alone essentially affect the sound of the tone. This may be starting to sound complicated, but it really isn't. The best way to convince you of this is to work out an example.

Fig. I-4. Fourier Synthesis is like Cooking!



The square wave we would like to make from sine waves is illustrated in Fig. I-5. The square wave is a periodic wave but we show one wavelength below. We purposely choose a difficult waveform, one with corners. *Fourier's Theorem* states that we can construct or synthesize this square wave from the harmonics that begin with the frequency of the periodic square wave. Let's call this frequency f. The fundamental therefore has the same frequency and thus the same wavelength of our square wave.

The method we employ in building up the square wave from the harmonics is more artistic that scientific. Physics and engineering majors learn mathematical methods to find the Fourier spectrum (recipe). Does that mean they understand it better? No. In fact, you will understand it better because you will see the square wave take shape as we go, rather than become overwhelmed by the obscurity of mathematics. The author speaks from experience. He finally understood Fourier analysis in graduate school when he did it the way you are about to see.

Fourier's Theorem directs us to the harmonics that build on the same frequency of the square wave we are trying to make. Therefore, the first harmonic is the sine wave with the same wavelength as our square wave. This first harmonic is shown in Fig. I-6. Let's use this as a reference for our amplitude and say it has amplitude 1. We go to our harmonic "cupboard" so to speak and use a full cup of our first harmonic.

It resembles our square wave a little. Doesn't it? You are probably not impressed with Fig. I-6 being a match for Fig. I-5. But we only used one harmonic (one ingredient). Things will shape up as we proceed (to cook).

Fig. I-5. Square Wave to Fourier Synthesize.

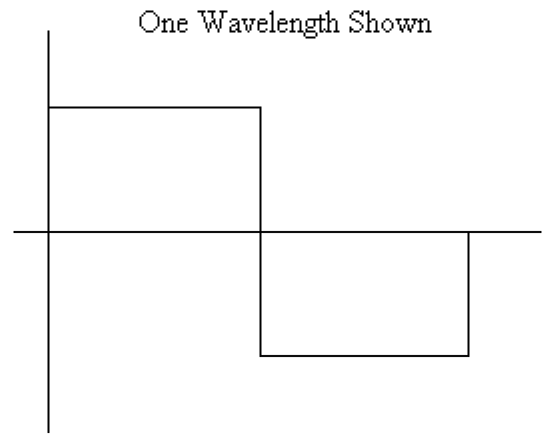


Fig. I-6. First Harmonic.

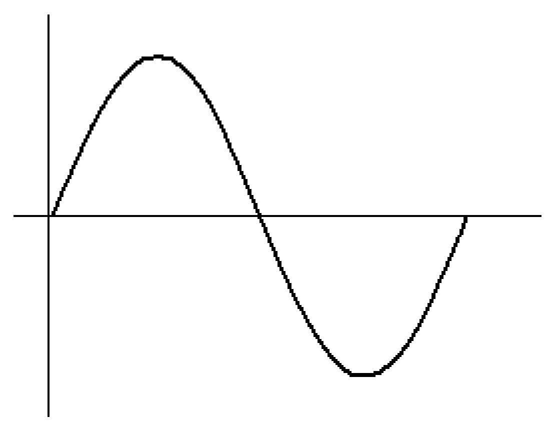


Fig. I-7. Corrections Needed for H1.

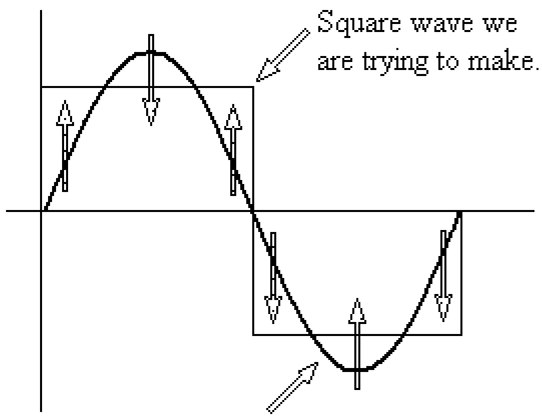


Fig. I-8. Adding H3 at 1/3 Strength to H1.

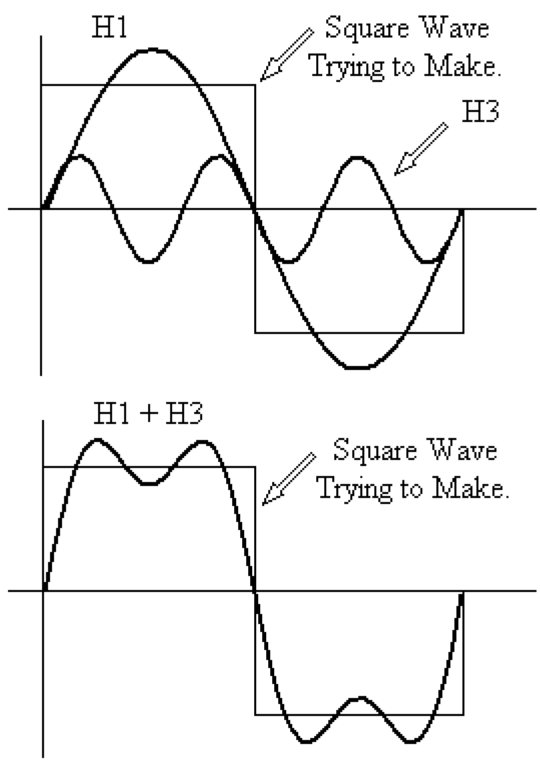


Fig. I-7 shows the shortcomings of our first harmonic H1. The square wave is included in the background so we can compare the two.

The first part of H1 (left edge) is too low. It needs to come up a bit. See the first arrow at the left in Fig. I-7. Similarly, the crest is too high. The crest needs to come down some. Then, the section just to the right of the crest needs to be taller. The trough needs similar corrections, but in reverse directions.

The vertical arrows in Fig. I-7 indicate all the major corrections necessary to improve the match with the desired square wave. The third harmonic has 3 crests and 3 troughs in just the right places to make these corrections.

See the upper diagram in Fig. I-8 for a sketch showing H1 and H3. We use H3 with an amplitude of 1/3. We might say we use the ingredient H3 with 1/3 of a cup. Note that we skip over ingredient H2. Remember, we do not have to use all the harmonics or ingredients for a specific recipe.

The result of adding H1 at full amplitude and H3 at an amplitude of 1/3 is given in the lower diagram of Fig. I-8. You might ask why an amplitude of 1/3 for H3. The recipe calls for 1/3 by trial and error. Actually, a mathematical procedure can be used to arrive at the precise value of 1/3.

Nevertheless, our sketch indicates that 1/3 is a good value for the amplitude of H3. Don't try to add the waves precisely; strive instead for a qualitative understanding of the addition. Note that H3 interferes constructively at its first crest. This shores up the left end a bit. Then note that the first trough of H3 interferes destructively with the crest of H1. It pulls the crest down. Study the effects of the addition on the other sections.

Fig. I-9. New Corrections and H5 to Assist.

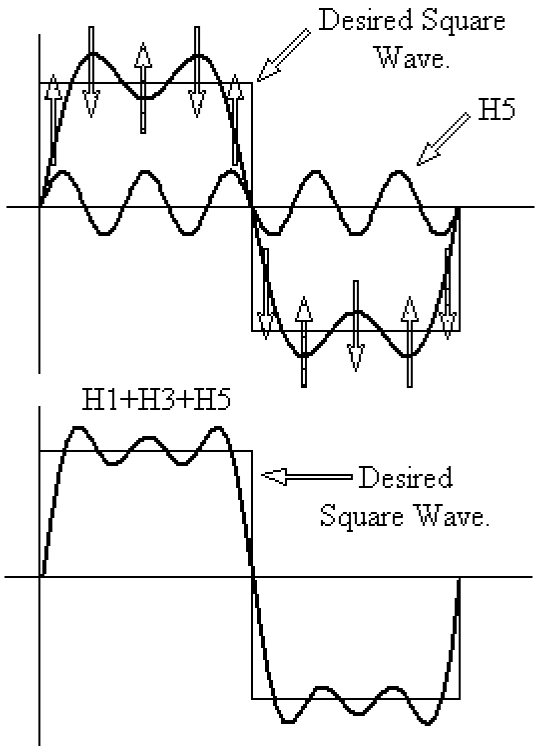
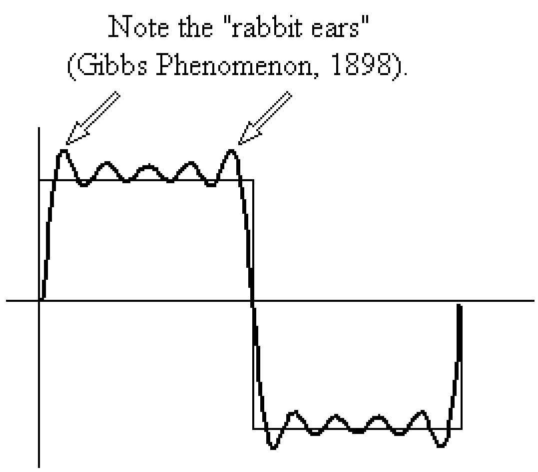


Fig. I-10. Sum of Odd Harmonics Up to H9.



We mark with vertical arrows the further corrections we need to make for our wave to look even more like a square wave. See the upper diagram in Fig. I-9. We find that we need 10 corrections, 5 up and 5 down. Note that they alternate. The fifth harmonic H5 has 5 cycles of crests and troughs. This harmonic is sketched in the upper diagram with an amplitude of 1/5. The corrections need to be gentle so we use this smaller amplitude for H5. Note that the 4th harmonic is skipped over for synthesizing a square wave.

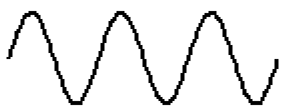
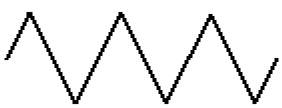
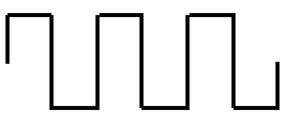
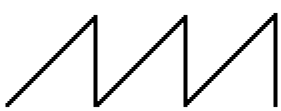
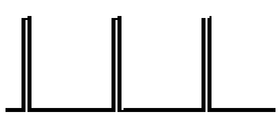
The result after adding the 5th harmonic to the sum of the fundamental and 3rd harmonic is given in the lower diagram in Fig. I-9. We are closer now in our approximation to the square wave. Observe that H1 alone has a single crest. The sum of H1 and H3 gives a "two-bump" crest and the sum of H1, H3, and H5 gives a "three-bump" crest. Notice also that the trough has these three small bumps but reversed. See if you can sketch the waveform resulting from adding in H7. This harmonic is used with an amplitude of 1/7. You should have 4 bumps in the first half of the wave. Fig. I-10 shows the result when H7 and H9 are included. The result has 5 bumps in the crest region. Note that we added 5 odd harmonics and have 5 bumps. The amplitude used for H9 is 1/9.

The sum wave begins to look more and more like a square wave. The prescription calls for using the odd harmonics, i.e., sine waves with frequencies f, 3f, 5f, 7f, 9f, and so on. The specific amounts to add (amplitudes) are 1, 1/3, 1/5, 1/7, 1/9, and so on respectively. As many more harmonics are added, the wrinkles get ironed out. The overshoot at the edges were discovered by Gibbs. They do not get ironed out; however each "rabbit ear" is squeezed shut as an infinite number of sine waves are added. The result is a perfect acoustical match.

We successfully constructed a square wave from the harmonic series. This is called *Fourier synthesis*. *Fourier analysis* is the breaking down of a periodic wave into its harmonics. Since any tone held by a singer or musical instrument is a periodic tone, it can be Fourier analyzed, usually by scientific instrumentation. The harmonics above the fundamental are referred to as overtones, as discussed earlier. Each harmonic is also called a partial (part of the complete tone).

The simple waveforms introduced earlier can be analyzed by theoretical methods of mathematical physics. The results for these simple geometrical waveforms are given in Table I-1 below (first nine harmonics). The partial-wave components (partials) are not always lined up with the fundamental for adding as in the case of the square wave. They need to be shifted left or right by various degrees, whatever the recipe calls for. The amount shifted is given by the phase. However, these are not listed since the more important ingredient in hearing is the amplitude information.

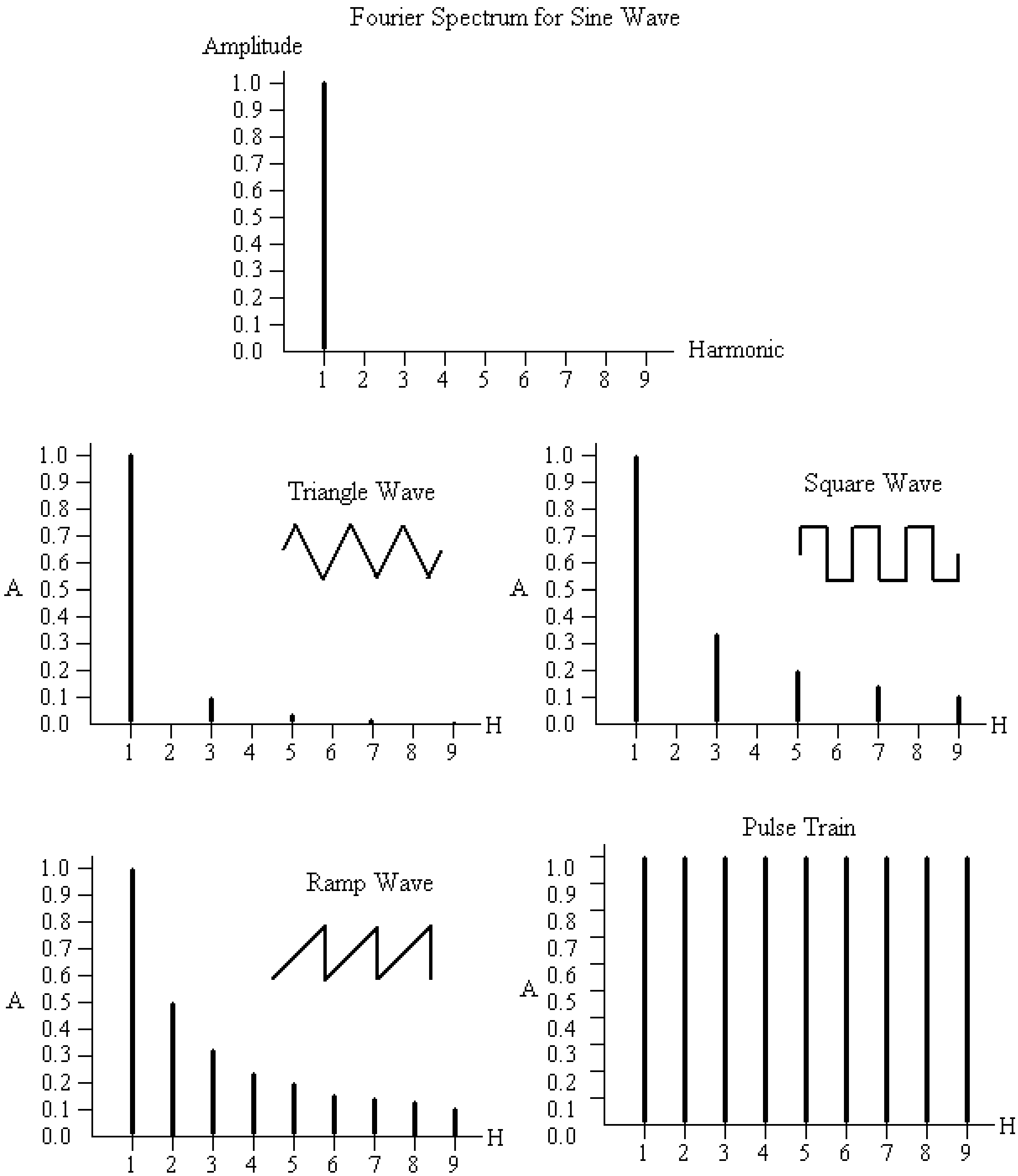
Table I-1. Five Periodic Waveforms and Their First 9 Fourier Amplitudes.

Waveform	Shape	H1	H2	H3	H4	H5	H6	H7	H8	H9	
Sine		1	0	0	0	0	0	0	0	0	Simple
Triangle		1	0	1/9	0	1/25	0	1/49	0	1/81	Weaker Partials
Square		1	0	1/3	0	1/5	0	1/7	0	1/9	
Ramp		1	1/2	1/3	1/4	1/5	1/6	1/7	1/8	1/9	Stronger partials
Pulse Train		1	1	1	1	1	1	1	1	1	Rich

Fourier Spectra

The results of Fourier analysis are most conveniently expressed in a bar graph. The plot of the relative amplitudes is often referred to as the *Fourier spectrum* for the periodic wave. The Fourier spectra for the five waveforms in Table I-1 are given in Fig. I-11. Note that the Fourier spectrum for the sine wave is just one harmonic, the fundamental or sine wave itself.

Fig. I-11. The Data in Table I-1 Expressed in Graphical Form.



Some Questions

Fill in the circles below for the correct answers.

Timbre	H2				H3				H4				H5			
	0	1/4	1/2	1	0	1/9	1/3	1	0	1/16	1/4	1	0	1/25	1/5	1
Sine	○	○	○	○	○	○	○	○	○	○	○	○	○	○	○	○
Triangle	○	○	○	○	○	○	○	○	○	○	○	○	○	○	○	○
Square	○	○	○	○	○	○	○	○	○	○	○	○	○	○	○	○
Ramp	○	○	○	○	○	○	○	○	○	○	○	○	○	○	○	○
Pulse Train	○	○	○	○	○	○	○	○	○	○	○	○	○	○	○	○

Check the appropriate boxes below.

Description	Sine	Triangle	Square	Ramp	Pulse Train
No Even Harmonics	☐	☐	☐	☐	☐
Same Amount of Each Harmonic	☐	☐	☐	☐	☐
The Amplitude of H4 is 1/4	☐	☐	☐	☐	☐
The Amplitude of H5 is 1/5	☐	☐	☐	☐	☐
The Amplitude of H5 is 1/25	☐	☐	☐	☐	☐

--- End of Chapter I ---

J. The Moogerfooger

The Physics of Sound and Music, Auditorium in Robinson Hall

October 24, 2002, Photo by former Office Assistant Ruth Maconi

In this chapter we will discuss some general descriptions of sound that are relevant for room acoustics. This material will build on our earlier studies of sound, mainly reflection. Then we will return to the Fourier spectrum, which we have just covered. We will see that some sounds such as bells and gongs include frequencies that are not part of the harmonic series. These sounds are not truly periodic so the harmonic series is not enough to describe them. Such a sound with additional frequency components in its spectrum is call an *inharmonic* sound, one not fully described by its harmonics. These sounds fall under the topic called *inharmonicity*.

Reverb

Reflection is one of the basic properties of waves. When sound waves reflect, we hear echoes. This is a very important property of sound in closed rooms and halls. The various reflections we hear define the acoustics of the room. A room excessive in echoes is not desirable. Enclosures such as basements and dungeons come to mind. Too little reflection gives a dry sound like we are out in an open field.

The speed of sound is 340 m/s = 1100 ft/s. Let's round this off generously to 300 m/s = 1000 ft/s. Sound travels 1 ft in one-thousandth of a second, i.e., 1 ms. It travels 3 ft, which is roughly 1 m, in 3 ms. Someone sitting a couple of meters away from you will hear your voice about 6 ms later. We can take this to be 5 ms since we are estimating. The sound waves leaving you also travel to and reflect off the walls, floor, and ceiling. If you have a rug, the reflections from the floor will be decreased significantly.

Sound hitting a wall nearly a couple of meters away, and reflecting to travel a similar additional distance to reach the hearer may get there 5 ms too late. Sound hitting the ceiling which is farther away than the nearby wall may arrive 10 ms too late. Sound going across the room to the far wall may reach the hearer 15 ms too late. All of the reflected sounds are softer of course. They arrive at the hearer within 20 ms. This makes the room *intimate*. If the first reflected sounds at a small concert hall start coming in to the observer before 20 ms, even though the reflections will continue, the concert hall is judged to be intimate, though not as intimate as an office.

We have learned to expect reflected sounds in rooms and halls. We also get reflections of reflected sound. This can go on for some time, especially in a dungeon. Such persistent reflected sound is called *reverberation* or *reverb*. A strict definition for reverberation or *reverberation time* is the time it takes for the reflections to die down to one millionth of the initial intensity heard by the observer. You might think this is too much to ask for. But our ears are not impressed unless great changes in sound occur. This is necessary if you want to be able to hear a whisper and later a band playing (with the same detector, your ear). Dropping a millionth in intensity is like going from a full blaring orchestra to soft background music or from a forceful voice to a whisper. We will see later that our ears are capable of hearing a change in intensity of more than one trillionth (i.e., a millionth of a millionth)!

To hear some reverb, cup your hands together and speak into them. Although reverberation time is typically too short for us to measure easily, we know when we hear it. You can hear the difference between talking into your hands and talking out in the open. You can imagine the sounds in caves, basements, and dungeons. These latter sounds have much reverberation.

There is a toy called the *Zube Tube* which has a spring about a meter in length inside a tube about the same length. As you talk into it, the sound waves excite the spring. The combination of an excited spring and a cylindrical cavity provides for reverberation. Springs also work in electronic circuits. Sending music in electronic form through a spring causes reverb. Reverb circuits have been designed to enhance the reverberation of the music we hear. You might have a reverb control on your stereo. The circuit introduces reflections electronically and the sound that comes out has more reverb in it. Your music sounds much better than it would if you depended solely on the natural reverb of your room.

Room Acoustics

Room acoustics is very complicated. The experience of sound in a room depends on so many things. The central feature to consider is reverb of course. However, reflections are dependent on where you are in the room, the distance you are from the source, the things in the room, the materials of the walls, etc. Certain materials absorb sound better than others. Special tiles are placed in practice rooms in UNCA's Department of Music to limit sound. These tiles absorb sound so that others in neighboring rooms and the outside halls will not be disturbed as much. Curtains and rugs also absorb sound relatively well. The smoothness of the walls is important. The shape of the room needs to be considered. Remember the dramatic effects in the whispering chamber. The height of the ceiling is crucial. The seats and the people also affect the outcome.

Much goes into the design of auditoriums as you can imagine. Since such planning is so complex, it is really an art, with design parameters evolving throughout the years. The reverberation time gives us a compact way to assess the acoustics of an enclosure. Since some frequencies may reflect better than others, reverb is often reported at specific frequencies such as 125 Hz (low range, bass), 500 Hz (midrange), and 2000 Hz (high range). We are fortunate that all frequencies travel at the same speed. Remember that the speed of sound depends on the medium. Otherwise, we would be in lots of trouble. Some sounds would then arrive out of sync with their supporting sounds.

Two basic descriptions of room acoustics are *intimacy* and *liveness*. We have touched on these already. They are defined below.

Intimacy - presence of reflections reaching hearer within 20 ms after the direct sound.

Liveness - presence of sufficient reverb.

Remember when we discussed the small room, reflections start arriving at the hearer 5 ms after the direct sound, surely less than 20 ms. We say the room is intimate acoustically, regardless of the conversation you are having. However, small rooms tend to be used for actually "intimate" conversations for this very reason.

Now let's go to a larger room like a school auditorium such as the one in *Lipinsky Hall* on campus. Does the hall satisfy the acoustic definition for intimacy? It doesn't have to excel in intimacy like an office, but reflections need to get to us around the 20 ms limit. Remember that sound travels 1 ft (1/3 meter) in 1 ms (millisecond or thousandth of a second).

Suppose we are in the back somewhere 100 ft from the performers. The direct sound gets to us in 100 ms (1 millisecond for every foot to travel). This is one-tenth of a second. The reflections better start arriving 20 ms after this. This means that some of the sound going in other directions better find something to reflect from so that the total distance of the longer path is around 20 ft. The indirect sound waves hitting a wall or the ceiling will travel too far (beyond the allowed 20 ft) by the time it reaches us (in the back) and thereby fail to meet the criterion for intimacy. We have to sit up front for the intimate experience.

On the other hand, the science auditorium has smaller dimensions than the auditorium in Lipinsky. The science auditorium has 180 seats, while the Lipinsky auditorium has about 600. The distance from the front to the back of the science auditorium is only about 40 ft. The sound gets to someone three-fourths of the way toward the back in about 30 ms. Other indirect sound waves have no trouble reflecting off the nearby walls or ceiling, taking paths less than an additional 20 feet. So we easily get reflections within 20 ms. We consider the auditorium "acoustically" intimate, although not as intimate as a smaller classroom.

Other acoustical descriptive words include *fullness*, *clarity*, *warmth*, and *brilliance*. Refer to Table J-1 below. If there is little reverb, we have clarity. This is good for playing Bach. The crisp, clear music of the baroque period (1600-1750) sounds best in rooms where the reverb times are less than 1.5 s. Good reverb times for music of the classic period (1750-1800) are in the range 1.6 to 1.8 s. Reverb times between 1.9 and 2.1 s are best for the full satisfying sounds of the romantic period (1800-1900).

Table J-1. Acoustical Descriptions for Different Amounts and Types of Reverb.

Little Reverb	Much Reverb		
Overall	Low Frequencies	High Frequencies	Overall
Clarity	Warmth	Brilliance	Fullness

For much reverb at low frequencies, the rich bass sounds envelop us with musical *warmth*. Such reverb can enhance the mellow sound of a cello playing in a small ensemble. Reverb at higher frequencies, i.e., *brilliance*, emphasizes instruments that play higher pitches (like flutes and piccolos). The word brilliance, like the others, is subjective. We use words by convention in order to help us describe room acoustics in some standardized way. In all, there are nearly 20 such terms. Two others are *ensemble* (ability of performers to hear each other) and *blend* (a satisfactory mixing of sound for the entire audience).

Reverberation data for four concert halls can be found in Table J-2. The reverberation time is given at three frequencies: 125 Hz (low range, bass), 500 Hz (midrange), and 2000 Hz (high range). The time is given in seconds. Note the shorter reverb times for Philadelphia's *Academy of Music* when compared to Boston's *Symphony Hall*. The great Leopold Stokowski (conductor of the *Philadelphia Orchestra* from 1912-1938) developed a masterful sustaining technique to counter the short reverberation times at the *Academy of Music*. On the other hand, Maestro Serge Koussevitzky of the *Boston Symphony* utilized an abrupt technique to compensate for the longer reverberation times at *Symphony Hall*. Their respective conducting techniques did not serve them well on visiting appearances due to the different characteristics of the halls. So they stopped the visits.

Stokowski conducted the music for the famous Walt Disney film *Fantasia* (1940). Under the direction of Stokowski and later Eugene Ormandy, the *Philadelphia Orchestra* became one of the world's finest orchestras. Eugene Ormandy was known for his interpretation of romantic music such as that of Rachmaninoff. There are old recordings of Eugene Ormandy conducting with Rachmaninoff at the piano, performing Rachmaninoff piano concertos. Ormandy was the conductor of the *Philadelphia Orchestra* from 1938 to 1980.

Table J-2. Acoustical Characteristics of Four Concert Halls (When Occupied).

Concert Hall	Location (City)	Year Built	Number of Seats	Reverberation Time in Seconds		
				125 Hz	500 Hz	2000 Hz
Academy of Music	Philadelphia	1857	2827	1.4	1.5	1.3
Symphony Hall	Boston	1900	2625	2.0	1.9	1.7
Carnegie Hall	New York	1891	2804	2.3	1.8	1.6
Kennedy Center	Washington	1971	2760	2.0	1.6	1.2

L. Beranek, *Concert and Opera Halls: How They Sound* (Acoustical Society of America, Woodbury, NY, 1996), for the above data and conducting story in text.

Inharmonicity

Inharmonicity refers to frequency components of a wave that do not fit into the harmonic series. For example, a 100-Hz ramp wave has Fourier spectral components such as 200 Hz, 300 Hz, 400 Hz, and 500 Hz. These are overtones and members of the harmonic family beginning with the fundamental 100-Hz sine wave. Imagine this ramp wave, but now with additional frequency components 171 Hz and 239 Hz. The wave is no longer a ramp. These frequencies are not in the harmonic series that starts with 100 Hz. These strange inharmonic additions are also called *inharmonic partials*. The overall sound is no longer periodic. If you keep the sound going electronically, the lack of periodicity is experienced as a lack of a well-defined pitch.

We will use a *balanced modulator*, also called a *ring modulator*, to illustrate inharmonicity. The balanced modulator accepts two waves and sends out two modified waves. The modification is understood best by considering input sine waves. When a balanced modulator accepts two pure (sine) input tones, it sends out the sum and difference tones. This is illustrated in Fig. J-1a below.

Fig. J-1a. Balanced Modulator.

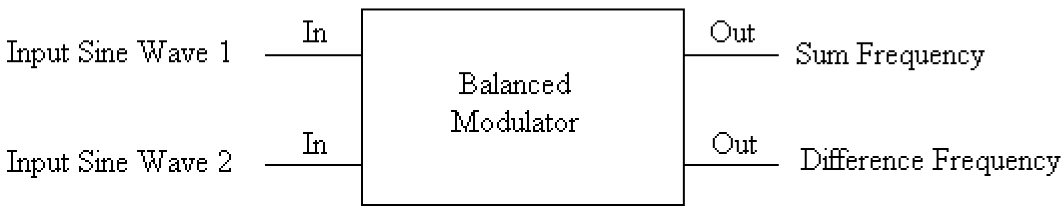
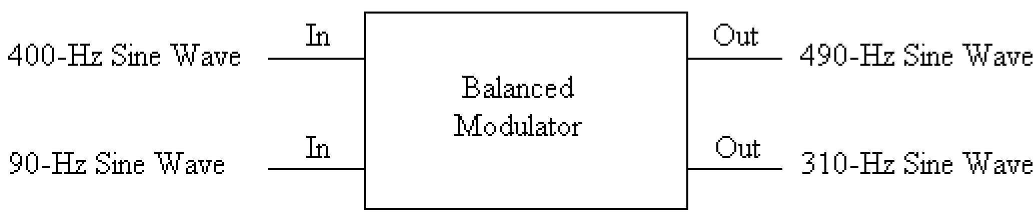


Fig. J-1b. Balanced Modulator Example.



A specific example with numbers is given in Fig. J-1b. The input frequencies are 400 Hz and 90 Hz. The output frequencies are the sum (400 + 90 = 490 Hz) and difference (400 − 90 = 310 Hz). It's important to note that all of these tones are sine waves. The rule applies to sine waves. If you send in one wave which is not a sine wave, then you must decompose it first into sine waves using Fourier analysis. Each harmonic is then added and subtracted with the other input sine wave. Our next example presents such a case.

Let's balance modulate a 600-Hz sine wave with a 50-Hz square wave. We need to think in terms of sine waves so we break the square wave into its harmonic spectrum. The square wave consists of odd harmonics, so we have f, 3f, 5f, and so on. Since f = 50 Hz, the first harmonic is 50 Hz. The third harmonic is three times this, which gives 150 Hz. The fifth harmonic is 250 Hz. Note that the amplitudes of the higher harmonics get smaller. In synthesizing a square wave (last chapter), we used an amplitude of 1/3 for the third harmonic and an amplitude of 1/5 for the fifth harmonic. Since these amplitudes get even smaller and smaller for the higher harmonics beyond the fifth, we can approximate the square wave stopping at H5.

The Fourier sine components for the square wave are listed below in Table J-3. Note that only the first few harmonics of the square are represented since the higher harmonics contribute even less to the sound. The even harmonics are missing for a square wave. Each harmonic of the square wave must be treated separately with the other input, the sine wave. The sum and difference for each harmonic is given in Table J-3. The results are given in Fig. J-2. The relative amplitudes of the original square-wave harmonics are preserved. The frequencies in the spectrum do not form part of a harmonic series. In fact, there is no fundamental. The spectrum is inharmonic.

Table J-3. Balanced Modulation of a 600-Hz Sine Wave and a 50-Hz Square Wave.

600-Hz Input Sine Wave	Square-Wave Harmonic (Hz)	Sum Wave Output (Hz)	Difference Wave Output (Hz)	Relative Amplitude
600	50 (H1)	650	550	1
600	150 (H3)	750	450	1/3
600	250 (H5)	850	350	1/5

Fig. J-2. Spectrum of Inharmonic Output from Above Table.

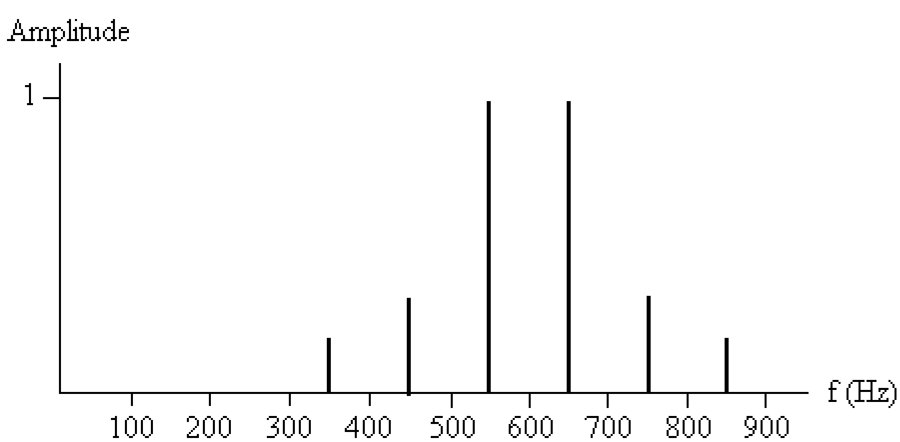


Fig. J-3a illustrates the input sine waves for the problem we just considered. The spectrum for the square wave and the sine appear together. The spectral components are the dotted vertical lines. Fig. J-3b shows the resulting output after balanced modulation. The frequency of each harmonic for the square wave (Wave 1) is added and subtracted from 600 Hz, the frequency of the sine-input wave (Wave 2). The spectral components (inharmonic) are the solid vertical lines.

Fig. J-3a. Input Waves into Balanced Modulator.

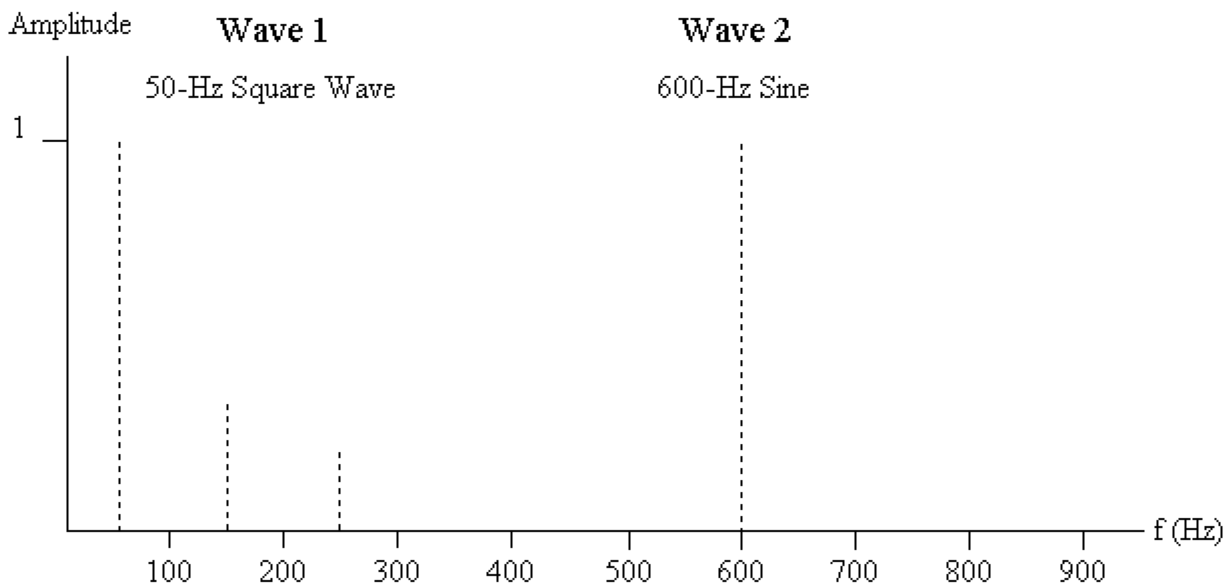
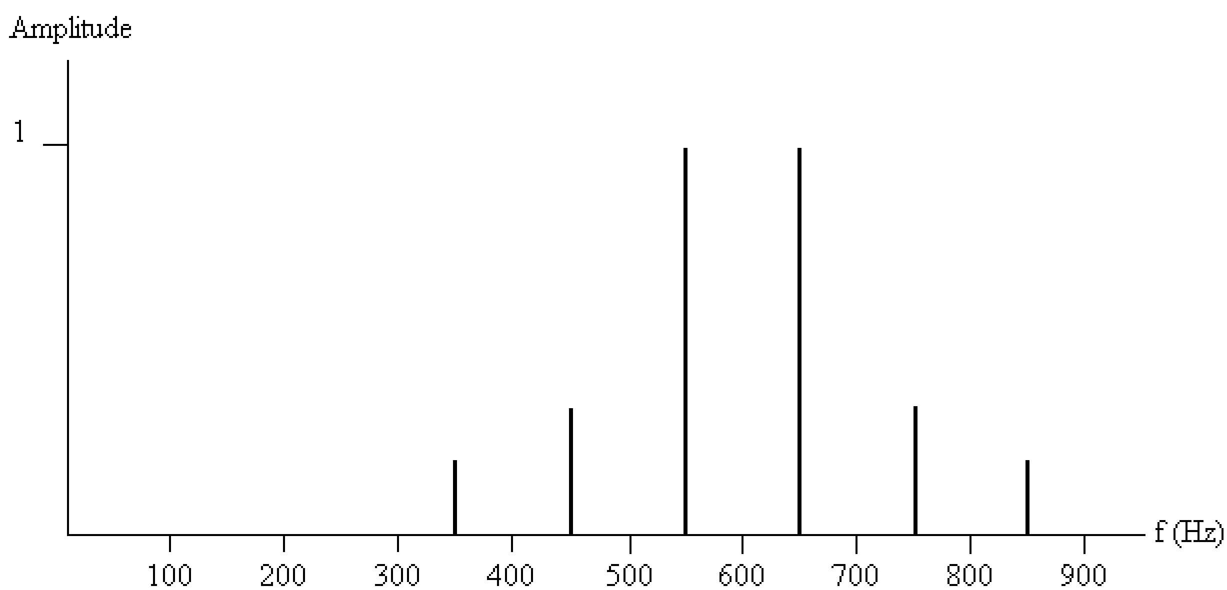


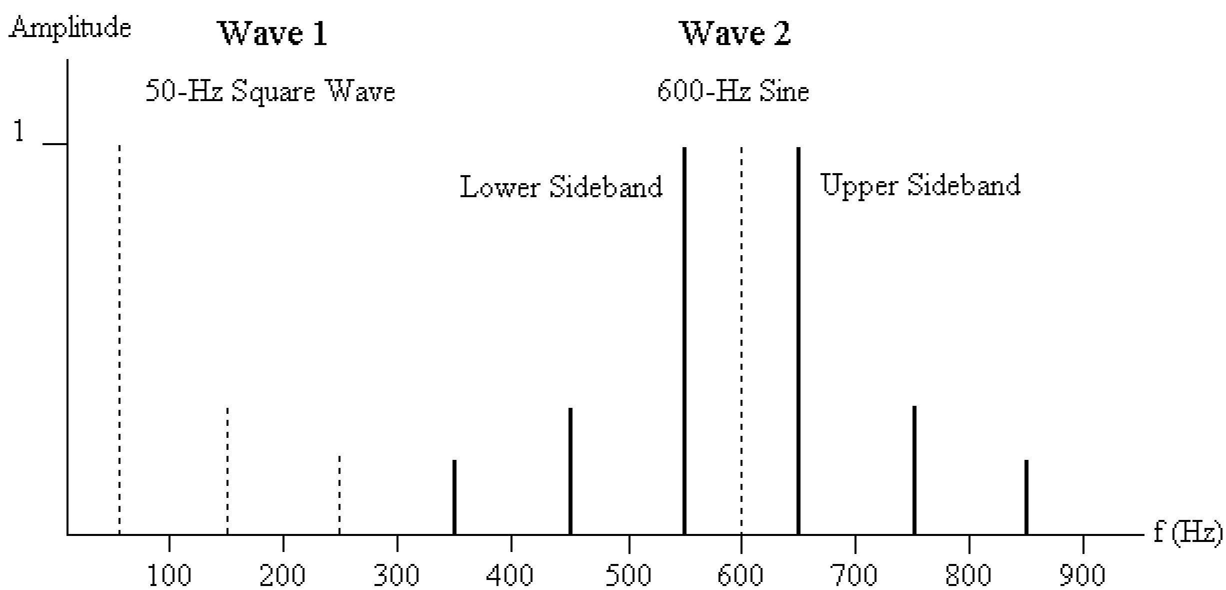
Fig. J-3b. Balanced-Modulated Output Spectrum for Above Input Waves.



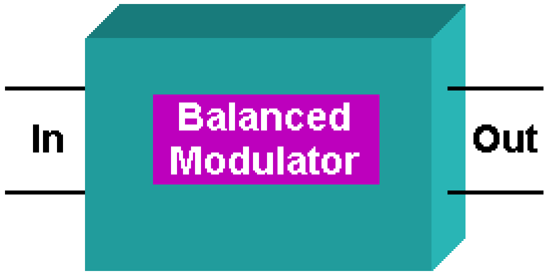
Finally we combine all the steps into one diagram, Fig. J-4 below. The input spectra are dotted, while the output spectrum is solid. Note the mirror-reflection symmetry about the 600-Hz sine wave. The group of frequency components to the right of the 600-Hz sine wave is called the *upper sideband*. The group of frequencies below the 600-Hz sine wave is called the *lower sideband*.

The output spectrum does not consist of a fundamental and its associated overtones as we find for periodic waves. In fact, as we noted, there is no fundamental. The spectral components 350, 450, 550, 650, 750, and 850 Hz do not exhibit the characteristic relationships of the harmonic series, i.e., f, 2f, 3f, 4f, etc. The balance-modulated sound is inharmonic. Any sound that is not periodic is technically inharmonic. However it is interesting to note sounds that are nearly periodic. Such examples of inharmonic sounds include bells, chimes, and gongs. The Moogerfooger 102 is built on principles of blalanced modulation. See your Power Notes for more on the Moogerfooger.

Fig. J-4. Balanced Modulation (50-Hz Square Wave and 600-Hz Sine Wave).



Some Questions

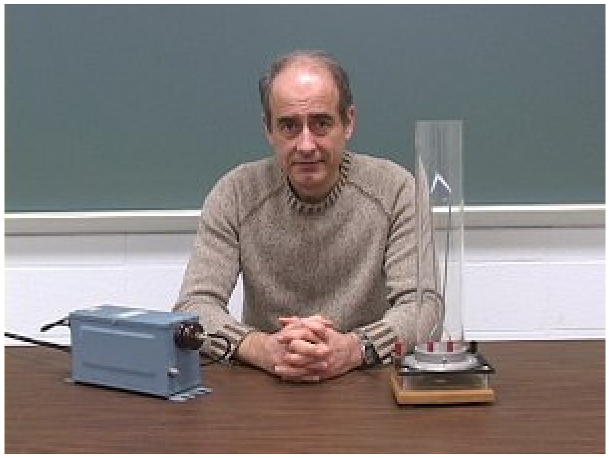


A 1050-Hz sine wave is balanced modulated with a 50-Hz ramp wave. Give the difference and sum frequencies for each Fourier component of the ramp wave balanced modulated with the sine wave. Enter amplitudes as fractions such as 1/2 when appropriate. Do NOT use decimals. Make sure you give the difference and sum frequency even though the amplitude may be zero. The zero amplitude means that these components will not be present in the output wave.

Part	Harmonic	Difference Wave (Hz)	Sum Wave (Hz)	Relative Amplitude	Hints
a	1				1050 - 50, 1050 + 50, 1
b	2				1050 - 100, 1050 + 100, 1/2
c	3				1050 - 150, 1050 + 150, 1/3
d	4				1050 - 200, 1050 + 200, 1/4
e	5				1050 - 250, 1050 + 250, 1/5

--- End of Chapter J ---

K. The Laws of E&M

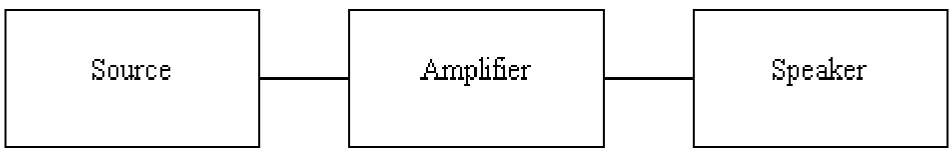


There are three basic stages of sound reproduction. These are illustrated in Fig. K-1. Dividing the challenging task of sound reproduction into modular steps is good engineering practice. We have division of labor and specialization of function. There is no duplication of jobs. Also, the source can then be interchanged. It might be a CD player, tape deck, or microphone.

The signal from any of these sources needs to be enhanced in order to drive a speaker membrane. This is accomplished in the second stage by the amplifier. In the third stage we find the speaker so we can hear the sound. A single speaker gives *monophonic* or *monaural* sound.

Two speakers, each with a somewhat different channel of sound information, provide for a three-dimensional experience of sound. This is *stereophonic* or *stereo*, which we are very accustomed to. A set of four speakers, each with a separate channel, is called a *quadraphonic* system. Most people are pleased enough with stereo and often settle for stereo systems.

Fig. K-1. The Three Basic Stages of Sound Reproduction.



Since the time of *hi-fi* (*high-fidelity*), when sound reproduction became quite realistic, but still monophonic, serious music lovers have taken pride in carefully picking out components for their sound systems. They might buy the best record player (e.g., in the 1950s) built by one manufacturer, the amplifier built by another, and speakers from still another.

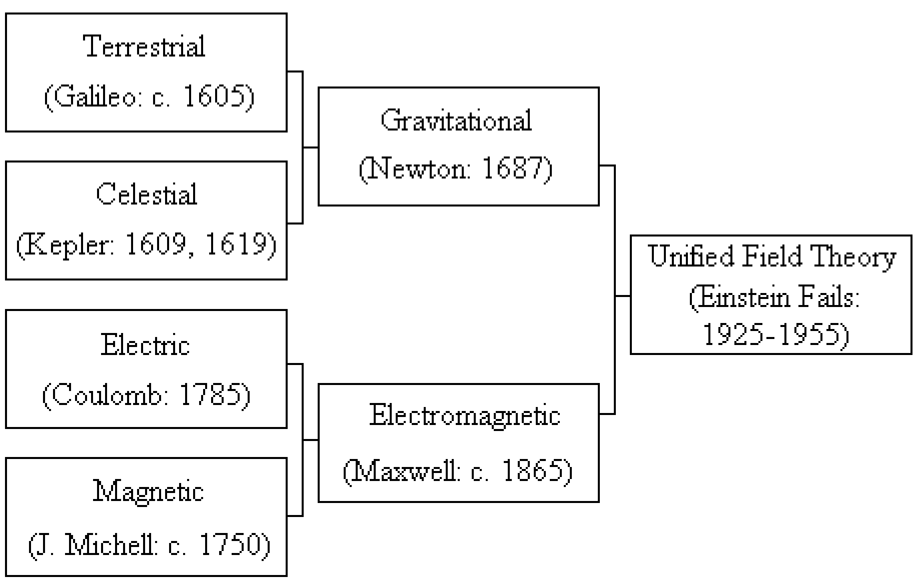
In the late 1950s, as stereo became accessible to the masses, thousands of monophonic records had to be replaced by the new stereo records. The standard setup was a turntable, amplifier, and two speakers. A tuner to pick up the new stereo radio stations was also essential. The older reel-to-reel tape recorder was handy for making good stereo tapes off the air. This would mean three source units: the turntable, tuner, and tape recorder. No one serious would dare buy a record changer that stacked records and dropped one automatically on top of the last one when playing the next record. The turntable, which can only handle one record, offered the best protection for your fine records like the *Angel* recordings of the Schumann four symphonies. You would want a good amplifier, capable of at least 35 watts per channel and two fine speakers. Each speaker unit would contain a large speaker for long-wavelength low frequencies and a smaller one for the higher pitches. Those with systems including 15-inch bass speakers would look down on those with 12-inch speakers. If you had 8-inch speakers, you were pretty far down the list. The wattage per channel was also a measure of status. One of the author's friends in graduate school had a 200-watt system in a small apartment during the 1970s.

We need to understand basic concepts in electricity in order to understand the 20th-century developments in sound reproduction. Electricity involves using a fundamental force of nature, the electric force, to our advantage. Let's look at basic forces in nature before pursuing our study of the electric force in some detail. Physicists have found very few fundamental forces in nature. You might say that these basic forces are responsible for holding the world together. Physicists have been amazed to discover that forces once thought to be different and separate, were found to be related in an intrinsic way.

For example, the forces that govern heavenly motion were once thought to be different from earthly forces. Kepler (early 1600s) discovered laws describing the celestial orbits of planets while Galileo, around the same time, found what appeared to be a different law describing terrestrial motion. Newton showed that the (*celestial*) force that keeps the moon in orbit around the Earth is the same (*terrestrial*) force that causes apples to fall to the ground. He called this the force of *universal gravitation*. Newton published this in his famous work, the *Principia*, in 1687.

Other forces under investigation, mainly in the 1700s and 1800s, were *electric* forces and *magnetic* forces. These were likewise shown to be manifestations of a single force (c. 1865), now called the *electromagnetic force*. The early historical development of discovering the unifying forces in nature is summarized in Fig. K-2. Einstein was so impressed by nature's unity of forces that he set out to search for the one force that unified all, the *unified field theory*. He failed to discover this. Also, there are two nuclear forces that need to be dealt with. There has been considerable success in unifying all the basic forces in nature except gravitation since 1967. Work is in progress now to unify all. String theory might be the answer. Of course, there is a chance that unification is impossible.

Fig. K-2. Unification of Basic Non-Nuclear Forces in Nature.



Engineering applications of gravitation and mechanical systems were made centuries before Newton presented his elegant mathematical laws in 1687. The associated inventions include the pulley, lever, and wheel-axle assembly. However, with the electromagnetic force, the engineering applications came after the theoretical laws were laid down. Mechanical systems can be seen and touched. We can handle levers and pulleys. We can experiment directly with our hands. But electricity is more subtle.

After the laws of electricity and magnetism were written down in 1865, the stage was set for inventors and engineers to develop an arena of new technological devices. A new source of power was unleashed, electrical power. Electrical engineering became a specialized profession. Electronics became an area of applied science. These developments made possible the science of sound reproduction. The laws of electricity and magnetism are essential for sound reproduction. Electronic circuits are joined with mechanical components to make microphones, tape decks, speakers, and other sound devices. At this point, we would like to learn more about this new duet of basic forces, the electric and magnetic forces, that makes all of this possible.

The electric force can produce *electricity*, the flow of charged particles. *Charge* is similar to mass in the sense that it is related to a fundamental force in nature. Mass is the source for gravity (or gravitation) and mass also responds to gravity. For example, the large mass of the Earth produces gravity near the surface of the Earth, and other masses, like rocks and baseballs, respond to this "force field" of gravity. The Earth and a baseball attract. We see the baseball move and not the Earth, because the baseball is light and the Earth is so huge.

Another fundamental force in nature besides gravitation, is the electric force. Charges create electric "force fields" and other charges respond to these fields. Once again, the lighter masses do more of the moving around. The electric force can be attractive or repulsive. There are two types of charges, designated as "+" and "−," making this possible. Two like charges, e.g., two plus or two minus charges, repel while unlike (a plus and a minus) attract.

The building blocks of matter are molecules and atoms. However, atoms are made up of a nucleus containing two types of particles (protons and neutrons) and a cloud of surrounding electrons. The elementary building blocks of matter are therefore the protons, neutrons, and electrons. These give us the key to understanding where charges come from. Table K-1 below lists these three elementary particles. The unit "amu" stands for *atomic mass unit*, a very small unit of mass.

Table K-1. Elementary Building Blocks of Matter.

	Proton	Neutron	Electron
Mass	1 amu	1 amu	0.0005 amu
Charge	+1	0	−1

In Table K-1 one unit of charge is taken to be the amount of charge on a proton. The masses of the proton and the neutron are about the same in value; however, both are much more massive than the electron (1836 electrons have about the same mass as 1 proton). The charges that move as electricity in wires are electrons. Electrons surround the nucleus in an atom. The number of electrons (minus charge) for any neutral atom is equal to the number of protons (positive charge) in the atom. The details of electron orbital motion and interaction with neighboring atoms are the subject of chemistry. In a metal wire, atoms are bound together forming a metallic solid. However, the outer electrons of the atoms are free to move around from one atom to another. This motion makes possible the production of electricity.

Electricity can be very dangerous. One should never play with electrical outlets at home. Electricity consists of small moving particles as we have seen. A current of these charges traveling in wires is the standard type of electricity. However, there is another kind of electricity called *static electricity*. There is no motion in this case (except for when sparks fly). To illustrate static electricity, rub a balloon against your arm and try to get it to stick on a wall (see Fig. K-3). The author stuck 100 balloons on walls in his second daughter's bedroom for her 4th birthday in 1990. Many of them stayed attached to the wall overnight. Rub a comb or ballpoint pen (plastic) against your sweater. Can you then pick up little bits of paper with the comb or pen? This is an example of the electric force at work.

Prepare two balloons by tying a string to each. Can you get the balloons to repel each other after rubbing them on your arm (see Fig. K-4). The balloons are said to be charged with static electricity. Each balloon picks up electrons (negative charges) when rubbed against your arm. Like charges repel each other. So the balloons repel each other because they have gained electrons. Ben Franklin called these charges negative, in the 1700s, before anyone knew that the charges were electrons. The electron was not discovered until 1897. Ben Franklin's choice forced scientists later to call the protons (with opposite charge) positive.

Normally, an object such as a balloon has just as many electrons as protons. The balloon is neutral. When you rub an object against another, sometimes electrons are transferred; the protons stay where they are. Two balloons with extra electrons repel since "likes" repel ("unlikes" attract). Rubbing silk with glass removes electrons from glass. Rubbing cat's fur with plastic removes electrons from the cat.

Fig. K-3. Static Electricity.

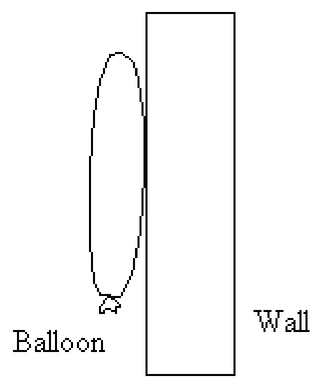
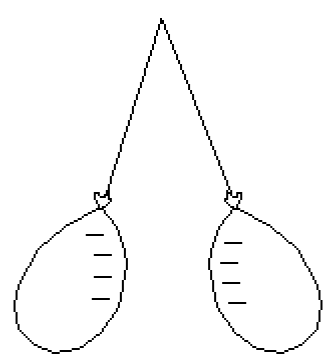


Fig. K-4. Electric Repulsion.

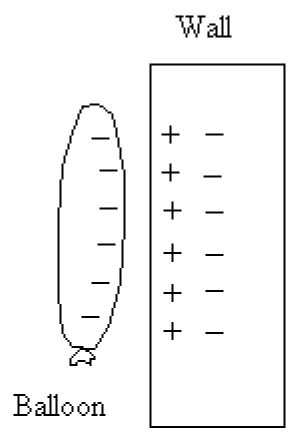


When you rub a balloon or comb against your hair, the balloon or comb picks up electrons, as we have noted. The negative charge on the balloon chases away electrons in the wall. These wall electrons are repelled by the electrons on the balloon. They move away from the edge of the wall (see Fig. K-5 at the right). The absence of electrons at the wall leaves the surface with a positive charge. The balloon's electrons are attracted to this positive charge and the balloon sticks to the wall. The electrons can't easily leave the balloon so the balloon stays stuck to the wall. If it is humid, the electrons can leave the balloon for the moist water droplets. The balloon then won't stay on the wall very long. The positive charge at the wall is induced by the balloon. This is called *electrical induction*. This also explains why a comb picks up bits of paper after you rub it on your sweater.

We can now consider magnetic forces by an analogy to electric force. Magnets attract and repel other magnets depending on the orientation of the ends of the magnets. We call one end north and one south, analogous to positive and negative. We find that "likes" repel. Two north poles won't pull together. They push each other away. Similarly two south poles repel. However, a north pole and south pole attract. See Fig. K-6 below. We have a similar rule here as the one we encountered in the electric case: "likes" repel and "unlikes" attract.

There is an important difference though. No one has ever found a magnet with one pole. All magnets have north and south poles. This is unlike charges, where we can find a sole plus charge such as the proton and a sole negative charge such as the electron.

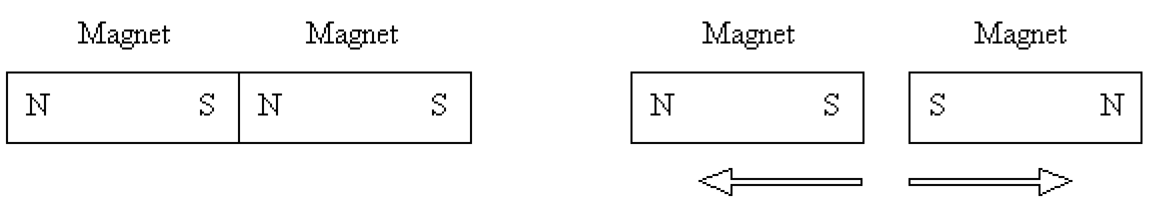
Fig. K-5. Electrical Induction.



Positive Charge Induced
on Surface of Wall.

Our next experiment will shed some light on why magnets have two poles. It will also show us that electricity and magnetism are related. However, some scientists still search for a magnet with one pole, a *monopole*. So far none have been discovered. This fact has become one of the laws of electricity and magnetism: *magnetic monopoles* do not exist! So if someone discovers one, that person will probably win a Nobel prize in physics.

Fig. K-6. Magnetic Attraction and Repulsion.



"Unlikes" Attract.
North and South Poles Stay Together.

"Likes" Repel.
The Magnets Move Away From Each Other.

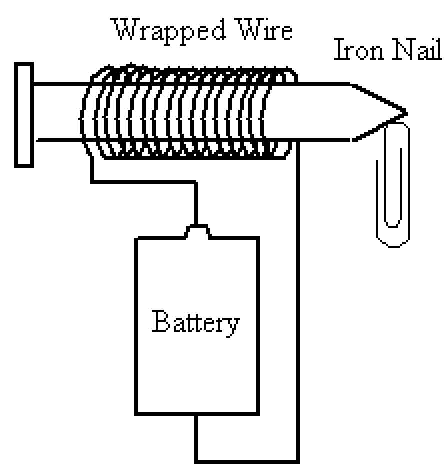
Speakers

Our next experiment will show us the connection between electricity and magnetism, give us insight why magnets have two poles, and provide the groundwork for understanding speaker systems.

If materials are available, take an iron nail and wrap thin insulated wire around it many times. Then connect each end of the wire to the ends of a battery, being careful not to touch the bare-wire ends or battery terminals. The wire and nail can get quite hot. If the insulated wire gets too warm, stop. Wrap more windings. Try to pick up a paper clip. The nail now acts like a magnet. See Fig. K-7 at the right. You have an *electromagnet*. You may find that a residual magnet remains after disconnecting the battery.

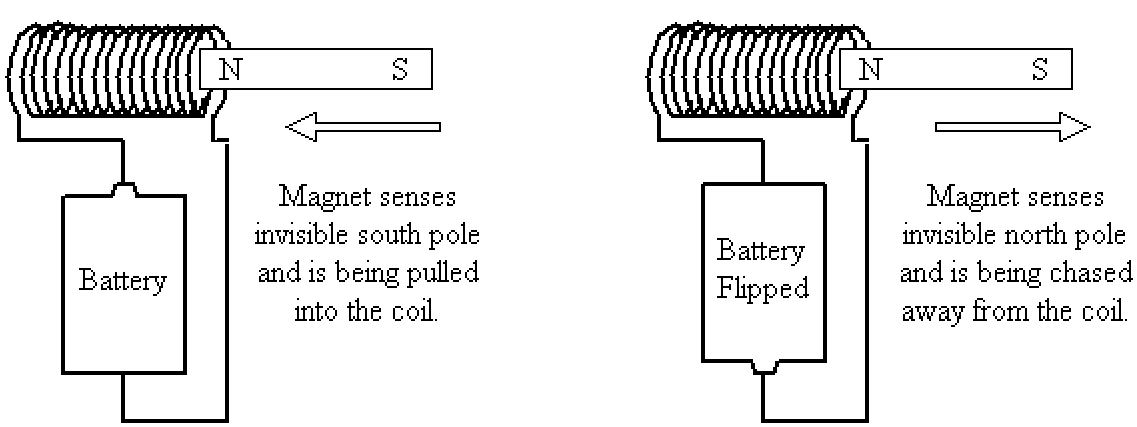
Our experiment shows that we can make a magnet using electricity. This suggests a unifying relationship between electricity and magnetism. It explains why scientists consider electric and magnetic forces as one fundamental (electro-magnetic) force rather than two. Now reflect on Einstein's struggle to look deeper into nature and find a similar relationship between gravitation and electromagnetism. If you don't see a connection, don't feel bad - neither did Einstein. And he studied the problem for the last 30 years of his life before he died in 1955.

Fig. K-7. Electromagnet.



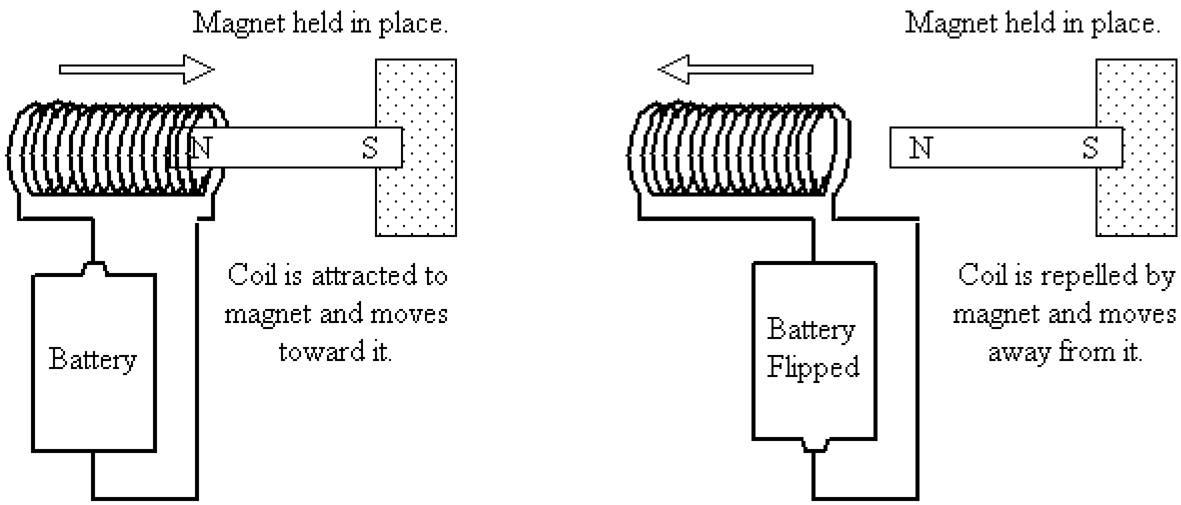
Now comes the real magic. We remove the nail and find that a compass responds in the vicinity of the ends of the wrapped wire. A force field exists although there is no magnet per se. We call this force field a *magnetic field*. We find that we can make a "ghost magnet" of the polarity of our choice by the way we attach the battery. See Fig. K-8 below.

Fig. K-8. Magnetic Fields.



In Fig. K-9 we perform a slight variation of our previous experiment. We fix the magnet by cementing the south pole into a wall. Now when the magnet is attracted to the "ghost magnet," it can't go toward it. So the coil comes to the magnet. The "ghost magnet" is free to move because the wires are not sturdy. When the battery polarity is reversed, the "ghost magnet" is repelled and moves away from the fixed magnet.

Fig. K-9. Magnetic Fields.



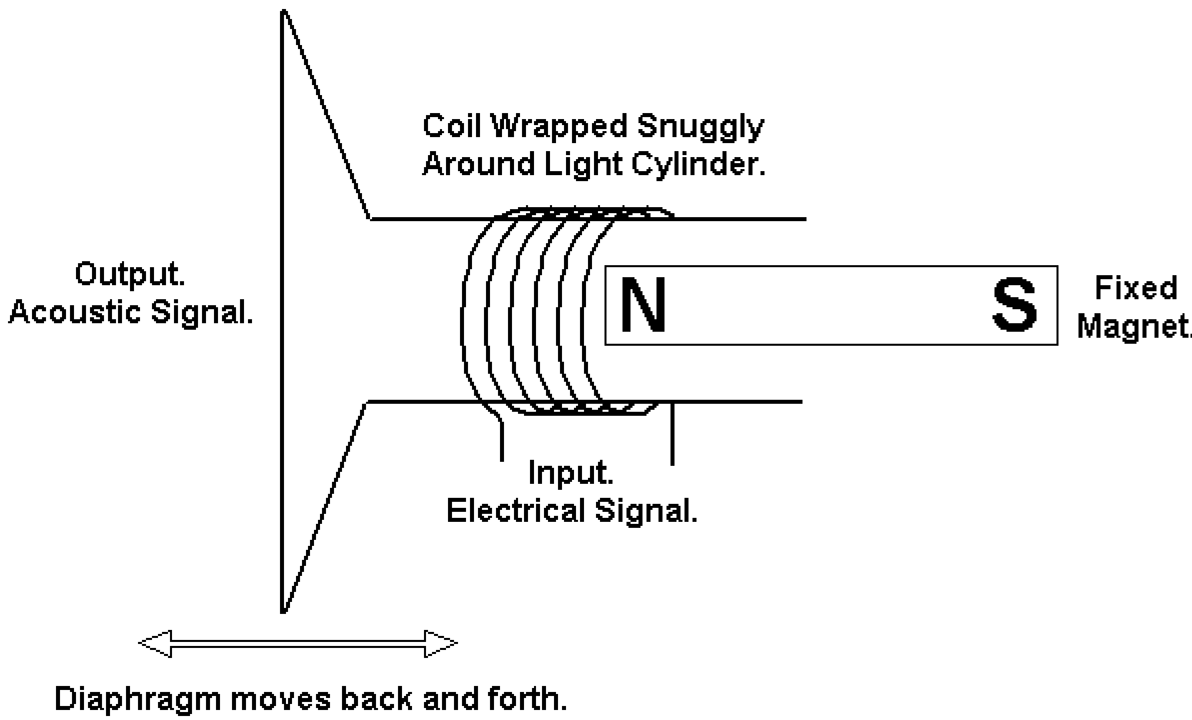
Newton discovered that for every action there is a reaction. When a force is applied, we can consider this the action, e.g., the "ghost magnet" pulls on the magnet. Then immediately there is a reaction force, the magnet pulls on the "ghost magnet." They both want to move. It's like the sun and Earth pulling on each other due to gravity. They both want to move. But the sun, which is more massive, resists moving (almost completely), while the Earth moves.

In our example, the fixed magnet cannot move, so the coil with its "ghost magnet" moves toward the magnet if the "ghost magnet's" south pole is facing the real magnet's north pole ("unlikes" attract). When the battery is flipped, the coil's north pole repels the magnet's north pole and the coil moves away. Our "ghost magnet" also suggests that a magnet should have two poles. The "ghost" does so because each end of the coil supports one pole. You can't have just one end of a coil.

Now imagine flipping the battery from right-side-up to upside-down back and forth 100 times a second. The coil would move left and right 100 times a second. Now attach a delicate membrane to the left end of the coil. This membrane will shake the air molecules 100 times per second and we will hear a 100-Hz bass tone. We have a *speaker*. We can't flip the battery that fast, but we can attach a pair of wires carrying electrical oscillations to the coil. We then convert an electrical signal to sound. That's what a speaker does!

We give the sketch for the speaker in Fig. K-10 below. It is the design for the common *dynamic speaker*. The input signal is electrical. Consider a periodic wave such as a sine wave. The electrical current oscillates in step with the crests and troughs of a sine wave. The crests push current in one direction, the troughs pull current in the opposite direction. The coil changes the magnetic field it produces in step with the crests and troughs. The changing magnetic polarity (north-south to south-north) results in pushes and pulls as the magnetic field interacts with the fixed magnet. The coil and cylinder vibrate back and forth. The diaphragm at the left vibrates the air and produces acoustic waves. We hear the sound of the sine wave.

Fig. K-10. Speaker Schematic.



Before we leave this section, we would like to point out that the physicist Ampère (1825) expressed in mathematical form how a current through a coil produces a magnetic field, a phenomenon first observed by Oersted in 1820. This is called *Ampère's Law*. It is the basic law of electricity and magnetism that is applied in the invention of the speaker.

Ampère's Law:
Current produces a magnetic field.

Microphones

Here we consider the invention of the *dynamic microphone*, the common microphone we encounter. There is a beautiful symmetry in nature that makes possible the invention of the microphone. The microphone is simply a speaker used in reverse. The laws of electricity and magnetism express this symmetry in nature. This means that if you talk into the diaphragm part of a speaker, the vibrating diaphragm will result in the production of an oscillating electrical signal. By this symmetry in nature, we mean that the cause-effect relationship, seen in the speaker, works backwards.

This counterpart law that says the reverse will work, was announced by Faraday in 1831 (Henry, an American, failed to publish his discovery in 1830). It is known as *Faraday's Law*. When you force the cylindrical coil to move back and forth near the fixed magnet, the coil "sees" the north pole of the magnet come toward it and then away from it. It "senses" a changing magnetic field. When the coil is away from the magnet, it "senses" little or no magnetic field. As it is suddenly brought over to the magnet, the coil envelops the magnetic field. The coil "experiences" a changing magnetic field inside itself. The result is a surge of current through the coil each time there is a change. When the coil leaves the magnet (decreasing magnetic field inside the coil), the current flows one way. When the coil envelops the magnet (increasing magnetic field inside the coil), the current generated in the coil flows the other way.

We can summarize Faraday's Law by saying that a changing magnetic field creates electricity in the surrounding coil. We are careful in the statement of Faraday's Law below to say "produces an electric field" because if there is no coil around, there is no electricity. There is the potential to produce electricity, a potential to push charges. A "ghost battery" is in the surrounding space, the electric field. Put that wire in place, and the "ghost battery" works on the charges in the wire to get them moving. You then get electricity.

Faraday's Law:

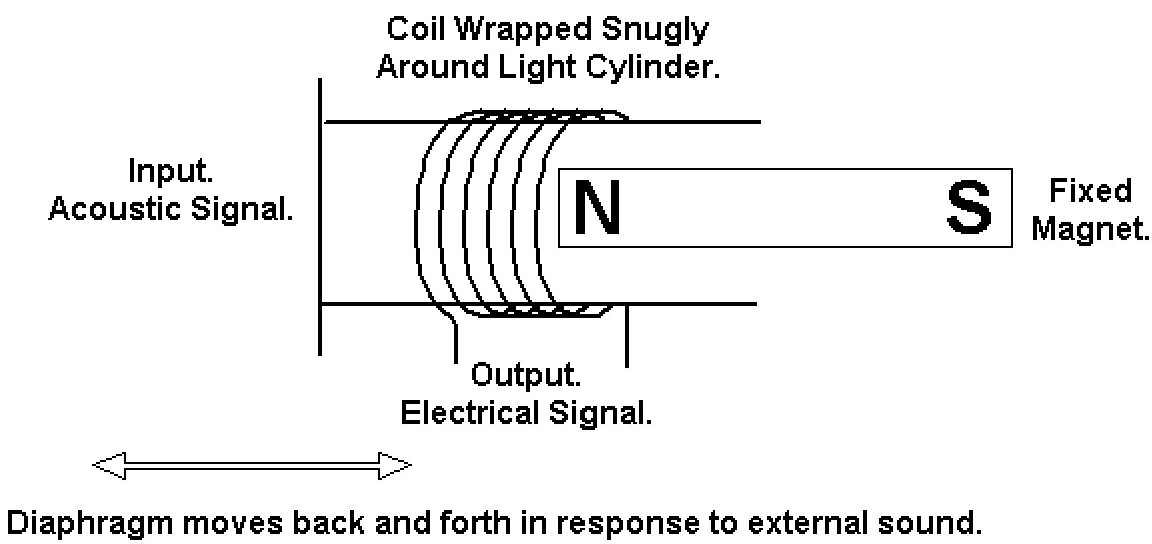
A changing magnetic field produces an electric field.

Faraday's Law got you to school today if you came in a car, truck, bus, cab, or motorcycle. Magnetic fields are created in a coil by your 12-volt car battery. Then, the magnetic field is rapidly turned on and off by a switch. A second coil surrounding the first coil "experiences" the sudden collapsing of the magnetic field. This extremely quick collapse generates electricity at 20,000 volts!

This voltage is distributed to your spark plugs according to the firing order for your car; and the 20,000 volts produce a spark across the small gap of the plug. The spark ignites the gas, which gas expands, pushing a piston that turns the mechanism that will eventually get your wheels turning. Faraday's Law enables you to go from 12 volts to the 20,000 volts.

Fig. K-11 below gives the schematic for the dynamic microphone. It is essentially the speaker diagram interpreted backwards. Someone speaks into the diaphragm. The diaphragm vibrates in step with the sound waves from the voice. The moving coil around the magnet produces a corresponding electrical output signal in the coil wires.

Fig. K-11. Microphone Schematic.



We leave this chapter by pointing out a further underlying beauty in nature. The stricter statement of Faraday's Law says that a "changing magnetic field produces an electric field." This electric field can give us electricity if a wire coil is brought nearby. Someone observed that Ampère's Law is not really the strict opposite. Ampère's Law states that a "current produces a magnetic field."

The strict opposite to Faraday's Law is that "a changing electric field produces a magnetic field." The physicist Maxwell observed this and combined this additional law with Ampère's Law around 1865. This completed the puzzle. The laws of electromagnetic theory were now complete. And we received a bonus! The fact that changing magnetic fields produce electric fields, which if changing, in turn produce magnetic fields has great consequence. It predicts the existence of electromagnetic waves. These waves include radio waves and visible light.

Table K-2 below summarizes the laws of electricity and magnetism. The four basic laws of electromagnetism are referred to as *Maxwell's Equations*. The Maxwell Equations predict the existence of electromagnetic waves. These are defined by their wavelength or frequency.

We have the familiar wave relation $\lambda f = c$, where c now stands for the incredible velocity of light (300,000 km/s or 186,000 mi/s). The electromagnetic (EM) spectrum of waves is broken into seven regions. From short to long wavelengths, these are gamma rays, x-rays, ultraviolet light, visible light, infrared light, microwaves, and radio waves.

For our study of sound reproduction, we are mostly interested in the radio end of the spectrum. The wavelengths of radio waves range from a few centimeters (microwaves) to hundreds of meters (AM Radio). Table K-3 lists the rich variety of radio waves and their frequencies.

Table K-2. Maxwell's Equations of Electromagnetism.

1. Coulomb's Law	Like Charges Repel, Unlike Charges Attract.
2. (No Name)	There are no Magnetic Monopoles.
3. Ampère's Law Maxwell's Addition	Currents Produce Magnetic Fields. Changing Electric Fields Produce Magnetic Fields.
4. Faraday's Law	Changing Magnetic Fields Produce Electric Fields.

Table K-3. Types of Radio Waves.

EM Radio Wave	Frequency Range
AM Radio	550 - 1600 kHz
CB (Citizen's Band)	26 - 27 MHz
International Short Wave	28 - 53 MHz
VHF Channels 1 - 6	54 - 88 MHz
FM Radio	88 - 108 MHz
VHF Channels 7 - 13	174 - 216 MHz
Ham Operators	216 - 469 MHz
UHF Channels 14 - 88	470 - 890 MHz
RADAR	300 - 30,000 MHz

--- End of Chapter K ---

L. Sound Systems

We address three more sound sources in this section. These are the record player, tape deck, and CD player. They represent three levels of improvement in sound reproduction. Faraday's Law and Ampère's Law will be important in our discussion of records and tape decks. We state applied versions of these laws in Table L-1. The two essential devices are the coil (electricity) and magnet (magnetism). Note again the interplay between electricity and magnetism. These are manifestations of the unified electromagnetic force of nature. Such unification makes possible the invention of diverse sound components.

Table L-1. Applied Versions of Two Basic Laws of Electricity and Magnetism.

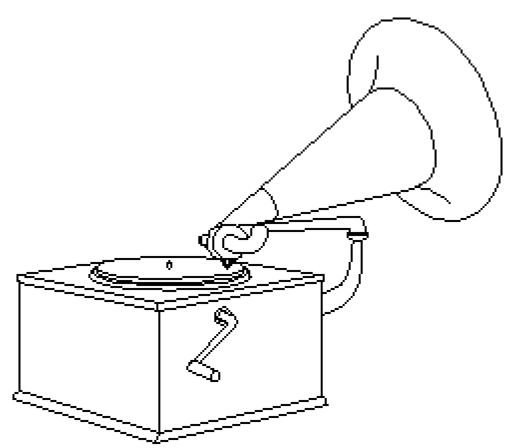
Law	Applied Version of Law
Ampère's Law	A changing electrical signal in a coil surrounding a magnet, causes relative motion between the coil and magnet.
Faraday's Law	Forced relative motion between a coil and inner magnet, generates a changing electrical signal in the coil.

Record Players

The record is our first practical method of storing sound. Sound vibrations are encoded as small hills and valleys on a disk. Thomas Edison invented the first model, a cylinder with grooves, in 1877. He coined the word *phonograph* at that time. However, the disk shape became the standard. The word *gramophone* was introduced in 1887 to distinguish the two. In America, the word phonograph is used to refer to both.

Fig. L-1 illustrates an old phonograph that is purely mechanical. You turn the crank and a needle vibrates to the bumps on the disk. The vibrations are amplified by the horn.

Fig. L-1. Old Mechanical Phonograph.



Early records had vertical hills and valleys. Later, one finds lateral cuts. We will illustrate the vertical cut and then move on to the two-channel stereophonic records.

The early records sounded poorly according to later standards because the hills and valleys in the grooves were cut using actual sound pressure itself. Starting around 1925, the cutting was done electrically. Cutting and playing back a record electrically brought about a significant increase in performance.

Engineers used the same electro-magnetic techniques we encountered with speakers (Ampère's Law) and microphones (Faraday's Law). Playing a record is the reverse of cutting a record. Fig. L-2 below illustrates the vertical or *hill-and-dale* cut. Here we keep the coil fixed and let a tiny magnet move. Remember that when there is a force between two bodies and one body is held fixed, the other moves. That's why we used the words "relative motion" for the description of the coil-magnet movements in our restated laws in Table L-1.

Fig. L-2a. Cutting a Monophonic Record (Ampère's Law).

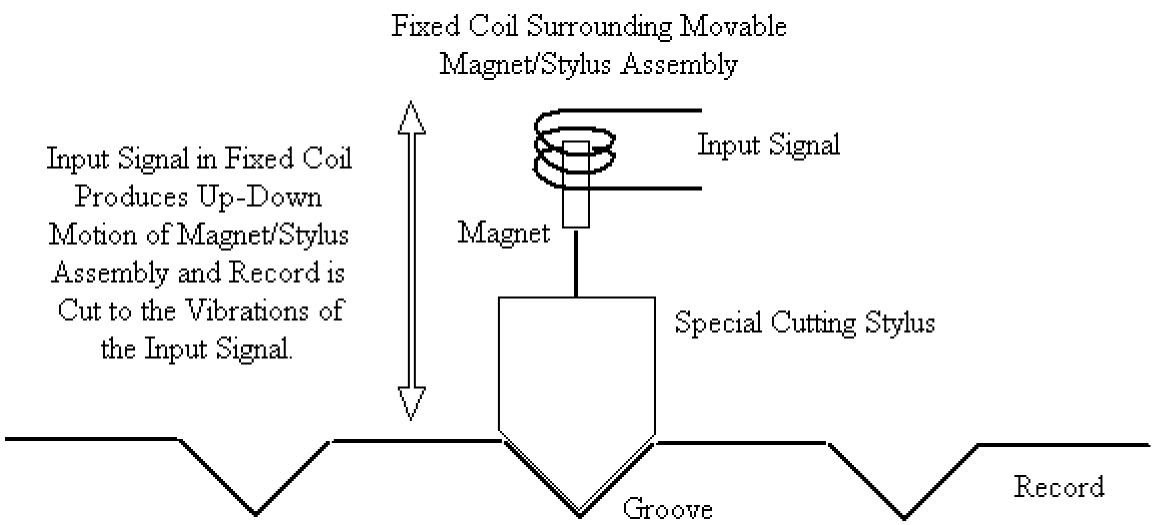
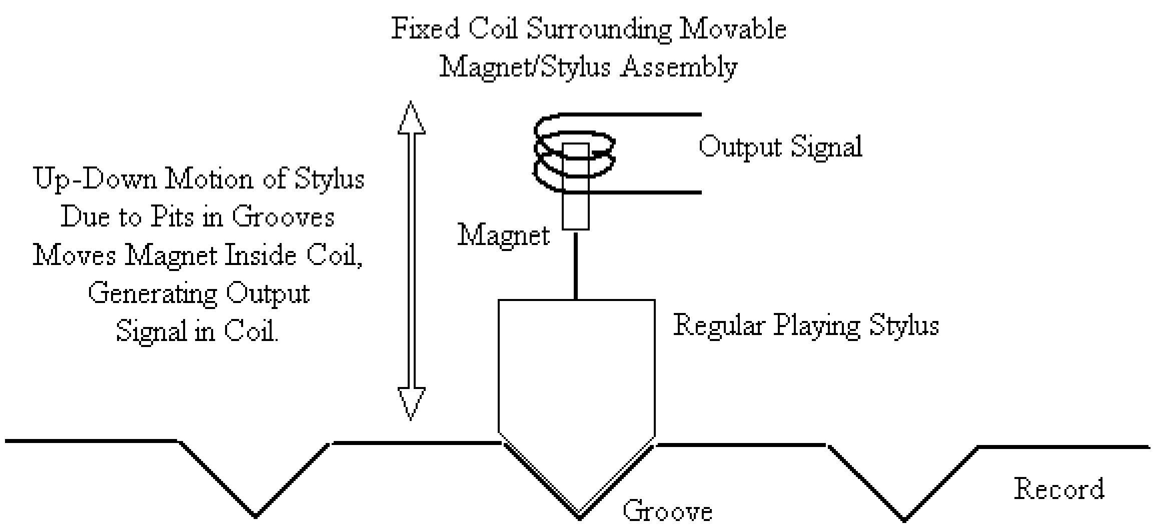


Fig. L-2b. Playing a Monophonic Record (Faraday's Law).



Monophonic or monaural records turned at various rates. The first popular standard records, made with a shellac compound, had a spin-rate of 78 rpm (rotations per minute). The later standards were the smaller 7"-diameter "45 (rpm)" with a large center hole, and the 12"-diameter long-playing (LP) records (33 1/3 rpm). Table L-2 lists data for these historical formats. The 45 became widespread for popular singles in the 1950s, while the LP was the standard for classical music.

Table. L-2. The Common Historical Record Formats.

Date	rpm	Type	Diameter	One Side	Material
< 1925	78	Mono	10"	5 min	Shellac
1948	33 1/3	Mono	12"	25 min	Vinyl
1949	45	Mono	7"	5 min	Vinyl
1958	33 1/3	Stereo	12"	25 min	Vinyl

The more expensive record players that appeared in the 1950s were able to run at three speeds: 78, 45, and 33 1/3. However, the 78s were phased out by this time. Then mono systems were outdated when stereo arrived in the late 1950s. The mono stylus assembly (cartridge) can only pick up one channel of information. Fig. L-3 illustrates the playback design of the stereo record. A stereo cartridge is needed to pick up two channels separately, the two series of undulations at 45° (see Fig. L-3). Stereo cartridges however can play a mono record. Why? The best stylus is a diamond needle, oval at the tip.

Fig. L-3. Playing a Stereophonic Record (Faraday's Law).

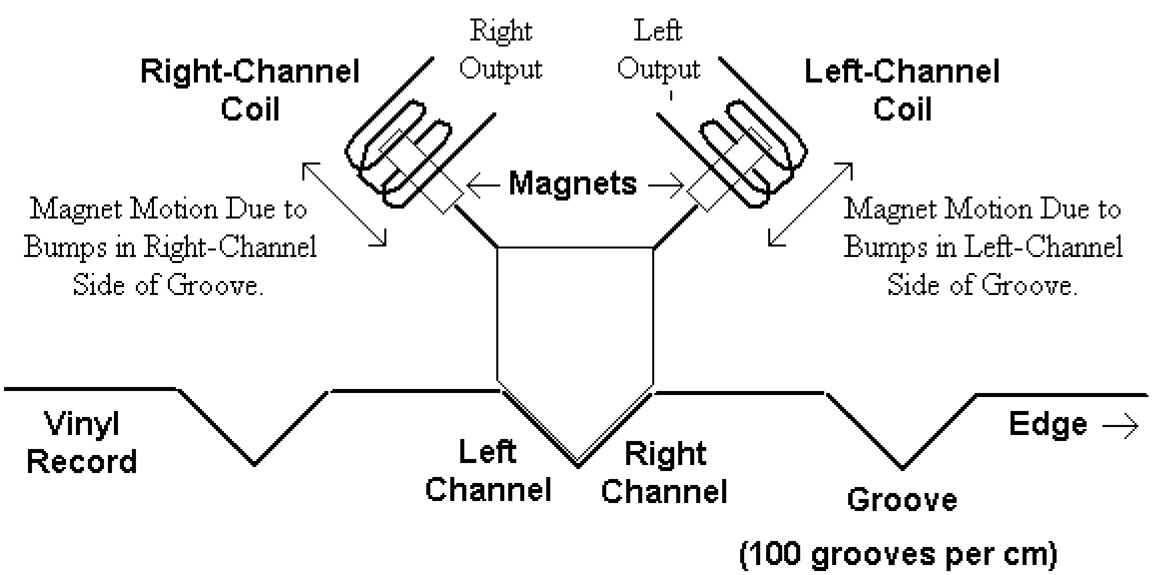


Fig. L-4 shows a side view of a turntable. It is important for the stylus not to press down on the record any more than is necessary. A fulcrum is used with a counterweight so that the heavy arm will not lean down very much on the record surface. The force of the stylus on the grooves is reduced to about 1 gram in this way. Physicists do not like to use the term "gram" for force or weight. Strictly speaking, gram is a unit of mass. Mass and weight are not the same. For example, your mass is constant, but your weight on the moon is 1/6 of your weight on Earth. However, by saying gram-weight, we are okay. One gram-weight is the weight of one gram of mass on Earth. At what gram-weight does the record player in Fig. L-4 track on the moon?

Fig. L-4. Use of Fulcrum and Counterweight to Reduce Tracking Force.

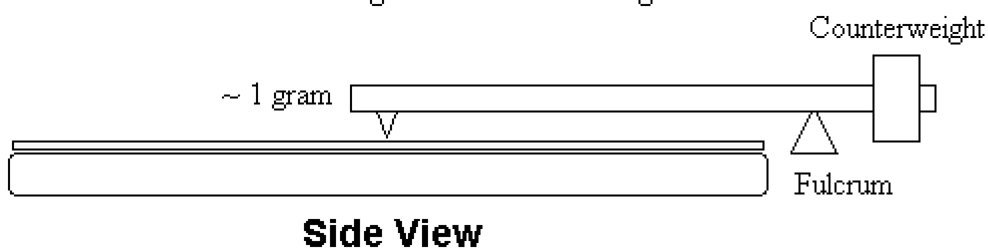
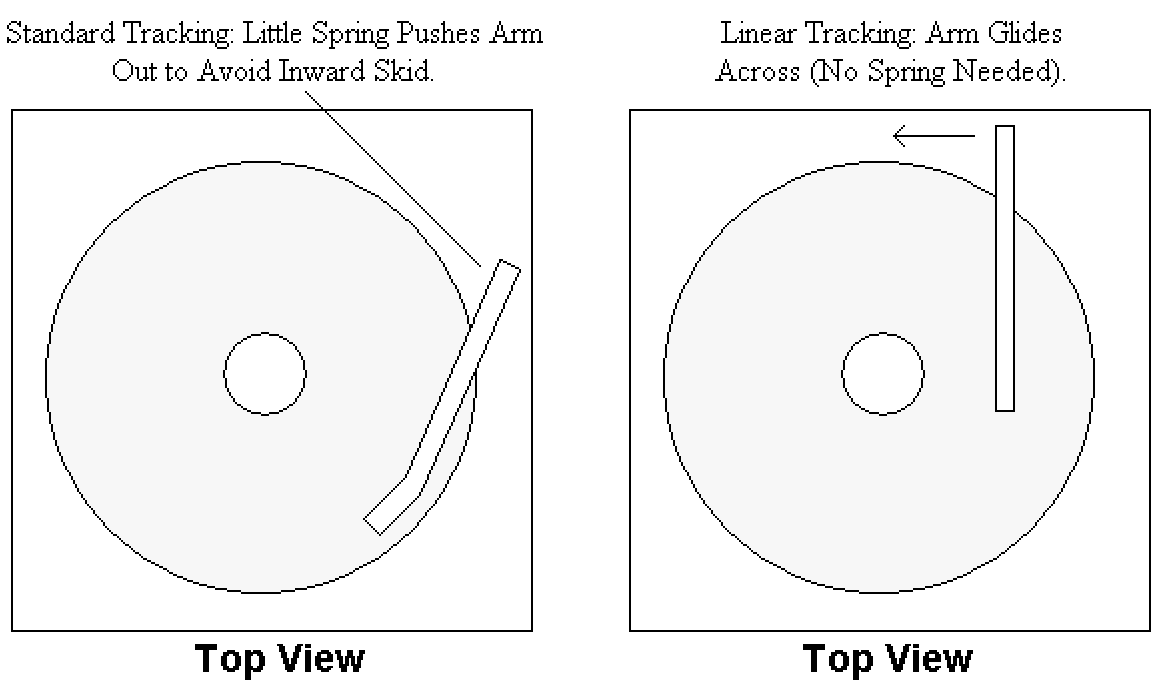


Fig. L-5 gives us a top view, illustrating two tracking methods. The record pulls the stylus end straight out in each case. This is fine for linear tracking, but causes the arm to "skate" inward toward the center in the usual arrangement. This "skating force" is balanced by a spring that supplies an outward force on the arm to compensate. This spring force is called the *antiskating force*.

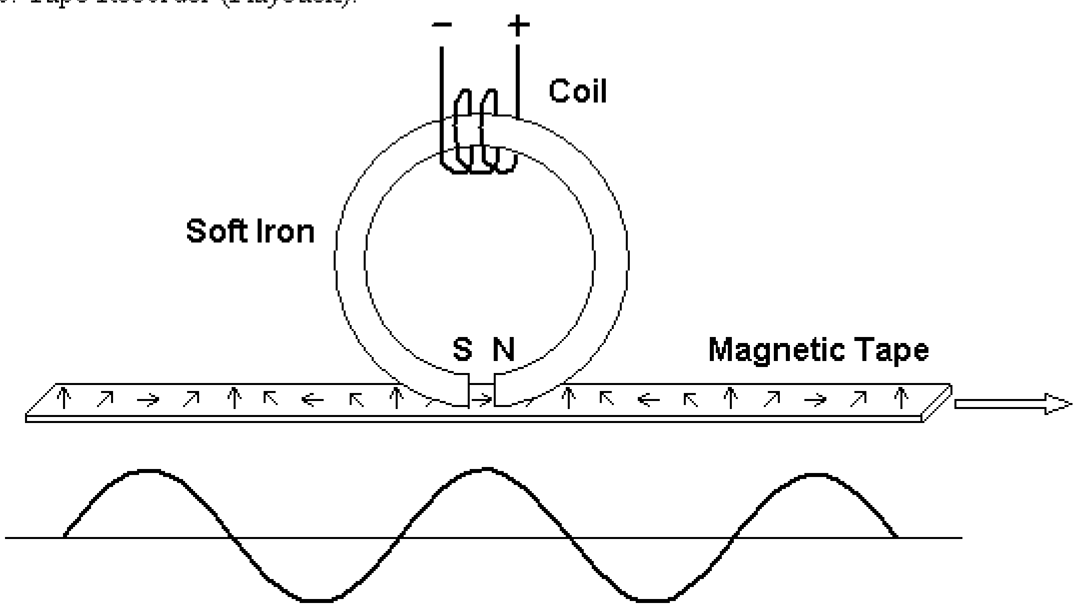
Fig. L-5. Arm Design for Tracking (Antiskating Force Needed in Standard Tracking).



Tape Recorders

Tape recorders became available in the mid 1950s due to the invention of plastic magnetic tape. The tape has tiny magnetic particles. These "baby magnets" are called *magnetic dipoles*. The magnetic coating that provides these are iron oxides or chromium oxides. We represent the magnetic dipoles as small arrows on the tape in Fig. L-6. Each arrow tip is a tiny north pole, while each tail is a south pole. The sizes are highly exaggerated in the figure.

Fig. L-6. Tape Recorder (Playback).



Sine wave encoded on above magnetic tape by orientation of magnetic dipoles (not to scale).

The dipoles in Fig. L-6 change direction in step with the sine wave illustrated. A maximum displacement (top of a crest) is represented by a dipole on the tape pointing to the right. Can you decode the other orientations? As the tape is moved across the playback head, the different magnetic orientations are picked up by the iron head. The iron is soft "magnetically," meaning that permanent magnetism will not occur.

The changing magnetic field is "sensed" by the surrounding coil at the other end. The changing magnetic fields induce electrical currents in the coil (Faraday's Law). The current changes in step with the magnetic field in the tape-head, which changes are in step with the changing orientations on the magnetic tape. To make a recording, the order is reversed. The coil in the record head (a 2nd head) receives electrical signals from a source. These cause magnetic changes in the core (Ampère's Law), which arrange the dipoles on the tape.

An erased tape is made by applying a high-frequency sine wave with the erase head (a 3rd head), so that the dipoles can't respond well. They become randomized. The early common tape recorders had two tape speeds: 3 3/4 inches per second (ips) and 7 1/2 ips (good for music). Cassettes use 1 7/8 ips, a speed unheard of for music in the 1950s. We will see why later.

CD Players

Compact disks (CDs) employ a new encoding technology, called digital technology. So far, we have encountered signals that vary continuously. This type of signal is called *analog*. Think of a meter with a dial, where the pointer can point to any value. On the other hand, *digital* signals are discrete. A piano is a digital system designed for your digits (fingers). You have 88 discrete choices of tones. A guitar or violin is an analog instrument because you can play between the regular tones by holding the string at any arbitrary point. The frets on a guitar assist you in using the guitar as a "digital" instrument.

Digital information consists of a series of numbers. Two digits, 0 and 1, are ideal for electronic processing and computers. Using two digits is called the binary system. Table L-3 gives the conversion from decimal to binary for the numbers from 0 to 15. Think of the odometer in your car. When you get to a mileage like 999, all the 9s turn to 0s the next mile and you get a fourth digit, starting at 1, i.e., 1000. With binary, you run out of digits quickly. After a binary 111, all the 1s turn to 0s and you get 1000. Look at the binary progression in Table L-3 as a changing odometer reading, where only two digits are available, 0s and 1s.

Another way to think of decimal numbers such as 3726 is to look at the position of the digits. The 3726 means 6 units plus 2 tens plus 7 hundreds plus 3 thousands. Reading right to left, you have units or 1s, 10s, 100s, 1000s, etc. You keep multiplying by 10 to get these numbers since there are 10 different symbols being used: 0, 1, 2, 3, 4, 5, 6, 7, 8, and 9. For binary, we multiply by 2 instead since we have only two different symbols at our disposal: 0 and 1. So instead of units, 10s, 100s, 1000s, etc. we have units, 2s, 4s, 8s, etc.

Table L-3. Binary Numbers.

Decimal	Binary
0	0000
1	0001
2	0010
3	0011
4	0100
5	0101
6	0110
7	0111
8	1000
9	1001
10	1010
11	1011
12	1100
13	1101
14	1110
15	1111

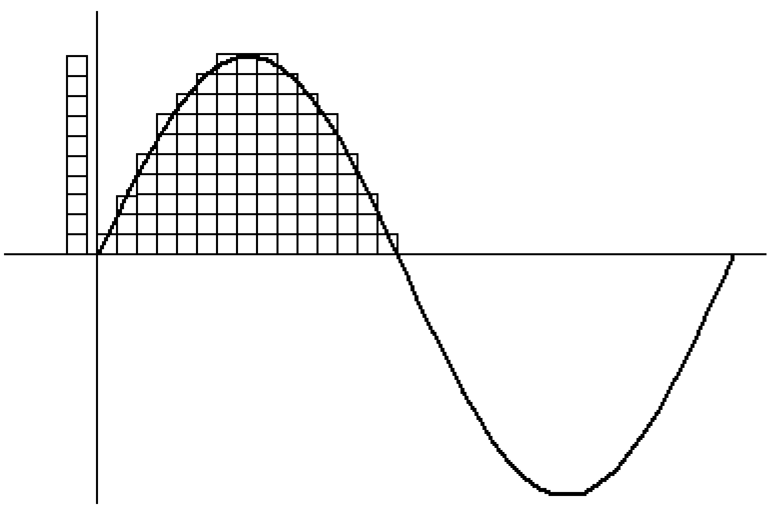
Therefore, the number 10 in binary indicates (reading from right to left) 0 units and 1 two, i.e., 2. Binary 11 indicates 1 unit plus 1 two or 3 in decimal. Similarly, 110 in binary implies 0 units plus 1 two plus 1 four, i.e., decimal 6. To avoid confusion with decimal numbers, a subscript 2 is used to indicate binary (base 2). We can write binary 110 as 110_2. As one last example, note that 1111_2 is equal to 1 unit + 1 two + 1 four + 1 eight, i.e., 15. Go through all the binary numbers in Table L-3, carefully checking each one in this way.

Now we are ready to digitize a wave. Fig. L-7 illustrates a sine wave. The sine wave varies smoothly. It is an analog signal. To digitize it we stack small boxes to approximate it. We count how many boxes we need for each small section of the wave. We list these results in the table under height. Then we convert to binary with the assistance of Table L-3.

The digitizing in Fig. L-7 is very crude. A real digital version would require boxes too small to count. The 0s and 1s for the coded information can be stored on a compact disk using a series of pits for the 1s. No pit can represent 0. These can be read by a laser. Compare a needle scraping in a groove (record) to a beam of light reflecting from pits in a groove of a compact disc (CD). The record wears out in a few years, but the CD is unaffected. The CD can last centuries!

Also, digital format can be read by computers. Computers can store music in digital form. They can also process the stored information, enhancing and modifying it. Records and tapes have limits to the range of loudness you can store. Hills and valleys can only get so high. Magnetic tape has saturation limits on how many dipoles can align together in a small region of tape. Digital methods are much superior.

Fig. L-7. Digitizing an Analog Signal.



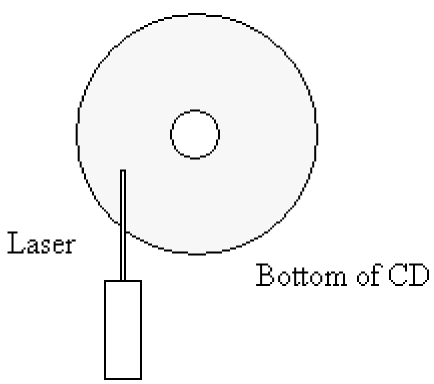
The actual boxes used are much smaller. The "horizontal box length" is taken to be about half the wavelegnth of a 20,000-Hz tone. The digital sampling rate is therefore about 40 kHz (actually 44.1 kHz).

Height	Binary
1	0001
3	0011
5	0101
7	0111
8	1000
9	1001
10	1010
10	1010
10	1010
9	1001
8	1000
7	0111
5	0101
3	0011
1	0001

Fig. L-8 depicts a compact disc. The laser shines light on the disc from below. The laser employs infrared light (IR). Infrared lies beyond the red end of the spectrum and is invisible. The IR light reflects from the pits on the CD. The reflected light is detected and analyzed. Data is read from the inside out as the CD varies its turning speed from 500 rpm to 200 rpm. The digital information is converted to an analog output signal which goes to the amplifier.

The first CDs became available commercially in 1983. The LP record, an important medium for stereo sound for 20 years (1960 to 1980), was in trouble. The phase-out period was underway.

Fig. L-8. Compact Disc.



The CD is about 5 inches in diameter. It can store 74 minutes of music. Many old recordings of famous musicians have been transferred to CDs. However, these were recorded with analog technology. These original works are often processed and remastered digitally. The original recording, intermediate processing and remastering stage, and the final format are summarized by three letters, each being either A (analog) or D (digital). A CD with the designation AAD means originally recorded with analog methods, processed and mastered as analog, then digitized for CD. The ADD indicates that the original analog recording has been digitally remastered before encoding on the CD. The combination DDD indicates a digital recording, digital processing, and final digital format. Of course, with a CD, the last letter is always D.

Let's return to the digitization of a sine wave. Refer again to Fig. L-7. The length of the rectangle is chosen so that two rectangles, side by side, extend to the wavelength of a 20,000-Hz tone. We say that we sample twice in the time-frame (period) of a 20,000-Hz tone. The s*ampling rate* is 40 kHz (actually 44.1 kHz). The height of the rectangle is chosen so that the distance from the trough to the crest of the "biggest wave" is given by the largest 16-digit binary number. This choice is called 16-bit sound. The largest 16-bit number is 1111 1111 1111 1111 which is 65,535. Since 0 is the first possible value, there are 65,536 numbers possible.

The amplitude is measured from equilibrium to crest, so the largest amplitude is one half this value, or 32,768. The loudness of a sound is given by the energy of the wave, which is related to the amplitude. The energy is essentially equal to the square of the amplitude. We state this without proof. It is saying that a water wave twice as tall has four times as much energy to hit you with. One 3 times as tall is 9 times more energetic. Therefore, the loudness range goes from 0 to 32,768 x 32,768. This is about 30,000 times 30,000 or 900,000,000, which rounds off to 1,000,000,000 (one billion). This is incredible dynamic range. Later we will learn that this corresponds to 90 decibels (90 dB).

--- End of Chapter L ---

M. Analog Electronics I

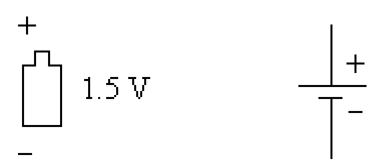
M-1. Flashlight

Our first circuit application is the flashlight. It is a simple electrical circuit with a battery, bulb, and wire or metal to connect the two. These components are discussed below. The first component we consider is the *battery*, the device that produces moving charge, i.e., electricity.

A battery gets charges moving for us. The battery separates charges chemically. Then the electric force serves as the force that makes the electrons move toward the positive region. A battery is sketched below in Fig. M-1 along with the symbol used by engineers for one battery cell. The minus is at the bottom, while the absence of electrons causes a plus region at the other terminal. If a wire were to connect both terminals (dangerous), electron flow through the wire to the plus terminal would dangerously exceed normal levels. This would be very bad for the battery for the flow would be too intense. The electricity must do some work, e.g., heat a bulb filament, so that the flow be within operating limits of the battery. Such a lower flow rate gives the battery time to keep up with the flow by constantly bringing the electrons internally back to the minus terminal. However, after the lifetime of the battery, the battery can no longer separate the electrons. We then say that the battery is dead.

A measure of the strength of the battery is given in *volts*. A typical flashlight battery has 1.5 V (1.5 volts); transistor radio batteries are 9-V batteries. If we stack batteries as in a flashlight cylinder, the voltages add. Two 1.5-V batteries in a flashlight result in a voltage of 3.0 V.

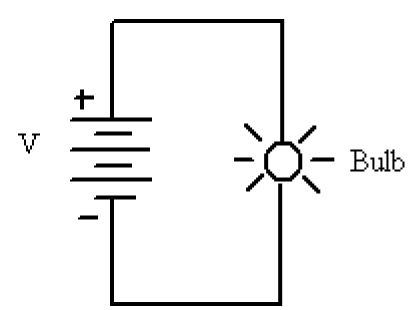
Fig. M-1. Battery and Symbol.



The simple circuit for the flashlight is given in Circuit 1. Suppose we have a bulb that will light up when 4.5 V is applied to it. We then stack three 1.5-V batteries in order to obtain 4.5 V. See the flashlight circuit for this bulb in Circuit 1 at the right.

The bulb consists of a fine tungsten filament surrounded by some inert gas (non-reactive). The filament heats up due to the current going through it. The lack of oxygen in the bulb prevents the bulb from burning up. The filament glows, giving off visible light and infrared light (heat).

Circuit 1. Flashlight.



Current is a measure of how many electrons pass a given point in one second. We need not concern ourselves with how many electrons move by per second. We simply use the convenient unit *ampère* or *amp*, abbreviated A. The letter I is used to designate current, e.g., I = 5 A (5 amps) .

1. *Direct Current* (DC). The current flows in one direction. The battery used in Circuit 1 supplies direct current in the circuit. A typical bulb in Circuit 1 may draw a current of 0.1 A (one tenth of an amp). It is convenient to use a smaller unit of current, thousandths of an amp or milliamps. The abbreviation for milliamp is mA. Since 0.1 A can also be written as 0.100 A, we readily see that we have 100 thousandths of an amp, i.e., 100 mA.

2. *Alternating Current* (AC). The current flows back and forth. This type of current is easy to generate by power companies. Our light bulb doesn't care which way the current flows. It would still light up if we used alternating current. Think of alternating current like a sine wave describing electrons pumping back and forth in a wire.

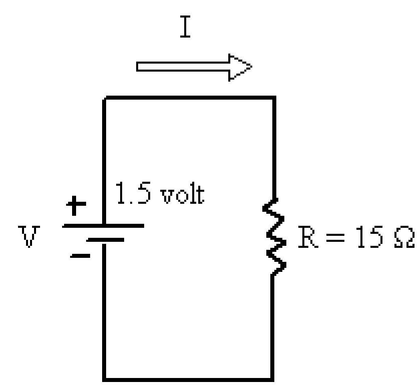
Resistance limits the flow of current in a circuit. Resistance is measured in *ohms*. The symbol for ohms is Ω. A *resistor* is a circuit element whose purpose is to provide resistance. Think of a resistor as a "clogged pipe with rags," where the "current of water" has trouble getting through. Before we see a dimmer circuit that will illustrate the importance of resistors, let's consider a simple circuit of a battery and resistor. This circuit is not useful for anything other than illustrating a basic principle of electronics: *Ohm's Law*.

Note that in Fig. M-2 below the current direction is pointed away from the plus side of the battery. This is a convention. The electrons actually move the other way. But since electrons are negative, we point the arrow in the opposite direction. This helps make the math simpler for engineers.

Ohm's Law relates the three quantities V, I, and R. The law states that V = IR. Think about this. You would expect it. If you keep the battery strength V constant, then less resistance (R) means more current (I) and vice versa. Recall that this logic is the same we used to discuss $v = \lambda f$, where the wave speed is constant. If you decrease λ, you increase f and vice versa.

Here is an example. For one cell, V = 1.5 volts. Consider a resistance of 15 Ω. Then, V = IR becomes 1.5 volts = I x (15 Ω). The current "I" must be one tenth, i.e., 0.1, A since one tenth of 15 is 1.5. We can also rearrange V = IR as follows: I = V/R = 1.5/15 = 0.1 A = 100 mA.

Fig. M-2. Current.



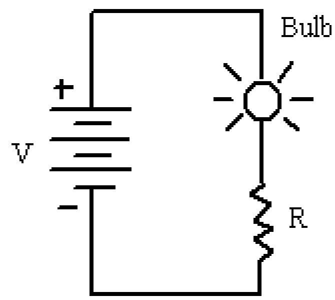
V = IR

I = V/R = 1.5/15 = 0.1 A

M-2. Dimmer

The circuit at the right (Circuit 2) makes the bulb glow dimmer due to the resistor R. Can you design a three-way dimmer circuit so that the light bulb has three settings? HINT: Modify the circuit so the pathway at the bottom has three possible upward turns, one with resistor R_1 (this can be the resistor with the large resistance that will cause the bulb to glow very dim), a second with resistor R_2 (this can be a resistor with less resistance), and a third upward path that is a "pure" wire (no resistance). Then use the switch symbol seen in C3. Have your switch come down from the bottom of the bulb, allowing it to choose one of the three pathways to complete the circuit.

Circuit 2. Dimmer.



M-3. Flash

Circuit 3 (C3) introduces a new circuit element called a *capacitor* (C). The capacitor can be thought of as consisting of two metal plates. The battery symbol shows two cells. This is the standard number of cells to show (in a circuit diagram) with any written value by the side of the symbol.

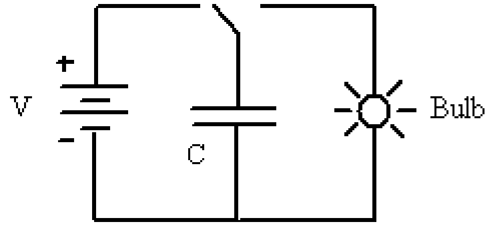
Note the switch symbol at the top. The switch can be thrown to the left or right. When the switch is thrown to the left, the capacitor is connected to the battery. Minus charges gather temporarily (–) on the bottom plate and the absence of electrons on the top plate makes the top plate positive (+).

The switch is then thrown to the right. Negative charges rush through the bulb to get to the other side (plus side) of the capacitor. Current flows through the bulb. However, since the capacitor loses its charge as electrons return to the top plate, the bulb is on for only an instant. The unit for capacitance is the *farad* (F), named after Faraday.

The capacitor is not a battery; it cannot remove electrons from the top plate and bring them back down to the bottom plate internally as a battery does. The capacitor only stores electricity and when this electricity is used up, the capacitor is neutral again (no net charge on either plate).

The rush of current is sudden and dies down quickly. If a resistor were in the charging left loop between the battery and the capacitor, the charging of the capacitor would take time since current flow would then be reduced to more of a trickle of flow due to the resistor.

C3. Flash.

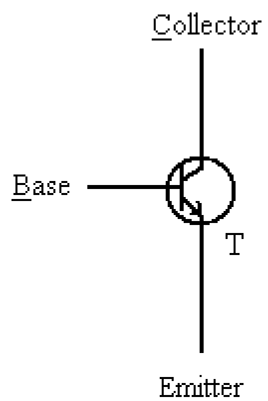


The circuit diagram depicts the standard use of two voltage cells to represent any voltage.

M-4. Delayed Light

A *transistor* is a circuit element with three wires protruding from a small container. The schematic of a transistor is seen in Fig. M-3. The three wires of the transistor are called the *base* (B), *collector* (C), and *emitter* (E). When a small amount of current enters the base (B) of the transistor, lots of current can then flow from the collector (C) to the emitter (E), provided that a battery is around nearby to supply the current. The small amount of current from B and the great amount of from C both leave the emitter E. We will not delve into the inner structure of a transistor.

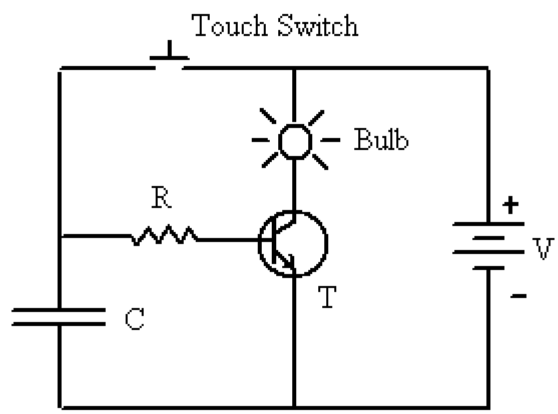
Fig. M-3. Transistor.



By carefully controlling the small base current, we can control the large currents capable of flowing from the collector to the emitter. Circuit 4 is a circuit that gives off light when the touch switch is pressed. When the touch switch springs back after release, the light stays on for awhile. After a certain time, the light then goes out automatically. Such a circuit is convenient for lighting a garage until you can drive the car out. After you're a block away or so, the garage light turns off by itself. However, Circuit 4 is a "doll-house" version of the real circuit.

When the switch is pressed, current rushes to the capacitor (C). The capacitor charges immediately since the path to it is clear. After the capacitor is charged, current trickles through the resistor activating the transistor. Current can now flow through the transistor, and the bulb glows. When the switch is released, it takes awhile for the charge on the capacitor to empty through the resistor. So the bulb stays on until the capacitor gives up its charge (discharges). After this, the capacitor is neutral, there is no current, and the bulb is off.

Circuit 4. Delayed Light.



We will see later how a transistor can be used as an amplifier in a sound system. Can you modify the above circuit so that one needs to hold the switch down for awhile in order for the capacitor to charge fully? HINT: Insert a resistor in the vertical section of the upper left loop. Now it takes time to charge the capacitor. The light does not go on immediately. However, when the switch is released, the light will stay on for a while as before.

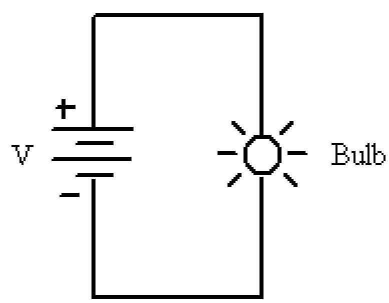
Let's review the four basic circuits we have encountered. These are reproduced in Fig. M-4 below. The simplest possible circuit is the flashlight. It consists of the minimum number of circuit elements in order to make a single circuit loop. You need a battery and one other element, in this case the bulb. There are only a few basic circuit elements engineers use as building blocks for circuit design, just as there are few notes in a musical scale.

Notes serve as building blocks for musical lines. For example, the common jazz blues-scale has only 6 notes, 3 of which lie outside the major scale we encountered earlier. The jazz improviser uses these notes in a traditional blues, perhaps only occasionally using other notes. Similarly, electrical engineers rely mostly on about 6 circuit elements: the battery (or power source), bulb (or some visual-display element), resistor, capacitor, transistor, and diode (which we haven't looked at yet). Observe how carefully each circuit below adds one more of the basic circuit elements.

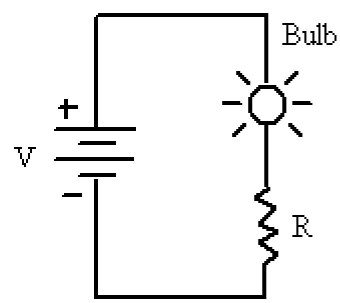
The dimmer should have a variable resistor to vary the light intensity. Sketch an upward slanted (to the right) arrow across the resistor in the second circuit below. This indicates a variable resistor. Variable resistors have a turning knob to vary resistance.

Fig. M-4. Review of Four Basic Circuits.

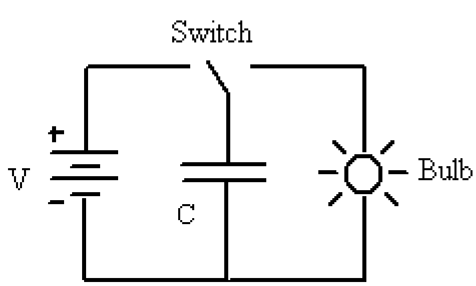
Circuit 1. Flashlight.



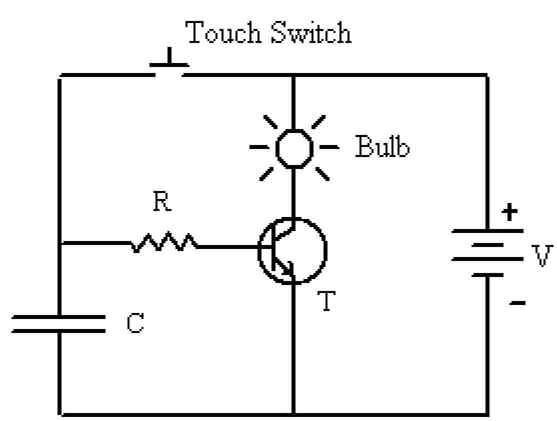
Circuit 2. Dimmer.



Circuit 3. Flash.



Circuit 4. Delayed Light.



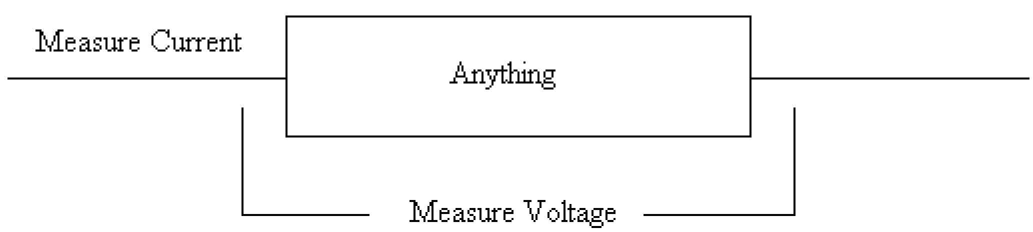
The source of energy for circuits is the battery or the electrical outlet on your wall. Batteries supply direct current (DC) while your wall outlet provides for alternating current (AC). The voltage of typical batteries is 1.5 V for the cylindrical variety and 9 V for the transistor batteries. The voltage from your wall outlet is about 120 V (110 or 115 V), alternating as a sine wave. Voltage for alternating current is defined as 70% of the amplitude.

Two aspects of electrical usage are important for cost analysis. These are the voltage you need and how much current you want to draw. If you need 3 V instead of 1.5 V, you need to buy 2 batteries and place one on top of the other. More voltage means more money. Also, if you tax the batteries, demanding lots of current, the batteries do not last as long. It costs to replace them. Engineers design bigger batteries with the same voltage so that more current can be supplied, but these bigger batteries are more expensive. Battery sizes for 1.5 volts include the common sizes AA, C, and D. You might need 8 size C batteries to get 12 volts (8 times 1.5 V) for a boom box.

To assess the cost, we need to know the voltage and the current. A simple index is obtained by multiplying these together. The result is called the *power* (see Fig. M-5 below). The unit for power is the watt, named after the famous Scottish inventor James Watt, who improved the steam engine in 1769. Watt's steam engine gave us the new form of power that led to the industrial revolution. However, the study of electricity was in its infancy then. We had to wait over another century for common uses of electrical energy in factories and homes. The light bulb was not invented until 1880 by Thomas Edison. It is appropriate to define the unit for power as the watt.

The power equals the voltage times the current ($P = VI$). This applies to both DC and AC circuits. We will give two examples, one for each of these types of circuits.

Fig. M-5. Calculation of Power Consumption.



Power = Voltage x Current

$$P = VI$$

Fig. M-6 illustrates a toy flashlight circuit with one cell. The current is measured to be 0.1 A. To measure current you simply break the wire and stick a meter in the path. The current then flows through the meter. You can measure voltage by changing the setting on the meter (multimeter) and touching each end of the battery (black probe to negative side, red to positive). The power is then given by the product. The answer is 0.15 watts, or 0.15 W. What is this power in milliwatts (mW)?

Fig. M-6. Power Example (DC).

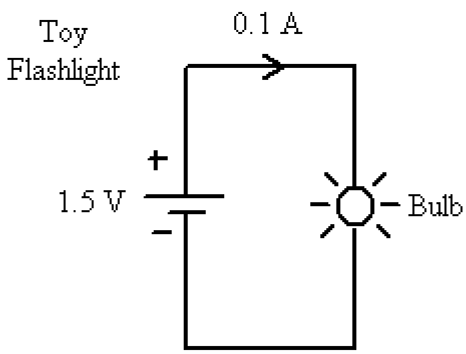
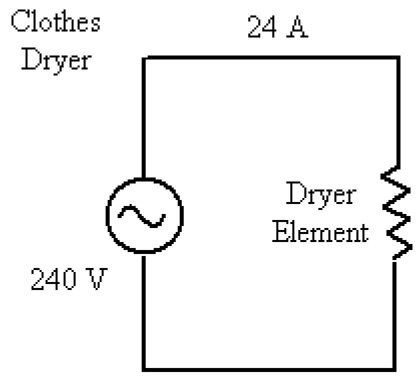


Fig. M-7 illustrates an alternating-circuit example. It is a clothes dryer. Note the circle with the wavy line inside. We use this symbol for an alternating voltage source. The wavy line is a sine wave. The voltage (about 120 volts) from your home outlet alternates as a sine wave at 60 Hz. Your clothes dryer is special, requiring two 120-volt outlets. The electrician wires up this combination, which takes a heavy-duty plug and wire to handle 240 volts. The dryer element is just a resistor that heats up. Note the large alternating current of 24 amps. The current oscillates at 60 Hz due to the alternating power source.

Fig. M-7. Power Example (AC).



$$P = (240 \text{ volts})(24 \text{ amps}) = 5760 \text{ watts}$$

The power consumption for the clothes dryer is 5760 watts, i.e., about 6000 W. This is a lot of power. The electrical company that supplies electricity to your home is interested in how long you keep this clothes dryer on. For example, suppose you dry your clothes for 2 hours. We say you use 12,000 watt-hours of electricity. We have multiplied again! The amount 12,000 watt-hours is more conveniently expressed as 12 kilowatt-hours (12 kWh).

Your electric company charges by the kilowatt-hour (kWh). Let's assume you are charged $0.10 for one kilowatt hour. Then, leaving your dryer on for 2 hours costs $1.20. This amount of money is what you pay for the energy used. The power gives you the level at which you use energy. It is a rate, analogous to speed. When you multiply speed by time, you get distance. Here, we multiply power by time to get the actual energy used.

Table M-1 below gives a list of common household electrical appliances and typical power consumptions. Usually there is a sticker on the device that lists the power. Sometimes voltage is listed (115 or 120) and the current instead of the wattage. You just multiply the given voltage value with the value for current in amps to get the power (wattage). The manuals for stereo and TV equipment also indicate the power consumption under the specifications section. Stereo "buffs" like specs. Unfortunately, due to the low level of science literacy in our culture, such information is often omitted in manuals for refrigerators, dishwashers, and other items. You have to look for the sticker on the appliance.

Due to those big values at the end of Table M-1 below for such household appliances as the electric hot water heater, the electric clothes dryer, and electric range/stove, the author switched to natural gas for his new home. Gas is available in the neighborhood. The electrical power for the gas water heater is 0 W. It is not plugged in. The same is almost true for the gas stove, but you need the lights, fan, and clock. During the 4-day blizzard/power outage in the early 1990s, many households were without heat, hot water, and a functional range/stove. With gas, the author enjoyed hot showers, hot spaghetti dinners, and heat - as the two gas fire-places cranked up living-room temperatures to 80°F!

Table M-1. Power Consumption for Appliances from the Author's Previous and Current Homes.

Electrical Appliance	Wattage	Electrical Appliance	Wattage
Night Light	4	Laser Printer	870
Record Turntable	5	Vacuum Cleaner	940
Yamaha Synthesizer	7	Coffee Maker	975
Tape Deck (no amp)	11	Dishwasher	1030
CD Player	14	Toaster	1050
VCR	27	Refrigerator	1150
Boom Box	38	Washing Machine	1180
Electric Typewriter	40	Oil Burner	1380
Notebook Computer	40	Hair Dryer	1400
TV (19" diagonal)	73	Popcorn Maker	1400
Stereo Receiver	130	Gas Furnace	< 1440
Desktop Computer	250	Hot Water Heater	< 4500
Blender	360	Clothes Dryer	< 6000
Garage-Door Opener	540	Central Air Cond.	< 6000
Microwave Oven	700	Range/Stove	< 12,000

Note the rather high power consumptions for devices that produce heat such as the coffee maker, toaster, popcorn maker, and portable hair dryer. The toaster is a simple circuit, essentially a power source connected to a resistor. The resistor gets warm as the current flows back and forth. The warmth heats and toasts the bread. The clothes dryer is basically the same idea, but doubling up on the voltage and using a motor to turn the drum that holds the clothes.

Central air conditioning is also high. Similar to the clothes dryer, it uses the doubled-voltage value of about 240 volts and draws much current. In comparison, electronic equipment such as tape decks, CD players, and VCRs have little power consumption. The TV or computer monitor demands more power, around 75 W, the same as a 75-watt light bulb. The power supply in a computer is typically 150 or 200 W to run the computer and peripherals. The power needed for the computer and monitor is around 250 W.

The author's portable computer (notebook) requires less, around 40 W. The night light uses the least power in the author's home. Calculate the monthly expense to keep the night light on, 10 hours per night for 30 days, if the cost per kilowatt-hour is a dime.

Electrical current is dangerous to the body. One must be very careful working with electrical appliances. Keep your hands dry. Water with its impurities conducts electricity very well. It acts like a wire. Never touch electrical components in circuits. Even if the circuit is off, capacitors can store much charge. It can still be dangerous. The third prong in an electrical wire connects the frame of the appliance to keep the outer surface neutral (at ground). If a wire or component becomes loose and touches the frame, current flows through the third prong and this should throw the circuit breaker, breaking the circuit at the source.

Once the author stuck his fingers into a wall outlet when he was 4 years old. He got shocked and cried for his dad. The wall outlet voltage is approximately 100 volts. The resistance for dry skin is in the range of 100,000 to 1,000,000 ohms (Ω). We will use 100,000 Ω. The current can be found from Ohm's Law: $V = IR$. We have $100 = I\ (100{,}000)$. The current $I = 100/100{,}000$. We cancel zeros to get $I = 1/1000$ A = 1 mA.

Now we check the health data to see if 1 mA is dangerous. A current of 1 mA is felt as a tingling sensation. However, if your skin is moist, the resistance is lower. For a resistance dropping by a factor of ten, the current goes up by the same factor. We then have 10 mA, beyond the 5-mA safe limit. You feel some shock. If you are 4 years old, this is quite an experience. Now if a baby sticks a finger in the mouth first, the resistance of the newly-wet skin goes way down and current soars upward.

At around 10-20 mA, sustained muscular contraction can prevent one from letting go if one grabs on to a live wire. This happened once to an electrician friend of the author. The electrician grabbed a live source with his hand. He couldn't let go. Luckily, he was standing on a ladder and jumped off using his legs. He fell, freeing his arm from the electrical source. Standing on a wet basement floor (electrical ground, neutral) and fooling with a power source can allow current to travel through the trunk of the body, through the heart.

Table M-2 below gives the physiological effects of various levels of current (60-Hz alternating like that coming from household outlets in the US) through the trunk of the body. The trunk is chosen so that the current passes through the heart. The heart is governed by natural biological forces electrical in nature. Introducing electrical current from outside can interfere with the normal beating of the heart. The dangerous currents are in the neighborhood of 100 to 300 mA. Note that for very high currents such as 6000 mA (i.e., 6 Amps) the heart freezes, then recovers. Remember though that the current is applied for only 1 second for the data in Table M-2.

The heart muscle gets confused when current is in the 100-300 mA range. It starts to beat irregularly. This effect is called *ventricular fibrillation*. It causes death. One can die within seconds or minutes. If an emergency team is nearby, the heart can be shocked again, this time sending in high current to freeze it. The hope is then to restart it beating normally.

The current path through the body is influenced by which part of the body touches the circuit and which other part of the body part acts as the ground. Current flows from higher voltage to lower voltage. The resistance of your skin/body plays a role. The usual amount of current drawn by the appliance you happen to stick your hand into is not the determinating factor. The important parameters are the voltage you touch and your resistance! You then use V = IR to calculate the current I.

Table M-2. Effects of Electrical Current on the Human Body.

Current (mA)	Effect after 60-Hz Alternating Current Goes Through Trunk of Body for One Second and then Stops.
1	Threshold of perception (tingling sensation).
5	Accepted as maximum harmless current.
10 - 20	"Let-go" current before sustained muscular contraction.
50	Pain. Possible fainting, exhaustion, mechanical injury.
100 - 300	Ventricular fibrillation (causes death).
6000	Heart freezes, then recovers.

Source: P. F. Leonard, "Characteristics of Electrical Hazards," *Anesthesia and Analgesia . . . Current Researches* 51, 797 (1972).

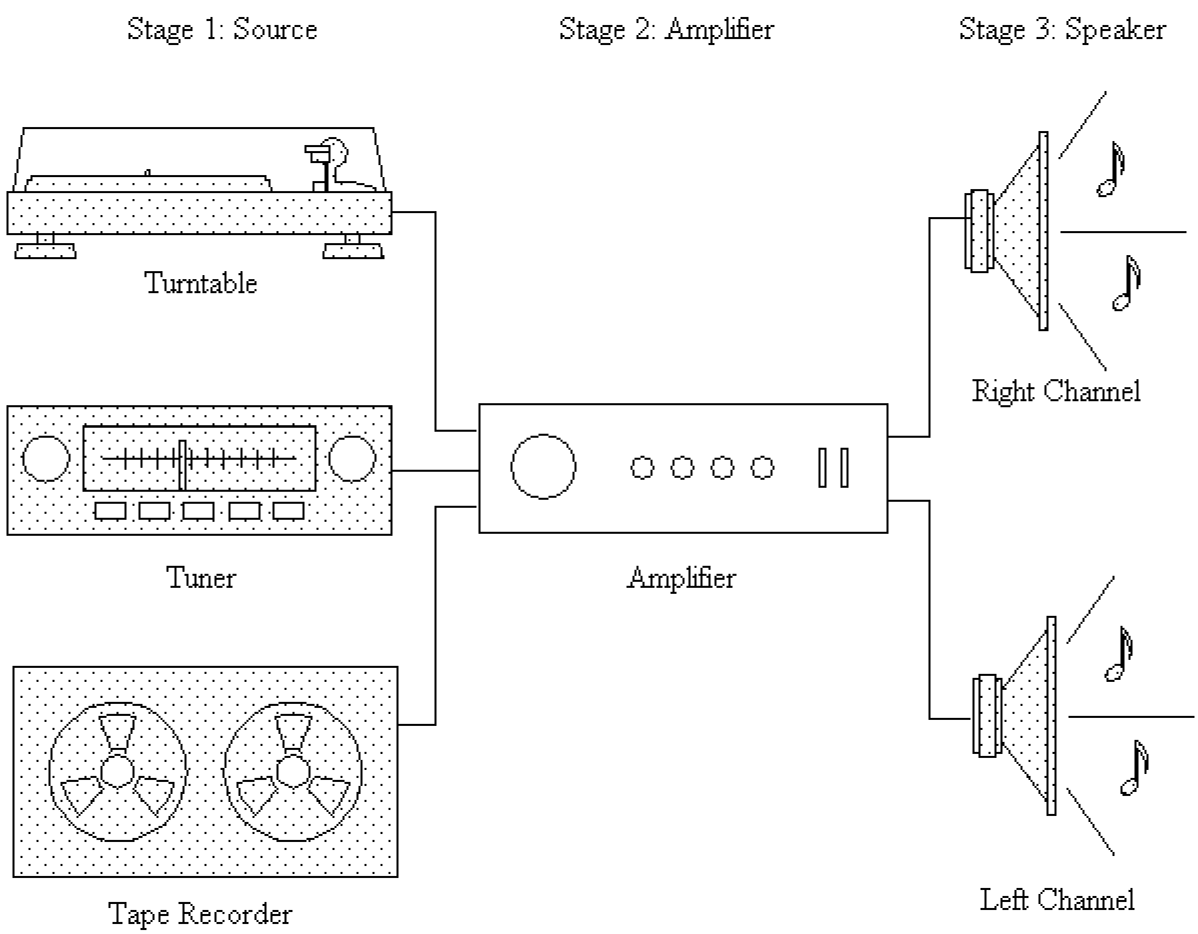
--- End of Chapter M ---

N. Analog Electronics II and Digital Electronics

Fig. N-1 illustrates the three basic stages of sound reproduction as it was realized in the stereo high-fidelity systems of the 1960s. The sources are the turntable (record player), tuner, and tape recorder (reel-to-reel). The amplifier is the common second stage. The third stage is the speaker, one for each channel. The hi-fi buff would have a collection of hundreds of LPs and dozens of tapes. The tapes would be recorded and played at 7.5 ips for high-fidelity. In order to get the *Rachmaninoff Third Piano Concerto* on one side of a tape, you need 40 minutes. A tape speed of 7.5 inches per second calls for a tape length of 7.5 x 60 = 450 inches for one minute. This is 450/12 = 37.5 feet, almost 40 feet for a minute of music. So you need about 1600 ft of tape. No problem. Tapes were sold on 7-inch diameter reels with 1800 feet and other lengths. You could get 2400-ft and even 3600-ft reels. How many minutes of music can you tape on one side of a 3600-ft reel at 7.5 ips?

We have already studied record players, tape decks, and speakers. In this chapter we investigate the amplifier and tuner. By the 1970s it was standard practice to purchase the tuner with the amplifier combined in one unit. Such a unit is called a *receiver*.

Fig. N-1. Hi-Fi System, 1960s.

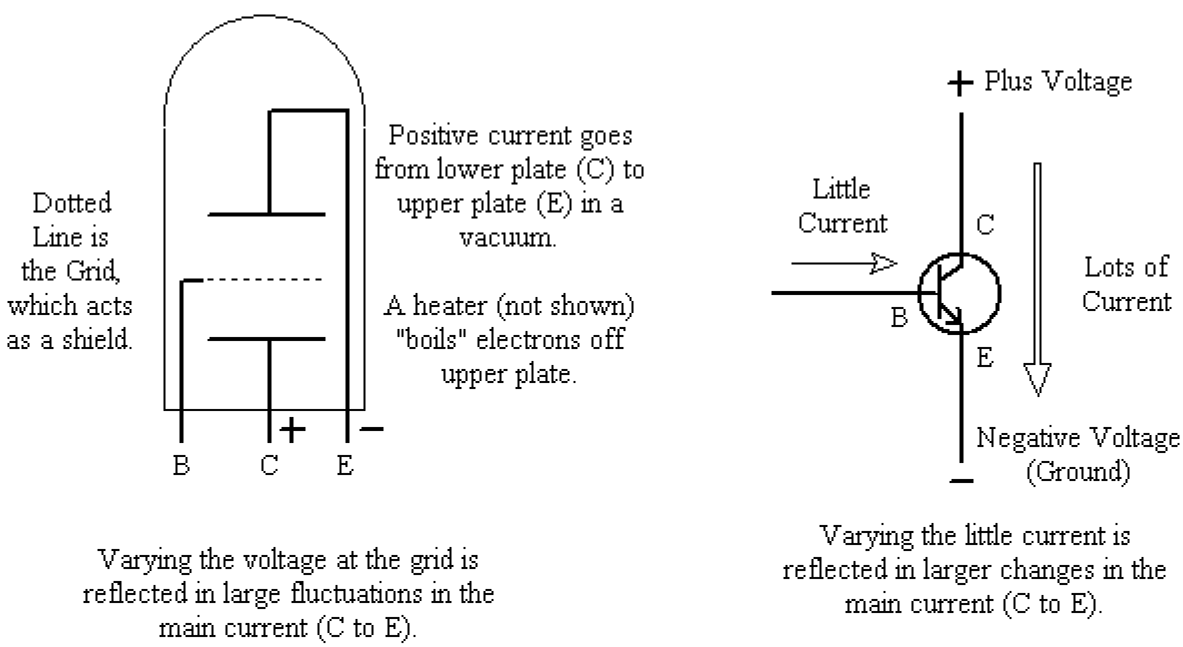


Amplifiers

We mentioned in the last chapter that we are not going to study the inner structure of the transistor, which is the semiconductor technology of solid-state physics. However, in this section we will present the old *vacuum tube* which played the role of a transistor. The transistor was discovered at *Bell Labs* in 1948. Later in the early 1970s, the integrated circuit (IC) could fit many transistors in smaller form on one chip. Today, we can have millions of tiny transistors in a chip we can hold in our hand. Such innovations made possible the desktop computer of the 1980s and beyond. All of these developments are not possible with vacuum tubes.

We can gain insight into the function played by a transistor by looking at its earlier counterpart, the *triode* vacuum tube. The triode was invented in 1907. Fig. N-2 below compares a triode to a transistor. Our aim is to understand how each functions as an amplifier.

Fig. N-2. Comparison of Triode Vacuum Tube (Left) and Transistor (Right).



There is a vacuum inside the tube in Fig. N-2. A heater (not shown) is used to boil electrons off the negative plate (connected to E), which plate is called the *cathode*. Electrons are also called *cathode rays*. Your TV has a cathode-ray tube that boils electrons off so they can be guided to the screen and light up phosphors. A current flows, by our convention, from C to E (opposite direction). This is a large current. Making the grid negative, discourages electrons to come toward it on their way to C. The grid can be thought of as a shield. Minor fluctuations in charge at B are then reflected in large changes in the main current from C to E. The triode amplifies the changes fed in at B.

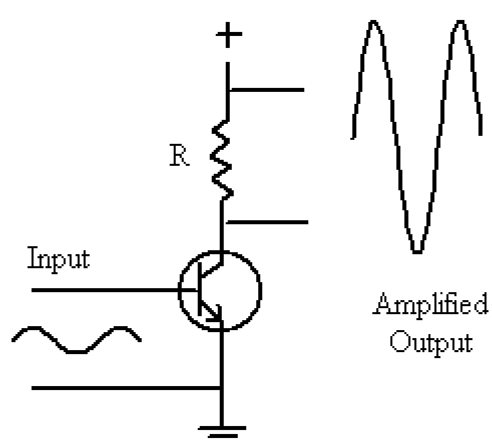
The transistor does essentially the same thing. A little current fed into the base B, activates the transistor so the pathway is now clear for lots of current to flow from C to E. Increase the little current, and you see a corresponding enhanced change in the main current. Think of the base as a stream that feeds into a structure that opens the dam to a river. A mechanical advantage is supplied by a fulcrum-counterweight structure (remember the record player) so that when the little stream pushes "a small trap door" open, the mechanism raises the flood gates for the river to flow. The plus and negative in our transistor diagram corresponds to differences in elevation. Plus means high elevation and negative means lower elevation, or *ground* in electronics terminology.

The river always wants to flow downstream but it can't because the main gate prevents it. Unless these gates are opened by the action of the little stream, nothing happens. If the little stream pushes the "trap door" only half way, then, the large volumes of the river water flow but at half strength. If the "back trap door" (the base B) is pushed in all the way, then the entire river current flows. Now play with the "back door" pushing it in and pulling it back rapidly. These small fluctuations are reflected in the larger changes of the river flow. The river current oscillates in step with the base stream oscillations. This is our amplifier action.

A small changing electrical signal fed into the base of the transistor or the triode produces a corresponding larger changing main current from the collector to the emitter. This larger varying current "to the tune" of our base-input signal is our amplified signal. The transistor is far superior to the vacuum tube. Transistors do not boil off electrons from a plate and therefore do not need heaters. Transistors are smaller. Therefore, devices made with transistors are smaller and cooler than those made with vacuum-tube technology.

Fig. N-3 illustrates the use of the transistor as an amplifier. The small input signal is a sine wave. Note that we need two wires for a signal. The bottom wire is ground and the top wire is the live wire on which our small voltage fluctuations occur. The small input oscillations are magnified in the main current fluctuations in step with it. In order to tap into this amplified signal, we attach a resistor on top of the transistor in Fig. N-3. When the current rushes down in step with the input wave, we get a voltage across the resistor according to V = IR. The resistor experiences large voltage fluctuations due to the large currents. By "grabbing electronically" on to this resistor, we tap off the circuit our amplified voltage signal.

Fig. N-3. Transistor as Amplifier.



Tuners

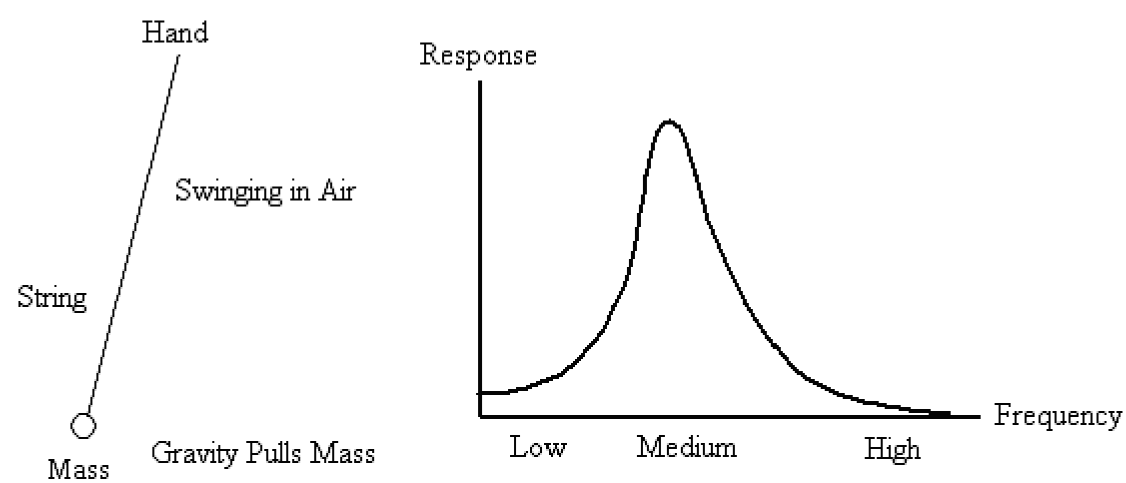
Tuners are radios without the amplifier and speakers. The circuit tunes in to receive a radio signal and decodes it. The tuner is another example of a source in our sound system. We have to amplify this weak source to get a strong output that can drive a speaker. We will focus on AM radio.

The AM tuner consists of two sections. First, the circuit that tunes in the radio wave and second, the part that decodes the signal. Remember, AM radio waves carry an encoded signal in the modulation of the amplitude.

The best way to understand the tuning circuit is by considering a mechanical analog, our old experiment shaking a ball on the end of a string (Fig. N-4). Let the shaking hand represent the incoming radio wave. Let the string represent an antenna. The signal we pick up is seen by the swinging ball. Refer to the resonance curve at the right in Fig. N-4. It's hard to get a response if the incoming frequency is too low or too high. However, for the middle frequency which we call the resonance frequency, we obtain the greatest response, i.e., amplitude for our swinging ball.

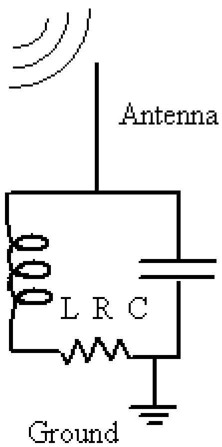
The ball always swings at the same frequency of the hand, but only for middle frequencies do we get a nice large sweep. We can say that the mechanical system is "tuned" to respond to the hand shaking at the resonance frequency. If the hand is shaking back and forth at a high frequency, this signal cannot get to the ball. But if we shorten the string, "tuning" our system to a different resonance frequency, we then can get a response. This is the mechanical analog of "tuning in" to get a specific radio station. Imagine many "invisible hands" trying to get the attention of the ball. Depending on the length of the string, one oscillating hand succeeds, the one shaking at the resonance frequency.

Fig. N-4. A Mechanical-Tuner Analogy.



Many signals in the air compete for the attention of your radio. However, the broadcasting frequency that succeeds is the resonance frequency you are tuned in to receive. The electrical analog to the mechanical resonance system is given in Fig. N-5. The incoming waves replace the moving hand. The incoming radio wave is an electromagnetic wave, a wave that electrons in an antenna can respond to.

Fig. N-5. Electrical Tuner Circuit.



L: Coil (Inductor), R: Resistor, C: Capacitor.

L acts as the mass (inertia), wants to keep the current going in same direction.
R is the resistance to the current (like the air in the mechanical case).
C is like gravity - too much charge on the plate wants to swing back.

The driving force (hand) is now the incoming radio signal. The antenna helps to "connect" this driving force to the circuit. The wire at the lower right is attached to a neutral body such as the Earth or chassis of a car.

The resonance frequency for the oscillating charges is determined by the circuit components L, R, and C. The movement back and forth of charges in response to the driving radio wave is analogous to the movement back and forth of the ball on the end of the string. If we pull the ball back and let it go on its own, it swings. If we force a lot of negative charges on one side of the capacitor, the charges will race through the circuit to the other side. But what keeps the ball going to overshoot the center position? The inertia of the mass of the ball.

Electrically, the coiled wire (the *coil*) does this. The reason is subtle. Consider the moment when the charges are flowing at full strength through the coil. Ampère's Law states that this current produces a magnetic field inside the coil. The charges begin to stop flowing through the coil when there aren't many left to flow. This decrease in current causes a decrease in the magnetic field produced by the current. The decrease in the magnetic field (a change) produces some electricity (Faraday's Law). This added electrical current gives the current "momentum" to keep going. So the charges overshoot the mark and charge the capacitor in the reverse way. Finally the current does stop.

The capacitor is then charged the opposite way. But now, the charges start moving again, this time the other way. The charges are moving to establish "equilibrium," i.e., no net charge on the capacitor. The whole process repeats in the reverse direction. They overshoot again and the capacitor is charged back the way it was before. After this, they start again, moving back the other way and so on. The charges "swishing" back and forth resemble our oscillating pendulum. The resistor in our circuit is analogous to the resistance of the air through which our pendulum swings.

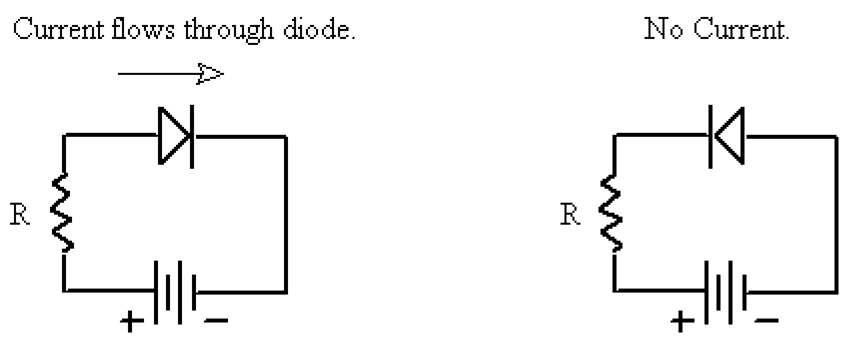
Let's continue discussing this subtle effect from different angles. When the charges start to flow, a magnetic field is produced (Ampère's Law). But going from no magnetic field to a magnetic field induces some current (Faraday's Law: changing magnetic field induces current). This electricity produced must oppose us. If not, we would be getting something for nothing. Before, when the magnetic field was decreasing, our current got a little kick (additional electricity) to keep the current alive for a little bit longer. Now, the magnetic field is increasing, so the electrical surge due to Faraday's Law wants to assist in keeping things the way they are - dead. Nature seems to work this way. It often opposes our desires.

If we want to decrease the magnetic field inside the coil by cutting back on the current, nature works against this by giving the current a little kick in the same direction the current is going. But this is the opposite of what we are trying to do. If we want to increase the magnetic field inside the coil by an increase in current, nature works against us by "kicking" in the opposite direction. In each case, nature is trying to preserve the status quo, against our wishes to change things. This opposition in the circuit is called *Lenz's Law*. It is analogous to Galileo's *Law of Inertia* (also Newton's First Law) which states that nature wants to keep objects doing whatever they are doing. If a mass is at rest, it wants to stay at rest. If it's moving, it wants to keep moving. You need a force like friction to stop a sliding box.

The electrical analog to the Law of Inertia is found in the behavior of the inductor (a coil). Current through a coil wants to keep doing whatever it's doing. So just as the mass of a pendulum overshoots equilibrium due to its inertia, the charges moving through an inductor overshoot and charge the capacitor in a reverse manner. Note that the coil is necessary because the coil allows for the interplay of Ampère's Law and Faraday's Law. This is deep physics!

The second part of the AM radio is the decoder of the amplitude modulation. The basic decoder is a *diode*. A diode is an electrical circuit element that lets current pass in only one direction. Fig. N-6 depicts a diode. Current passes through along the direction indicated by the triangle. Current trying to go the other way is blocked (right diagram in Fig. N-6).

Fig. N-6. Diode.



A diode lets current pass in one direction only.
The resistor R limits current to safe values.

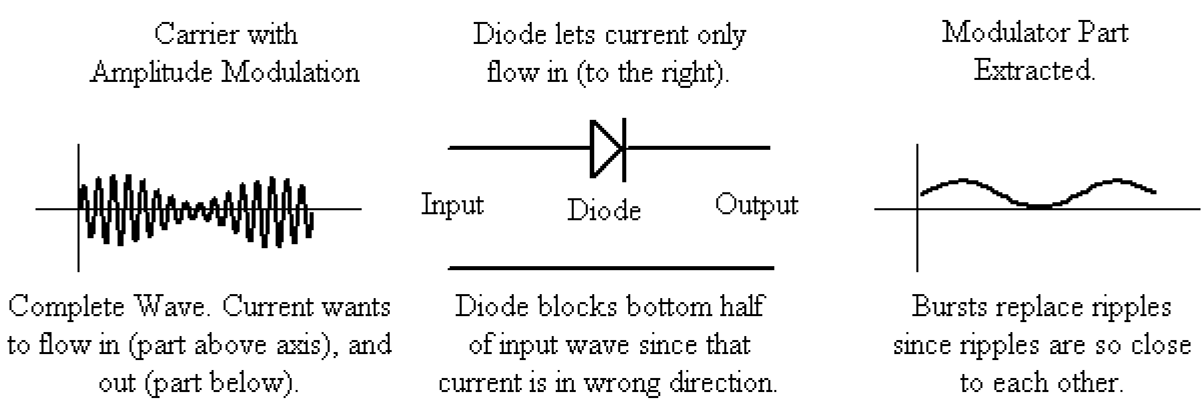
Fig. N-7 shows how a diode can be used as a decoder for amplitude modulation. The incoming wave is oscillating to and fro. It wants to quickly push current into the diode and then pull current back in the opposite direction. The diode passes the "to" part and blocks the "fro." Current can only pass through a diode in one direction. Therefore, the output is just the top part of the input wave, that part representing pushing current into the diode. The part representing the pulling of current back (below the horizontal axis) is blocked.

Since the carrier wavelength is so short (ripples) compared to the modulation of the amplitude (long wavy contour of amplitude), we sketch the output with these ripples merged. The result is the wavy contour of the amplitude. But do you remember why the carrier wavelength is so short and the modulation wavelength long?

The carrier wave is a radio wave. For an AM radio station, a typical frequency is 1310 kHz (*WISE*). This is 1.3 MHz, i.e., about 1 million times a second. The amplitude varies at an audio frequency. Take a sine wave of 100 Hz (a bass tone). That is only 1 hundred times a second. Let's compare 100 with 1,000,000. Well, one million is 10,000 times more! That means the radio carrier wave wiggles back and forth 10,000 times before the amplitude undergoes one crest and trough! So our picture at the left in Fig. N-7 is exaggerated. Fig. N-7 shows about 16 ripples for our carrier wave as it goes through about one cycle and a half of amplitude variation. There should be thousands of ripples instead.

The output wave is a result of the current that went through the diode in the correct direction. This is the upper part of the input wave. We can consider the output as bursts of current since the carrier waves (ripples) are closely packed (thousands) and all the current now is in the same direction. These bursts occur at the modulator frequency. This is our sound wave, i.e., the wave at 100 Hz. We have decoded (demodulated) the AM radio signal. We send this to an amplifier and then to a speaker to hear it.

Fig. N-7. Diode Used as Demodulator.

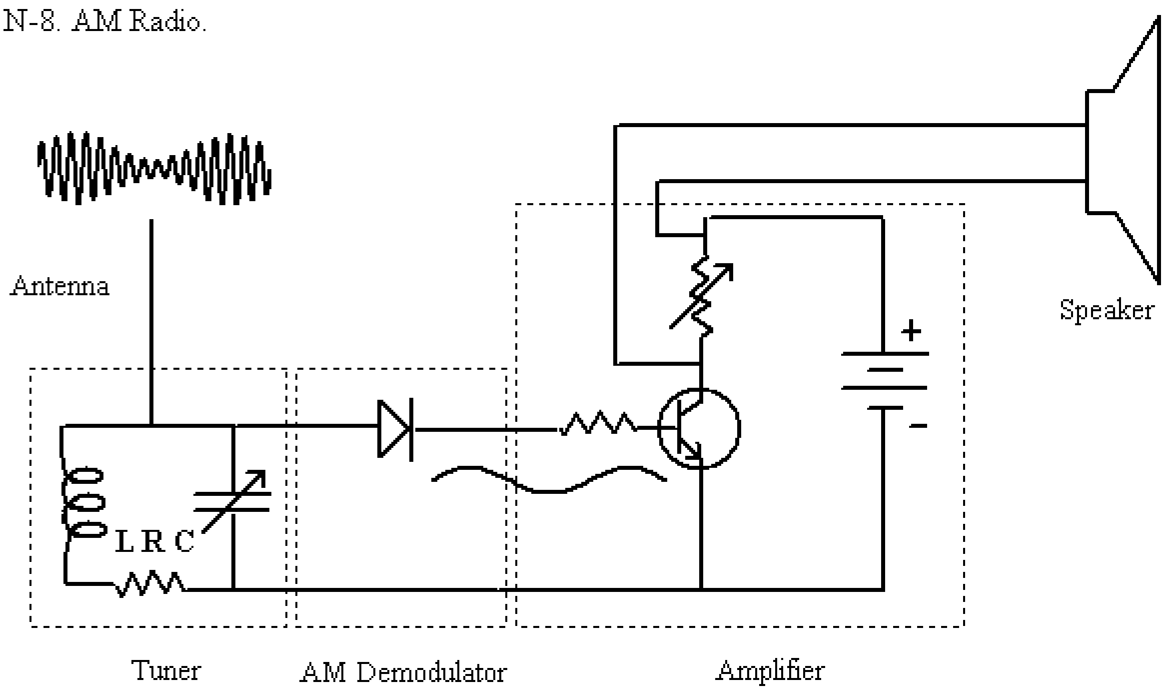


Now we are ready to put all of this together: the tuner, the demodulator, the amplifier, and the speaker. The result is Fig. N-8, one big impressive diagram. Focus your attention on each component. These have already been discussed. If you are confused, localize the difficulty. Then review that section in the text in order to understand that particular part. Note that the amplifier circuit includes a variable resistor so you can control the volume of the output signal. The capacitor likewise has a variable control. You also know the inside structure of a speaker from an earlier chapter. So you really understand quite a bit about the circuit in Fig. N-8. The ground wire (see Fig. N-5) is not shown in Fig. N-8. Sketch it in.

In the old days (the 1950s) it was common to buy a tuner, consisting of the tuner section and the demodulator. You would really be buying two circuits since your tuner would be able to pick up AM and FM radio waves. This consisted of one of your source components. Remember, the source component is stage one in sound reproduction. The amplifier is stage two. That would be your second component. Then you would buy speakers for the third stage.

Manufacturers began combining the tuners with the amplifiers. These combo units are called receivers. Today, it is very common to buy a receiver. You get the radio and amplifier. This assumes the higher-end product where components are specialized. Of course, you can buy a radio which includes everything, tuner, amplifier, and speaker(s). The author had a 6-transistor pocket AM radio as a kid in 1963, complete with everything. The 6 transistors allowed for stages of amplification. The little radio used a 9-volt battery and could pick up Houston or Cincinnati from Philadelphia at night as AM radio waves bounced off the upper atmosphere (the *ionosphere*, which is higher at night).

Fig. N-8. AM Radio.

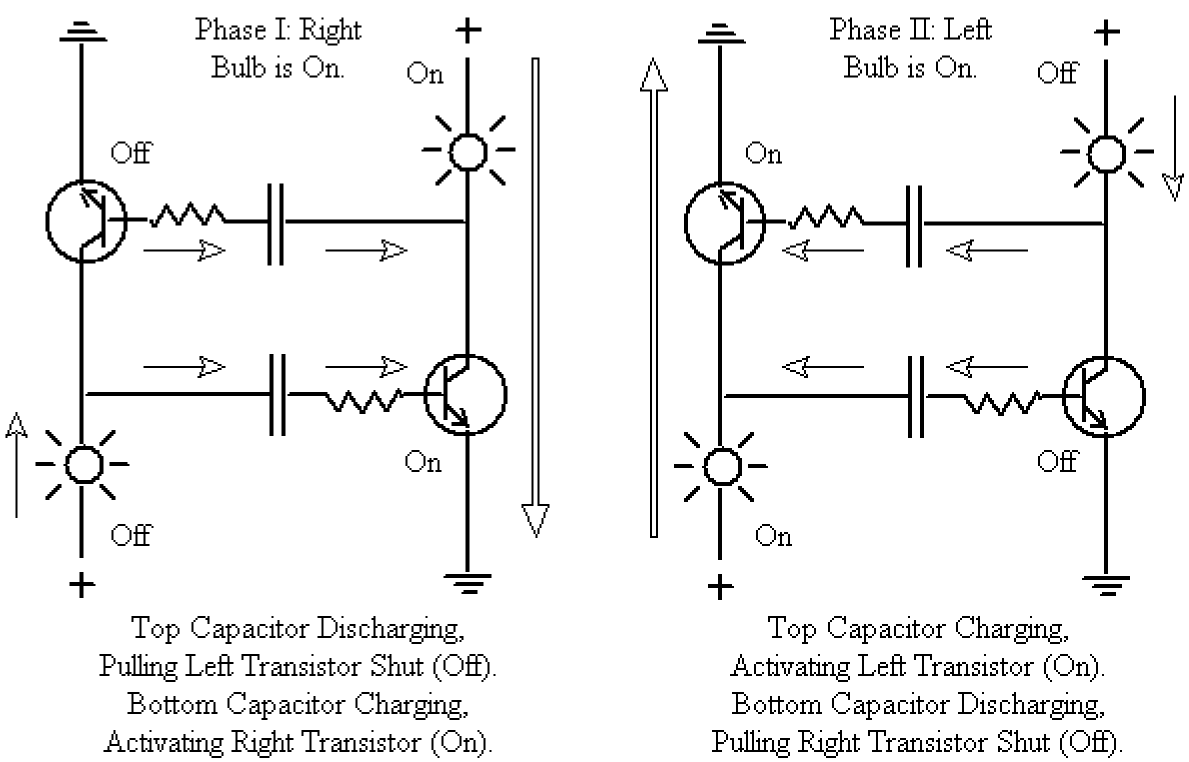


Oscillators (Optional Section)

Oscillator circuits are important for both radio frequencies and audio frequencies. An oscillator is used to produce the carrier waves in radio broadcasting. Oscillators are also used at audio frequencies to produce tones in music synthesizers. In the spirit of this text, the simplest possible oscillator circuit is given in Fig. N-9. Try to get an overall understanding for what the circuit does. Shift your eyes back and forth between the diagrams below. The current changes direction. The bulbs take turns going on and off. Only one is on at a time.

The battery for the circuit is not shown directly. The plus and minus (ground) symbols indicate where to connect to the battery terminals. In the left diagram, the bottom capacitor is charging. The current going through the left bulb is too weak to light it up. The top capacitor is discharging, "sucking" current from the left transistor (pulling the "trap door" the wrong way a little). The left transistor is off. When the lower capacitor is fully charged, the reverse (right diagram) takes place. The left bulb goes on; the right bulb goes off. This circuit is called the *astable multivibrator*. We can replace the bulbs with resistors. Then tap off one of these resistors and we have our oscillator. The particular resistor-capacitor combination chosen for the charging circuits determines the frequency of the oscillator.

Fig. N-9. Oscillator Circuit.



Digital Electronics

In digital electronics, the voltage can be in any of two states: on or off. We will consider 1 volt for on and 0 volts for off or simply 1 and 0. Digital electronics is the foundation of the digital computer. The computer only knows 1 or 0; however, zillions of these give the computer the ability to do anything imaginable. Below we introduce the logic of using 1s and 0s.

NOT (INVERTER)

The first example of a digital-logic component is the NOT or Inverter. Have you ever changed your mind about something or said to a friend “You look good. NOT!” You negate or invert your comment. We change Yes to No or vice versa. See Table N-10a.

Table N-0a. Inverter Logic

Before	**After**
No	Yes
Yes	No

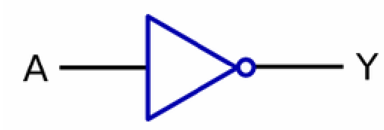
We will take 1 = Yes and 0 = No. You have some position on an issue, but after some reflection, you change your mind. Then the table is given as below. It is called a *Truth Table*. It lists all the cases. We refer to "Before" as the "Input" or "In" for short and "After" as the "Output" or simply "Out." Then we label “In” as A and “Out” as Y.

Table N-0b. Truth Table for Inverter Logic

A	**Y**
0	1
1	0

We would also like to have a visual representation of this logic in symbolic form. We are led us to our first digital-electronics symbol, also called a *Gate*.

Fig N-10. Inverter Gate Symbol



AND

Consider two qualities you must have in your date; otherwise, you do not date the person. Perhaps, you insist on a date that is attractive to you in some way and one that has polite manners. This is an example of an AND condition. We summarize this logic in the following truth table.

Table N-1a. AND Logic

Attractive in Some Way	**Polite Manners**	**Decision to Date**
No	No	No
No	Yes	No
Yes	No	No
Yes	Yes	Yes

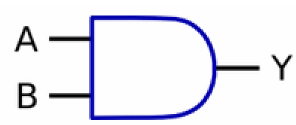
We generalize this logic by letting A = "Attractive in Some Way" and B = "Polite Manners." Then we take 1 = True and 0 = False. The Decision is the Output or Out.

Table N-1b. Truth Table for AND Logic

A	**B**	**Y**
0	0	0
0	1	0
1	0	0
1	1	1

The AND Gate symbol is given below.

Fig. N-11. AND Gate



OR

Consider two qualities, where only one is necessary for your decision to be true. Perhaps you are at your grandmom's house and she offers you lunch. Choice A = sandwich and Choice B = soup. But Grandmom says you can have both if you want. Then, the condition of eating lunch is satisfied by eating the sandwich, having the soup, or accepting both.

Table N-2a. OR Logic

Eat Sandwich?	Have Soup?	Did You Have Lunch?
No	No	No
No	Yes	Yes
Yes	No	Yes
Yes	Yes	Yes

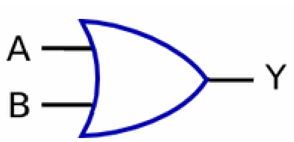
The corresponding generalized truth table is below.

Table N-2. Truth Table for OR Logic

A	B	Y
0	0	0
0	1	1
1	0	1
1	1	1

The OR Gate symbol is in Fig. N-12.

Fig. N-12. OR Gate



XOR

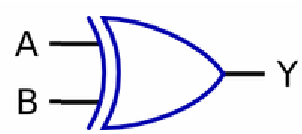
Now comes the "Exclusive OR." Grandmom says you can only have one for lunch: soup or sandwich. There is just not enough food to go around for everyone. In this case, the logic is summarized in Table N-3.

Table N-3. Truth Table for XOR Logic

A	B	Y
0	0	0
0	1	1
1	0	1
1	1	0

The XOR Gate is below. Remember the XOR by thinking both choices are "excluded: you can have one or the other, but not both.

Fig. N-13. XOR Gate



Can you think of other scenarios for the XOR?

NAND

Consider the NAND as the negation of the AND. You just flip the answers of the AND to their opposites. See Table N-4a.

Table N-4a. Truth Table for NAND Logic

A	B	Y
0	0	1
0	1	1
1	0	1
1	1	0

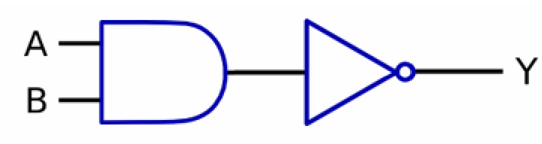
Below is another way of looking at your date requirements (see Table N-4b).

Table N-4b. NAND Logic

Attractive in Some Way	Polite Manners	Decision to Stay Home
No	No	Yes
No	Yes	Yes
Yes	No	Yes
Yes	Yes	No

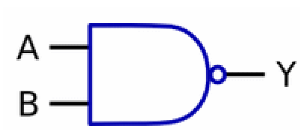
We can write the negative of the AND output with the inverter. See Fig. N14a.

Fig. N-14a. Constructing NAND Logic



The abbreviated form for Fig. N-14a is Fig. N-14b.

Fig. N-14b. NAND Gate



NOR

Let's return to our visit to Grandmom's for lunch where she serves soup and sandwich. Suppose we ask the question "Did you fast at Grandmom's Luncheon?" The answers to this question are listed in Table N-5a.

Table N-5a. NOR Logic

Eat Sandwich?	Have Soup?	Did You Fast?
No	No	Yes
No	Yes	No
Yes	No	No
Yes	Yes	No

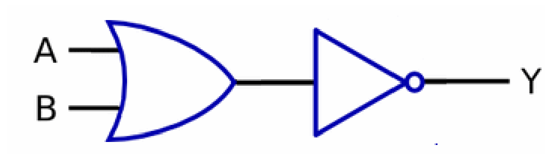
The general truth table is Table N-5.

Table N-5. Truth Table for NOR Logic

A	B	Y
0	0	1
0	1	0
1	0	0
1	1	0

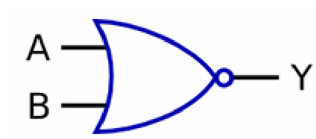
This table is the inverted version of the OR. Compare the output Y columns. You find that the NOR has the opposites compared to OR. We illustrate this in Fig. N-15a.

Fig. N-15a. Constructing NOR Logic



The abbreviated form for Fig. N-15a is Fig. N-15b.

Fig. N-15b. NOR Gate



You just place a little "bubble" on the end of the OR. Sometimes, NAND and NOR are referred to as a bubbled AND and bubbled OR respectively.

XNOR

Let's flip the outputs for the XOR. We then get the truth table in Table N-6.

Table N-6. Truth Table for XNOR Logic

A	B	Y
0	0	1
0	1	0
1	0	0
1	1	1

Can you think of an example where this logic would apply? You can have nothing or both. It's an all or nothing deal! How about maintaining balance on a plank where A = "Hold a Weight in Outstretched Left Arm" and B = "Hold a Weight in Outstretched Right Arm." When are you balanced?

My students taught me this next example. Let A = 1 mean A is in love with B and let B = 1 indicate that B loves A. Then the acceptable scenarios are the Platonic relation A = B = 0 or the love relation A = B = 1. The unrequited love cases have Y = 0. These relationships are undesirable.

You might ask why we don't write this one as NXOR with the N out if front? Well, we just don't. It would be too hard to pronounce it that way.
The corresponding digital circuit symbols are shown in Fig. N-16a and Fig. N-16b.

Fig. N-16a. XNOR Logic

Fig. 17. Summary of Digital Logic Gates

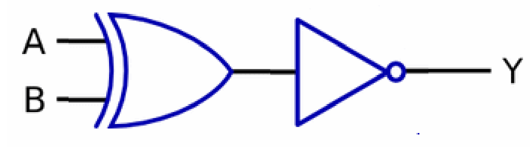
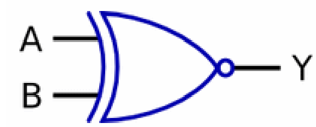
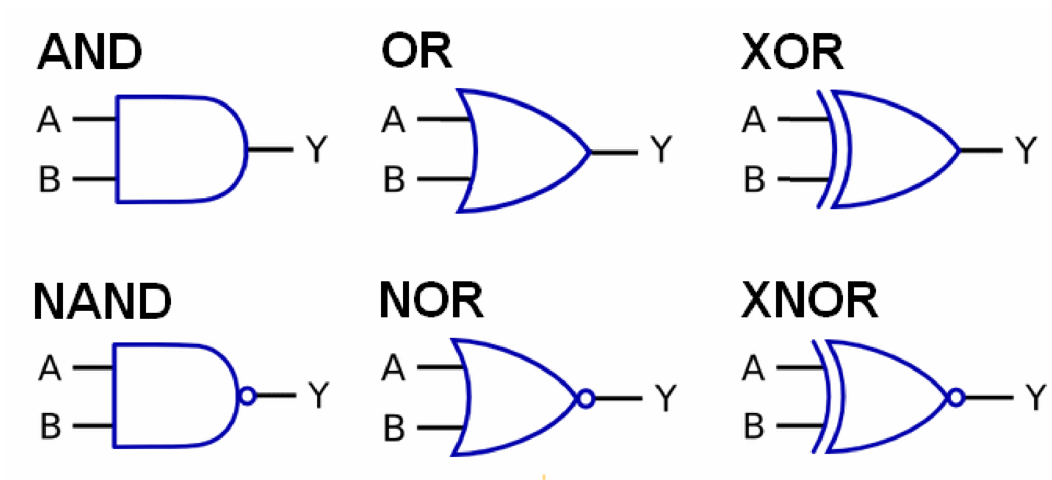


Fig N-16b. XNOR Gate



Check out the super summary below of all the gates (Fig. 17).



AND

A	B	OUT
0	0	0
0	1	0
1	0	0
1	1	1

OR

A	B	OUT
0	0	0
0	1	1
1	0	1
1	1	1

XOR

A	B	OUT
0	0	0
0	1	1
1	0	1
1	1	0

NAND

A	B	OUT
0	0	1
0	1	1
1	0	1
1	1	0

NOR

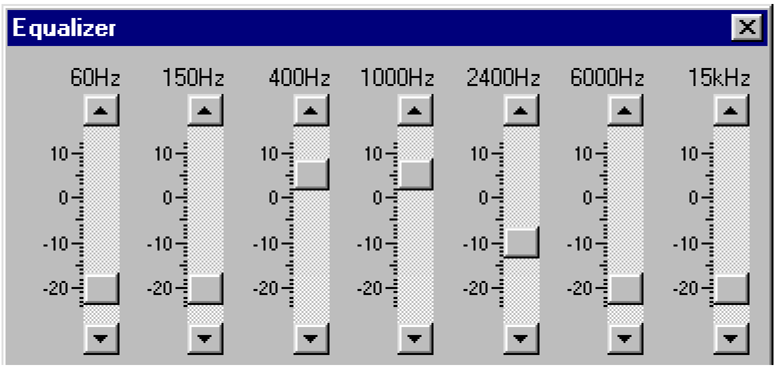
A	B	OUT
0	0	1
0	1	0
1	0	0
1	1	0

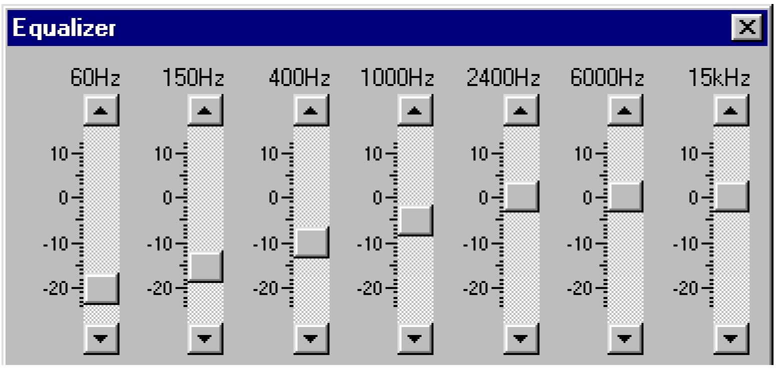
XNOR

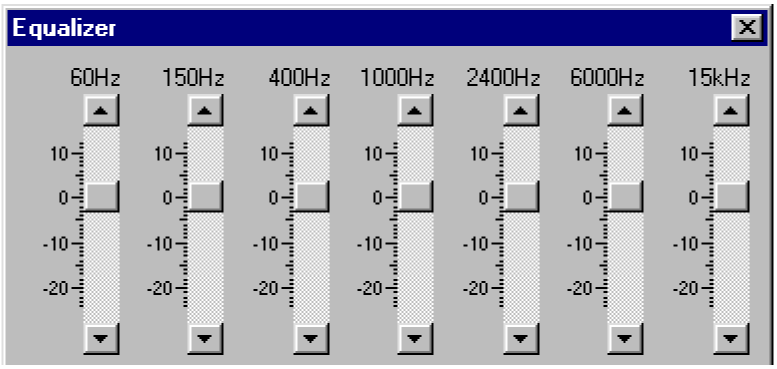
A	B	OUT
0	0	1
0	1	0
1	0	0
1	1	1

--- End of Chapter N ---

O. Signal Processing







The author's father once told the author around the late 1950s that no tape or recording could be better than the original. Something is lost in the recording. Nothing is perfect. There is no such thing as perfect high-fidelity reproduction. However, the author's father did not live to see the 1980s and beyond, where signal processing challenges this assertion. Here is a specific example. Suppose the original lacks reverb and is too weak in the bass. Then we correct these deficiencies in the processing stage before making the master. The recording is enhanced! You process the signal at home when you play around with bass and treble controls. If you have an equalizer (to be discussed), you process your signal even more.

Philosophically, the original is not reproduced exactly. However, we can get "garbage" along with the good stuff. If we can eliminate some of the "garbage" during the intermediate processing stage, then can we say that the final product is better? The debate over whether a recording is "better" than the original may remind you about our earlier discussion concerning the existence of sound when a tree falls in a forest.

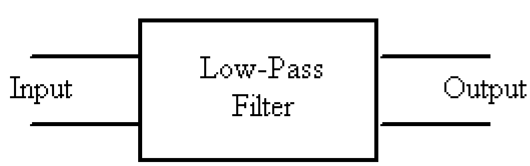
But suppose the source is a digital synthesizer or computer. The information is encoded as 1s and 0s on a diskette. If we copy the diskette, the copy is indeed an exact reproduction of the original! And suppose a digital recording of an artist who hits 5 wrong notes is made and then the mistakes are corrected during the digital-processing stage? The author is sure that his dad did not have these tricks in mind when he made his statement during the early days of stereo and tape recording.

Filters

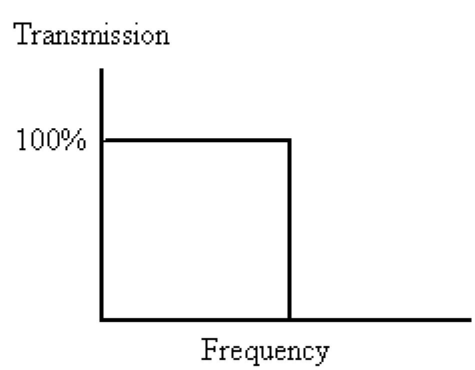
One of the essential features of signal processing is the use of filters. Filters can reduce frequencies that are not as important as others. An *active filter* boosts frequencies transmitted by combining a filter with a dedicated amplifier for the filter. A filter without combined amplifier assistance is called a *passive filter*.

Fig. O-1 illustrates an ideal filter that filters out high frequencies beyond a certain point. The graph at the right gives the percentage of the amplitude that gets transmitted for each frequency. Real filters do not cut off so abruptly.

Fig. O-1. Ideal Low-Pass Filter.



Input sine waves of different frequencies.
See what comes out.



We send in sine waves of different frequencies and see how much of the original wave comes out. We find that 100% of the wave gets through if its frequency is low. If the frequency is higher than a certain frequency, we get nothing (0% transmitted). The filter is ideal. Since actual filters are not "cliffs" that drop off so perfectly, we would find that some intermediate sine waves get partially transmitted. However, when we get to high enough frequencies, they would not get through at all.

Fig. O-2 presents us with a specific ideal filter (labeled with numbers) that transmits low frequencies. We say the filter is a *low-pass filter* because it "passes" low frequencies. Note that sine waves less than 275 Hz get transmitted and sine waves beyond 275 Hz cannot pass through the filter. This boundary value of 275 Hz is called the *cutoff* frequency. Sine waves with frequencies beyond 275 Hz get "cut off." Fig. O-2 analyzes what happens to an incoming 50-Hz ramp wave. Note that the filter rule applies to sine waves. So we need to decompose the ramp wave into sine waves for analysis (lower left diagram). The first harmonic is 50 Hz, the second 100 Hz, and so on. The relative amplitudes we know from our earlier chapter on Fourier analysis. All harmonics below 275 Hz pass through the filter (see lower right diagram).

Fig. O-2. Sending a Ramp Wave Through An Ideal Low-Pass Filter.

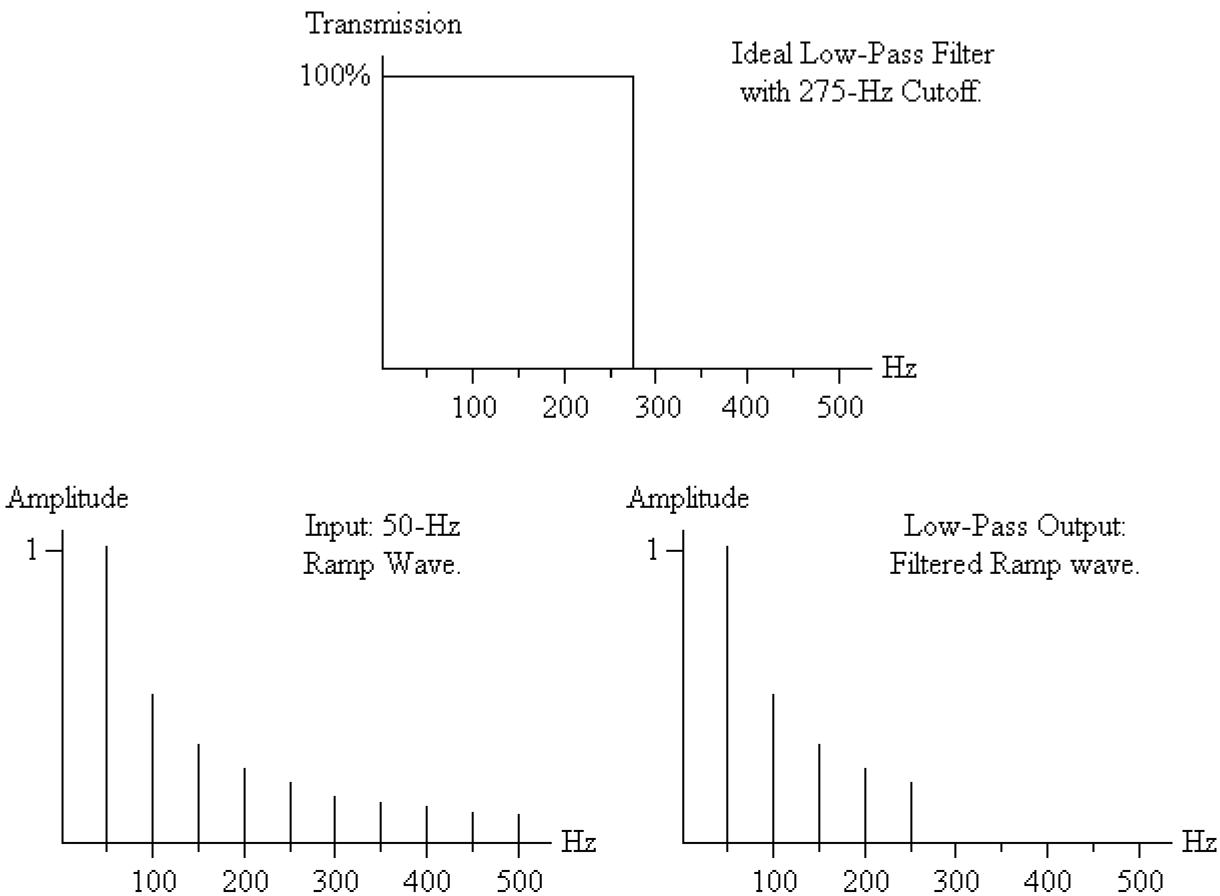


Fig. O-3 illustrates an ideal *high-pass filter*. This filter passes high frequencies instead of low frequencies as we saw earlier. Once again, we send in sine waves to determine the transmission rule for the filter.

Fig. O-3. Ideal High-Pass Filter.

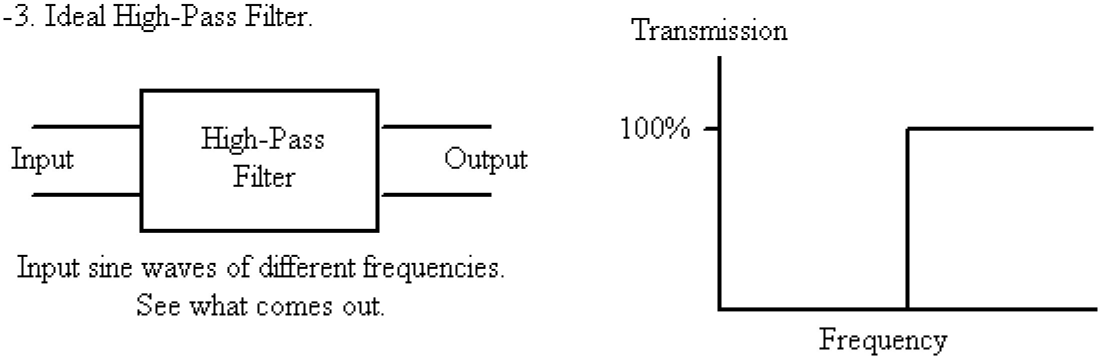


Fig. O-4 gives a specific example of an ideal high-pass filter using our ramp wave again. This time, the higher harmonics of the ramp wave pass through.

Fig. O-4. Sending a Ramp Wave Through An Ideal High-Pass Filter.

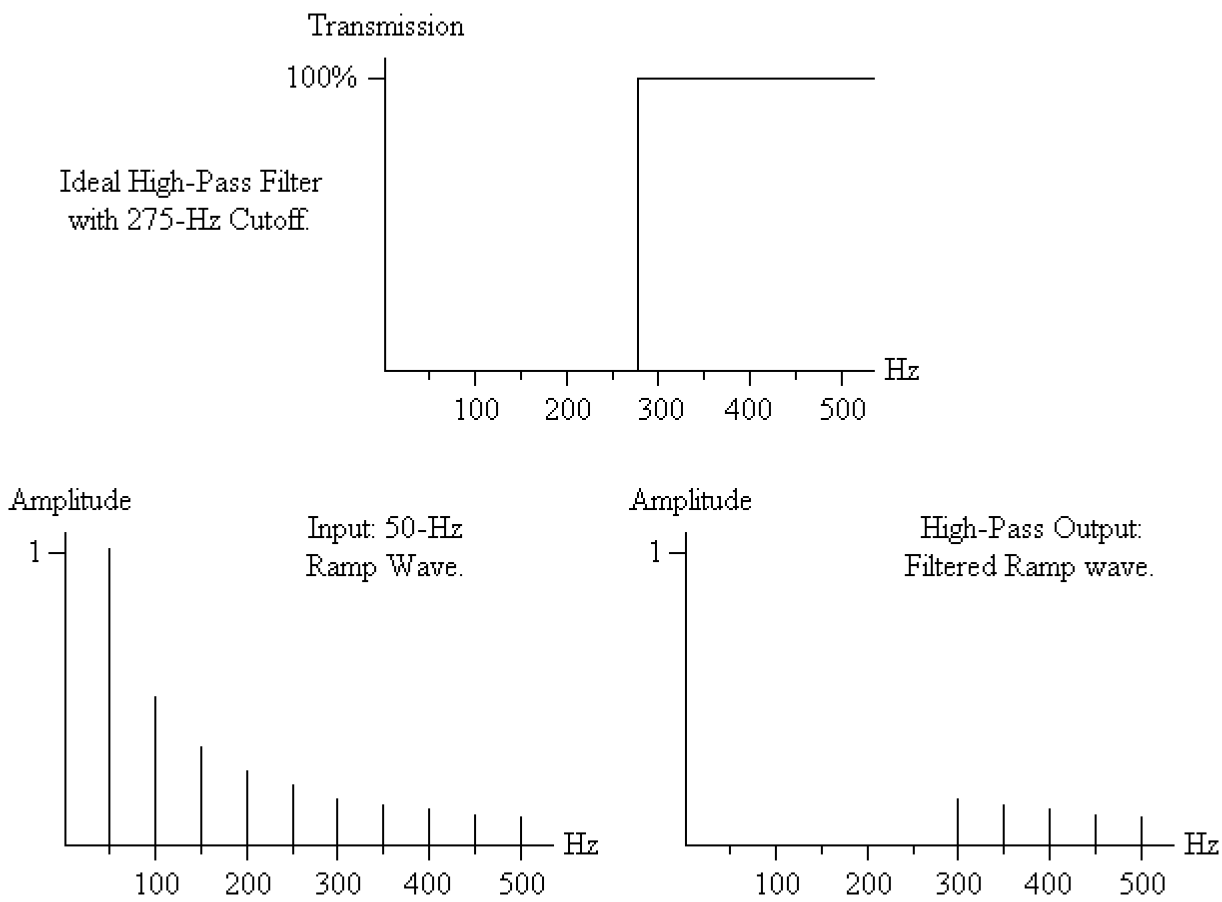


Fig. O-5 illustrates an ideal *bandpass filter*. This filter passes middle frequencies. We have a low cutoff and a high cutoff.

Fig. O-5. Ideal Bandpass Filter.

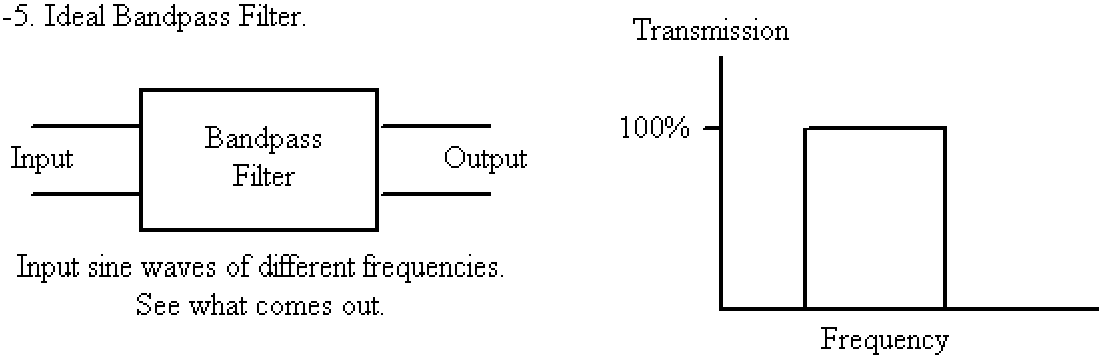
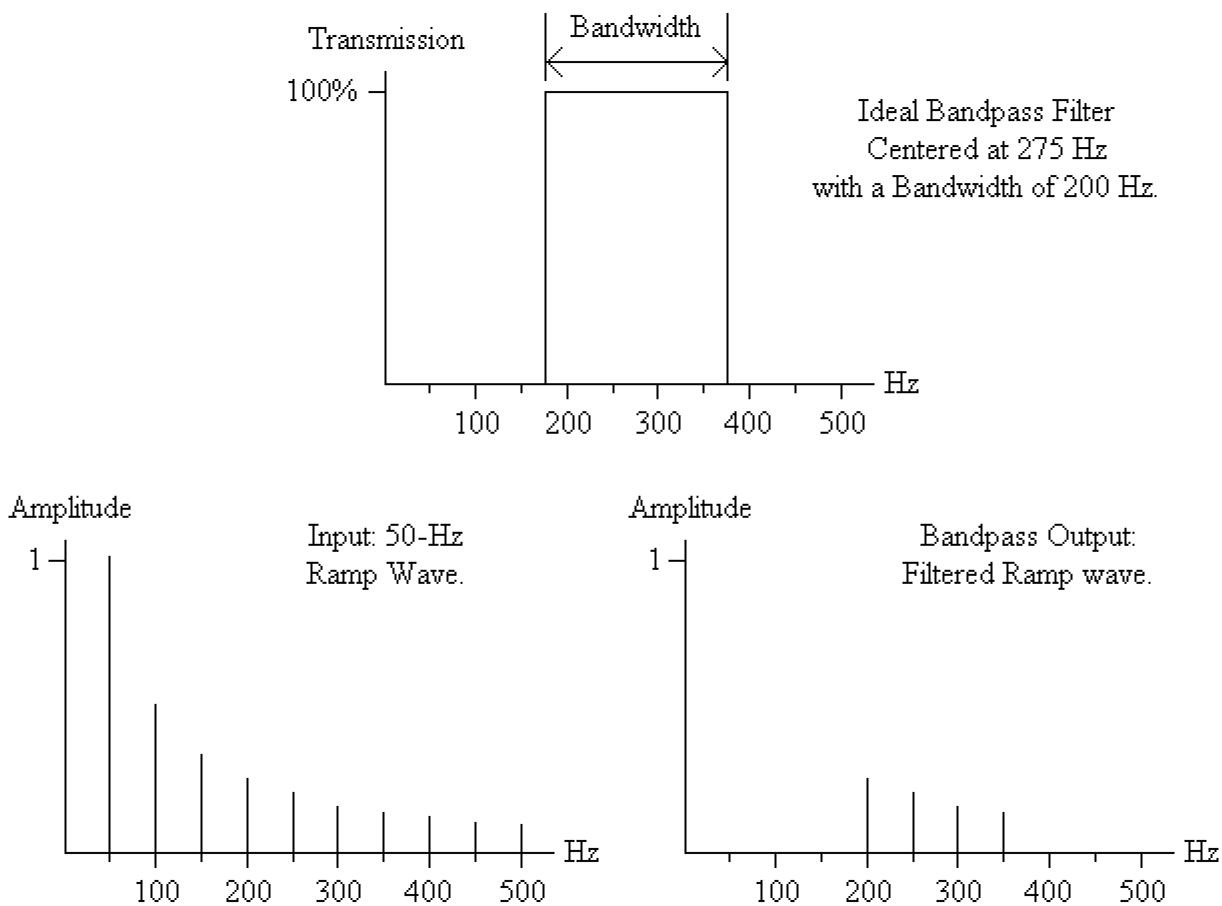


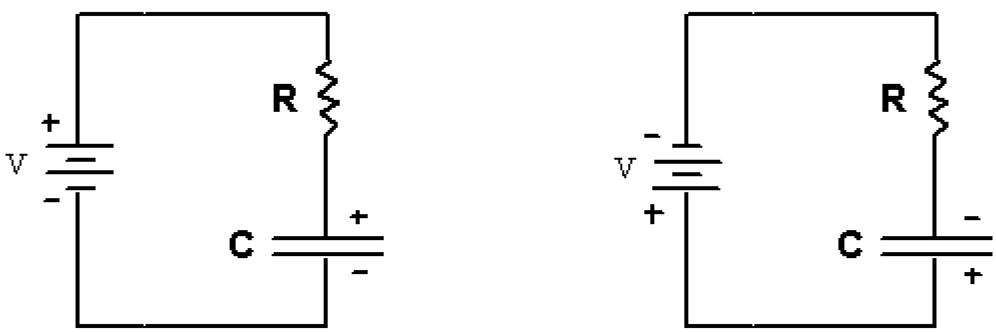
Fig. O-6 gives a specific filter. It is convenient to refer to the bandpass filter by its central frequency and then give the width of the filter (the *bandwidth*). The filter in Fig. O-6 is centered on 275 Hz and it has a 200-Hz bandwidth (175 to 375). It passes frequencies within this band.

Fig. O-6. Sending a Ramp Wave Through An Ideal Bandpass Filter.



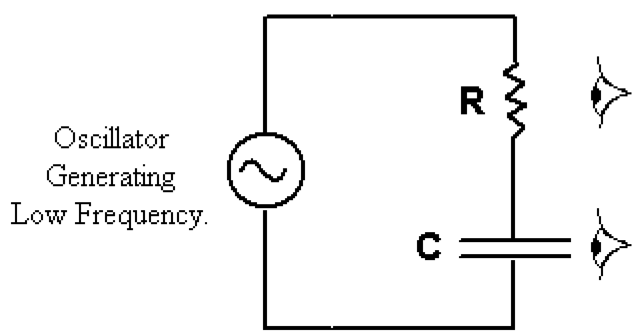
A simple filter can be made with one resistor and one capacitor (RC circuit). To understand why, we review the concept of charging a capacitor. In Fig. O-7 we charge a capacitor two ways. In the left case, electrons (negative) gather on the bottom plate of the capacitor as they try to get to the positive side. The absence of electrons at the top makes the upper plate positive. The battery is reversed in the right case so the charge on the capacitor is reversed. A key idea is that it takes time to charge the capacitor in either case because of the resistor. Now imagine flipping the battery orientation back and forth. This is basically what the oscillator does in Fig. O-8.

Fig. O-7. Charging a Capacitor.



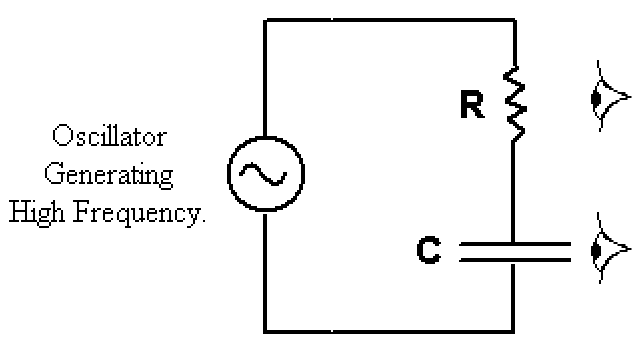
Plus on Top Capacitor Plate. Plus on Bottom Capacitor Plate.

Fig. O-8. Sending an Oscillating Signal into an RC Circuit.



Changes are so slow that very little current ever flows through R. This small I means small voltage signal (V = IR) across R.

The low frequency gives the capacitor time to charge one way and then the other. The signal is seen at the capacitor.



The rapid surges in current are seen at R but charge never gets a chance to collect on the capacitor.

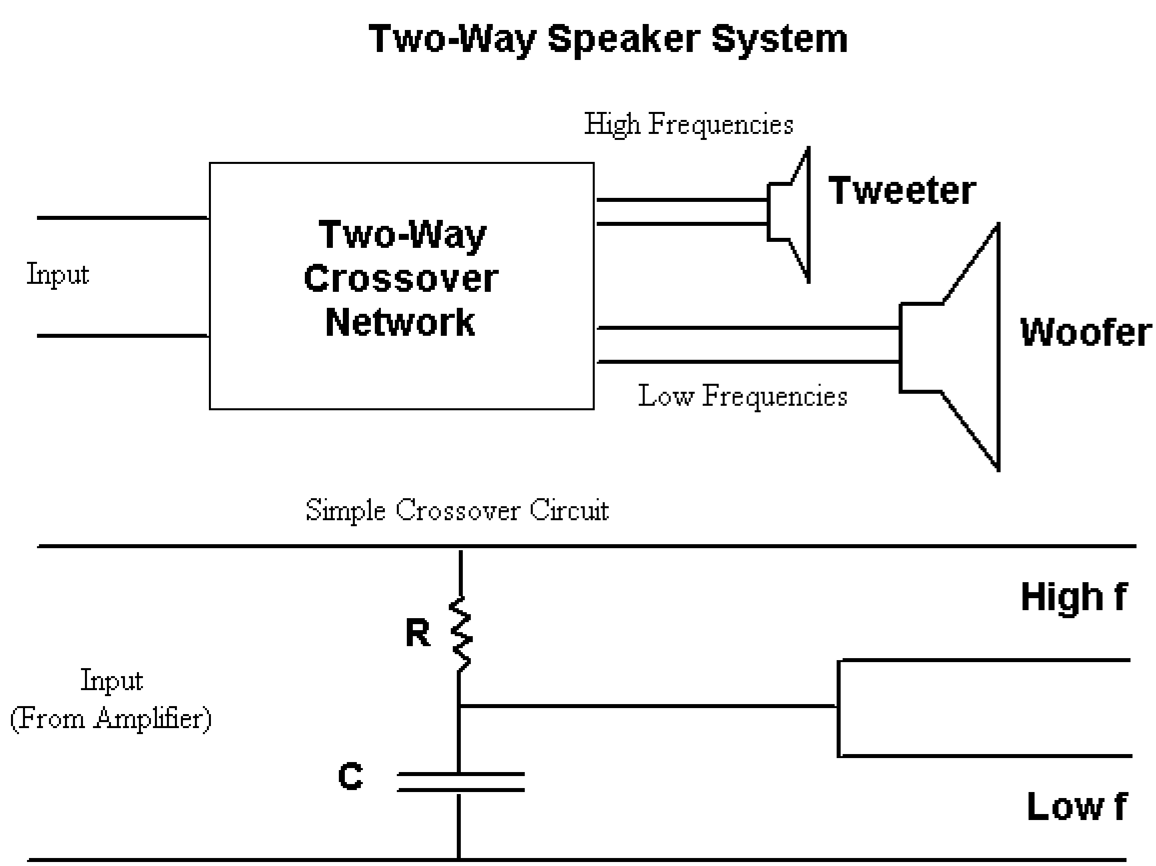
Rapid changes do not allow the capacitor to charge. The signal is not seen at the capacitor.

Carefully study the comments in Fig. O-8. From these, we see that the low-frequency oscillations are directed to the capacitor, while the high-frequency oscillations are picked up at the resistor. The RC circuit splits the frequencies. By choosing specific numerical values for the resistance and the capacitor, the frequency where the division occurs can be chosen.

In a two-speaker unit, this simple RC circuit can be used to direct low frequencies to the larger bass speaker and send the high frequencies to the smaller speaker. This circuit is called a *crossover circuit*. The low frequencies "cross over" to the bass speaker, called a *woofer*. Remember this by thinking that "woof-woof" is a low-frequency sound. The woofer is large and can support longer wavelengths. The high frequencies "cross over" to the smaller speaker, which is called a *tweeter*. Remember this by thinking that "tweet-tweet" is a high-frequency sound. The tweeter better supports short wavelengths. The size of a vibrating system is related to the wavelength.

Long strings produce bass, short ones high pitches. Similarly, large speaker membranes can support long wavelengths very well. Small speakers handle the short-wavelength high tones better. Fig. O-9 illustrates a crossover network which routes the low and high frequencies to the appropriate speakers. Compare the lower diagram in Fig. O-9 with Fig. O-8.

Fig. O-9. Crossover Network for Two-Speaker System.



We are now ready to give RC circuit diagrams for the low-pass and high-pass filters. Our *crossover network* can serve as a guide. Fig. O-10, reproduces our *crossover network*. To make a low-pass filter, focus on the lower section with the capacitor. Throw out the wire at the upper right of the resistor. Then bend the resistor down to the left and straighten out the wire to its left. The result is the low-pass filter in Fig. O-11. For the high-pass filter, discard the wire to the lower right of the capacitor. Bend the capacitor up and straighten out the wire to its left. Then flip the whole thing from top to bottom to obtain the circuit in Fig. O-11. For alternating signals, it doesn't matter that the circuit is flipped from top to bottom.

Fig. O-10. Basis For Low-Pass and High-Pass Filters.

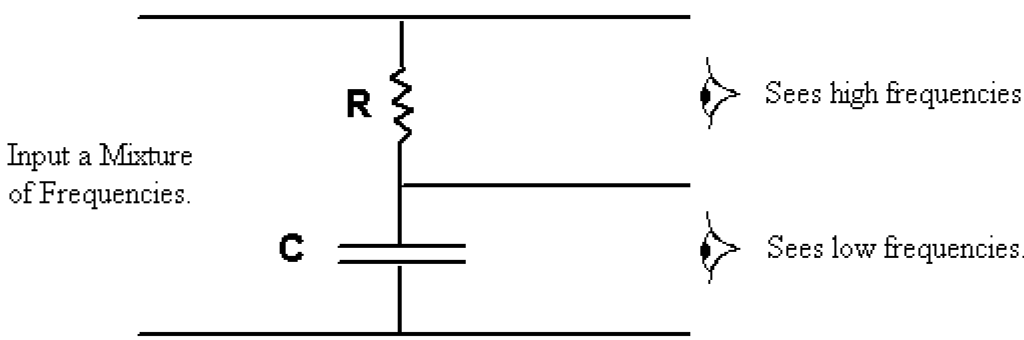
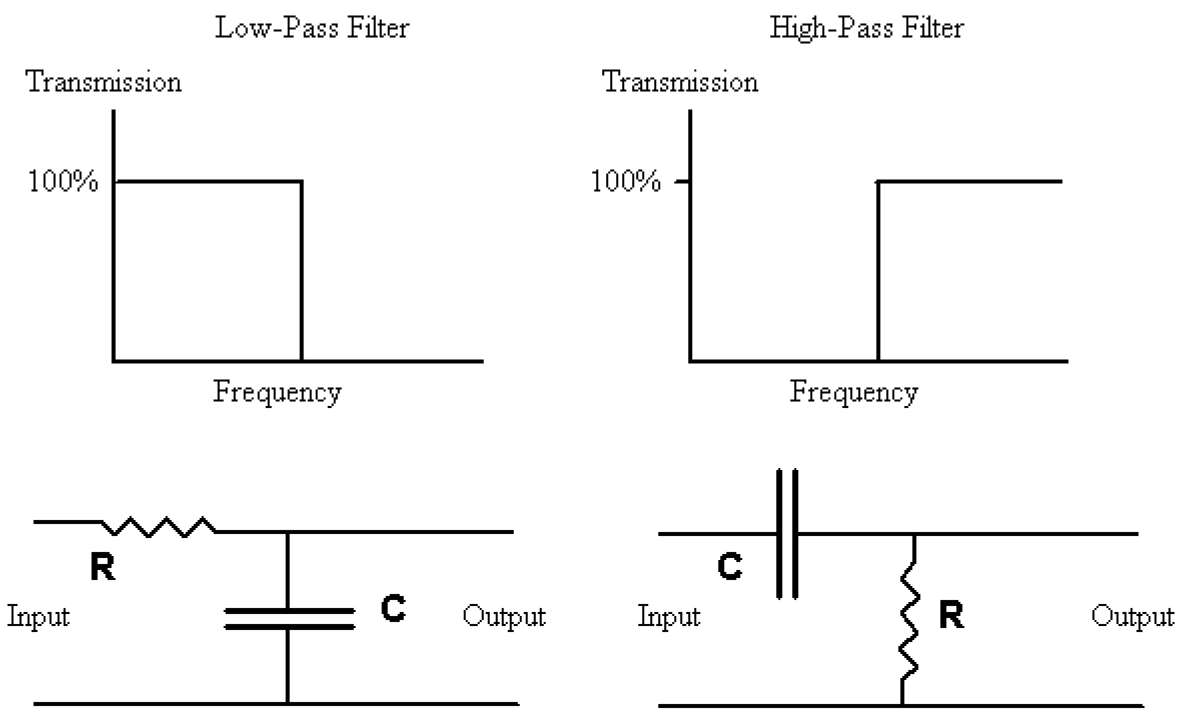


Fig. O-11. Low-Pass and High-Pass Filters.



We can obtain a bandpass filter by combining a low-pass filter with a high-pass filter provided that there is common overlap between the two. See Figs. O-12 and O-13.

Fig. O-12. Low-Pass and High-Pass Filter to be Combined to Form a Bandpass Filter.

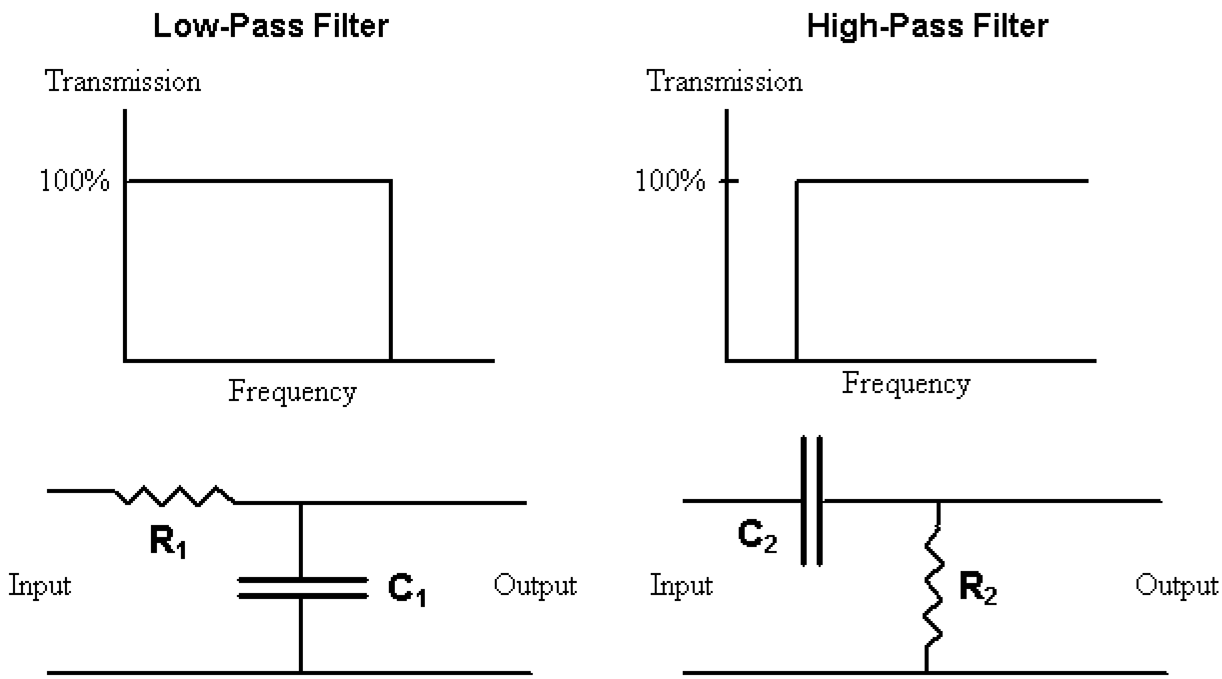
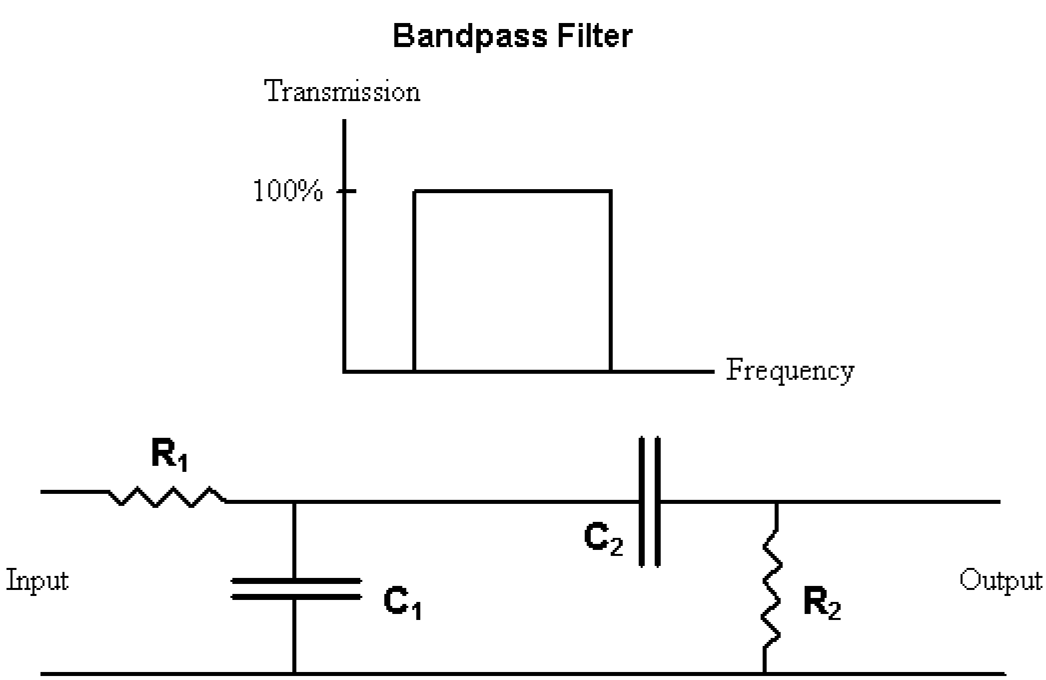


Fig. O-13. Bandpass Filter Made by Combining Above Filters.



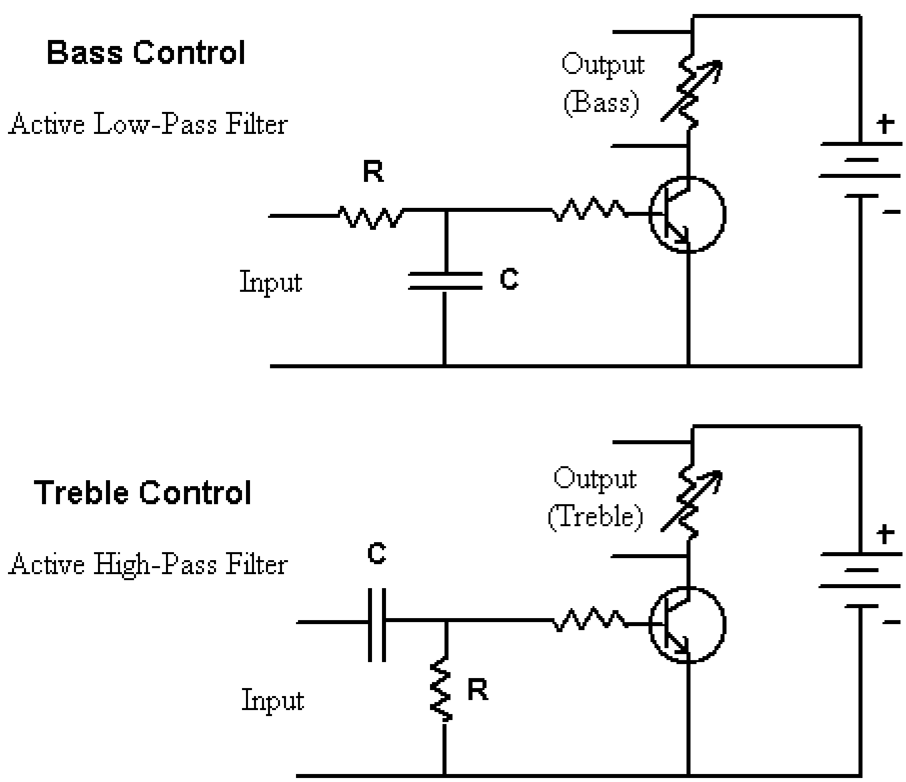
Equalizers

Equalizers are filter circuits that enable us to achieve a more "equal" sound in our homes or elsewhere. The equalizer can help to correct for deficiencies in the original sound or correct for less-than-ideal room acoustics in our homes or cars. The simplest filter circuits that offer some control are the bass and treble controls. The *bass* control enables us to modify the lower-frequency half of the audio spectrum while the *treble* control allows us to adjust the higher-frequency half.

A bass control and treble control are found in Fig. O-14. The same input audio signal is sent through each of these filters. The output of each filter is sent to an amplifier to boost the filtered output. This makes the filter an active filter. The RC-circuit alone is a passive filter, one that filters without any subsequent amplification as part of the filtering process. Active filters amplify the filtered output as this is more desirable. It gives us control of the volume (loudness).

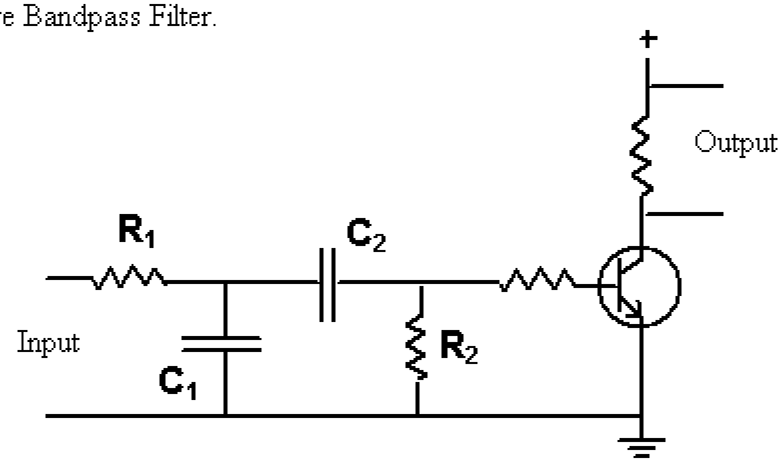
The amplifier unit has a control that adjusts the volume overall. Amplifiers usually have bass and treble controls. These respectively provide control over the lower and higher frequency regions of the spectrum. We can boost the bass or treble independently.

Fig. O-14. Bass and Treble Controls.



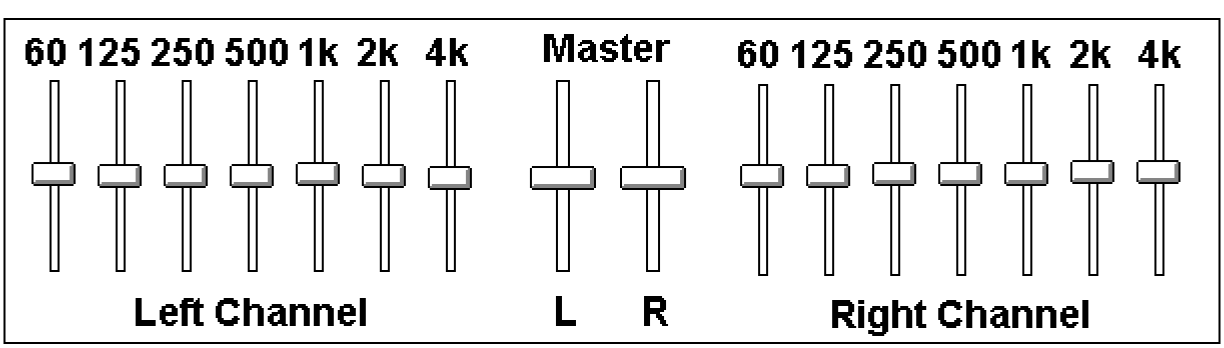
The bass and treble controls let us adjust the lower and upper parts of the audio spectrum. Think of the range of audio frequencies split into two parts, the lower from 20 - 500 Hz, and the upper from 500 - 20,000 Hz. Now break the audio spectrum into many small bandwidths and let an active bandpass filter control each section. We then have far more control of the sound. An active bandpass filter is given in Fig. O-15. Note that a power source (battery) is indicated by the "+" symbol and ground.

Fig. O-15. Active Bandpass Filter.



An equalizer is an audio component that gives control over more than two regions of the spectrum. Active bandpass filters are employed to boost specific frequency bands. A seven-band equalizer is illustrated in Fig. O-16. The audio spectrum is broken up into 7 regions. A better equalizer is one that breaks the spectrum into 12 regions or more. Note how the central frequency for each region progresses by doubling the previous frequency. We can adjust the amplified levels of each of these bandwidths by sliding levers up and down. Adjustments can be different for left and right channels. Specific settings depend on the music played, room acoustics, and our personal preferences. Some equalizers include a reverb control.

Fig. O-16. Seven-Band Stereo Equalizer.



Dolby

We consider a special kind of signal processing now, one aimed mainly at reducing noise on playing tapes. Noise is heard on playing a blank tape. The noise is particularly dominant above 5 kHz. This is due to room temperature, which gives energy to the magnetic dipoles ("baby magnets") on the tape. They do not align correctly due to this energy.

Fig. O-17. Noise on Tape Playback.

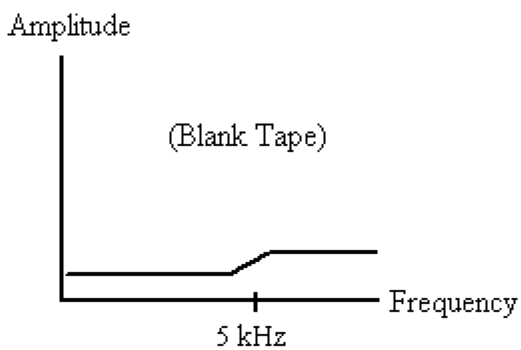


Fig. O-17 indicates the amplitude of noise on playback. *Noise* is the term used when virtually all frequencies are present. Make a "shhh" sound. This is noise. A release of steam is another example. In Fig. O-17 we find some presence of all frequencies. However, there is a greater amount for frequencies over 5 kHz. Due to the larger amount of high frequencies, the tape noise sounds more like a "hiss." In fact, it is called *tape hiss*. The dipoles do not truly get randomized when we erase the tape. Similarly, they do not align 100% correctly when we record.

You can cool the system since thermal agitation due to heat is the culprit. But you probably do not want to get into cryogenics. Temperatures have to get really low. Dolby thought of a very simple way to approach the problem in the 1960s. The first method, called *Dolby-A*, was developed for professional use. A commercial version for the average consumer, *Dolby-B*, became widely available in the 1970s. The basic idea is to simply use an active filter to boost the high frequencies when you are recording, then filter them on playback. Remember, hiss is relevant only on playback. You can't tape hiss. So boosting on taping just raises the level of the music at high frequencies. Then on playback, use a filter that reverses your boost at high frequencies to bring the music back to where it should be.

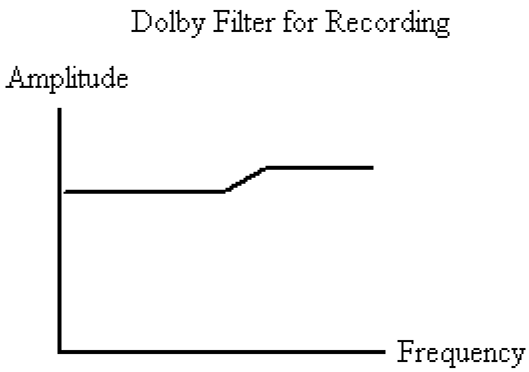
The key here is that you are lowering all high frequencies on playback. This includes the playback hiss. The high-frequency noise is lowered along with the taped high pitches on playback. The high pitches move down to levels where they should be and the hiss moves down also. The result is less hiss.

The Dolby processing circuit for playback matches in reverse what the recording circuit does. You boost high frequency on taping and then you lower high frequency on playback. In order to use Dolby during recording, you must tape with Dolby activated on your tape recorder. There is a switch to set Dolby to on. Then you must play back the tape with Dolby on.

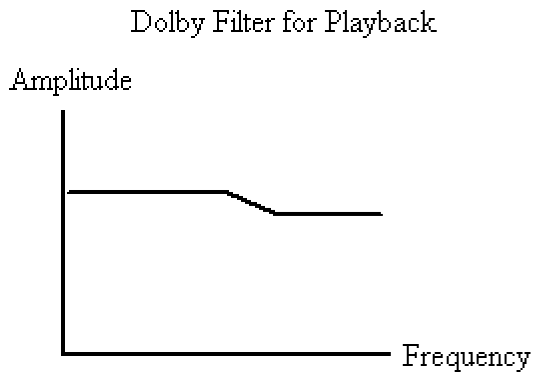
Playback with Dolby on activates the playback circuit which applies the proper correction for the high frequencies. Leave the Dolby switch to the on position and automatically the right Dolby processing circuit is used. Some people like playing Dolby tapes with Dolby off. You then have the hiss but the high frequencies are enhanced from the original taping. The enhancement helps mask the hiss (cover it up). However, you are then hearing the high frequencies louder than that intended by the musicians.

Fig. O-18 puts into graphical form many of things we have been discussing. Dolby boosts high frequency on recording and compensates for this on playback. Note how the music and noise get equalized in the final diagram which indicates what we hear.

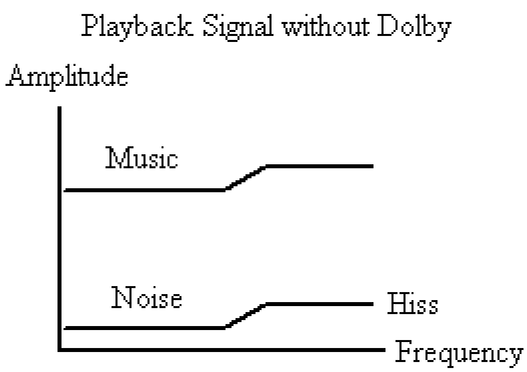
Fig. O-18. Dolby Filters for Recording and Playback.

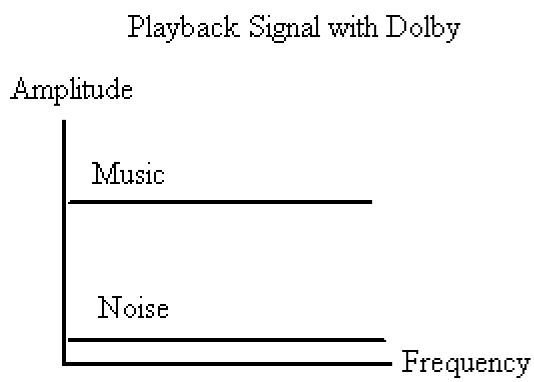


Playback tape noise is not relevant during recording. So we boost the high end in preparation for suppressing high frequencies later. Lowering the intensity of the high end to the proper level during playback will drive the noise that arises during playback down.



During playback, the high frequencies are suppressed somewhat. This is okay since the high frequencies were boosted during taping. The noise that appears on playback gets "wasted" as the high frequencies (music) are reduced to their proper levels (see immediately below).





In the diagram immediately above at the left, the tape was made with Dolby but is incorrectly being played back without the Dolby playback filter. High frequencies were taped higher so they are louder on playback without the Dolby filter. Note that noise now creeps in during the playback, especially over 5 kHz (the high end). With proper playback (diagram to the right), the hiss is lowered.

There are a few Dolby Noise Reduction Systems. The common commercial form of Dolby is Dolby B. Dolby A is a more elaborate form, used in professional recording. In Dolby B, the high-frequency end above 5 kHz is enhanced ten-fold. On playback, the high frequencies are suppressed ten-fold to compensate. This takes the noise down to one-tenth its usual value. We refer to such a drop as 10 decibels (10 dB).

Dolby proposed in 1983 using two Dolby B systems back to back to get a hundred-fold suppression of noise. On the decibel scale which we will learn about later, a hundred-fold drop is a 20 dB decrease.

Back in the old days when such sophisticated processing circuits didn't exist, faster tape speeds had to be used. Cassettes play at 1 and 7/8 ips (inches per second). There is much hiss at this speed. Tape decks in the 1950s typically had the higher speeds: 3 and 3/4 ips (twice the cassette speed) and 7 and 1/2 ips (four times the cassette speed). Playing tape hiss twice as fast doubles the frequency (pushes it up an octave). So 5 kHz gets pushed to 10 kHz by doubling the speed. Double again and you get 20 kHz. Fine recordings could be made in the early days with tape speeds at 7 and 1/2 ips. However, you need much tape, a big reel compared to a small cassette.

If you suggested taping music at cassette speed in 1960, people would think you were joking. One would only record speech at such a low speed. In fact, the professionals used 15 ips for higher-quality musical recordings. However, 25 years later, the improvements in signal processing made it possible to have excellent recordings at cassette speeds. You need little tape because it moves across the tape head so slowly. Now, the *Rachmaninoff Third Piano Concerto* (40 minutes) needs 400 ft instead of 1600 ft of tape (at the 7.5 ips we discussed earlier). Therefore, the cassette can be small. However, cassettes are no longer as popular as they once were due to compact discs.

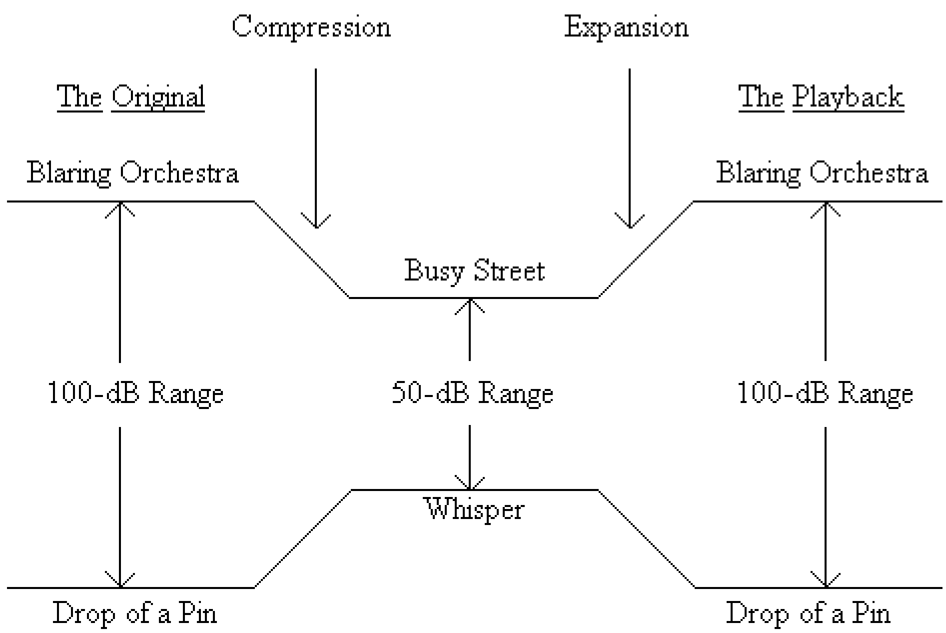
The dbx Compander

Tape is not able to handle the range of loudness levels we find in real life. From an extremely soft violin playing solo (barely audible) to the blare of the full orchestra, the decibel change is about 100 dB. A tape can't record this difference. The tape just gets saturated. Tapes can successfully record a range of 50 dB. The decibel scale is tricky. This doesn't mean half the orchestra. Think of a range of 50 dB as going from a whisper to a busy street.

There is an ingenious way to capture a sound as quiet as a drop of a pin and also the full orchestra. A processing circuit compresses the 100 dB range to 50 dB as the sound is recorded on the tape. The tape can faithfully store the compressed range of 50 dB. The various loudness levels are "pushed together" for recording. Then on playback, another processing circuit expands the signal from the tape. Fig. O-19 illustrates the idea. The original sound is compressed and the compressed version is recorded. The recording is expanded on playback to recover the full range of the original sound.

The system described in Fig. O-19 is called the *dbx* system. It does wonders for curing tape hiss. The tape hiss appears at the "whisper level" before expansion. When tape hiss appears on a blank tape without dbx or Dolby, you hear it like a whisper at normal volume. Expansion takes the "whisper" down to the level of a "drop of a pin." The hiss is essentially gone! Can you hear the drop of a pin with almost any other sound going on?

Fig. O-19. The dbx System.

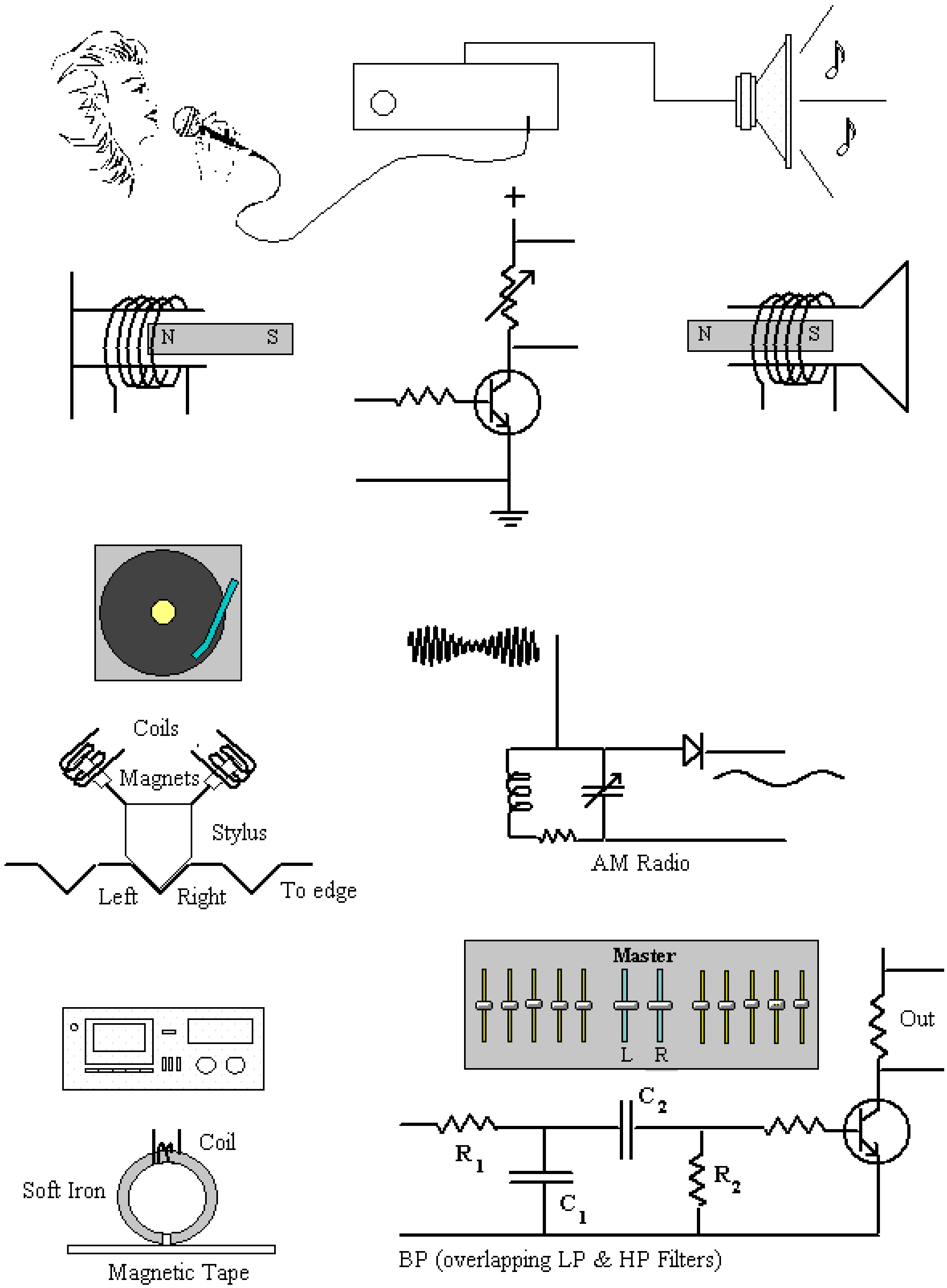


Like Dolby, dbx requires the dbx switch to be on during recording and during playback. It is interesting to take a regular recording without dbx and try playing it back with dbx. The soft sounds are made too soft and the loud sounds too loud.

When the music goes from soft to loud, this change is expanded by the dbx processing circuit. It sounds as if the music is moving quickly right at you due to the sudden increase in sound. Then the reverse is perceived when the music drops. It appears as if the source of the music quickly retreats to a great distance.

Finally, Fig. O-20 summarizes most of the sound components we have covered in this text. Connect the output of the microphone schematic to the input of the amplifier circuit. Then connect the amplifier output to the speaker. Filter abbreviations are used: LP for low-pass, BP for bandpass, and HP for high-pass. Refer to the appropriate places in the text for detailed drawings and explanations.

Fig. O-20. Summary of Basic Sound Components.



--- End of Chapter O ---

P. Moog Synthesizer I

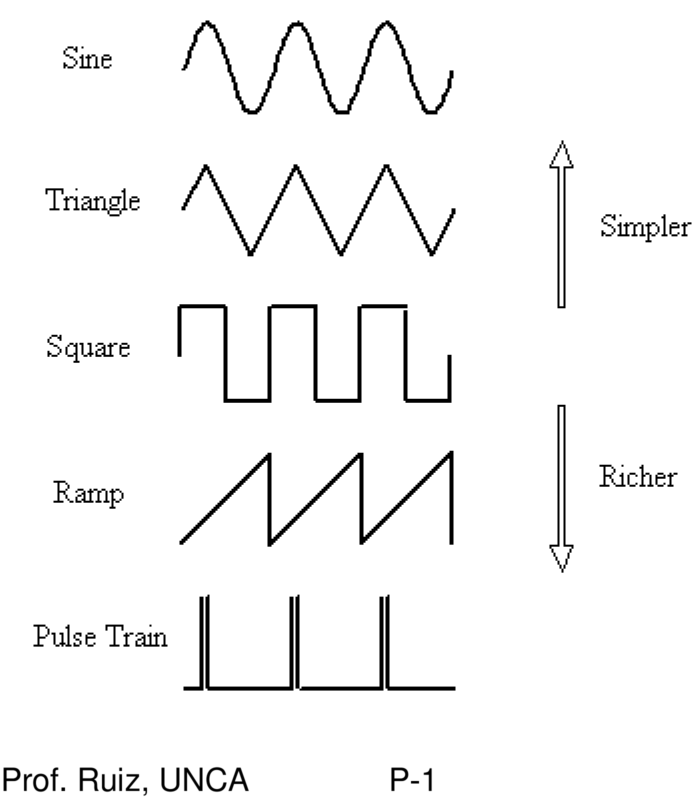
The music synthesizer was invented in the early 1960s by Robert Moog. Moog came to live in Leicester, near Asheville, in 1978 (the same year the author started teaching at UNCA). Although, there were examples of electronic music before 1960, the key discovery of Moog was *voltage control*. Moog used voltage to control sound characteristics. Consider loudness. You can turn a volume control up with your hand. This may take a half of a second. However, if the volume is turned up electronically, it can be up in a thousandth of a second. The result is a plucking sound. The use of electronic modules dedicated to different tasks along with voltage control brought about the music-synthesizer revolution of the recent generation.

We take a modular approach in covering the synthesizer. Each section introduces a new module. In this text, symbols are taken from standard electrical engineering convention. As each new module is introduced, we combine it with the other modules covered before it. In this way, the architecture of the synthesizer unfolds before us in an elegant and satisfying manner. Discussion focuses on the early basic modular synthesizer, where you have to patch all the connections yourself. This provides us with an excellent foundation.

Voltage-Controlled Oscillator (VCO)

An oscillator produces a basic waveform that can later be modified by other modules. Five fundamental waveforms produced by electronic circuits are given in Fig. P-1. We have encountered them before. These waveforms range from simple sounds to rich ones.

Fig. P-1. Five Basic Oscillator Waveforms.



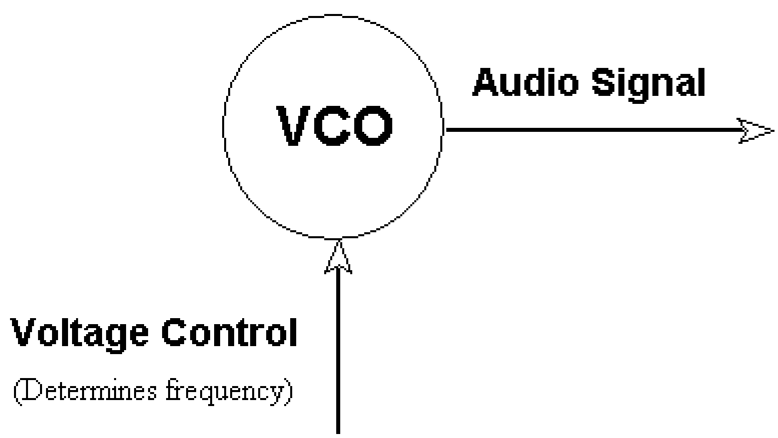
The circuit symbol for an oscillator is a circle. We use the circle in this text to represent the VCO (see Fig. P-2). Another convention in this text is to let audio signals travel from left to right and control signals enter the modules they control from below the module. In Fig. P-2, the voltage entering from below determines the frequency of the audio signal. The voltage controls the oscillator (VCO). For example, the control voltage usually ranges from 0 to 10 volts. Low-frequency audio signals with frequencies like 60 Hz may require 1 volt, while a 16 kHz-tone may need 9 volts.

The audio signal must be sent to a standard amplifier/speaker unit to be heard. This last step is omitted in our synthesizer diagrams. It is assumed that this final connection is provided. Note the simplicity in function of the VCO. Its task is very specialized and simple.

You can assume that the choice of different waveforms are made by a switch. In the early modules you actually insert a wire, called a patch, into the desired jack on the VCO. If you are communicating by a synthesizer "formula" for a friend to reproduce your sound, you can include the waveform instructions on the diagram. You may simply sketch the waveform inside the circle. You can do this underneath the letters "VCO" or sketch it in place of the letters. The circle indicates oscillator and the waveform inscription inside the circle reinforces the idea that the circle is an oscillator.

The modular diagram can serve as a way to communicate to others interesting sounds you have synthesized. The complete diagram or recipe is also called a *patch*. Today, with so many preset patches, we do not have much need for recipes. A modern synthesizer often has over 100 preset sounds called *voices*, more than enough for the average user. Many users do not want to "discover" or "invent" their own sounds, but rather prefer to use those offered by the manufacturer. Different makes of synthesizers vary in how much the user can do in the way of making new voices.

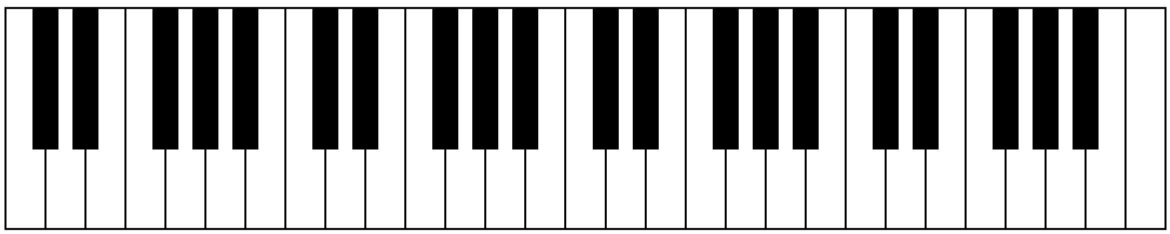
Fig. P-2. The Symbol for the VCO.



Keyboard (KBD)

The keyboard (see Fig. P-3) is the device that usually controls the oscillator. However, it can be a guitar or other instrument. Due to the popularity of the keyboard, many musicians who play other instruments have some knowledge of the keyboard. The piano has always been a fundamental instrument for musicians as it gives an easy visual representation of harmony, it offers accompaniment capability for singers, and it is readily available.

Fig. P-3. Keyboard.



The symbol for the keyboard is given in Fig. P-4. We abbreviate the names for all modules, usually with three letters. A rectangle is the symbol we choose for a control module. When a key is pressed, two things happen. First, the control voltage assumes a voltage value depending on which key is pressed. This voltage enables the VCO to sound the right pitch. The control voltage continues after the key is released. This feature is called *sample and hold*. Remember that the task assigned for each module is kept as simple as possible.

Second, the trigger voltage is activated when a key is pressed. It is deactivated when the key is released. We will see later how to use this. Think of it as a binary on-off or "yes-no" voltage ("yes" if any key is held down, "no" if all keys are up).

Fig. P-4. The Symbol for the Keyboard (KBD).

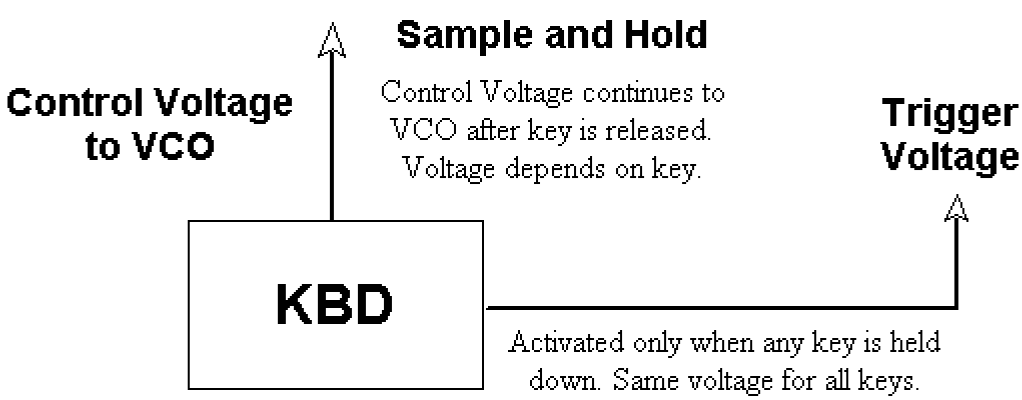


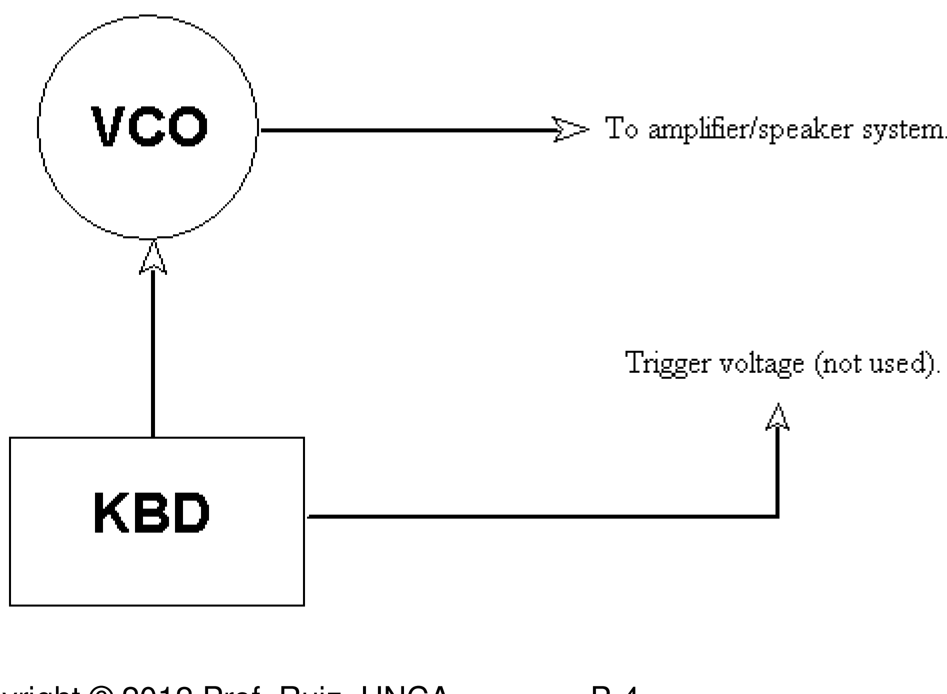
Fig. P-5 illustrates the keyboard (KBD) working together with the voltage-controlled oscillator (VCO). The keyboard control voltage tells the oscillator which frequency it should oscillate at. The waveform is chosen manually by you. You flip a switch at the VCO or plug the audio cable into the desired opening on the VCO for the waveform of your choice. The trigger voltage is not used in Fig. P-5. The arrangement in Fig. P-5 produces a tone when a note on the keyboard is pressed. When the key is released, the tone is still made by the VCO due to the sample-and-hold feature.

Note that you can only use one note at a time. You can't play two notes at once. The reason is simple. There is only one oscillator, i.e., one VCO. Additional oscillators are needed if you want to play more than one note simultaneously. We will stick with one VCO in our diagrams so that we can master all the fundamental features of the modular synthesizer.

Playing more than one note simultaneously is called *polyphony*. A polyphonic synthesizer has several oscillators in order to do this. The common synthesizer today offers at least 8-note polyphony. This is usually enough for a performer. With two hands you can hit 10 notes at one time and exceed this number by letting the thumb handle 2 notes. But this is not typical. For synthesizers with computer capability (MIDI), the voices are independent and the computer can play different instrument voices at the same time.

Composers may need more independent voices such as 16. Many MIDI synthesizers are 28-note polyphonic. The popular Roland XP-50 workstation has 64-voice polyphony. However, some sounds require the synthesizer to use more than one of its voices to produce it. For all practical purposes, this reduces the actual note polyphony. You need to consult the manual to see how many voices are needed to produce a sound of your choice. This can amount to a significant reduction on some synthesizers. Check the manual and listen to the sounds before purchasing an expensive unit.

Fig. P-5. KBD Controlling VCO.



Voltage-Controlled Amplifier (VCA)

The voltage-controlled amplifier is an amplifier that is controlled by voltage. Instead of turning it on with your hands, you can connect a 1.5-volt battery to it. More voltage means more amplification. Two batteries in series (one on top of the other) supplies 3 volts and the sound is louder. But of course, we don't use batteries. There is another module (to be taken up in the next section) that is used to control the VCA. Remember that each module specializes in a small task.

Fig. P-6 illustrates our symbol for the VCA. We choose a triangle since the triangle is the standard symbol for an amplifier in electrical engineering. An audio signal is sent into the amplifier and a modified signal is sent out. The amplitude of the signal is modified by the input control voltage. Remember our convention that audio signals progress from left to right and control signals are drawn controlling from below. The control voltage has values in the range of 0 to 10 volts like the VCO. Control signals can be steady or changing. The audio signals leaving the VCO are always periodic waveforms with frequencies in the range of human hearing. The VCO electrical oscillations are therefore always alternating in current (AC).

The voltage amplitudes for the AC audio signals are low, in the millivolt (mV) range like typical sources. The VCA controls the amplitude to achieve many interesting electronic effects rather than boosting it in the usual fashion to drive a speaker. You still need the regular amplifier. Once again, we do not show the last two stages in sound systems: the amplifier (2nd stage) and the speaker stage (3rd stage). All of our synthesizer diagrams depict the source stage (1st stage). Other sources we have studied include the microphone, CD player, and radio. Your standard stereo amplifier (receiver) has auxiliary inputs to handle an additional source such as a synthesizer.

Fig. P-6. The Symbol for the VCA.

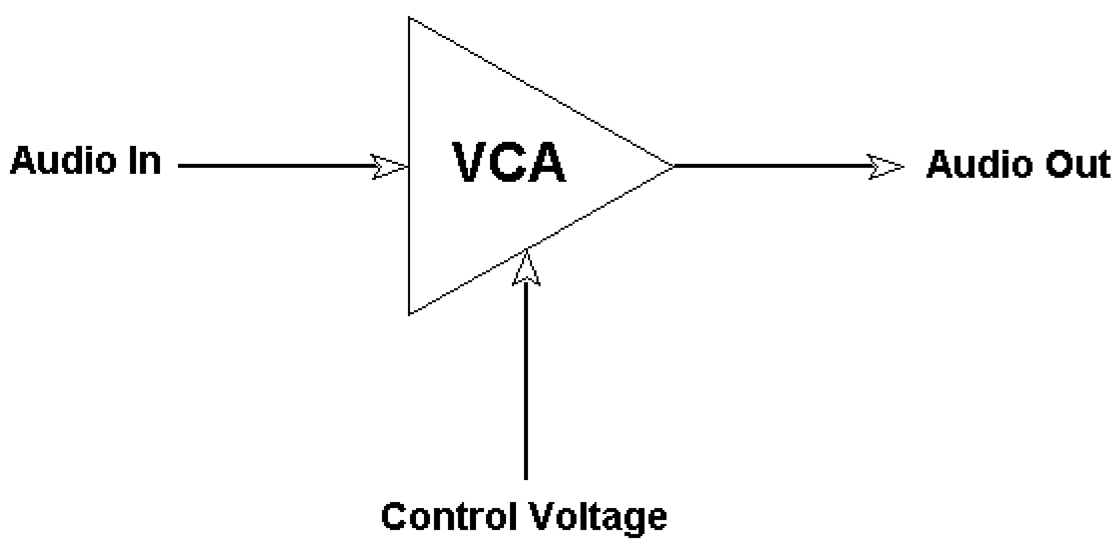
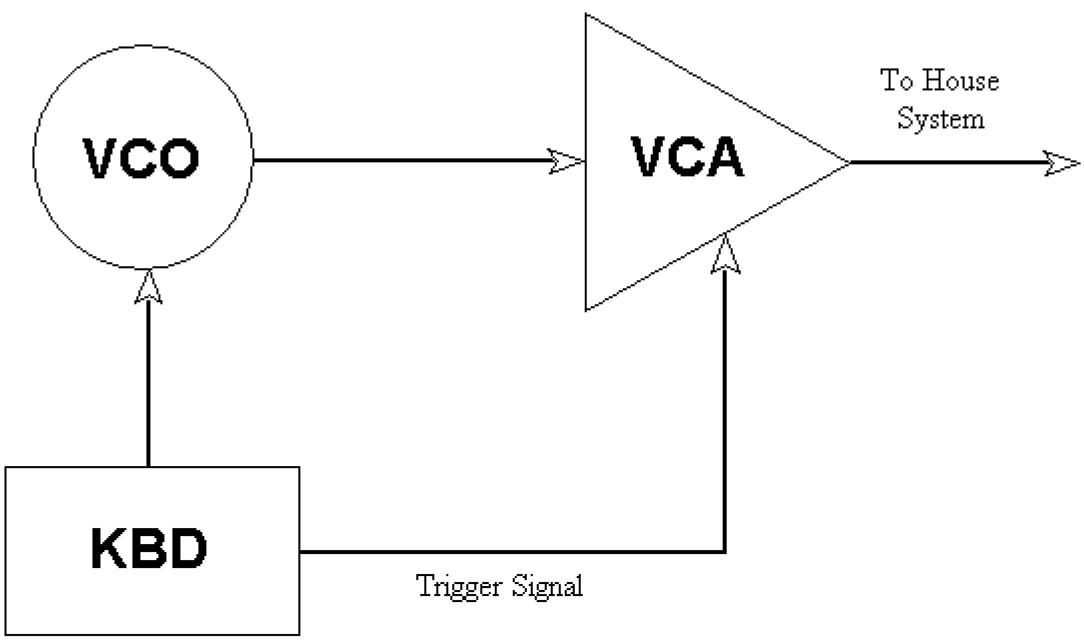


Fig. P-7 illustrates the simplest way to combine the VCO, KBD, and VCA. The keyboard controls the oscillator as before. However, now we take the trigger voltage and apply it to the amplifier. Remember that the trigger voltage is either on or off, depending on whether a key is held down or not. The signal is the same for all keys. Pressing any key results in the same "on-voltage" in the trigger line.

Pressing a key then does two things. The keyboard sends a control voltage to the VCO to tell it which frequency to oscillate at. This voltage depends on the key. A key to the left on the keyboard sends a low voltage to the oscillator and the oscillator produces a bass tone. Keys to the right on the keyboard send higher voltages, resulting in treble tones. The keyboard sends out a second control voltage, this one to the VCA. This voltage is the same for any key. When a key is pressed, the trigger voltage informs the VCA, which then amplifies the incoming signal. When the key is released, the amplifier cuts off the incoming signal.

When the amplifier cuts the incoming signal off from going to the output, the input signal is still there trying to get through. This is a result of the sample-and-hold feature of the keyboard. It is important because the VCA may want to do more things to the incoming wave. In our simple case it cannot. But if we have a more elaborate control voltage entering the VCA than the simple trigger voltage, we obtain more interesting sound effects. We take this up in the next section. For now, our control is a simple on-off. We hear an abrupt start when a key is pressed. The tone is steady when the key is held down. When released, the tone ceases just as abruptly as it started.

Fig. P-7. Simplest Arrangement for VCO, KBD, and VCA.



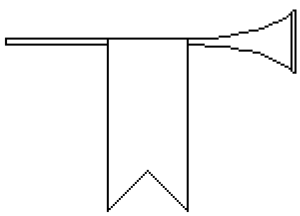
Envelope Generator (ADSR)

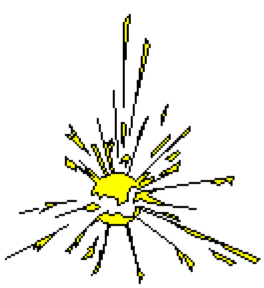
The control offered by the trigger voltage is not very satisfactory. The trigger voltage is not designed to feed directly into the VCA. Its purpose is to activate another module, the envelope generator, which in turn controls the VCA in a much more sophisticated manner. The trigger voltage turns the VCA on abruptly when a key is pressed (see previous section).

The turn-on phase of a sound is called the *attack* phase. When a key is released, the trigger turns the VCA off abruptly. This last phase of the sound is called the *release* phase. If the key is held down, there is an additional phase to the sound, the *sustain* phase. For quickly pressing and releasing a key, the trigger control on the VCA provides us with a simple analysis for the sound: an abrupt or short attack and an abrupt or short release phase. When you strike a bell, the attack phase is abrupt but the release phase of the sound is long.

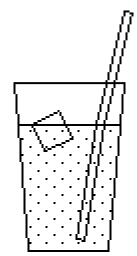
Table P-1 lists four sounds depending on whether the attack and release is either abrupt or gradual. There are four possible basic combinations. We are not considering a sustain phase at this time. The sound takes an amount of time (attack) to reach a maximum level, then immediately begins to die away (release).

Table P-1. The Attack and Release Phases of Sound.





Attack	Release	Sound
Abrupt	Abrupt	Staccato
Abrupt	Gradual	Explosive
Gradual	Abrupt	Sucking
Gradual	Gradual	Swishing



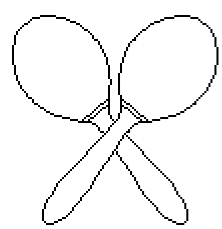


Fig. P-8 illustrates four phases for a sound. The sound begins during the attack phase. Then there is a *decay* to the sustain level. The sustain level continues until the key is released, at which time the release phase begins. This four-part method is very powerful in shaping a multitude of sounds. The wave being shaped in Fig. P-8 is a square wave. Fig. P-9 illustrates the four phases of the envelope or outline. The amplitude shaping here is a form of amplitude change or modulation, however, the change doesn't keep repeating. The square wave can be considered to be the carrier wave. The modulator wave shape is the envelope contour (Fig. P-9).

Fig. P-8. Shaping the Amplitude of a Square Wave.

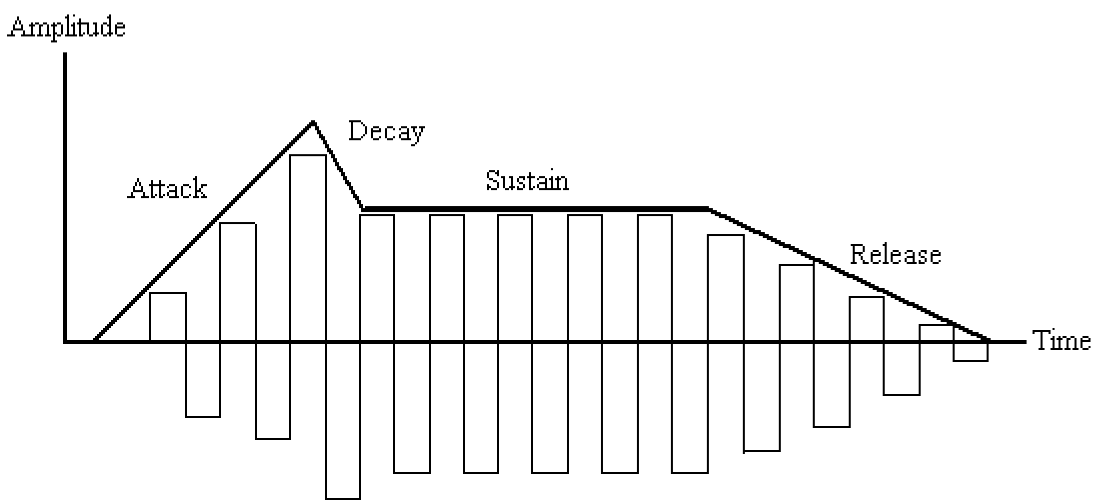


Fig. P-9. The Four Phases of the Envelope.

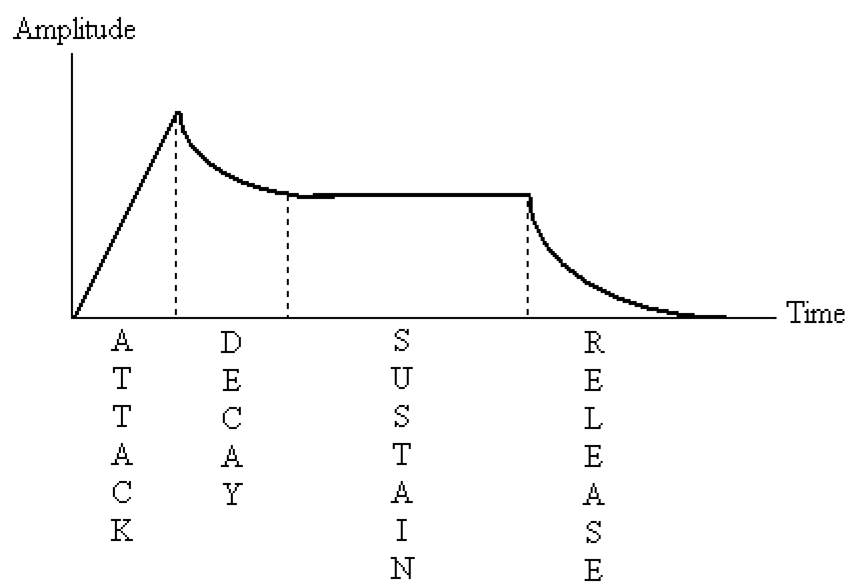


Fig. P-10 at the right gives the symbol for the envelope generator. It is a controller so a rectangle is used. The ADSR is controlled by the trigger, and the ADSR in turn controls the VCA. You press a key and the trigger "fires up." This tells the ADSR to go through a preset program. It immediately turns on the VCA for the duration that's set previously for the attack phase. Think of 4 knobs on the ADSR unit, one for each phase.

An attack setting of 1 ms gives a percussive start, while a setting of 1 s is a long beginning, like slowly playing a harmonica. The decay phase follows for the time set for it. This phase brings the sound level down to the preset sustain value. The ADSR continues sending out the sustain voltage for the sustain phase until you release the key. Upon release of the key, the trigger voltage goes off. The ADSR begins the release phase, turning the VCA off according to another preset value.

The settings (preset) for the attack, decay, and release are time values, usually between 1 ms and 1 s. The sustain setting is a voltage value. The time for this phase is determined by how long a key is held down. Fig. P-11 illustrates the VCO, KBD, VCA, and ADSR working together. A variety of musical sounds can now be synthesized by shaping the available waveforms.

Fig. P-10. Symbol for ADSR.

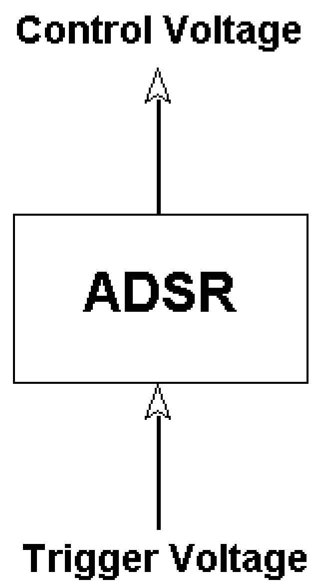
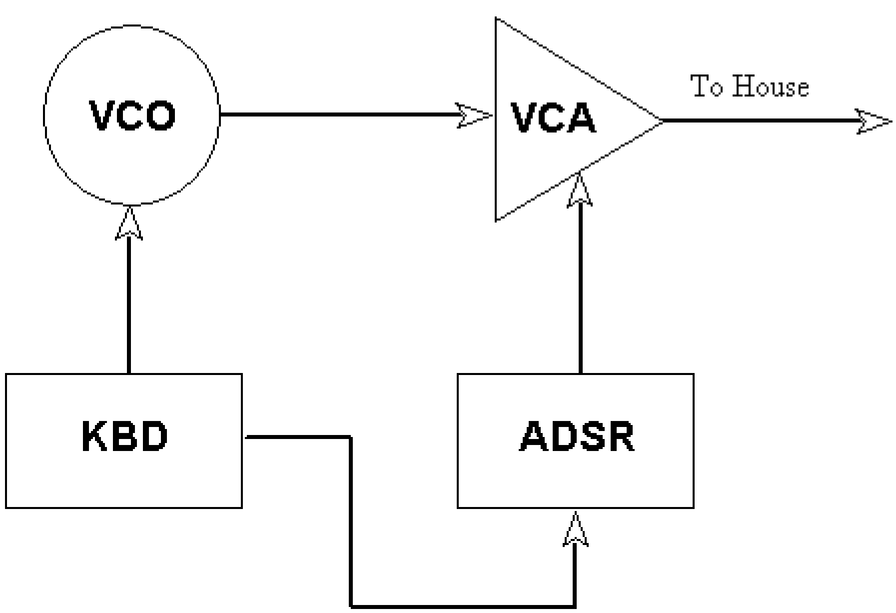


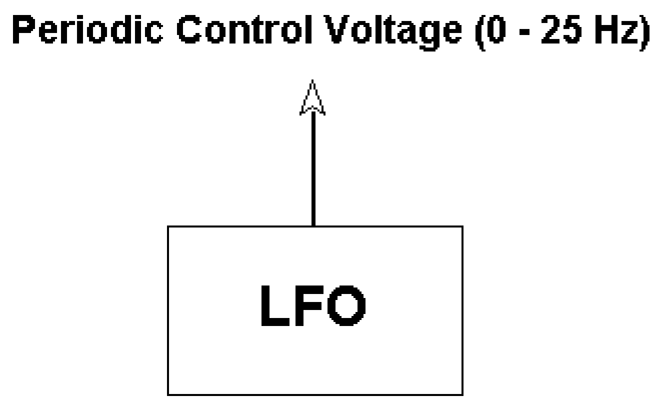
Fig. P-11. The VCO, KBD, VCA, and ADSR.



Low-Frequency Control Oscillator

The ADSR shapes the carrier waveform once for each key press. We mentioned that this change is, in a sense, a modulation. You might say that the ADSR performs an aperiodic modulation, while the more usual type of modulation is periodic modulation. We have discussed the three basic types of modulation: amplitude modulation, frequency modulation, and timbral modulation. These types of modulation can be achieved electronically. A special controller is dedicated to performing periodic changes on waveforms. This unit operates at the low frequencies, as we expect, for the usual modulation discussed in a previous chapter (0 - 25 Hz). See Fig. P-12.

Fig. P-12. Symbol for Low-Frequency Control Oscillator.



In Fig. P-13 below, the LFO is used to obtain the two familiar types of modulation, AM and FM. The LFO simply controls the appropriate module in each case. The LFO has settings for your modulating frequency and the sweep range. For example, you can sweep the VCO over small ranges for vibrato or large ones for siren effects.

Fig. P-13a. Tremolo.

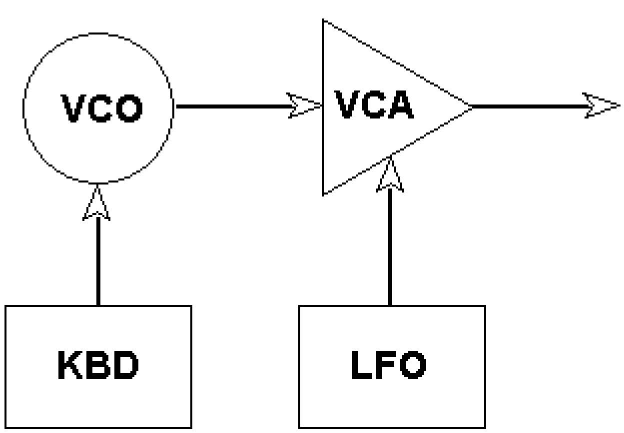
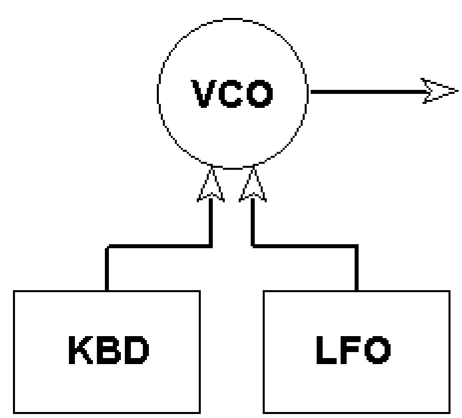


Fig. P-13b. Vibrato.



Note that two signals go into the VCO in our example for vibrato. The VCO can accommodate these signals. The voltages are added together. Then the sum controls the VCO. The control voltage from the keyboard sets the base frequency or tone. Perhaps this voltage is 3 V. The voltage from the LFO then provides an additional fluctuating voltage to it. For a vibrato, this extra periodic voltage may vary a little, e.g., fluctuating between 0 and 0.1 V. The tone raises its pitch a little and then lowers it back and so on. The rate at which this occurs is determined by the frequency setting on the LFO. For sweeping siren sounds, the LFO voltage may vary from 0 all the way up to 1 V and back down again periodically.

More practical arrangements for synthesizing a tremolo or vibrato are built from the basic arrangement of the VCO, KBD, VCA, and ADSR working together. Then the LFO is brought in to provide the appropriate modulation. These patches or arrangements are shown in Figs. P-14a and P-14b. The LFO generates its own frequency. Nothing ever is sent to the LFO. We use the rectangle for the LFO because it produces a control voltage rather than an audio signal like the VCO.

Fig. P-14a. Playing a Tune with Tremolo.

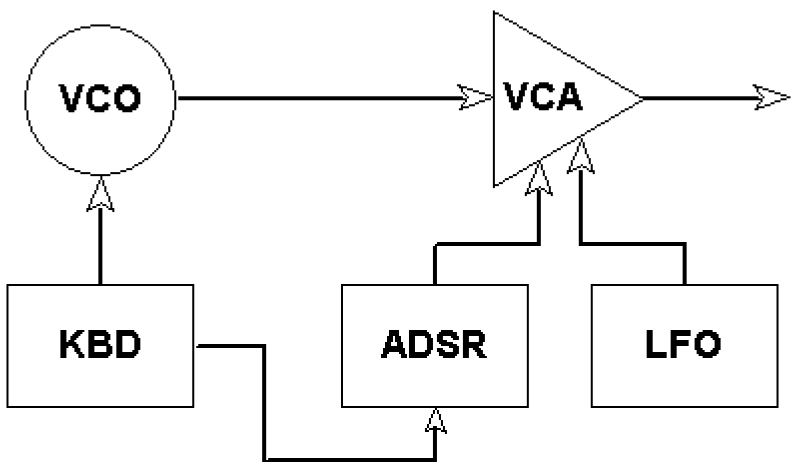
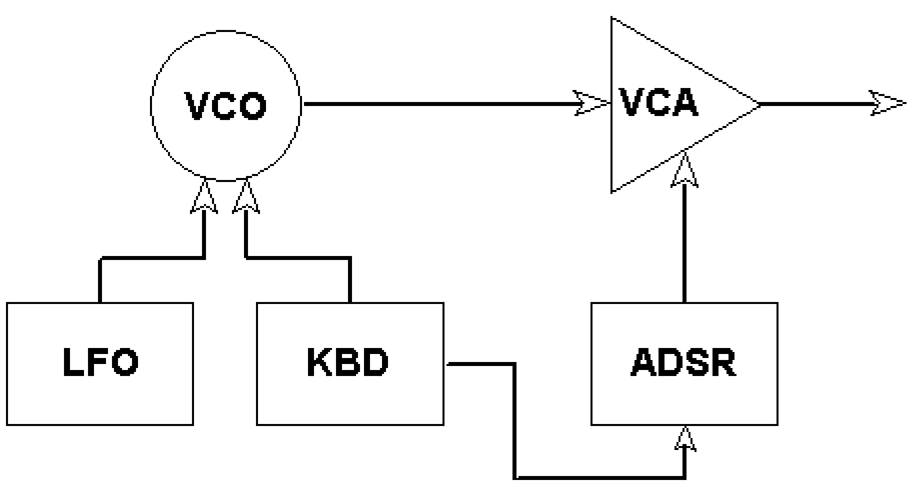
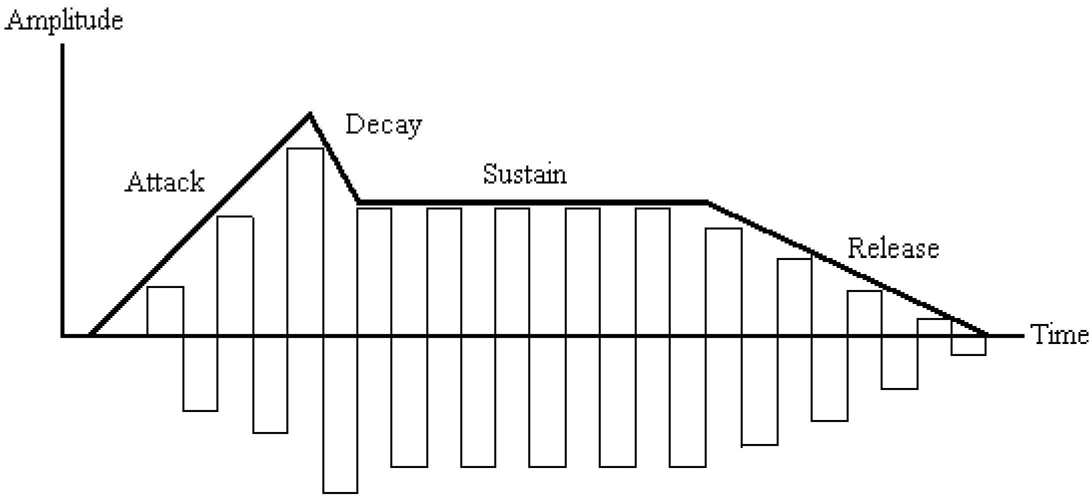


Fig. P-14b. Playing a Tune with Vibrato.



An Exercise



Description	T	F
There are four cycles of the square wave tucked inside the attack phase (see above).	◯	◯
The sustain phase always contains the greatest number of wavelengths.	◯	◯
A couple of milliseconds is an example of an abrupt attack time.	◯	◯
The sustain phase is longer than the release phase in the above diagram.	◯	◯
The release phase is longer than the decay phase in the above diagram.	◯	◯
For a real audio example, the above wavelength must be made extremely shorter.	◯	◯

--- End of Chapter P ---

Q. Moog Synthesizer II

In the previous chapter we introduced five synthesizer modules: the voltage-controlled oscillator (VCO), the keyboard (KBD), the voltage-controlled amplifier (VCA), the envelope generator (ADSR), and the low-frequency oscillator (LFO). In this chapter we introduce two more: the noise generator (N), and the voltage-controlled filter (VCF). These seven modules are the basic modules of the modular synthesizer.

Noise Generator (N)

The engineering definition of noise is a sound that has a mixture of all frequencies. When there is fairly equally-perceived distribution of all frequencies, we have *white noise*. This name comes from an analogy with light. Newton discovered that white light consists of all the colors in the visible spectrum. If you shine white light through a prism, the light breaks up into the spectrum of colors from violet to red. Violet has the shortest wavelength and highest frequency. The violet light is nearly twice the frequency of red light.

We see about an octave in color. All the colors within this octave mixed together appear white to our eyes. With sound, we hear many octaves. The piano alone has 7 octaves. We can hear a few additional octaves beyond the piano. The frequency range for hearing is often reported as 20 to 20,000 Hz. However, most people lose it somewhere between 10,000 Hz and 20,000 Hz. A fairly equal representation of sound from 100 Hz to 10,000 Hz is taken to be white noise.

An excellent example of white noise is the roar of the ocean. The turbulence of the water at the beach or shore is so varied that frequencies of sound across the audio spectrum are produced. Some claim that putting a seashell to your ear reproduces the ocean sound, having captured it mysteriously in some way. You do hear sound when you place a seashell to your ear, and it is white noise. The white noise is not due to the ocean though; it is noise resulting from random movement of air molecules due to temperature. You can get a similar effect by holding a glass to your ear or cupping your hands over an ear.

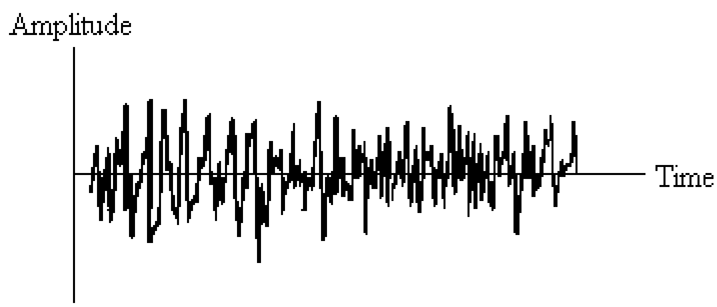
Steam is another good example of noise. You need to be careful if sounds of steam get loud because all frequencies are represented. This has the potential to damage your hearing. Technicians working near steam turbines in power plants wear ear protection. Another example of white noise is the noise a fan makes. The yells and cheers of a baseball or football game heard from afar approximate white noise. Far away you get a better mix of the sound. Can you imitate the distant sound of cheers by making sounds with your mouth?

We are more sensitive to higher pitches than low ones. The noise that sounds fairly uniform is actually somewhat lacking in lower frequencies. If we really have equal amounts (by the reading of a meter) for all frequencies, the noise sounds a little deeper. Engineers use the term *pink noise* for such noise since it appears to have a greater presence of low frequencies. Going back to the analogy with light, white light with a presence of more low-frequency red is pink.

Noise is illustrated in Fig. Q-1. This figure represents a snapshot of noise as it appears on an oscilloscope. The mixture of so many wavelengths produces a chaotic pattern that constantly shifts on the oscilloscope. The many frequencies present in noise are evident by the many different wavelengths superimposed. The low frequencies have long wavelengths, while high frequencies have short wavelengths. The combined waveform is aperiodic. The wave patterns do not repeat. Therefore, there is no defined pitch as we find in periodic waves. Fig. Q-2 below illustrates some examples of noise.

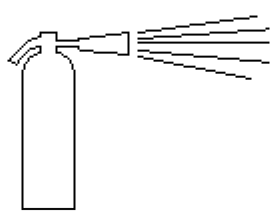
Fig. Q-1. Oscilloscope Showing Noise.

White Noise - all frequencies together.

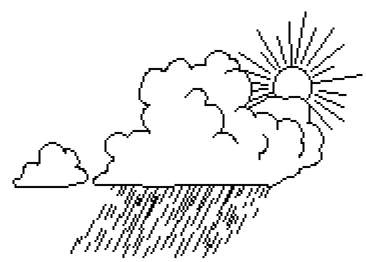


Note that the waveform is aperiodic.

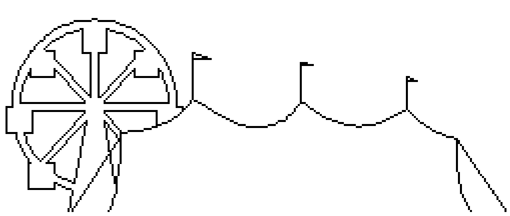
Fig. Q-2. Some Examples of Noise.



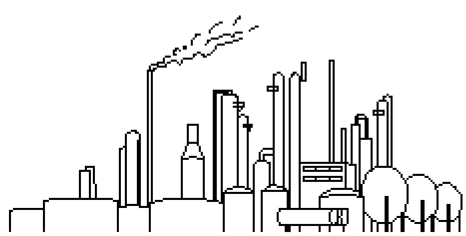
Fire Extinguisher in Action



Rain



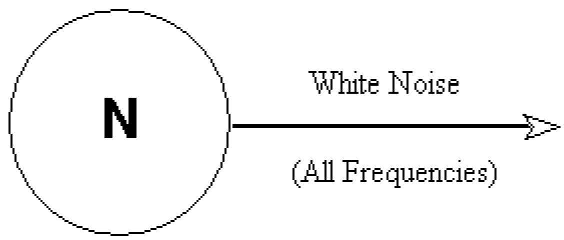
Cheering Crowd at the Circus



Machines in a Factory

Noise can be made electronically by forcing a current into a transistor the wrong way. This causes microscopic havoc. If you overdo it, you destroy the transistor. But in smaller amounts, electronic noise is produced. This is all we want for the noise source. Our noise generator does just this and nothing more. The symbol for the noise generator is given in Fig. Q-3 below.

Fig. Q-3. Symbol for Noise Generator.



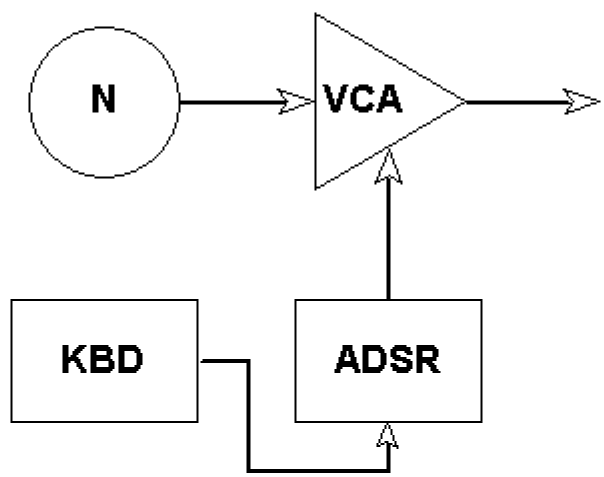
The circle is used for the noise generator because it is a module that generates audio signals. The VCO also generates audio signals. However, there are some differences. The noise generator has no input control signal. It always produces noise. The reason we need to control the VCO is that the VCO produces one frequency at a time. We need to tell the VCO which particular frequency to produce by playing a key of our choice on the keyboard. Since the noise source N produces all frequencies in a random mixture, the noise generator needs no instructions.

You might wonder why the noise source is always on. But we want this. Recall that the VCO always sends out whatever the last key instructs it to do. This is sample and hold. Likewise the noise source keeps sending out white noise. Each module is asked to do very little. We need all the modules working together to produce the final desired outcome.

See Fig. Q-4 for a simple example. It doesn't matter which key is pressed. The effect is the same. Each key produces the same trigger.

Quick attacks and gradual releases can synthesize explosions. Explosions have many random frequencies (noise). Gradual attacks and abrupt releases synthesize sucking on a straw (random turbulent motion of the liquid produces noise). It's best to just press and quickly release the key for these. If a key is held down, steady-state noise is maintained.

Fig. Q-4. Simple Use of Noise.

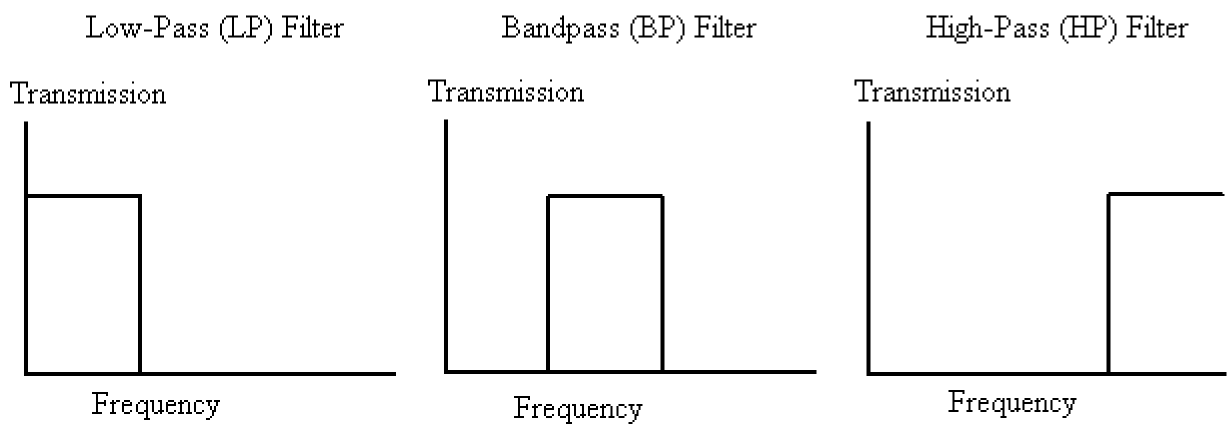


Voltage-Controlled Filter (VCF)

The last module of the basic seven units that comprise the modular synthesizer is the voltage-controlled filter. The filter allows us to modify the waveform, shape the timbre. The filter can alter the Fourier spectra of our basic periodic waveforms that originate in the VCO. The filter can also modify the sound from the noise generator. We can filter out high frequencies and approximate pink noise.

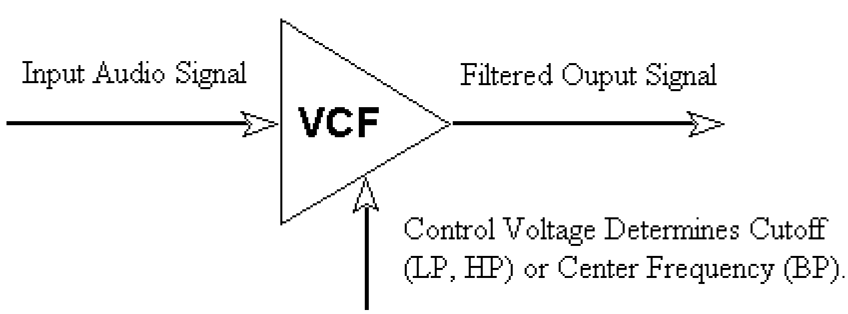
When we employ light with colored filters, we get different colors. The color we see is the color transmitted by the filter. A blue filter transmits blue light. We might say that white noise going through an electronic filter produces "colored noise." The three basic filter types are reviewed in Fig. Q-5 below.

Fig. Q-5. Review of the Three Basic Filter Types.



The filter graphs above give the transmission percentage for a sine wave at each frequency entering the filter. Periodic complex waves must be decomposed into their Fourier spectra in order to analyze which harmonics can pass through. The symbol for the VCF is given in Fig. Q-6.

Fig. Q-6. Symbol for Voltage-Controlled Filter.

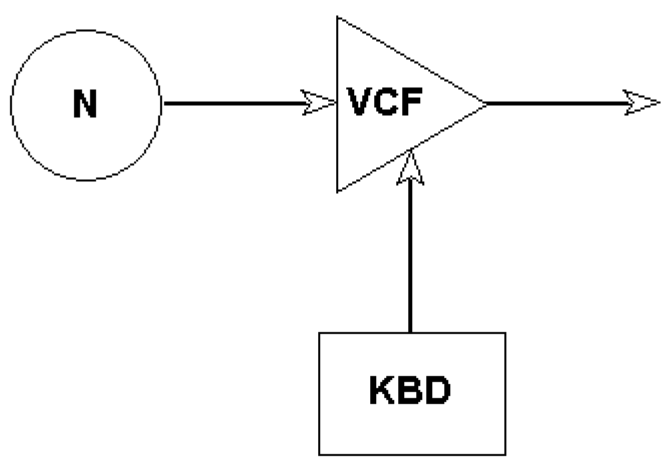


The symbol for the voltage-controlled filter (VCF) is a triangle. The triangle is the symbol used for the amplifier, which modifies the audio signal's amplitude overall. The filter works on individual spectral components. It modifies the amplitudes of the partials that make up a periodic waveform. It modifies noise, filtering out some sine-wave frequency components, passing others. Therefore, the triangle is the logical choice. Think of the triangle symbol as one that represents a device that accepts an audio signal and modifies it. The amplifier and filter do this. Active filter circuits have amplifiers incorporated in them. We can consider the VCA and VCF as members in the same family, the family of audio-modifier modules.

The voltage-controlled signal that enters the filter symbol from below fixes the cutoff frequency if the filter is low-pass (LP) or high-pass (HP). The control determines the center frequency for bandpass filters. Now we can obtain "colored noise." We choose a bandpass filter and use the keyboard to pick where the central frequency should be. See Fig. Q-7. If the central frequency is low, the filter lets a band or window of low frequencies pass. If a middle key is played, the noise band near the middle of the audio spectrum is highlighted. For the keys near the top of the keyboard, the noise band is centered on high frequencies, giving more of a "hiss."

The noise generator in Fig. Q-7 sends out "white noise" to the VCF. The VCF filters out frequency components of the noise, producing an output of "colored noise." Since we are using a bandpass filter, you may carefully pencil in BP to the lower left of the letters VCF inside the filter triangle. This reminds us to use the correct filter when we look up our "recipe" for interesting colored-noise effects in the future, i.e., Fig. Q-7. When a key is released, the control voltage corresponding to the key just released is still going to the VCF. This is due to the sample-and-hold characteristic of the KBD. Since the noise keeps sending out an audio signal, you keep hearing the filtered noise. You can use the VCA and ADSR, as we will show, to control the shape or envelope of the sound. Different settings on the ADSR then provide for a rich variety of noise effects.

Fig. Q-7. Synthesizing "Colored Noise."

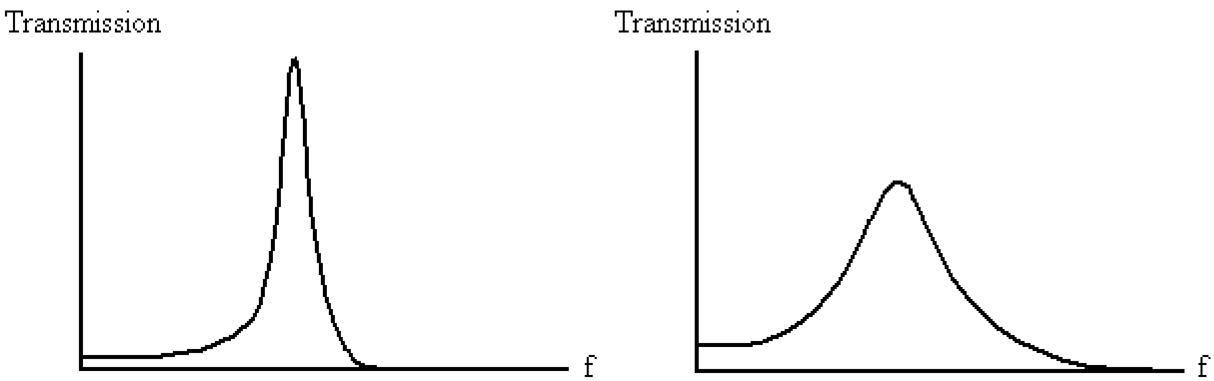


A special kind of bandpass filter can pass only a very narrow band of frequencies. This filter is called a *resonance filter*. It is essentially the resonance-tuner circuit we encountered in the AM radio; however, now we are dealing with audio frequencies. Our first encounter with resonance involved mechanical resonance very early in this text. Fig. Q-8 presents us with the familiar resonance curve again. But this time, two graphs with different features are given. Both are in the family of resonances. Specific choices for circuit elements can give the usual tall and narrow resonance-response curve or a short and wide one.

In radio electronics, a tuner with a tall and narrow response band is said to have much selectivity. The resonance circuit is very selective in its response. If the incoming frequency is not within a very narrow band of frequencies, there is very little or no response. This is desired since neighboring stations aren't picked up at the same time. Such a tuner is also more sensitive since the response at resonance is so great. This is another desired feature in tuner resonance circuits.

Resonance filters that are tall and narrow are said to have a high *Q-value* (Quality-value or Quality-factor). They are very selective in the frequencies that they pass. They also amplify these special frequencies very much. So the filters can be said to be sensitive to the frequencies near the resonance frequency.

Fig. Q-8. Resonance Filters.



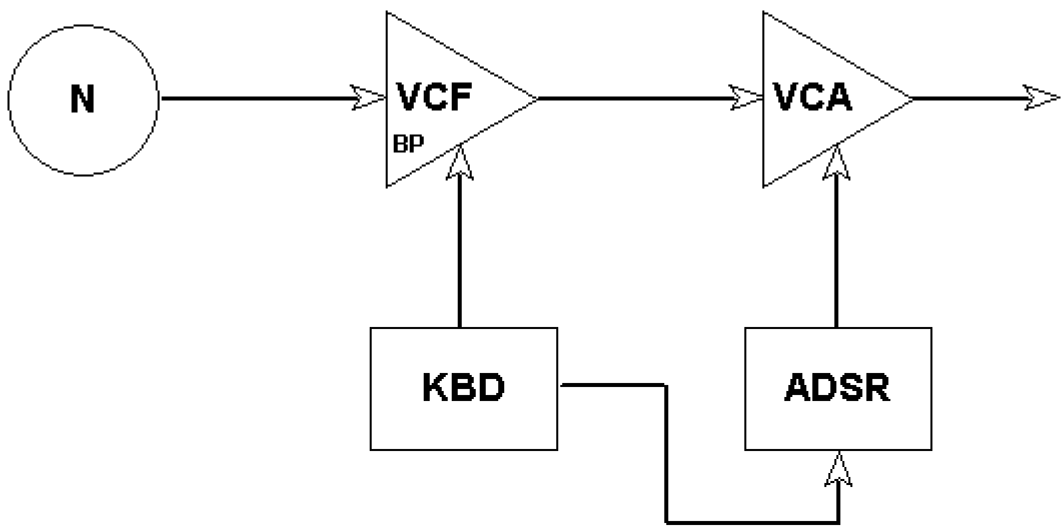
High Q-Value:
Tall and Narrow.

Low Q-Value:
Short and Wide.

A resonance filter with a high Q-value passes a narrow band of frequencies. The emphasis of such a narrow band of frequencies produces a dull tone in the noise. When you whistle, especially not so good, there is a lot of noise mixed in with the whistling tone. A group of such whistlers is even a better example. We can use a narrow bandwidth filter to synthesize a group of whistlers. Fig. Q-9 gives the arrangement. We need the VCA and ADSR now in order to cut the sound off as we release the keys. In this way, we can play a tune.

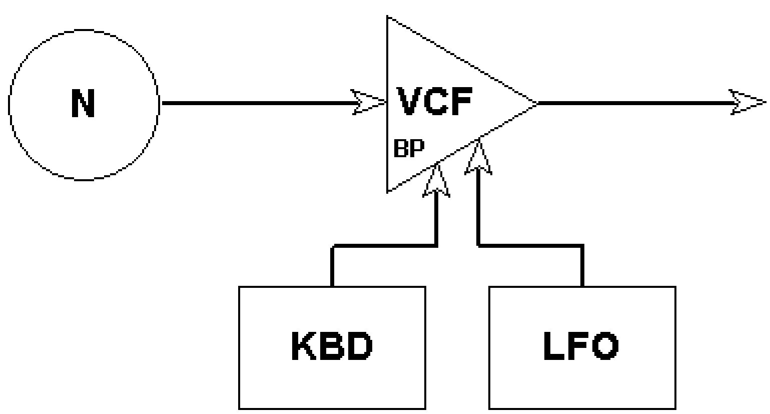
The key on the keyboard determines the central frequency of the bandpass filter in Fig. Q-9. If the bandpass filter is a resonance filter with a high Q-value, the filter can narrow the broadband white noise down to noise in the neighborhood of the resonance frequency of the filter. A tone with surrounding noise is produced. We approximate a group of whistlers, where noise comes from the air rushing in the mouth along with the whistling tone. We can synthesize a group whistling the tune *The Bridge on the River Kwai*. The VCA and ADSR supply the amplitude shaping of each whistled note.

Fig. Q-9. Synthesizing a Group of Whistlers (BP Filter with High Q-Value).



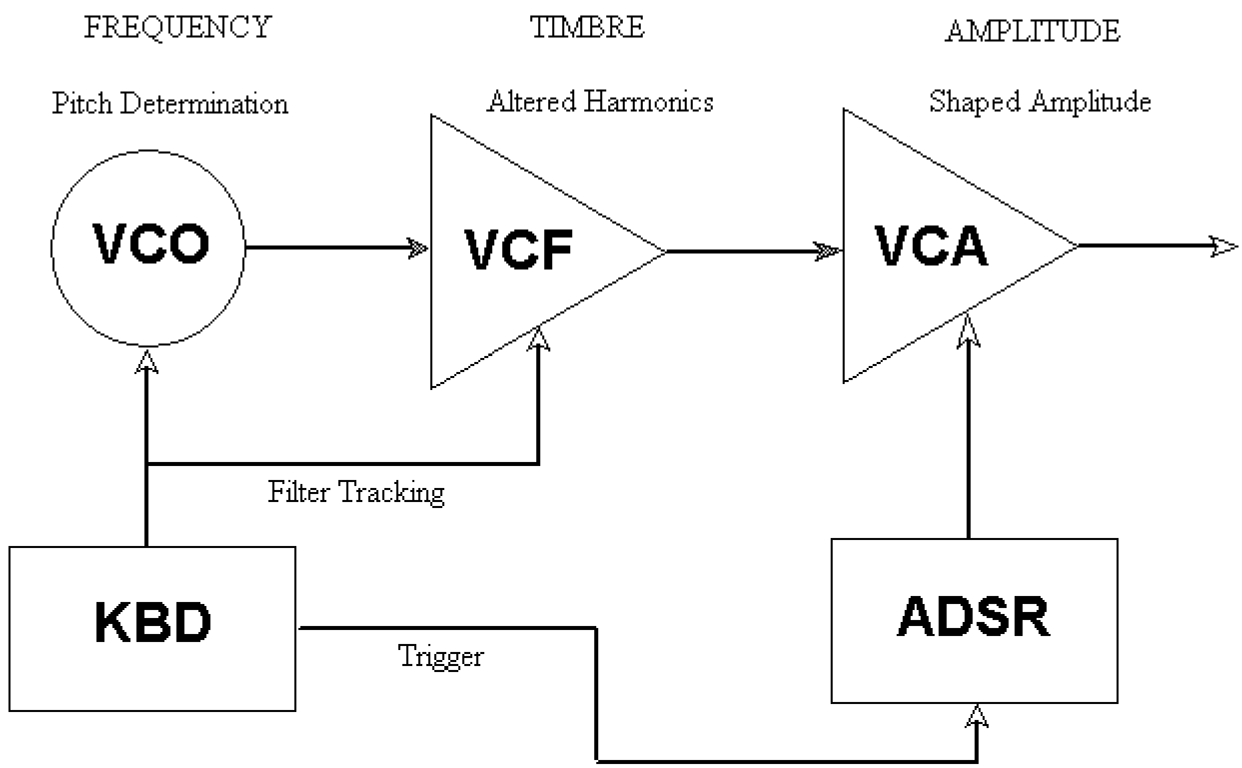
The arrangement in Fig. Q-10 below produces howling wind. It does not need a VCA and ADSR since the sound may continue on its own in this case. The KBD determines the central frequency for the bandwidth and the LFO shifts the entire bandwidth up and down, giving the howling-wind effect. Playing different keys gives variety to the sound.

Fig. Q-10. Synthesizing Howling Wind.



The most basic set of synthesizer modules for producing a musical tone consists of the following five modules: the VCO, KBD, VCA, ADSR, and VCF. The other two modules we studied, the LFO and N are used for special effects. The LFO can add tremolo or vibrato effects. Noise can be employed to synthesize explosions. The five modules needed for producing musical tones are depicted in Fig. Q-11 with their proper relationships to each other. Fig. Q-11 serves as an excellent review of modular components. It also introduces one new concept, filter tracking.

Fig. Q-11. The Standard Modular Arrangement for Producing a Musical Tone.



First we introduced the VCO and KBD. The VCO produces the audio signal with a frequency determined by the KBD. Think of these two modules working together as a pair. They address one of the three basic characteristics of a periodic tone, the frequency. The other two fundamental characteristics are the amplitude and timbre. The amplitude shaping is accomplished by the VCA and ADSR. Consider these as a pair working together. The amplitude shaping gives us the playing of a single note. Otherwise, the sound would continue indefinitely. The shaping of the third main aspect of a periodic tone, the timbre, is accomplished by the filter. However, it needs a control partner too. It uses the KBD. The KBD controls both the VCO and VCF.

There is good reason for the KBD to control both the VCO and VCF. If the cutoff frequency for the filter were to be fixed by some other control source, then some tones might be totally cut out. As you played up the scale, a low-pass filter could cut your high frequencies off. You want the filter cutoff frequency to follow you.

A specific example will illustrate this important point. Consider using the pulse train as the oscillator waveform. This is set manually before playing the synthesizer. The pulse train has a Fourier spectrum where each harmonic is present with equal amplitude. Consider also choosing a low-pass filter and an appropriate fixed-control voltage (from a power supply) so that the filter lets through the first three harmonics of the pulse train when we play *Do*. We obtain a unique sound, a pulse train with only the first three harmonics. Now if we start playing up the scale (*Do*, *Re*, *Mi*, etc.) the frequency of our pulse train gets higher and higher. It quickly gets beyond the low-pass filter cutoff and nothing comes through at the higher end of the scale. Playing up the scale and having your sound disappear is not good.

Now if the filter cutoff can move up the scale with us, then the filter can keep passing the lower harmonics. As the fundamental gets higher in frequency going up the scale, so does the cutoff frequency of the filter. To enable the filter cutoff to move with the note we play, we simply tap off the KBD control voltage and send it to the filter in addition to sending it to the VCO. The filter can now track our base frequency. With filter tracking, we preserve the filter's ability to modify our pulse train over the entire keyboard.

Even with filter tracking, the filter will not have exactly the same effect on our pulse train everywhere along the scale. Maybe an additional harmonic gets through here and there. However, the basic modification of passing lower harmonics is preserved. Timbres on real instruments change somewhat at low and high ends of the musical scale anyway. So we should not be too hard on our filter if it also varies somewhat in how it shapes the spectrum for different notes.

Note how the arrangement in Fig. Q-11 has three main vertical sections. The first, consisting of the VCO and KBD, focuses on frequency. The second region, consisting of the VCF with filter tracking made possible by voltage control from the KBD, is concerned with timbre. The third region, consisting of the VCA and ADSR, attends to the amplitude. The modification of the three essential features of periodic tones is accomplished by voltage control in each case. Note the first two letters "VC" for three of the modules. The trio of voltage-controlled modules reflects the fundamental nature of periodic-wave characteristics: frequency, timbre, and amplitude. To control these by voltage is the landmark discovery of the music synthesizer. Finally, see Figs. Q-12a and Q-12b, which introduce the LFO in order to give the synthesized musical tones a tremolo or vibrato effect.

Fig. Q-12a. Adding Tremolo to the Standard Arrangement.

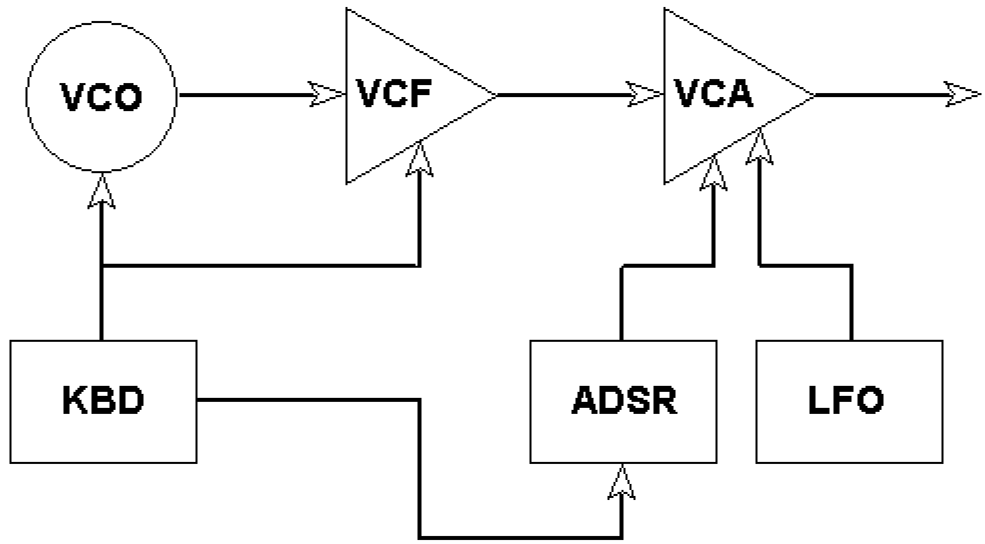
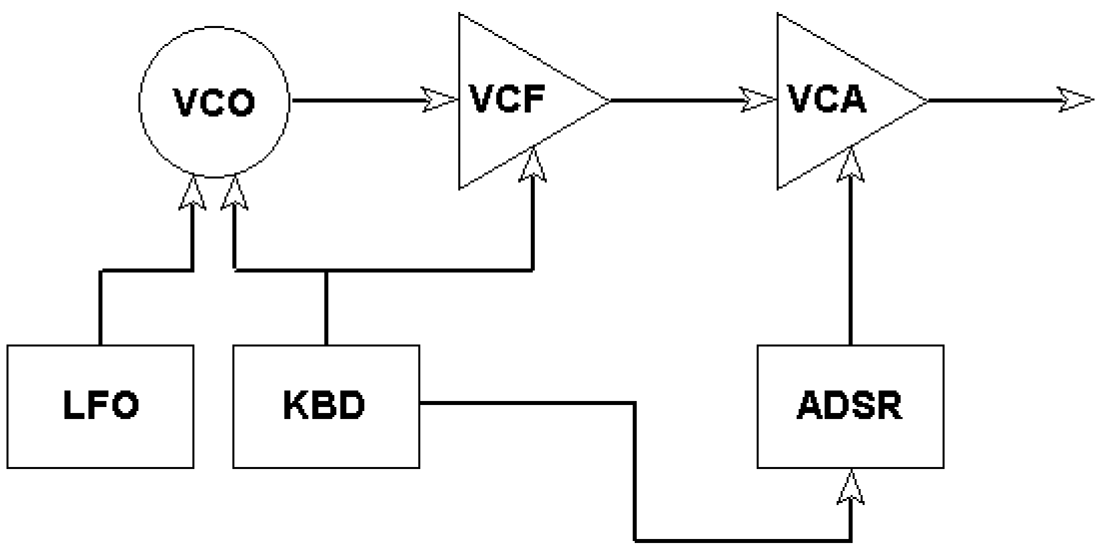


Fig. Q-12b. Adding Vibrato to the Standard Arrangement.



--- End of Chapter Q ---

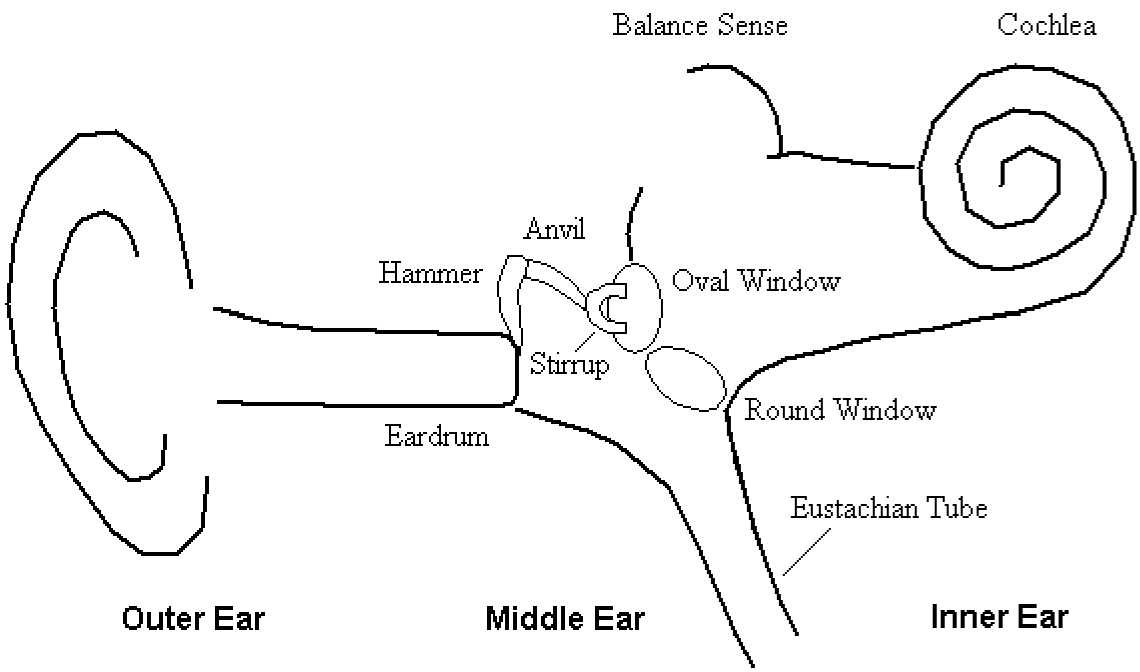
R. The Ear

We turn now to the study of the human ear. In this chapter we consider the biological components of the ear. We learn how they work in order to detect sound waves. Then, in following chapters, we look at the psychology of perception and the medical area of hearing loss, audiology.

Structure of the Ear

Fig. R-1 below illustrates the human ear. The ear is divided into three parts or sections. These are the *outer ear*, *middle ear*, and *inner ear*. The sounds enter the ear in a gas medium. We have acoustic waves in air. The waves travel to the eardrum. The eardrum is the boundary between the outer and middle ear.

Fig. R-1. Schematic of the Human Ear.



The eardrum is a membrane that vibrates in response to the incoming sound waves. It is analogous to the membrane in a microphone that oscillates in step with the sound it picks up. The eardrum transmits the vibrations to the middle ear. The middle ear consists of three tiny bones: the *hammer*, *anvil*, and *stirrup*. The vibrating eardrum vibrates the hammer. The motions of the hammer are transmitted to the anvil, and then to the stirrup. In this sequence, the bones act as a lever system. The vibrations are amplified in the process and directed to the oval window. The amplification occurs since the applied force at the oval window acts over a smaller area when compared to the area of the eardrum, the location of the input force. Muscles dampen the bones if necessary. However, sudden loud sounds may cause damage to the middle ear.

The *eustachian tube* connects the middle-ear section to the region below the back of your mouth. This is important in order to equalize pressure. When there is no sound, the pressure on either side of the eardrum should be the same. When sound compressions and rarefactions enter, the eardrum vibrates about its equilibrium position. If you go to a higher altitude, the air pressure is less. The eardrum wants to push outward due to the lower pressure outside and greater pressure inside. However, if the outside pressure can be communicated to the middle ear behind the eardrum, a new equilibrium pressure can be established.

The reverse occurs when you come down from a higher altitude to a lower one. The increased pressure at lower altitude pushes in on the eardrum since now the average pressure behind the ear is less. When the greater outside pressure is communicated to the middle ear via the eustachian tube, the equilibrium pressure is raised.

The adjustments in pressure must not occur instantly, otherwise, the eardrum wouldn't work at all. You want the rapid changes in pressure of a sound wave to vibrate the eardrum. It does this because the pressure changes rapidly on each side as the sound vibrates the eardrum. We only want to make a correction to the pressure behind the eardrum when the overall average pressure, i.e., equilibrium pressure, changes. Therefore, the response time for the pressure adjustment is greater. It may take a few seconds or longer. When the adjustment occurs, you might hear a "pop." You might describe it as your ears "popping."

If you are "stuffed up" due a cold or allergy, your ears may have trouble adjusting for equilibrium pressure changes. The build up of pressure on either side of the ear can be quite painful, especially on airplanes. The ear is challenged on airplanes. Although attempts are made to keep cabin pressure constant, there are typically changes in the pressure when raising to higher altitudes and coming down. Smaller low-flying planes do not maintain cabin pressure as well as larger commercial jets. Some people suggest swallowing to help open passageways in the eustachian tube. Others recommend chewing gum.

The author's father-in-law (*the Colonel*) was a flight instructor during World War II. His method calls for you to cover your nose openings with your thumb and forefinger, close your mouth, and try to blow outward through your covered nose. The internal build-up of pressure helps to open the pathway in the eustachian tube.

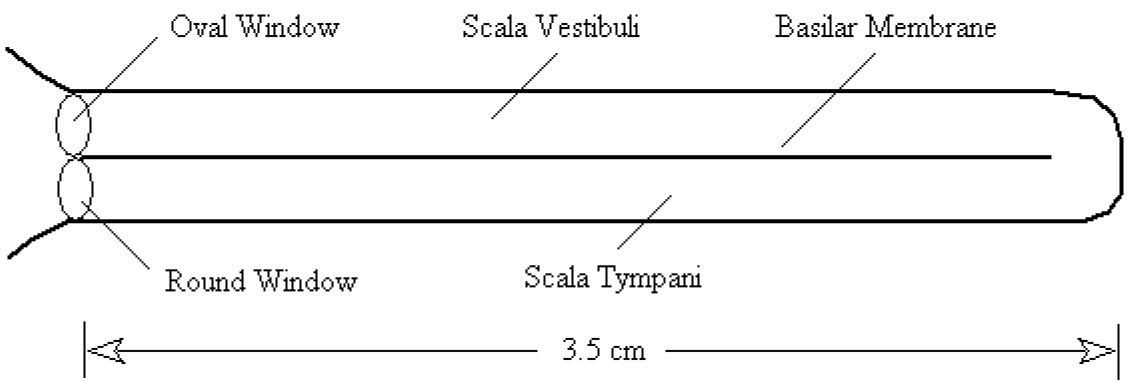
The stirrup is attached to a structure called the oval window, the boundary between the middle and inner ear. The stirrup vibrates the *oval window*. On the other side of the oval window is the inner ear with fluid. The vibrations now travel as conduction in a liquid. The pressure waves in the coiled *cochlea* (KOE-klee-uh) result in detection of sound. The inner ear also houses our sense of balance. However, the balance sense is independent of hearing. The balance structure contains semicircular canals (not shown in Fig. R-1) with another fluid which responds to our orientation. Here, fluid actually shifts around, stimulating hair cells. In the cochlea, pressure waves in the fluid stimulate hair cells.

The Inner Ear

To acquire an understanding of the cochlea of the inner ear, we will enlarge it in stages. However, first we unwrap it. The unwound cochlea is about 3.5 cm in length (one inch and a half). Magnifying the unwrapped cochlea gives the result in Fig. R-2 below. Sound enters at the oval window. The oval window transmits the sound waves into the upper cochlear chamber called the *scala vestibuli*.

The oval window is an oval opening covered with a membrane on each side. It is exaggerated (too large) in Fig. R-2. The window itself does not take up the entire end of the upper chamber. Sound travels down the scala vestibuli, crosses and is detected along the basilar membrane (the place depending on the frequency), and remnant sound waves travel through the *scala tympani*, the lower chamber to the round window.

Fig. R-2. Unwound Sketch of the Cochlea.



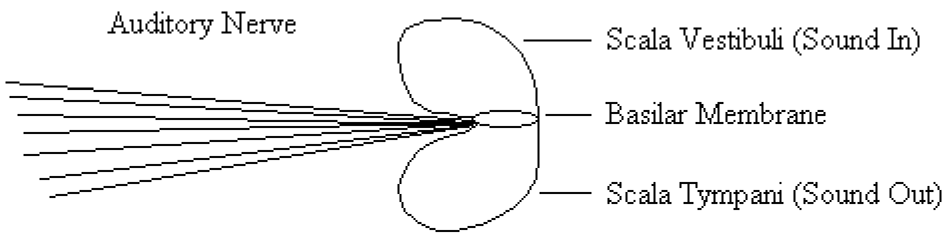
Sound Travels Down the Top Chamber, Around the End, and Back Up the Bottom Chamber.

The *round window* is a small round opening covered with a membrane on each side. It is located at the end of the path that the sound takes. It gives way to the pressure vibrations, preventing strong reflections that would occur if the end of the path were a solid wall. The size of the round window is exaggerated in Figs. R-1 and R-2. It does not take up the entire end of the lower chamber.

The *basilar membrane* lies between the upper and lower chambers in Fig. R-2. The sound is detected here. The basilar membrane is stiff at the left end near the oval window. This causes high frequencies to be detected there. Stiff structures vibrate more quickly than supple ones. The right end of the basilar membrane is not as firm and responds more readily to low frequencies. The slight disturbance of specific regions of the basilar membrane detects the different frequencies. The basilar membrane connects to nerves along its length. A cross section of the unwound cochlea found in Fig. R-3 shows this bundle of nerves called the *auditory nerve*.

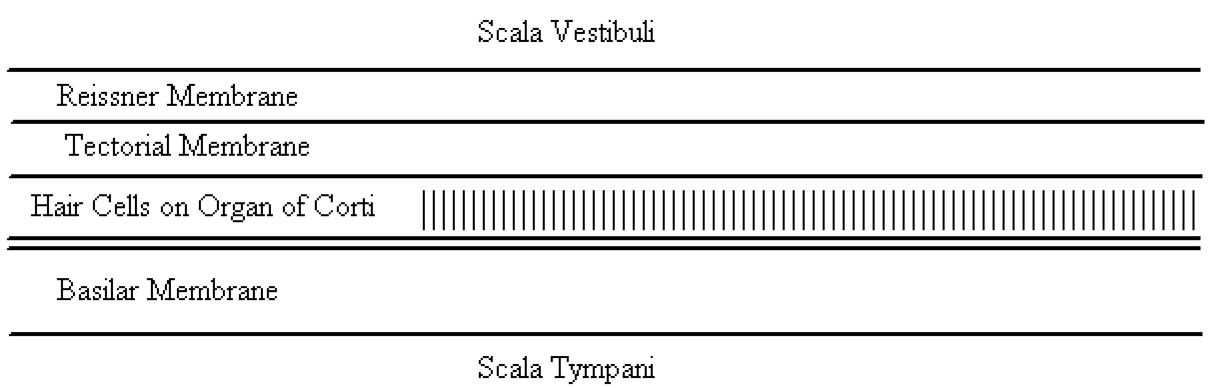
Fig. R-3 is a cross section of a cochlea that has been unwrapped. Think of the cochlea now as a hot dog with threads coming out from the left middle side throughout its entire length. Then cut the hot dog in half and look at the end that is cut. This is what you are seeing in Fig. R-3. To get the original cochlea you need to put the other piece back and roll it up. In Fig, R-3 you can look down the top and bottom chambers of the cochlea. The basilar membrane lies in the middle region. However, some structures in the middle have been omitted so far. To investigate further the structure of this central region of the cochlea, we magnify once again.

Fig. R-3. Cross Section of Unwound Cochlea.



The central region of the basilar membrane is blown up in Fig. R-4 below. The basilar membrane supports the *organ of Corti* (CORE-tee) which contains over 20,000 hair cells. As one region of the basilar membrane responds to a specific frequency, those hair cells in the vicinity move as a result of the disturbance of the underlying section of the basilar membrane. The moving hair cells get "tickled" by the *tectorial membrane*. Each hair cell is connected to a nerve member of the auditory nerve. The stimulated hair cells send electrical impulses to the brain by way of the auditory nerve. The louder the sound, the greater the movement, and a more intense electrical signal is sent along the auditory nerve. The *Reissner membrane* is a protective covering. Note the neighboring scala vestibuli and scala tympani in Fig. R-4.

Fig. R-4. Enlarged View of Basilar-Membrane Region.



The position of the stimulated hair cell along the basilar membrane determines the frequency perceived by the brain. Remember that the basilar membrane has various degrees of stiffness. Different sections of the basilar membrane move for different incoming frequencies. The stiffer parts near the oval window respond to higher frequencies while the more supple sections at the far end move under the influence of low frequencies.

In order for the ear to be able to detect frequencies as low as 20 Hz and as high as 20,000 Hz, the various frequencies detected are spaced in a special way. Imagine dividing the length of the basilar membrane into 10 equal steps. Figs. R-5a and R-5b illustrate two spacing rules. The spacing in Fig. R-5a is *arithmetic* or *linear*. You add the same amount every step of the way as you proceed from right to left. In Fig. R-5a we add 20 for each step. After 10 steps, we have moved 200 units. Since we started with 20, the final value is 220. The range is not very great.

Fig. R-5a. Linear Spacing Greatly Restricting Range.

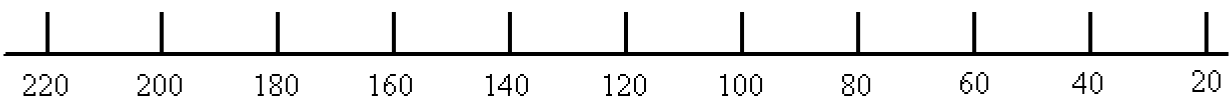
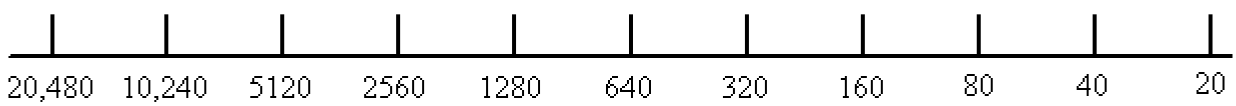


Fig. R-5b. Nonlinear Spacing of Frequency Detection Along Basilar Membrane.



On the other hand, the spacing in Fig. R-5b is *nonlinear*. Equal spacings do not correspond to equal increments. The particular nonlinear spacing in Fig. R-5b is *geometric* since every time you make a step, you multiply by a fixed number instead of adding a fixed amount. In Fig. R-5b, we multiply by 2 for every step, i.e., double each previous value. Let each vertical line in Fig. R-5b represent a bundle of hair cells on the organ of Corti. The many more hair cells that lie between these localized groups are omitted for ease of analysis. Then the numbers are the frequencies detected along the basilar membrane.

The social scientist Malthus drew attention to the concepts of arithmetic and geometric rates in his analysis of population around 1800. He proposed that population growth is geometric but improvements in food supplies are arithmetic as time goes on. Consider a chess board as another example. If you put 1 penny on the first square, 2 on the second, 4 on the third, and so on, roughly how much money would you need for the 28th square? How much for the 38th square?

A slightly smaller range of frequencies to consider spans from 30 Hz to 16,000 Hz. This stays away from the extremes 20 Hz and 20,000 Hz. The lowest note on the piano is about 30 Hz, while the highest note is approximately 4,000 Hz or 4 kHz. Frequencies beyond 4 kHz sound real shrill. See Fig. R-6 below for frequencies listed from 30 Hz to 16 kHz. We double at each step as before. Note that when we double 60, we write 125 instead of 120. We do this because doubling 125 gives the nice number 250, which we can double to get 500. The spacing in Fig. R-6 represents the spacing of the frequencies as detected along the basilar membrane. Remember that the "k" is shorthand for 1000.

Fig. R-6. Two Ranges of Frequencies.

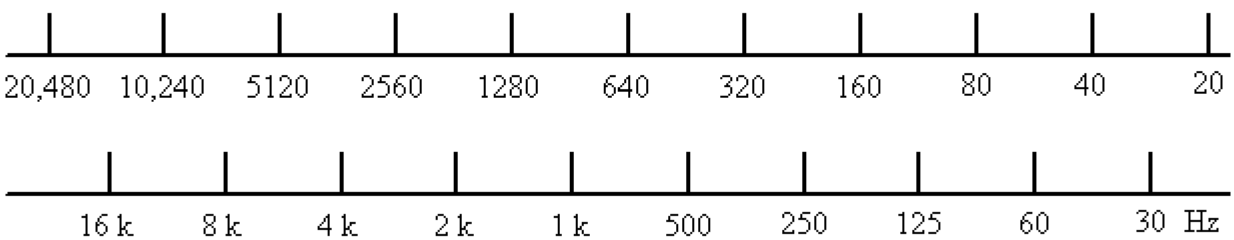
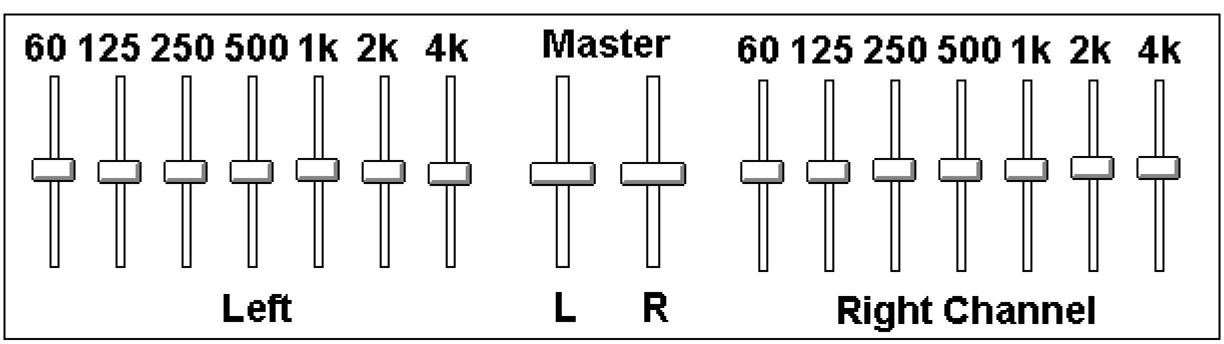


Fig. R-7 illustrates the equalizer we encountered earlier. Compare the numbers on the equalizer with our new series of frequencies in the lower line in Fig. R-6. The seven-band equalizer offers 7 frequency bands in which you can enhance the sound from your source. The equalizer can be considered as part of the 2nd stage in sound reproduction.

We learned about the functional similarity between the amplifier and filter. Both are modifiers. An active filter in an equalizer also incorporates an amplifier as part of the filter circuit itself. The center frequencies of the equalizer bandpass filters are arranged in jumps corresponding to the spacing of frequencies along the basilar membrane. The frequencies span the pitches on the piano except for the lowest octave.

Fig. R-7. Equalizer.



--- End of Chapter R ---

S. Perception

This chapter builds on the biology of the previous chapter. The emphasis here is on perception. This topic falls in the area of perceptual psychology. The brain receives the information from the ear by way of the auditory nerve. The sense of sound is perceived by the brain. If we adhere to the strictest definition of sound, which includes external vibrations and internal perception, then there must be a brain for sound to exist. In this viewpoint, the very definition includes the experience of sound.

The Place Theory of Hearing

The *place theory of hearing* states that the place where hair cilia get stimulated along the basilar membrane determines the perceived frequency. This is supported by observation. However, this "place observation" cannot explain some phenomena. Playing a specific tone excites the hair cilia in the corresponding region along the basilar membrane. Playing the tone louder stimulates the hair cilia more and this is perceived as louder.

However, experiments in the 1930s revealed that different pitches carefully matched for loudness were not perceived to be equally loud. In other words, amplitude doesn't solely determine loudness. Pitch influences loudness. We know this from experience since we find high-pitched tones loud and irritating. You might try to defend the place theory of hearing by saying that it is easier to stimulate the stiff region of the basilar membrane. Therefore, high-pitches will of course be perceived to be louder. But maybe this is not the correct analysis. Perhaps other factors and the brain play a role.

There is another observation that presents difficulties if we try to explain everything by the place theory of hearing. Psychologists have found that we can perceive raises in pitch when tones are increased in loudness. We might defend the place theory of hearing by reasoning as follows. If you increase the loudness, the hair cilia shake so much that you get some neighbors shaking also. Since we assumed above that the stiffer end of the basilar membrane responds better, we expect to excite some neighbors at the higher-frequency end more. This raises the tone. But is it enough? Wouldn't the original hair cilia be shaking more wildly and overshadow the higher pitch. These issues are being studied today.

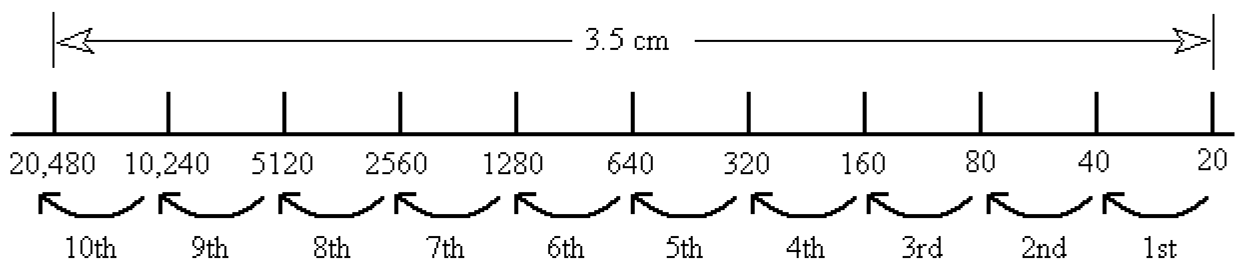
You get the point. We are in gray areas on some of these questions. So although the place theory of hearing is based on the observed spacing of frequency sensitivity along the basilar membrane, it does have limitations in explaining some perceptual phenomena.

Don’t be discouraged by this. Compared to physics, psychology is really hard. The brain and ear are far more complex than compression waves traveling through an elastic medium such as a slinky or air. Also, physics has been around longer, since the 1600s. Psychology as a separate discipline began more recently in the 1800s. We may have to wait centuries before understanding some perceptual subtleties really well.

Let's focus on some simple aspects of the place theory of hearing that we know to be true. Various frequencies are detected along the basilar membrane. Every equal step along the basilar membrane (approaching the oval window) results in doubling the frequency. Therefore, each step corresponds to a pitch increase of an octave. We can count the steps to go from 20 Hz to 20,000 Hz, doubling the frequency each time. We arrive at 10 steps or octaves.

Frequencies along the basilar membrane are seen the Fig. S-1. Most people will have trouble hearing 10 octaves. If we use our more practical range of 30 Hz to 16,000 Hz, we obtain 9 octaves. Note that in either range we use, each doubling step is still an octave. We just start at a different frequency (30 Hz) and make 9 steps instead of 10. But since you need all 10 steps starting from 20 Hz for the entire 3.5-cm length of the basilar membrane, we can say that each octave corresponds to 1/10 this amount, which is 0.35 cm or 3.5 mm.

Fig. S-1. The Ten Octaves of Human Hearing and Positions Along the Basilar Membrane.



The Decibel Scale

We saw how the place theory of hearing gives us a basic understanding of our perception of frequency. Frequency is one of the three fundamental characteristics of periodic waves. The other two are amplitude and timbre. The place theory of hearing also provides us with the essential mechanism for detecting amplitude. The hair cilia, responding at a particular place along the basilar membrane determined by the frequency, get stimulated more when the amplitude of the sound is increased. The greater stimulation of the hair cilia gets sent to the brain through the auditory nerve.

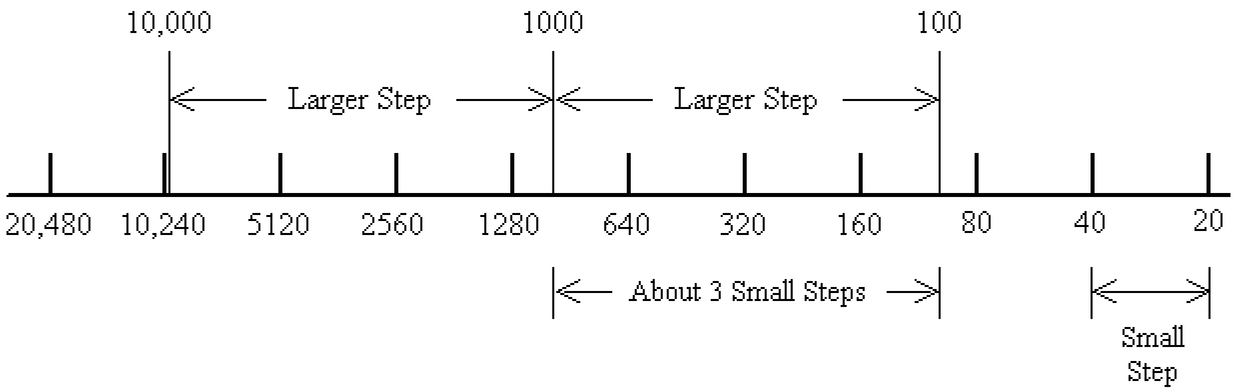
The ear detects a very impressive range of frequencies, from 20 to 20,000 Hz (about 10 octaves). Likewise, the ear detects an impressive range of amplitudes. We can hear the barely audible sound made as a pin drops onto a soft cushion. We can also hear the full blast of an orchestra. We have to be careful that we do not expose ourselves to sounds that are too loud. These can damage our ears. We will take that subject up more fully in our next chapter where we discuss hearing loss.

The way the ear is able to detect so many levels of loudness is due to the fact that it is stubborn in responding to a new level of loudness. The new stimulus must be much greater than the previous one to hear an appreciable increase. We encountered a similar idea in frequency detection. To stimulate the next neighboring group of hair cilia (another 3.5 mm along the basilar membrane), we need to really turn up the frequency. We need to be in the next octave. See Fig. S-2 for another sketch of the frequencies along the basilar membrane. Every time we make a 3.5-mm step along the basilar membrane, the frequency response doubles. In this way, the basilar membrane can pack the tremendous response range of 10 octaves into a total length of only 3.5 cm.

The secret in understanding the strategy at work along the basilar membrane is to realize that equal steps mean you multiply instead of adding. We studied this earlier with the small step size of 3.5 mm. Every small step of this size results in doubling. However, you can take bigger steps. See Fig. S-2 for a larger step size. The secret still applies, but now you multiply by a different number. This number is 10 for the larger step size in Fig. S-2. This "secret" when applied to the psychology of perception is known as *Weber's* (VAY-ber's) *Law*.

The organism will not perceive an equal-step jump in perception unless the original stimulus is multiplied. To get the ear to respond at equal small steps along the basilar membrane, you double the frequency at each step. To get the ear to respond to equal large steps along the basilar membrane, you multiply the frequency by 10 at each step. Note that three of the smaller steps in Fig. S-2 are necessary to make one larger step in Fig. S-2.

Fig. S-2. Weber's Law: Equal Steps Indicate Multiplication.



Weber's Law is usually discussed within the context of loudness. The same idea applies. If you want roughly equally-perceived jumps in loudness, you need to multiply how many sources of the sound you have at each step. In fact, Weber's Law applies in this approximate way to the stimulation of all five senses. Table S-1 lists the application of Weber's Law to the five senses. We employ in Table S-1 the two different step sizes used in Fig. S-2, i.e., multiplying by 2 and multiplying by 10.

There is nothing "sacred" about choosing multiplying factors of 2 and 10 in Table S-1. Weber's Law applies to any number. It just means that the size of the perceived jump will be different. Remember that 3 of the small "doubling" steps equal one of the larger "tenfold-increasing" steps. Refer again to Fig. S-2 to impress this on your memory. This can be also be understood from observing that doubling three times (2 x 2 x 2 = 8) gives us approximately a tenfold increase.

Table. S-1. Weber's (VAY-bers) Law Applied to the Five Senses.

Sense	Small Equally-Perceived Jumps	Large Equally-Perceived Jumps
Sight	Candle Flames: 1, 2, 4, 8, 16.	Candle Flames: 1, 10, 100, 1000.
Hearing	Dropping Pins: 1, 2, 4, 8, 16.	Dropping Pins: 1, 10, 100, 1000.
Taste	Grains of Salt: 1, 2, 4, 8, 16.	Grains of Salt: 1, 10, 100, 1000.
Smell	Drops of Perfume: 1, 2, 4, 8, 16.	Drops of Perfume: 1, 10,100, 1000.
Touch	Strands of Hair: 1, 2, 4, 8, 16.	Strands of Hair: 1, 10, 100, 1000.

The study of stimuli in terms of math and physics is the subfield of perceptual psychology called psychophysics. Fechner (FECK-ner), considered the founder of experimental psychology, came up with a mathematical formula that embodies Weber's Law. Fechner was a physicist and early psychologist. Weber and Fechner both worked in the 1800s during the birth of psychology as a discipline. Weber's Law (or Fechner's mathematical equivalent) is not an exact law; however, it is useful as a starting point in analyzing perception. *Fechner's Law* states that a response is proportional to the logarithm of the stimulus: $R = k \log S$. What's this? Logarithms? Don't worry. You already understand the law if you understand Table S-1. The formula is just the mathematical way of writing the information found in the table.

The area of perception is one of the most challenging applications of mathematics and physics. It is part of the subject of experimental psychology. We are trying to come up with ways, using numbers, to describe how one responds to a stimulus. The detection system offers important clues. Here is where biology and physics come into play. The stiffness of the basilar membrane and its vibrating response to incoming sound is a case in point. Getting a mathematical handle on the perception of stimuli is called *scaling*. Our task now is to scale the perception of loudness. We will present the historical scaling based on Fechner's Law (also Weber's Law). The result is the *decibel scale* we use today as a practical way to measure sound levels. We sidestep working explicitly with Fechner's scaling formula, just as we avoid detailed equations elsewhere in this text. However, it should be stressed that although mathematics is important, it's even more important to understand what's behind the mathematics. What follows is the essence of the historical scaling law for the stimulus-response of loudness.

The sound-level scale is given in Table S-2. It extends the reasoning of Table S-1, where Weber's Law is applied to the five senses. In both tables, we consider dropping pins. The scale numbers are simply counting numbers for the large-sized steps that now continue on for 14 phases. Each perceived jump (step) signifies a tenfold increase in the actual number of pins that drop. The threshold of human hearing is taken to be the sound made by the drop of a pin on a soft cushion. Sound-level meters are designed to get accurate measures of levels. The examples in Table S-2 are approximate. Note the fundamental feature of Weber's Law. The scale rating proceeds by equal jumps or increments while the actual number of sources (pins) for the stimuli goes up by a tenfold increase at each step.

The unit for the scale numbers is the *bel*, named after Alexander Graham Bell. Bell invented the telephone (1876) and made contributions to the study of sound. His mother was deaf and so was his wife. The third column in Table S-2 gives the number of *decibels*. The metric prefix *deci* means one tenth. One tenth of a bel (0.1 bel) is one decibel (1 dB). The decibel is a smaller unit so you need more of them to make up the larger bel. Compared to the reference drop of one pin, a decibel level of 140 is letting 100 trillion pins fall an equivalent distance on the appropriate surface material.

Table. S-2. The Decibel (dB) Scale.

Dropping Pins	Scale	dB	Example
1	0	0	Drop of a Pin
10	1	10	Breathing
100	2	20	Gentle Breeze
1,000	3	30	Whisper
10,000	4	40	Quiet Office
100,000	5	50	Library
1,000,000	6	60	Conversation
10,000,000	7	70	Busy Street
100,000,000	8	80	Factory
1,000,000,000	9	90	Nearby Subway Train
10,000,000,000	10	100	Machine Shop
100,000,000,000	11	110	Construction Site
1,000,000,000,000	12	120	Rock Band
10,000,000,000,000	13	130	Pneumatic Riveter
100,000,000,000,000	14	140	Nearby Jet

The number of pins used to make the sound gives us the intensity from the point of view of physics, not perception. The perceptual scale is the compressed scale that goes from 0 to 14 bels or 0 to 140 dB. We can consider our engineering method of measurement as noting the number of pins we drop to make the sound. So we can consider the number of pins we drop as our intensity. The bel scale is the sound-level scale to approximate our perception of sound. It can quickly be obtained by counting how many zeros there are after the 1 in the number of pins dropped. Therefore, for 100 pins we have 2 (since there are 2 zeros after the 1); for 1000 pins we have 3, and so on.

Finally, to get the decibel column, multiply by the number of bels by 10. This is the prescription given by Fechner's Law: $R = k \log S$. You obtain the stimulus (S) - the number of pins you drop. The "log" is the instruction to count how many zeros are after the 1. Then you multiply by k, which is our multiplier 10. Engineers like to write I_r instead of S. I_r stands for relative intensity; we always compare to the drop of 1 pin. Also, the Greek letter β (beta) is used instead of R (R is used for resistance). We will use β to stand for the sound-level response in dB. Then, Fechner's Law for sound level can be written as $\beta = 10 \log I_r$.

Engineers get precise with the standard of intensity for the drop of one pin. Imagine a drop-of-the-pin sound that is sustained. The energy coming to the area of your eardrum each moment must be equivalent to one trillionth of the energy of a one-watt light bulb falling on an area of one-meter square. This is indeed a small amount of energy. The pin, the cushion, falling height, and distance away all affect the sound level. See Fig. S-3 for a quick overview of approximate sound levels.

Fig. S-3. Overview of Sound Levels (dB).

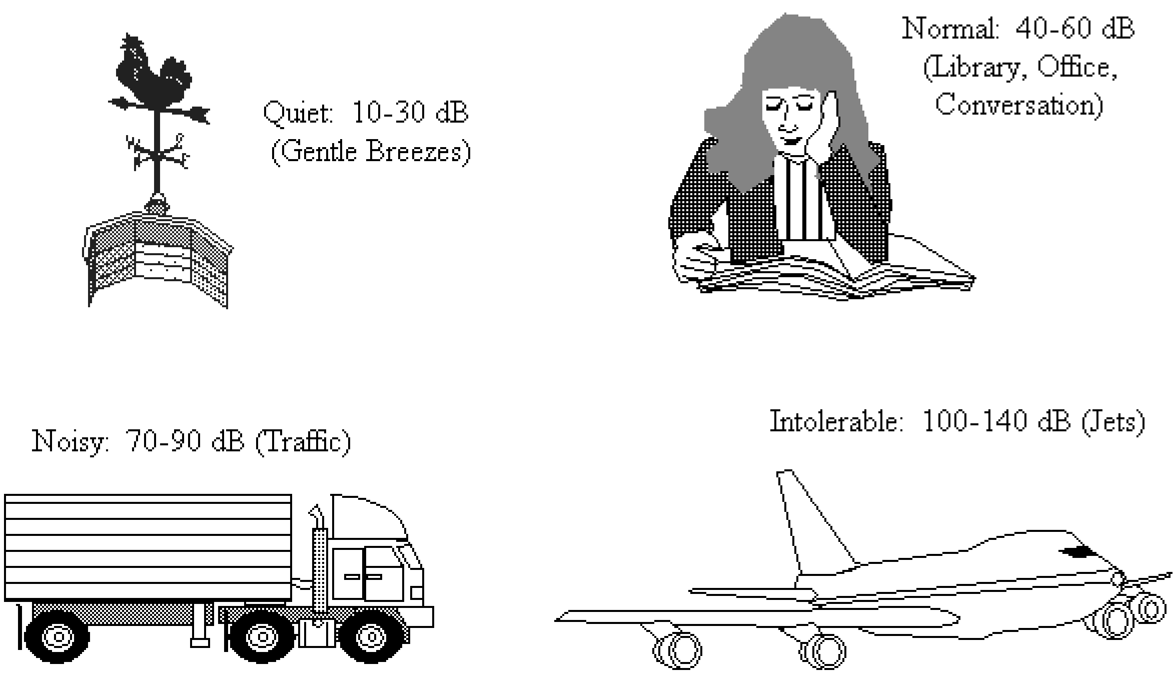


Table S-3 relates sound levels to the language used by composers to indicate on the music score how loud music should be played. These are called dynamic markings. They instruct the performer how softly or loudly to play specific passages and notes. These instructions to performers are traditionally given in Italian. The decibel equivalents given in Table S-3 are approximate. Performers know that sound levels are especially subjective due to an interesting feature of our perceptual process. We perceive a loud sound to be extra loud if it is preceded by silence. So if you have a *ff* passage coming up, play extra softly a little before. Our perceptual dependency on what comes before the sound is just one of the subtleties that makes perception a challenging subject to study.

Table. S-3. Approximate Sound Levels for Musical Dynamic Markings.

dB	Dynamcis	English	Italian
30	*ppp*	Extremely Soft	Pianississimo
40	*pp*	Very Soft	Pianissimo
50	*p*	Soft	Piano
60	*mp*	Moderately Soft	Mezzo Piano
	mf	Moderately Loud	Mezzo Forte
70	*f*	Loud	Forte
80	*ff*	Very Loud	Fortissimo
90	*fff*	Extremely Loud	Fortississimo

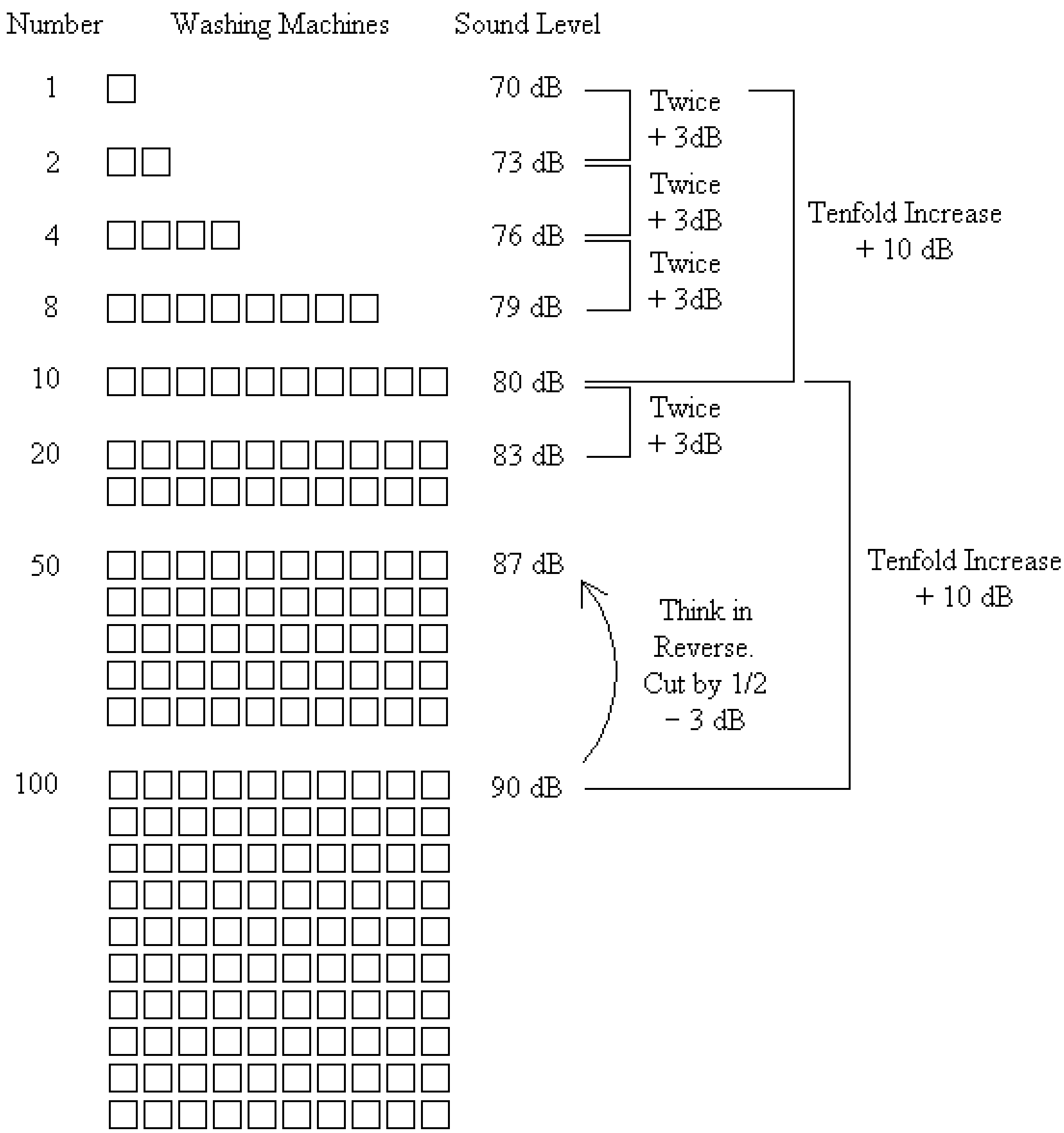
Table S-4 below gives two handy rules for determining sound levels when we increase the number of sources. You may recognize that these rules are the expressions of Weber's Law as applied to the decibel scale. Remember our steps along the basilar membrane. There, one large step (tenfold increase) was equal to about three smaller ones (twofold increases). Doubling the number of sources represents the smaller step size in loudness (add 3 dB), while increasing the number of sources by a factor of ten is our larger step size (add 10 dB).

Table S-4. Two Simple Rules for Determining Sound Level.

Increasing the Number of Sources	Corresponding Change in dB
Two of the original sound sources.	Add 3 to your original decibel level.
Ten of the original sound sources.	Add 10 to your orginal decibel level.

A working example of the rules found in Table S-4 is given below in Fig. S-4. We start with one washing machine at 70 dB. Of course we need to be at the right distance. Assume that we can have more machines at the appropriate distance. Every time you double the amount of machines, you add 3 dB. Every time you multiply the number of machines by 10, you add 10 dB. To get the level for 50 machines, step from 1 machine (70 dB) to 10 machines (80 dB), then to 100 machines (90 dB), and cut the final number of machines in half. You then subtract 3 dB instead of adding. With our two rules, you can determine so many cases. You can easily estimate the levels for other amounts in between. For example, 7 machines produces about 78 dB. Why would 5 machines give 77 dB? Do not make the common careless mistake and state that 2 machines would be 2 x 70 dB = 140 dB. Note that 100 machines produce only 90 dB.

Fig. S-4. Example in Determining Sound Levels.



The study of how frequency relates to loudness was undertaken in the 1930s. The reference for the sound-level scale (dB) is a 1000-Hz tone. Think of a precise experiment where we do not drop pins but use flutes that produce a barely audible 1000-Hz tone at a given distance. Then, at the proper distance, 1 such flute gives 0 dB, 10 flutes give 10 dB, 100 flutes give 20 dB, and so on. If we work with another frequency, subjects perceive a different "loudness spectrum."

So we modify our table from the very early chapter concerning the physical and perceptual characteristics of sound. This table is reproduced in Table S-5. However, we add a qualifier to emphasize that the correspondence is approximate.

Table S-5. Approximate Relationships Between Physics and Psychology.

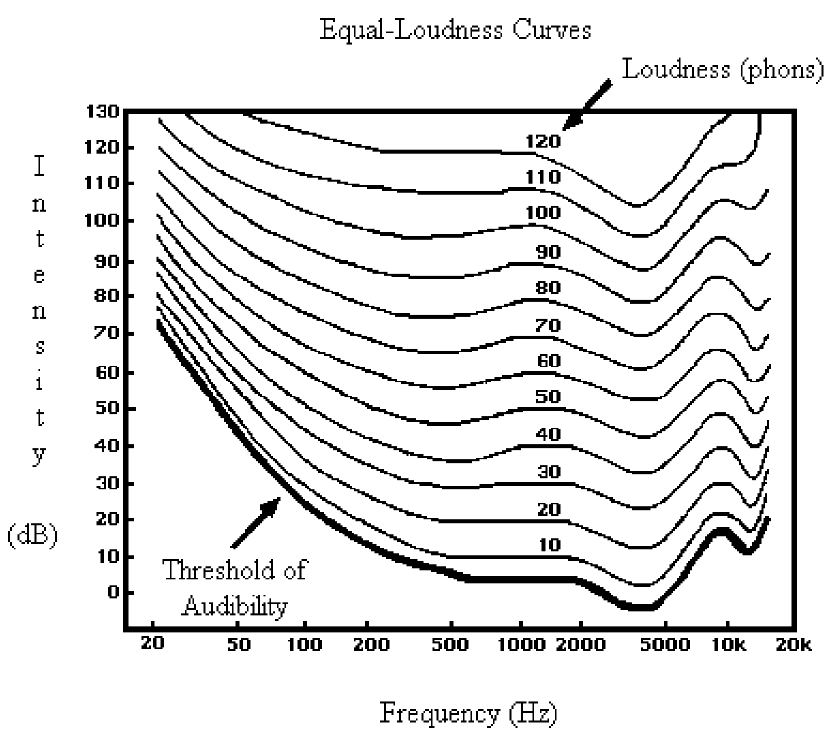
Physical	Perceptual
Amplitude	Loudness
Frequency	Pitch
Waveform	Timbre

Fletcher and Munson (1933) made a study of the perception of equal loudness and how the sensitivity of the ear varies across the frequency spectrum. They started with the sound-level scale which assumes a 1000-Hz tone. They then presented subjects with different pitches. Imagine replacing the 1000-Hz flute with one at 500 Hz. We then play one of these, then 10, then 100, and so one. Of course, in practice one uses a tone generator and controls the energy output to simulate the series of cases, 1, 10, 100, 1000, and so on. Fletcher and Munson found that their subjects perceived different loudness levels for the different frequencies played at the same level according to a scientific instrument. For example, if we employ the 1000-Hz flutes, we get a threshold response when one such flute is played. Now if we switch to a 50-Hz instrument (bass tone), we need 10,000 instruments to just get the subject to hear anything. This is 40 dB higher! We are less sensitive to bass tones than we are to 1000 Hz.

The Fletcher-Munson experiment carefully starts with a set pitch. We push up the decibel level, monitoring it on a scientific instrument, until the subject hears something. This establishes the threshold for the pitch. We then draw a curve across the audio spectrum which represents thresholds (see the lowest curve in Fig. S-5).

Other equal loudness curves are determined using 1000 Hz as the reference. By definition, the perceiver is in agreement with the sound-level meter at 1000 Hz. This phase of the experiment begins with the 1000-Hz reference tone along with the pitch other than 1000 Hz. The reference is set to a decibel level according to the sound-level meter. For example, the 1000-Hz tone might be set to 30 dB. Then, the other tone with a different frequency is presented to the hearer. The hearer adjusts its volume so that the different frequency matches the loudness of the fixed 30-dB reference of 1000 Hz. Tones that sound the same in loudness, are found to have different decibel levels according to the meter. In our above example, 40 dB at 50 Hz sounds as loud as 0 dB at 1000 Hz. To avoid confusion, it is said that they have the same value in *phons*, the subjective scale.

Fig. S-5. Fletcher-Munson Curves.



Find any point along any curve in Fig. S-5 as follows. First choose a specific curve, then a point along that curve. Suppose you choose the 30-phon curve. You then slide along this curve to any point. Consider stopping at the point to the left, corresponding to 50 Hz (horizontal) and 60 dB (vertical). This point tells us that in order to hear a tone of 50 Hz at the same level as the reference 1000-Hz tone at 30 dB, we need to make the 50-Hz tone 60 dB. In other words, 50 Hz at 60 dB has the same loudness as 1000 Hz at 30 dB. Each is said to have 30 phons. Note that the dB-value and phon-value agree at the 1000-Hz reference for all levels of intensity.

As we move to the outer limits of human hearing, the curves rise. Focus on the lowest curve, the threshold curve. This curve describes barely audible sounds. The threshold curve gets higher at each end of the spectrum. Note the enhanced sensitivity near 3000 Hz. Here all the curves dip down. The ear canal is like a small pipe and has a resonance frequency near 3000 Hz. The ear canal amplifies sound near 3000 Hz as a resonance effect. The threshold for low frequencies is high. This difficulty in hearing low bass tones is actually good. Otherwise, we would hear the low-frequency sounds made inside our bodies. Since the ear is not very sensitive in the low-frequency range, sound-level meters have a special weighting mode (A-weighted) that discounts lower frequencies. Meters also usually have fast and slow response modes, the slow response giving more or less an averaged sound level.

Other Perceptual Phenomena

1. Masking.

When more than one sound is perceived, the louder sounds are heard more easily. Therefore, it is possible for a loud source to overpower a soft one. This can happen to the point where we no longer hear the soft one. This is called *masking*. We all have experienced the trouble of hearing something soft because something else is louder and distracting.

White noise can help mask sounds. White noise presents us with all frequencies. We have encountered such examples as the fan, sound of the ocean, and rain. Putting on a fan helps some people go to sleep due to the masking effect. Distracting sounds are covered up by the soothing even-distribution of all frequencies.

2. Periodicity Pitch.

Masking is an example where loud sounds prevent us from hearing other sounds. Here we see that certain sounds can cause us to hear other sounds not originally present in the source. The place theory of hearing cannot explain why we perceive tones that are not present in the sound waves. Consider sending a 200-Hz sine wave to the ear along with a 300-Hz tone. The brain perceives the 200-Hz and 300-Hz tones. It recognizes that these two tones can be thought of as the second (H2) and third (Hz) harmonics relative a 100-Hz sine wave. The brain registers at a lower level of intensity the fundamental at 100 Hz (H1), the *periodicity pitch*. A frequency-analysis of the incoming waves is done by the ear-brain system, establishing and perceiving the fundamental!

We saw that each periodic tone can be represented by a Fourier spectrum of harmonics. Most of the time the fundamental is the strongest component. The ear-brain expects the fundamental to be there and puts it in if it's not. This explains why we hear bass better than we should from a small 2-inch speaker. The low fundamental tones are lost to some extent, but the ear-brain system supplies them. The ear-brain knows they should be there.

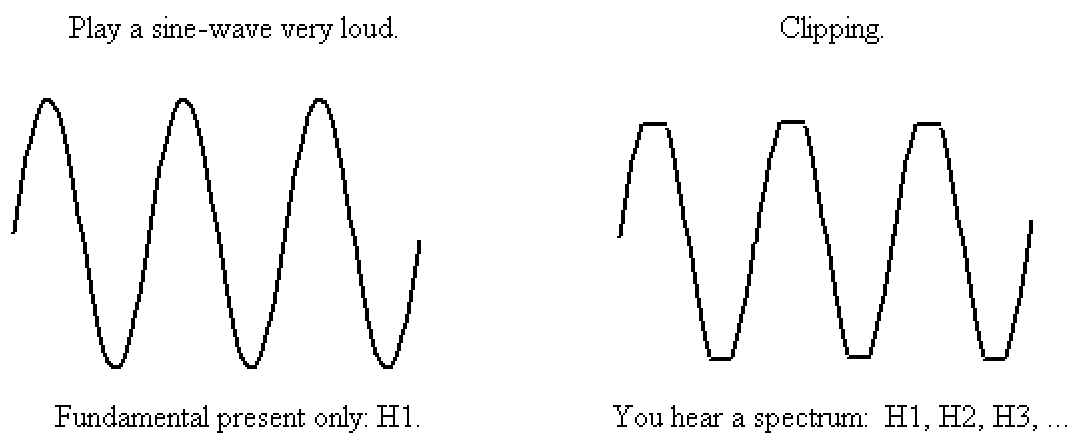
Now consider a rephrase of our earlier question about a tree falling in a forest. Is "sound" present for a fundamental tone if it is heard but there is no source making that frequency?

A *periodicity theory of hearing* has been developed based on observations that we hear fundamental tones not present in the incoming sound. In such a theory, Fourier analysis of incoming waves are relevant. Both the place theory of hearing and the periodicity theory of hearing are important in providing for a more complete picture of hearing.

3. Aural Harmonics

Another fascinating case of hearing components of sound not present in the original sound is experiencing aural harmonics. A sine wave has one harmonic, the fundamental (H1). However, if you play it loud enough, the eardrum can't vibrate through the distance it needs to. You get clipping of the wave as the eardrum reaches its limits. The information sent to the middle and inner ear is now no longer a sine wave. Therefore, you perceive overtones (*aural harmonics*), frequency components not in the original sound entering the ear. See Fig. S-6

Fig. S-6. Aural Harmonics Due to Clipping.



The amplitude of the sine wave is so great in Fig. S-6 that the eardrum cannot faithfully reproduce it. The wave gets distorted. If we turn up the volume too high, the system is not able to handle it. Distortion can occur in making tapes if we tape the source with the amplifier setting too high. The strength of the amplified sound is indicated in recording equipment by a sound-level display to guide us. Whenever an electronic component distorts the shape of a sine wave, harmonics of the sine wave appear. This is referred to as *harmonic distortion*.

4. Combination Tones.

Playing two very loud sine waves causes us to hear additional tones beyond those discussed above. We hear the original tones, the sum and difference tones at low levels, and possibly even more tones. For example, if we play a 500-Hz tone very loud with a 700-Hz tone, we hear 1200-Hz (sum) and 200 Hz (difference). The lower tone may be difficult to hear due to our lack of sensitivity to low pitch.

The resulting frequencies are the *combination tones* made by combining the original frequencies. Other combinations found by adding and subtracting various multiples of the original frequencies may also be heard. How can you use beats to determine if a 1200-Hz tone is heard when a 500-Hz tone is played loudly with a 700-Hz tone?

5. Binaural Effects.

Binaural effects are phenomena that result from our having two ears. Just as two eyes (binocular vision) give us an excellent sense of three dimensions, two ears provide us with a better three-dimensional sense of hearing. With two ears, we can more easily tell from which direction a sound comes. Sounds at our left do not reach the right ear as well. The brain constantly compares the sound level at each ear to give us a perception of our surroundings. For long-wavelength bass tones, the brain relies more on a comparison of phases. When a compression reaches the closer ear, there is a delay before the compression reaches the farther ear due to the extra distance. So different parts of the wave cycle reach each ear at any given time.

The important role of the brain in processing signals from the auditory nerve is evident in the following experiment using two ears. Earphones are employed to send a different signal into each ear. When the different tones are close in frequency, we hear beats. Even when care is taken to play the tones softly to eliminate any bone conduction in the skull, beats are still perceived. The conclusion is that the beats occur in the brain. When we usually hear beats, the waves combine physically outside the ear. The pressure waves add. The result is a fluctuation in the sound wave itself. You can hear it with one ear.

When the tones are separated, they cannot physically add together. Each tone enters a different ear. However, the combination of these signals from the auditory nerve of each ear to the brain is processed in the brain. The brain effectively combines the waves in a way similar to the physical addition of wave amplitudes. The beats experienced are perceptual or psychological rather than physical. Do they really exist? Is there sound if a tree ... ?

--- End of Chapter S ---

T. Audiology

Our focus in this text started with physics. This led us into many engineering applications in electronics. After looking at the technological developments in sound reproduction and electronic synthesizers, we turned to biology. Then we examined the psychology of perception. This chapter takes us into an area related to medicine.

Audiograms

Audiograms are medical records of an ear's ability to hear sound. The study of hearing loss is called *audiology*. It is a fairly recent field of study. It originated after World War II to assess the hearing losses of war veterans. The government provides disability payments for veterans that suffer injuries, many of which severely curtail employment opportunities. The coordinated effort to study the science of hearing loss led to the use of the word audiology in 1946.

Table T-1 lists the ranges for human hearing in terms of sound level and pitch. As we have studied, these features rely mainly on the amplitude and frequency of the sound respectively. Since the timbre is a result of overtone frequencies, testing amplitude (sound level) and frequency for sine waves suffices.

Table T-1. Hearing Ranges: Sound Level and Frequency.

Characteristic	Range	Typical Problem Regions
Sound Level (β)	$0 \le \beta \le$ "140 dB"	Outer and Middle Ear
Frequency (f)	20 Hz $\le f \le$ 20,000 Hz	Inner Ear

Quotations appear with 140 dB in Table T-1 because this limit is not relevant in testing. You are tested for the lower limit. If you can hear levels near 0 dB, you can hear louder sounds. With frequency, things are different. The different frequencies need to be tested.

Hearing impairment can result from problems in each of the three regions of the ear. Problems in the outer and middle ear tend to affect all frequencies. There can be blockage in the ear canal or damage to the bones in the middle ear. Problems in the inner ear can be selective, effecting some frequencies and not others. Perhaps only one region along the basilar membrane is not functioning properly. One suffers loss in hearing those frequencies corresponding to the problem area. The result is that some frequencies are not heard unless they are loud. If an appreciable percentage of the basilar membrane is not working properly, sounds lose clarity. Doctors can remove obstructions such as wax in the outer ear. Operations can often repair the middle ear. However, difficulties in the inner ear are hard to correct. Hearing aids that amplify everything do not really help.

Data from testing frequency for hearing thresholds can be easily plotted on a two-dimensional graph. One axis is chosen for frequency, the horizontal. The other axis, the vertical, is chosen for the sound level of the hearing threshold. See Fig. T-1 for a grid ready for data. The frequencies tested do not include the entire range of human hearing. The focus is on frequencies present in speech. Speech falls mostly between 250 Hz and 4000 Hz. The intent of hearing-loss assessment is to determine whether an individual can function in society without a "hearing challenge." Therefore typical testing is restricted to the frequency range 125 Hz to 8000 Hz (See Fig. T-1).

The limits of the usual hearing test correspond to an extra octave on each side of the range for most speech. Hearing loss in this range can present one with difficulties in communication. The audiogram is a diagnostic record for this range. If you can pass the test everywhere up to 8000 Hz, you pass. Of course you may have a serious loss at 10,000 Hz and beyond. But for practical purposes you can understand speech and you are judged to be satisfactory. This was the government's original concern in measuring the hearing of veterans. If you are a veteran, the question is can you understand speech? Communication is crucial for employment. The government doesn't care if you can't fully appreciate the richness of the high notes in the Sibelius *Violin Concerto* because you miss out on some of the overtones.

The audiogram is a card on which the response measurements are recorded in a grid similar to Fig. T-1. The frequencies jump by octaves. Pure tones (sine waves) are presented to the ear. The softest sound level heard at each of the main frequencies is noted. Since the ear is not uniformly sensitive to all frequencies, bass tones are boosted to compensate. Normal thresholds are then 0 dB for each of the tested frequencies.

Fig. T-1. Audiogram Grid.

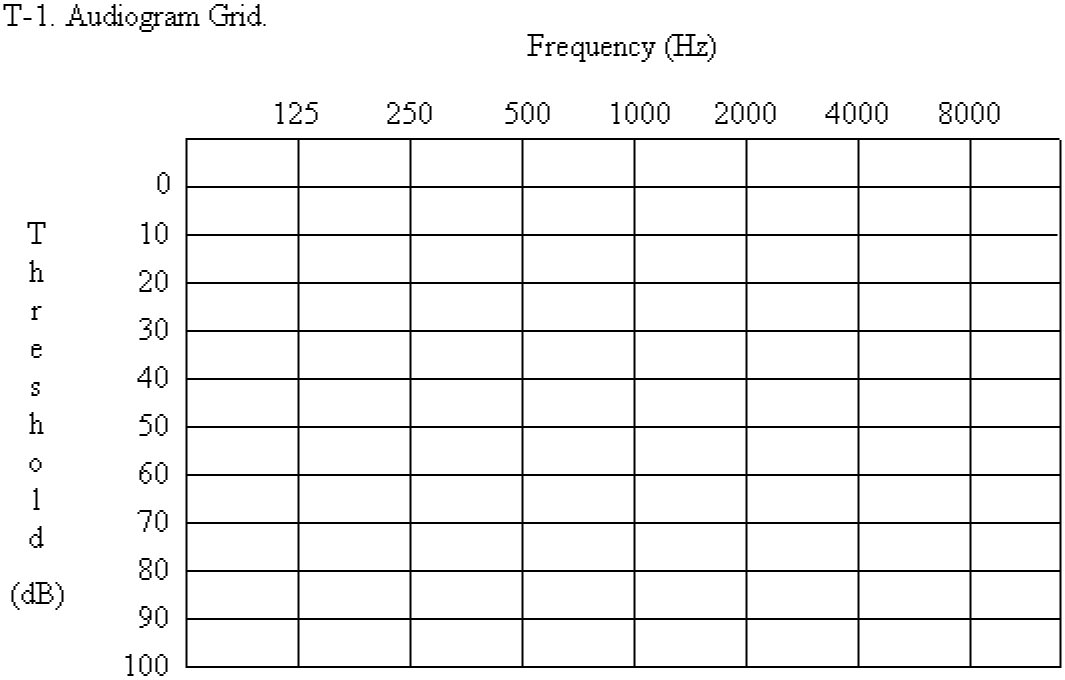
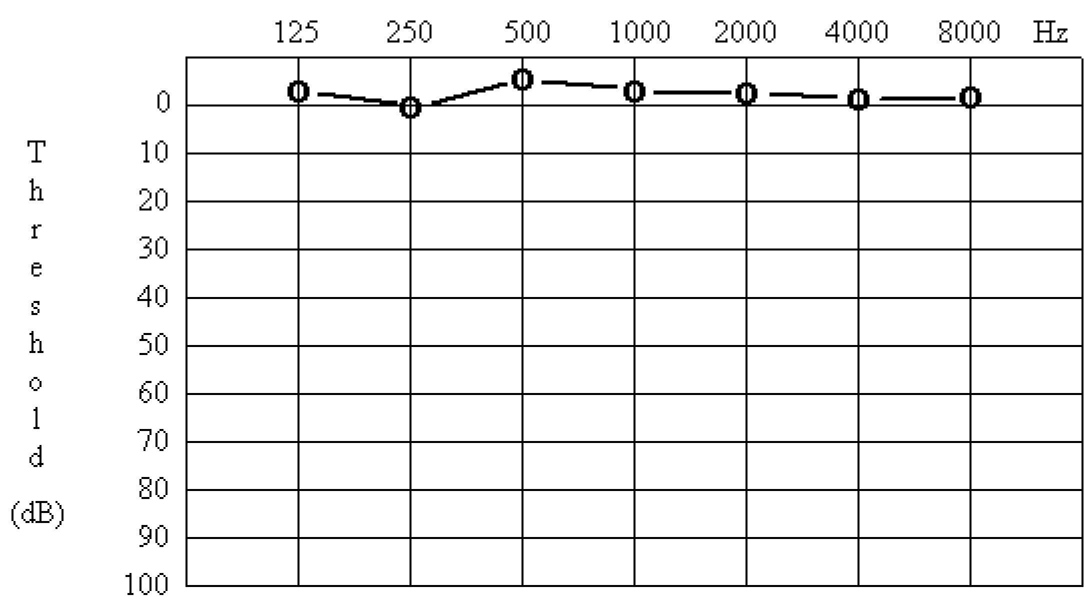


Fig. T-2 depicts an audiogram for a normal ear. The measured thresholds are connected by straight lines. The ear hears as low as 0 dB for the frequencies across the spectrum. In fact, this particular ear exceeds the normal a little. Younger people may be able to score better than 0 dB for many of the frequencies. The zero reference is a statistical average. Some people have better hearing than the average. A “–3 dB” means that an ear can hear a sound softer than the drop of a pin on a soft cushion. The minus 3 signifies half the intensity. We might approximate this by dropping half a pin or a smaller pin. A “–5 dB” is even better.

This is analogous to some people having better vision than normal. The normal acuity is 20/20. You see at 20 ft what you should see at 20 feet. But some people can see at 20 ft what normal people see at 15 ft. They are better. Their vision is given the rating 20/15. A few people can see at 20 ft what normal folks see at 10 ft. Their vision is 20/10. As an example, it is said that baseball great Babe Ruth could read a far license plate when his friends couldn't determine the color of the plate.

Hearing thresholds may be a little better than 0 dB or a little worse. As long as the threshold is near 0 dB, we consider the response normal. Some consider 20/25 vision normal. These people need to be at 20 ft in order to read letters they should be able to discern at 25 ft. However, this is very close to normal. A vision of 20/30 may still be good enough to pass a driver's test without needing glasses. Similarly, someone's typical hearing thresholds may be +5 dB for some frequencies; other people may score around –5 dB. We might say that at +5 dB one has a slight loss just as we might say that a person with vision of 20/25 or 20/30 has a slight visual-acuity impairment (without glasses). It is easy to correct for the common forms of visual impairment by prescribing glasses since the difficulty lies with all frequencies of light, the image. It is more difficult with the ear since some frequencies may be fine, others not.

Fig. T-2. Audiogram of a Normal Ear.



Hearing Loss

Fig. T-3 below shows some degrees of hearing losses. A slight loss indicates that sound levels need to be louder than 0 dB to be heard, but not greater than 10 dB. For example, a 5-dB threshold indicates a slight loss. A mild loss requires sound levels to be near 15 dB to be heard. A threshold near 25 dB denotes a moderate loss.

Fig. T-3. Degrees of Hearing Loss.

A threshold of 30 dB across the audio spectrum indicates that any sounds below 30 dB cannot be heard. This means that an individual with a 30-dB threshold cannot hear at all the drop of a pin on a soft cushion (0 dB), breathing (10 dB), a gentle breeze (20 dB), or a faint whisper (a little less than 30 dB). The whisper at 30 dB is barely audible. The hearer is not able to understand the whispered message. It is too faint to make out phrases and sentences.

Table T-2 indicates how a person with a 30-dB threshold (across all frequencies) hears. We simply call 30 dB our new threshold. The hard-of-hearing person hears 30 dB as the average person hears 0 dB. The hard-of-hearing person hears 40 dB as 10 dB and so on.

The descriptions for slight, mild, and moderate losses refer to the general population. In settings where there are severe hard-of-hearing individuals, all of the above impairments are considered virtually normal. Schools for the deaf and hard-of-hearing can have individuals with hearing thresholds of 100 dB.

Table. T-2. Hearer's Perceptions with 30-dB Threshold Compared to Normal Perceptions.

dB	Normal Perception	Hearer with 30-dB Threshold
0	Drop of a Pin on a Soft Cushion	Not Heard.
10	Breathing	Not Heard
20	Gentle Breeze	Not Heard
30	Whisper	Drop of a Pin on a Soft Cushion
40	Quiet Office	Heard as Breathing
60	Conversation	Heard as a Whisper
90	Nearby Subway Train	Heard as Conversation
120	Rock Band	Heard as Nearby Subway Train

Fig. T-4 gives a rough sketch of a possible audiogram for Beethoven as he was going deaf during his adult years. Beethoven turned 30 in 1800 and around this time realized he was going deaf. Beethoven's uniformly high threshold (low line on audiogram) could have been due to the onset of a middle-ear problem. Middle-ear disorders can often be corrected today through drugs or surgery. Beethoven's case is still being debated today. Eventually, Beethoven's audiogram dropped lower and lower. His genius allowed him to continue composing music after he was profoundly deaf! For example, he composed his famous *Ninth Symphony* afterwards. He completed this monumental work in 1824. The last movement incorporates a choir. Here Beethoven sets Schiller's *Ode to Joy* to music.

Fig. T-4. Hypothetical Audiogram for Beethoven in His 30s (early 1800s).

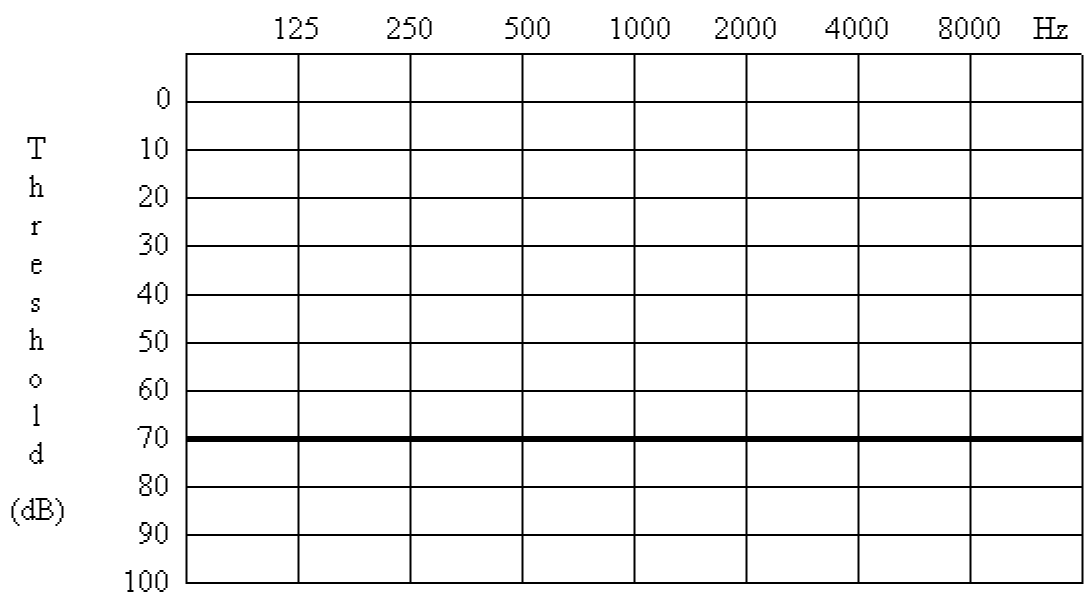


Fig. T-5 illustrates the audiogram for a rock musician. Rock music is usually played at sound levels harmful to the ear. Rock music is often 120 dB. On stage it can be 140 dB. We can actually feel inside us sound levels at about 120 dB and above. The left ear ("X" in Fig. T-5) of our rock musician is worse than the right ear ("O") because of the proximity of the speaker to that ear during practice and performance in the particular band. Loss typically begins at the high frequencies. A relevant factor may be the fact that the hair cilia responding to high frequencies are close to the oval window, the entry point for the sound. We also know from the Fletcher-Munson curves that the ear is more sensitive at the high-frequency end.

Fig. T-5. Audiogram of Rock Musician (Left Ear "X", Right Ear "O").

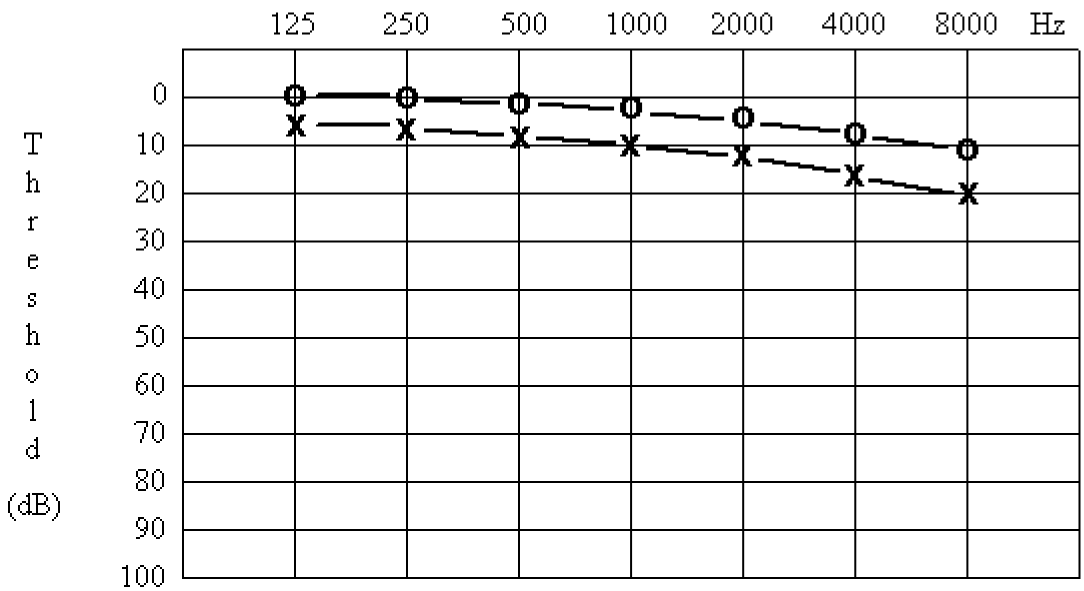
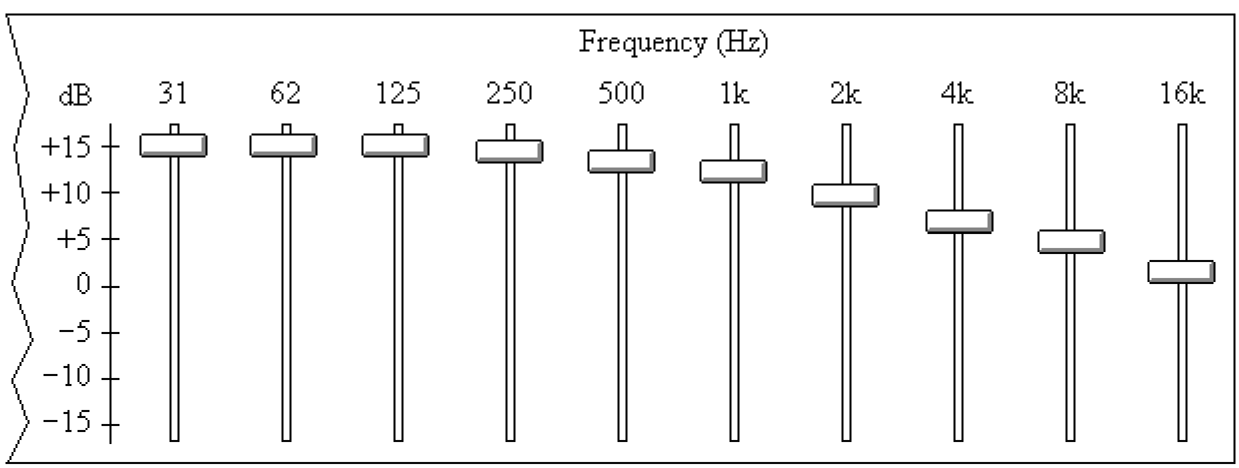


Fig. T-6. Using a Frequency Equalizer to Simulate Hearing Loss.



Cutaway of Equalizer (Right Channel).

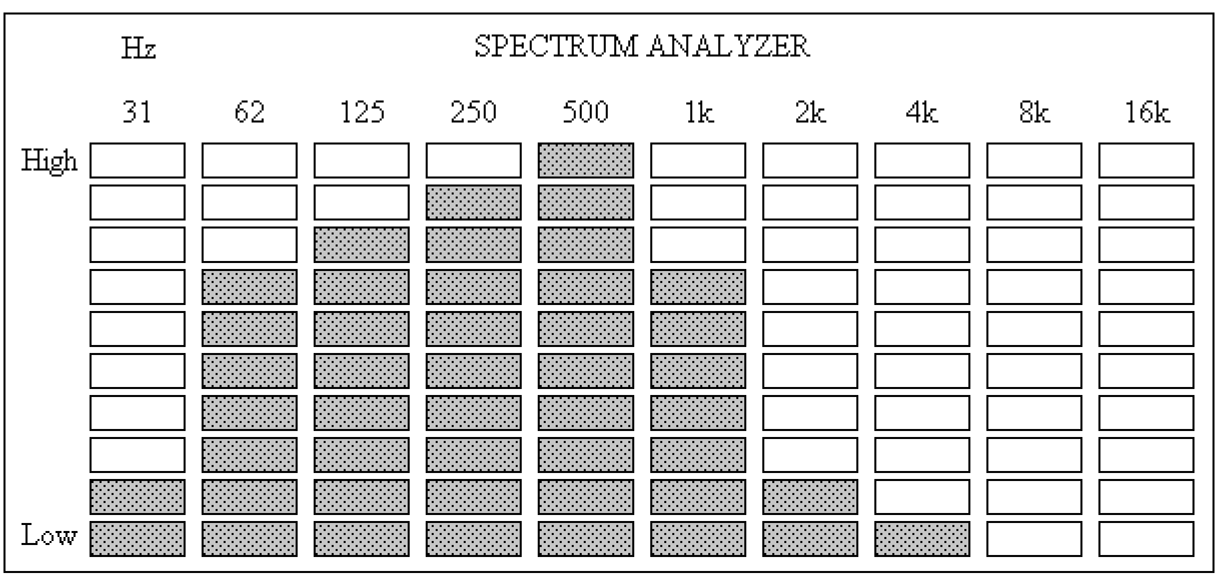
In Fig. T-6 we employ a frequency equalizer to simulate hearing loss. We studied equalizers in an earlier chapter. The equalizer is a signal-processing unit that consists of an array of active filters. The equalizer in Fig. T-6 is a 10-band frequency equalizer. The cutaway illustrates the right channel. Rather than use the equalizer to "equalize" frequencies for our specific listening environment, we use the processing unit to "unequalize" frequencies in this example.

The frequency equalizer in Fig. T-6 allows for boosting frequency bands up to +15 dB or filtering them down to –15 dB. We can set all the bands up to +15 dB. We then lower the master volume control to compensate. Now we have the choice of knocking any frequency band down to –15 dB, which is 30 dB lower than the +15 setting. Therefore, we can achieve up to a 30-dB loss. We arrange the slider controls to imitate the audiogram of our rock musician. This simulates the hearing loss.

The better frequency equalizers like our 10-band equalizer in Fig. T-6 have spectrum analyzer displays (see Fig. T-7). The spectrum analyzer is usually located in the center of the front panel. It displays with rapidly moving lights the current frequency distribution of the sound we are hearing. The sound typically changes constantly as we hear speech or music. The analyzer displays the amount we hear in each frequency band from moment to moment. A snapshot of the spectrum analyzer display appears in Fig. T-7. Note the absence of activity at the higher-frequency end of the spectrum. This is a result of our setting of the slider bars in Fig. T-6. The analyzer in Fig. T-7 reflects what a person with a hearing loss hears.

With specialized laboratory filters we can simulate more severe hearing losses. Our commercial stereo equalizer gives us control of 30 dB for the frequency bands. This is more than satisfactory for the usual use of an equalizer - to make the adjustments necessary in accommodating room acoustics.

Fig. T-7. Spectrum Anaylzer on Frequency Equalizer.



Damage to the inner ear is usually permanent. Once the hair cilia are destroyed, they cannot recover. One should always take precautions when around nearby loud sound levels. It is advisable to wear ear protection when working with power equipment such as weed eaters, lawn mowers, and power saws. You should not expose yourself to sound levels near 80 dB and above for long periods of time. Note that a rock band playing at 120 dB is 10,000 times more intense in terms of energy than an 80-dB sound. Or to look at it another way, a rock band cranking away at 120 dB is equivalent to ten thousand other bands playing at 80 dB. How do we know this from the way the decibel scale works? Before we conclude this chapter we discuss two more common forms of hearing loss.

1. Presbycusis (prez-bee-KUE-sis). *Presbycusis* is the natural hearing loss at high frequencies that accompanies age. A child may be able to hear 20,000 Hz and beyond. The young adult at age 20 (assuming no loss due to exposure to loud sounds) may hear 18,000 Hz and beyond. Then after a decade, this may drop to 15,000 Hz. In later years, it may dip to 12,000 Hz or 10,000 Hz. However, all may still pass the hearing test since that examines only to 8,000 Hz.

It is interesting to conjecture how much of our normal hearing loss is due to aging versus our exposure to higher levels of noise that accompany an industrialized society. One study found that older people at age 70 in a third-world country had hearing as good as the average 30-year-old in an industrialized society. This study suggests the importance of factors other than age.

2. Tinnitus (tin-NIGH-tis). *Tinnitus* is an internal perception of sound when there is no sound present at all. It is sometimes referred to as a ringing or a buzzing tone. Tinnitus can be caused by infections or medication. However, the most serious form is due to damaged hair cilia in a narrow frequency band. A sudden loud exposure such as a gunshot can destroy some hair cilia and damage neighboring ones that partially recover. Fig. T-8 shows an audiogram for localized destruction of hair cilia.

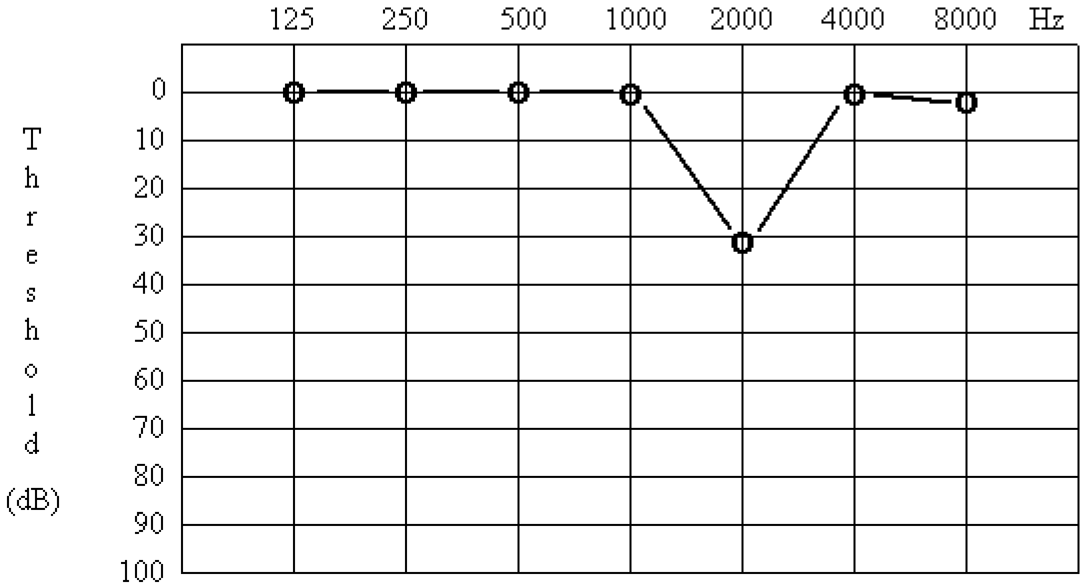
The ear described in Fig. T-8 tested normal for all frequencies except 2000 Hz. There is damage in this vicinity. The strange thing about the ear-brain system is that a person with the audiogram in Fig. T-8 may permanently hear a tone or ringing of frequencies in this region. The ear can't detect these frequencies. Yet there is an ever-present internal sensation of these. When it is quiet externally, the ringing is most noticed. These individuals have to learn to bear with this sound. They also have to learn to sleep with it.

The dip in the audiogram of Fig. T-8 is called the *acoustic trauma notch*. Perhaps the ear hears an explosion at close range. The sound is intense. Most of the hair cilia recover but a small group does not. They are permanently out of commission. The audiogram then shows the acoustic trauma notch. The individual may hear a ringing with a pitch or pitches corresponding to the damaged hair cilia along the basilar membrane.

3. Conductive Loss. When an individual has a *conductive hearing loss*, the audiogram shows a problem across the entire frequency spectrum like the audiogram in Fig. T-4.

4. Sensory Neural. A *sensory neural hearing loss* is one that involves damage to hair cells in the inner ear. This damage can involve a region of the hair cells. In the frequency regions where there is no damage, the individual will hear normally and the audiogram level will be up near 0 dB. See the audiograms in T-5 and T-8.

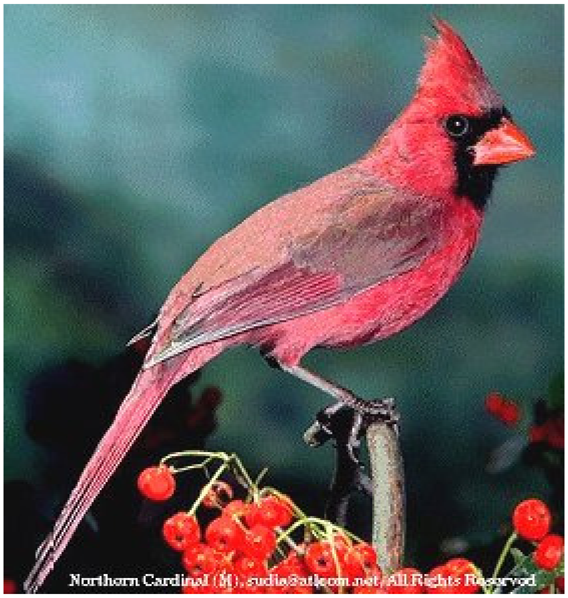
Fig. T-8. Audiogram (Notched) for an Ear with Tinnitus Caused by Destroyed Hair Cilia.



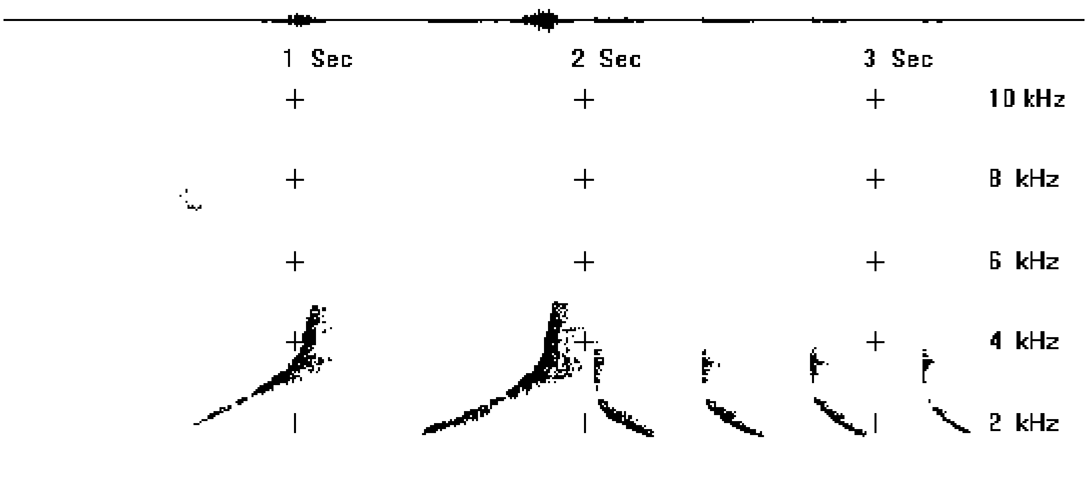
--- End of Chapter T ---

U. Spectrograms

Northern Cardinal



Courtesy Dr. Sudia



Courtesy Gregory J. Kunkel

The *spectrogram* is a special plot of sound over a short time frame. We have seen that audiograms use two axes to incorporate on a graph two basic properties of sound. The properties tested in an audiogram are amplitude (sound level) and frequency. The third basic feature of sound, the timbre, is understood by amplitudes and frequencies of the spectrum. The spectrum analyzer illustrates this. A periodic wave has a Fourier spectrum of harmonics. The frequency of the periodic complex waveform sets the base frequency for its Fourier spectrum.

However, the spectrum analyzer responds to all sounds, both periodic and aperiodic. An aperiodic sound does not have a defined frequency. So we can't express such a sound in terms of a harmonic series. Noise is a good example. It contains all frequencies. Inharmonic tones such as those produced by balanced modulation present another example. These contain non-harmonic spectral components. Nevertheless, a frequency spectrum can still describe these sounds. It just means that we need to put more in the spectrum than we have to for periodic waveforms.

The problem is this. Aperiodic tones do not remain the same as time goes on. They can be crashes, speech, music in progress, etc. For a sketch or graph to reflect these changing waves, we need to consider a time axis. This pushes us to three variables: amplitude, frequency, and time. This implies a graph in three dimensions, the axes being length, width, and height. We will see that the spectrogram is an ingenious way to reduce such a graph to two dimensions! To understand how this is accomplished, we once again return to our square wave.

Fig. U-1 illustrates the square wave we encountered earlier in our discussion of envelope shaping. Focus your attention on the sustain phase. The sustain is constant in time for the duration of the sustain phase. This is the easiest part of the wave in Fig. U-1 to consider. We will translate this section of the wave, step by step, into a spectrogram. Before considering this full-blown spectrogram which incorporates amplitude, frequency, and time information, let's look at combinations of two parameters.

First, take amplitude and time. The amplitude of a wave is a measure from equilibrium to the maximum height. The maximum heights of the square wave during the sustain phase are constant. The periodic square wave changes in displacement from crest to tough, but the boundary height (amplitude) for the crests is fixed. Fig. U-2a is a graph of amplitude (vertical axis) as a function of time (horizontal axis). For any time along the horizontal, the amplitude is seen to be the same. The value of the constant amplitude could be read on the vertical axis if there were a scale of numbers there. This amplitude is the net result of the harmonics that combine to form the overall square wave.

In Fig. U-2b we choose to plot amplitude (harmonics) and frequency. This plot is our familiar decomposition of the wave in terms of its Fourier spectrum, similar to what the spectrum analyzer on an equalizer does. The equalizer spectrum analyzer is not as detailed because there, groups of frequencies are thrown together into a series of bands. However, the equalizer shows the changing spectrum in time by providing us with spectral bars that flicker at different heights. Fig. U-2c is obtained by folding Fig. U-2a down and replacing the total amplitude with the harmonic amplitudes.

Fig. U-1. Square Wave Shaped by Envelope Generator.

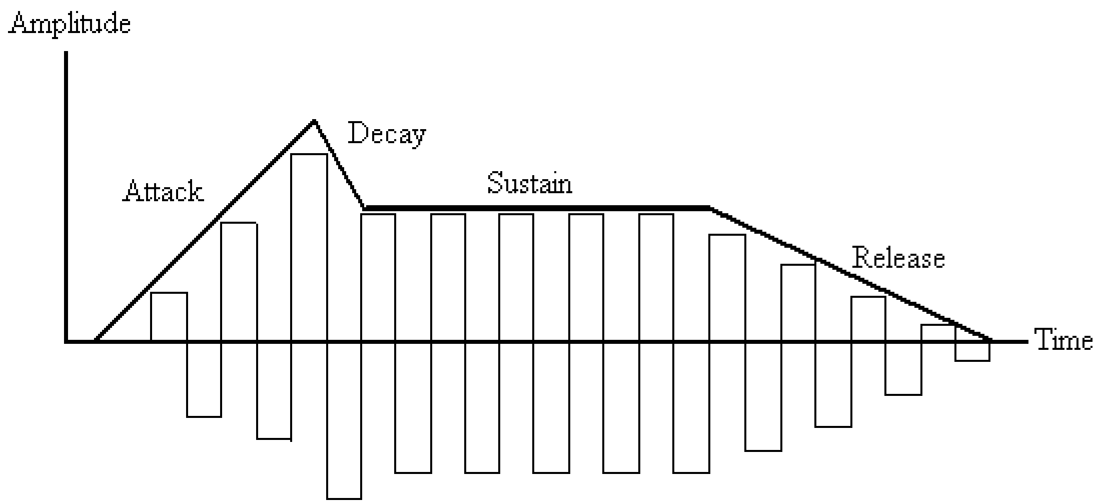


Fig. U-2d is obtained from Fig. U-2c by looking at the edges of the amplitude tops. Taller amplitudes are indicated by darker lines in Fig. U-2d. In this way, three dimensions are collapsed into two.

Fig. U-2a. Total Amplitude and Time.

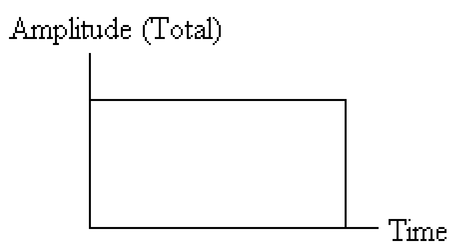


Fig. U2-b. Harmonic Amplitudes and Frequency.

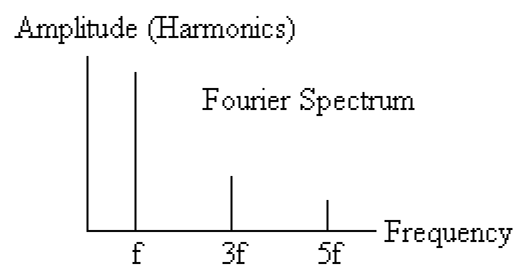


Fig. U-2c. Harmonics and Time.

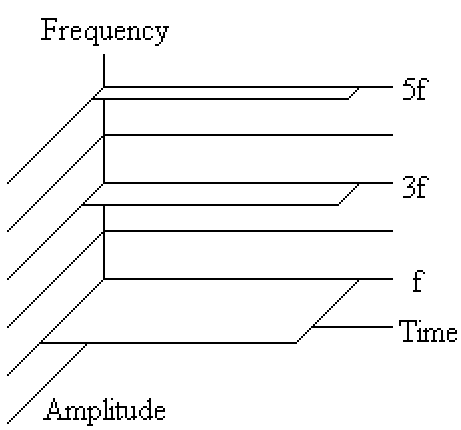


Fig. U-2d. Spectrogram.

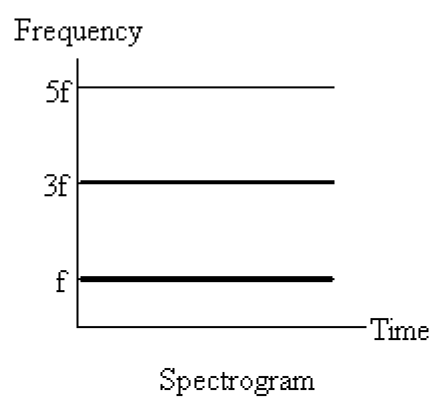


Fig. U-3 illustrates noise spectrograms. The spectrogram for white noise contains all frequencies. Frequencies are indicated along the vertical axis in a spectrogram. Therefore, all frequencies get shaded in for white noise. Shading proceeds to the right as long as the white noise is present. Moving along the horizontal axis of our graph represents movement in time.

A spectrogram of high-frequency noise describes a hissing sound. Tape hiss, which begins at around 5 kHz, is a good example. Low-frequency noise is heard more as a sigh. Let out a deep breath. Different shades of darkness represent different sound levels. The shading in Fig. U-3 is uniform for each case. White noise actually has more emphasis in the higher frequencies. The spirit of the spectrograms that follow is approximate.

Fig. U-4 is a set of spectrogram sketches. Many of these have been adapted from spectrograms made at Bell Laboratories, appearing in F. Alton Everest, *Acoustic Techniques for Home and Studio*, 2nd ed. (TAB Books, Blue Ridge Summit, PA, 1984). An excellent source of bird spectrograms can be found in Chandler S. Robbins, Bertel Bruun, and Herbert S. Zim, *A Guide to Field Identification: Birds of North America* (Golden Press, New York, 1983).

Fig. U-3. Noise Spectrograms.

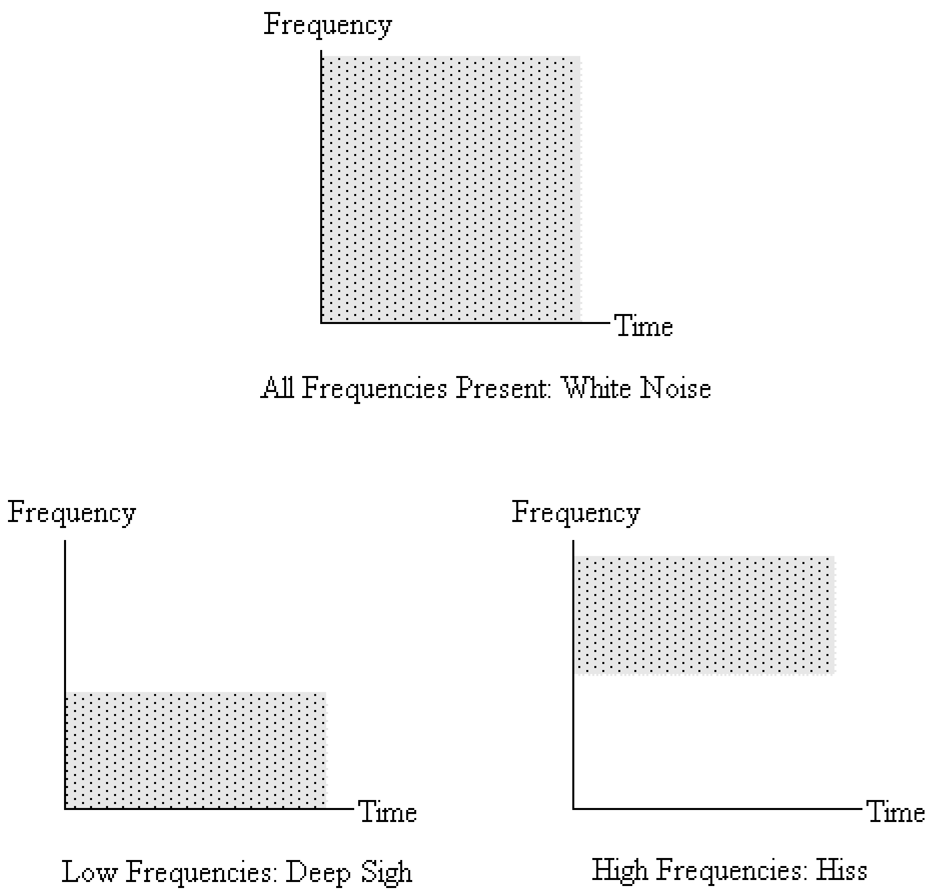
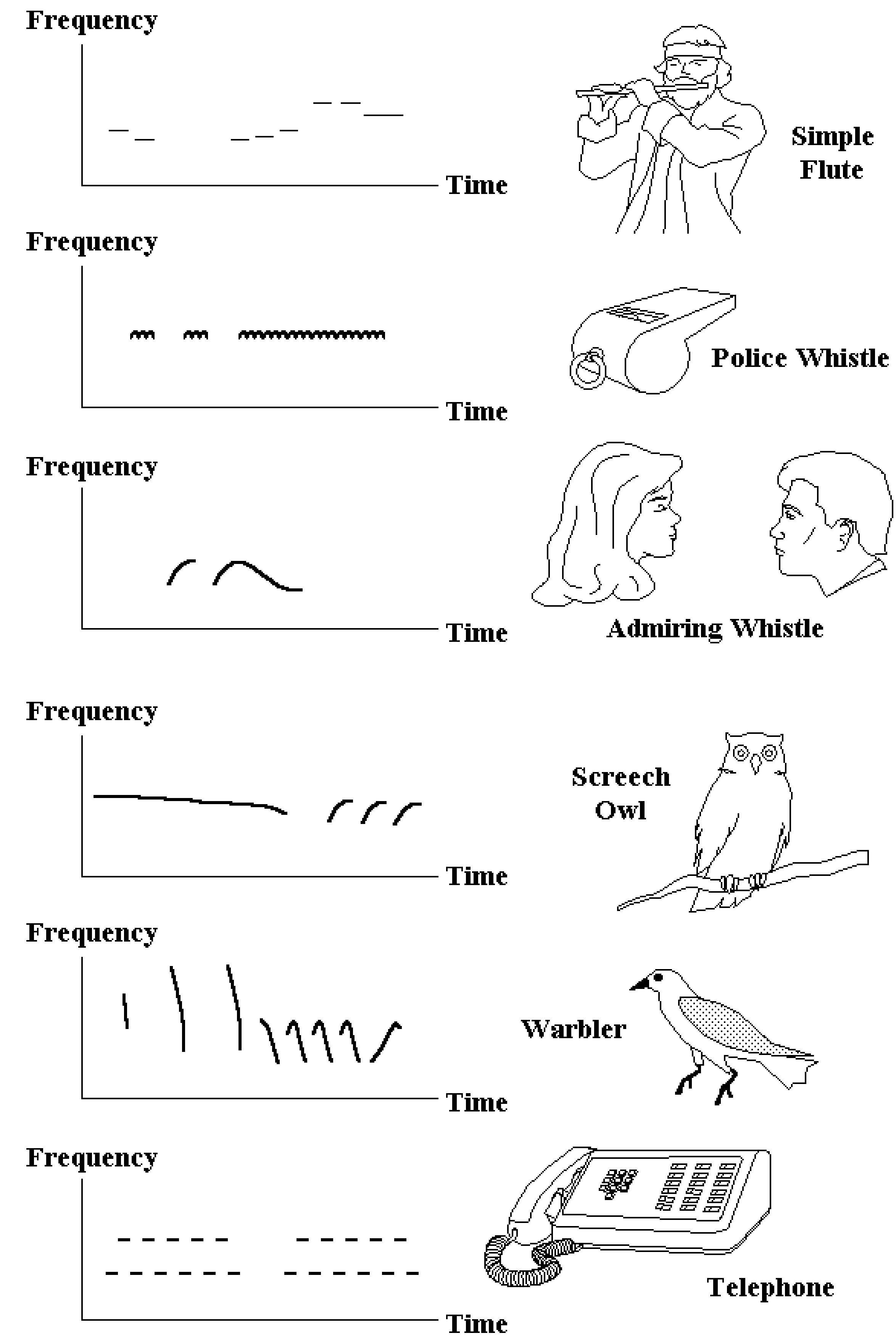
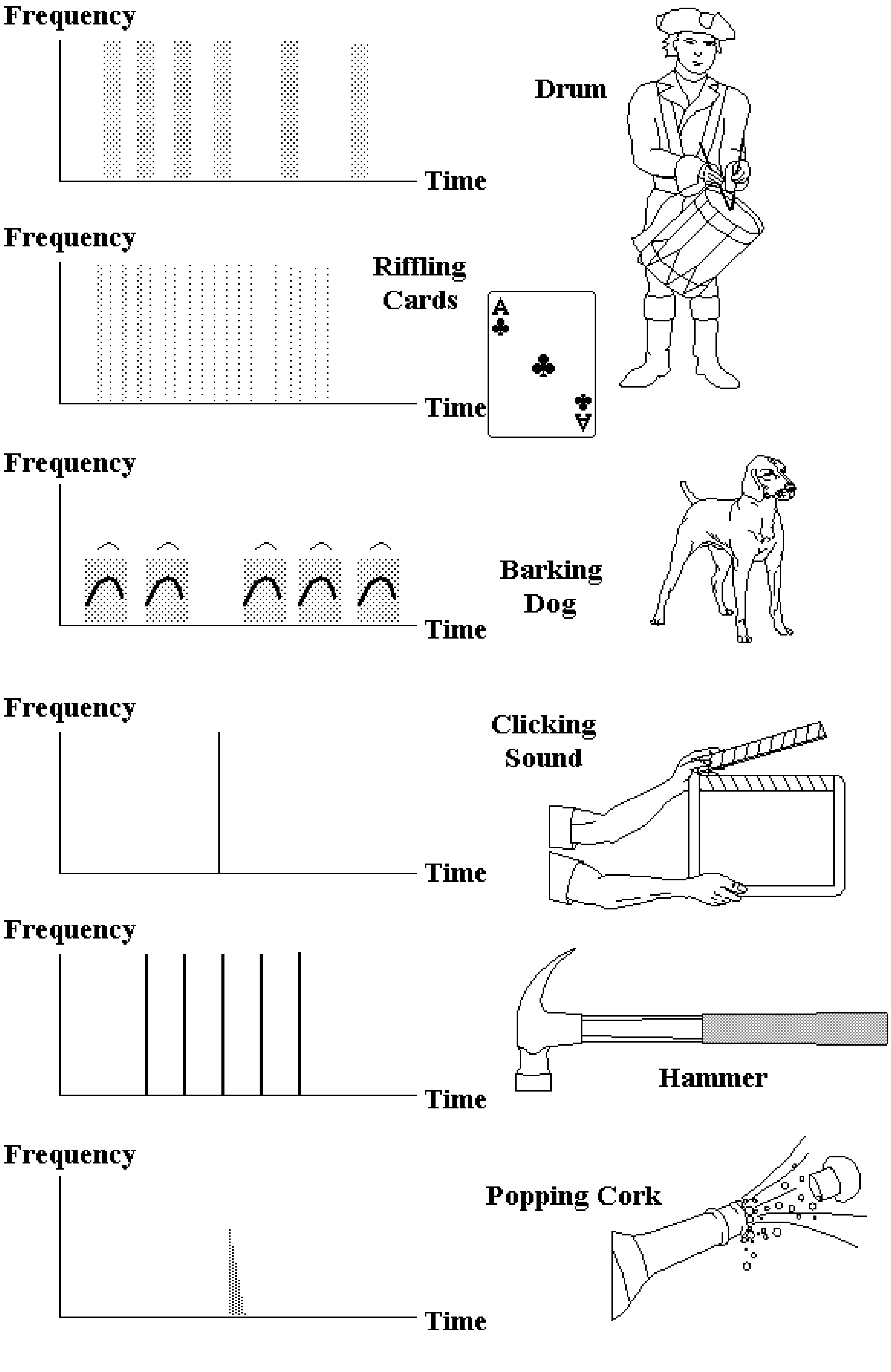
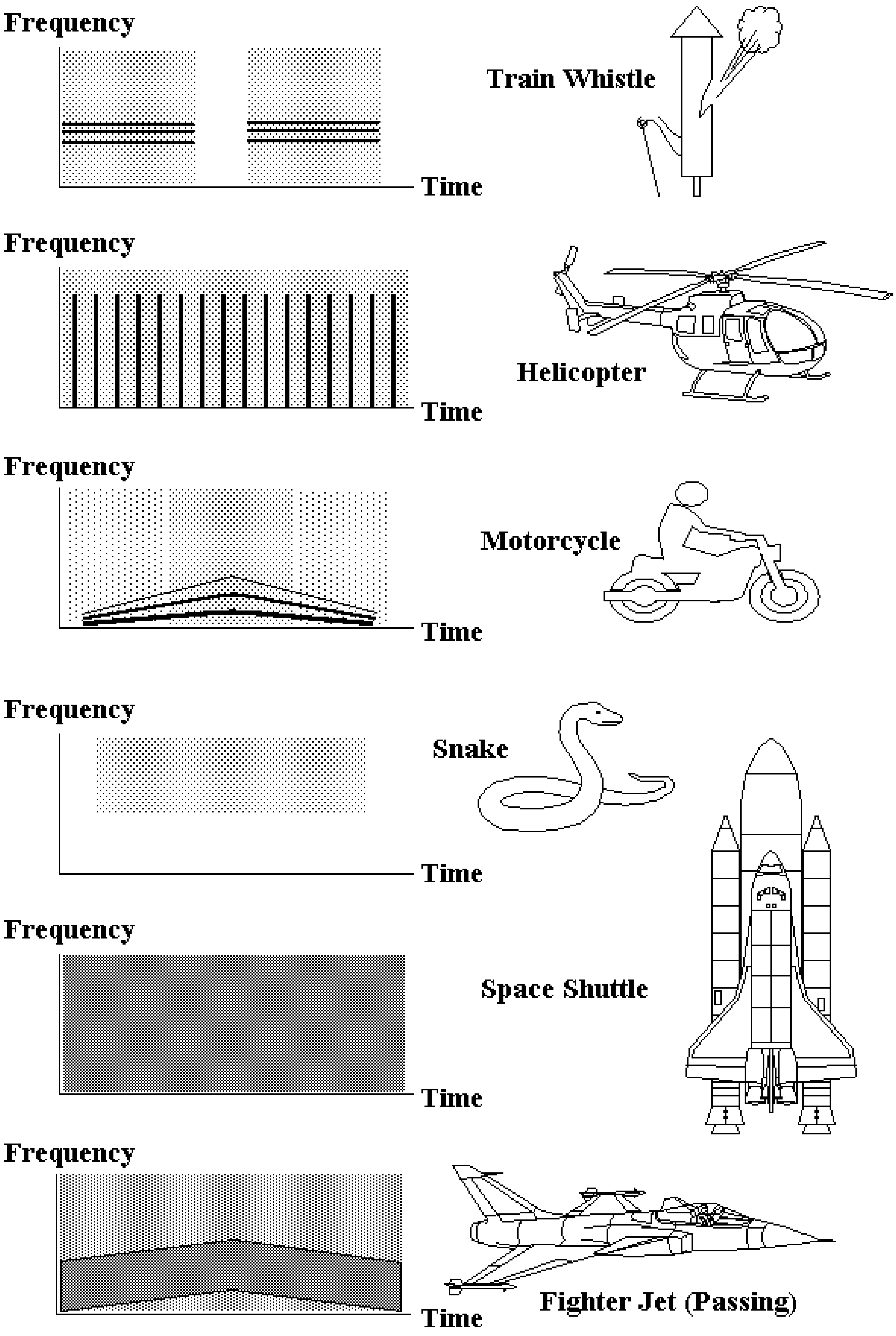
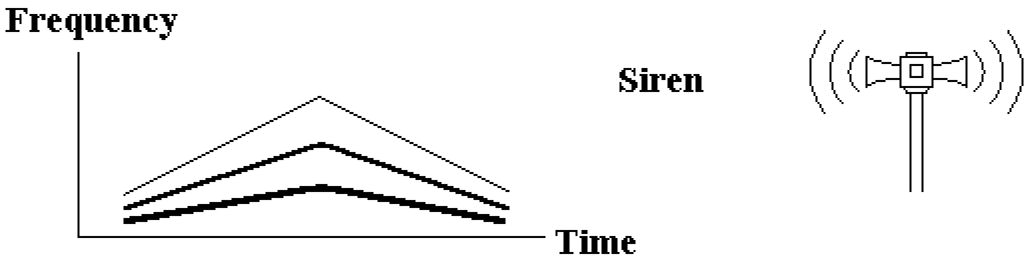


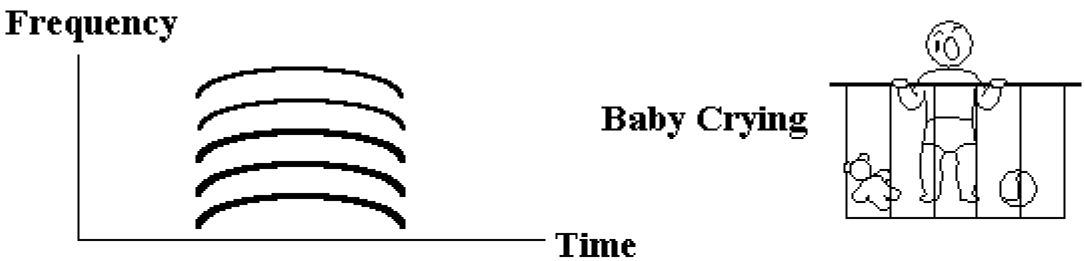
Fig. U-4. Spectrogram Sketches.

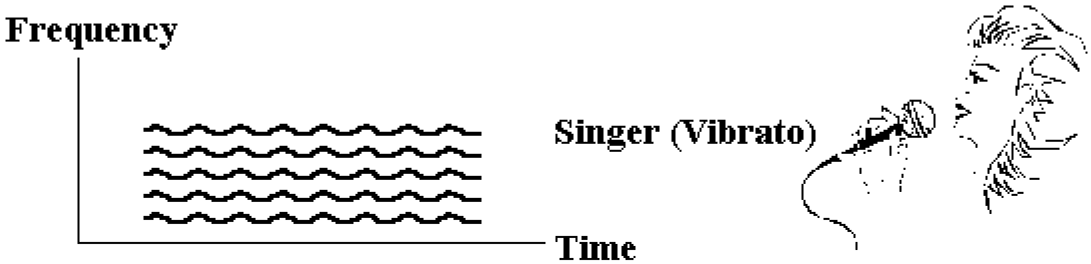


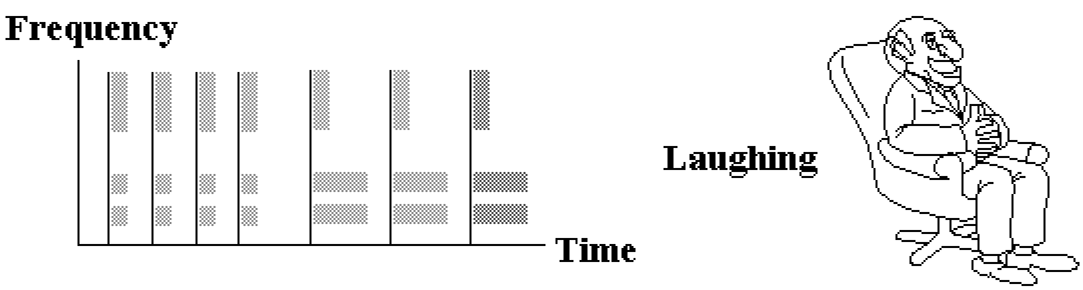

Frequency
Time
Drum
Frequency
Time
Riffling
Cards
Frequency
Time
Barking
Dog
Frequency
Time
Clicking
Sound
Frequency
Time
Hammer
Frequency
Time
Popping Cork

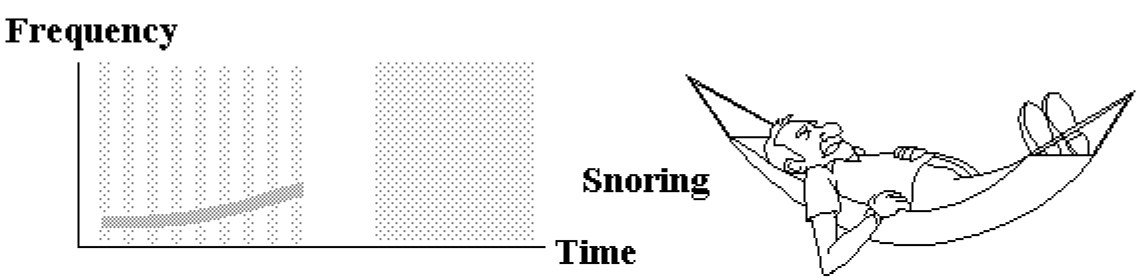

Frequency
Time
Train Whistle
Frequency
Time
Helicopter
Frequency
Time
Motorcycle
Frequency
Time
Snake
Frequency
Time
Space Shuttle
Frequency
Time
Fighter Jet (Passing)

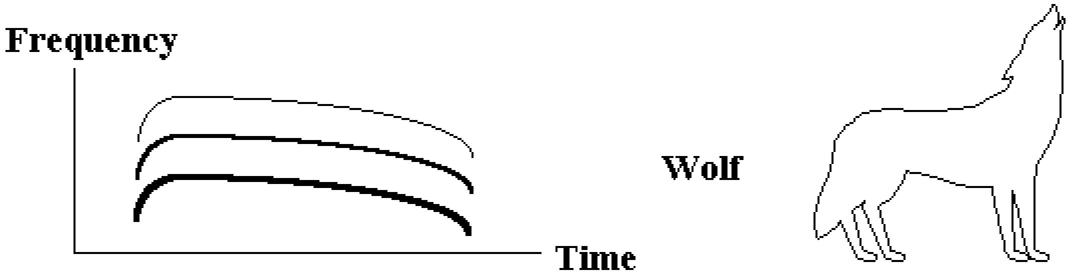

Frequency
Siren
Time


Frequency
Baby Crying
Time


Frequency
Singer (Vibrato)
Time


Frequency
Laughing
Time

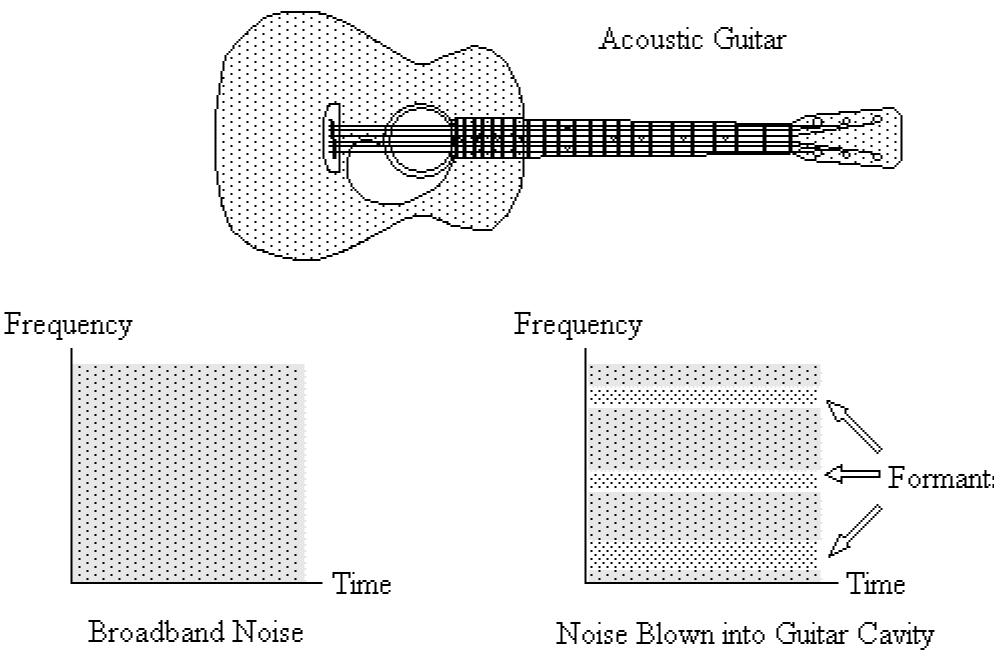

Frequency
Snoring
Time


Frequency
Wolf
Time

Specific bands of frequency are enhanced in many systems such as our voice and musical instruments. A vibrating mechanism produces frequencies. Then some of these frequencies get enhanced by resonances in the system. We consider an acoustic guitar in order to illustrate this. We perform an experiment with our guitar. We do not play it at all since we want to investigate what the cavity and wood do to the sound. We blow white noise into the guitar cavity and analyze the resulting sound. The guitar modifies the white noise. It enhances certain frequency bands due to resonance. The inner cavity of the guitar has more complex resonances of air vibration than a long narrow pipe.

The wood of the guitar also vibrates in response to the noise. The wood resonance vibrations depend on the characteristics of the wood. Certain bands of frequencies of the white noise are amplified. The spectrogram of the final sound is the white-noise spectrum with the enhanced frequency bands. See Fig. U-5. These enhanced regions are called *formants*. The formant regions are determined by the shape and size of the guitar cavity, the type of wood and other materials used in the construction.

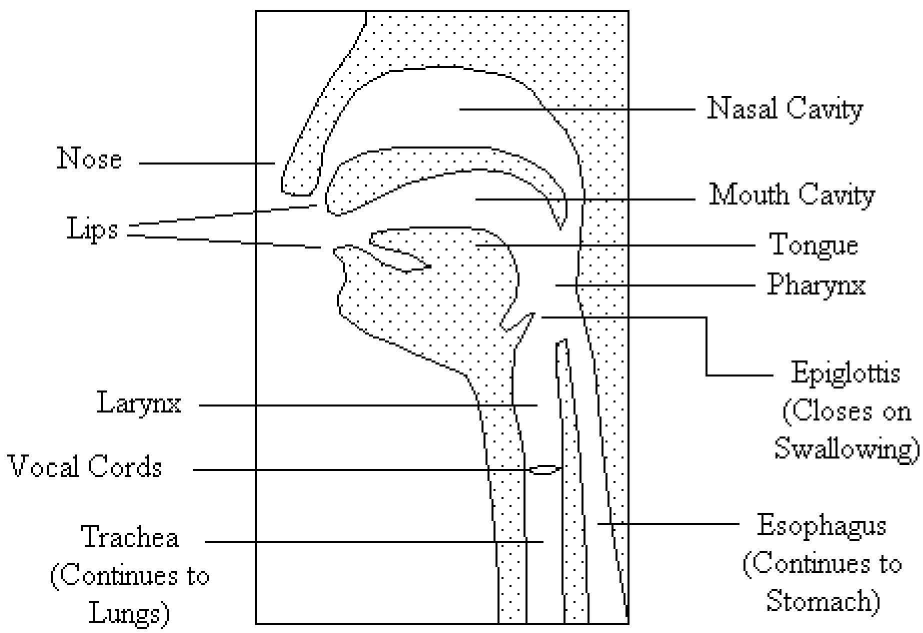
Fig. U-5. Guitar Formants.



During the normal playing of the guitar, the strings are used to produce the initial vibrations of sound. The strings produce periodic tones. Each pitch has a fundamental and associated overtones. Those overtones falling in formant regions of the guitar get enhanced. The tone becomes richer. The final sound is unique to the make of the guitar. A similar analysis can be made with the violin. The final rich spectrum, dependent on the formants, is once again, a unique timbre.

The human voice has formants that vary, depending on the shape of the vocal system. The cavities involved are the larynx, pharynx, mouth, and nasal cavity (see Fig. U-6). These depend on the size of the person. Also, the individual can control a considerable portion of the vocal system in order to change these dramatically. Colds and congestion influence the nasal cavity. The vibrations originate at the vocal cords. The sound is enhanced by the neighboring cavities in a similar way the acoustic-guitar cavities bring out the sound of a vibrating guitar string. Formants are relevant.

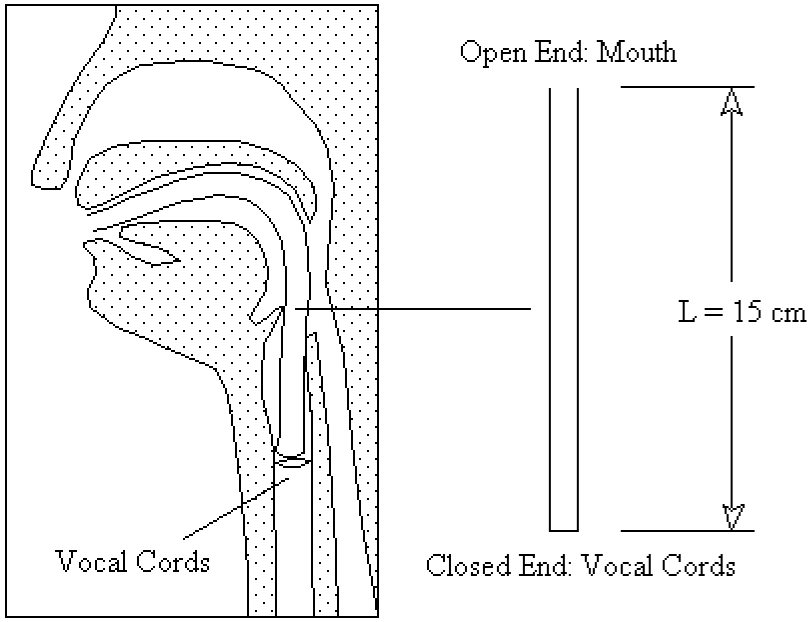
Fig. U-6. Human Vocal System.



We once again turn to biology. It is obvious from Fig. U-6 that the human voice system is quite complex. We would like to have a simple model. Here is where physics comes in. We can take the vocal cords to be the closed end of a pipe, the pipe extending all the way to the mouth opening. See Fig. U-7. Look upward, place your thumb on your Adam's apple and touch your lips with your forefinger. Take a ruler and measure this distance between your thumb and extended forefinger. The result varies from person to person, roughly correlating with height. A typical value is around 15 cm.

A singer can produce a range of pitches with considerable control over the voice system. We can determine very approximately where the formant regions are in the frequency spectrum. The formant frequencies are related to the natural frequencies of vibration in the cavity of the vocal tract. In our model we approximate this tract as a closed pipe. Therefore, the vocal formants will be set by the resonances of a closed pipe. These resonance frequencies are odd harmonics.

Fig. U-7. Modeling the Vocal Track as a Closed Pipe.



The steps for determining the resonance frequencies of the closed-pipe vocal tract are given below. The key issue is finding the fundamental. We know that the resonances are the odd harmonics. Once we know the fundamental frequency "f" as a number, the odd harmonics 3f, 5f, 7f, and so on are then known explicitly. These are the vocal formant regions of our model.

1. The length of the closed pipe. $L = 15$ cm

2. The fundamental wavelength. One quarter wave is fitted to the length. $\frac{1}{4}\lambda = L$

   The wavelength is four times the length. $\lambda = 4\ L = 4 \times 15\ \text{cm} = 60\ \text{cm}$

3. The frequency. Wavelength x frequency = speed. $\lambda f = v = 340$ m/s

   Express sound speed in centimeters per second since wavelength is in cm (must be consistent). $(60\ \text{cm})\ f = 34{,}000$ cm/s

   Divide 34,000 by 60 to find frequency. $f = (34{,}000 / 60) = 3400 / 6 = 1700 / 3$

   $f = 1700 / 3 = 567$ Hz (but we will use 500 Hz since our model is very approximate).

   The next two modes are then 1500 Hz (3f for H3) and 2500 Hz (5f for H5).

4. Vocal Formants. The first three vocal formants are 500 Hz, 1500 Hz, and 2500 Hz.

Vocal formants depend very much on the shape of the cavities in the vocal system. For the ideal closed narrow pipe, we find approximately 500 Hz for the first vocal formant. An adult's first vocal formant can vary a couple hundred hertz either way from 500 Hz, depending on the sound produced. The second formant can vary over an even wider range. Since vocal formants are dependent so much on the sounds we pronounce, the best way to determine vocal formants in specific cases is from spectrograms. Some professional singers have been found to have a strong formant region between 2500 and 3000 Hz. This formant is called the *singer's formant* or *singing formant*. It assists in projecting the voice. Is it a coincidence that trained singers can achieve this formant, which happens to be right where our ears are most sensitive (consulting the Fletcher-Munson Curves)?

Simplified spectrograms are given in Fig. U-8 for six vowels spoken with a steady voice at a fundamental of 150 Hz. The steady voice producing a vowel sound is virtually a periodic tone. We can then analyze the spectrum in terms of harmonics. The equally-spaced horizontal lines are the harmonics H1, H2, H3, etc. for the 150-Hz wave. They have frequencies 150 Hz (1f), 300 Hz (2f), 450 Hz (3f), 600 Hz (4f), etc. The spacing between harmonics is simply 150 Hz since each time you rise to the next harmonic, you add another 150 (i.e., f). Note the richness of the overtones. Some of the sounds have over 20 harmonics.

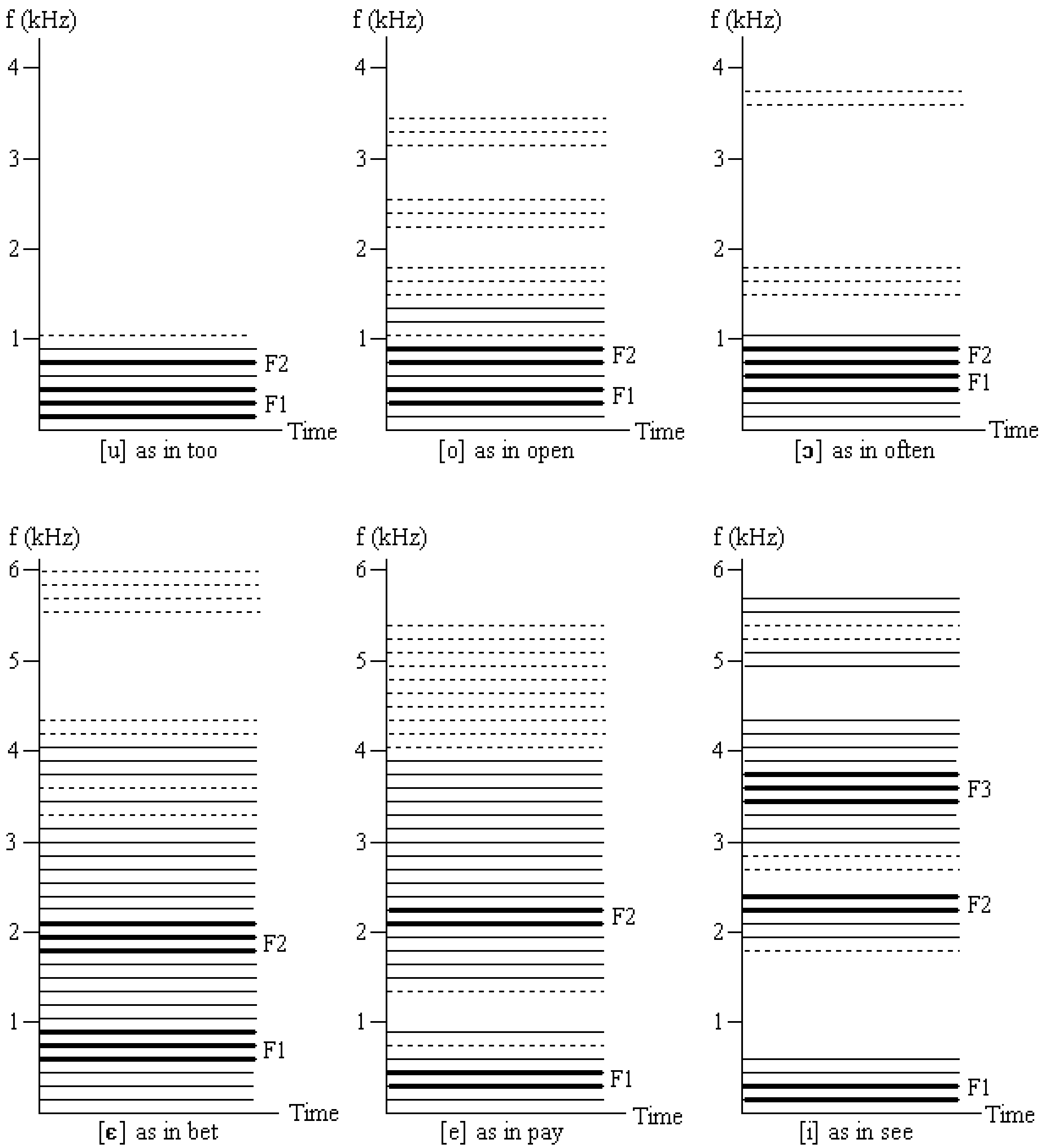
The more intense harmonics are darker. Sometimes a harmonic is not intense enough to see. The formants are indicated as F1, F2, etc. These are enhanced harmonics due to the shape of the internal cavities of the vocal system. The first enhanced region is called the first formant region and so on. There is no correlation in many cases to the formants of our ideal pipe since the voice can change formants by inner shaping of resonance cavities.

However, note that the [i] sound has formant regions near 500 Hz, 2500 Hz, and 3500 Hz. These correspond to formant regions of our model (the 1st, 5th, and 7th harmonics of the simple closed-pipe model of the voice system). But the formant corresponding to the 3rd harmonic (1500 Hz) has been suppressed. The 3rd harmonic has been filtered out by the voice system. The shape of the system for this vowel makes it difficult for the 3rd harmonic to resonate. The 2nd formant region (F2) for [i] is near the 5th harmonic of the "closed-pipe" (2500 Hz), which is close to H16 of the 150-Hz wave being produced (making H16 and also H15 more intense). We know 2500 Hz is close to H16 because the frequency of the 16th harmonic is 16 x 150 Hz = 2400 Hz.

The symbols employed as pronunciation guides for the sounds in Fig. U-8 are from the International Phonetic Alphabet (IPA), devised in the late 1800s. The IPA consists of basic units of speech called *phonemes* (FOE-neams) that can be put together to indicate how any word in almost any language is pronounced. The IPA is used by teachers of the hard-of-hearing, speech pathologists, actors, singers, radio and TV announcers, and others. A singer armed with the IPA can know how to sing a phrase in a foreign language. The IPA enables you to pronounce words in other languages you don't understand, if you have mastered the basic sound units (about 40) of the IPA.

The data in Fig. U-8 comes from doctoral research in an area where psychology, physics, and medicine converge. You should consider it an achievement that you can understand and appreciate such advanced research data.

Fig. U-8. Spectrograph Sketches for Six Vowel Sounds Spoken with a Steady Voice.

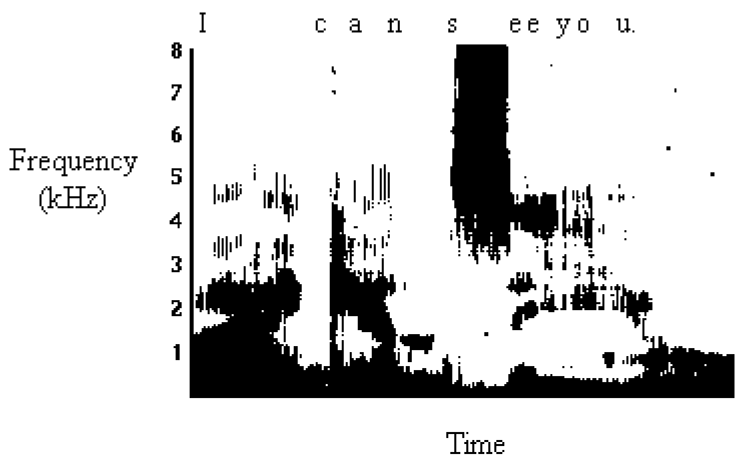


Sketches are adapted from spectrograms found in Colin Painter, *An Introduction to Instrumental Phonetics* (University Park Press, Baltimore, 1979).

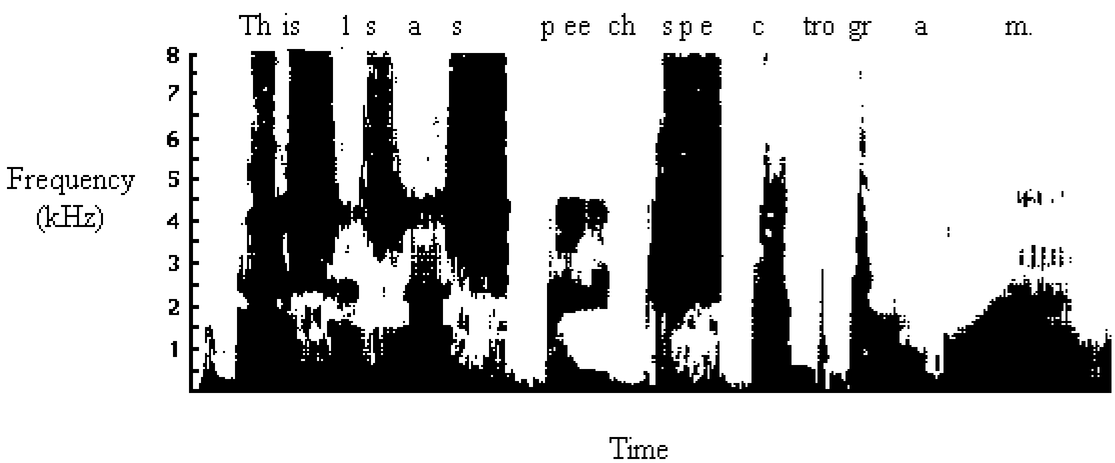
Two speech spectrograms can be seen in Fig. U-9. The pictures are examples of those produced by a sound *spectrograph*. The spectrograph analyzes the sound by playing it over and over again in order to scan the frequencies up to 8 kHz. A drum spins and an ink pen records the presence of frequencies on a paper rolled on the drum. When finished, the unwrapped paper becomes the spectrogram.

Fig. U-9. Speech Spectrograms.

Spectrogram of the spoken sentence: I can see you.



Spectrogram of the spoken sentence: This is a speech spectrogram.



Reference: F. Alton Everest, *Acoustic Techniques for Home and Studio*, 2nd ed. (TAB Books, Blue Ridge Summit, PA, 1984).

--- End of Chapter U ---

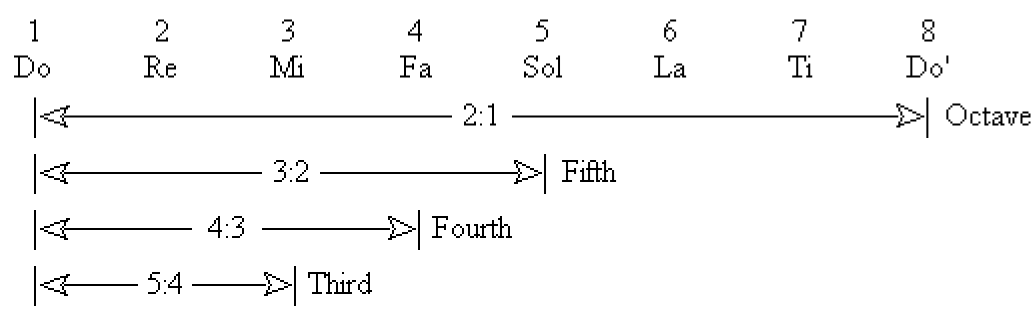
V. Musical Temperament

We saw in the last chapter how the human voice system can produce a rich variety of sounds. Earlier, we learned how engineering electronics can generate sound and modify it. We also investigated the storing of sound on media such as records, tapes, and CDs. In the final chapters we turn to the musical production of sound with traditional instruments, until the very last chapter. It is fitting that most of the final phase of this text be dedicated to that which historically has provided our current culture with the rich esthetic experience of musical art.

The Major Scale

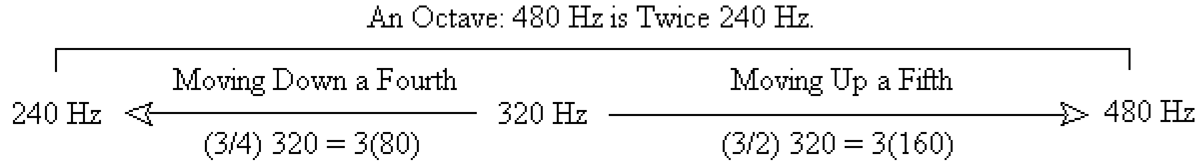
The major scale is depicted in Fig. V-1 below. The frequency ratios are indicated for a perfect octave, perfect fifth, perfect fourth, and perfect major third. These intervals provide for the most consonant combinations of tones after the unison. The octave is so close to the sound of the unison (two identical notes sounding) that we proceed to the next ratio (the 3-to-2) to serve as a foundation for a system of music theory, the *cycle of fifths*. Movement by fifths is very pleasing. The traditional way to end a piece is to move from the fifth (*dominant*) to the root (*tonic*), achieving a sense of coming home and completion. Such a harmonic change is called a *cadence*.

Fig. V-1. The *Just Major Scale* and Most Consonant Tones.



Jazz musicians employ the *cycle of fifths* often. Once the author had a heated debate with a sax player of 15 years who claimed it was the *cycle of fourths* instead. Finally the author realized that the *cycle of fourths* is in a sense the same as the *cycle of fifths*. If you move up by a fifth and call the new note *Do*, moving down by a fourth gives the *Do* that is an octave lower. Since *Do* defines the key, you have a transition to the same key in each case. Fig. V-2 illustrates this connection. To move up by a fifth (3:2), multiply by 3/2. To move down a fourth (4:3), use 3/4 instead of 4/3 (in order to get a fourth lower).

Fig. V-2. Relationship Between Moving Up a Fifth and Moving Down a Fourth.



The Twelve-Tone Scale

We are going to analyze the major scale, which is the standard eight-tone scale we become accustomed to in grade school. Our analysis will show that the "perceived jump" from note to note is not the same. Rather than rely on our ears to tell us this, we will reach this conclusion from mathematical analysis, using arithmetic. We will conclude that there is room for more notes in the major scale since some of the jumps are about twice as great as others. We can stick in an extra note here and there so that the perceptual jump from each note to the very next is the same. Recognition of this fact by hearing the notes is the experimental approach. Both methods agree. Good science requires that theory support experiment and vice versa.

Fig. V-3 takes us through the analysis step by step. First we start with the major scale. We list the perfect ratios for the 8 tones relative to the first note *Do*. This version of the major scale (the just major scale), as noted in an earlier chapter, is also called the *just diatonic scale*. We next express the frequency ratios as fractions. Note that these fractions are greater than one. A fraction is simply one number divided by another. As an example, consider the ratio 3:2. We write this ratio as 3/2, getting ready to multiply our base frequency "f." The base frequency is the frequency we choose for *Do*. Therefore, we see (3/2)f in Fig. V-3.

Next we take 240 Hz for *Do* as we did before. This makes the arithmetic easier. For the fifth, (3/2)f, the result is (3/2)240 = 3(120) = 360 Hz. These have been worked out before in the text when we first established the just diatonic scale. You might want to review that chapter at this time. The frequencies reproduced in Fig. V-3 are the same. To make our analysis even simpler, we divide each frequency by 10. Then, 240 becomes 240/10 = 24. A zero is knocked off each frequency. This particular realization of the just diatonic scale is too low to be practical, but it serves our purpose. As an exercise, start with 24 Hz and work out all the other frequencies using the appropriate ratios.

The next row compares adjacent tones. The first two frequencies (24 Hz and 27 Hz) give a ratio comparison of 27/24. The next row expresses this ratio in reduced form: 27/24 = 9/8. Note the importance of ratios. The perceived jumps in frequency are based on ratios. Remember our steps along the basilar membrane are organized by ratios (equal steps of 3.5 mm for each 2:1 frequency ratio). If our 9/8 were 8/8 instead, we would have the same note. The 9/8 has an additional 1/8 beyond unity (i.e., 1). We use this 1/8 to denote the "extra contribution," the extra part beyond 1. We then consider 1/8 and 1/9 essentially the same size. If one pie is cut into 8 pieces and a second pie cut into 9 pieces, could you tell the difference between a 1/8-size slice and a 1/9-size slice? We also replace 1/15 by 1/16 since these are even closer to being the same size.

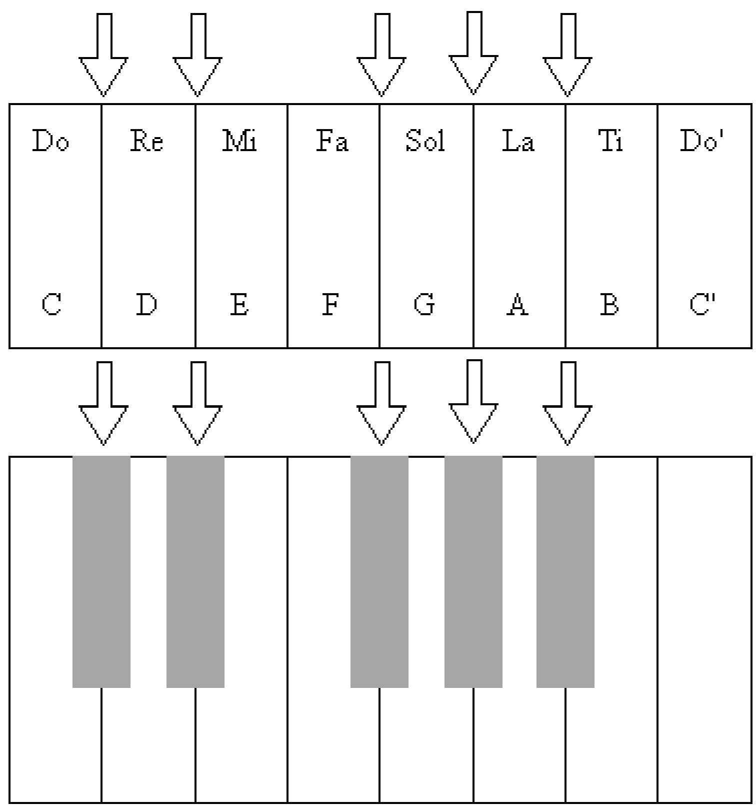
We are now ready to draw the big conclusion. There are 5 bigger "pieces of the pie" and 2 smaller pieces. We cut the 5 big pieces in half so every new piece resulting has the same size. This introduces 5 more tones. These are the 5 black keys appearing on the keyboard for each octave. We have "derived" the black keys!

Fig. V-3. Adding the Black Keys to the Eight-Tone Scale.

The eight-tone scale.	Do	Re	Mi	Fa	Sol	La	Ti	Do'
Just Diatonic Scale.								
Perfect ratios.	1:1	9:8	5:4	4:3	3:2	5:3	15:8	2:1
Ratios to multiply f.	f	(9/8)f	(5/4)f	(4/3)f	(3/2)f	(5/3)f	(15/8)f	(2/1)f
Take f = 240 Hz.	240	270	300	320	360	400	450	480 Hz
Divide by 10 for low example.	24	27	30	32	36	40	45	48
Adjacent ratios.	27/24	30/27	32/30	36/32	40/36	45/40	48/45	
The adjacent ratios reduced.	9/8	10/9	16/15	9/8	10/9	9/8	16/15	
Extra contributions.	1/8	1/9	1/15	1/8	1/9	1/8	1/15	
Approximate extra contributions.	1/8	1/8	1/16	1/8	1/8	1/8	1/16	

There are 5 big 1/8-steps (called whole steps) and 2 small 1/16-steps, half the size (called half steps). We can split the 5 whole steps so that going from note to note is a half step in each case. This introduces 5 more notes, the "black keys."

The last note here is considered to be the first note of the next octave. It is the starting note for the pattern to repeat an octave higher. Therefore our new scale has 12 notes.



Now we have the 7 notes from *Do* to *Ti* and 5 additional notes. This gives us a *twelve-tone scale*. We consider the note an octave higher than *Do* (i.e., *Do'*) as the beginning of the next 12 tones. For a moment, retreat to our major scale. Each step which has an extra contribution of 1/8 is called a *whole step* or *whole tone*, while the steps with the extra 1/16 contribution are called *half steps* or *semitones*. In the new twelve-tone scale, all the steps are equal. You make 12 half steps in going from *Do* to the *Do* that is an octave higher (*Do'*). To step by whole tones on the twelve-tone scale, just skip a note at each step.

Our prior restriction to the major scale limits us in playing songs since many tunes use the additional notes we have added. We can give a formula for the major scale. From your starting note you proceed to by first making a whole step. You make a whole step by skipping the very next note (whether it is a black key or white key) and land on the note after the one you skip. For a half step, you go to the very next note. The formula for the major scale is: whole-whole-half-whole-whole-whole-half (see Fig. V-4). Note that this formula consists of two whole-whole-half sections joined by a whole step or connection in the middle. The total number of steps in the scale is 7.

Centuries ago some mystics found profound meaning in the formula for the major scale. We noted that Pythagoras was a mystic and mathematician. Nearly 2000 years later, Kepler (1600) likewise felt that mystical secrets of the universe were to be found in numbers and formulas. In a sense, the professional physicist is not too far from this point of view. The secrets to understanding nature can be expressed in beautiful mathematical form. But the mystics went further. The seven steps of the scale meant much more. The number 7 was considered sacred. We see this theme often in different historical settings: the 7 days of the week, the early 7 celestial bodies of the crystalline spheres (Moon, Mercury, Venus, Sun, Mars, Jupiter, and Saturn), the 7 sacraments, the 7 colors of the spectrum, and the 7 steps of the musical scale.

Some mystics gave meaning to the two half steps in the major scale. To them, these represented a break from the usual progression of whole tones. They applied this in everyday life by saying that all efforts following from an original aim (*Do*) can get sidetracked in two key places. One is after we get started and the other is at the very end.

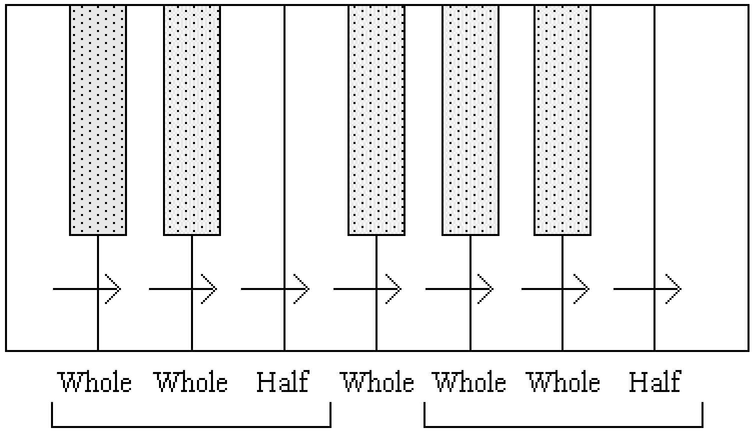
Have you every worked on a goal that proceeded smoothly for awhile (*Do-Re-Mi*) and then you reached a challenge (the half step from *Mi* to *Fa*)? Most people quit at this point. The cleaning of the room does not get completed, the term paper remains unfinished, you don't read the entire novel.

However, if you apply a conscious effort at the challenging point (*Mi-Fa*), you go on smoothly again for awhile (*Sol-La-Ti*) until the very end. You can still fizzle out. The modern-day version of this law is *Murphy's Law* (originating in electrical engineering): "If something can go wrong, it will." And it usually does so at the least expected places - after we just start and think things are going well, and right when we think we are about to be finished.

A scale with no half steps is the *whole-tone scale*, which Debussy liked (see Fig. V-4). Another is the *double-diminished scale*, popular in jazz improvisation. Abbreviations are used for whole (W) and half (H) steps in the example for the double-diminished scale in Fig. V-4.

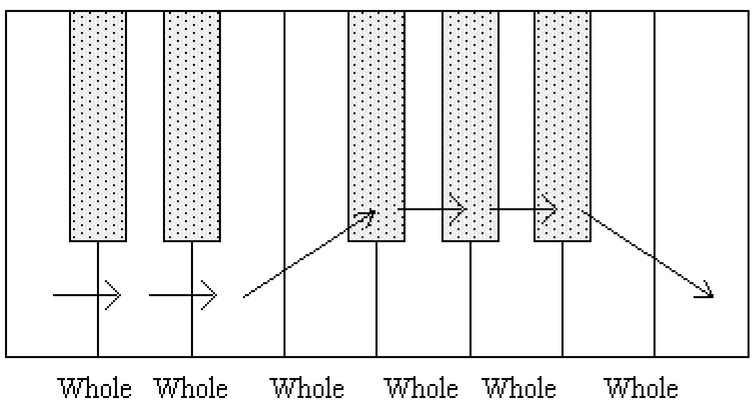
Fig. V-4. Three Formulas to Obtain Three Different Scales.

Major Scale: Do-Re-Mi-Fa-Sol-La-Ti-Do'.

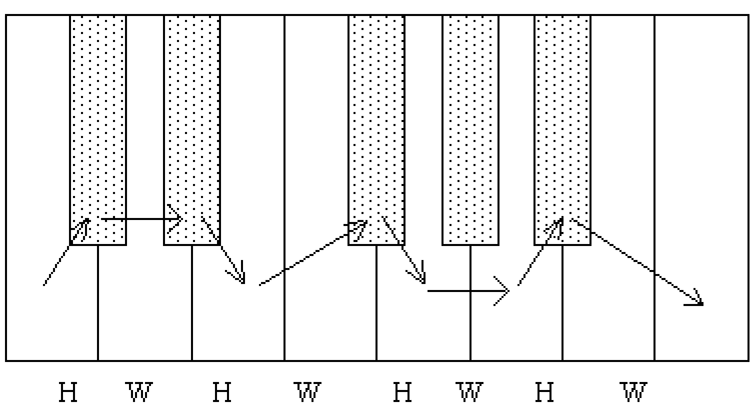


Two Similar Sections Joined by a Whole Step in the Middle.

Whole-Tone Scale, Used Often by Debussy (1862-1918).



Double-Diminished Scale, Used in Cocktail Lounges.



There are many other scales with different formulas. Go to a piano and pick out the scales we have described: the *major scale*, the *whole-tone scale*, and the *double-diminished scale*. Use Fig. V-4 for assistance. Then try the *natural minor scale* (whole-half-whole-whole-half-whole-whole). The *chromatic scale* is the scale obtained by playing all the notes (half, half, half, etc.). The modern composer Schoenberg (SHERN-berg) liked the complete *twelve-tone scale* and devised lines using the tones once and only once. These lines are called *twelve-tone rows*.

They sound strange (modern) since a note can't be used more than once in the musical line. Try writing a sentence that uses each letter of the alphabet once and only once. It's impossible. But you can write a sentence that uses all the letters of the alphabet: "The quick brown fox jumped over lazy big cats." Can you think of another that uses less than 37 letters? How about just a group of words, following the modernist Schoenberg, where the meaning can be cryptic?

Equal Temperament

Earlier in our treatment of whole steps, we considered that 1/8 and 1/9 were essentially equal. We did the same for 1/15 and 1/16 (the half steps). We would like to have more precise definitions for half steps and whole steps. Historically the difficulty with tuning to perfect ratios presented problems. If you start with a different key to play a scale, the frequency ratios are not preserved in the new key. Temperament refers to the specific choices we make for the frequencies in our scale. The uniform manner in which frequencies are chosen in *equal temperament* is described in this section.

The whole steps in our just diatonic scale have slightly different frequency ratios since really 1/8 is not exactly the same as 1/9. The equal-tempered scale solves this problem by making all half steps precisely the same in such a way that by the time you reach the octave, the frequency has doubled. The perfect-frequency ratios are given up in favor of equal-frequency ratios between adjacent tones. The only perfect interval remaining is the octave. The fifths are no longer perfectly 3:2, the fourths no longer perfectly 4:3, etc.

The remaining task is to find the magic ratio for the half step satisfying the criterion that 12 half steps give a perfect octave (2:1). This problem is identical to determining the annual interest rate (applied once yearly) needed so that your money doubles in 12 years. Each half step is analogous to each year. The growth in frequency from our example of 240 to 480 takes 12 steps. We want the growth rate to be the same from step to step. Growth in frequency is analogous to growth in money.

In banking, if you save your money, you get back the original amount plus interest. If your annual interest rate is 8% for the year, $100 earns you $8 at the end of the year. We assume interest is applied once yearly. Better deals apply interest earnings more than once a year. For example, if interest is compounded quarterly, every 3 months one applies 2% interest. This is one-fourth of the yearly rate so technically it is still 8%. However, applying the appropriate percentage more often results in a better deal for you, because your money starts to grow after 3 months.

Consider an interest rate of 10% and an initial amount of $100. After the year, the interest is $10. Our new amount is $110. We have the original $100 plus the $10 interest. We leave the $110 in for another year. The interest the following year is 10% of $110. This is $11. Note that we not only receive an interest of $10 for the $100, but an additional $1 for the $10 of interest we made the first year. We are getting interest on interest in the second year. This is good news. To get the next year amount just multiply the last amount of money by 1.1.

Table V-1 gives growth patterns for several interest rates and a starting value of $240. Values in the table are rounded off. This is unlike what banks do. They keep fractional amounts over a penny. In the table, everything is carried from year to year in the calculator, rounding off annual amounts for table entries. Which case approximates the *just-diatonic* growth pattern at the far right of the table? We can cleverly answer without looking at the table. We know from before that each half step corresponds to approximately an excessive 1/16. This fraction is about 0.06 in decimal form. That is 6% interest!

Table. V-1. Growth of $240 Until Twice the Amount is Reached.

Just-Scale Amounts (Which interest percentage comes closest?)

	Interest Rates						
Time (or Step)	4%	6%	8%	10%	12%	14%	
Before 1st Year	240	240	240	240	240	240	240
After 1st Year	250	254	259	264	269	274	
After 2nd Year	260	270	280	290	301	312	270
After 3rd Year	270	286	302	319	337	356	
After 4th Year	281	303	327	351	378	405	300
After 5th Year	292	321	353	387	423	462	320
After 6th Year	304	340	381	425	474	527	
After 7th Year	316	361	411	468	531		360
After 8th Year	328	383	444	514			
After 9th Year	342	405	480				400
After 10th Year	355	430					
After 11th Year	369	456					450
After 12th Year	384	483					480

Our only demand is to match the 480 after the 12th year.

Using the 6% interest for the frequencies means that our new frequency is the old one plus 6%. To obtain the 6%-increase we multiply the old frequency by 0.06 or 1/16. The new frequency (original + 6%-increase) can be obtained by multiplying the old frequency by 1 + 0.06, i.e., 1.06. However, we see that by the time we get to twelve steps or years, the answer is a little too high. In Table V-1, the amount for 240 after 12 years is 483 instead of 480. So 1.06 is slightly too high.

We want to know that special number that we can multiply something by 12 times and arrive at twice our starting value. Let's call the special number "a." Then, multiplying 12 times, using the dot-symbol "·" for the multiplication symbol, we have

$$a \cdot a \cdot a \cdot a \cdot a \cdot a \cdot a \cdot a \cdot a \cdot a \cdot a \cdot a = 2$$

Mathematicians call "a" the 12th root of 2. The answer is a little less than 1.06 as we expect. It is given below.

$$\sqrt[12]{2} = 1.059463\ldots$$

This is the number we multiply any frequency by to get the next one a half step higher. It corresponds to an interest of a little over 5.9%. It is almost 6%. How accurate in theory does this number need to be? This question is answered by perceptual psychologists. They study how close two frequencies need to be before we judge them to be the same. They research stimuli in general such as loudness, color, or taste. The difference between two stimuli where the stimuli cease to appear the same is called the *just noticeable difference* (*JND*). The just noticeable difference for frequency depends on where along the audio spectrum we are being challenged to make the match.

Notice that when high notes are played on the piano, it is more difficult to tell them apart compared to notes in the middle range. On the average, the JND for frequencies in the range of musical instruments is about 1 Hz. With a starting frequency of 1000 Hz, in order to get accuracy of 1 Hz, we can use 1.059. This gives 1059 Hz. The author once wrote a computer program to generate all the equal-tempered frequencies on the piano. One should always use the best accuracy possible for the 12th root of two, find all the frequencies, then round to the nearest hertz at the end.

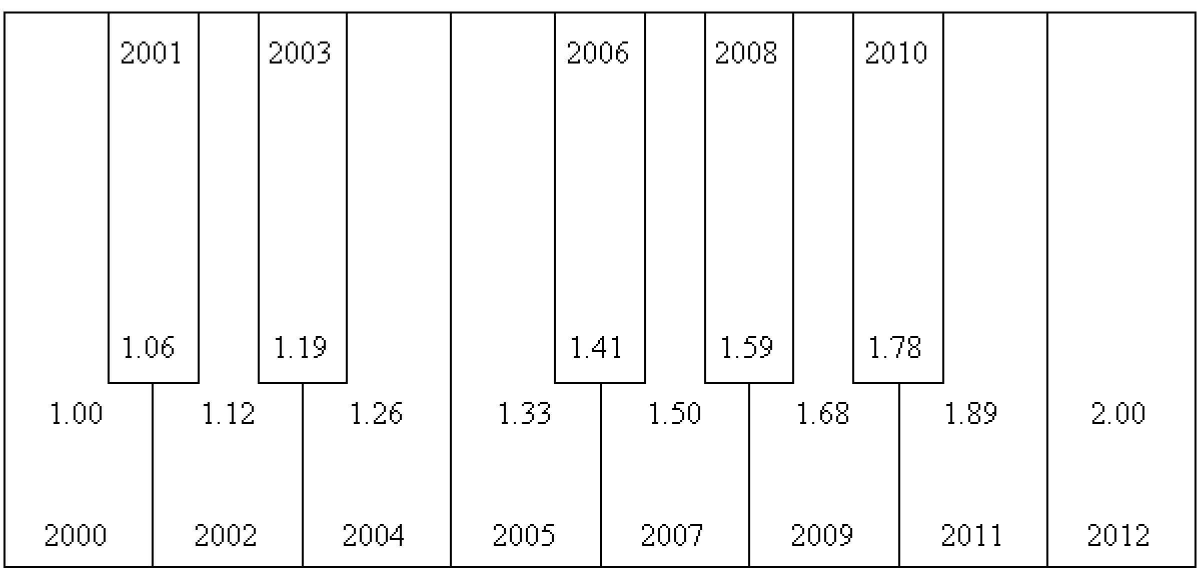
Piano tuners establish an equal temperament for the middle octave, then pretty much tune the other notes proceeding by octaves. Traditionally tuners have used their ears. Some today use electronic devices for assistance. There is debate as to which can provide the best tuning. There is no question that a well-trained tuner can do a superb job without electronic aid. Some argue that electronic devices get in the way when you strive for quality tuning.

Transposing

We would like to construct an equal-tempered set of 12 tones from which any other equal-tempered set, starting on a different first note, can be determined. So we choose 1 Hz for our first note and proceed. If you want to start with 300, then you multiply each tone of the sample set by 300. Banks do this also with loans. For example, they give values based on $1000 for home loans. If you want to borrow $105,000 and find out your interest payment, then you multiply the interest payment based on $1000 by 105.

To maintain an analogy with finance, let 1 stand for $1.00. Then, for our annual interest rate of 5.9%, we earn almost 6 cents interest after one year. The bank will give you 5 cents interest and keep the fraction of a penny. But you might say 90% of a penny is so close to a penny. Too bad! The banks round down. They make much money this way. However, when we apply the interest formula below, we round in the standard way. After one year, we have $1.06. We obtain this by multiplying our $1.00 by 1.059. The 1 in 1.059 gives us back our original dollar and the 0.059 part gives us the 5.9 cents of interest. For the 12 years or steps, we multiply by 1.059 twelve times. Actually more decimal places were used in the calculator and numbers rounded off last to get the results in Fig. V-5.

Fig. V-5. Growth of $1.00 Savings for 12 Years at 5.9463% Interest.



We can now apply Fig. V-5 to our favorite starting point, 240 Hz. Or, we might say we would like to put $240 in the bank and leave it there for 12 years. Our money will double to $480 after 12 years. We multiply each of the values in Fig. V-5 by 240 to find out how much our money is worth year by year. We "transpose" the values in Fig. V-5 to a new starting point. Musicians transpose to other keys musically by playing in different keys rather than calculating frequencies. The concept is similar.

The eight notes of the just diatonic scale are compared with equal-tempered frequencies in Table V-2. A calculator was employed that used an extremely accurate value for the 12th root of two for the equal-tempered values. Then, frequency values were rounded off to the nearest one tenth of a hertz. The starting frequency for each scale was set to 240 Hz. Since the equal-tempered scale preserves octave ratios of 2:1, the ending notes are in exact agreement. The "black keys" of the equal-tempered scale are not listed since they are not present in the just diatonic scale.

Table V-2. Just Diatonic Scale Compared to Equal-Tempered Frequencies.

Degree in Scale	1	2	3	4	5	6	7	8
Name of Degree	Do	Re	Mi	Fa	Sol	La	Ti	Do'
Just Diatonic (Hz)	240	270	300	320	360	400	450	480
Equal-Tempered (Hz)	240.0	269.4	302.4	320.4	359.6	403.6	453.1	480.0

The equal-tempered scale has a perfect ratio only at the octave. Perfect intervals in other places are lost. Remember that the JND in frequency at midrange is about 1 Hz. Therefore, the frequency difference of 0.6 Hz for the 2nd degree of the major scale is hardly noticed. However, the 2.4-Hz difference for the 3rd degree exceeds the 1-Hz tolerance. The 3rd degree on the equal-tempered scale is slightly sharp (i.e., higher in frequency) relative to a perfect major third. The 1st and 8th degrees for the just diatonic and equal-tempered scales are in perfect agreement. The 2nd, 4th, and 5th degrees are very close. The 3rd, 6th, and 7th degrees are not as close.

To compare the scales in the next octave, double every frequency in Table V-2. Consider the 6th degree. In Table V-2, the difference is 403.6 − 400 = 3.6 Hz. For the scale an octave higher, the difference between the 6th degrees is twice this: 807.2 − 800 = 7.2 Hz. Things are worse. This trend continues. However, we lose our keen frequency discrimination in the highest octave of the piano.

Musical Range

The piano is an excellent guide for studying musical range. The piano has the largest range of all instruments except for some organs. The letter names for the notes are very convenient in discussing musical range. See Fig. V-6 for the letter names of the major scale. Piano students learn them by first remembering that "C" is the key to the left of the pair of black keys. The pattern repeats on the piano. Find any place where there are two black keys grouped and there is a "C" to the immediate left. The "D" is the key that's between the two black keys, "E" is to the right. The "F" is to the left of a group of three black keys and so on. If you are not a musician, next time you are near a piano, see how fast you can hit all the "D" notes on the piano from bottom to top.

Fig. V-6. The Letter Designations for Notes.

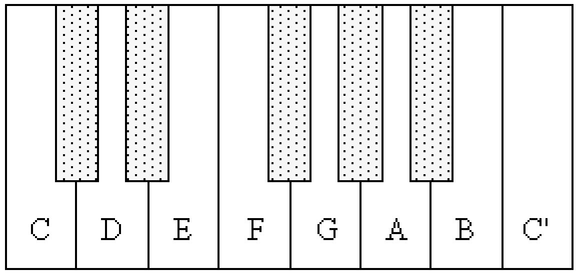


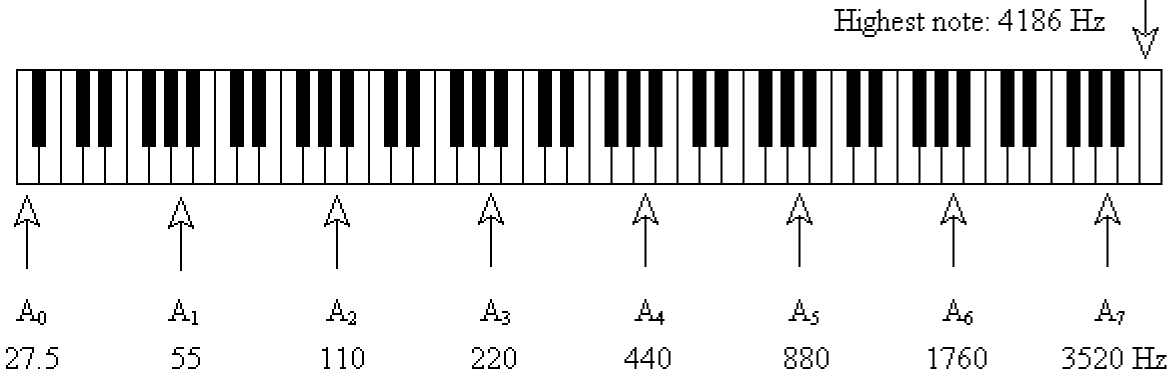
Fig. V-7 displays the entire piano keyboard. The first note (lowest) on the piano is an "A." We use the convention that calls the first "C" on the piano C_1. The note A_1 is the "A" in the major scale that starts with C_1. Therefore, we refer to the very first "A" on the piano as A_0.

Note that there are 7 complete scales starting with a "C." Since we added the black keys to the tones of the major scale, each complete scale is the chromatic scale with 12 notes. You can play a major scale starting on any of these 12 notes if you follow the formula for a major scale. There are 12 such scales, 7 for the white keys and 5 for the black keys.

Pianists spend hours learning to play these rapidly with both hands. The 7 octaves of the 12-tone chromatic scale give us 7 x 12 = 84 keys. There are three additional keys at the bottom and the sole "C" at the top. The uppermost "C" supplies the highest 12-tone scale with its resolution *Do*. Therefore, there are 84 +3 + 1 = 88 keys on the piano.

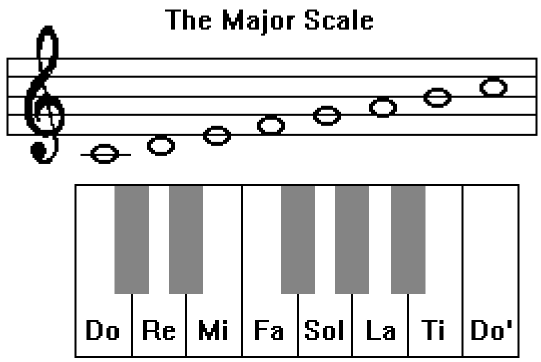
The last item remaining is to fix the frequency of one note, to get started. Equal-temperament does the rest. The standard is to set A_4 to 440 Hz. This is the note the first violinist hits on the piano when the orchestra tunes up for a piano concerto. You double 440 to get A_5 and halve 440 to get A_3 and so on in Fig. V-7. Multiplying 440 Hz by 1.059 gives the next highest key, the black key to the right of A_4. We refer to this key as $A^{\#}_4$. The symbol "#" is called a sharp symbol. We say "A-sharp-4." Dividing 440 by 1.059 gives us the black key to the left of A_4, called $A^b{}_4$ or "A-flat-4." Note that $A^b{}_4$ is also the same as $G^{\#}_4$. *Sharp* means move one half step to the right; *flat* means move one half step to the left. The piano frequencies range from 27.5 Hz (A_0) to 4186 Hz (C_8).

Fig. V-7. The Piano as Reference for Musical Range.



An Exercise

What are the frequencies for the “just diatonic” major scale if the first one is 120 Hz? Be sure to be able to work these out using the ratio for each of the intervals.



Part	Degree of Scale	Name of Degree	Frequency (Hz)
a	1	*Do*	120
b	2	*Re*	
c	3	*Mi*	
d	4	*Fa*	
e	5	*Sol*	
f	6	*La*	
g	7	*Ti*	
h	8	*Do'*	

--- End of Chapter V ---

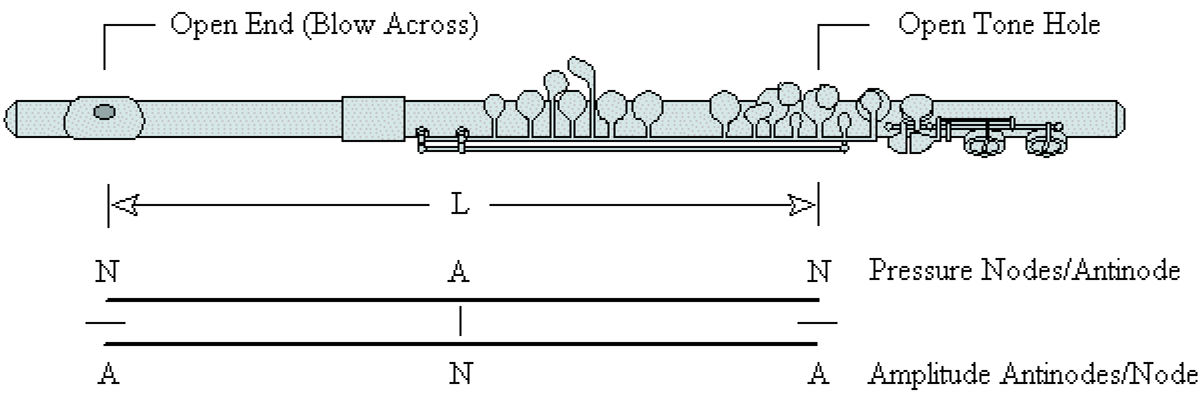
W. Woodwinds

We now take up the study of the musical instruments. We consider in this chapter and the next two, the three main sections of the orchestra: woodwinds, brass, and strings. We choose this order because it corresponds to the manner in which they appear on an orchestral score of music. In the chapter on strings, we discuss percussion, which is also very important.

Flutes

1. Flute: The Physics. The flute is an open pipe. Fig. W-1 illustrates an orchestral flute. The player blows across the left end. It serves as one of the open ends of the pipe. The other open end is determined by which tone hole is open. The effective length of the open pipe changes. Fig. W-1 leads us step by step in arriving at the length needed for a mid-range tone, E_4. The value for the speed of sound at freezing temperatures is used to simplify the arithmetic. This is okay since we are after an approximate answer anyway. It is interesting to discover that the length is approximately 50 cm, a length that humans can carry.

Fig. W-1. Flute.



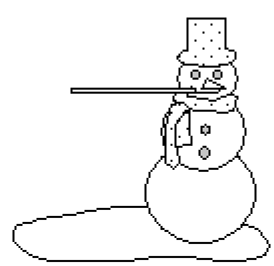
For the Fundamental: $\lambda / 2 = L$ $\lambda f = v = 340$ m/s

Example calculation: What is L for E_4, i.e., 330 Hz?

Step 1. Find Wavelength from Frequency: $\lambda f = 330$ m/s (Use speed at 0°C, 32°F.)

λ (330 Hz) = 330 m/s

$\lambda = 1$ m



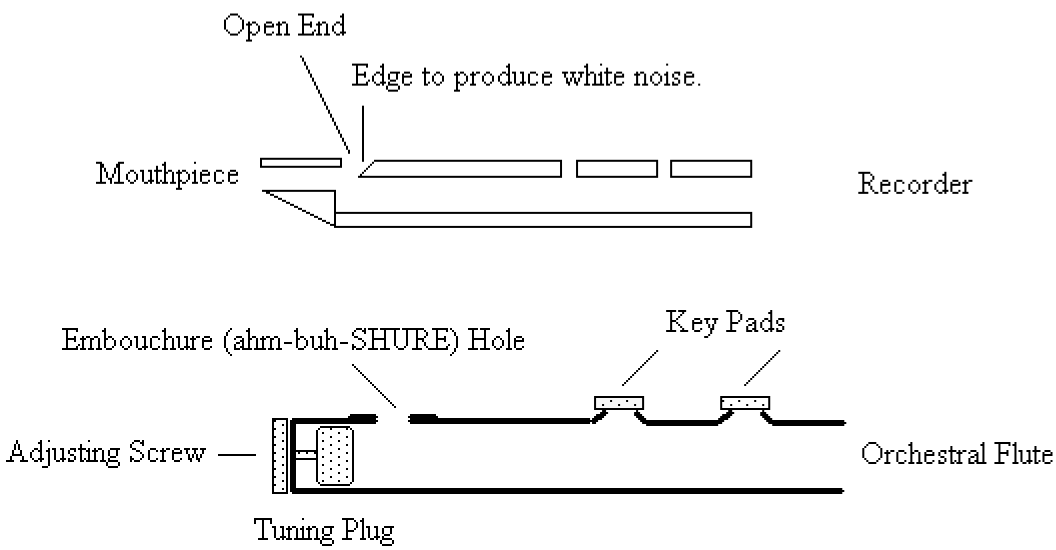
Step 2. Find Pipe Length from Wavelength: $L = \lambda / 2$

$L = (1 \text{ m}) / 2 = 0.5 \text{ m} = 50 \text{ cm}$

2. Flute: Producing the Tones in the Scale. Twelve effective lengths of the pipe are needed to obtain the 12 tones in the chromatic scale. One length is achieved by using the far end of the open pipe. Then, 11 tone holes are needed to produce shorter "effective pipe lengths" to obtain the other 11 pitches. The orchestral flute has a few additional holes to make playing easier. The tone holes provide for notes within one octave. Driving the pipe to resonate at higher harmonics extends the range. Fig. W-2 illustrates an older flute (the recorder) and the orchestral flute.

Many common variety flutes resemble the recorder. You blow into one end and use your fingers to cover holes. To open a hole, a finger is released. The number of holes needs to be reduced unless you have 12 fingers. The author had a toy flute with 8 holes in 5th grade. The additional tones in the chromatic scale could be produced by using the technique of covering half a hole.

Fig. W-2. Recorder and Orchestral Flute.

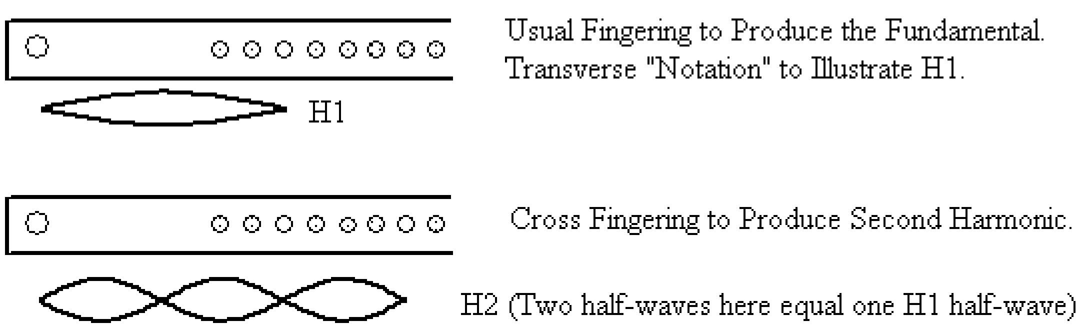


The source of energy for the flute is supplied by the jet of air made by the player. Blowing across the embouchure of an orchestral flute produces white noise. The pipe goes into resonance. Putting the mouth completely over the mouthpiece in the case of the recorder presents a problem. The air stream travels straight through the tube like a straw, unless there is an edge with another opening to produce turbulence. The air flow coming into contact with the edge produces the resonance.

The orchestral flute has a tuning plug that enables the player to slightly change the length of the flute. The change in pipe length shifts the frequency slightly. The resonance is affected by the blowing pressure, air-jet length, and area of lip opening. The player can control the sound level in this way. The pitch can either be slightly modified by these or drastically by exciting a higher harmonic.

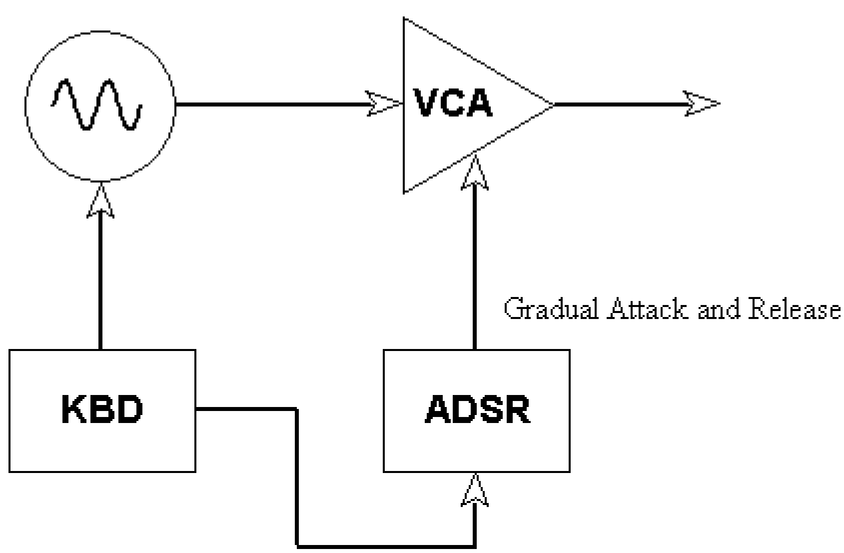
3. Flute: Extending Musical Range. The flutist can also obtain higher pitches by carefully arranging for additional openings to encourage more nodes. This technique is called cross fingering. In Fig. W-3 there is an extra key pad raised. The node positions support a pipe resonance at the second harmonic (H2). Here 2 half-waves fit into the same length (lower diagram) compared to the one half-wave of H1 (upper diagram). This technique and the earlier ones mentioned allow coverage of the 3 octaves C_4 to C_5, C_5 to C_6, and C_6 to C_7.

Fig. W-3. Cross Fingering to Excite Next Harmonic.



4. Flute: Synthesis. A flute-like sound can be synthesized using a sine wave (see Fig. W-4). The synthesized sound is approximate; however, not bad for imitating a toy flute. Improvements can be made by adding a little white noise to the sound to imitate breath and another oscillator to add a touch of the first overtone. These additions require more modules and a mixer.

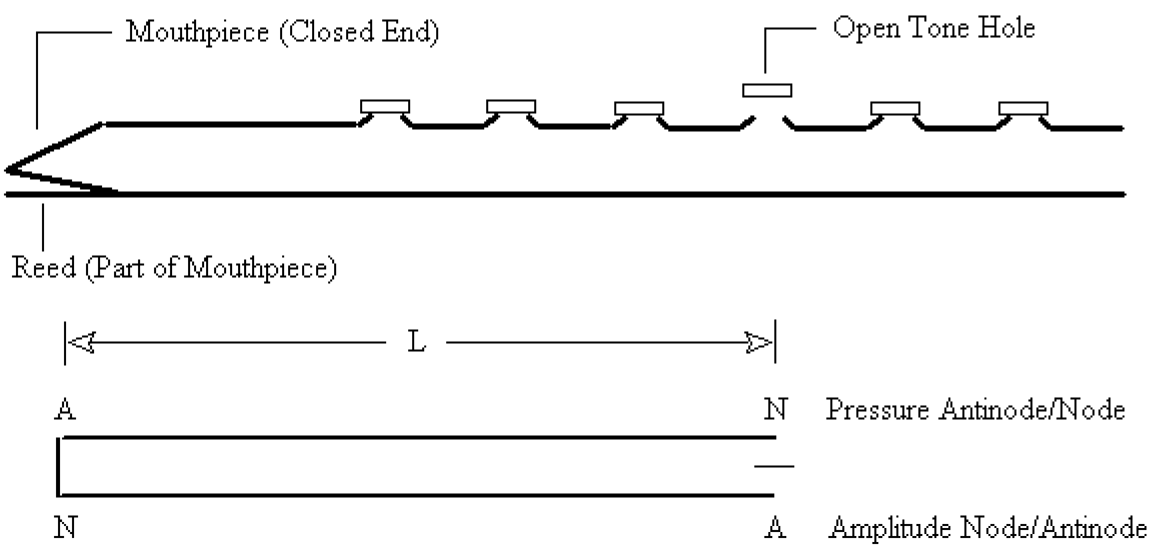
Fig. W-4. Synthesizing a Flute-Like Sound.



Clarinets

1. Clarinet: The Physics. The clarinet mouthpiece is different from the flute. The player's mouth covers the end of the mouthpiece on the clarinet. Therefore, this end is closed. Fig. W-5 illustrates modeling the clarinet as a closed model. The reed vibrates, supplying the energy for the resonance. The effective length of the closed pipe is from the mouthpiece to the opening at the tone hole in Fig. W-5. The wavelength for the fundamental is given as 4L for a closed pipe of length L since one quarter-wave fits along the pipe length. The fundamental frequency produced is one octave lower than that for an open pipe of the same length. Therefore, the E_4 (330 Hz) we obtained with a 50-cm open pipe in the flute analysis now drops an octave lower to E_3 (165 Hz) for a closed pipe of the same length. Details of the explicit calculation are found in Fig. W-5.

Fig. W-5. Clarinet Schematic.



For the Fundamental: $\lambda / 4 = L$ $\lambda f = v = 330$ m/s (Freezing Temperature)

Example calculation: Find the frequency if L = 50 cm (the length for E_4 on the flute).

Step 1. Find Wavelength from Pipe Length: $\lambda / 4 = L = 50$ cm

$\lambda = 4\,L = 200$ cm

$\lambda = 2$ m

Step 2. Find Frequency from Wavelength: (2 m) f = 330 m/s (using $\lambda f = v = 330$ m/s)

f = 165 Hz (Octave lower than the flute case.)

2. Clarinet (Fig. W-6): Producing the Tones in the Scale. For an open-pipe instrument, 11 tone holes and the 1 all-tone-holes-shut position cover the 12 notes of the chromatic scale. Higher harmonics are produced to reach into the next octave or two.

There is a problem with a closed pipe since the next harmonic is the third harmonic, not the second. Therefore we need tone holes to span the notes from H1 to H3. Fig. W-7 illustrates the physics of tone holes for open and closed pipes. The total pipe length (all holes shut) is used for the lowest note. The tone holes are needed to fill in the notes in between H1 and H2 for open pipes, H1 and H3 for closed.

Fig. W-6. Clarinet

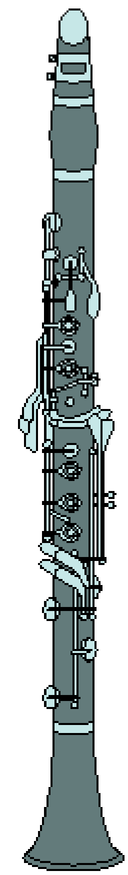


Fig. W-7a. Tone Holes Needed for Open Pipe (11).

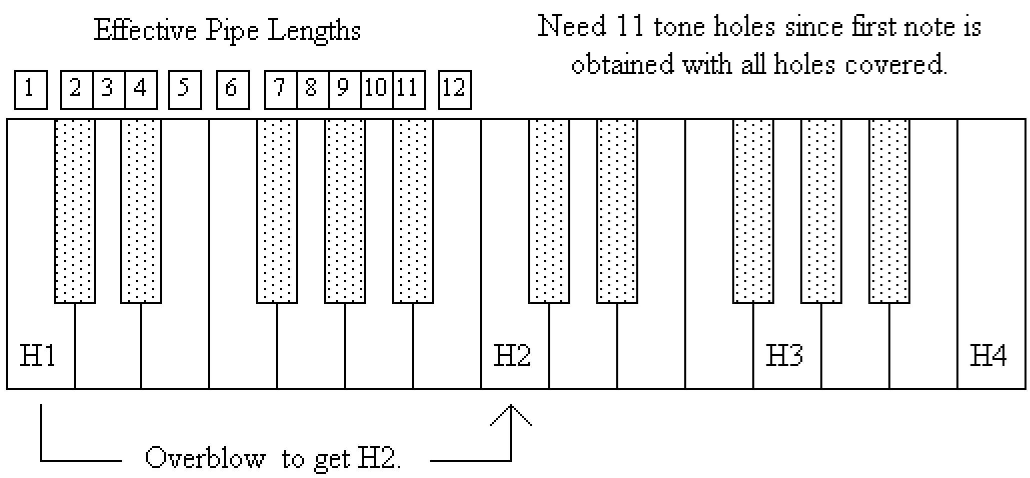
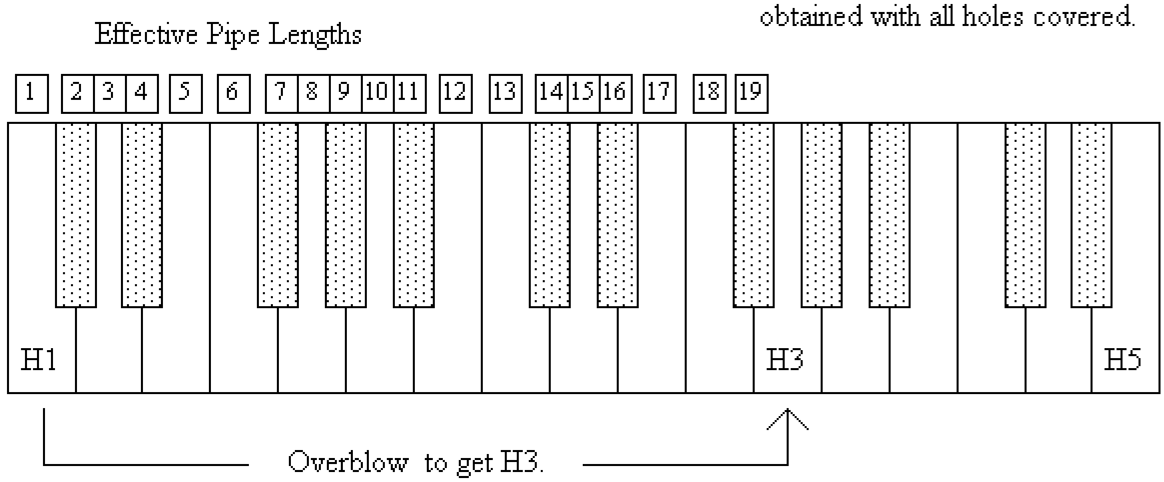
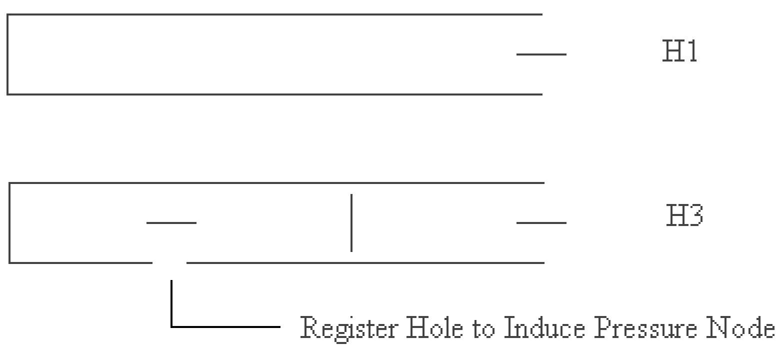


Fig. W-7b. Tone Holes Needed for Closed Pipe (18).



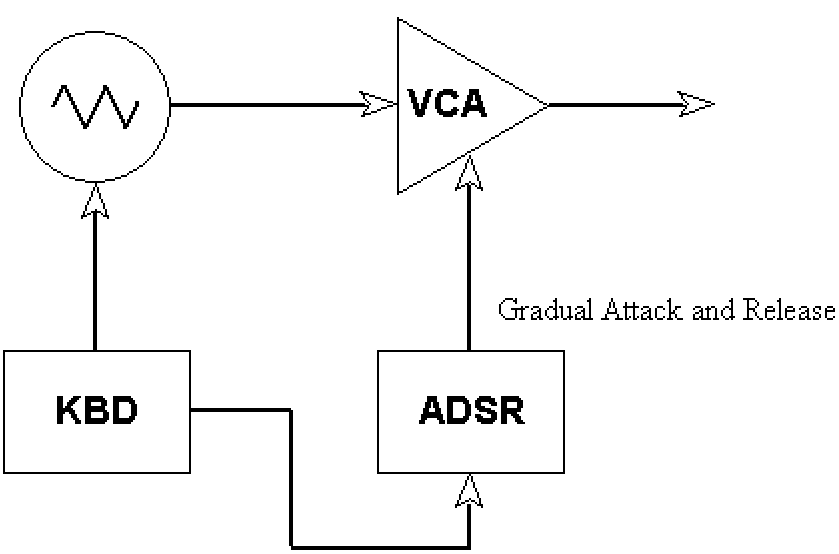
3. Clarinet: Extending Musical Range. The clarinetist gets assistance from a register hole to produce the third harmonic (see Fig. W-8). Cross fingering is also used to play in the higher octaves. The combination of these and blowing techniques enable the clarinet to extend to 3 octaves. The common orchestral clarinet, the B-flat clarinet, easily spans the 3 octaves D_3 to D_4, D_4 to D_5, and D_5 to D_6. The first note D_3 is almost an octave lower than the flute's first note C_4.

Fig. W-8. Register Hole to Induce The Third Harmonic.



4. Clarinet: Synthesis. To synthesize a clarinet-like sound, we first note that the clarinet functions, in our model, as a closed pipe. Closed pipes produce odd harmonics. The resonance in the lowest register consists mainly of the fundamental. To achieve a fundamental with some small presence of odd overtones, we use a triangle wave or square wave. The Fourier spectra for these contain only odd harmonics.

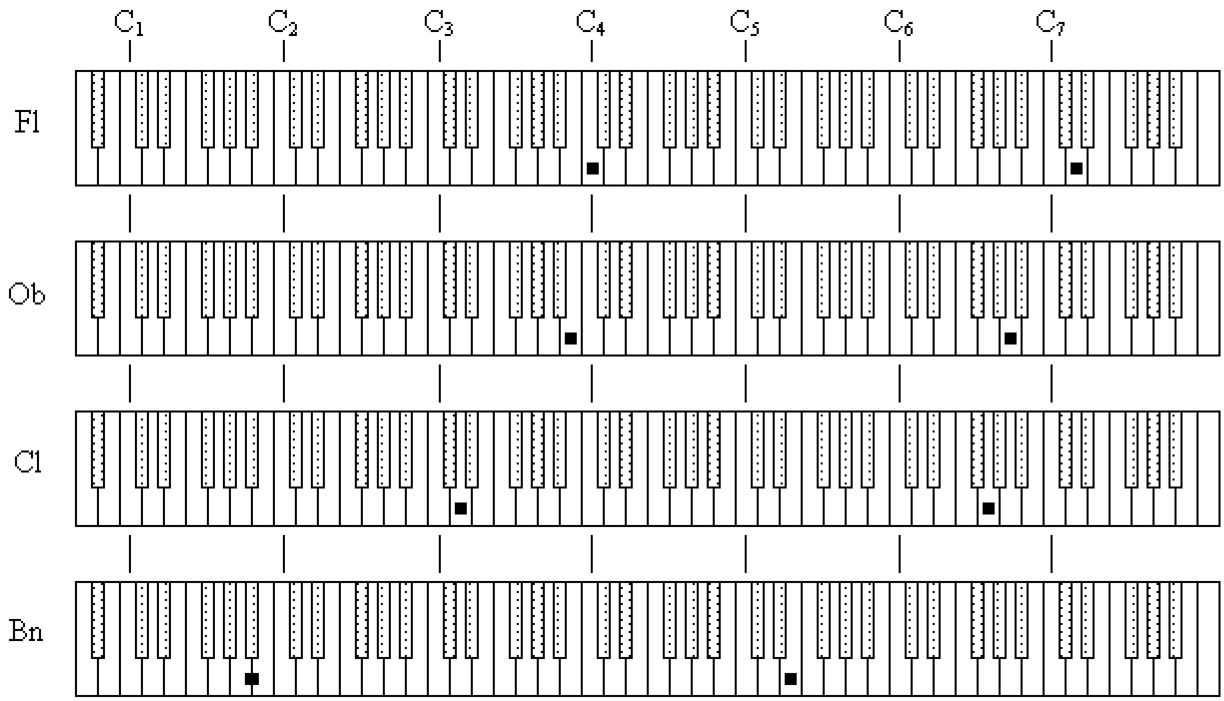
Fig. W-9. Synthesizing a Clarinet-Like Sound.

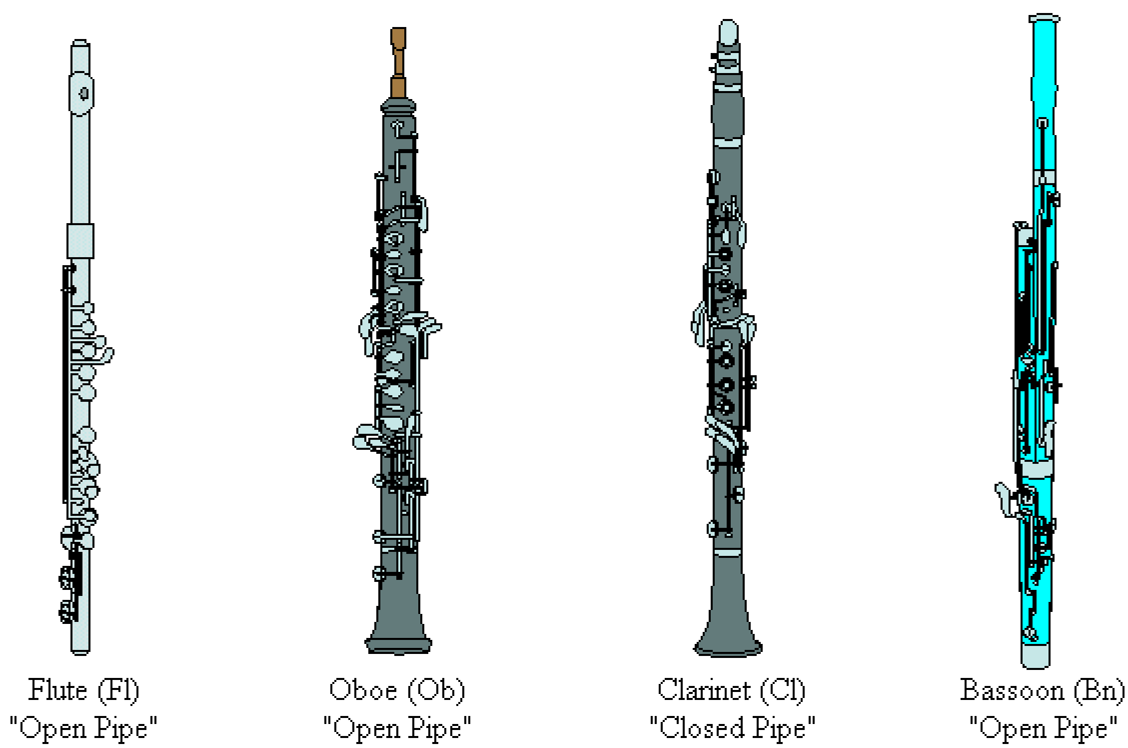


The Woodwind Choir

The "choir of woodwinds" is shown below. in Fig. W-10. Think of this analogy with a choir of singers: flute (soprano), oboe (alto), clarinet (tenor), and bassoon (bass).

Fig. W-10. Woodwind Ranges.





Not to scale.

The oboe and bassoon have double reeds. This makes each function more like an open pipe rather than a close pipe. Fill in the circle of the appropriate choice for each below.

Description	Flute	Clarinet	Oboe	Bassoon
Lowest Range	○	○	○	○
Closed Pipe	○	○	○	○
Double Reed Like the Bassoon	○	○	○	○
Single Reed	○	○	○	○
Can Reach the Highest Note	○	○	○	○
Range Closest to that of the Flute	○	○	○	○

--- End of Chapter W ---

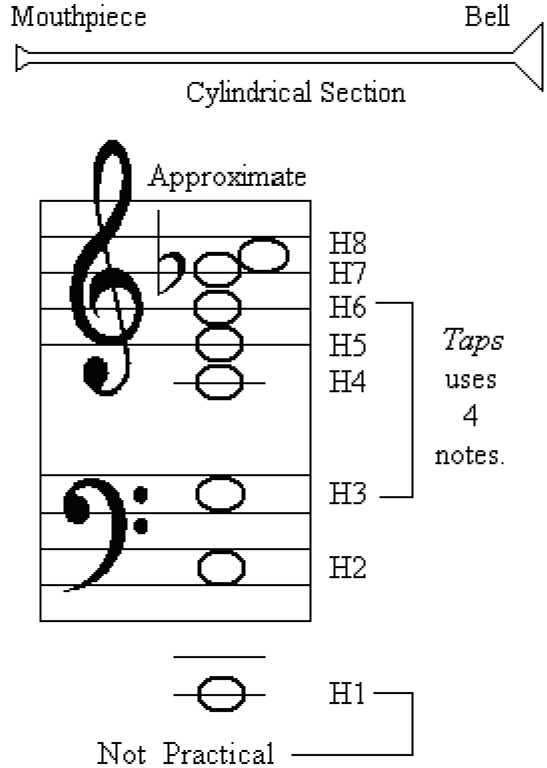
X. Brass

Early brass instruments were pipes of fixed lengths (see Fig. X-1). Today four very common brass instruments are found in the orchestra: the French horn, trumpet, trombone, and tuba. These each have a mouthpiece, cylindrical section, and bell end.

The trombone externally extends to increase length, while the trumpet uses valves to include more internal pipe sections in the resonance. The French horn and tuba likewise use valves. Focusing on the trombone and trumpet will give us an understanding of the two different methods employed by the brass instruments to obtain different pipe lengths for different notes.

Due to complicated effects, all harmonics are present for brass instruments; however, the fundamental is hard to access for typical use. For practical purposes, the player is restricted to the overtones (Fig. X-1) for a fixed-length pipe. Songs such as *Taps* were composed using just overtones for the older brass instruments.

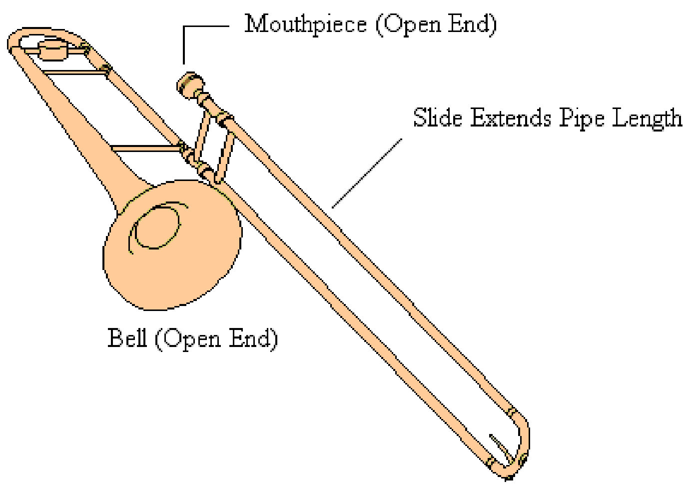
Fig. X-1. Early Horn.



Trombones

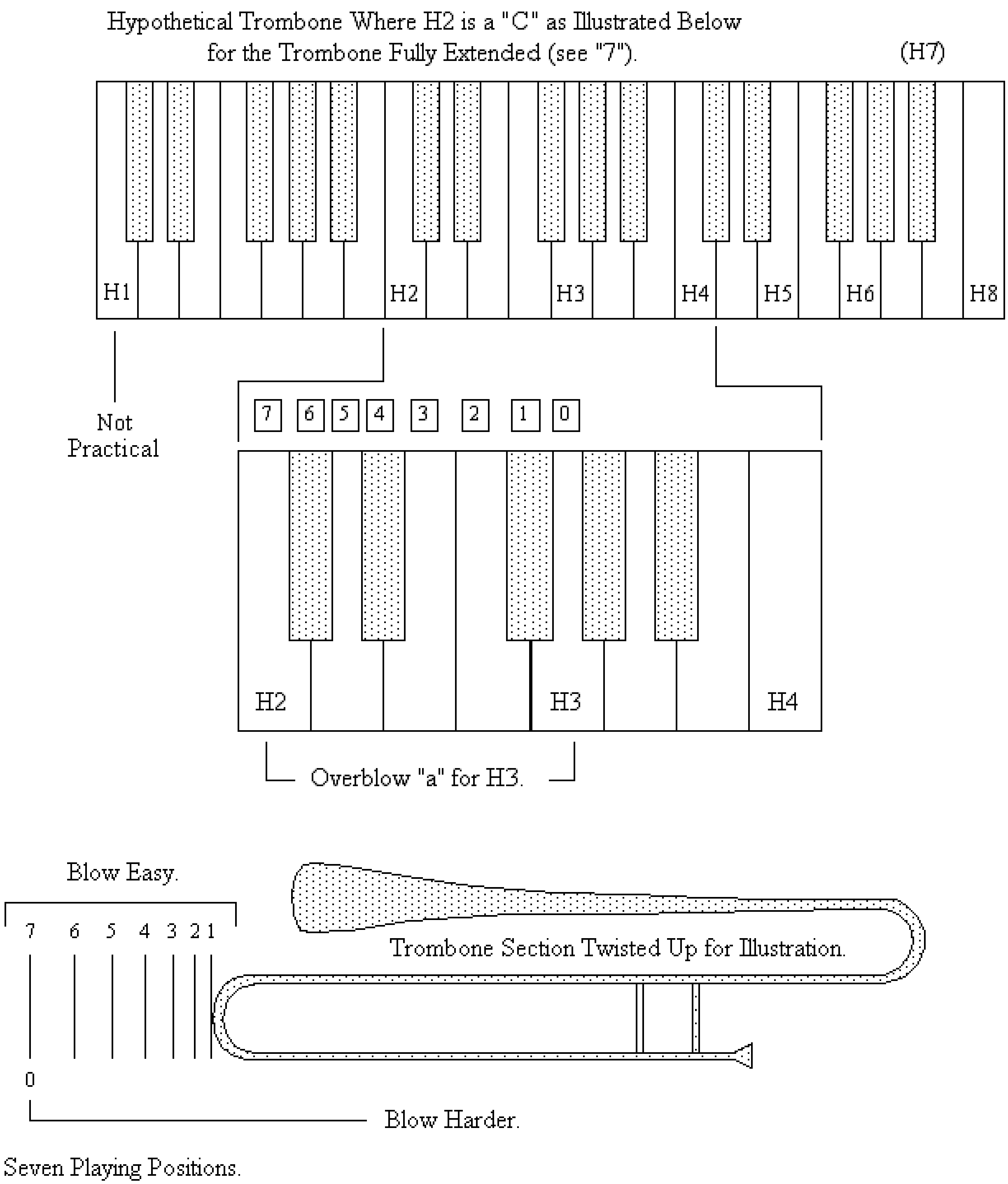
1. Trombone: The Physics. Fig. X-2 illustrates how the restriction to one set of overtones is removed in the trombone by the slide. Pipe length is adjustable to obtain more tones.

Fig. X-2. Trombone.



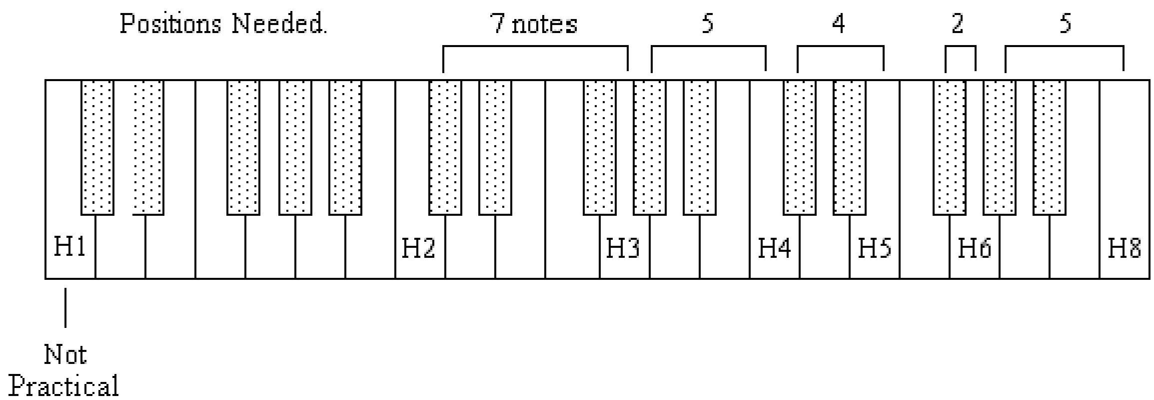
2. Trombone: Producing the Musical Tones in the Scale. We have transposed the harmonics to fall on convenient notes in Fig. X-3. We choose H2 to be a "C" for the fully extended trombone. The length is reduced to fill in the notes of the chromatic scale between H2 and H3 indicated on the keyboard of Fig. X-3.

Fig. X-3. Trombone Playing Positions.



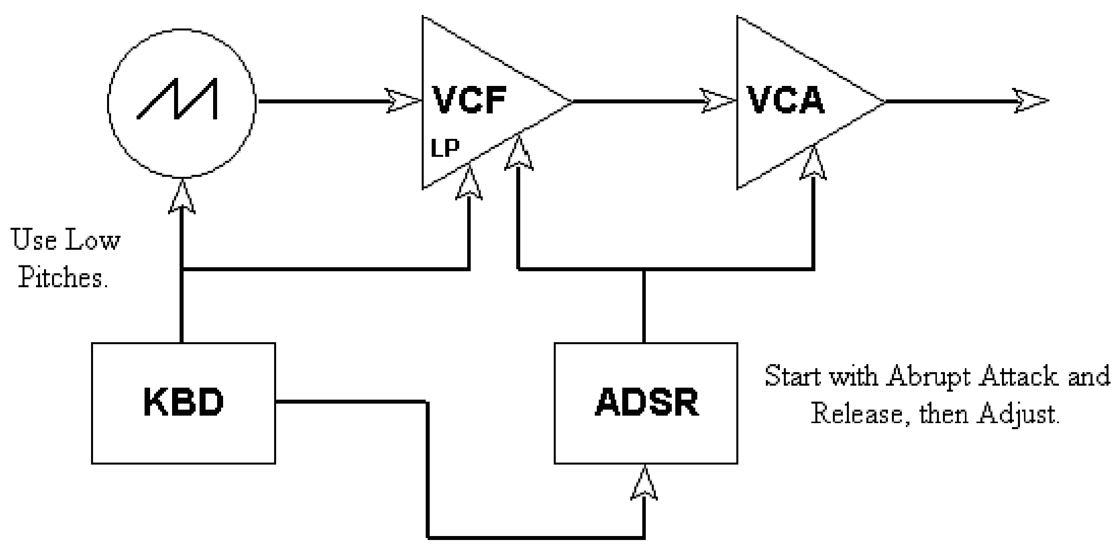
3. Trombone: Extending Musical Range. In Fig. X-3 the extended positions increase the pipe length by our equal-tempered 6%-"interest" amount. This drops the tone a half step each time. Note how the increments in length grow from position 1 to 7. This is gaining "interest on interest." Think of the 7 positions as the capability to achieve 7 half steps. By blowing harder into the trombone, harmonic jumps are achieved. The playing positions are used to fill in the half steps from harmonic to harmonic. Fig. X-4 indicates how the number of half steps decreases as you go higher. Note that H7 is avoided. The trombone can reach 3 octaves in this way. The real trombone starts on an "E_2," not a "C."

Fig. X-4. “Hypothetical” Trombone (H2 on “C” for Position 7) Extending Its Range.



4. Trombone: Synthesis. To synthesize a trombone-like sound we choose the "raspy" ramp wave (see Fig. X-5). The ADSR shapes the filter and the VCA. The timbral change gives a characteristic brass "vah-rump" sound. The ADSR is varied to get the desired effect. The lower harmonics are emphasized due to the low-pass filter.

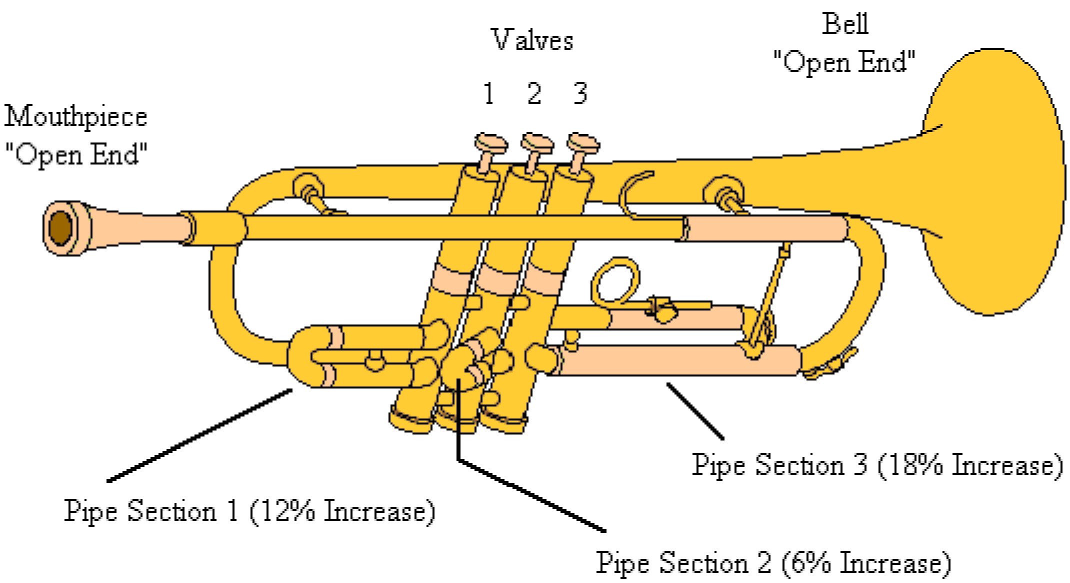
Fig. X-5. Synthesizing a Trombone-Like Sound (Staccato Effect)



Trumpets

1. Trumpet: The Physics. The trumpet uses valves to increase the length of the pipe. Pressing a valve increases the internal pipe length (see Fig. X-6). When no valves are pressed, the trumpet has its minimum or base length. Valve 1, when pushed down, allows a pipe section to join the internal path, which increases the base length by approximately 12%. The three valves provide increases of 6%, 12%, and 18%. These are multiples of our "interest rate" of 6%. They allow for frequency changes of a half step, whole step, and whole + half step respectively. Note that the smallest percentage increase goes with valve 2, the one in the center.

Fig. X-6. Trumpet.

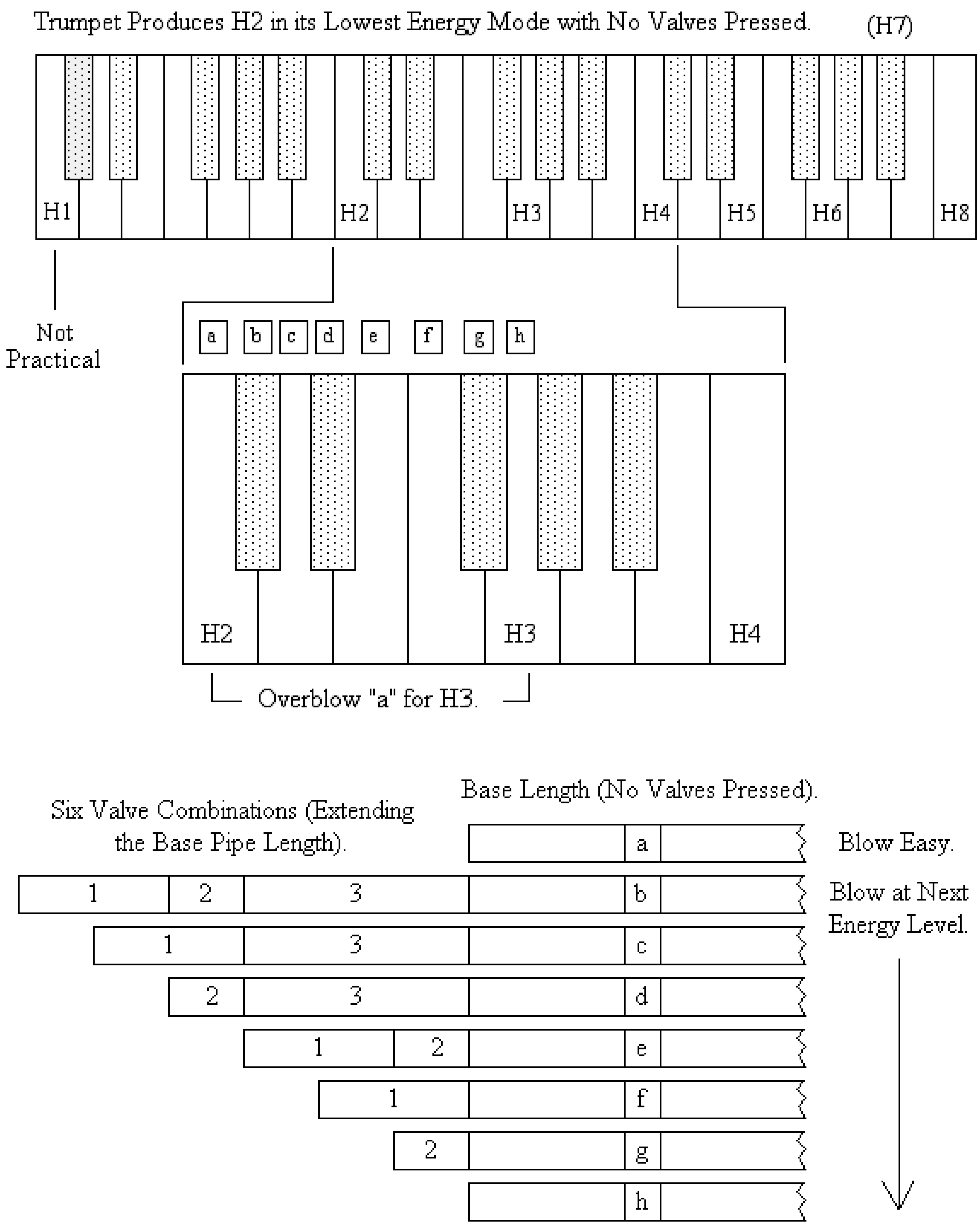


Why 3 valves? We know from the trombone that a base length can give us H2 and H3. So we need 6 additional playing positions to fill in the gap between H2 and H3. To reach these 6 half steps going down from H3, we need to increase the base length by (forgetting little extra gains by the "interest-on-interest" effect) 6%, 12%, 18%, 24%, 30%, and 36%.

The little additional percentages needed by the gain of "interest-on-interest" effect can be compensated by blowing technique. How can we obtain these 6 percentages with the minimum number of building blocks? The answer is 3, choosing as units 6%, 12%, and 18%. You get 24% by adding the 6%-unit to the 18%-unit, 30% by combing the 12%-unit with the 18% unit, 36% by adding all three units: 6% + 12% + 18% = 36%. This is a very elegant application of basic mathematics and physics to instrument design. In practice, the 18%-unit is saved until you need 24%. To achieve 18%, the 6%-unit is combined with the 12%-unit. It gives a better match due to the ways the precise percentages are chosen in order to minimize the "interest-on-interest problem."

2. Trumpet: Producing the Musical Tones in the Scale. The trumpet has 7 basic playing positions, like its cousin the trombone, to span the notes from H2 to H3. See Fig. X-7. The H2 on the keyboard corresponds to base length rather than full "extended" length as it did earlier for the trombone. We do this because Fig. X-7 is an actual trumpet.

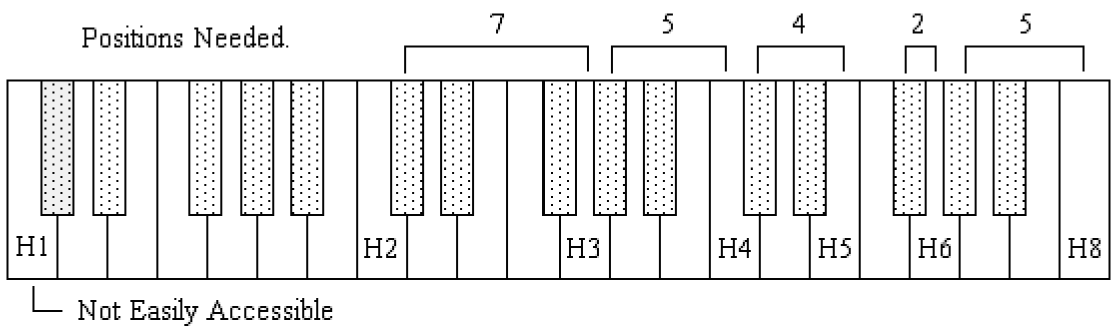
Fig. X-7. Trumpet Playing Positions.



3. Trumpet: Extending Musical Range. Blow the trumpet with no valves pressed (position "0") hard enough to obtain H3 (see Fig. X-8). Now press all 3 valves (1-2-3) to lower the tone 6 half steps (which is roughly a 36% increase in length, 6% per half step). This gives the note to the immediate right of H2 in Fig. X-8.

Proceed to decrease pipe length by 6% each time to raise the pitch note by note to H3. Fig. X-7 gives these valve configurations, which decrease the length by 6% each time: 1-3 (total pipe length is 30% beyond base length), 2-3 (24%), 1-2 (18%), 1 (12%), 2 (6%), and 0 (0%). Now blow harder to get H4 (no valves used). You have skipped the tones between H3 and H4. So we need to drop 4 half steps to get to the note immediately right of H3. This is 4 x 6% = 24% (increase in length), valve configuration 2-3. We then proceed with 6% decreases in length to march up to H4. These are 1-2 (18%), 1 (12%), 2 (6%), and 0 (0%). Blow even harder with no valves and get H5. To get the 4 notes from the note after H4 to H5, we just need 1-2 (18%), 1 (12%), 2 (6%), and 0 (%). Then, blow harder to get H6. Can you figure out which configurations are next. If so, you know how to play the trumpet theoretically!

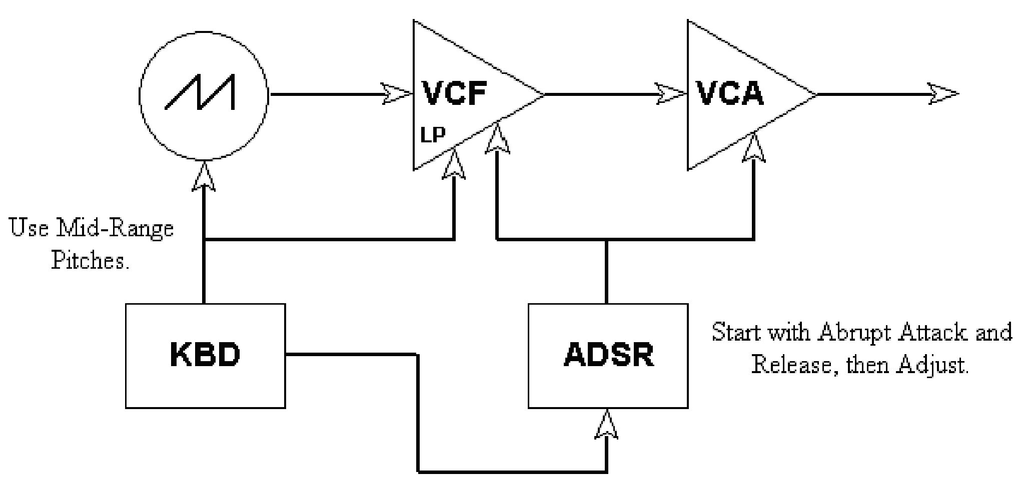
Fig. X-8. Extending the Trumpet's Range by Overtones and Values.



4. Trumpet: Synthesis. The arrangement (Fig. X-9) is the same as that for the trombone except we use mid-range pitches.

We obtain lower harmonics by employing the low-pass (LP) filter. Remember that the ramp wave is rich in harmonics.

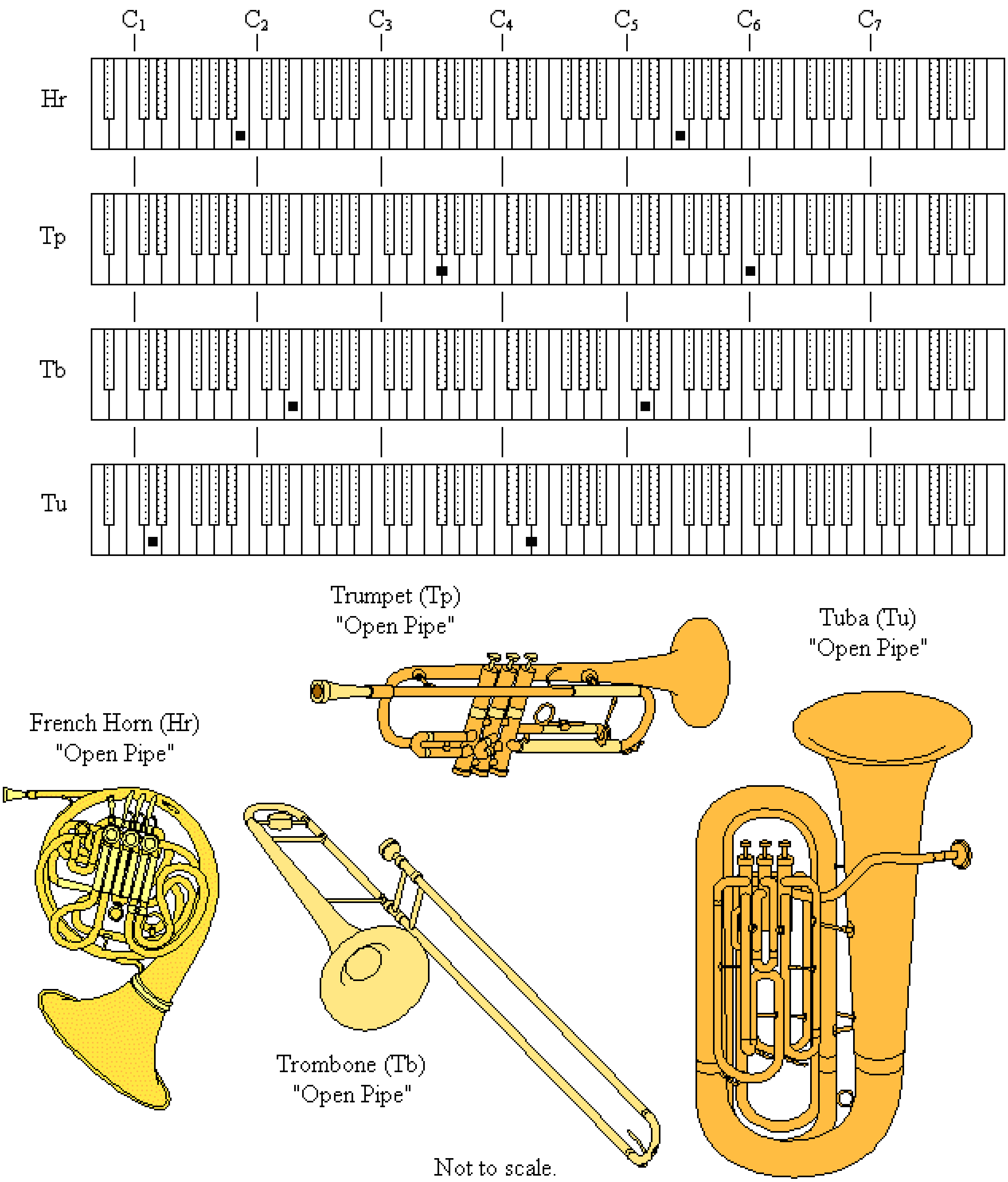
Fig. X-9. Synthesizing a Trumpet-Like Sound (Staccato Effect).



The Brass Choir

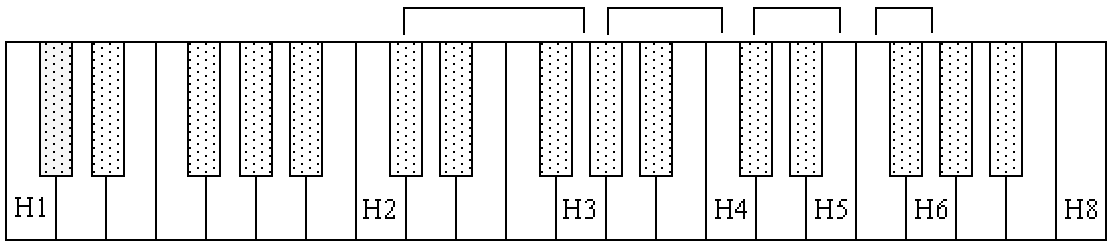
The "choir of brass" is shown below. in Fig. X-10. Think of this analogy with a choir of singers: trumpet (soprano), horn (alto), trombone (tenor), and tuba (bass). The convention is to place the horn at the top even though the trumpet reaches higher notes.

Fig. X-10. Brass Ranges.



An Exercise

Give the number of tone holes or playing positions needed to span each of the intervals in the table below. Note that the last note in each span is obtained by blowing harder on the configuration of the instrument needed to obtain the lower harmonic listed for each span.



Description	Number	Interval
Flute Tone Holes to Fill Gap Between H1 and H2		Octave
Clarinet Tone Holes to Fill Gap Between H1 and H3		Octave + Fifth
Brass Playing Positions to Span from H2 to H3		Fifth
Brass Playing Positions to Span from H3 to H4		Fourth
Brass Playing Positions to Span from H4 to H5		Third
Brass Playing Positions to Span from H5 to H6		Minor Third

--- End of Chapter X ---

Y. Strings and Percussion

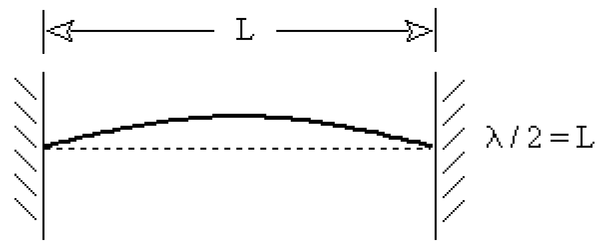
We now come to the application of strings in musical instruments. We begin with the violin family, the string section of the orchestra. Then we consider the guitar. The fundamental string physics we studied earlier applies to these. With the piano, strings get complicated. There is far more tension in the strings. The strings are stiffer, to some extent like a rod. Also, there are string wrappings about other strings for the lower tones. These features make the piano somewhat more complicated. Finally, we will briefly discuss drums and cymbals.

The Violin Family

The violin family brings us to the last major section of the orchestra. The first two, woodwinds and brass, deal with pipe physics. The violin and its cousins employ strings. In fact, the section of the orchestra with these instruments is called the string section or simply strings. Strings are employed the most by composers overall. The audience never seems to get tired of the nice blend of harmony strings offer. The woodwinds come second in use and the powerful brass last. The string family consists of the violin, viola, cello, and bass (also called the contrabass). Each member of the string family has 4 strings. We will consider the orchestral stringed instruments together.

1. Violin Family: The Physics. The vibrating string supplies the basic physics of the violin family. The fundamental has a wavelength equal to twice the length of the string. One half-wave fits between the fixed ends for the first mode of vibration (see Fig. Y-1). The string has the same overtone series as the open pipe.

Fig. Y-1. Vibrating String.

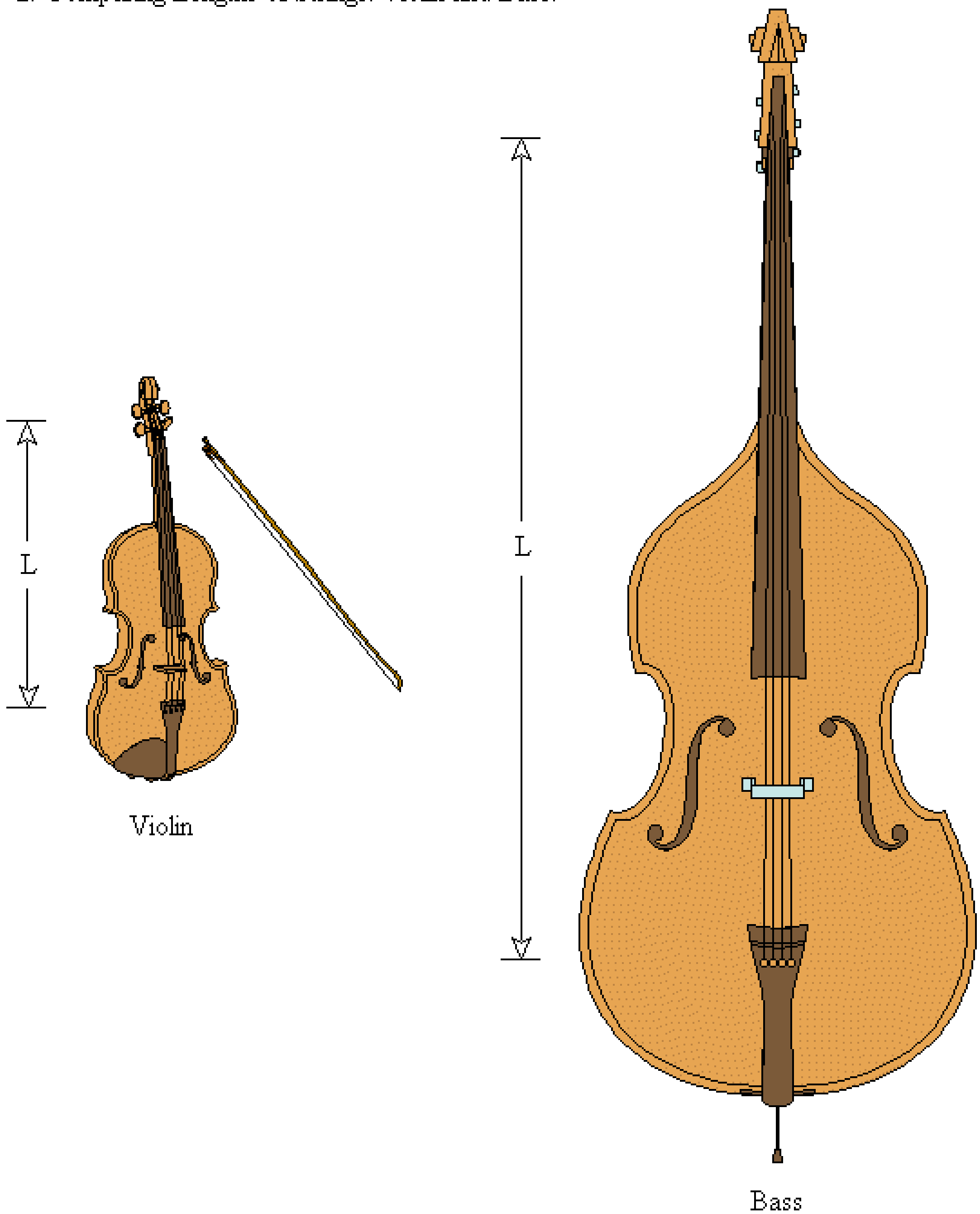


The violin is usually played by drawing a bow across the string. This supplies the energy for the string to resonate. The interaction between bow and string generates predominantly the fundamental. However, a rich amount of overtones is also produced. The violin cavity and wood enable us to hear the sound easily. They also enhance the overtones that fall into formant regions.

The overtones enrich the sound. Overtones are not used the way they are in woodwinds and brass instruments. There, one uses specific resonances of the higher harmonics. Playing the woodwinds at the fundamental mode is referred to as playing in the first register. Then, playing at the second harmonic by either cross fingering or using a register hole is called playing in the second register. You can carefully hold the center of a violin string gently and pluck it with your free hand. You will excite the second harmonic. But this is not done in practice. Instead, the violin has four separate strings to reach higher registers of pitch.

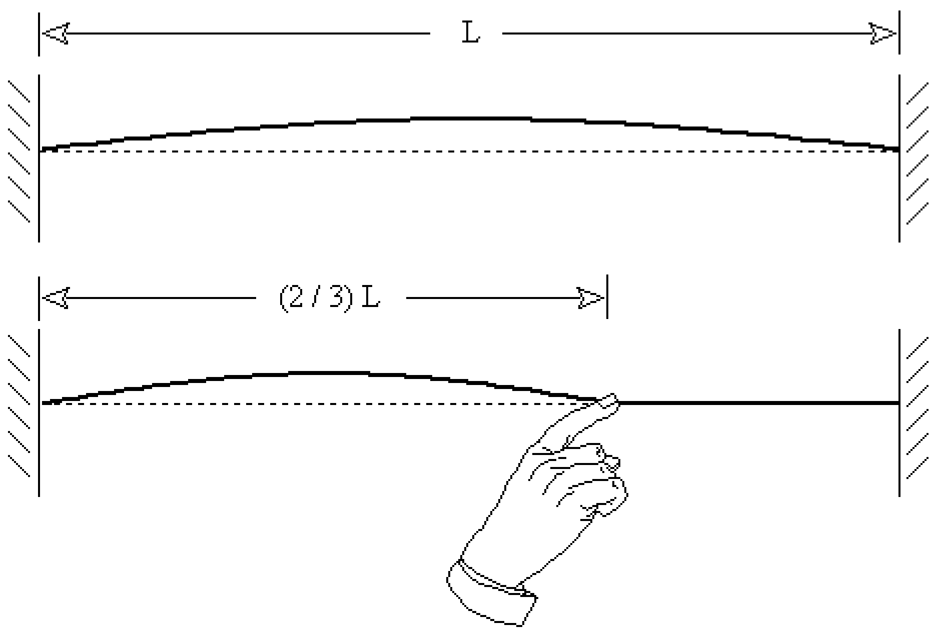
Fig. Y-2 illustrates the practical version of Fig. Y-1. The length of the string on the violin is compared to the length of a string on the bass (contrabass). The long string of the bass produces the low pitch we expect from a bass. This follows from Mersenne's First Law.

Fig. Y-2. Comparing Lengths of Strings: Violin and Bass.



2. The Violin Family: Producing the Tones in the Scale. Different pitches are obtained on a violin by placing a finger along the string. This shortens the section of the string undergoing the vibration. See Fig. Y-3 for an example where the string is shortened to two-thirds its original length. The new fundamental has a shorter wavelength. The wavelength is scaled down to two-thirds of the original value. The violinist can move up the string in this way to produce the half steps of the chromatic scale.

Fig. Y-3. Using a Finger to Obtain a "Shorter String Length."



The ratio of 2:3 for the wavelengths in Fig. Y-2 corresponds to the interval of a fifth. Recall that whenever you shorten the wavelength, the frequency increases. The frequency ratio is 3:2. The numbers are reversed for the frequency. Pythagoras measured such string-length ratios for the consonant intervals (see Table Y-1).

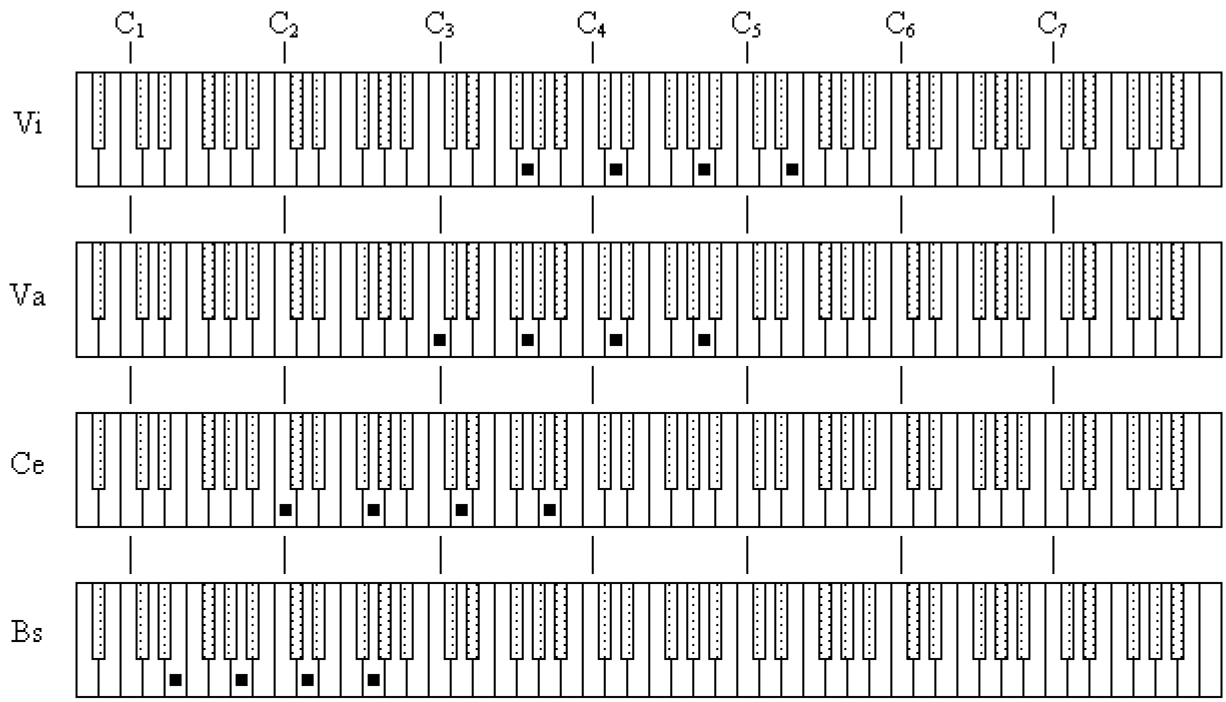
Table Y-1. Ratios of Frequencies, Wavelengths, and String Lengths for Common Intervals.

		Ratios		
Tones	Interval	Frequency	Wavelength	String Length
Do-Do	Unison	1:1	1:1	1:1
Do-Do'	Octave	2:1	1:2	1:2
Do-Sol	Fifth	3:2	2:3	2:3
Do-Fa	Fourth	4:3	3:4	3:4

3. The Violin Family: Extending Musical Range. Stringed instruments have more than one string to extend the range. See Fig. Y-4 for the four strings found on the violin (Vi), viola (Va), cello (Ce), and bass (Bs, or contrabass). The spacing interval is a fifth for the violin, viola, and cello. Three octaves are easily within reach this way.

The frequencies get lower overall as the instruments go from violin to bass. The cello and bass have longer strings to produce bass notes. The strings get so long for the bass, it's too much of a stretch (of the arm) to play many different notes on one string. The smaller spacing of a fourth cuts down on the number of notes each string must be responsible for.

Fig. Y-4. The Four Strings on the Violin, Viola, Cello, and Bass.



4. The Violin Family: Synthesis. To synthesize a very approximate violin-like sound, we observe that the string is pulled by the bow due to friction, then quickly slips back, is pulled again, and slips back, etc. like a ramp wave. A ramp wave is chosen for synthesizing bowing in Fig. Y-5. A gradual attack is employed for approximating a gentle approach of the bow. An abrupt release is used to reproduce a sudden release of the bow. Of course, the violin need not be played this way. We are only choosing one of the many characteristic ways to play the violin. One serious limitation of our synthesis is the lack of enhanced overtones that occur in the violin due to the formants.

On the other hand, it is relatively easy to approximate the plucking sound of a bass (see Fig. Y-6). The plucking of the string produces a sound much closer to a sine wave than bowing. The formants aren't as important then, since the sine wave has no overtones.

Fig. Y-5. Synthesizing a Very Approximate Violin-Like Sound (Bowing).

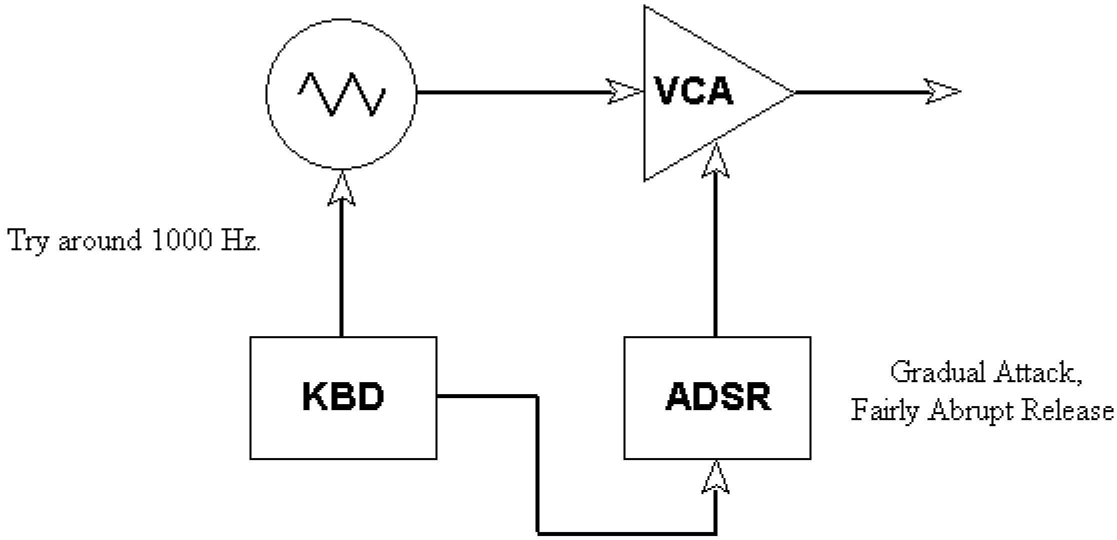
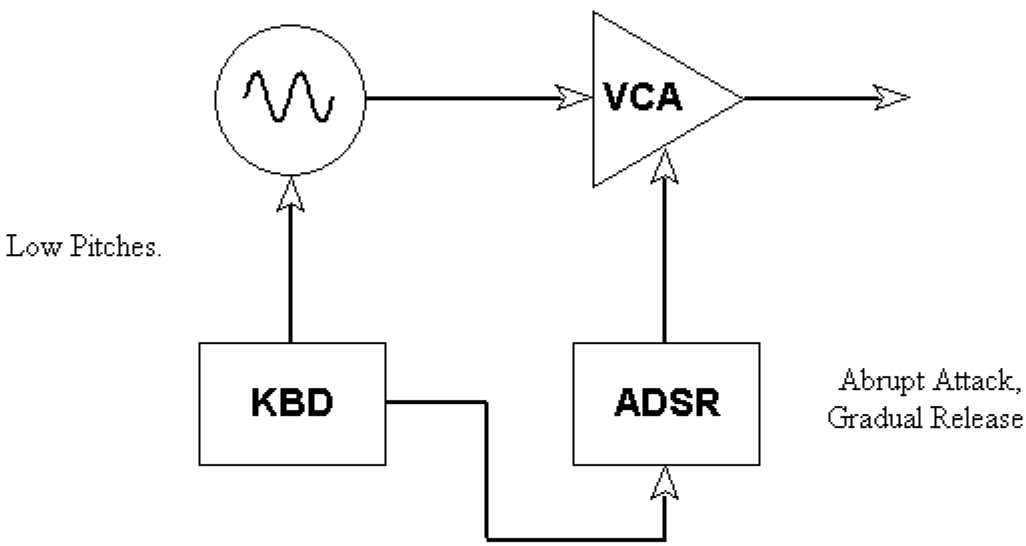


Fig. Y-6. Synthesizing a Plucking-Bass Sound.

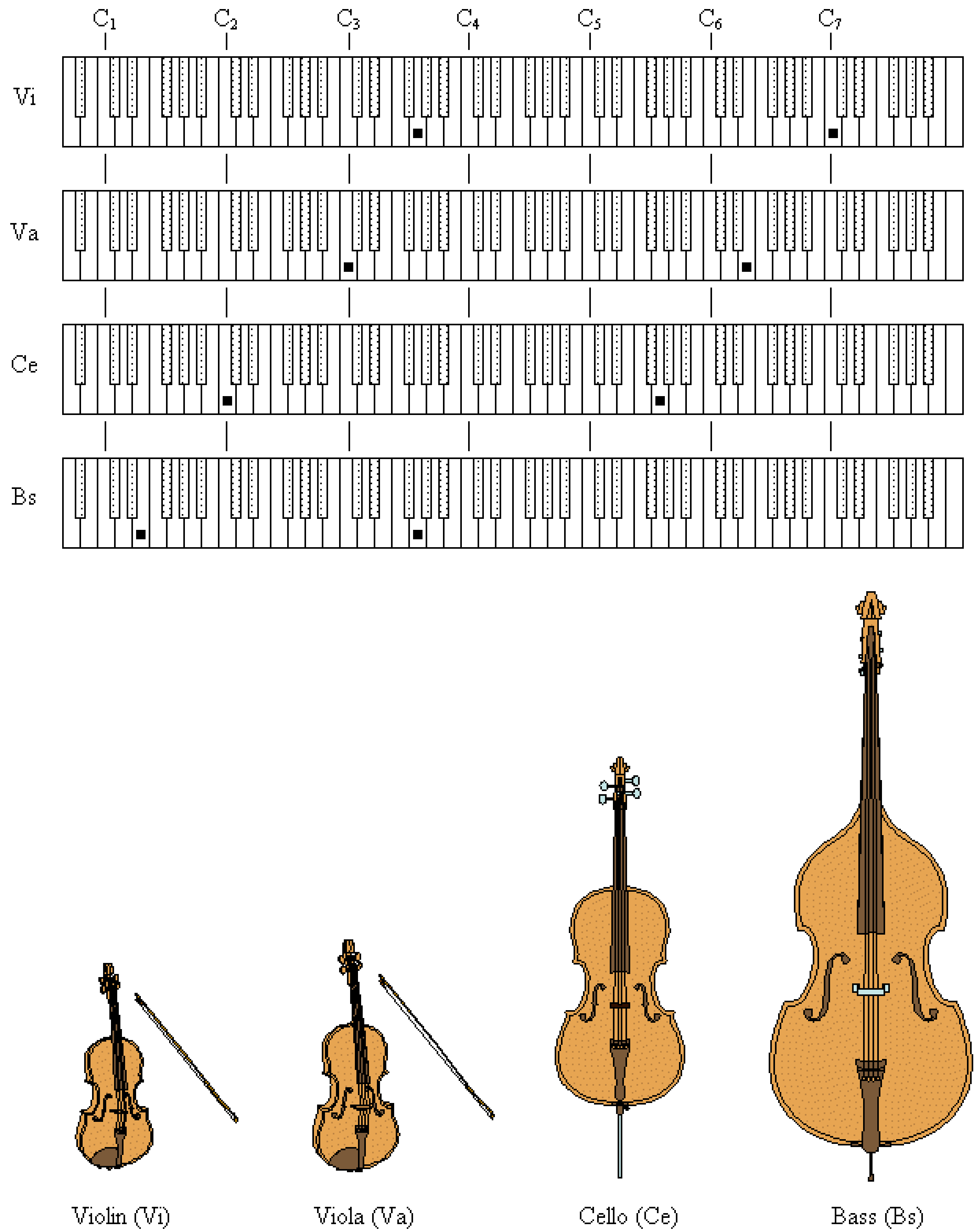


An orchestra typically has 30 violins, divided into two sections, the first and second violins. There are approximately a dozen violas, a dozen cellos, and eight or ten cellos. The strings blend so well together due to their similar structure. They are used most of the time in musical compositions. The strings can be employed for long lengths of time because the performers do not get tired easily. Woodwind and brass players need to occasionally "catch a breath."

The String Choir

Fig. Y-7 illustrates the string choir of the orchestra. Think of this analogy with a choir of singers: violin (soprano), viola (alto), cello (tenor), and bass (bass).

Fig. Y-7. String Ranges.



A Review

What is the harmonic number n for the wave shown in the photo? __________

b. How many half-waves are there in the above photo? __________

c. How many half-waves are there in harmonic 2? __________

d. How many half-waves are there in harmonic 1? __________

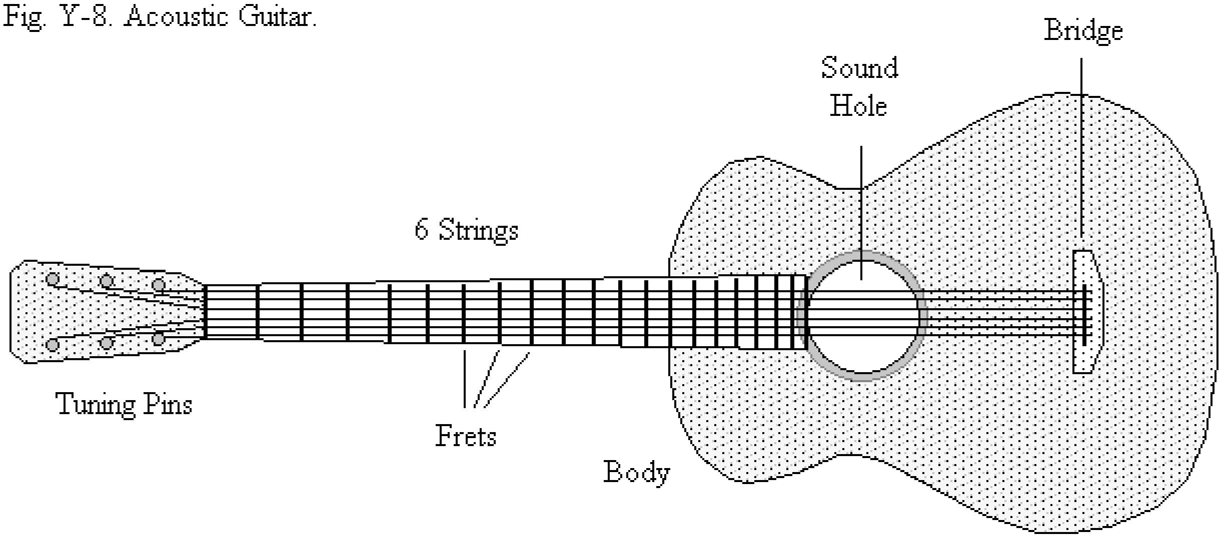
e. How many nodes are there in harmonic 8? __________

f. How many antinodes are there in harmonic 8? __________

Guitars

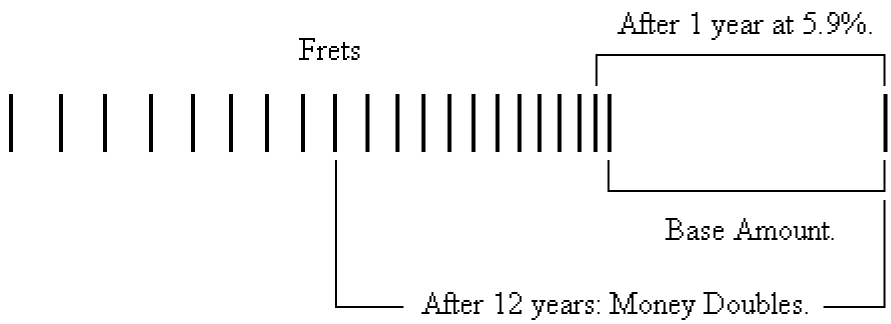
The guitar is one of the most popular instruments. It is light, portable, and offers excellent accompaniment to singers. It is used for classical, popular, folk, country, and rock music. Popularity soared in the 1950s with the fame of Elvis Presley and continued in the 1960s due to early rock groups such as the *Beatles*. The acoustic guitar is not electric. The sound is brought out by the body, cavity, and sound hole. See Fig. Y-8. The energy is supplied by plucking or strumming. The strings resonate. The vibrations are transmitted to the body by the bridge. The wood and cavity enable us to hear the sound. Their formant regions enhance some of the overtones. Tuning pins allow for control of tension. According to Mersenne's Second Law, greater tension produces higher pitch.

Fig. Y-8. Acoustic Guitar.



The guitar has frets (ridges) which enable the performer to easily find the correct positions to effectively shorten the strings for specific tones. Fig. Y-9 illustrates the spacings for the frets. Each step from fret to fret is a half step. We use our banking analogy with an investment interest of 5.9%. We start at the far right and move leftward. In this way, the far left-fret position gives the lowest frequency. This is an outcome of the inverse relation between frequency and wavelength we analyzed in the last section.

Fig. Y-9. Fret Positions and Banking Analogy.



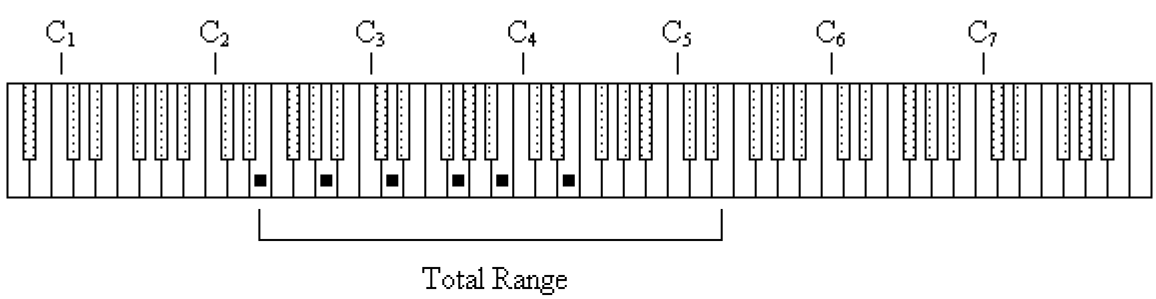
Artisans crafting stringed instruments with frets over the centuries discovered a simple guideline in spacing. They started construction at the end near the tuning pins and placed each fret at 1/18 the distance of the remaining section of string. After you construct a fret, you forget about the string to the left (using the orientation in Fig. Y-8). You measure from the last fret you finished to the bridge. Then proceed to the right 1/18 of this new distance and install the next fret. Since the remaining string distance keeps shrinking, the distances between the frets get smaller and smaller.

We can arrive at this rule from theory. Pick any fret in Fig. Y-8. The distance from the fret to the bridge represents our savings according to Fig. Y-9. By moving to the right from any fret, we reduce the length of the string, or go back in time for our investment analogy. This means we lose some interest. How do we take interest away? First, let's find out what the applied interest is as a fraction. The value 5.9% is about 1/15 or 1/16 as we approximated it earlier from the just diatonic scale. But we must be more precise now. We are no longer working with the just diatonic scale but the equal-tempered scale.

The decimal corresponding to 5.9% is 0.059, just like 50% corresponds to 0.50. The fraction 1/16 = 0.0625 is too large. The better fraction is actually 1/17 since 1/17 = 0.0588 (very close). When we proceed into the future (moving from the bridge to the tuning pins) we increase our length by 1/17 year by year. Fix your attention on any fret. The next fret to the left increases the effective string length by 1/17. The new longer string length is now 18/17 (the original 17/17 plus the extra 1/17). Now to go backwards, we need to make a cut from 18/17 to 17/17. If we start with 18/17, in order to get 17/17, we need to cut off 1/18 of the 18/17-length. We amazingly arrive at the 1/18-rule. This is a subtle use of arithmetic to arrive at a "golden rule" of guitar design.

The guitar extends its range with the help of additional strings. The 6 strings, along with the total range for the guitar, are illustrated in Fig. Y-10. There is also a 12-string guitar. Other plucked instruments include the dulcimer (American folk version has 3 or 4 strings), banjo (4 or 5 strings), lute (4 or more strings), sitar (7 or more strings), zither (30 to 40 strings), and harp (almost 50 strings). All except the harp have frets. The harp is also unique in that it is considered a standard instrument in the symphony orchestra even though it is not used in many works.

Fig. Y-10. Guitar Strings and Total Range.



We saw earlier that a sine wave can be used to synthesize the plucking of a bass. A square wave provides for a richer sound, more comparable to what we hear on a guitar or banjo. Fig. Y-11 illustrates a modular diagram to synthesize such a sound. The resulting sound is more like that of a banjo. The best results use a waveform between a square wave and a pulse train. Square waves have crests that are 50% of the total waveform. The best results are found with crests between 10% and 25%. Increasing the pulse in the range between 25% and 33% produces a harpsichord sound.

Fig. Y-11. Synthesizing the Plucking Sound of a Banjo.

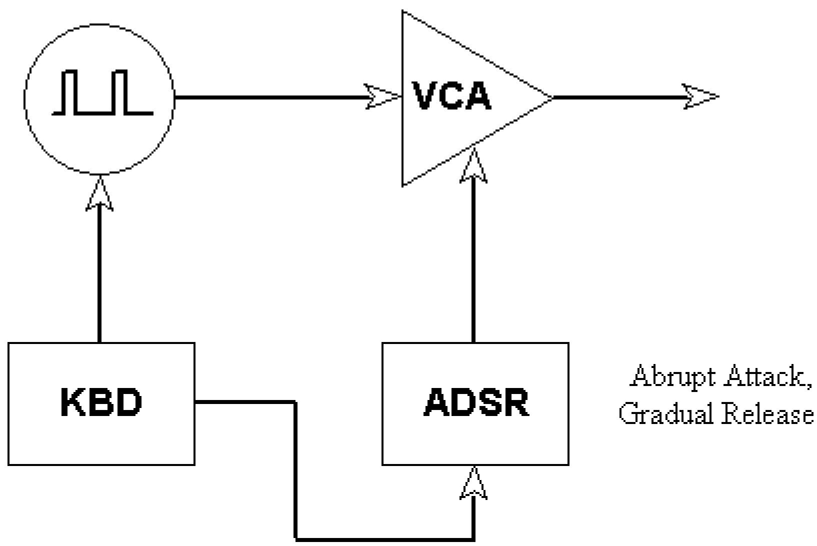
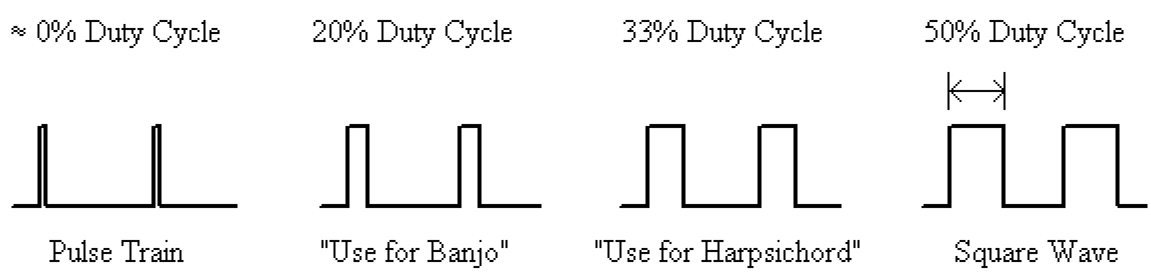


Fig. Y-12 shows some waveforms with different pulse widths. The pulse width compared to the wavelength is called the duty cycle. A square wave has a duty cycle of 50%. The pulse train has a theoretical duty cycle of 0%. The Fourier spectra for the 20% and 33% cases below are more complicated than those for the pulse train and square wave, which we have encountered earlier.

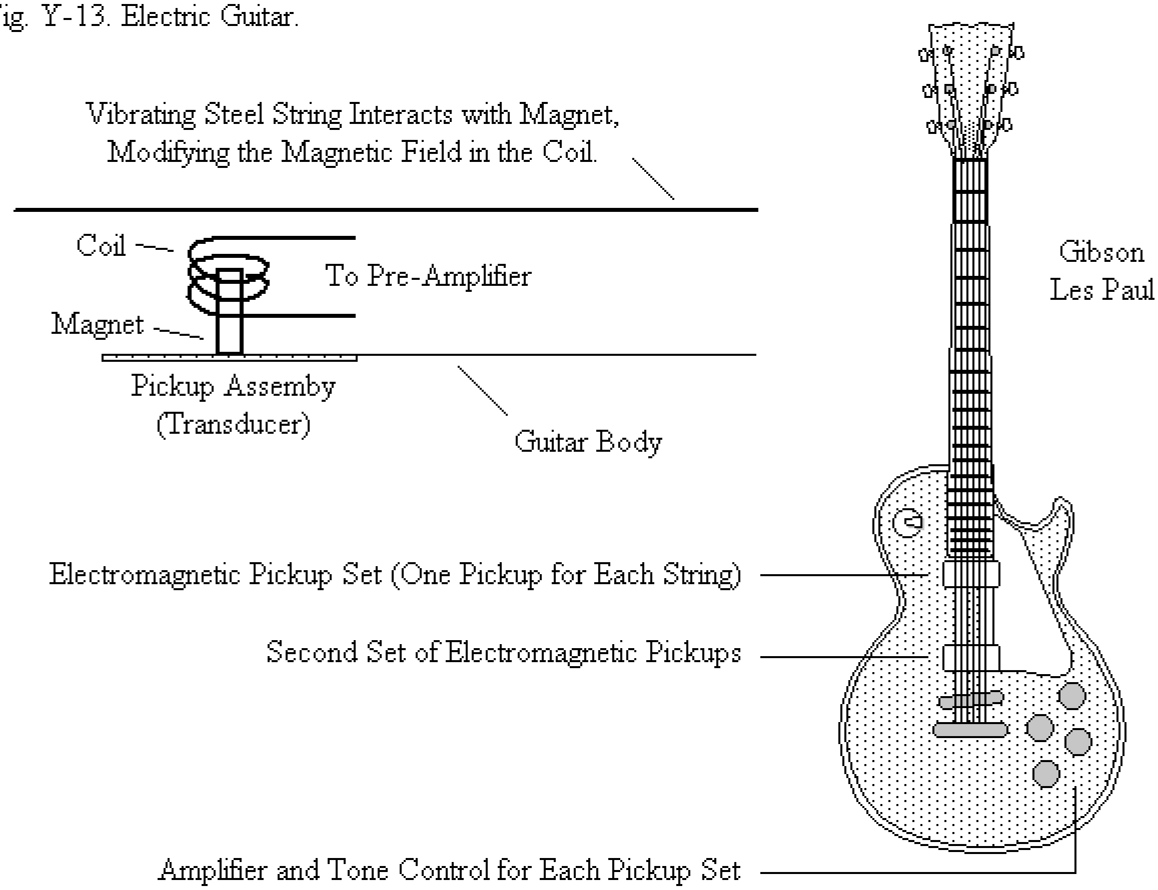
Fig. Y-12. Pulses and Duty Cycle.



The electric guitar amplifies sound by electromagnetic pickups. Fig. Y-13 illustrates the procedure. The metallic strings vibrate near coil-magnet assemblies. The string is not attached to the coil. However, the metal string influences the magnetic field inside the coil as it gets near it, due to the presence of the pickup's permanent magnet. The fluctuating magnetic fields are in step with the vibration of the string. These generate electrical currents in the coil. The electrical signal is sent to an amplifier inside the guitar. The internal amplifier is a *preamplifier*, whose output goes to the regular amplifier for driving the speakers.

There are usually two sets of pickups. Each set contains 6 pickups, one for each string (electric bass guitars have 4 strings). The upper set picks up mostly the fundamental while the lower set registers overtones better. The combination of the two signals gives the resulting sound. The acoustic body is not needed. This method is superior to attaching a coil gently to the wood surface to pick up vibrations. Such pickups are called *pressure transducers*. Pressure variations are converted into electrical signals. The wood plays the role that the diaphragm does in a microphone in such an arrangement. The electric guitar does not use such body vibrations. Therefore its outside design has evolved into something quite different from the acoustic guitar. The development of the modern electric guitar goes back to the work of musician-inventor Les Paul in the 1940s. The manufacturer *Gibson* worked with Les Paul in the 1950s to produce top-quality electric guitars.

Fig. Y-13. Electric Guitar.



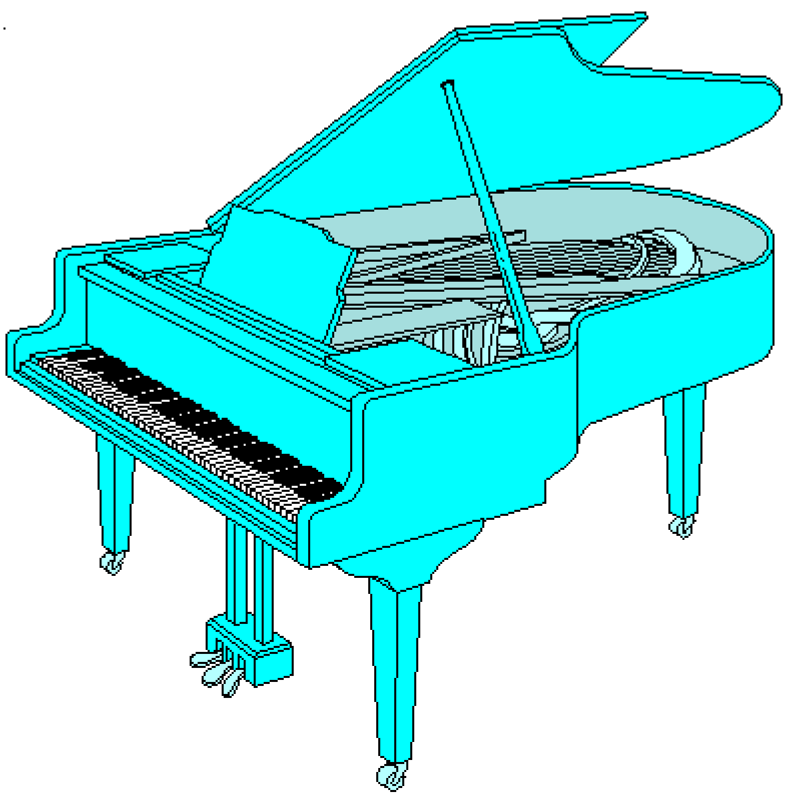
The Piano

Hammers strike the strings in a piano. The piano (see Fig. Y-14) is considered a percussion instrument. The strings are under great tension. A tuning instrument is needed to change the string tension. A lower string consists of a metal string wound on another metal piece. The lower notes have one string, middle tones have two, and the highest strings have three. The strings get shorter (Mersenne's First Law) and thinner (Mersenne's Third Law) toward the top.

The middle octave is tempered equally. Then other strings are tuned by octaves. Piano strings present some tuning challenges. Piano strings are thicker in order to withstand the greater tension. Such strings lose some flexibility and begin to behave to some extent like metal rods. The higher modes of excitation for a metal rod do not correspond to the overtone series of strings and open pipes. Therefore, the second harmonic of the piano string falls a little beyond twice the frequency of the fundamental. To avoid having the note an octave higher beat with the slightly-raised second harmonic of the lower octave, the piano tuner stretches the tuning of the higher octave. The higher octaves on the piano get tuned slightly sharper and sharper, the lower octaves flatter and flatter.

It is apparent that a tuning expect is needed. Pianists do not tune their own pianos unless they happen to be piano tuners also. The piano was invented (early 1700s) since its precursor, the harpsichord, played over a very limited range of loudness (*dynamics*). The new instrument was called the *pianoforte* (soft-loud). The research and development by *Steinway* in the 1800s contributed significantly to the modern piano of today.

Fig. Y-14. Piano.



Drums and Cymbals

Drums are vibrating membranes. The two-dimensional surface of a membrane is much more complicated than the narrow string or pipe, both of which can be considered as one dimensional. The kettledrum is illustrated in Fig. Y-15. A set of 2, 3 or 4 usually make up the timpani in an orchestra. Each is tuned to a pitch. The membrane produces the pitch. The cavity below enhances the sound. The inside is not slender like a pipe, but wide. Therefore, thin-pipe physics does not apply. In thin pipes, standing waves are set up; in wide containers, the mass of air swishes around. The result is a characteristic low-emphasis of frequencies (see Fig. Y-16). This jar-type of resonator is called a *Helmholtz resonator*, which we encountered earlier in discussing speaker design.

Fig. Y-15. Kettledrum.

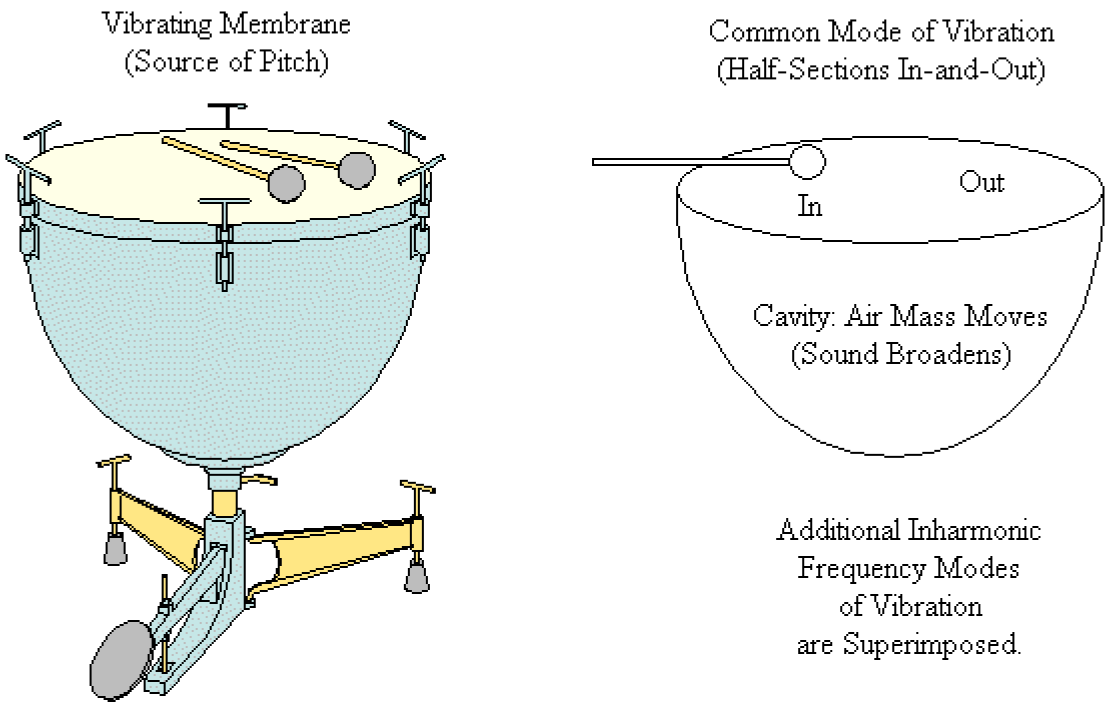
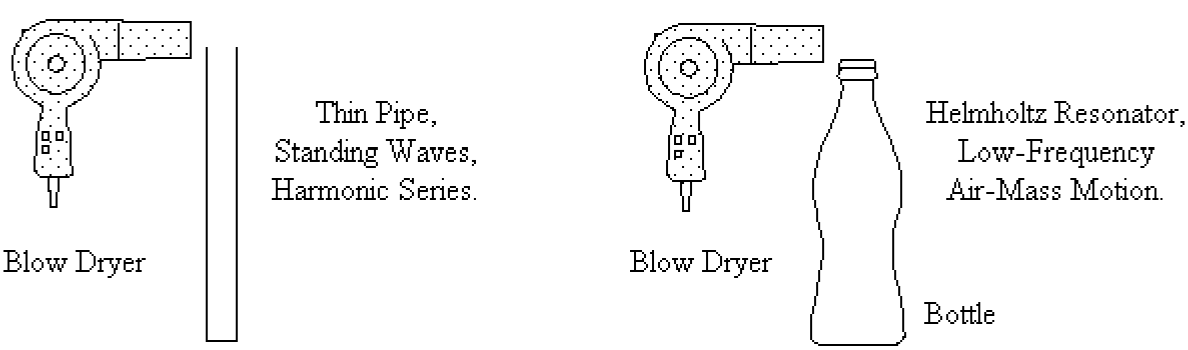


Fig. Y-16. Thin-Pipe Resonator Compared with Helmholtz Resonator.



Other drums do not produce a pitch. See Fig. Y-17 for examples of such drums. These drums produce broadband noise. The cymbals produce a concentration of high-frequency noise.

Fig. Y-17. Band Drum Set.

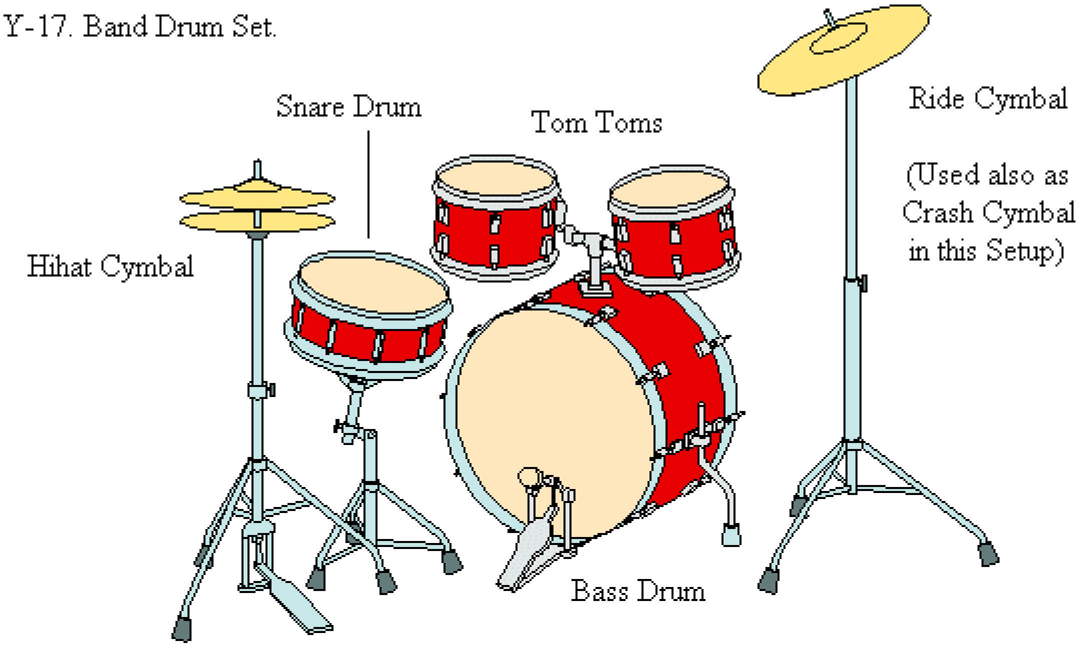
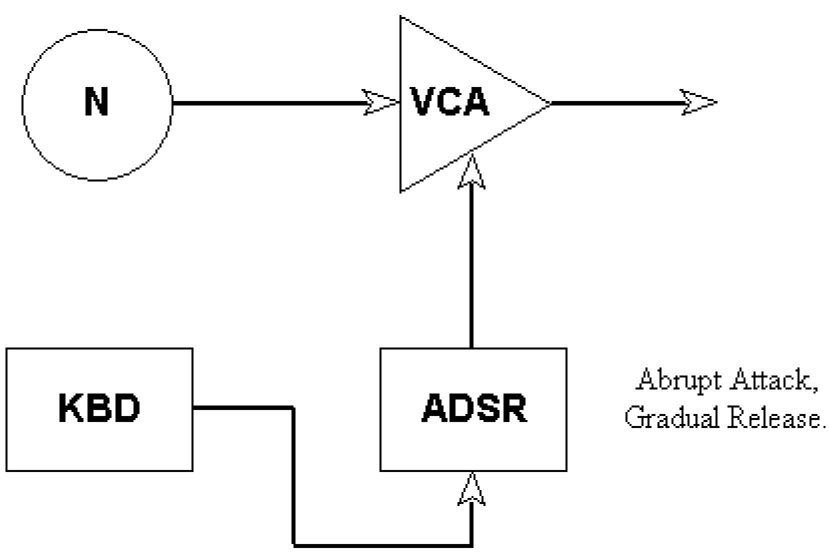


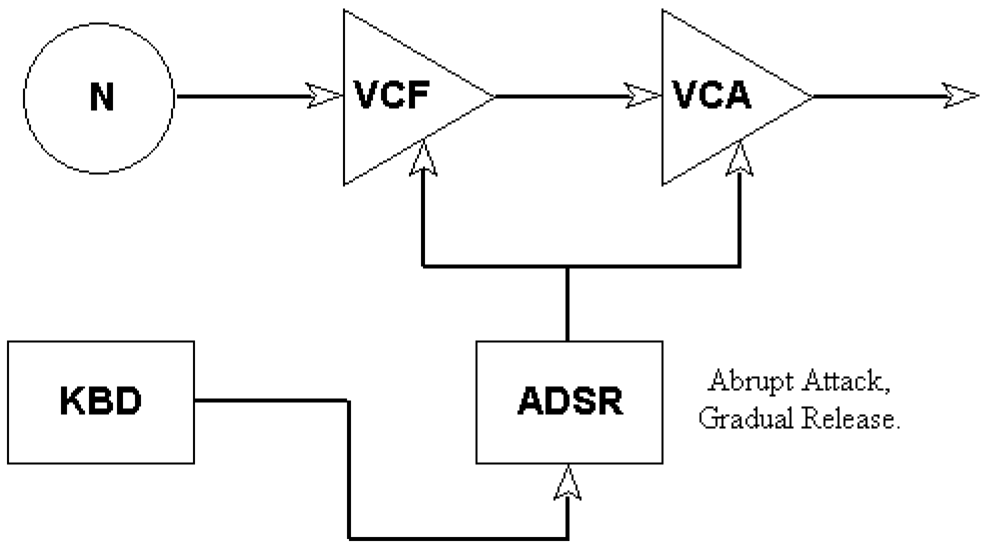
Fig. Y-18 illustrates a modular-synthesizer arrangement to produce band drums. A low-pass or high-pass filter can be used to highlight the lower or higher frequencies accompanying large or small drums. No filter-tracking should be employed since the keyboard is only used as a trigger for the ADSR.

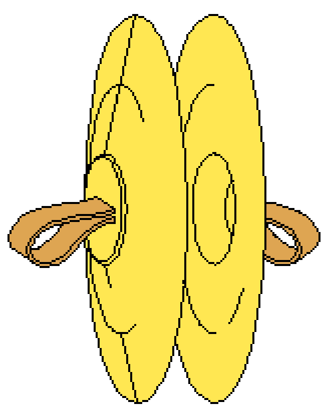
Fig. Y-18. Synthesizing Band Drums.



The cymbal produces a crash. The sound changes rapidly. A sweeping filter produces changes in the spectrum of a sound. Fig. Y-19 employs a sweeping filter to simulate a fleeting change in the broadband spectral characteristics of the noise. Better control can be achieved with two ADSR units, where one sweeps the filter while the other shapes the amplitude. However, the spirit of our modular diagrams is to assume we have one of each unit to work with.

Fig. Y-19. Approximating a Cymbal Sound.





Some Exercises

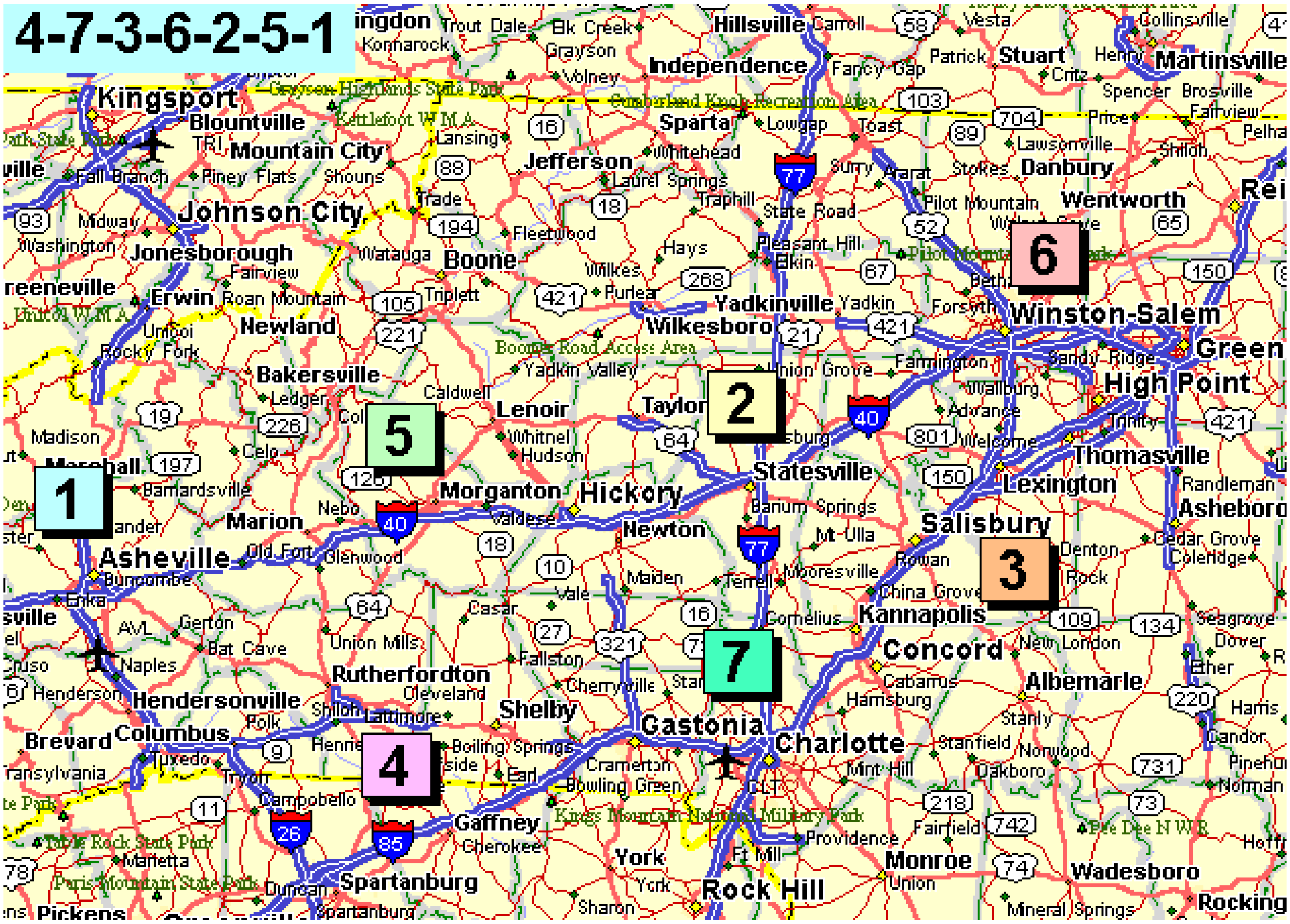
Statement	T	F
Drums typically have abrupt attacks.	○	○
Cymbals produce sounds with many combined frequencies.	○	○
Drums and cymbals have long sustain phases (ADSR).	○	○
Air mass moves in kettledrums.	○	○
The kettledrum resonates like a closed pipe.	○	○
The kettledrum is a Helmholtz resonator.	○	○

Answer below either true or false for synthesizing a cymbal sound.

Statement	T	F
Choose the noise generator for the source.	○	○
Choose a ramp wave for the source.	○	○
Do not use a VCA.	○	○
Use the LFO to control the VCA.	○	○
Use the VCF if "colored noise" is desired.	○	○
Use a gradual release on the ADSR.	○	○

--- End of Chapter Y ---

Z. Circle of Fifths



Consonance

Fig. Z-1 below shows the most consonant intervals after the 1:1. The best blend is the unison, where you compare a note with itself, i.e., the 1:1. The next best is the octave with its 2:1 ratio: you go from Do to "Do junior," the beginning of the next generation. So this is still in the family. The first outside one's family is the fifth (3:2).

Fig. Z-1. Consonant Intervals.

```
1       2       3       4       5       6       7       8

Do      Re      Mi      Fa      Sol     La      Ti      Do'

|<-------------------------------- 2 : 1 ------------------------------->| Octave

|<---------------- 3 : 2 ---------------->| Fifth

|<----------- 4 : 3 ----------->| Fourth

|<------ 5 : 4 ----->| Third
```

This is the secret behind the circle of fifths. You move by a fifth each time. Most of traditional western music moves by a fifth. The same goes for popular music and jazz.

The Basic Idea of the Circle of Fifths

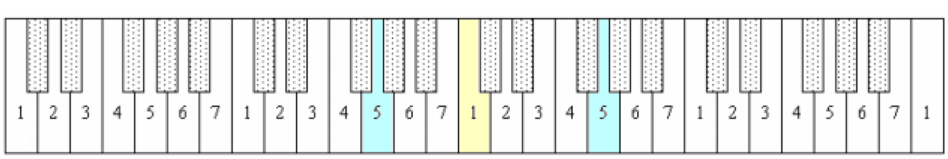
The *circle of fifths*, also called the *cycle of fifths,* is the key to understanding harmonic changes. You move by a fifth from Do to Sol harmonically. Then you consider Sol your starting point and move a fifth again. You continue in this fashion. It is like moving from your Facebook page to your friend's page. Then you go to a friend of your friend. We shortly discover which degrees of the scale mark the addresses of these Facebook members. First though we analyze an argument your instructor had with a jazz musician.

Many years ago your instructor got into an argument with a jazz musician who insisted that the best harmonic changes followed the *circle of fourths* and not fifths. The argument became quite intense. But then your instructor reasoned that his musician friend must be right too. How can both be right? Your instructor finally discovered the answer.

See Fig. Z-2. If you go up a fifth from the middle note labeled 1, you get to the 5th degree of the scale. But if you go down a fourth, you get to the 5th degree of the scale in the octave lower.

So moving up by a fifth is equivalent to moving down by a fourth. In either case you get to the 5th degree of the scale.

Fig. Z-2. The Fifth and the Fourth.



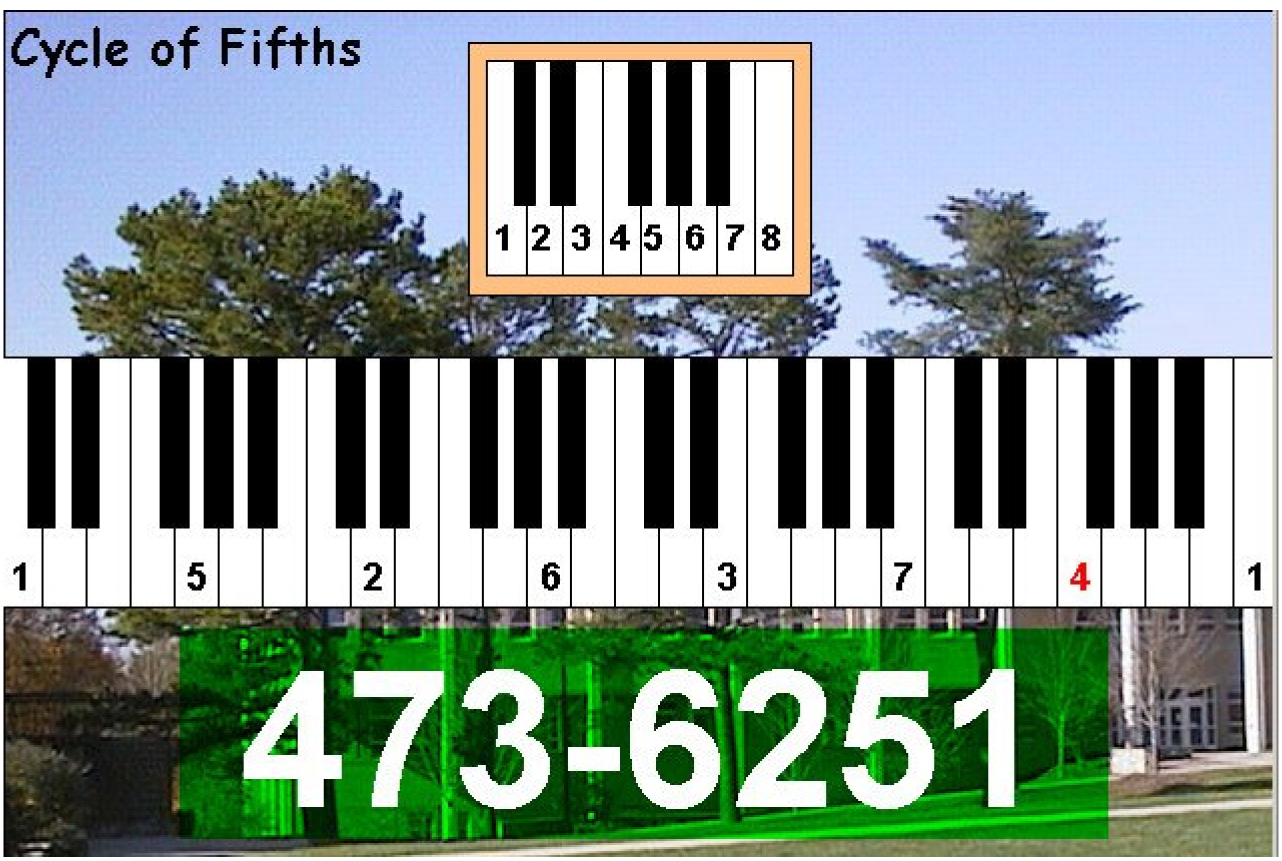
Be careful here that when you move by a fifth that you count 7 half steps. When you move by a fourth, you count 5 half steps. Here is a mathematical proof that moving up by a fifth is the same as moving down by a fourth. Start with 120 Hz so that we can work with a specific frequency. Then your fifth higher is (120) 3/2 = (60)(3) = 180 Hz. If you go down a fourth, you need to deal with the 4:3 ratio in some way. Going down means we use 3/4. Therefore, (120)(3/4) = (30)(3) = 90 Hz. But this is an octave lower than 180 Hz. Therefore, we are on the same degree of the scale, a Sol in each case.

Here's a more abstract version. Go up a fifth means 3/2. Go down a fourth means 3/4. But this latter one is an octave lower. Why? Double it and you find 2(3/4) = 3/2, the higher Sol. So both the lower and higher notes are Sols. Remember the importance of seeing things from more than one vantage point.

A Portion of the Circle of Fifths

Start on the 1 in the large keyboard of Fig. Z-3 and count 7 half steps to arrive at the 5. Use the cute small keyboard as a reference to identify the degrees of the scale. Keeping counting by 7 half steps to get all the cases but cheat in going from 7 to 4 making that only 6 half steps, i.e. the tritone. We do this so that we can finish up with the 1 with a total of eight numbers. So we break the cycle between 7 and 4 to pull this off. Note that the last interval (4 to 1) is a fifth. We do the break because popular songs are typically written in groups of 4 units called measures. So now we have two such units. Read the numbers backwards and you get the two units: 1-4-7-3 and 6-2-5-1. When you repeat this, the 1 appears twice in a row: 1-4-7-3-6-2-5-1-1-4-7-3-6-2-5-1-1. Remember this by the phone number 473-6251 and note that you stay on the 1 for twice as long.

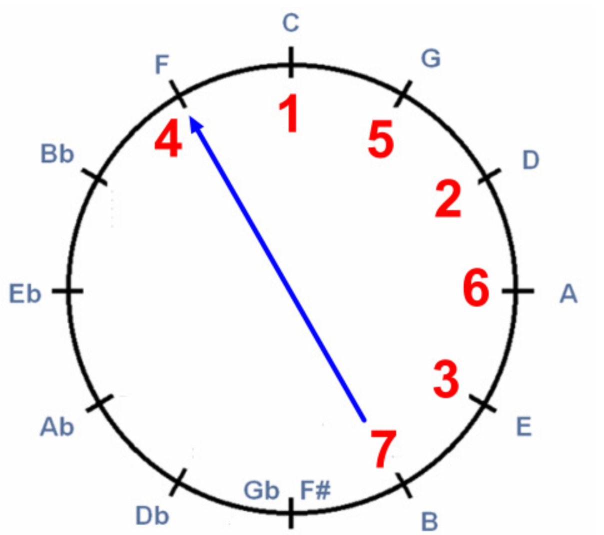
Fig. Z-3. Portion of the Circle of Fifths.



The Full Circle of Fifths

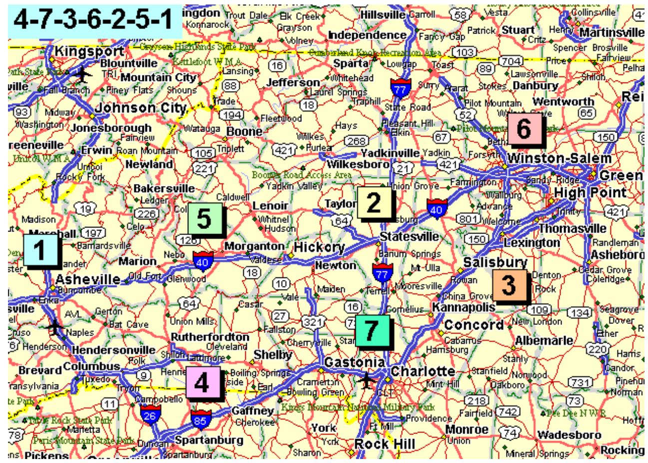
The full circle of fifths is given in Fig. Z-4 with letters. The musicians like to use a circle with labels of the twelve unique notes of the keyboard. Don't worry about the letters. Some of these will be our black keys. We cheat in going from the 7 to the 4. That is powerful enough to analyze so many songs as we do in class. See Fig. Z-5 for a map analogy.

Fig. Z-4. Circle of Fifths and Our Portion.



Note that the subsection we pull out, which has to have the tritone splice (7-4) is our 4-7-3-6-2-5-1. Part of this is 6-2-5-1 and a part of the latter is 2-5-1. Some of the songs we analyze need only these parts. The larger cycle is like a powerful physics formula and the smaller pieces are simple components of the master rule.

Fig. Z-5. Map Analogy: Traveling to Friends of Friends.



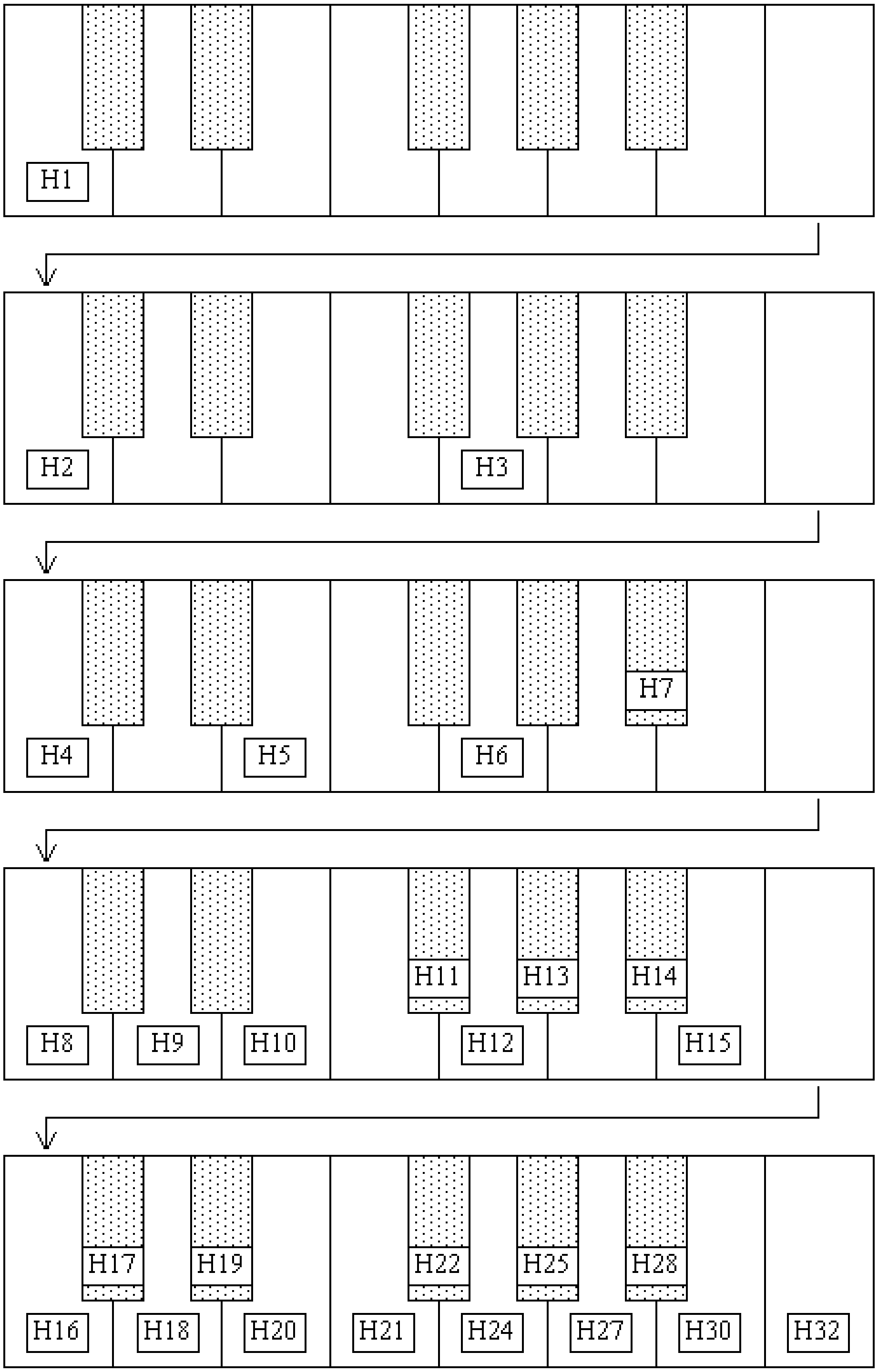
We need to stress that these movements by fifths musically refer to the harmony and not the tunes. The great Russian composer Rimsky-Korsakov (1844-1908) pointed out the importance of orchestrating a harmony with a blend of compatible notes. These are often chosen from the harmonics that go with the root or base note.

So what was Rimsky-Korsakov referring to when he wrote in his *Principles of Orchestration* that "the resonances of different harmonic parts must be equally balanced." Of course, *physics*!

--- End of Chapter Z ---

Appendix: *Harmonic Mysteries*

Placement of Harmonics on the Nearest Notes.



1. The Mystery of the 1st Octave: *Bass*.

2. The Mystery of the 2nd Octave: *Cycle of Fifths*.

3. The Mystery of the 3rd Octave: *Chords*.

4. The Mystery of the 4th Octave: *Scales* (Formulas).

5. The Mystery of the 5th Octave: the complete *Chromatic Scale* (H23, H26, H29, H31 not Listed).

6. The Mystery of the 6th Octave: *Microtonality* (exceeding the 12-tone scale), already suggested in the 5th mystery by H23, H26, and H31.

Glossary for Physics of Sound and Music (Ruiz)

Version March 9, 2020

acoustic guitar – musical instrument made of wood with 6 strings, frets, and a sound cavity

acoustic traumatic notch – an audiogram with a hearing loss where the loss is only in a small band of frequencies – thus the dip in the audiogram plot (graph), an upside-down V-notch

acoustical – relating to sound

acoustic waves – sound waves traveling in air, fluid, or solid

acoustics – the study of sound

ADSR – see envelope generator

alternating current (AC) – current that flows back and forth in a wire or circuit element

Ampère's law – electrical current through a coil produces a magnetic field inside the coil

amp (also ampère) – the unit for current; amp is abbreviated as A

amplifier – device that increases the amplitude of an electrical wave

amplitude – the measure of the wave strength from the equilibrium (center line) to the highest point of a wave. Sometimes engineers like to measure from the very bottom to the very top and thus obtain twice this value. To avoid confusion we refer to the later as "anti-peak-to-peak" amplitude or simply as "peak-to-peak" amplitude. Do not confuse this with measuring horizontally from one peak to the next, in which case you obtain the wavelength.

amplitude modulation (AM) – the change of the amplitude of a wave

analog signal – a signal that can take on any numerical value between its minimum and maximum values

AND – the result for $Y = A \cdot B$ with $Y = 1$ only if both $A = 1$ and $B = 1$

antenna – a metallic structure (can even be a single wire) used for the purpose of picking up an electrical signal. The incoming electromagnetic wave makes electrons in the antenna wiggle in step with the incoming wave.

antinode – a place on a wave that undergoes maximum change, e.g., displacement antinode means maximum changes in displacement (movement) at that point; pressure antinode (in a pipe) means maximum changes in pressure

antiskating force – the force that a little spring exerts outward on the bent record player arm so that the arm is not pulled in to the center of the record as the record is played anvil – the middle bone in the middle ear

aperiodic – not periodic

Armstrong, Louis – famous historical jazz trumpet player

attack – the time for the sound to reach a maximum level from its start. A short or abrupt attack is characteristic of an explosive or plucking sound.

audio in – phrase used to describe the input audio signal in electrical form entering a circuit section or module

audio out – phrase used to describe the output audio signal in electrical form leaving a circuit section or module

audio signal – electrical signal that when sent to an amplifier and speaker can be heard

audiogram – a medical record of an ear's ability to hear where hearing threshold in dB is plotted along the downward vertical axis and frequency along the horizontal axis at the top of the plot

audiology – the study of speech and hearing loss

auditory – pertaining to the perception of sound

auditory nerve – the "biological wire bundle" through which electrical signals are carried to the brain from the organ of Corti

aural harmonics – harmonics introduced by the ear due to distortion of the original waveform if the original sound is too loud or if there is some imperfection in the ear/brain system

Autumn Leaves – song from 1930s serving as our signature song in a minor key for the 4-7-3-6-2-5-1 progression in a minor mode or key

Bach, Johann Sebastian – baroque composer, master at using the cycle of fifths

balanced modulator – a device that accepts two sine waves as input and then sends out two sine waves, but one with the sum frequency and the other with the difference frequency

balanced modulation – process whereby the sum and difference frequencies of two sine waves are produced

baffle – a barrier with a hole in it so that a speaker can be inserted, thus preventing the out-of-phase waves leaving the rear from destructively interfering with the waves leaving the front of the speaker

bandpass filter – a filter that passes frequencies from some lower limit to some higher limit bandwidth – the difference of the maximum frequency and minimum frequency that defines the range of permitted frequencies in a bandpass filter

Baroque Period – period in European music from 1600-1750 characterized by grand long melodic lines

Bartók, Béla – modern Hungarian composer who lived in Asheville for 3 months due to health reasons, use of notes spaced by fifths to obtain an eerie beautiful sound

basilar membrane – membrane of length 3.5 cm in the cochlea on which sits the organ of Corti, which organ detects sound waves. The basilar membrane has varying degrees of stiffness, increasing as one nears the oval window. The membrane responds by resonance where the higher the frequency of the incoming sound wave, the closer to the oval window the sound is detected. Each octave corresponds to 3.5 mm and we can hear 10 octaves.

bass – low frequencies; also, the stringed instrument that produces the lowest pitches in the string family

bass control – amplifier control for low-pass filter so that you can adjust the strength of the lower frequencies

bassoon – a woodwind instrument that acts as an open pipe and has the lowest musical range of the four common woodwinds: flute, oboe, clarinet, and bassoon

battery – a device that separates plus and minus charges inside it and thus can produce a direct current when inserted in a circuit

beats – the pulsations that occur when two waves have nearly the same frequency

beat frequency – the pulsation frequency that occurs when two waves have nearly the same frequency. The beat frequency is given by the difference of the frequencies. The frequency of the actual tone undergoing the pulsations is given by the average of the frequencies of the two original waves.

Beethoven, Ludwig van – a transition composer that bridges the classic and romantic periods in music. He helps usher in the romantic period with music heroic in proportion.

bel – a unit of sound level (loudness)

bell jar – jar used to place a bell inside and pump out the air so that no sound can be heard. It is used to demonstrate that sound needs a medium to travel through while light does not (since you can see the bell when there is no air inside the jar).

Berlioz, Hector – romantic French composer who wrote his *Symphonie Fantastique* as sexual sublimation for his beloved Harriet Smithson since she rejected him. What happened after she heard the symphony? He introduces the Dies Irae them in the 5th movement of his symphony, which movement is called *Dream of a Witches' Sabbath*. He also includes Church bells, an example of inharmonicity.

Bernstein, Leonard – conductor who invited Louis Armstrong and other black musicians in the mid 1950s to perform *St. Louis Blues* with composer W. C. Handy and his wife in the audience. Bernstein also discovered the black pianist André Watts from Philadelphia.

Berry, Chuck – a prince of rock 'n roll, whose *Mabellene* (1955) neutralizes the blues with all 1-chords off and on throughout the piece

binary – two-valued, either 0 or 1, true or false

binary number – a number using 1s or 0s such as 1101, where reading from right to left you have 1s, 2s, 4s, 8s, etc. Therefore 1101 is equivalent to 1(8) + 1(4) + 0(2) + 1(1) = 13.

binaural beats – beats perceived in the brain when two pitches close in frequency are played separately into each ear low in volume to avoid bone conduction. The beats are still perceived in the brain even though there are no physical beats present. The waves do not mix in space, but only in the brain.

binaural effects – perceptual sound effects possible since we have two ears such as the perception of sound direction

blue noise – high-frequency noise

blues – song following the pattern of 12 sections (measures or bars) with harmonic content 1-4-1-1, 4-4-1-1, 5-4-1-1 where substitutions for harmony are allowed.

blues scale – a scale of six notes developed to blend well with the blues harmonic chord progression

Brahms, Johannes – romantic composer, study orchestration with Robert Schumann, lived with the Schumanns for a time, Uncle to the Schumann kids, composer of the *Academic Festival Overture*

brass – the orchestral instruments made of metal based on open-pipe physics. The four brass instruments of the orchestra are the French horn, trumpet, trombone, and tuba.

bridge – on a guitar, the place where the strings are fastened opposite to the end that contains the tuning pins

brilliance – term in acoustics when reverb is present for high frequencies but not much is there for low frequencies

broadband noise – white noise

Can't Take My Eyes Off You – song from 1960s serving as our signature song in a major key for the 4-7-3-6-2-5-1 progression. The mid section uses the progression.

capacitor – electrical component (in its basic design, two plates) that can store electricity

Carlos, Wendy – renown musician on the synthesizer. Her record “Switched-On Bach” which used the Moog Synthesizer in the late 1960s created a sensation.

carrier wave – the wave upon which a modulation is applied

cartridge – name given to record player component, fitting at the end of the arm, that translates the mechanical vertical motions of the stylus into electrical signals via the Faraday principle. The cartridge consists of the stylus at its bottom and two tiny magnets that can move inside coils at the top.

cello - the instrument in the orchestral string family that has a lower musical range compared to the viola but a higher one compared to the bass

Classic Period in Music - the time from roughly 1750 - 1800. The music, in contrast to the grand baroque period that preceded it, is known for its Simplicity, Order, Balance, elegance, and Restraint (SOBER). Two of its major composers were Haydn and Mozart.

CD – a compact disc

CD player – a device that converts binary data on a CD in the form of pits into sound using laser light to reflect off the pits

charge – an electrical plus or electrical negative

chord – the dressing up of a degree of the scale so that you obtain a harmonic group of tones to accompany a melody. Chords can be labeled such as the 1-chord, meaning a chord built on the first degree of the scale. The simple major chord is built using Do-Mi-Sol simultaneously – a triad. When you add a low bass note (H1), then Do-Mi-Sol become H4, H5, and H6 relative to your H1. Think of chords are including harmonics that go with a given degree of the scale.

chorus effect – the effect whereby a group of sound sources are producing the same sound, e.g., singers in a choir singing the same song

chromatic scale – the equal-tempered twelve-tone scale

circle of fifths – same as cycle of fifths

circuit – an electrical system with wires and electrical components

circuit element – an electronic component of an electrical system. The basic circuit elements are the battery (V), bulb (B), resistor (R), capacitor (C), coil or inductor (L), transistor (T), and diode (D).

clarinet – a woodwind instrument that acts as a closed pipe and has a lower musical range than the oboe and a higher musical range than bassoon

clarity – term in acoustics for little overall reverb

closed pipe – a pipe closed on one end and open on the other where we designate the pipe length as L. The natural modes of the longitudinal vibrations are the odd harmonics and they have frequencies f_1, $3f_1$, $5f_1$, etc., where f_1 is the frequency of the first harmonic (fundamental). The corresponding wavelengths are $\lambda_1 = 4L$, $\lambda_3 = 4L/3$, $\lambda_5 = 4L/5$, and so on. For each frequency f and its associated wavelength λ the wave relation is always true: $v = \lambda f$, where v is the speed of waves in the medium inside the pipe, the medium usually being air.

cochlea – the coiled region of the inner ear than contains the auditory detection mechanism

coil – a wire wound in the form of circles forming a cylindrical structure with empty space inside

colored noise – noise resulting after white noise is passed through a filter

Coltrane, John – jazz saxophonist from North Carolina who brought a modern dissonant "Stravinskyesque" sound to jazz. An example is his version of the Rodgers and Hammerstein *My Favorite Things*, which inspired the *Doors* for their improvisational section to *Light My Fire*.

compact disk – CD, a disk that contains digital information such as audio, video, text, images, etc.

complex periodic wave – any periodic wave that is not a sine wave

compression – a squeezing together of the medium in a longitudinal wave, analogous to the crest of a transverse wave

consonance – pleasing combination of tones. The five most consonant intervals in order from most consonant to least are 1:1 (unison), 2:1 (octave), 3:2 (fifth), 4:3 (fourth), 5:4 (third).

constructive interference – interference where the crest of one wave lines up with the crest of the other and so do the troughs. We obtain a bigger wave and the waves are said to be in phase.

contrabass – the bass stringed instrument, also simply called bass

control voltage – voltage used to control a synthesizer module

Coulomb’s law – like charges repel, unlike charges attract

crescendo –musical term for a gradual increase in loudness

crest – the part of a wave above the “sea level” reference line of the wave

current – the flow of electricity

cutoff frequency – the frequency in a filter which determines whether a signal can pass through a filter or not

cycle of fifths – the movement by fifths through the 12 notes in the scale. Songs typically use portions of the cycle.

cymbals – metallic plates slightly conical that crash together producing noise with a strong presence of high frequencies

damped wave – a wave that reduces its amplitude to zero as time goes on damped harmonic motion – harmonic motion that decreases in amplitude such that eventually the motion stops

decay – the time it takes for the sound to drop in amplitude after the attack phase until it reaches the sustain level

decrescendo – musical term for a gradual decrease in loudness

De Morgan's Theorems – First Theorem: $\overline{A \cdot B} = \overline{A} + \overline{B}$, which translates as NOT(A AND B) equals (NOT A) OR (NOT B), Second Theorem: $\overline{A + B} = \overline{A} \cdot \overline{B}$, which translates as NOT(A OR B) equals (NOT A) AND (NOT B)

dbx compander – tape circuit that compresses the signal strength when recording so that the 100-dB dynamic range is squeezed into 50 dB in order for the tape to handle it – then on playback, the circuitry expands the signal to achieve the full 100-dB dynamic range. Otherwise, the limited magnetic tape cannot faithfully produce a range of 100 dB on its own – you would reach a saturation point similar to an "overexposure" in photography.

Debussy, Claude – French impressionistic composer who composed the orchestral work *La Mer* (The Sea)

deci – the metric prefix for 1/10, e.g., a decibel is 1/10 of a bel

decibel (dB) – a measure of the loudness of a sound equal to 1/10 of a bel

decibel Scale – scale where a tenfold increase of sound sources translates to an additional 10 dB on the decibel scale and a twofold increase in sound sources results in an additional 3 dB on the decibel scale

degree of the scale – the location of the note in the major scale where Do is 1, Re is 2, Mi is 3, Fa is 4, Sol is 5, La is 6, Ti is 7, and Do' is 8.

delayed light – a circuit that allows a light bulb to stay on for a short time after you release the switch

demodulator – a circuit section that extracts the transmitted information contained in the modulated carrier wave

destructive interference – interference where the crest of the first wave lines up with the trough of the second and the trough of the first lines up with the crest of the second, thus canceling each other out. The sum wave is zero. The waves are said to be out of phase or 180 degrees out of phase.

devil's tone – a tritone interval, so called because of the perceived tension in the sound of the interval. A tritone is often used in the harmony of the 5-chord to heighten the tension to resolve to the 1-chord.

diatonic scale – the just diatonic scale, i.e., the major scale tuned to perfect ratios – Do (1:1), Re (9:8), Mi (5:4), Fa (4:3), Sol (3:2), La (5:3), Ti (15:8), Do' (2:1)

Dies Irae (The Day of Wrath) – the haunting Medieval theme in a minor key inspired by the Biblical Day of Judgment

digital sampling rate – the rate at which one samples the value of signal. For CD quality, the sampling rate needs to be approximately double the highest frequency we can hear. Therefore the sampling rate used is 2 x 20,000 Hz = 40,000 Hz = 40 kHz.

digitize – to convert analog information into digital information (strings of 1s and 0s)

diffraction – the bending of a wave due to passing through an opening, around a corner, or around an obstacle. The opening or obstacle size must be comparable to the wavelength of the wave.

dimmer circuit – circuit with a battery, light bulb, and resistance arranged in a single loop. The bulb glows dimmer due to the presence of the resistor.

diode – a circuit element that allows electricity to pass only in one direction

direct current (DC) – current that flows in one direction in a circuit section

discrete signal – a signal that can only take on specific values between its minimum and maximum value. When the unit increments are really small, the signal appears to be continuous (analog).

displacement – change in position, e.g., when the medium is moved away from its natural position (equilibrium) forming a crest (above equilibrium, i.e., positive displacement) or trough (below equilibrium, i.e., negative displacement)

displacement antinode – a place where maximum movement of the medium occurs

displacement node – a place where no movement of the medium occurs

dissonance – the opposite of consonance

Dolby – the system by which you boost the higher frequencies when you record on a cassette tape (Dolby recording filter) and then you play the tape back through a filter that reduces the higher frequencies back to normal (Dolby playback filter). The playback filter also reduces the annoying high-frequency hiss that is characteristic on playback without the filter.

Doppler effect – the change in pitch due to the relative motion between the sound source and the observer. The pitch is higher when the source and observer approach each other and the pitch is lower when the source and observer move away from each other.

driven oscillation – an oscillation resulting from an agent (driver) forcing a system to oscillate. The system being driven (the "drivee") oscillates at the same frequency as the driver but the amplitude of response depends on the frequency.

drum – percussive instrument with a vibrating membrane, i.e., a vibrating two-dimensional surface, producing a noise with an abrupt attack. A kettledrum also includes a tuned pitch since the cavity of the drum in this case acts as a Helmholtz resonator, i.e., encouraging the air mass to swish around and produce a low-pitch tone.

duty cycle – the percentage of the wavelength taken up by the width of a rectangular crest or pulse. A pulse wave with a 50% duty cycle is a square wave; a pulse wave with a small duty cycle is a pulse train wave.

dynamic range – the range of loudness from softer to louder sounds

dynamics – range of loudness or capability to produce on a musical instrument or electronic system both soft and loud sounds

eardrum – membrane that vibrates when sound enters the ear, serving as the boundary between the outer and inner ear

earplug – small padded material placed in the outer ear to decrease the sound level. Good ones can decrease sound levels by 30 dB and more.

echo – the reflection of sound from an object or surface

echolocation – locating objects by reflecting sound waves off the objects. A bat sends out high-frequency sound waves that reflect off objects and then return to the bat.

electric field – force field due to the presence of a charge. If the charge is positive, then the force field is such that a negative charge will be pulled in towards the positive charge and a positive charge will be repelled. The force field gets weaker as you move farther and farther away from the charge.

electric guitar – guitar that employs the Faraday principle to convert the vibrating metallic strings into electrical signals that are then amplified and sent out through speakers

electrical force law – Coulomb's law: like charges repel, unlike charges attract

electricity – moving charges

electricity and magnetism – phrase used to describe the four physical laws of electricity and magnetism and their study

electromagnet – magnet formed by passing current through a coil than surrounds a piece of iron or other suitable substance

electromagnetic pick-up – see pick-up

electromagnetic pick-up set – see pick-up set

electromagnetic wave (EM wave) – a transverse wave requiring no medium to travel in since a changing electric field creates a magnetic field (Ampère's law extended) and a changing magnetic field creates an electric field (Faraday's law). The wave propagates itself. All EM (pronounced E and M) waves, whether they are visible or invisible, travel at the speed of light (300,000 km/s).

electron – an elementary particle with charge –1 that serves as a building block in the make-up of atoms. The electrons exists in the region than surrounds the inner positive nucleus. It is the electrons in outer atomic shells of metals that travel as electricity in wires made of metals such as copper.

electronics – the use of electrical components to build devices such as radios

Ellington, Duke – jazz band leader, composer, pianist. His *Blues in Orbit* (1958) gives a modern version of the blues for the early years of the Space Age. His *Satin Doll* is built on the 2-5 and 2-5-1 sequence from the circle of fifths.

embouchure – the opening that serves as one open end of the orchestral flute over which the performer blows air to make the flute resonate

envelope – the amplitude shape of the sound spanning the attack, decay, sustain, and release phases

envelope generator (ADSR) – the synthesizer module that controls the attack time, decay time, sustain level, and release time of an audio signal and starts the process when the trigger voltage is received

equal-tempered scale – the twelve-tone scale where each note is a half-step interval away from each of its adjacent notes

equal temperament – tuning the notes on the piano so that all adjacent notes, i.e., all half steps, have the frequency ratio given by the twelfth root of 2

equalizer – device containing a group of bandpass filter circuits with amplifier controls so that you can adjust the sound level in each frequency band

equilibrium – the state of the medium when no waves are present

Eustachian tube – tube connecting the back of the mouth to the middle ear serving for pressure adjustments when we climb to high altitudes or return to low altitudes

Faraday’s law – a changing magnetic field inside a coil produces electrical current in the coil

fifth - the musical interval from Do to Sol, which outlines the beginning of “Twinkle, Twinkle Little Star,” which is also an interval of 7 half steps.). In the Just Diatonic Scale with perfect ratios, the interval of a fifth is 3:2.

filter (electrical) – a device that allows the passage of electrical waves with a certain frequency range

filter tracking – technique used in synthesizers where the filter cutoff shifts its frequency to adjust for the note you are playing, thereby producing a similar altering of the harmonics for all the notes you play

Fitzgerald, Ella – historical jazz singer who sings our signature 1-6-2-5 song *Heart and Soul*. She also appeared in a *Memorex* commercial breaking a glass with her amplified voice.

flash – name for a circuit where a capacitor first touches a battery to store electricity and then the capacitor touches a light bulb to release the stored electricity, making the bulb flash briefly

flashlight – a circuit with a battery and light bulb arranged in a loop

Fletcher-Munson Curves – a series of graphs published in the 1930s that shows the equivalent hearing threshold, and equivalent loudness levels, over all frequencies and loudness in human perception. Each equivalent curve has a distinct phon value. The phon value is set to agree with the decibel value at 1000 Hz.

flute – a woodwind instrument that acts as an open pipe and has the highest range of the four common woodwinds: flute, oboe, clarinet, and bassoon

fourth – the musical interval from Do to Fa, e.g., the beginning of “Here Comes the Bride,” which is also an interval of 5 half steps. In the Just Diatonic Scale with perfect ratios, the interval of a fourth is 4:3.

formant – an enhanced frequency region in a spectrogram due to the resonance structure of the system generating the sound. Resonances of some of the frequencies produced result in enhancements of those frequencies.

Fourier analysis – the breaking down of a complex periodic wave into its harmonics, including the amounts of each harmonic present (the harmonic amplitudes)

Fourier spectrum – the bar graph you get for the Fourier recipe of a particular periodic wave where the amplitude of each harmonic is represented along the vertical axis and the harmonic number or frequency along the horizontal

Fourier synthesis – the construction of a complex periodic wave by adding the appropriate amounts of its harmonics and lining the harmonics up correctly (the phases)

Fourier’s theorem – any periodic wave with frequency f can be built by using sine waves with frequencies f, 2f, 3f, 4f, and so on by adding appropriate amounts of each (and possibly needing to shift the phases of some of the harmonics before you add them)

French horn – a brass instrument with a second lowest musical range of the brass, the lowest being the tuba

frequency – the number of times something repeats during a time interval. The common example used often is the number of cycles per second, written as 1/s and defined as hertz, i.e., Hz.

frequency modulation (FM) – the change of the frequency of a wave

fret – a ridge on a guitar to mark the place where one should place a finger to obtain a note of the scale fullness – term in acoustics used when reverb is present for all frequencies

fundamental – the first harmonic in the harmonic series

Garner, Erroll – jazz pianist who played completely by ear, never learning to read music.

gating – an example of amplitude modulation where you apply a square-wave modulator to a sound wave such that you hear the sound during the crest-phase of the modulator and do not hear the sound during the trough-phase of the modulator

generator – device that generates electricity via the Faraday principle, e.g., moving a magnet in and out of a coil generates alternating current in the coil

Gillespie, Dizzie – jazz trumpet player, colleague of Charlie Parker, who together were forerunners of the bebop era (a fast-paced jazz era with 2-5 sequences inserted in the blues as well as dissonant improvisations).

golden rule – the rule that in constructing a guitar, the next fret should be placed 1/18 of the distance from the current fret to the bridge

graph – a plot of data where the vertical axis represents one characteristic or parameter and the horizontal axis another such as distance and time respectively. One can then say that we are plotting distance against time.

green noise – mid-frequency noise

Greensleeves – song dating back to Elizabethan times (1580). Modern harmonization makes this song a perfect fit for the 4-7-3-6-2-5-1 in the minor mode or key. The tune is popular today as “What Child is This?” during Christmas time.

ground – the part of the circuit where the minus side of the battery is found

guitar – instrument with 6 stings and a series of frets

hair cell – one of about 20,000 cells on the organ of Corti for the detection of sound waves of various frequencies from 20 to 20,000 Hz

half step – the musical interval where the ratio is given by the twelfth root of 2, which we can remember as the rounded-off value of 1.06. Going up a half step in frequency is analogous to gaining 6% in interest so that your original $100 becomes $106.

half wave – one half of the wavelength, e.g., a crest or trough of a sine wave

hammer – the first bone in the middle ear, the bone that attaches to the eardrum

Handy, William Christopher (WC) - jazz composer known as the "father of the blues"

harmonic – a sine wave with frequency in the series f, 2f, 3f, etc. where f is the first frequency harmonic motion – natural motion with smooth crests and troughs described by a sine wave such as that made by a mass attached to a spring. The same as simple harmonic motion.

harmonic series – the harmonics f, 2f, 3f, and so on, sometimes designated as H1, H2, H3, …

Haydn, Franz Joseph – composer of the classic period in music and known as the "father of the sonata"

hearing loss – a condition where the hearing threshold at any frequency tested is greater than 0 dB. This means that the frequency must be played louder than 0 dB for you to hear it.

hearing threshold – the dB level at which the sound needs to be produced so that you can barely hear it. One is typically tested at 125 Hz, 250 Hz, 500 Hz, 1000 Hz, 2000 Hz, 4000 Hz, and 8000 Hz. The normal threshold is 0 dB for all frequencies tested, which is analogous to 20/20 in vision. Results for each ear are plotted on an audiogram.

Heart and Soul – song from the 1930s that serves as our signature song for the 1-6-2-5 progression from the circle of fifths

Helmholtz resonator – a device shaped like a big empty apple cider jug. When you blow across the top, the entire air mass swishes around producing a very low pitch. This is unlike the longitudinal standing waves produced in narrow pipes.

hi-fi – short for high fidelity

high fidelity – a term dating back to the 1950s meaning that the reproduction of sound with vinyl records, early tape recorders, and the radio was of true fidelity – faithful reproductions compared to what was available prior to this time

high-pass filter – a filter that passes high frequencies

hiss – high-frequency noise, typically 5000 Hz and above

Holiday, Billie – famous historical jazz singer known as Lady Day

horn – a metal instrument based on open-pipe physics. See brass and the French horn.

house system – the external amplifier and speaker system in a room or auditorium

impressionism – period in the latter part of the 19th century which aims for an impression rather than detailed focus such as a catchy tune or sharp image in a painting. The two giants are the two Claudes: Debussy in music and Monet in painting. This period is included in romanticism for piano competitions but in the humanities and arts it is considered a separate period: romanticism more like 1800-1850 and impressionism more like 1850-1900.

inductor – another name used for a coil circuit element

inertia – the property that an object remains at rest or moves at constant velocity unless acted upon by a force. Sometimes used interchangeably with mass.

inharmonic partials – frequencies that are not part of the harmonic series

inharmonicity – term to describe a sound that consists of frequencies that are not in the harmonic series f, 2f, 3f, etc. for the tone or sound. Examples include bells, gongs, and noise.

in phase – two waves with the same wavelength are such that the crest of one wave lines up with the crest of the other wave. The phase shift of one with respect to the other is 0 degrees.

inner ear – the region of the ear beyond the oval window which transmits sound via fluid conduction and from which electrical signals are sent to the brain

interference – the effect due to superimposing two waves, i.e., adding two waves together

interval – the musical jump or span from one note to another, e.g., a fifth is the interval you span going from Do to Sol

intimacy – term in acoustics for the presence of reflections reaching the hearer within 20 ms after the direct sound reaches the hearer

Jobim, Antonio Carlos – Brazilian composer known as "Tom" who composed many bossa nova songs, e.g., *One Note Samba* which has a charming pair of 2-5-1 sequences in its middle section.

just diatonic scale – the major scale tuned to perfect ratios – Do (1:1), Re (9:8), Mi (5:4), Fa (4:3), Sol (3:2), La (5:3), Ti (15:8), Do’ (2:1)

Khachaturian, Aram – modern Armenian composer who used extensively the half step and small intervals, giving his music a non-western element, characteristic of Armenia

kettledrum – a drum that also includes a tuned pitch since the cavity in this case acts as a Helmholtz resonator, i.e., encouraging the air mass to swish around producing a low-pitch tone.

keyboard (KBD) – unit that sends out along the voltage-control wire a different voltage for each key that is pressed and sends out along the trigger control wire a common signal when any key is pressed down. The voltage-control wire connects to the VCO and the trigger control goes to the ADSR.

Laura – movie from 1944 where composer David Raksin uses four 2-5-1 progressions in the key song for the movie, also named *Laura*.

Lissajous figure – a stationary pattern formed when a horizontal wave motion is combined with a vertical wave motion because the frequency ratio can be expressed with whole numbers such as 1:1, 3:2, or 4:3

liveness - term in acoustics for the presence of sufficient reverb

longitudinal standing wave – a harmonic vibration in a pipe so named because the “dancing pattern” has a “kind of stationary characteristic” with its fixed displacement nodes and moving antinodes in between low-frequency oscillator (LFO) – synthesizer module that generates a periodic control voltage in the range from 0 to 25 Hz that is used to modulate a carrier wave

low-pass filter – a filter that passes low frequencies

lower sideband – the inharmonic frequency components on the left side of center in a spectrum due to balanced modulation

LRC circuit – a circuit consisting of an inductor (coil), resistor, and capacitor. L = inductor, R = resistor, C = capacitor.

kilo – the metric prefix for 1000, e.g., 1 kilosecond = 1000 seconds

kilowatt hour – useful for electrical costs since it includes the wattage (current you draw times the voltage you are using, i.e., P = IV) and also the time (hour). The typical charge is about 10 cents per kilowatt hour, i.e., 10 cents per kWh.

larynx – the space in the vocal track lying above the vocal folds

linear tracking – term used in record players where the arm is straight and thus there is no skating force to deal with

longitudinal wave – a wave where the medium moves parallel (or antiparallel) to the direction of propagation of the wave

loudness – the perceived strength of an acoustic wave as experienced by the ear/brain system where stronger perceived waves are said to be louder

LP – long-playing record. The LP record turns slowly at 33 and 1/3 rotations per minute in contrast to records turning at 45 rotations per minute and the very early ones that turned at 78 rotations per minute.

Mach speed – the speed in units where the value “1” stands for the speed of sound in the medium under consideration. At standard temperature and pressure, Mach 1 = 340 m/s = 1125 ft/s = 750 mi/h.

magnetic dipole – a magnet with its north and south pole. Also, small “baby magnets” residing on a magnetic tape.

magnetic force law – like poles (two norths or two souths) repel and unlikes attract

magnetism – term to describe magnets and their physical nature

magnetic field – force field that surrounds a magnet or a field that is produced when you send current through a coil with nothing inside the coil. The magnetic field is produced in the space inside the coil for such a situation. The farther you get away from the source of the field, the weaker its effects.

major key (song in) – a song that sounds happy because the third degree of the scale is used in the song as well as in the root harmony (1-chord)

major scale – Do, Re, Mi, Fa, Sol, La, Ti, Do’, which in the equal-tempered scale has these adjacent intervals: Do-Re (whole step), Re-Mi (whole step), Mi-Fa (half step), Fa-Sol (whole step), Sol-La (whole step), La-Ti (whole step), Ti-Do’ (half step)

Mancini, Henry – composer of film music. He employs two consecutive 5th intervals effectively in his theme for the movie *Condorman*.

Marsalis, Wynton – famous trumpet player who gives a nice analogy of playing the blues with playing within the boundary of a basketball court. Marsalis also likes to give the Albert Murray description of blues as a vaccination against real sadness.

mass – “stuff” that makes up any object. See inertia.

masking – the covering up of sound by one sound as in the sounds of a fan drowning out a conversation

medium – the environment such as air or a spring through which a wave can travel

Mersenne’s laws – three string laws: 1) longer strings have lower pitches, 2) greater string tension means higher pitch, and 3) heavier strings have lower pitches

microphone – a device that translates the mechanical motion of a diaphragm into an electrical signal via Faraday’s law

middle ear – the region of the ear from the eardrum to the oval window consisting of three tiny bones that vibrate in step with incoming sound waves

milli – the metric prefix for 1/1000, e.g., 1 millisecond = 1/1000 of a second

minor key (song in) – a song that sounds mysterious or sad because the root harmony (1-chord) is built using the minor third (tone a half-step lower than the third) and the song may use the minor third in the melody line

minor third – the musical interval from Do to a half step lower than Mi, i.e., an interval equal to three half steps

modular synthesizer – electronic circuits designed in modules that can synthesize and manipulate sound characteristics. See synthesizer.

modulation – changing a property of a wave such as its amplitude or frequency

modulator – the wave that modulates a carrier wave in some way

monopole – term that refers to a single north magnetic pole or single south magnetic pole, either of which have never been found alone. Magnets always come with a north and south pole.

moogerfooger – a balanced modulator. See balanced modulator and balanced modulation.

mouth cavity – open passage beyond the mouth that serves as a resonance chamber producing formants in the sound made by the vocal system

Mozart, Wolfgang Amadeus - composer of the classic period in music. A child prodigy and super genius composer often used as a "yardstick" to measure other composers.

Murphy's law – law named after electrical engineer, which law is commonly stated as “If something can wrong, it will.” Engineers make back-up systems because of this law and it is prudent to do so in real life situations, e.g., never let things go to the last minute.
Murray, Albert – author of *Stomping the Blues*, blues is like an antidote or vaccine.

Moog, Bob – inventor of the Moog Synthesizer. He spent the last 25 years of his life mostly in Asheville and associated with UNC Asheville. His company Moog Music is in downtown Asheville.

Moog synthesizer – synthesizer invented by Moog. See synthesizer for more about Moog as inventor of the synthesizer.

Moogerfooger – a balanced modulator developed by Bob Moog

music synthesizer – see synthesizer

musical range – the range of tones that can be produced by a musical instrument or singer. The musical range for the piano is about 7 octaves from the lowest note at 27.5 Hz to the highest pitch at 4186 Hz.

musical scale – a discrete set of tones from which one can compose a tune. Examples include the major scale, the common minor scale, blues scale, whole-tone scale, pentatonic scale (black keys only on the piano).

“Musician's Scale” – term used by your instructor to refer to the major scale (Do-Re-Mi-Fa-Sol-La-Ti-Do')

musical temperament – see temperament

NAND – the NOT of AND which can be written as $Y = \overline{A \cdot B}$

nasal cavity – open air passage beyond the nose that serves as a resonance chamber producing formants in the sound made by the vocal system

neurosensory hearing loss – hearing loss of specific frequencies due to damage of hair cells in the inner ear

neutron – an elementary particle with charge zero that serves as a building block in the make-up of the nucleus of atoms node – a place on a wave that does not change, e.g., displacement node means no displacement; pressure node (in a pipe) means no change in pressure

noise – term to describe the presence of all frequencies from 100 Hz to 10,000 Hz

noise generator (N) – synthesizer module that generates all frequencies (noise)

NOR – t he NOT of OR which can be written as $Y = \overline{A + B}$

NOT – the value $Y = \overline{A}$, where Y = 1 if A = 0 and Y = 0 if A = 1

oboe – a woodwind instrument that acts as an open pipe and has a slightly lower musical range than the flute and a higher musical range than the clarinet

octave – the musical interval from Do to Do' which starts the song "Somewhere, Over the Rainbow," which is also an interval of 12 half steps. The two notes form an interval of an octave if their frequency ratio is 2:1.

ohm – the unit of resistance; ohm is abbreviated by the Greek letter Ω (capital omega)

Ohm's law – the law V = IR, where V is the voltage, I is the current, R is the resistance and R is constant. However, the rule V = IR can always be applied whether R is constant or not.

open pipe – a pipe open on each end where we designate the pipe length as L. The natural modes of the longitudinal vibrations are called harmonics and they have frequencies f_1, $2f_1$, $3f_1$, etc., where f_1 is the frequency of the first harmonic (fundamental). The corresponding wavelengths are λ_1 = 2L, 2L/2, 2L/3, and so on. For each frequency f and its associated wavelength λ the wave relation is always true: v = λ f, where v is the speed of waves in the medium inside the pipe, the medium usually being air.

OR – the result for Y = A + B with Y = 1 if either A = 1 or B = 1

orchestra – large group of instruments including the strings, woodwinds, brass, and percussion

organ of Corti – the organ in the cochlea that detects sound waves and from which the auditory nerve goes to the brain

oscillation – the generic term for vibration or production of one cycle of a periodic wave

oscilloscope – an electrical measuring instrument that sweeps out a picture of an electrical wave

oval window – the boundary between the middle ear and inner ear

overtone – any harmonic above the fundamental (1^{st} harmonic). Thus the first overtone is the 2^{nd} harmonic, the second overtone is the 3^{rd} harmonic and so on.

overtone series – the overtones: H2, H3, H4, H5, and so on. When you include the fundamental with the overtone series you get the harmonic series.

out of phase – two waves with the same wavelength where the crest of one wave lines up with the trough of the other wave. The phase shift of one with respect to the other is 180 degrees.

outer ear – the region of the ear from the ear lobe to the eardrum

Parker, Charlie "Bird" – jazz saxophonist who incorporated the 2-5 in the blues formula

partial – any harmonic in the harmonic series H1, H2, H3, .. (f, 2f, 3f, …)

peak-to-peak amplitude – a measure equal to twice the amplitude

perfect ratio – a ratio of two whole numbers such as 2:1, 1:2, 3:2, 2:3, etc.

period – the time it takes to complete one cycle of anything that repeats such as a periodic wave

periodic wave – a wave pattern that repeats

periodicity pitch – the fundamental perceived by the ear/brain system even if no fundamental is present since H2 and H3 imply a periodicity of a fundamental H1. So the ear/brain system puts in the fundamental during the perceptual processing even though it is not physically there.

Peterson, Oscar – Jamaican-Canadian jazz pianist who had phenomenal technique. A Franz Liszt of Jazz Piano.

phase – the horizontal shift of a wave, where 360 degrees represent a shift of one wavelength. A phase of 180 degrees means you have shifted a wave by one-half wavelength so that a crest has moved over to where a trough was initially located. A phase shift of 90 degrees means you have shifted the wave by one-quarter wavelength.

phon – a unit that designates the same loudness value. A value of zero means we can barely hear it. Humans are not that sensitive to low pitches. Therefore, 0 phons at 20 Hz has to be 70 dB according to a scientific meter for us to barely hear it. The phons are matched with dB values at 1000 Hz. Therefore, 60 phons = 60 dB for perception at 1000 Hz.

“Physicist’s Scale” – term used by your instructor to refer to the scale you would get using just the harmonics: f, 2f, 3f, …

piano – percussive instrument with 88 keys and 88 hammer/string units (single strings for low pitches, double for intermediate pitches, triple for the highest pitches), all under high tension. The musical range is from 27.5 Hz to 4186 Hz, a span of a little more than 7 octaves.

pianoforte – name for the piano, meaning soft-loud in Italian. The piano was an innovation capable of soft and loud sounds depending on how hard you hit the keys. It was therefore named pianoforte, which was shortened to piano.

pick-up – a small coil near the metallic string of an electric guitar that picks up changing magnetic fields as the nearby metallic string vibrates. The vibrating string disturbs the magnetic field inside the coil, thus inducing electrical signals in the coil via Faraday’s principle.

pick-up set – a set of pick-ups so that each electric guitar string has its own pick-up. You can have more than one set of pick-ups on the guitar, where a second pick-up set is placed at another location along the strings.

pink noise – noise with a greater presence of low frequencies

pipe – see open and closed pipe

pitch – the perception of the frequency of a sound wave where higher frequencies are said to have higher pitch

physics – the study of the fundamental properties and laws of matter and energy

place theory of hearing – the idea that the location of the detection of sound along the 3.5-cm basilar membrane depends on the frequency. Each octave of 10 octaves is detected over a distance of 3.5 cm / 10 = 3.5 mm of the basilar membrane with the higher and higher frequencies being detected on the stiffer and stiffer sections of the basilar membrane. These stiffer sections are closer to the oval window.

plot – graph

pole – either end of a bar magnet: a magnetic north pole or magnetic south pole

power – the product of the current and voltage, i.e., P = IV. Your electrical company charges you for the power (which includes how much current you draw and at which voltage). But in addition, they charge you for the time duration you use it.

power supply – a device you plug in and obtain voltage so that you do not have to use batteries. A typical power supply in a lab is one that ranges from 0 to 5 volts, producing direct current.

preamplifier – an amplifier than enhances a very weak signal, which in turn is further amplified by another amplifier and then sent to a speaker for everyone to hear the sound

presbycusis – natural hearing loss of high frequencies as one ages

pressure antinode – a place where maximum change in pressure occurs

pressure node – a place where no change in pressure occurs, e.g., the open end of a pipe which is free and always takes on the atmospheric pressure of the surrounding environment

Prokofiev, Sergei – Russian modern composer using dissonance for beauty and innovative use of the 9th in form a chord to replace the usual classical chord ending with the 8th

propagation – the traveling of a wave. To propagate is the same as to travel in this context.

proton – an elementary particle with charge +1 that serves as a building block in the make-up of the nucleus of atoms pulse – a wave disturbance that does not repeat

pulse-train wave – a wave with a narrow pulse in each wavelength. The Fourier amplitudes are 1, 1, 1, 1, 1, 1, 1, 1, 1, etc. for the Fourier components H1, H2, H3, H4, H5, H6, H7, H8, H9, and so on

pulse wave – a wave consisting of a "rectangular building" and "courtyard." If the building takes up half the wavelength, you have a square wave. If the building is very narrow, you have a pulse train wave.

pulse-width modulation (PWM) – a form of timbral modulation where the pulse-width of a rectangular-shaped wave changes its width

pulse-width – the width of a rectangular-shaped wave crest

Q-value – a measure of how tall and thin the resonance graph is. High Q-value means tall and narrow width; low Q-value means short and wide.

quality factor – see Q-value

Rachmaninoff, Sergei – Russian composer who had a nervous breakdown after a poor performance of his first symphony. He was cured by the Moscow physician Dr. Dahl who kept telling Rachmaninoff positive things over and over again daily for more than three months. Rachmaninoff then wrote his Second Piano Concerto and dedicated it to Dr. Dahl.

radio – a device that receives electromagnetic waves and extracts the sound information for us to hear. It at least contains a tuner and a demodulator. It may also include an amplifier and speaker so that it is a self-contained unit with no need for an external amplifier and speaker.

Rainey, Ma - early jazz singer known as the "mother of the blues"

ramp wave – a wave with a ramp waveform. The Fourier amplitudes are 1, 1/2, 1/3, 1/4, 1/5, 1/6, 1/7, 1/8, 1/9, etc. for the Fourier components H1, H2, H3, H4, H5, H6, H7, H8, H9, etc.

Ravel, Maurice – French impressionistic composer

RC circuit – a circuit consisting of a resistor and capacitor, characteristic of filter circuits. R = resistor, C = capacitor

receiver – a console sound component that contains a radio and amplifier as well as inputs to the amplifier for a record player, tape deck, or CD. You may be able to connect all three at the same time but you will still need to buy a speaker system in order to hear anything.

record player – device that converts the mechanical vibrations of the stylus (as it rides over the hills and valleys of a vinyl record) into electrical signals via Faraday's law

reflection – the bouncing of a wave off a surface

rarefaction – a stretching of the medium in a longitudinal wave, analogous to a trough of a transverse wave red noise – low-frequency noise

refraction – a change in direction of a wave due to a change in the wave properties of the medium

Reissner membrane – protective membrane above the tectorial membrane in the cochlea

release – the time it takes for the sound to go from its sustain level to zero after the key is released

resistor – circuit element that limits the flow of electricity

resonance – the phenomenon occurring when a driven oscillatory system gives the greatest amplitude of response. The frequency at which this occurs is called the resonance frequency.

resonance circuit – the LRC circuit, the electrical analogy of the mechanical resonance system. L = inductor (coil), R = resistor, C = capacitor

resonance curve – the plot or graph of the amplitude response (vertical axis) versus the frequency (horizontal axis)

resonance filter – bandpass filter with a transmission graph that resembles the resonance curve

resonance frequency – the frequency that results in the greatest response of a driven oscillating system

reverb – echo. Also, a synthesizer module (REV) that adds reverb to the sound.

reverberation time (RT) – the time it takes for the sound level to drop 60 dB

ring modulator – same as balanced modulator, so called because the basic circuit design in the first models resembled a ring

Romantic Period in Music - roughly the time spanning 1800 - 1900. This period reacts against the simplicity and order of the classic period that preceded it. Romantic music can be excessive in nature, exaggerated, and over the top.

round window – the end of the sound path in the inner ear and located below the oval window

Saint-Saëns – romantic composer, use of many harmonics in his Third Symphony that includes an organ, composer of *Danse Macabre* (*The Skeleton Dance*) with its dramatic use of the tritone

sample and hold – phrase used to describe the fact the voltage in the control-voltage sent out by the keyboard in a synthesizer retains its value even after the key is released

sampling rate – see digital sampling rate saw tooth wave – a ramp wave

scala tympani - lower chamber in the unwound cochlea through which sound passes after being detected by vibrating a section of the basilar membrane

scala vestibuli – upper chamber in the unwound cochlea through which sound first enters before being detected by the basilar membrane

Schumann, Clara Wieck - super talented daughter of piano teacher Friedrich Wieck who was groomed by her father to be one of the best pianists in Europe. Also, romantic composer and wife of Robert Schumann, who also studied piano with her dad.

Schumann, Robert - romantic composer known for his bipolar nature producing many works during his manic phases and attempting suicide twice during depression stages. He died in an insane asylum, believed to be the result of contracting syphilis in his youth. He married his former piano teacher's daughter Clara.

scope – abbreviated form for oscilloscope, often used by personnel in physics and electronics labs

second – the musical interval from Do to Re, e.g., the beginning of the song “Doe a Deer, a Female Deer, …,” which is also an interval of 2 half steps (1 whole step). In the Just Diatonic Scale with perfect ratios, the interval of a second is 9:8.

semitone – a half step

seventh – the musical interval from Do to Ti, which was prominently used by John Williams in the theme to the movie *Superman* in 1978, which is also an interval of 11 half steps.). In the Just Diatonic Scale with perfect ratios, the interval of a seventh is 15:8.

sidebands – the inharmonic frequency components on either side of the center in a spectrum due to balanced modulation

simple harmonic motion – natural motion with smooth crests and troughs described by a sine wave such as that made by a mass attached to a spring. The same as harmonic motion.

sine wave – the motion made by a mass attached to a spring, the simplest and most natural waveform. The Fourier amplitudes are 1, 0, 0, 0, 0, 0, 0, 0, 0, etc. for the Fourier components H1, H2, H3, H4, H5, H6, H7, H8, H9, and so on. In other words, you just have one Fourier component – the fundamental H1.

sixth – the musical interval from Do to La, e.g., the beginning of the song “My Bonnie Lies Over the Ocean,” which is also an interval of 9 half steps. In the Just Diatonic Scale with perfect ratios, the interval of a sixth is 5:3.

skating force – the force on a bent record player arm that tends to pull the arm towards the center of the record as the record is played. This is counteracted with the antiskating outward force due to a little spring attached to the arm.

shock wave – a wave with a large "V" (in two dimensions) or "cone structure" (in three dimensions) that is created when an object travels faster than the wave speed in the medium. An example is a speeding motor boat making a V-formation in the water.

Smith, Bessie - early jazz singer known as the "Empress of the Blues"

SONAR – SOund Navigation And Ranging. One can determine the depth of water by noting the time it takes a sound wave to leave the boat to reach the bottom and reflect back to the boat. You need to use the speed of sound in water, which is 1500 m/s.

sonata - "classic" form in music consisting of exposition, development, and recapitulation. In the exposition there is the primary theme followed by a contrasting secondary theme.

sonic boom – an acoustic shock wave

source – term used for source of sound or source of electrical signal

speaker – device that converts electrical oscillations into the mechanical vibrations of a diaphragm via Ampère's law

spectrogram – a plot of frequencies on the vertical axis against time on the horizontal where louder sounds appear with thicker lines. If you view a spectrogram on a computer, it is possible that louder sounds are represented with brighter lines and/or different colors.

Spivey, Victoria - early jazz singer and known for singing "Black Snake Blues"

square wave – a wave with a square crest and upside-down square trough waveform. The Fourier amplitudes are 1, 0, 1/3, 0, 1/5, 0, 1/7, 0, 1/9, etc. for the Fourier components H1, H2, H3, H4, H5, H6, H7, H8, H9, and so on.

staccato – sound with an abrupt attack and abrupt release

standing wave – any single harmonic vibration on a string or in a pipe so called because the "dancing pattern" has "kind of a stationary characteristic" with its stationary displacement nodes and moving displacement antinodes in between each pair of displacement nodes

static electricity – excess charge on an object which can then attach itself to another object, e.g., rubbing a balloon on your arm will enable a balloon to stick to a wall due to static electricity

stirrup – the third bone in the middle ear, the bone that connects to the oval window

Stravinsky, Igor – Russian born modern composer with innovative rhythm and use of clashing harmonics (1/2 step apart) in his *Rite of Spring* (1913)

string – used as an ideal model for string physics. A string of length L is fixed on each end. The natural modes of the transverse vibrations are called harmonics and they have frequencies f_1, $2f_1$, $3f_1$, etc., where f_1 is the frequency of the first harmonic (fundamental). The corresponding wavelengths are λ_1 = 2L, 2L/2, 2L/3, and so on. For each frequency f and its associated wavelength λ the wave relation is always true: $v = \lambda f$, where v is the speed of waves on the string.

strings – instruments in the string family where vibrating strings are used to make sounds: violin, viola, cello, and bass. In the typical orchestra you might find a group of 10 violins categorized as the first violins, another 10 violins categorized as the second violins, about 8 violas, 10 cellos, and 6 basses.

stylus – a needle (best being diamond) that rides over the hills and valleys of a vinyl record

sum displacement – addition of the displacements for two waves or more, remembering that positive heights are above the “sea level” reference line and negative heights are below. You need to combine the displacements including the relevant plus or minus signs during your addition.

supersonic – faster than the speed of sound, i.e., greater than Mach 1

sustain – the level for the sound as the user keeps a finger on the key

synthesizer – electronic device producing sound where one of the inventors was Robert Moog (1934-2005). Moog was a resident of Asheville for roughly his last 25 years of his life and former Artist in Residence at UNC Asheville. He was also a former adjunct professor in our Department of Physics.

tape recorder – a device that converts a signal stored on a magnetic tape into electrical signals via the Faraday principle. The signal is stored on the tape by different orientations of “baby magnets” (magnetic dipoles) during the length of the tape. When a recording is made, Ampère’s principle is used to convert electrical signals to magnetic storage by aligning “baby magnets” on the magnetic tape as the tape is pulled across the recording head.

tape speed – the speed of tape in a tape recorder. Cassette tape speed is 1 and 7/8 inches per second, which we often approximate as 2 inches per second. The standard speeds in reel-to-reel tape recorders are double and quadruple that for the cassette. These standard speeds are 3 and 3/4 inches per second and 7 and 1/2 inches per second. There are even faster tape speeds such as 15 inches per second. The faster tape speeds allow for better sound quality.

Taps – a theme that is composed using only the harmonics H3, H4, H5, and H6

Tatum, Art – Black American Jazz Pianist legally blind since childhood due to early cataract complications. Supreme technique at the piano in a flamboyant “romantic” style. A Franz Liszt of Jazz Piano.

Tchaikovsky, Peter – romantic Russian composer known for his long melodic lines, long dramatic endings, e.g., use of the rapidly alternating 1-chord and 5-chord at the end of a work. Disastrous marriage, no consummation, because he was gay. Use of inharmonic sounds with bells and cannons in his *1812 Overture*.

tectorial membrane – membrane just above organ of Corti that interacts with hair cells on the organ of Corti as sound is detected by the cochlea temperament – a prescription for tuning, e.g., in equal temperament, one adjusts the frequency ratios from one note to the next so that each ratio is a constant 1.059…, the twelfth root of 2

theremin – early electronic device invented by Leon Theremin where an eerie pitch is produced depending on where your right hand is relative to a vertical metal rod. The closer to the rod, the higher the pitch. The left hand controls the loudness – when near a metal ring with your left hand, the sound level diminishes.

Theremin, Leon – inventor of the Theremin

third – the musical interval from Do to Mi, e.g., the beginning of the "Marine's Hymn," which is also an interval of 4 half steps (2 whole steps).). In the Just Diatonic Scale with perfect ratios, the interval of a third is 5:4.

timbre (or timber) – the perception of the waveform of a wave, which allows us to distinguish one instrument from another such as a flute from an oboe

timbral modulation (TM) – the change of the timbre of a wave

tinnitus – internal perception of sound as a ringing or buzzing when there is no sound present, occurring at places on the frequency spectrum where there is an actual hearing loss

tracking force – the force of that the stylus exerts on a vinyl record, the typical value being the weight of one gram of mass

trachea – the space in the vocal track lying below the vocal folds

transistor – circuit element that can act as a switch and amplifier. It has two input wires and one output wire. One input (B = Base) accepts a tiny amount of current and this activates the transistor so that lots can flow from the other input (C = Collector) and out the common output (E = Emitter).

transmission – the passing of an electrical signal through a filter or other circuit element

transposing – the act of changing the key of a song or scale from one starting frequency (Do) to a new starting frequency. All relative relationships in the song or scale are kept so that the song or scale is recognizably the same but higher or lower in pitch.

transverse wave – a wave where the wave disturbance moves sideways (perpendicular) to the direction of propagation of the wave

treble control – amplifier control for a high-pass filter so that you can adjust the strength of the higher frequencies

tremolo – the musical term for an amplitude modulation, i.e., periodic changes in loudness levels alternating between louder and softer levels

triad – musical term for three tones played simultaneously as a harmony, i.e., a chord triangle wave – a wave with a triangle crest and upside-down triangle trough waveform. The Fourier amplitudes are 1, 0, 1/9, 0, 1/25, 0, 1/49, 0, 1/81, etc. for the Fourier components H1, H2, H3, H4, H5, H6, H7, H8, H9, and so on.

trigger voltage – voltage control sent from keyboard (KBD) to the envelope generator (ADSR) to initiate the ADSR sequence

tritone – an interval corresponding to three whole steps

trough – the part of a wave below the “sea level” reference line of the wave. The “sea level” line is the horizontal equilibrium line drawn through the middle of the wave.

trombone – the brass instrument without values that extends in order to produce different notes and having a musical range a little smaller than that of the French horn

trumpet – the brass instrument with three valves and the highest musical range in the brass family

tuba – the brass instrument with three valves that has the lowest musical range in the brass family

tuner – a resonance electrical circuit that can tune in to pick up a radio signal. You tune in by changing either the capacitor value or inductor value (for the coil) in the circuit.

twelfth root of 2 – the number “a” such that when taken as a factor 12 times and multiplied, you obtain 2, i.e., a x a x a x a x a x a x a x a x a x a x a x a = 12. The value for a = 1.05946309…, which in class we take to be 1.06.

tweeter – speaker with a small light membrane to support the rapid motion of high frequencies, i.e., waves with short wavelengths

twelve-tone scale – the scale with twelve tones, each one half-step from each other

two-way speaker system – see two-way crossover network

two-way crossover network – a circuit that accepts an input signal and directs the low frequencies to the woofer and the high frequencies to the tweeter

“Twirl-a-Tune” – a corrugated plastic toy often going by various names which produces harmonics when twirled, beginning with the 2^{nd} harmonic (H2) and reaching higher harmonics as it is twirled faster and faster

tympani – a set of typically three kettledrums, one usually tuned to the 1 (first degree of the key of the piece being played), another to the 5, and the third to the 4 or whatever the composer calls for

ultrasound – sound with a frequency above the range of human hearing, i.e., above 20,000 Hz. Also, the image of a fetus made by ultrasound.

unison – the name given when two identical pitches are played together. Their ratio is 1:1. An example is going from Do to Do, the same note.

unit – in physics, this designates the word that goes with the number when you make a measurement, e.g., for a weight of 120 pounds (120 lb), the value is 120 and the unit is lb

upper sideband – the inharmonic frequency components on the right side of center in a spectrum due to balanced modulation

vacuum tube – a tube with air pumped out and electrical properties that serves as a component in a circuit. Two common vacuum tubes are the tube version of the diode and the tube version of the transistor.

VCA – see voltage-controlled amplifier

VCF – see voltage-controlled filter

VCO – see voltage-controlled oscillator

velocity – technically the speed you are going AND the direction you are going; however, often just used to represent your speed without a concern for the direction

vibrato – the musical term for a gentle frequency modulation where the frequency changes do not vary too far from the original pitch. The result is a quivering pitch characteristic of singers.

viola – the instrument in the orchestral string family that has a lower musical range compared to the violin but a higher one compared to the cello

violin – the instrument in the orchestral string family that has the highest musical pitches compared to the others such as the violas or cellos

vocal cords – see vocal folds

vocal folds – the vibrating biological component producing sound in the human vocal system, also referred to as vocal cords. The vocal-folds end is approximated as a closed pipe end.

vocal formants – formants in the sound spectrogram produced by the uniqueness of one's vocal system that includes the resonance cavities of the mouth and nose. The simple closed-pipe model of the vocal tract for a male adult gives a closed-pipe length of 15 cm and a fundamental of roughly 500 Hz. This is the first formant region or first formant, also called the first principal formant region or first principal formant. The second formant region corresponds to the next harmonic in a closed pipe, the third harmonic (1500 Hz). The third formant region is the fifth harmonic (2500 Hz) and so on.

volt – the unit for voltage; volt is abbreviated as V

voltage – the effective strength of a battery's ability to produce current

voltage-controlled oscillator (VCO) – synthesizer module that accepts a voltage that then determines the frequency of the electrical wave that is produced. A VCO can typically be set to produce a few different basic waveforms such as sine, triangle, square, ramp, and pulse train waves.

voltage-controlled amplifier (VCA) – synthesizer module where the amplitude of the audio signal entering the amplifier is changed according to the value of the control voltage applied to the amplifier

voltage-controlled filter (VCF) – synthesizer module where the timbre is altered as the audio signal entering the VCF is passed through a filter with the cutoff or central frequency determined by the VCF control voltage. The filter type (LP, BP, HF) is set by a switch on the filter.

warmth – term in acoustics when reverb is present for low frequencies but not much is there for high frequencies

wave – a traveling disturbance

waveform – the shape of one pattern of a periodic wave

watt – the unit for wattage

watt hour – see kilowatt hour

wattage – the power, given by the product of the current and voltage (P = IV)

Watts, André – outstanding black pianist from Philadelphia discovered by Leonard Bernstein

wavelength – the distance corresponding to one pattern of a periodic wave

whispering chamber – an elliptically-shaped room where sound from one of two special points (each called a focus) gets reflected by the chamber walls so that the reflected wave heads towards the other special point (the second focus)

white noise – fairly equal presence of all frequencies from 100 to 10,000 Hz. White noise is named after an analogy with white light since white light consists of all frequencies of colors.

whole step – two half steps

whole tone scale – the six-note scale where each note is a whole tone step away from its two closest neighboring notes

Williams, John - composer of music for film. In *Superman* he employs the dissonant 7th interval in an effective way.

wire – electrical component made of metal such as copper that allows for the passage of electricity. The resistance of a wire in a circuit can be assumed to be zero, i.e., R = 0 ohms. This assumption breaks down if the wire gets to be extremely longer and longer. Wires in a circuit are typically very short.

woofer – speaker with larger membrane to support the slower motion of low frequencies, i.e., waves with long wavelengths

woodwinds – originally made of wood, instruments where the performer blows air against an edge or reed to excite the pipe into resonance. The basic woodwind types in the orchestra are the flute, oboe, clarinet, and bassoon. There are two of each in the standard orchestra.

XNOR – the NOT of XOR

XOR – the Exclusive OR. For A and B as input, you get a 1 if either A or B is 1 but not both.