pt

TERC

PHYSICS OF TECHNOLOGY

COORDINATED BY AMERICAN INSTITUTE OF PHYSICS

THE LOUDSPEAKER

Sound Waves and Wave Models

THE LOUDSPEAKER

A Module on Sound Waves and Wave Models

TERC

J. Edward Neighbor, Northeastern University
Ronald S. Newbower, Harvard Medical School

John W. McWane, Project Director

MCGRAW-HILL BOOK COMPANY

NEW YORK
ST. LOUIS
DALLAS
SAN FRANCISCO
AUCKLAND
DÜSSELDORF
JOHANNESBURG
KUALA LUMPUR
LONDON
MEXICO
MONTREAL
NEW DELHI
PANAMA
PARIS
SÃO PAULO
SINGAPORE
SYDNEY
TOKYO
TORONTO

The Physics of Technology modules were produced by the Tech Physics Project, which was funded by grants from the National Science Foundation. The work was coordinated by the American Institute of Physics. In the early planning stages, the Tech Physics Project received a small grant for exploratory work from the Exxon Educational Foundation.

The modules were coordinated, edited, and printed copy produced by the staff at Indiana State University at Terre Haute. The staff involved in the project included:

Philip DiLavore . . . . Editor
Julius Sigler . . . . Rewrite Editor
Mary Lu McFall . . . . Copy and Layout Editor
B. W. Barricklow . . . . Illustrator
Stacy Garrett . . . . Compositor
Elsie Green . . . . Compositor
Donald Emmons . . . . Technical Proofreader

In the early days of the Tech Physics Project A. A. Strassenburg, then Director of the AIP Office of Education, coordinated the module quality-control and advisory functions of the National Steering Committee. In 1972 Philip DiLavore became Project Coordinator and also assumed the responsibilities of editing and producing the final page copy for the modules.

The National Steering Committee appointed by the American Institute of Physics has played an important role in the development and review of these modules. Members of this committee are:

J. David Gavenda, Chairman, University of Texas, Austin
D. Murray Alexander, DeAnza College
Lewis Fibel, Virginia Polytechnic Institute & State University
Kenneth Ford, University of Massachusetts, Boston
James Heinselman, Los Angeles City College
Alan Holden, Bell Telephone Labs
George Kesler, Engineering Consultant
Theodore Pohrte, Dallas County Community College District
Charles Shoup, Cabot Corporation
Louis Wertman, New York City Community College

This module was written and tested at the Curriculum Development Laboratory of the Technical Education Research Centers, Inc.

The authors wish to express their appreciation for the help of many people in bringing this module to final form. The criticisms of various reviewers and the cooperation of field-test teachers have been most helpful. Several members of the staff of the Technical Education Research Centers also deserve special recognition for their contributions. They are:

Richard R. Lewis . . . . Apparatus Design
Mary A. Heffernan . . . . Graphic Composition
John W. Saalfield . . . . Illustration

In addition, special thanks go to our physics consultants:

Nathaniel H. Frank, Massachusetts Institute of Technology
Ernest D. Klema, Tufts University

The Loudspeaker

Copyright © 1975 by Technical Education Research Centers. All rights reserved. Printed in the United States of America. No part of this publication may be reproduced, stored in a retrieval system, or transmitted, in any form or by any means, electronic, mechanical, photocopying, recording, or otherwise, without the prior written permission of the publisher.

Except for the rights to material reserved by others, the publisher and copyright owner hereby grant permission to domestic persons of the United States and Canada for use of this work without charge in the English language in the United States and Canada after January 1, 1982. For conditions of use and permission to use materials contained herein for foreign publication or publications in other than the English language, apply to the American Institute of Physics, 335 East 45th Street, New York, N.Y. 10017.

ISBN 0-07-001732-8

1 2 3 4 5 6 7 8 9 0 E B E B 7 8 3 2 1 0 9 8 7 6 5

PREFACE

ABOUT THIS MODULE

Its Purpose

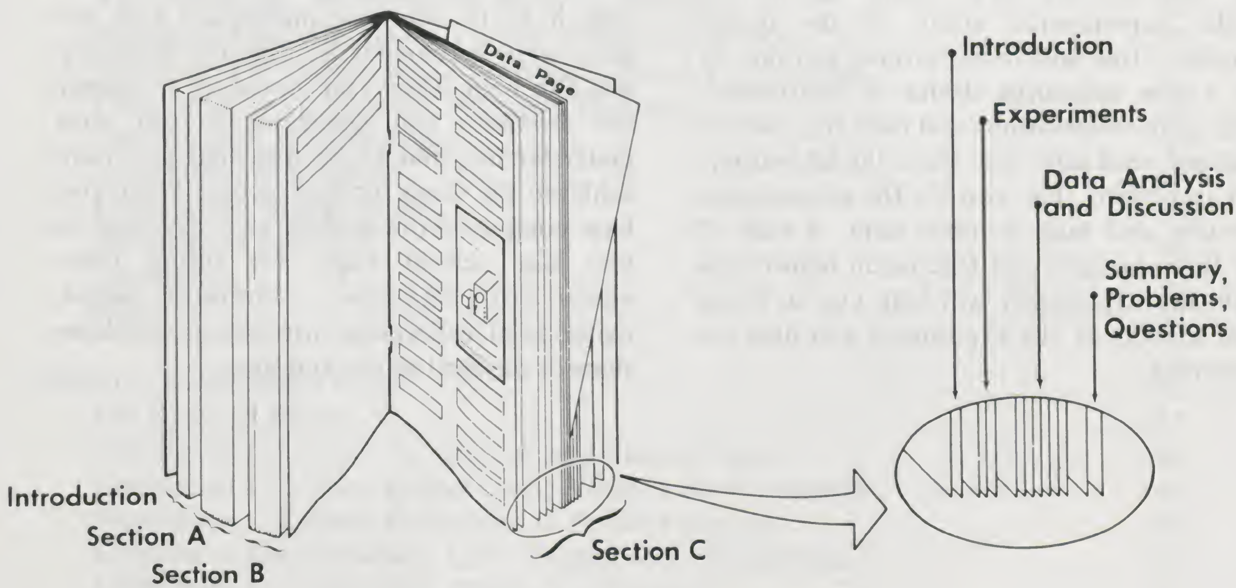
The purpose of the Physics of Technology program is to give you an insight into the physical principles that are the basis of technology. To do this you are asked to study various technological devices. These devices have been chosen because their operation depends on or illustrates some important physical phenomenon. In this module the device is the loudspeaker. Its design and use are based on the properties of sound.

The PoT program has adopted a modular format with each module focusing on a single device. Thus you can select only those modules that relate to your own interests or areas of specialty. This preface highlights some of the features of the modular approach so that you may use it efficiently and effectively.

Its Design

The *Introduction* explains why we have selected the loudspeaker to study and what physical principles will be illustrated by its behavior. Learning *Goals* are given, as well as *Prerequisite* skills and knowledge you should have before beginning. The three *Sections* of the module treat different aspects of the device. They are of increasing difficulty, but each can be completed in about one week.

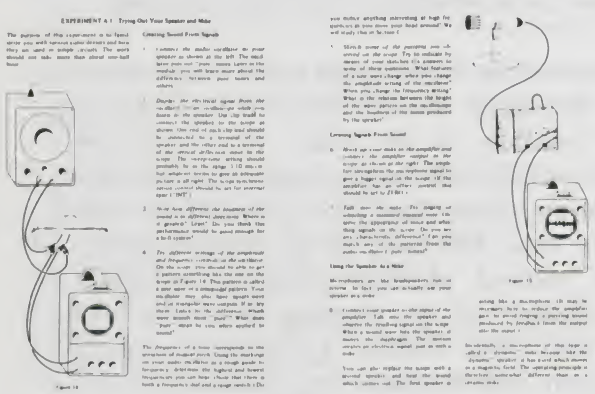
Each section begins with a brief *Introduction* to the topics treated and how they relate to the behavior of the device. The *Experiments* follow and take about two to three hours. Tear-out *Data Pages* are provided to record your data. The body of the section describes the method of *Data Analysis,* including a *Discussion* of the physical principles which explain your results. A *Summary* ends the section with *Problems* and *Questions* you can do to test your understanding.



HOW TO USE THIS MODULE

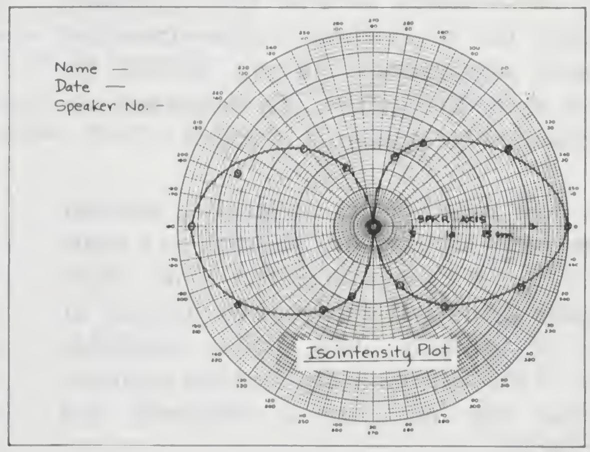
To Begin

This module has been written so that it can be quickly and easily scanned. That is, you can get the gist of the ideas and experiments by simply flipping from page to page, reading only the headings and italicized words, and looking at the illustrations. We suggest that, before you begin a section or an experiment, you scan through it in this way so that you will know where you are going.



The Data Analysis

The data you take will generally have to be graphed before it can be analyzed. Graphing and graphical analysis are essential parts of experimental science. And understanding graphs is important for technology since technical information is often presented graphically. For these reasons, and since the discussion of your results will be centered around your graphs, it is important that you prepare them clearly and accurately.



The Experimental Activities

The heart of a Physics of Technology module is the experimental study of the device behavior. This will often involve learning to use a new measuring device or instrument. Since your observations and data may not be analyzed until *after* you leave the laboratory, it is important that you do the experiments carefully and take accurate data. A scan of the Data Analysis and Discussion *before* you begin the experiment will help you to know what aspects of the experiment and data are important.

Review

When you finish a section you should again scan it to be sure you understand how the ideas were developed. Reading the *Summary* will also help. Then you should try to answer the *Problems* and *Questions* to test your understanding and to be sure that you have achieved the Goals for that section. When you have completed the module, you may want to tear out certain pages for future reference: for example, conversion tables, methods of calibrating instruments, explanations of physical terms, and so on.

TABLE OF CONTENTS

The Loudspeaker

INTRODUCTION: WHY STUDY LOUDSPEAKERS?

Everyone Uses Speakers

Loudspeakers are found in telephones, TV's, tape recorders, Hi-Fi's, AM-FM radios, intercoms, and so on. Count them: A typical family may own ten or more loudspeakers! Surely speakers are the most common "household appliances," next to light bulbs and motors.

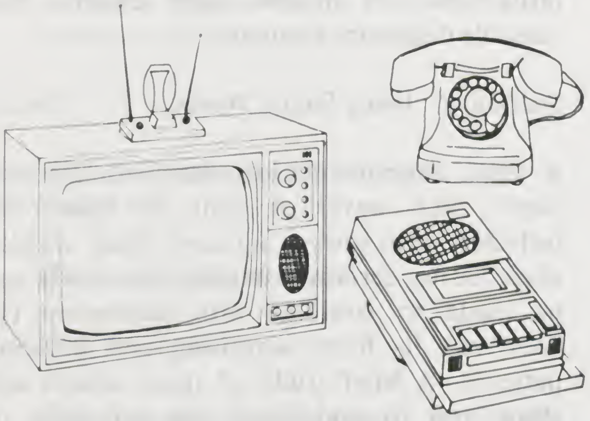


Figure 1. Everyone uses speakers.

Loudspeaker Principles Have Wide Application

Although there are many different types of loudspeakers, they all produce sound from electrical signals.

Devices which convert power from one form to another are called *transducers.* To convert electrical power to mechanical power (of a vibrating surface), most speakers use electric currents and magnets. Thus, knowledge of how speakers work helps in understanding some of mankind's most important related devices–including motors and generators.

The Principles of Sound Have Wide Application

Many of the physical properties of sound are shared by light, x-rays, TV and radio signals, ocean swells–even earthquakes. All of these are *waves,* although what is "waving" is different in each case: perhaps water, air, or rock. All waves have some properties in common, and their behaviors are essentially similar. Thus, studying one type of wave is like studying them all at once.

Sound itself is important in daily life for communication and entertainment, as well as for protection. In addition, sound and *ultrasound* (sound too high-pitched for human ears to hear) have increasing uses in technology: for drilling, for cleaning, for shaping metal, and for non-destructive testing.

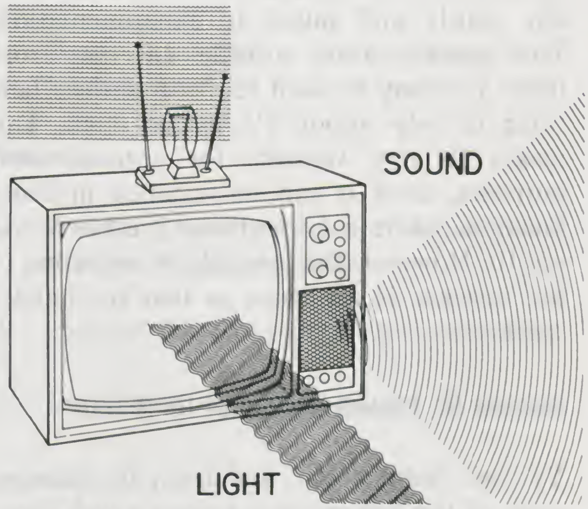


Figure 2. All waves have properties in common.

WHAT WILL YOU LEARN?

Section A: Generating and Detecting Sound

Using common sound sources (a radio or tape recorder for music and voice, and an *audio oscillator* for "pure tones"), you will first explore briefly the types of sound accompanying various electrical signals. Each signal in turn will be used to drive a loudspeaker.

Then a careful, but not difficult, experiment will reveal the physics underlying this most familiar and useful "machine." You will measure small motions of the speaker using a sensitive optical technique.

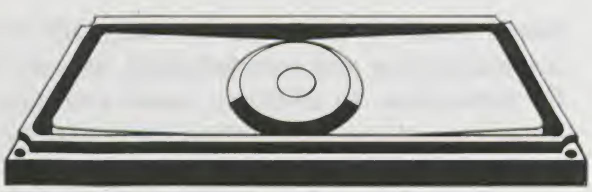


Figure 3. Planar speaker.

For convenience in your experiments, you will very likely use a "planar" speaker, as illustrated in the module. Its design is unusually sturdy and suited to experimentation. This speaker looks a little different from those you may be used to, because the whole thing is only about 1½ inches thick. This makes it very versatile for entertainment purposes, since it can be installed in many locations where a conventional speaker would not fit. However, the principle of operation of this speaker is the same as that for bulkier speakers.

Section B: Measuring Sound Radiation

You will learn about, and learn to measure, some of the performance factors which determine speaker quality. You will learn about factors like *efficiency* and *linearity*. When it comes to shopping for "high fidelity," such factors can make a difference of hundreds of dollars in the prices of seemingly similar units. Special attention will be paid to the *directionality* of a speaker's radiation, and how that is measured. Besides being of basic interest, this should help you to understand some practical considerations affecting the quality of your own home music system.

In this section you will also learn more about sound itself, including the meaning and importance of concepts such as *frequency* and *acoustic power.* You will find that the power needed to create most familiar sounds is exceedingly small. This should give you an increased respect for your own hearing apparatus (ears) as an amazingly sensitive and versatile detector of sound.

Section C: Using Sound Waves

A little experimentation shows that sound waves, like waves of light or water, are reflected from various surfaces. They are also absorbed by certain materials. The waves can be made to combine with themselves (to *interfere*) to form surprising and intricate patterns. A brief study of these effects will allow you to understand the principles of some important sonic devices. These include echo rangefinders, ultrasonic sensors (burglar alarms), and sonar.

The emphasis in this section is less on the loudspeaker and more on sound as such. You will measure the wavelength of sound and determine the speed of sound. The experiments in this section are more open-ended than in the preceding sections (and perhaps less precise). Therefore, you may be called upon to use your own imagination somewhat in figuring out what to do.

GOALS

The two general goals of this module are 1. to give you an understanding of the physical principles underlying the design of the loudspeaker, and 2. to give you an understanding of the physical behavior of sound itself.

Loudspeaker

Regarding the loudspeaker in this module, you should gain a knowledge of:

The conversion of electrical energy into sound energy.

The magnetic force law.

The directionality of sound radiation.

The technical specifications of a loudspeaker: power output, efficiency, linearity, and frequency response.

Sound

This module should provide you with a knowledge of:

The generation of sound.

The detection of sound.

The parameters of sound: amplitude, intensity, loudness, frequency, wavelength, and speed.

Reflection and interference.

Standing waves.

Some uses for sound and sound systems.

This module also introduces several important measurement techniques. Especially useful is the general method of "null measurements." Furthermore you will learn about the "optical lever" as a means for amplifying and detecting very small motions.

At the end of this module you should be able to demonstrate your understanding by doing the following things:

Section A:

1. Describe the basic properties of sound.
2. Describe methods for the generation, propagation, and detection of sound.
3. Explain the basic principles of the electrodynamic loudspeaker. Sketch the elements of such a speaker.
4. Measure the motion of the speaker, using an optical method.
5. Measure the force produced by the voice coil, using a null method.
6. Calculate the magnetic field from the force, the current, and the coil-wire length.

Section B:

7. Measure the radiation pattern of a loudspeaker, and produce an "isointensity plot."
8. Explain the basic specifications of a loudspeaker.

Section C:

9. Predict the behavior of reflected sound in some simple situations.
10. Explain the basic parameters of sound.
11. Describe some technical uses of sound.

PREREQUISITES

To complete this module, you will need a working knowledge of basic algebra. Some knowledge of electric current and its measurement using an ammeter would also be helpful.

Certain experiments in the module make use of the oscilloscope. If you are not acquainted with the scope, the necessary introduction is provided by an Appendix at the back of the module. (On the other hand, the scope is not essential for the main parts of the work.)

In Section A some discussion is included dealing with the idea of a magnetic field. This discussion is very brief but self-contained. To learn more about the subject, you may wish to make use of an additional Physics of Technology module, *The Solenoid,* which deals primarily with magnetism.

SECTION A

Generating and Detecting Sound

WHAT IS SOUND?

Vibrations of the Air

Most of us know that the air exerts a pressure on all the things in it. This pressure is called "atmospheric pressure," and it is often mentioned in the TV and radio weather reports.

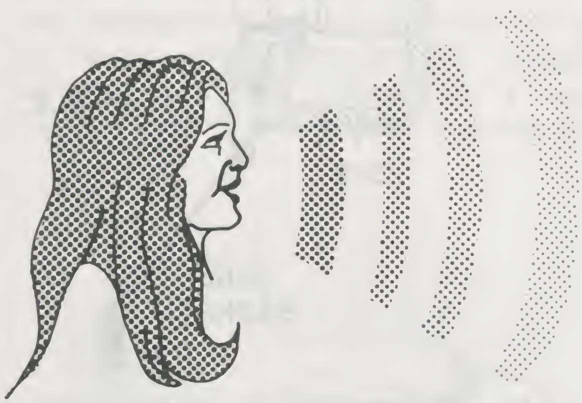


Figure 4.

In a sound wave, the air pressure changes by a very small amount. For audible sounds this happens hundreds or thousands of times each second. First the air pressure increases slightly above the atmospheric pressure. Then it decreases slightly below this value. Then the pressure increases and the cycle is repeated again and again. The air *density* and *temperature* also vary in the same rhythm. This pattern of changing pressure and density spreads rapidly outward from the sound source, just as ripples on a quiet pond spread from the place where a pebble was thrown in. And just as with ripples on a pond, the passing of a sound wave through the air results in small back-and-forth motions (vibrations) of the air *molecules.*

Vibrations of Other Materials

Sound can travel through other materials, such as solids or liquids. The effects are similar, involving slight back-and-forth vibrations of the molecules.

In particular, sound transmission through water is used by ships (sonar) to locate other ships and submarines. Powerful sound generators send high frequency sound waves outward from the ship. Sensitive microphones detect the later return of reflections (echoes) from other objects. The time delay and the direction of the echoes locate the object.

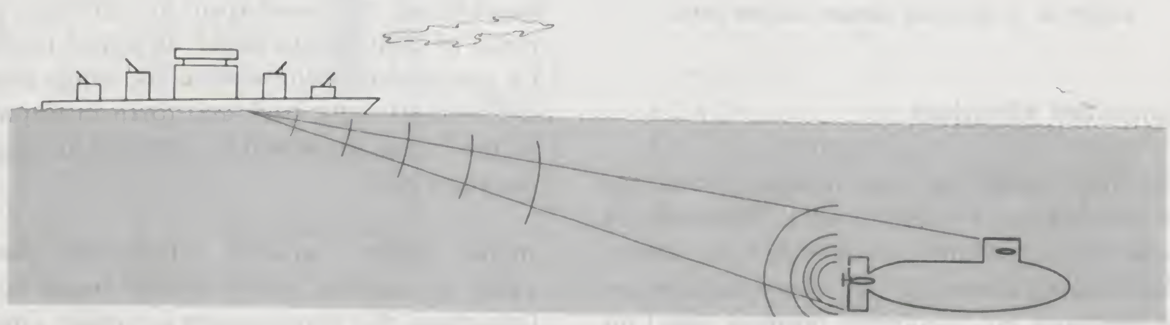


Figure 5.

HOW IS SOUND CREATED?

Sudden Changes of Pressure

Any event causing a very rapid *compression* or *expansion* of the air at one place is followed by a sound pulse traveling outward. Examples include the clapping of hands, a thunder clap, or the boom of a cannon. The sound of a hand clap arises from a sudden mechanical compression of air. The thunder clap and cannon boom begin with an expansion of air due to rapid heating.

Natural Vibrations

Most sounds are caused by the very rapid back-and-forth motion of a solid object, which in turn sets the surrounding air in motion. Watch a guitar string or a drum head in operation, and this vibration will be apparent. Another, less obvious, example is human speech, which is caused by the vocal cords in the throat as they vibrate.

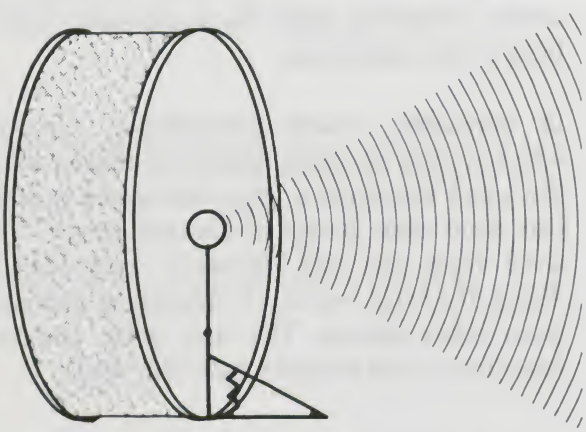


Figure 6. A vibrating surface creates sound.

Controlled Vibrations

Electrical signals are used to cause vibrations of the surface of a loudspeaker. This makes it possible to not only *produce* but also *reproduce* sounds almost at will. Everyone is aware of the huge entertainment business based on this fact. Less well known is the rapidly expanding industrial technology of sound and ultrasound.

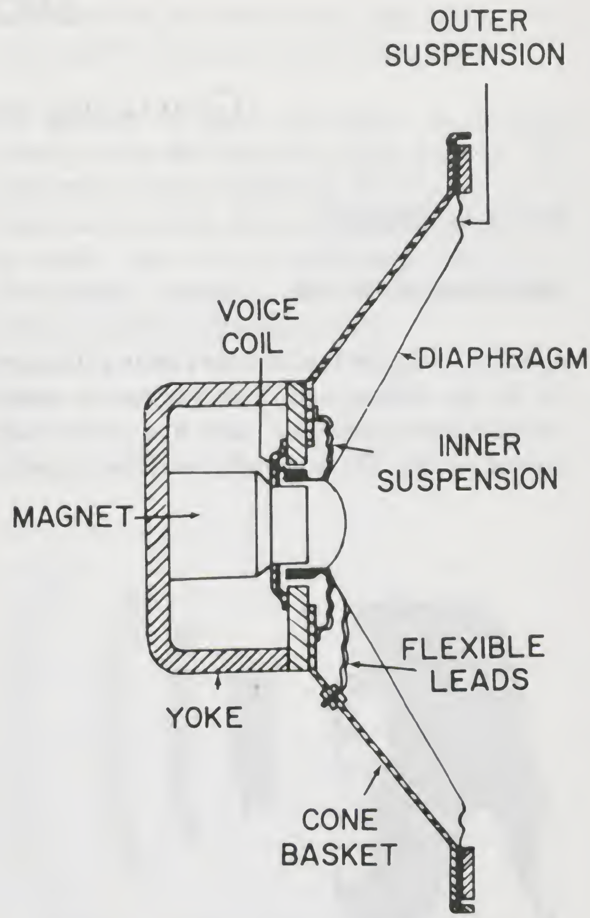


Figure 7. Construction of a cone loudspeaker.

Construction of a Typical Speaker

The figure above shows a cross-sectional view of a typical dynamic loudspeaker, the most common type. The vibrating surface (the *diaphragm*) is cone-shaped for stiffness and made of light-weight paper. It is held in place by corrugated cloth suspensions, which allows the cone to move back and forth in response to the forces generated by currents flowing in the voice coil.

In the "planar" speaker, a flat, rigid, plastic panel is used in place of the paper cone. Otherwise, the construction is similar. One of your first tasks will be to verify that the parts of your speaker are basically the same as those shown here.

HOW IS SOUND DETECTED?

A Typical Microphone

Most microphones are like loudspeakers run in reverse. In a loudspeaker, an electrical signal forces a diaphragm to vibrate, creating sound. In a "mike," the sound makes the diaphragm vibrate, and this creates an electrical "signal." This electrical output can be amplified, recorded, or broadcast as desired. Our own ears work in much the same way as does a microphone.

The figure below illustrates how a simple *ceramic* microphone works. Small motions of the diaphragm are transferred by the pin to the special ceramic element. The ceramic material develops a voltage, like a tiny battery, when it is bent. Such materials are called *piezoelectric* (pronounced "pea-ā′-zo-electric" or "pye-ee′-zo-electric").

The Human Ear

In the ear, the diaphragm is the eardrum, and the pin is a series of three bones in the middle ear called the *hammer, anvil,* and *stirrup.* Their function is to transfer vibrations of the eardrum to the *cochlea* (inner ear). The output of the inner ear goes to the brain as an electrical signal via the auditory nerve.

Notice how similar the working of a human ear is to the working of a microphone. The ear also converts sound into electrical impulses, which then travel to the brain for processing or storage.

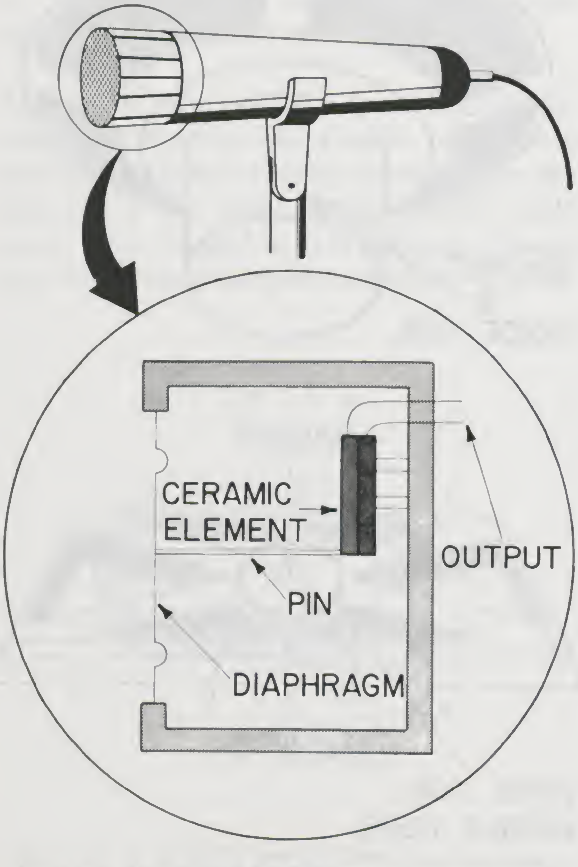


Figure 8. Construction of a ceramic microphone.

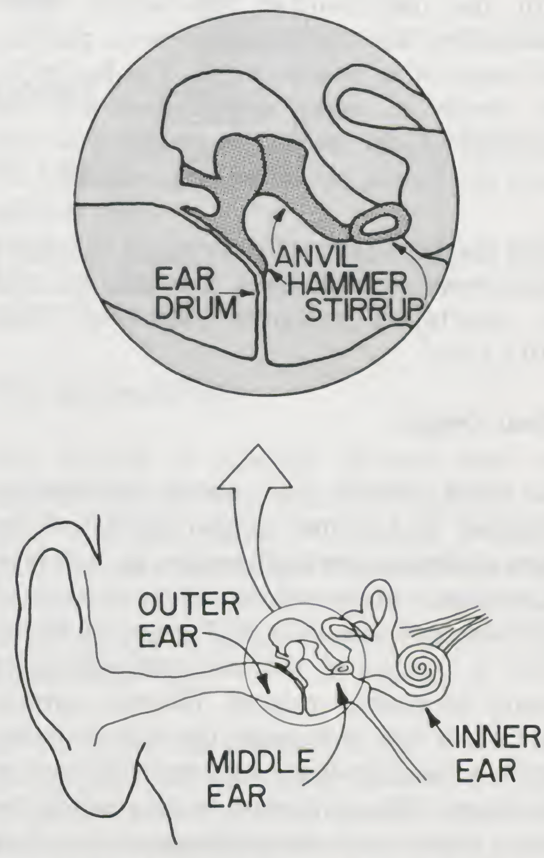


Figure 9. Construction of the human ear.

EXAMINING YOUR LOUDSPEAKER

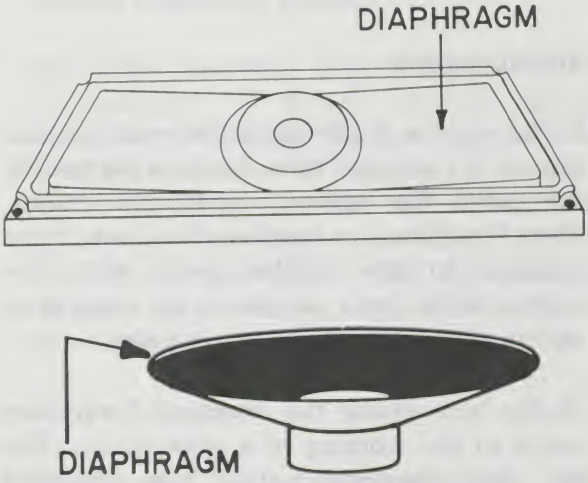


Figure 10.

An Example of Modern Technology

Recent years have seen impressive developments in materials technology. Revolutionary manufacturing techniques have been coupled with the invention of new alloys, semiconductors, liquid crystals, ceramics, plastics, and many other new materials. This has led to the design of many new devices and the re-design of old ones. The speaker you will study is a simple, yet interesting, example.

Your speaker's plastic diaphragm is *expanded polystyrene.* This material is largely air, and its density is extremely low–less than $0.012\ g/cm^3$.

Planar Design

As noted before, the planar loudspeaker functions in a manner similar to that of the more common cone loudspeaker. In each type of speaker, a voice coil consisting of a coil of wire wrapped around a stiff paper or plastic form is positioned between the poles of a strong permanent magnet. Electric currents applied to the coil cause the coil to move, which in turn produces motion of the cone or diaphragm. The diaphragm motion causes the air to vibrate and this produces sound. You will see how this works in your experiments.

Speaker Components

1. *Examine your speaker carefully.* Identify the various parts shown in Figures 10 and 11. By comparing your speaker with the figure of an ordinary cone loudspeaker, shown previously, you can see that, while the construction is basically the same, the planar speaker is simpler and more rugged.

2. *Look for the voice coil in the magnet gap behind the diaphragm.* This is barely visible. However, if a dismantled version of the speaker is available, you can see how the coil fits into the gap.

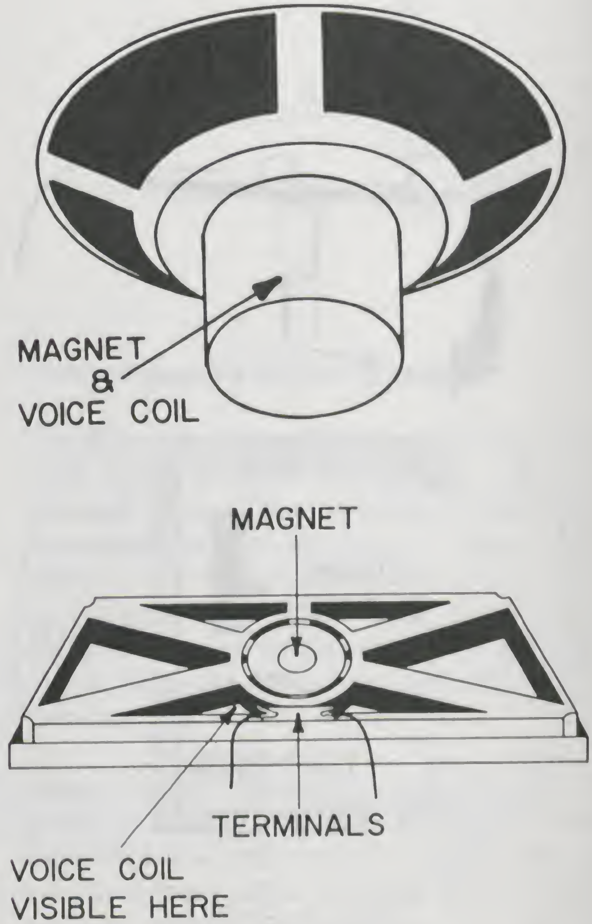


Figure 11.

What Is a Field?

Nearly everyone has noticed that a magnet mysteriously affects the space around it so that, if pieces of magnetizable material (like iron or steel) are placed nearby, they experience a pull. We describe this by saying that the magnet surrounds itself with a "field of magnetic influence," or magnetic field for short. Of course, just giving the effect a name does not really explain anything. But a name makes it easier to discuss.

You Can Detect a Field with a Compass

While it cannot directly tell you the *strength* of a magnetic field, a compass can tell you that a field is present and what its *direction* is. The magnetic needle in a compass always aligns itself with the field as much as it is free to. Of course, the needle is only free to rotate in one plane, the orientation of which depends on how the compass is held.

The sketch below shows a familiar horseshoe magnet, indicating the pattern of so-called *field lines.* These are simply lines showing the direction a very small compass needle would point, when located at each position in space, if it could rotate completely freely.

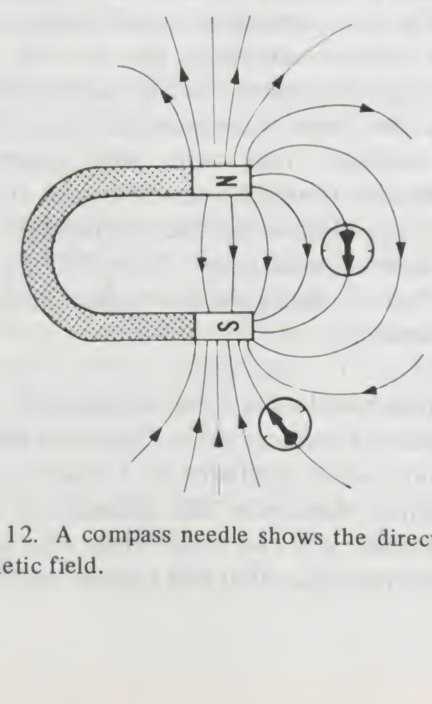


Figure 12. A compass needle shows the direction of a magnetic field.

A magnetic compass is sensitive enough to tell you the direction of the earth's very weak natural field. This is, of course, its intended use. It can also be used to reveal the directions of your speaker's field at different points near the speaker.

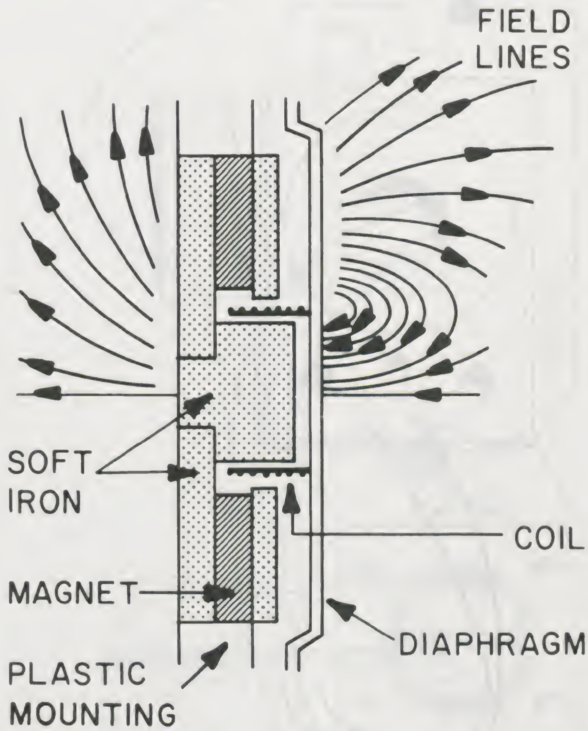


Figure 13. Cross-section of speaker's magnet, showing some field lines. (Lines for the lower half were left off for clarity.)

The Speaker's Field

Try moving a compass around near the speaker magnet. Note that the field there is so great that the needle is strongly deflected from its usual northerly orientation. You can manually rotate your compass case around until the needle is able to point more or less along a field line.

Slowly move the compass farther away from the magnet. The field gets weaker. At what point do you think the speaker's field has faded to a strength equal to that of the earth's field? How did you find this point?

EXPERIMENT A-1. Trying Out Your Speaker and Mike

The purpose of this experiment is to familiarize you with various audio devices and how they are used in simple circuits. The work should not take more than about one-half hour.

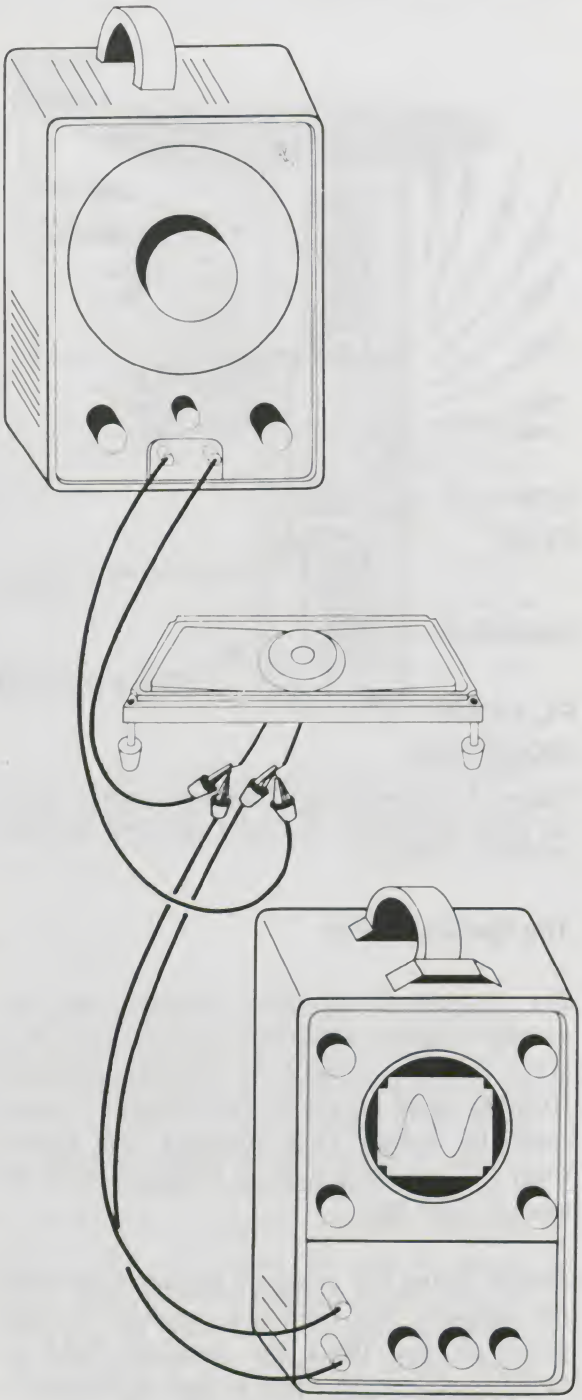


Figure 14.

Creating Sound from Signals

1. *Connect the audio oscillator to your speaker* as shown at the left. The oscillator puts out "pure" tones. Later in the module, you will learn more about the difference between pure tones and others.

2. *Display the electrical signal from the oscillator on an oscilloscope while you listen to the speaker.* Use clip leads to connect the speaker to the scope as shown. One end of each clip lead should be connected to a terminal of the speaker, and the other end to a terminal of the *vertical deflection* input to the scope. The *sweep-time* setting should probably be in the range 1–10 ms/cm, but whatever seems to give an adequate picture is all right. The scope synchronization control should be set for *internal sync* ("INT").

3. *Note how different the loudness of the sound is in different directions.* Where is it greatest? Least? Do you think this performance would be good enough for a hi-fi system?

4. *Try different settings of the amplitude and frequency controls on the oscillator.* On the scope, you should be able to get a pattern something like the one on the scope in Figure 14. This pattern is called a *sine wave* or a *sinusoidal* pattern. Your oscillator may also have square-wave and/or triangular-wave outputs. If so, try them. Listen to the difference. Which wave sounds most "pure"? What does "pure" mean to you when applied to sound?

The *frequency* of a tone corresponds to the sensation of musical *pitch.* Using the markings on your audio oscillator as a rough guide to frequency, determine the highest and lowest frequencies you can hear. (Note that there is both a frequency dial and a range switch.) Do you notice anything interesting at high frequencies as you move your head around? We will study this in Section C.

5. *Sketch some of the patterns you observed on the scope.* Try to indicate by means of your sketches the answers to some of these questions: What features of a sine wave change when you change the amplitude setting of the oscillator? When you change the frequency setting? What is the relation between the height of the wave pattern on the oscilloscope and the loudness of the tones produced by the speaker?

Creating Signals from Sound

6. *Hook up your mike to the amplifier and connect the amplifier output to the scope, as shown at the right.* The amplifier strengthens the microphone signal to give a bigger signal on the scope. (If the amplifier has an offset control, this should be set to ZERO.)

7. *Talk into the mike. Try singing or whistling a sustained musical note.* Observe the appearance of voice and whistling signals on the scope. Do you see any characteristic difference? Can you match any of the patterns from the audio oscillator ("pure" tones)?

Using the Speaker as a Mike

Microphones are like loudspeakers run in reverse. In fact, you can actually use your speaker as a mike.

8. *Connect your speaker to the input of the amplifier.* Talk into the speaker and observe the resulting signal on the scope. When a sound wave hits the speaker, it moves the diaphragm. The motion creates an electrical signal, just as with a mike.

   You can also replace the scope with a second speaker, and hear the sound which comes out. The first speaker is acting like a microphone. (It may be necessary here to reduce the amplifier gain, to avoid *ringing*, a piercing sound produced by *feedback* from the output into the input.)

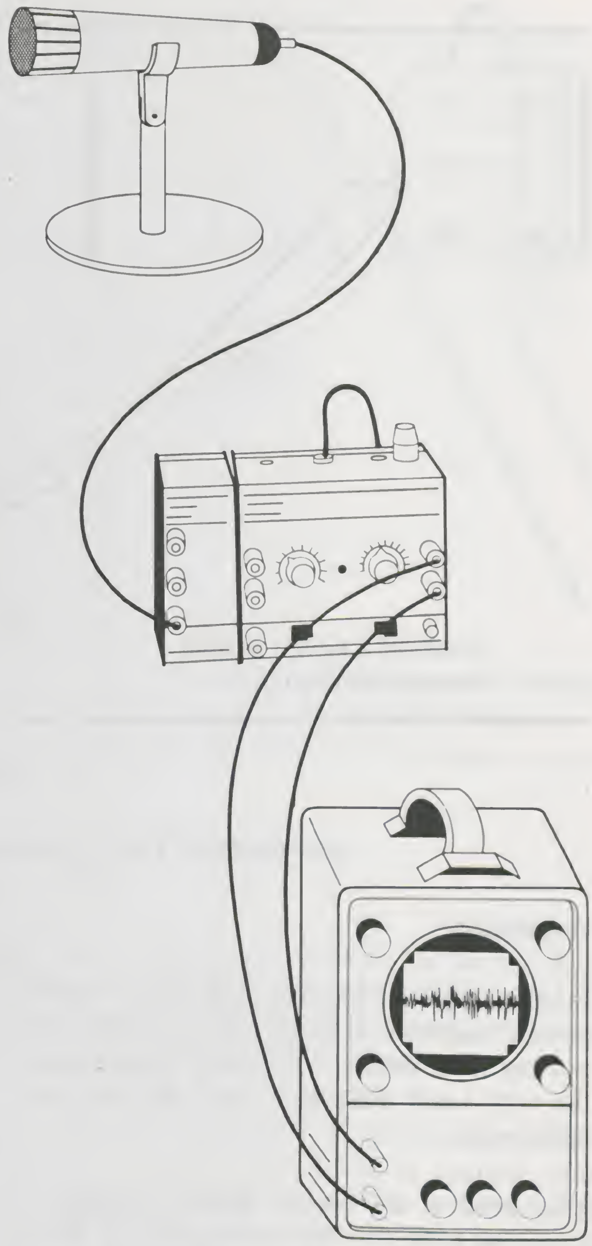
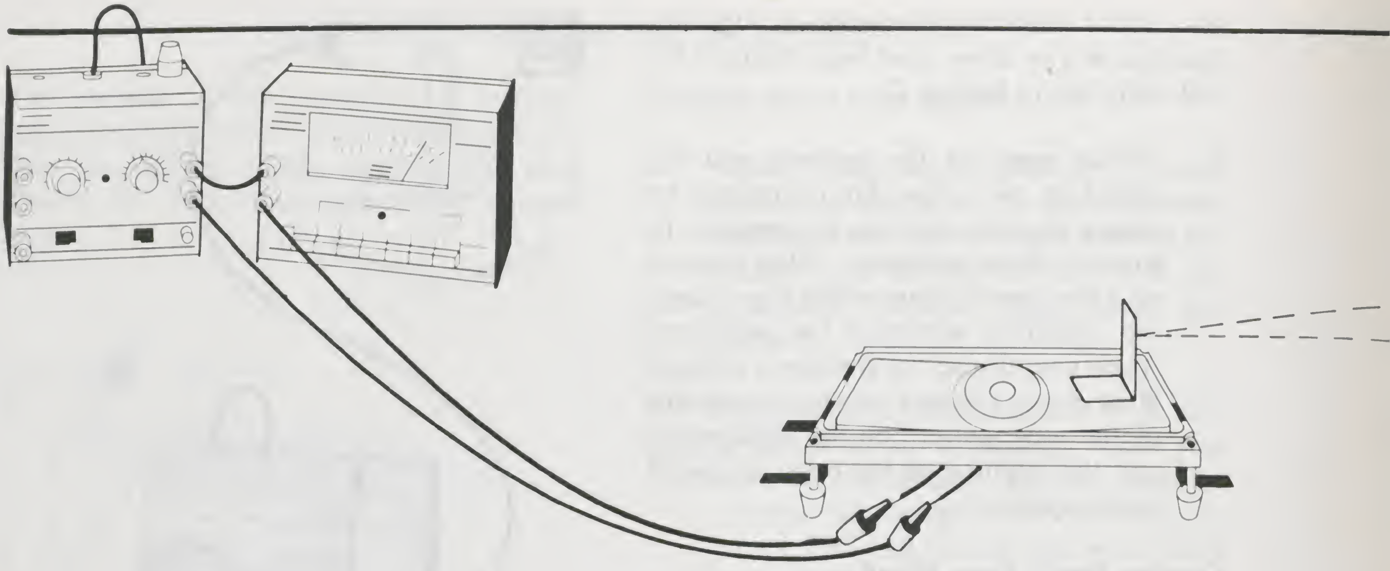


Figure 15.

Incidentally, a microphone of this type is called a "dynamic" mike because, like the "dynamic" speaker, it has a coil which moves in a magnetic field. The operating principle is therefore somewhat different than in a ceramic mike.



Connect a power supply through an ammeter . . . to the speaker with the mirror attached.

Figure 16.

EXPERIMENT A-2. Measuring Motion of the Diaphragm

Preparation

In this experiment you will put a steady electric current through the speaker coil, causing it to deflect. You will then measure the very small deflection with the help of a light beam.

You need a DC-power supply capable of supplying from 0 to 10 V and up to 0.5 A. An ammeter or multimeter with a 0–1 A range will measure the DC current.

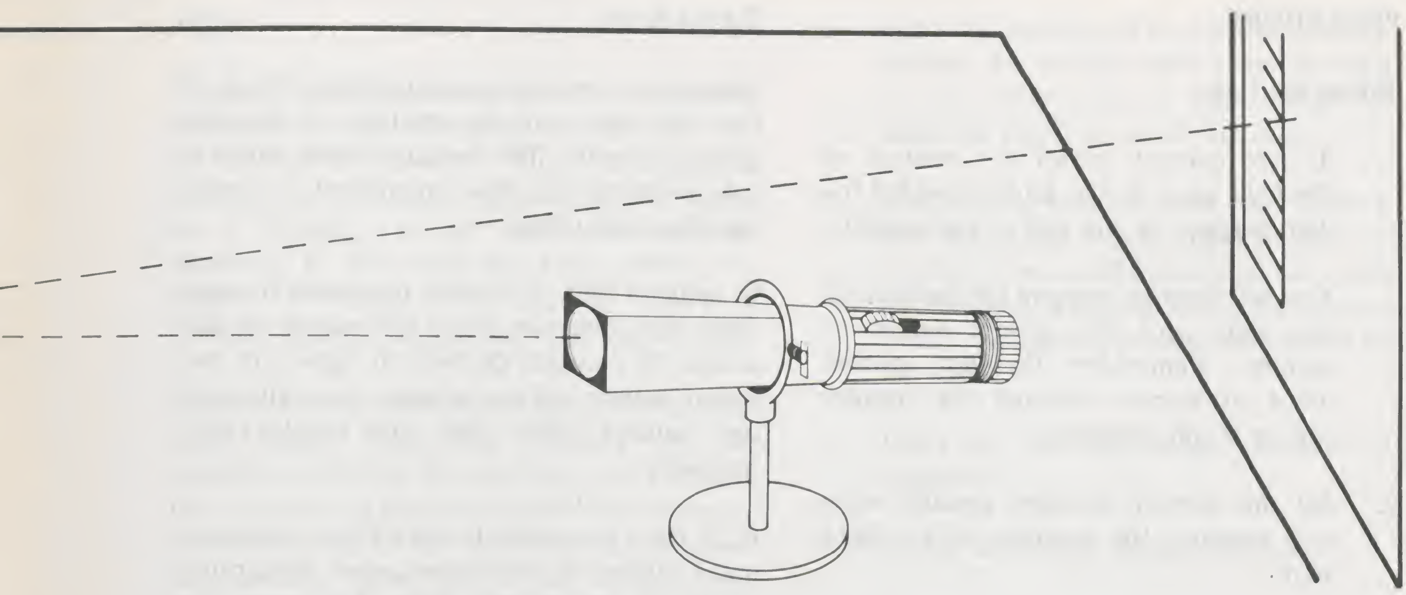
You also need a light source and a mirror to reflect the light to a screen. The screen may be just a piece of paper taped to the wall. The image on the screen should not be too blurred, because you will be trying to measure its small motions. To get a focused image, you should use a curved mirror ("spherical" mirror), which acts like a lens. Then no other lenses are needed, and almost any light source will do. For instance, you can use a flashlight or even just a bare bulb operated from a battery.

Setup

1. *Hook up the circuit shown in Figure 16, using clip leads. Adjust the supply voltage* until a current between 0 and about 0.5 A registers on the ammeter.

CAUTION: *Do not exceed 0.5 A. Too much current can damage the coil.*

2. *Set the speaker flat on your table with the front pointing up. Place the small mirror* on the L-shaped bracket in the speaker diaphragm in the position shown. Carefully tape the speaker to the bench as shown, so it doesn't move accidentally.



CAUTION: *The speaker diaphragm is soft and easily damaged. DO NOT USE FORCE. Handle the speaker lightly. Tape it down only by the edges.*

3. *Adjust the current up and down.* As the diaphragm moves up and down, the mirror deflects through a small angle. Try observing this small motion of the mirror so you understand what is happening.

4. *Mount the light source.* This can be done on a ring stand with a clamp. Set it up near the speaker and on the same table. Aim it at the mirror.

5. *Use a sheet of white paper as a screen, and locate the reflected light beam* about a meter from the speaker. When you have located it, hang the sheet of paper on a stand or tape it to a wall or other convenient support. Focus the image of the light source by adjusting the source or changing distances to get the clearest possible spot on the screen.

   You will need to measure the deflection of the light beam as current changes. Adjust the current up and down to get a feel for how far the light beam moves. *Reverse* the two wires to the speaker and try again. The motion should now be in the opposite direction.

6. *Now tape a ruler to the screen* in such a position that you can measure the motion of the light spot throughout the full range you have just observed. The motion of the light spot follows the motion of the diaphragm, but it is much larger and easier to measure without disturbing the diaphragm. That is the reason for this setup.

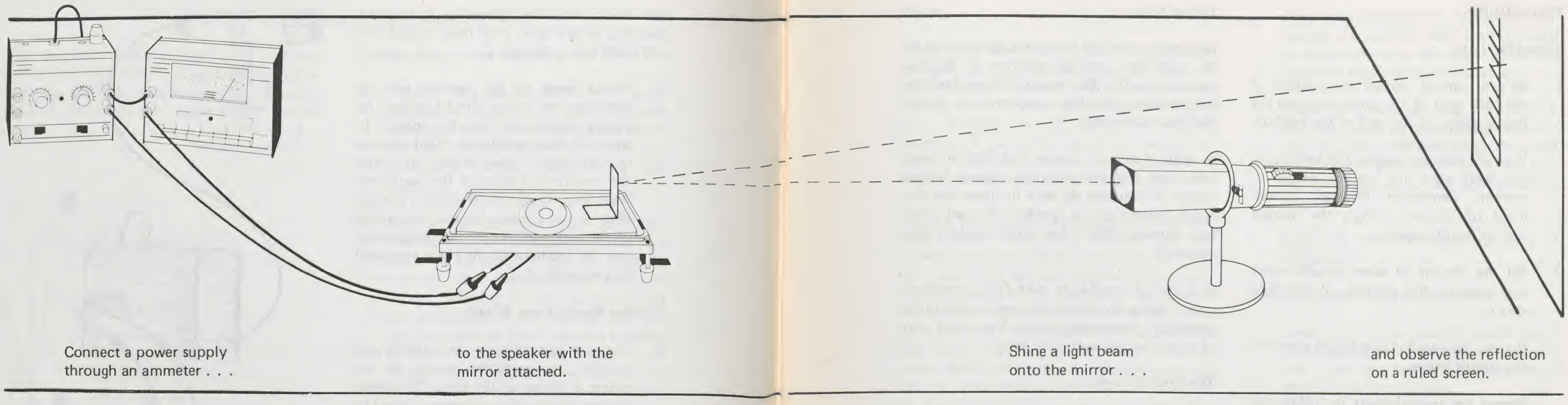


Figure 16.

EXPERIMENT A-2. Measuring Motion of the Diaphragm

Preparation

In this experiment you will put a steady electric current through the speaker coil, causing it to deflect. You will then measure the very small deflection with the help of a light beam.

You need a DC-power supply capable of supplying from 0 to 10 V and up to 0.5 A. An ammeter or multimeter with a 0–1 A range will measure the DC current.

You also need a light source and a mirror to reflect the light to a screen. The screen may be just a piece of paper taped to the wall. The image on the screen should not be too blurred, because you will be trying to measure its small motions. To get a focused image, you should use a curved mirror ("spherical" mirror), which acts like a lens. Then no other lenses are needed, and almost any light source will do. For instance, you can use a flashlight or even just a bare bulb operated from a battery.

Setup

1. *Hook up the circuit shown in Figure 16, using clip leads. Adjust the supply voltage* until a current between 0 and about 0.5 A registers on the ammeter.

CAUTION: *Do not exceed 0.5 A. Too much current can damage the coil.*

2. *Set the speaker flat on your table with the front pointing up. Place the small mirror* on the L-shaped bracket in the speaker diaphragm in the position shown. Carefully tape the speaker to the bench as shown, so it doesn't move accidentally.

CAUTION: *The speaker diaphragm is soft and easily damaged. DO NOT USE FORCE. Handle the speaker lightly. Tape it down only by the edges.*

3. *Adjust the current up and down.* As the diaphragm moves up and down, the mirror deflects through a small angle. Try observing this small motion of the mirror so you understand what is happening.

4. *Mount the light source.* This can be done on a ring stand with a clamp. Set it up near the speaker and on the same table. Aim it at the mirror.

5. *Use a sheet of white paper as a screen, and locate the reflected light beam* about a meter from the speaker. When you have located it, hang the sheet of paper on a stand or tape it to a wall or other convenient support. Focus the image of the light source by adjusting the source or changing distances to get the clearest possible spot on the screen.

You will need to measure the deflection of the light beam as current changes. Adjust the current up and down to get a feel for how far the light beam moves. *Reverse* the two wires to the speaker and try again. The motion should now be in the opposite direction.

6. *Now tape a ruler to the screen* in such a position that you can measure the motion of the light spot throughout the full range you have just observed. The motion of the light spot follows the motion of the diaphragm, but it is much larger and easier to measure without disturbing the diaphragm. That is the reason for this setup.

PROCEDURE

Taking the Data

1. *At zero current, record the position of the light spot* in the table provided for that purpose at the end of the module.

   You will want to measure the position of the light spot for several values of current. *Remember: Do not exceed 0.5 A of current through the speaker coil, or it will overheat.*

2. *Set the current* to some specific value and measure the position of the light spot.

   Record the data in the table, as shown in the sample below.

   Repeat the measurement for other current values up to 0.5 A.

3. *Reverse the clip leads* to the speaker terminals and repeat the series of measurements. Record these currents as *negative* currents, with a minus sign in front of them in the table.

Taking Notes

Remember, you are measuring the position of the light spot, not the position of the diaphragm directly. The two are related, but it is not necessary for this experiment to figure out the relationship.

In spite of this, it is wise to record in your notes the distance from the mirror to the screen. If you ever do wish to figure out the actual motion of the speaker, you will need that number. (Why? Are other numbers also needed?)

It is always sensible to take fairly complete notes during an experiment, even concerning seemingly unnecessary facts. You never can be sure what you may need later!

The Optical Lever

We have remarked that a light spot is used here because its motion is easier to measure than the very small motion of the diaphragm. Why is the motion of the light spot larger? A clue to this is contained in the name of the method. Such a way of *amplifying* small motions is called an "optical lever" method.

TRIAL NUMBER	CURRENT (AMPERES)	POSITION (ARBITRARY UNITS)
1	0	17.2
2	0.05	18.3
11	−0.05	[illegible]
12	− 0.10	15.0
13	− 0.15	13.9
		12.8

Figure 17. Sample table.

Units

When you make a measurement, it is usually essential to pay attention to the *units* of measurement involved. For instance, a number such as 267, representing the height of a building, is not much use without knowing if the units are feet, meters, or inches. Obviously, knowing the units is usually as important as knowing the number.

In your present experiment however, you can learn many interesting features of the speaker's behavior by knowing just the *relative* amounts of motion for different currents. Thus, positions of the light spot may be used to *represent* deflections of the speaker, but with the *units* of deflection not known. For this reason, any convenient units (arbitrary units) will do. For instance, the spot positions may be read directly from the scale attached to the paper screen, as we have suggested.

Plotting Your Data

You can plot your data on a piece of ordinary graph paper. It will be necessary to first put numbers on the position axis. The range should be just enough to include all of your position readings for the light spot.

1. *Begin the labeling of the position axis by putting the position value corresponding to zero current at the origin.* This will make the graph centered vertically.

2. *Decide on a suitable scale for the position axis.* Make a choice of divisions–for example, one small division on the graph for 1 mm on your ruler–so that the plot will fit on the paper, and yet will not be too small.

3. *Label the axis with major divisions of positions.*

4. *Do the same with the current axis.*

5. *Locate the correct point for each set of current-position values, and mark it with a dot.* The dots can be enclosed in small circles for easier visibility.

6. *Draw the best straight line you can through the data points.* Use your judgment to fit the points as well as possible. The best straight line should have roughly as many points above it as below. Check any points which seem significantly out of line to see whether they are misplotted.

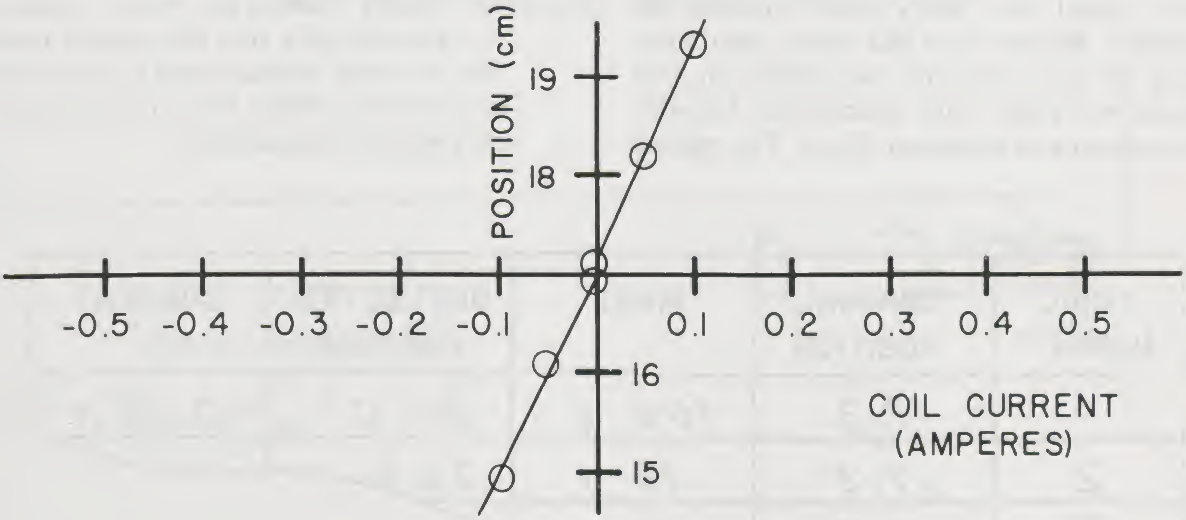


Figure 18. Sample graph.

EXPERIMENT A-3. Measuring the Force of the Voice Coil

In this experiment, the amount of force exerted on the diaphragm by the voice coil will be determined for various values of current. This will be done by using the unknown voice coil force to *cancel* a known force (a weight placed on the speaker). Thus there will be *no net displacement* of the diaphragm at the time of the measurement.

Procedure

Again a light spot will be used to observe the diaphragm motion. The aim will be to cancel (or "null out") any observed deflection of the spot by a proper adjustment of the current. Measurement of this sort is often referred to as a *null measurement.* In a null measurement, some unknown variable is measured by cancelling it with a known variable. What is the unknown variable in this measurement? What is the known variable?

Handling the Weights

Place each weight directly in the center of the diaphragm, so that it is pushed down without any tilting. It is better to use one weight at a time, rather than combinations. *Be sure that the speaker is not shifted in this process.* It should be taped down.

Why should you worry about shifting the speaker? Because if it did move, then later, when the spot deflection was nulled out, you would not really have cancellation between the known and unknown forces. The reading for "zero deflection" would have changed. This kind of problem in a null measurement is called *zero drift.*

Taking Data

1. *Record the position of the light spot* with zero current in the voice coil before placing any weight on the diaphragm. Call this the *original position.* Use the same position scale as you used in the previous experiment. Use the data table provided at the back of the module for recording your data.

2. *Add a mass to the speaker, and record its value.* Be sure to note the units. Record the position of the light spot, calling it the "deflected position."

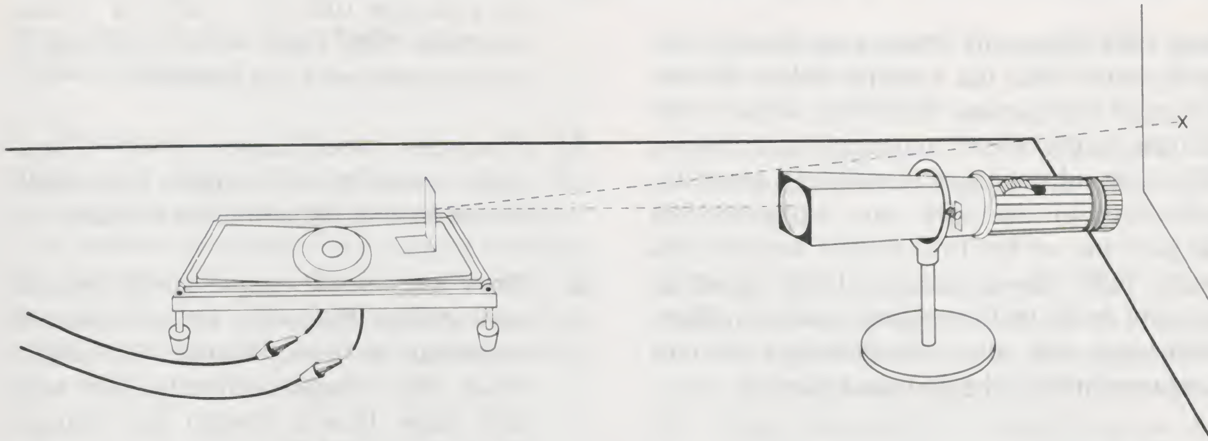
3. *Turn on the current, and adjust its value* until the light spot returns to the *original* position. *Record* the current needed. This concludes the data-taking for that particular weight.

4. *Turn off the current, remove the mass, and repeat the whole process* with another mass. Repeat with five or six different masses from 0 to 100 g. As a check on whether you have avoided accidentally moving the whole speaker, you should verify that the original position for each measurement is about the same as the previous value. *If not, repeat the previous measurement.*

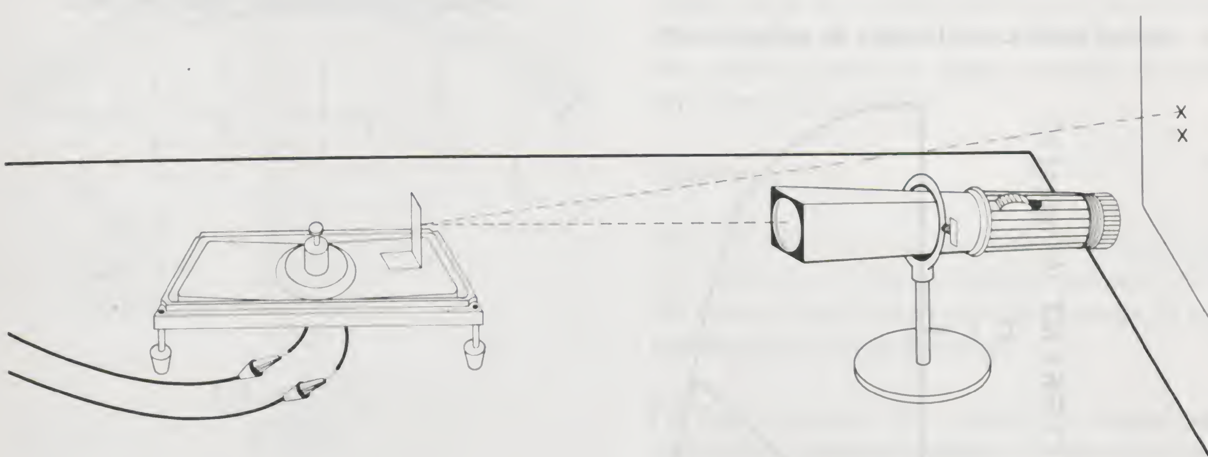
TRIAL NUMBER	ORIGINAL POSITION	MASS	DEFLECTED POSITION	CURRENT
1	17.2	100 g	24.0	0.48 A
2	17.3	75 g	22.4	
3	17.3	50 g		

Figure 19. Sample data table.

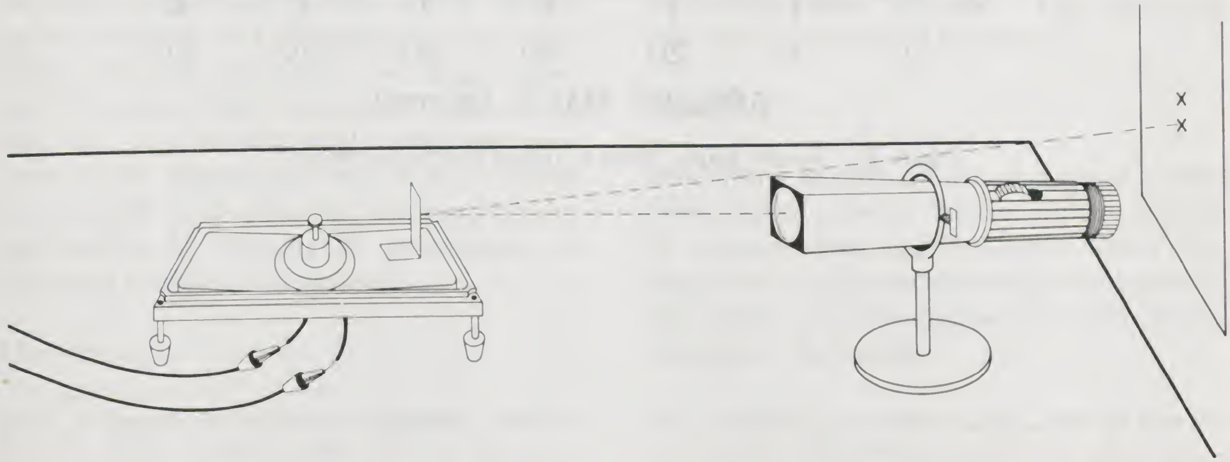
1. With zero current, record position of spot.



2. Add 100-gram (100-g) mass; record new position of spot.



3. Adjust current to <u>null out</u> deflection. Record current.



4. Repeat with other masses, down to 10 g or less.

Figure 20.

Graphing the Data

Your plot of current versus mass should look pretty much like the example below. Notice that now there are no "arbitrary units," even though, in the actual experiments, you have still been observing an arbitrary quantity, deflections of the light spot. However, this has gone out of the final results, because you made "null" measurements, always adjusting the spot back to the original position. Other advantages and some disadvantages of null measurements will be discussed shortly.

1. *Draw the horizontal and vertical axes* on your graph, leaving some room at the edges for scales.

2. *Label both axes.* Be sure to include units on both axes this time. Remember that, as a general rule, the results of a measurement only make sense if the unit is known along with the number.

3. *Plot each combination of current and mass* as a point on the graph. Draw small circles around the points for visibility.

4. *Draw a smooth, simple curve through your points.* Naturally, any such curve is supposed to pass through the origin, since zero current certainly goes with zero mass. Does a straight line through the origin "fit"? The answer may require some judgment, because there is bound to be a little scatter in the experiment values, due to errors of measurement.

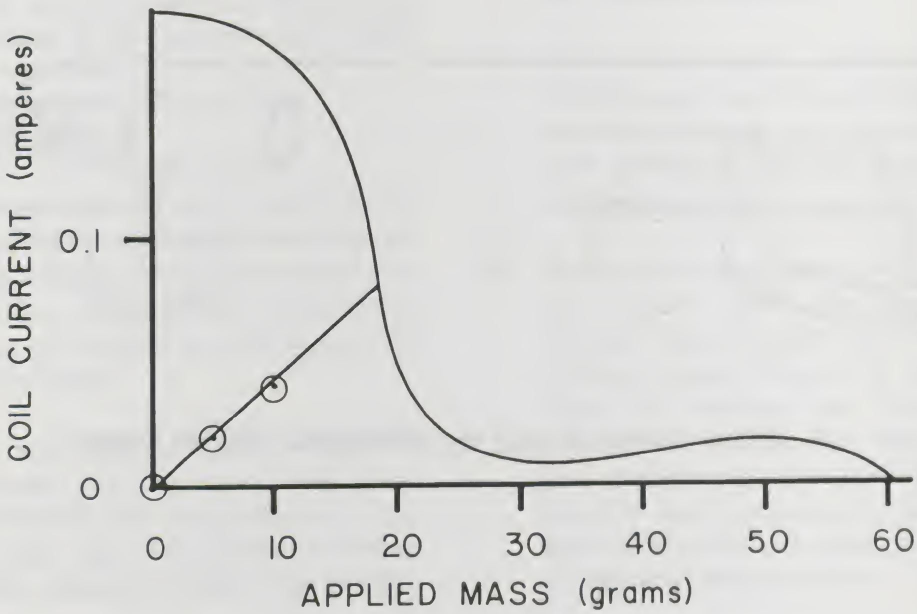


Figure 21. Sample graph. Does a straight line fit the points?

What Can You Say about Deflections of the Diaphragm?

In Experiment A-2, when the current in the voice coil changed to a new value, the diaphragm quickly moved to a new position. This position resulted from a balance between the *magnetic force* on the diaphragm, due to the current, and the *elastic forces* caused by stretching and bending of the diaphragm away from its "natural" position.

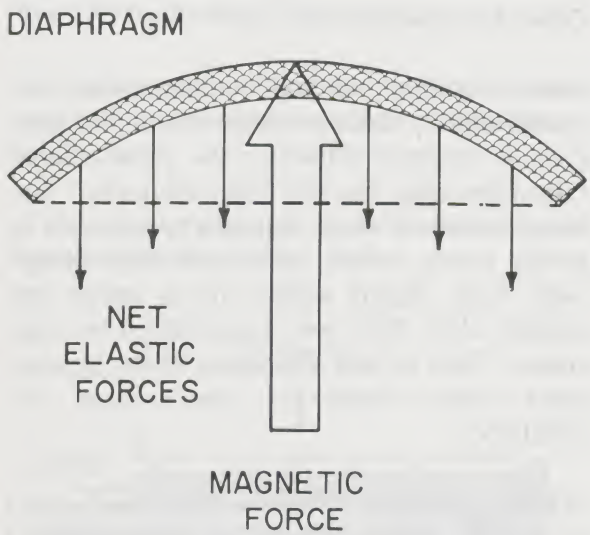


Figure 22. A balance of forces.

Both the magnetic and elastic forces depend on the position in a complicated way, especially when the deflections are large. However, in spite of this fact, and in spite of the fact that you never actually measured positions of the diaphragm itself—only positions of the light spot—you can get some useful information by looking at your graph of light-spot position versus current.

Linearity

Does a graph of deflection against current prove to be a straight line? Yes, approximately. The deflection is proportional to the current. This means that increasing the current by a given amount always gives the same increase in deflection, no matter how great the current is—within reasonable limits, of course. Look again at your own graph. Can you see any evidence that your graph is not a straight line?

Linearity in a loudspeaker (the fact that a straight line is produced) affects its *fidelity*, the accuracy with which it can reproduce a sound from an electrical signal. If the signal gets bigger, the deflection should increase in a proportional way. More importantly, if the signal really represents two or more distinct signals (from different sounds) added together, then the overall deflection should just be the *sum* of deflections corresponding to the various parts. A linear response insures this. Can you see why?

Polarity

Does the direction of the deflection reverse if the direction of the current is reversed? Yes. Do you see any differences in behavior in the two directions of deflection?

Paying attention to polarity is important when more than one speaker is used in a hi-fi system. The speakers must be connected so that they move in the same direction at the same time. This is called *phasing* the speakers. They must move "in-phase." The reason for this will be explored in Section C.

Sensitivity

How much motion of the diaphragm results from a given current? This is a little difficult to compute (see the Problems). The calculation involves the location and orientation of the mirror on the speaker, as well as the distance to the screen.

The *sensitivity* of the loudspeaker is one of the factors which determines how much of an electrical signal it takes to produce a sound of a certain loudness.

NULL MEASUREMENTS

Measuring Forces on the Diaphragm

When you measured the *deflection* of the speaker diaphragm in Experiment A-2, both of the opposing *forces* were of unknown strength: the magnetic force and the elastic force.

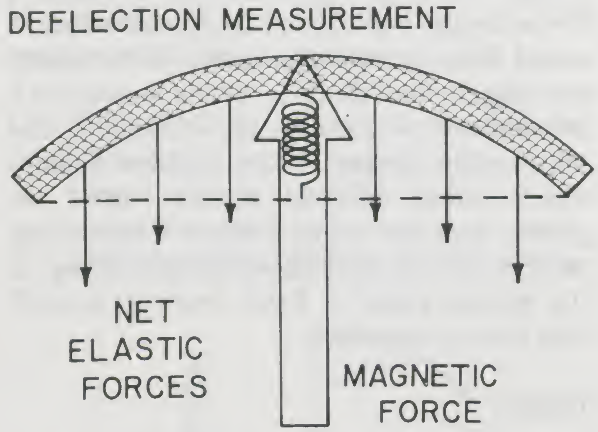


Figure 23. Both the net elastic force and the magnetic force are unknown.

To be able to say something about one of the unknown forces, for instance, the magnetic force, you must balance it against a known force. Naturally, you must make sure that the other unknown force is *not* acting then. This

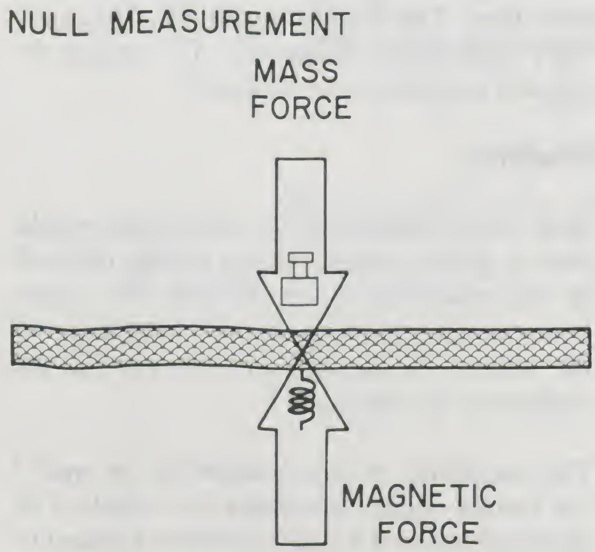


Figure 24. The force exerted by the mass is known, and from this the magnetic force can be found.

is the idea behind the *null measurement* in Experiment A-3.

In that measurement, the elastic force played no role because the diaphragm was always brought back to its original position after the weight was applied. Thus, no bending or stretching was taking place. The elastic force was *nulled out* of the picture.

The *unknown* magnetic force from the current then was balanced by the *known* force of the weight and was equal to the force of that weight.

Other Examples

Many examples of null measurements are quite familiar. *Balance* scales, such as the large ones in doctor's offices or the small ones in chemistry labs, use the "null principle." Stability is one of their features, as opposed to *spring* scales, whose calibration may change with time. Spring scales are so much less reliable that their use is prohibited in commerce. That is why the scales at the grocery store often display a label saying NO SPRINGS.

Another example involves the measurement of voltage. You may have used a *voltmeter* at one time or another. This works by causing the unknown voltage to drive a current through a resistor. The meter really measures this current. It is a deflection measurement. By contrast, a null measurement of voltage can be made using a *potentiometer.* This cancels the current from an unknown source using another current from a known source. In this way any current drain from the unknown source is avoided.

Advantages of Null Measurements

In general, null measurements avoid uncertainties associated with deflections: unknown variables, limitations of range, and unwanted changes of the measured system (such as having the voice coil moving out of the magnet gap, or drawing additional current from the voltage source).

Converting Electrical Signals to Sound

As you have seen, loudspeakers are devices which change electrical energy into sound energy. Devices which convert energy from one form to another are called *transducers.* Various principles of physics are involved in the energy-conversion process in different kinds of loudspeakers. The most important is the following.

The Electrodynamic Principle

A fundamental interaction exists between electric currents and magnetic fields. Stated briefly, it is:

A current-carrying wire experiences a force in a magnetic field.

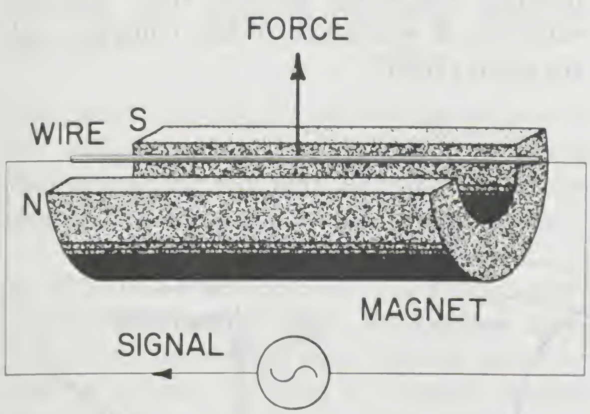


Figure 25.

In *electrodynamic* loudspeakers (called "dynamic" speakers for short), the wire forms a coil–the voice coil–which is attached to the diaphragm and moves it. The current in the voice coil comes about because of the electrical signal.

The force you determined in Experiment A-3 was equal to the electrodynamic force exerted by the speaker's magnet on the voice coil.

Converting Sound to Electrical Signals

Earlier in this module you learned that microphones are like loudspeakers in reverse. In fact, in most simple intercom systems, the speakers serve both functions. The person on each end has one box which contains a dynamic speaker and some simple electronics. By listening to his speaker, a person can hear sound; by talking into it, he can create signals.

You used a speaker as a microphone in Experiment A-1. You hooked it up directly to an amplifier and talked into the speaker. The output signal was visible on the oscilloscope.

The Principle of Induction

The output signal arises because a diaphragm, moved by sound, moves a coil in the magnetic field. The movement of the coil through the magnetic field causes (induces) a voltage in the coil. This phenomenon is called the Principle of Induction. It is the reverse of the Electrodynamic Principle.

Other Energy-Converting Devices: Motors and Generators

The *electrodynamic principle* is the cause of forces in an electric motor. The forces needed are large, so many more turns of wire are wound on the coil. Also, rotary motion is usually more desirable than back-and-forth motion; therefore motors are designed so that electrical energy is converted to rotational energy.

Electric generators operate on the *principle of induction.* In a generator, the motion of many turns of wire through a magnetic field causes current to flow. This current can then be used for any desired purpose.

The physical principles which apply to loudspeakers and microphones also apply to motors and generators.

CALCULATING THE MAGNETIC FIELD

The Field in the Gap

The "gap" is the space between the poles of the magnet. The voice coil is centered in the gap. As the figure shows, the magnetic field lines in the gap are radial. That is, they point directly in toward the center of the magnet along a radius in each case.

The Magnetic Force Law

You can calculate the *field strength* in the magnet gap using the measured force on the voice coil.

The sketch in Figure 26 shows part of the coil, emphasizing its position in the field. The current is at right angles to the field lines everywhere in the gap.

Whenever the current and the magnetic field are at right angles, a simple equation gives the force F on the wire in terms of the value of current i and the field strength B.

$$F = B \times i \times L$$

Here L is the total length of wire. The units are indicated in the sketch. The field-strength unit is one tesla (1 T), which is a strong field. The earth's magnetic field (which causes a compass needle to point north) is less than 10^{-4} T. The magnetic field is assumed to have the same strength at every point of the coil.

Calculate the Field

Follow the steps in the worksheet at the back of the module to calculate B for your speaker's magnet. The magnetic field is assumed to have the same strength at every point of the coil. For the length ℓ, use 3.2 m. This is the approximate total length of the coil winding, as can be discovered by unwinding one. You must convert forces measured in grams to newtons. To do this, convert each value measured to kilograms and multiply the result by 9.8. How does the value for B which you obtain compare with the earth's field?

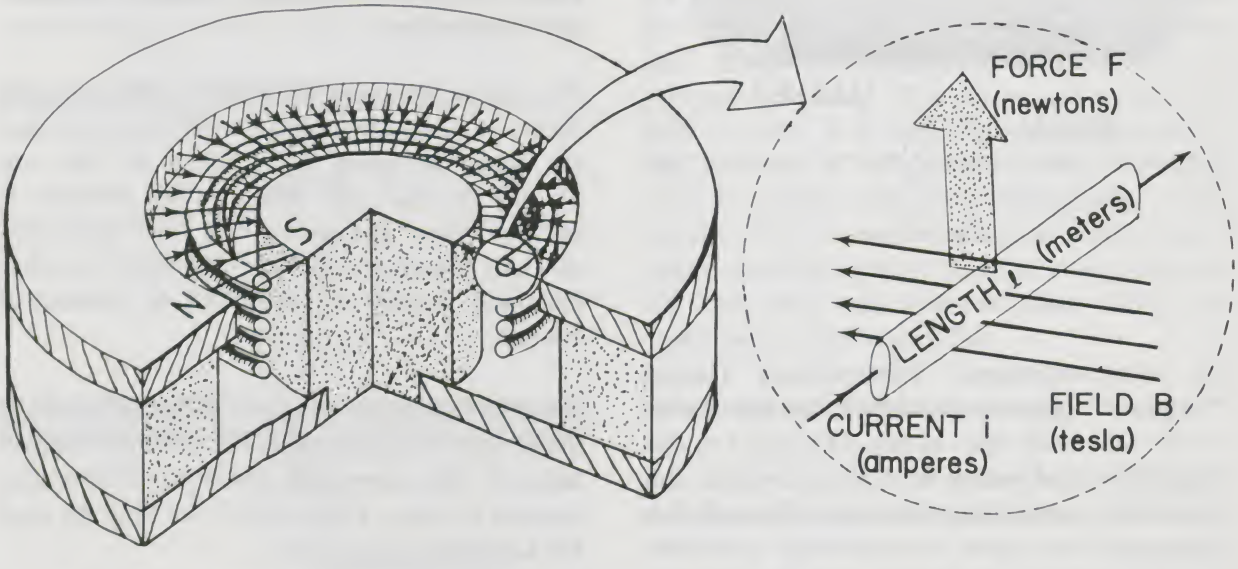


Figure 26. A cutaway view of a loudspeaker magnet, indicating some of the field lines and a few turns of the voice coil. A magnified view of part of one turn is also shown.

SUMMARY OF SECTION A

In Section A you were introduced to the basic ideas of sound, its generation and its detection. You examined a loudspeaker and located its basic elements, the *voice coil*, the *magnet gap*, and the *diaphragm*. You experimented with various electrical signal sources, becoming familiar with the parameters of *amplitude* and *frequency*.

You then did two experiments with the *electrodynamic* force of a loudspeaker voice coil, using an optical lever to magnify the motion. You first measured the *deflection* of the diaphragm produced by the force; then you measured the force itself, using a *null* measurement technique.

You thus observed the basic properties of this speaker, its linearity, its polarity, and its *sensitivity*. You learned how a null measurement allows determination of the electromagnetic force of the voice coil without knowing any details of the elastic forces of the speaker.

You learned about the *electrodynamic principle*, which has broad applications, and using that knowledge you were able to calculate the magnetic field which exists in your speaker.

In addition to working with the loudspeaker, you learned how a simple microphone operates. You found that your speaker could be run in reverse, forming a dynamic microphone, which operates on the *principle of induction*. The principle of induction can be regarded as the reverse of the electrodynamic principle.

QUESTIONS

1. Explain why the optical-lever technique is useful for measuring the deflection of the speaker diaphragm.

2. Explain the advantages of null measurements.

3. What property of a loudspeaker suffers if a graph of its motion versus applied current is not a straight line?

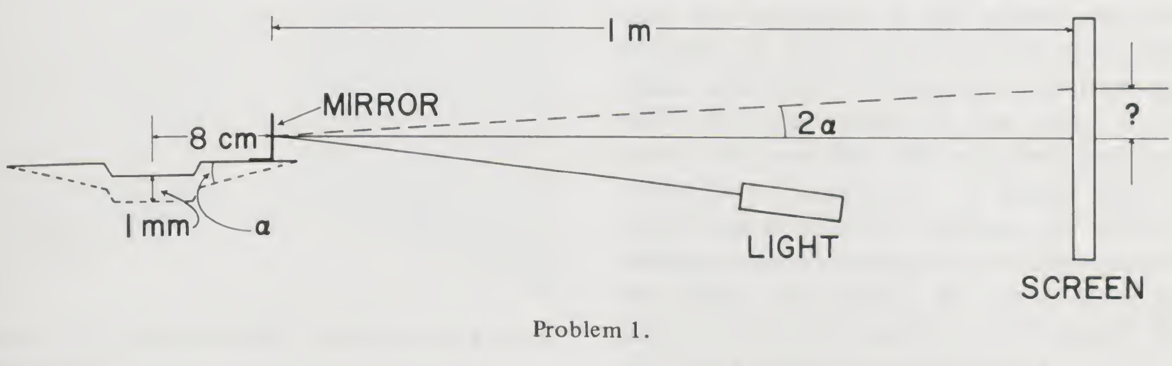
4. The electrodynamic principle is a relationship between what three quantities?

5. State the principle of induction. In what sense is this the reverse of the electrodynamic principle?

6. In what way is a dynamic loudspeaker similar to an electric motor? How is a dynamic mike like an electric generator?

PROBLEMS

1. Using geometry from the sketch below, calculate approximately how much deflection of the voice coil is needed to cause a 12-mm deflection of the spot on the screen. (Note that there are similar triangles of different sizes involved.)

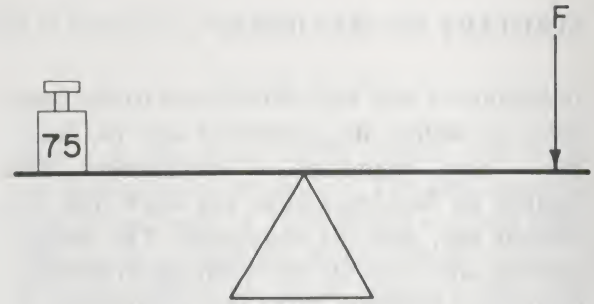


Problem 1.

2. If you have a coil made of 1 m of wire in a magnetic field of 4 T, what force will be produced by 1 A of current?

3. If you count two layers of wire, with 50 turns in each layer, on a 1-cm diameter voice coil, how many centimeters of wire must you use for your calculations?

4. If a certain force is balanced in an *equal-arm* beam balance by a 75-g weight, what is the force in newtons?



Problem 4.

SECTION B

Measuring Sound Radiation

SOUND SOURCES ARE NOT ISOTROPIC

Isotropic means "equal in every direction." Most sources radiate sound more in one direction than in others. They are called *anisotropic.* What shape would an isotropic source have to be? Probably spherical, since it would have to radiate in a perfectly spherical pattern. No one has actually constructed a vibrating spherical source. But designers of loudspeakers often try to build as isotropic a source as possible.

Ordinary Loudspeakers

Simple speakers, like your planar loudspeaker, radiate more frontwards and backwards than sideways. This is because the back-and-forth motion of the diaphragm compresses and decompresses air mainly on the front and back surfaces. So it is reasonable to expect that you hear less sound at the side than at the front.

Earlier, we suggested that you listen to your speaker from different angles to detect any differences in loudness. In this section you will measure these differences.

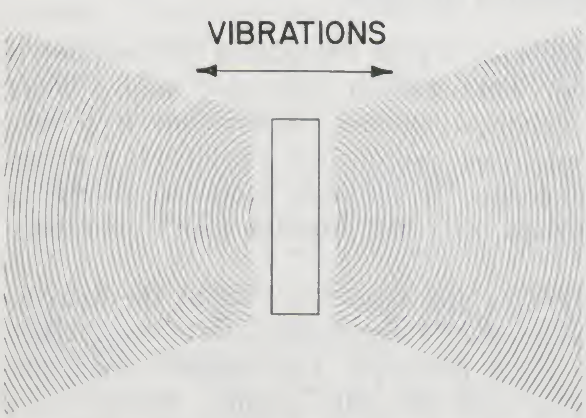


Figure 27. Sound is loudest in front and back of your speaker.

Loudhailers

Many devices are intentionally anisotropic. For instance, electronic megaphones, called *loudhailers,* are designed to amplify the sound of a person's voice and project it straight ahead as far as possible, so that it can be heard at a distance. Since sound is energy, any sound going off to the side is "wasted energy."

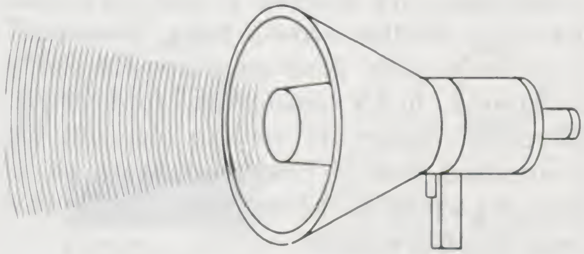


Figure 28. Loudhailer.

Hi-Fi Speakers

High fidelity loudspeakers are carefully made versions of basic speakers, put in cases called *enclosures* which are designed to improve their performance. They are also *anisotropic* radiators. This is undesirable, since it means that the loudness of the sound you hear depends on your position in the room. Even worse, the shape of the radiation pattern may differ for high notes and low notes. In this case, one says that the radiation pattern is "frequency dependent." It means that the sound has a different character, as well as a different overall loudness, in different areas of the room. Of course, the room itself also plays a role in the quality of hi-fi sound. You will study this further in Section C.

DETECTORS ARE ALSO ANISOTROPIC

Ordinary Microphones

As you know, microphones are transducers that work in the opposite way from speakers. All microphones convert sound energy to electrical energy. It is not really surprising that microphones are also anistropic; they are more sensitive to sound coming from one direction than from another. Sometimes this is desirable and sometimes it isn't. A microphone in the center of a table, used for recording voices all around the table, should be as *isotropic* (or *omnidirectional*) as possible.

Directional Microphones

These mikes are designed to pick up sounds from one location without being "swamped" by other sounds. They are purposely highly *anisotropic.* In TV broadcasting, for example, directional mikes are used to pick out the quarterback's voice from a distance, as he gives his signals at the line of scrimmage in a noisy football stadium.

Some of the most sophisticated modern technology has gone into microphones for spying. There are claims that a whispered conversation can be picked up by such devices at distances up to half a mile!

Figure 29. A directional microphone.

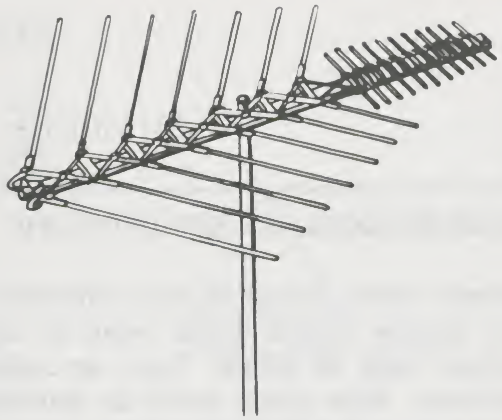


Figure 30. A directional TV antenna.

Other Examples of Anisotropy

Earlier it was mentioned that sound is a wave. Other types of waves include light, radio or TV signals, x-rays, etc. All waves have common properties. One of these is *anisotropy*—the property of having different strength, or different character, in different directions. For instance, flashlights or searchlights put out much stronger waves straight ahead than behind or to the sides. These are like loudhailers which use light instead of sound.

The extreme example of a directional light source is the modern *laser.* The beam from a laser may spread out so little that it can be detected even after traveling from earth to the moon and being reflected back to earth.

Similarly, a telescope is like a directional microphone. Both are extremely anisotropic detectors of waves.

Likewise, if your FM radio or TV reception is not good enough, you may need a more directional antenna. Again, this is a very anisotropic sensor. If it is pointed in a certain direction, it responds more sensitively to signals from that direction than from others.

Perhaps you have heard about the extreme example of a directional radio antenna, called a *radio telescope.* This operates much like your own FM radio antenna, although on a much larger scale. Some radio telescope antennas are as large as a football stadium and roughly the same shape.

MEASURING PATTERNS OF SOUND

Knowing the pattern of radiation from a source of waves (sound, light, radio, or what-have-you) is often of practical importance. In this section you will learn some ways to measure and describe patterns of sound radiation. You will record a pattern using a *polar plot* of your measurements.

A hi-fi loudspeaker provides a good example of the use of such polar plots. These are often furnished by the manufacturer, along with other data, to show how the speaker radiates in various directions.

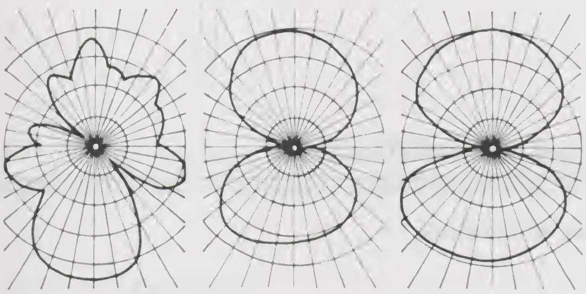


Figure 31. Radiation patterns for a typical speaker at different frequencies.

You Need the Right Detector

Your ear, of course, is a sensor which can *detect* sound, but it isn't good at *measuring* it. You need a detector which converts sound energy to a proportional amount of electrical energy. This allows you to use the large variety of instruments available for measuring electrical signals, including meters and oscilloscopes. As you know already, the right detector is a microphone.

Electrical energy is a convenient form for handling a signal for amplification and measurements. Thus it is very common practice to convert information into electrical signals, whether the information is about sound, changing temperature or pressure, or the speed of an engine. That is because electrical energy is easily manipulated; it can be carried on cables, amplified, displayed on a scope, or stored on a tape recorder.

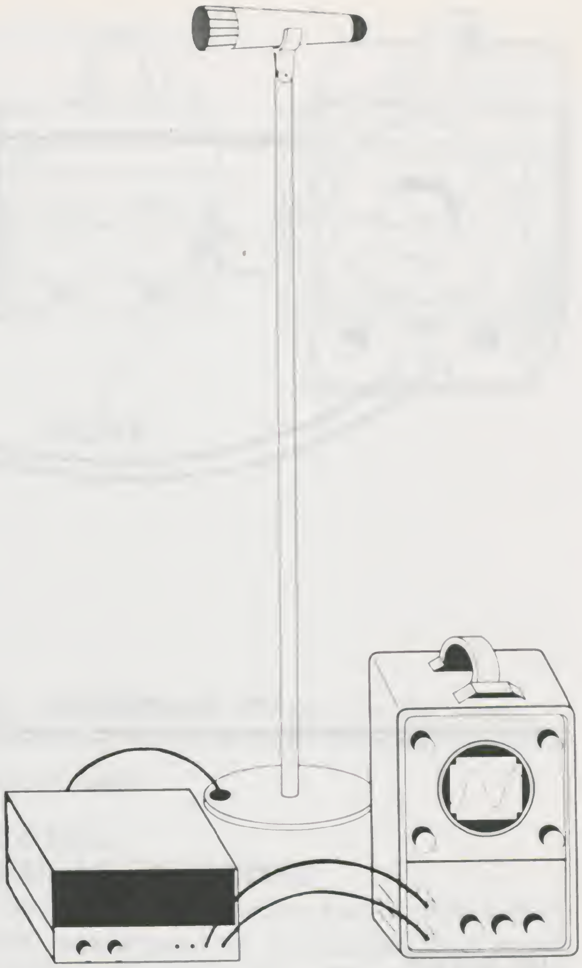


Figure 32. Microphones convert sound to electrical signals.

You Need Careful Procedure

The sound pattern of a given source is usually changed by its surroundings. (That is the purpose of speaker enclosures.) Therefore, in sound measurements, caution is needed to avoid having the room or the apparatus (or the experimenter himself) disturb the sound pattern.

Background noise is also a problem. To avoid background noise and echoes, sound professionals work in special quiet rooms, called *anechoic chambers.* These rooms have walls that absorb almost all sound which strikes them. You will not be able to use an anechoic chamber, but you should, if at all possible, do the following experiment in a reasonably quiet room.

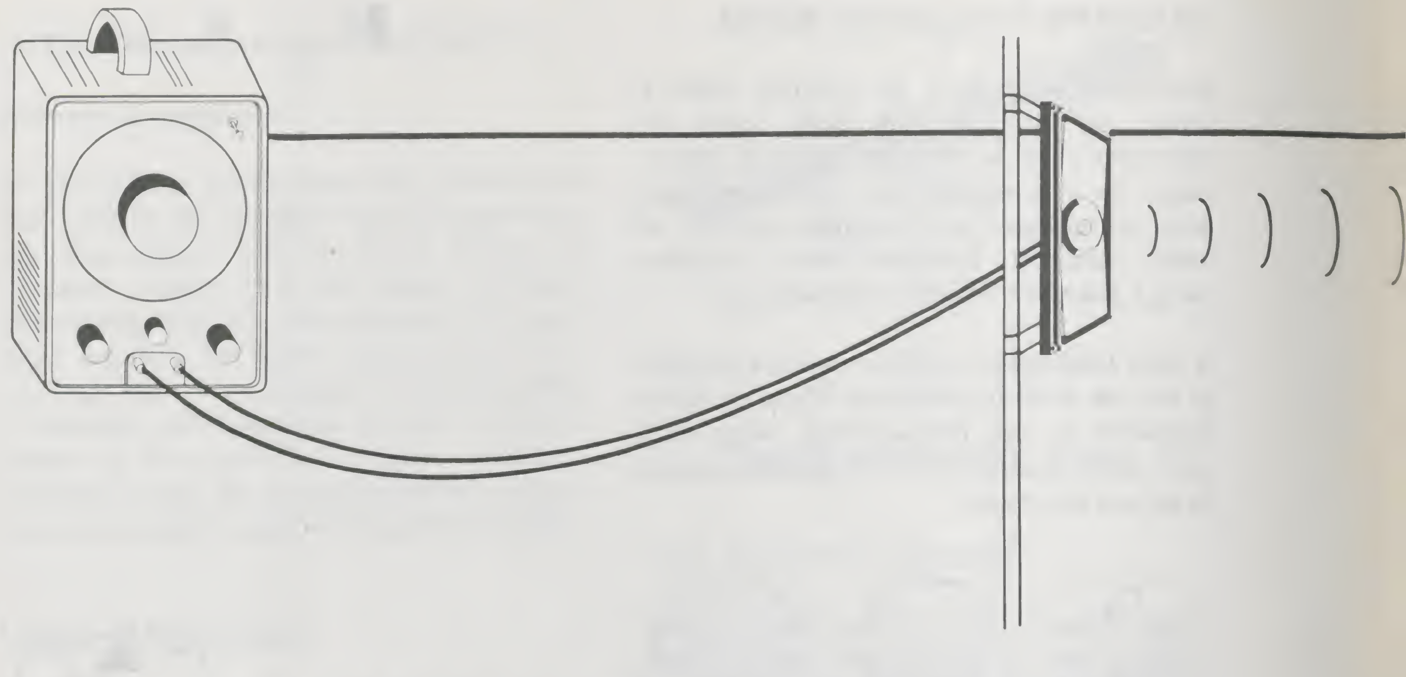


Figure 33. Summary of the setup.

EXPERIMENT B-1. Measuring the Pattern of Radiation

THE SETUP

The general idea of this experiment is to move the microphone around the speaker in a big circle, measuring the strength of sound radiation at different angles, in front, at the sides, and in back of the speaker. You will measure the properties of an *unbaffled* speaker, a speaker without an enclosure, backing, or mounting plate of any kind. In other words, you will study *the basic, unaltered properties of the speaker.* By the way, you might think for a moment about what limitations there are in your experimental setup that make this claim of studying the unaltered properties of the speaker not entirely accurate. After doing Section C, you should have an answer to this.

Oscillator

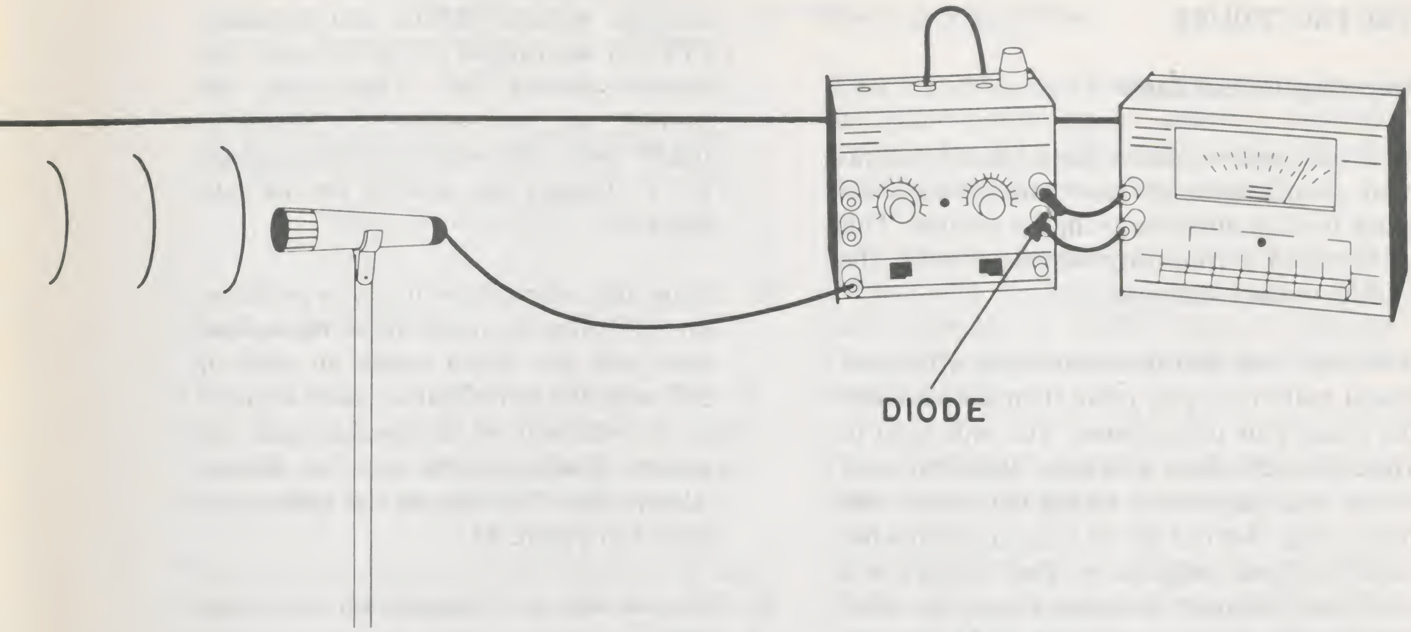
You will need an audio oscillator as a signal source. An oscillator produces a steady "pure" sinusoidal signal.

The frequency and strength of the output of the oscillator are adjustable. *Start with a frequency of 400 hertz (Hz).* Such a low frequency will bother your neighbors' experiments less. *Also, start with a low sound intensity level.* The sound should be clearly audible but certainly not loud or noticeably distorted.

Speaker and Microphone

Connect the oscillator to the speaker with clip leads. Mount the speaker vertically on a stand as shown. Keep as much of the surrounding area as possible clear of obstacles.

Mount the microphone on its stand so that it is at the same height as the center of the speaker. Both should be about one meter above the table top, if possible. The microphone will be used as the sensor to measure the intensity of the speaker's sound pattern at different locations around the speaker.



Then, connect the mike . . . to the amplifier and converter . . . and the DC meter.

AC-to-DC Converter

Connect the microphone output cable to the input of the preamplifier. Connect the output of the amplifier, through *an AC-to-DC converter,* to the DC meter. The converter changes the *alternating current* signal produced by the microphone into a *direct current* signal which can be measured by a voltmeter. The alternations of the signal produced by the microphone correspond to the rapid back-and-forth vibrations of air.

The simplest way to convert the AC signal to DC is to pass it through a device which allows current to flow in one direction only. Such a device is called a *rectifier.* In this method half of the alternating signal is rejected, while the other half is observed, using a DC-measuring instrument.

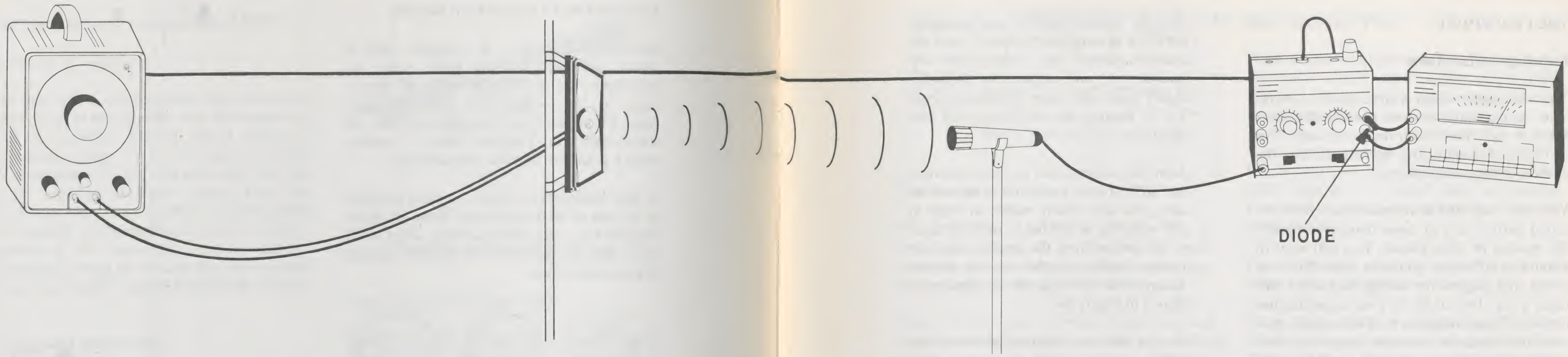
In your experiment, the converter consists of a single *diode rectifier* plugged into the output of the amplifier. The meter which follows the diode averages out any rapid variations which remain in the rectified signal from the amplifier.

DC Meter

The meter allows you to read numbers which are proportional to the amplitude of sound striking the microphone. This is because the size of the direct-current signal produced by the converter is proportional to the size of the original AC signal. It may be much bigger, because it is amplified, but it increases and decreases in a way which is proportional to the input signal.

The meter RANGE should be set to 2 V. Then adjust the amplifier OFFSET so the meter shows a very small reading (at most, 0.1 V) with no sound coming to the mike.

If you have an oscilloscope, it can be used as a replacement for the converter *and* the meter. You can see the size of the AC voltage directly on the scope. The full peak-to-peak voltage variation on the scope should be several millivolts if everything is hooked up right, if the sound level is comfortable, and if the microphone is about 25 cm from the speaker. If you do not know how to measure voltage with an oscilloscope, ask for help.



Use the oscillator. . . to drive the speaker.

Then, connect the mike . . . to the amplifier and converter . . . and the DC meter.

Figure 33. Summary of the setup.

EXPERIMENT B-1. Measuring the Pattern of Radiation

THE SETUP

The general idea of this experiment is to move the microphone around the speaker in a big circle, measuring the strength of sound radiation at different angles, in front, at the sides, and in back of the speaker. You will measure the properties of an *unbaffled* speaker, a speaker without an enclosure, backing, or mounting plate of any kind. In other words, you will study *the basic, unaltered properties of the speaker.* By the way, you might think for a moment about what limitations there are in your experimental setup that make this claim of studying the unaltered properties of the speaker not entirely accurate. After doing Section C, you should have an answer to this.

Oscillator

You will need an audio oscillator as a signal source. An oscillator produces a steady "pure" sinusoidal signal.

The frequency and strength of the output of the oscillator are adjustable. *Start with a frequency of 400 hertz (Hz).* Such a low frequency will bother your neighbors' experiments less. *Also, start with a low sound intensity level.* The sound should be clearly audible but certainly not loud or noticeably distorted.

Speaker and Microphone

Connect the oscillator to the speaker with clip leads. Mount the speaker vertically on a stand as shown. Keep as much of the surrounding area as possible clear of obstacles.

Mount the microphone on its stand so that it is at the same height as the center of the speaker. Both should be about one meter above the table top, if possible. The microphone will be used as the sensor to measure the intensity of the speaker's sound pattern at different locations around the speaker.

AC-to-DC Converter

Connect the microphone output cable to the input of the preamplifier. Connect the output of the amplifier, through *an AC-to-DC converter,* to the DC meter. The converter changes the *alternating current* signal produced by the microphone into a *direct current* signal which can be measured by a voltmeter. The alternations of the signal produced by the microphone correspond to the rapid back-and-forth vibrations of air.

The simplest way to convert the AC signal to DC is to pass it through a device which allows current to flow in one direction only. Such a device is called a *rectifier.* In this method half of the alternating signal is rejected, while the other half is observed, using a DC-measuring instrument.

In your experiment, the converter consists of a single *diode rectifier* plugged into the output of the amplifier. The meter which follows the diode averages out any rapid variations which remain in the rectified signal from the amplifier.

DC Meter

The meter allows you to read numbers which are proportional to the amplitude of sound striking the microphone. This is because the size of the direct-current signal produced by the converter is proportional to the size of the original AC signal. It may be much bigger, because it is amplified, but it increases and decreases in a way which is proportional to the input signal.

The meter RANGE should be set to 2 V. Then adjust the amplifier OFFSET so the meter shows a very small reading (at most, 0.1 V) with no sound coming to the mike.

If you have an oscilloscope, it can be used as a replacement for the converter *and* the meter. You can see the size of the AC voltage directly on the scope. The full peak-to-peak voltage variation on the scope should be several millivolts if everything is hooked up right, if the sound level is comfortable, and if the microphone is about 25 cm from the speaker. If you do not know how to measure voltage with an oscilloscope, ask for help.

THE PROCEDURE

Preparing the Lab Table

You will need to clear a large space to work. You should have at least one meter clear space in each direction from the speaker. This is to avoid having any objects disturb the motion of the sound waves.

You also may find that your arms affect the sound pattern as you move them around near the speaker or microphone. You will want to avoid that effect as you take data. Put your books and papers for taking data some distance away. Do not set up your apparatus too close to your neighbor's. Two meters is a minimum distance between setups to avoid interference. Also, you should avoid unnecessary noise.

Taking Data

1. *Place a protractor* at the base of the speaker stand, so that you can determine angles as you move the microphone around the speaker.

2. *Place the microphone about 25 cm away from the speaker.* Place it right on a line in front of the speaker. This line is called the speaker's "axis." All angles are measured relative to that axis line.

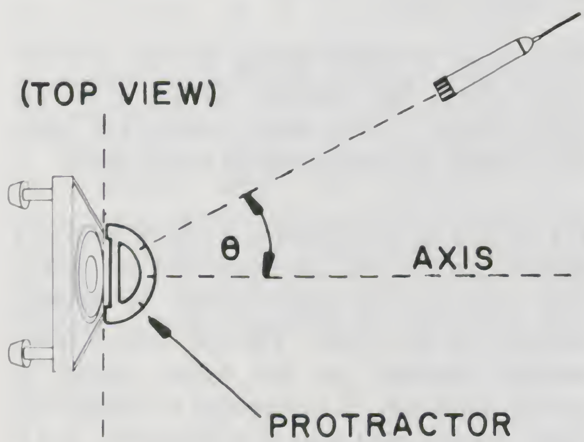


Figure 34. Measuring the angle to the microphone.

3. *Set the meter RANGE and amplifier OFFSET as outlined previously, with the speaker turned off. Then turn the speaker on, and adjust the amplifier GAIN until the meter reading is about 1.5 V.* Record this reading on the data sheet.

4. *Move the microphone to a new position, say 20°* off-axis, and *move it in and out along the line which makes an angle of 20°* with the axis. That is, move it closer to or farther from the speaker until the *voltage reading is the same as before.* Always aim the mike at the speaker, as shown in Figure 34.

5. *Record this new distance* on your data sheet.

6. *Repeat this procedure,* increasing the angle by 20° each time. You are looking, in each case, for the *distance* at which the *same voltage output* (the same sound level) is achieved. Obviously, as the sound gets weaker at certain angles, the microphone will have to be brought closer, and the distance will be smaller.

Recording Data

You will make a table of values of angles and corresponding distances, for one specific voltage output (sound intensity). Angles between 90° and 270° are *behind* the speaker. You will have to reposition the protractor to make measurements there, and make the necessary subtractions or additions to figure out the angles with reference to the axis. The angle increases as you move *counterclockwise* around the circle.

When you reach 360° you are back to the initial position. Do your measurements for 0° and 360° agree reasonably well? If not, repeat some of the other measurements to find any errors.

ANGLE θ	DISTANCE (cm)
0°	25
20°	
40°	

Figure 35. Sample data table.

The Background Noise

One nuisance you have to put up with in these experiments is the background noise from the other experiments in the room and from people moving around and talking.

Once everyone gets a speaker hooked up and running, the background noise from the other setups should be fairly constant. This is a great help. Since you are taking data by always looking for the *same* output signal, a constant background noise will affect each of your measurements the same way.

For example, the microphone signal may be 4 mV at the initial location. Let us imagine you turn off your speaker, and find that 1 mV of signal is due to the background sound from everyone else's experiment. Thus the difference must result from the sound from your speaker, which is added to the background noise. Now at a later location, you adjust the distance to obtain the same output voltage, 4 mV. If the background noise is the same, then you know the sound from *your* speaker is again the same. Graphing these results, you get a plot of positions of equal intensity. Such a plot is called an *isointensity* plot.

This is just another relative measurement, like the null measurement of Section A. Unwanted factors, such as the sensitivity of the microphone and the background noise level cancel out. Of course, voice and other *transient* noises (that is, ones which change) do not cancel out, and you should not let them confuse your data.

The Isointensity Plot

After the lab session is over, plot your data on the *polar graph paper* which is provided. Connect the data points with a smooth curve to make the pattern more obvious.

Note that each radial line on the graph is labeled with its angular measurement. Each circle should be labeled with a distance. Choose a suitable scale for labeling the concentric circles.

Plot each pair of values for angle and distance. *Draw a small circle around each point.*

Draw a smooth curve to represent the shape of the pattern of points. Some points will be "out of line" due to experimental error.

These data make up an *isointensity* plot—a plot of all points around the speaker at which the intensity is the same.

The shape of the plot shows the *pattern* of sound distribution around the source. It indicates the nature of the speaker output. The closer to the origin (center) the plot goes, the weaker the sound radiation was in that direction.

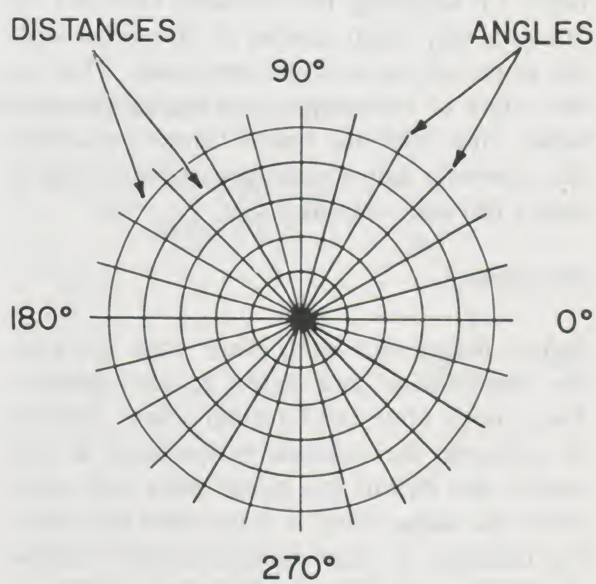


Figure 36. Polar graph.

EXPERIMENT B-2. Measuring the Effect of Distance

The idea that the loudness of sound decreases as one moves farther away from the sound source is familiar from everyday experience. It is perhaps not quite so obvious why sound (or indeed any other wave phenomenon) loses intensity as it moves away from the source. Also, what determines the rate of reduction as waves expand to cover larger and larger distances?

In this experiment, you will determine how the intensity of sound from your speaker decreases with increasing distance along the axis of the speaker. Measurements will be made using the same equipment as before. In the discussion, you will find that your results can be explained on the basis of *energy conservation.*

Setup

Start with the setup used at the beginning of Experiment B-1. The microphone should be placed a little nearer to the speaker, perhaps 10 cm away (on-axis). The sound level should be at a comfortable value, with the oscillator frequency set at about 400 Hz.

Begin by adjusting the amplifier OFFSET to obtain a very small reading on the meter with the sound off, as outlined previously. This has the effect of subtracting out the background noise. Then with the sound turned up, adjust the converter gain so that the meter reading is nearly full scale–about 2.0 V.

Procedure

Before taking any data, place your ear near the microphone and listen to the speaker. Then move your head farther away, and try to estimate the changes in loudness of the sound. See if you can guess what will occur when the same thing is done with the mike. For instance, at what distance is the loudness one-half as great? One-fourth? One-tenth?

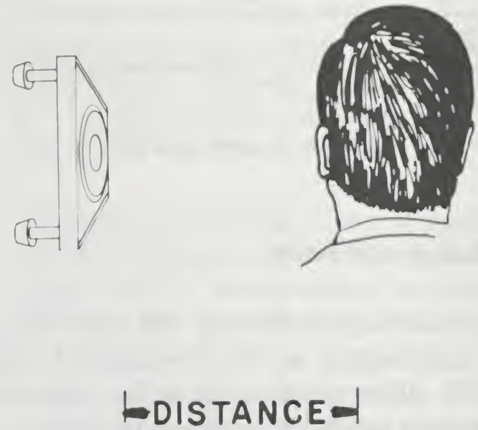


Figure 37. Estimating loudness with your ear.

Now measure microphone voltage versus distance along the axis. Record your data in the table provided. Put a title on the data sheet. Label the columns for distance and voltage.

Increase the distance in small steps, about 10 cm each, recording the voltage at each distance. Again, when recording voltage, do so at a moment when no transient noises, such as talking, are present. Continue the measurements out to at least one meter from the speaker. At the end of your data-taking, check to be sure the meter still reads zero with your speaker off.

Plot your data on a piece of linear graph paper. How well do the results agree with what you estimated?

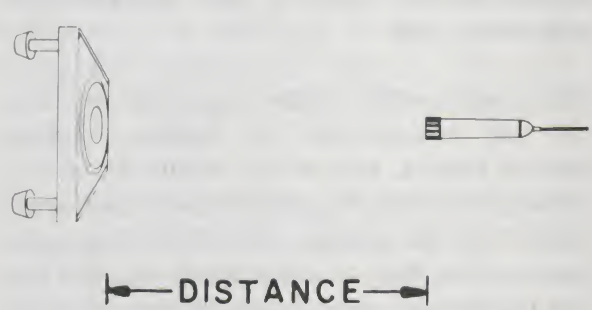


Figure 38. Measuring intensity along the axis.

EXPERIMENT B-3 (OPTIONAL). Family of Isointensity Plots

Repeat Experiment B-1 a couple of times, starting with a different frequency or a different distance of the microphone from the speaker. This will provide a *family* of isointensity plots, which will show how the behavior varies with frequency and distance as well as how it depends on the angle. You can use the same data table and polar graph paper as before. You might try a frequency of 5000 Hz, for example.

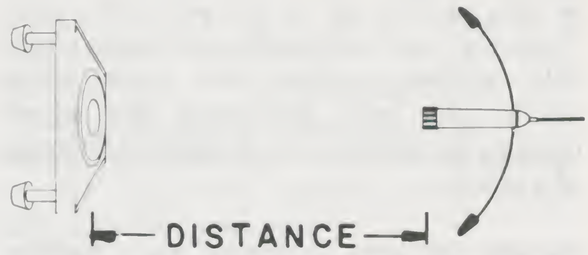


Figure 39. Repeat the first experiment using a different distance or frequency.

EXPERIMENT B-4 (OPTIONAL). Directional Character of the Mike

Measure the microphone pickup pattern by doing an experiment similar to what you did with the speaker. The easiest way to do it is to keep the microphone located in a constant sound field, for example right in front of the speaker, and rotate the mike through various angles. Be sure to keep the part of the mike at which the sound enters in the same position (Figure 40). Record *angles* and *voltage* outputs. Record your data, and plot it on a spare piece of polar graph paper. What are the differences between this plot and your earlier plots?

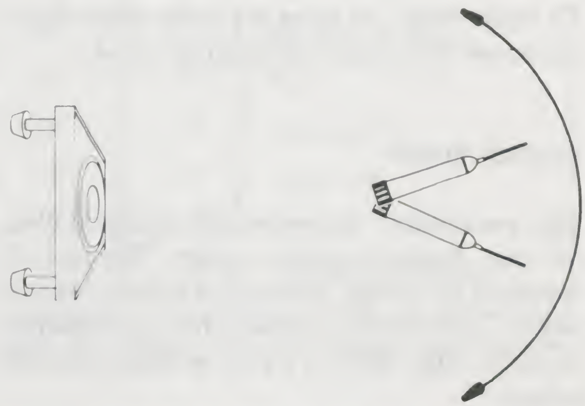


Figure 40. Try rotating the mike itself.

EXPERIMENT B-5 (OPTIONAL). Spherical Radiators

Measure the pattern of a "baffled" speaker by mounting your speaker over a hole in a large plywood sheet. Repeat the isointensity plot on the front side of the board. What has changed? Why could such an arrangement be called a "spherical radiator"? In the jargon of high fidelity, a speaker enclosure which makes it impossible for any sound from the rear to reach the space in front is called an "infinite baffle." This can be done very easily, for instance, by mounting the speaker in a closet door. Such a mounting improves the radiation pattern.

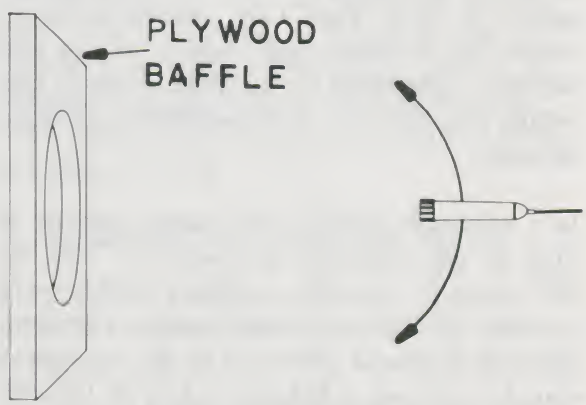


Figure 41. Mount the speaker on a large baffle.

In this module up to this point, the term "intensity" has been used several times. Probably without worrying too much about details, you have understood that sound intensity is related to the strength or loudness of a sound.

Actually the term "intensity" has a meaning which is different from "loudness," although the two terms are closely related. Being careful to distinguish them will lead to a better understanding of your experiments, especially Experiment B-2. Also, it will reveal an interesting fact about human hearing.

To begin with, we must say a bit more about the *power* involved in generating sound.

Acoustic Power

The production of sound (or anything else, for that matter) requires energy. The *rate* of radiation of energy in sound is called *acoustic power.* Like electric power, this is measured in watts. One watt is a rate of one joule per second.

The power represented by most familiar sounds is not large. For instance, speaking at a conversational level generates only about ten millionths of a watt (10^{-5} W). Thus the power in sound generated by all of the ten million people in New York City, talking at once, would be no more than enough to run one ordinary light bulb! If you think about it, this means that your ear is an incredibly sensitive detector.

In a very soft whisper, the power may be as little as one billionth of a watt (10^{-9} W). At the opposite extreme, shouting loudly may produce 10^{-3} W of acoustic power. For comparison, the most powerful of all man-made sounds, that of a Saturn rocket at lift off, radiates about 40 million (4×10^7) watts.

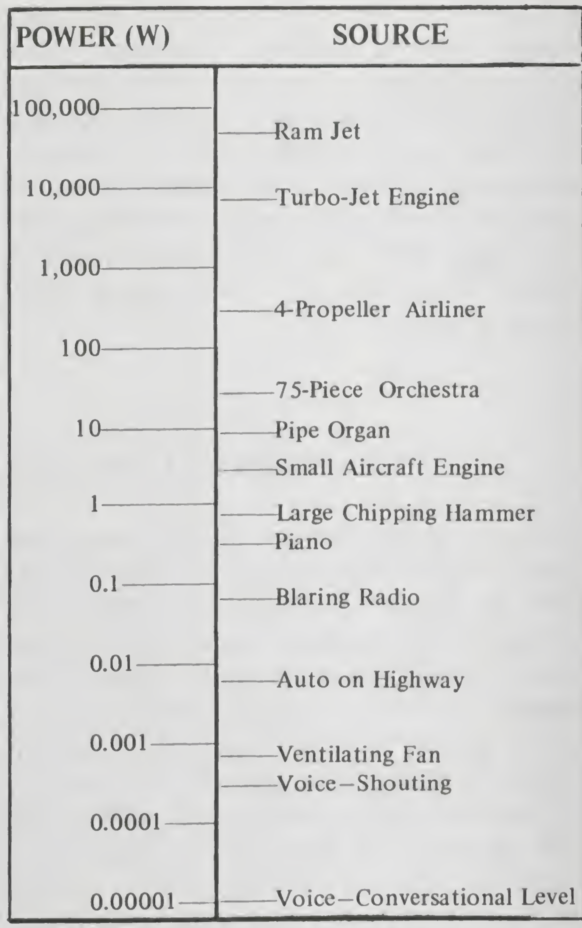


Figure 42. Typical values of acoustic power. The power scale is *logarithmic,* to squeeze in the very large range of values.

Intensity

Usually the *intensity* of a sound is of more interest than the total radiated power. Intensity is the wave power crossing any given small area in the sound field, divided by that area. It is measured in units of watts per square meter (W/m^2).

A detector of sound, such as a microphone or your ear, picks up only that part of the sound power which crosses the detector's own area. If the sound intensity is I, then:

$$\text{Power detected} = I \times \text{area}$$

The Loudness of Sound

In contrast to intensity, *loudness* is what your mind *thinks* the strength is, based on what your ears tell it. Of course, different people have different hearing sensitivity, so the term "loudness" refers to an average person's hearing.

Surprisingly, hearing experiments have shown that loudness is proportional to the *logarithm of the intensity* of sound. This means that each additional multiplication by ten in intensity is just an increase of one more unit in loudness. This "logarithmic behavior" of the ear is very important, since it affects the design of all things which are used to produce sound.

The Decibel Scale

The unit of loudness mentioned above, representing one decade (factor of 10) in intensity, is called the *bel.* Zero bels is defined as the loudness an average person can just begin to hear—the *threshold of hearing.* By convention, the intensity of this sound level is taken to be exactly 10^{-12} W/m^2.

The bel is a large unit, so it is commonly divided into ten subdivisions called *decibels* (dB). Thus an *additive* increase of 10 dB (or one bel) is a *factor* of ten increase in intensity. Likewise an additive increase of 20 dB is a factor of 100 in intensity, and so on. Thus the noise of a train at 80 dB loudness has 100 times the intensity of a cocktail conversation of 60 dB.

It takes a while to get used to working with a decibel scale. But it is very commonly used in specifications for audio equipment. That is because such equipment is designed for appreciation by humans, and it therefore makes sense to use a scale which corresponds to the sensitivity of human hearing.

In the table below, we show you some common sounds and their locations on the decibel scale. The scale spans a range of a factor of 10^{12} in intensity, so it shows how versatile the ear is. It can handle a huge range of levels, allowing us to hear amazingly weak sounds without being overloaded by ones that are 10^{12} times more powerful. Thus, the ear has a large *dynamic range.*

INTENSITY (W/m^2)	LOUDNESS (dB)	SOUNDS
10^{0}	120	Threshold of pain
10^{-2}	100	Jet plane at 30 m
10^{-4}	80	Train
10^{-6}	60	Conversation
10^{-8}	40	Quiet radio
10^{-10}	20	Whisper
10^{-12}	0	Threshold of hearing

Figure 43. Typical values of intensity and loudness.

Notice some of the interesting properties of a logarithmic scale. The *difference* in loudness between 100 dB and 120 dB is the same as the difference between 20 and 40 dB, even though the first case involves 10^8 times more intensity than the second. Notice on the scale below that a doubling of intensity does not produce a doubling of loudness. Instead, the increase in loudness is proportional to log 2, or 0.3. That means an increase of 0.3 bels, by definition, or 3 dB.

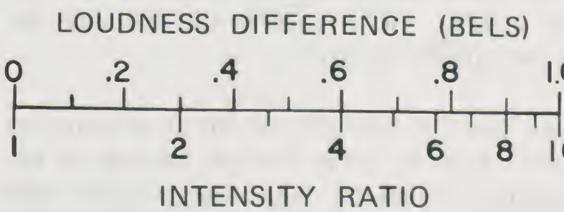


Figure 44.

EXAMINING YOUR RESULTS

At this point, you may have realized why your ear is not a good guide to predicting microphone results, as in Experiment B-2. The ear measures loudness, while the mike output reveals something else; something which is more closely related to the sound intensity. That something is the sound *amplitude.*

Amplitude

The *amplitude* of sound is the amount of vibration in the wave, measured in units of pressure difference. For instance, as the wave travels along, if the air pressure varies above and below atmospheric pressure by a maximum of $10\ N/m^2$, then that is the sound amplitude: $10\ N/m^2$.

An amplitude this large, by the way, represents an extremely loud, even painful, sound. Since normal atmospheric pressure is about 10^4 times larger, you can see that we are dealing with extremely small pressure changes in sound.

A microphone can be used to measure sound amplitude because the voltage produced by a mike is usually proportional to the pressure change. Thus the strength of the electrical signal, as indicated on your meter for example, is a direct measure of the amplitude of the sound.

The Data

Your meter readings in Experiment B-2 produced a graph like the one sketched in Figure 45. This shows that the sound amplitude goes down quite rapidly as you move away from the speaker. (Why doesn't the loudness decrease equally rapidly?)

Note that the readings do not go to zero, no matter how far away you go, because of the amplifier "offset" adjustment and also because of the constant background noise.

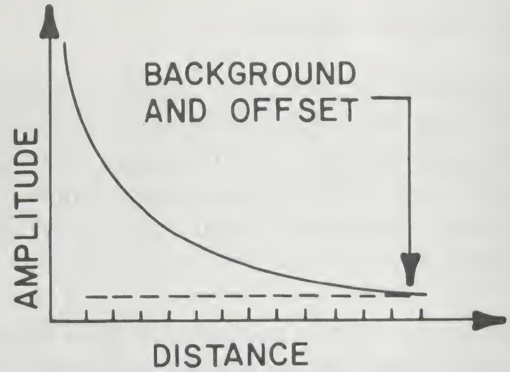


Figure 45. Amplitude decreases rapidly with distance.

How far out does it make sense to continue your graph? Only until you can no longer see significant changes in the true signal above this final voltage level. Probably that happened at a distance between one and two meters in the actual experiment.

Replotting the Amplitude

Something interesting about the amplitude behavior is shown by making a new type of graph. *Replot your data using the inverse distance $1/d$ instead of d itself.* This new plot should look like Figure 46. You see essentially a *straight line* graph of amplitude versus inverse distance. Naturally this starts from the arbitrary offset level at $1/d = 0$ (infinite distance), but above that level the increase is basically *linear* as the detector approaches the speaker from reasonably large distances.

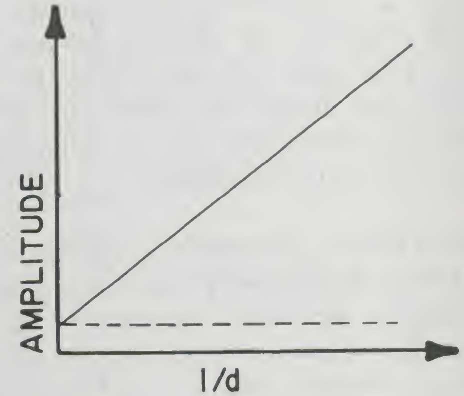


Figure 46. The plot is nearly linear with inverse distance.

The Ideal Behavior

If your results were not perfect because of unwanted noise, reflections, interference, etc., that is of minor concern. The ideal behavior of the amplitude with inverse distance is exactly linear for waves expanding outward in all directions from a single source. That is: $A \propto 1/d$, where A is the amplitude and d is distance from the source. The symbol $\propto$ stands for "is proportional to."

Intensity and Amplitude

The preceding simple result has a simple explanation. To seek it, we must first express the result in terms of the intensity, rather than amplitude. This is possible because of the following general rule: intensity is proportional to the square of the wave amplitude. In symbols:

$$I \propto A^2$$

So, if the amplitude of a wave doubles, the intensity quadruples. If the amplitude is halved, the intensity is quartered, and so on.

The Inverse Square Law

To find the ideal behavior of the intensity, simply combine the preceding relations. Square both A and $1/d$ in the first relation, finding $A^2 \propto 1/d^2$. Then, using the second relation, replace A^2 with I. The result is called the *Inverse Square Law of Radiation*, which reads as follows:

$$I \propto \frac{1}{d^2}$$

Or, in words, intensity is proportional to the inverse distance squared. The inverse square law has some limitations, but it represents the simplest type of behavior possible for expanding waves. They *must* weaken with increasing distance, and they do so more *or* less in this simple fashion. To see that, you must recognize that waves carry energy as they travel, and that energy must be conserved.

Conservation Explains the Behavior

Energy can be neither created nor destroyed, but only converted between various forms. The total amount of energy remains constant. In physics, this is what we mean by energy conservation. The forms of energy include light, heat, motion (kinetic energy), and so on. As you have learned in this module, sound also represents one form of energy.

As sound waves expand, the energy in them is spread out over an ever-increasing area. In fact, the area increases like d^2 (where again d is the distance from the source). To see this, think of the surface of a large balloon being blown up to larger and larger diameters. The formula for its surface area is $\pi d^2/4$.

As a balloon expands, it gets thinner, because the mass is conserved. Likewise, the energy of an expanding wave is spread over a larger and larger area more and more "thinly" because of energy conservation. If you think about it, this "thinness" is measured by the *intensity*. So indeed the intensity goes down as the area covered by the sound goes up.

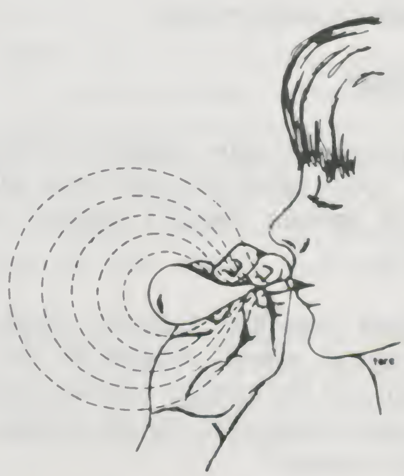


Figure 47. Because its mass is conserved, a balloon gets thinner as the area gets larger.

LOUDSPEAKER SPECIFICATIONS

Anyone shopping for a loudspeaker meets a bewildering variety of types and prices. So-called "hi-fi" speakers, for example, may range in price from only a few dollars to as much as a thousand dollars. What is the real difference between speakers? How can a buyer make a sensible choice, according to his needs?

Figure 48. High-fidelity speaker system.

To help answer these questions, manufacturers provide data sheets, called *technical specifications* (or "specs"), which tell you about a speaker's strengths and weaknesses. These can be quite complicated, and may demand a fair amount of study if one is to avoid being misled. Basically, however, one need only answer a few questions about a loudspeaker's performance.

Power Specs

As you know by now, sound is energy. The speaker specs which are concerned with the way the speaker handles energy are the following.

1. *Power Handling Capacity.* How much electrical power, applied to the voice coil, can the speaker tolerate before the sound is distorted and the speaker possibly damaged?

2. *Efficiency.* What fraction of power used to drive the speaker is converted to sound (not heat, etc.)?

If the electrical power driving the speaker is P_{in}, and the sound power output is P_{out}, the efficiency is defined as the *ratio*:

$$E = \frac{P_{out}}{P_{in}} \times 100\%$$

For the speakers you have seen, this is usually about 1% or less, which of course means that almost all of the input power is "wasted." In hi-fi systems for the home, that is not important, since even 1 watt of sound power in an ordinary room is excessive. The low efficiency of a cone speaker is more than offset by its strong points: uniform frequency response and linearity.

In public-address applications, these priorities are reversed. To fill a stadium or a theater with sound may require tens of watts of acoustical power. Thus 1% efficiency is intolerable, since the amplifier power requirement would be thousands of watts! Fortunately, another type of loudspeaker, the *horn* loudspeaker, is commonly 20 to 50% efficient. For this reason, horns are the speakers of choice for P.A. systems.

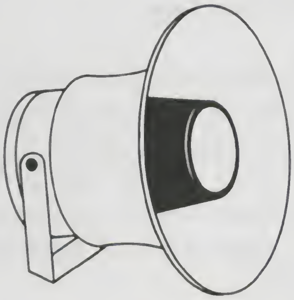


Figure 49. Public-address speaker.

Frequency Specs

A *pure* musical tone involves perfectly *sinusoidal* vibrations of the air pressure. The number of complete vibrations per second is called the *frequency* of the tone. The unit of frequency is the hertz (abbreviation Hz). One hertz is just one vibration per second.

The ear does not respond to frequencies

below about 20 Hz or above roughly 20,000 Hz. This range of hearing varies from person to person, and usually decreases greatly with age.

Of course, most sounds are not pure tones at all. Rather, they are complex patterns of air vibration with no single frequency. However, *any* sound can be formed from a number of pure tones all added together, much as the individual instruments of an orchestra, each playing single notes, combine to make the complex sound of a symphony. The individual tones which make up a given sound are called its *spectrum.*

One of the most important requirements for high-fidelity speakers is to reproduce the entire spectrum of audible sounds. This leads to the next specification:

3. *Frequency Response.* If the speaker is driven with pure tones, how much does the efficiency vary with the frequency of the pure tones?

Speaker manufacturers usually provide graphs which illustrate the frequency response of a speaker. These show the sound output intensity or loudness versus frequency for a constant input power. A typical plot for the planar speaker you have used is shown in Figure 50.

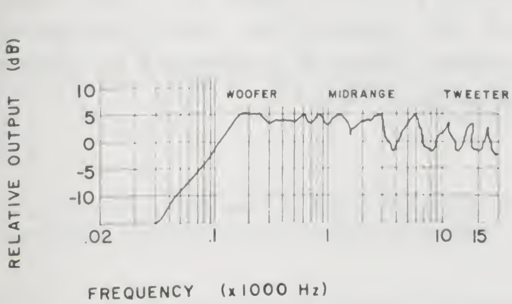


Figure 50. Frequency response of a planar speaker.

Three kinds of speakers are usually found in a hi-fi speaker system, each one handling a different range of frequencies. Figure 51 illustrates the frequency responses of these three speaker types. *Woofers* are quite big, perhaps 12 inches in diameter, and *tweeters* are smallest, perhaps only 3 inches in diameter. Your planar speaker is primarily a midrange unit, also useful as a general purpose speaker.

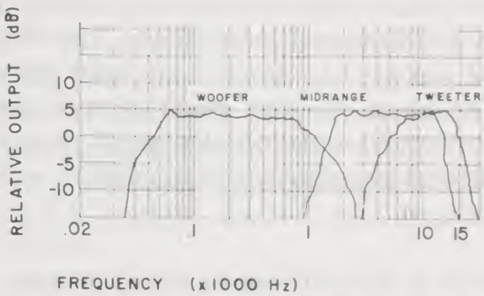


Figure 51. Frequency response of special purpose speakers.

Other Specs (Optional)

Several other types of specifications occur which should be listed. While these will not be discussed at any length here, you will recognize that they relate to some of your other work in the module.

4. *Directivity.* How much does the loudness of the radiated sound depend on direction?

The directivity is usually illustrated with polar plots which show the loudness versus direction at a *constant distance* from the speaker. Thus, although such plots look very similar to your own isointensity plots, the meaning is a little different. Since the directivity depends on the frequency, often two or three plots are provided (for low, middle, and high ranges).

5. *Linearity.* If the speaker is driven with one pure tone, how pure is the output sound? (The failure to produce a pure output is called *harmonic distortion.*)

6. *Transient Response.* If the input signal suddenly changes (either in pitch or loudness), how quickly and faithfully does the output follow?

SUMMARY OF SECTION B

In Section B you examined the properties of your loudspeaker in more detail. It is an *anisotropic* sound source. To measure the *directionality* of its sound radiation, you used a microphone as a sensor. Caution was required to obtain valid data in the presence of background noise and interference. The data obtained was used to produce an *isointensity plot.* The shape of that graph showed the pattern of sound radiation. A separate experiment showed the decrease in amplitude with distance.

Amplitude is proportional to sound pressure. *Intensity* is proportional to sound power and to the square of the amplitude. *Loudness* is proportional to the logarithm of intensity. It is measured in *decibels.* Zero decibels corresponds to the threshold of human hearing.

Using this knowledge, you can examine more critically the specifications of commercial speakers and speaker *systems* (sets of speakers in *enclosures*). Common specifications include plots of *frequency response* and *directionality.* Frequency is measured in *hertz.* The ear responds only to sound in the range roughly from 20 Hz to 20,000 Hz.

QUESTIONS

1. Sketch an isointensity plot for an *isotropic* sound source.

2. Sketch an isointensity plot for a highly directional microphone used for picking out the signals of the quarterback in a football stadium.

3. Explain why it is undesirable for a speaker to become more directional at high frequencies than it is at low frequencies.

4. Explain what information you get from a *family* of isointensity plots in addition to the information supplied by a single plot.

5. When taking data for an isointensity plot of the speaker output, why should the microphone always point *at* the speaker as it is moved to various positions around the speaker?

PROBLEMS

1. How much more *intensity* is there in the sound of a jet plane at 30 m (100 dB) than there is in a whisper (20 dB)?

2. If you put 100 W of electrical energy into a loudspeaker and obtain 0.5 W of sound energy, what is the speaker's efficiency, and what type of speaker is it most probably?

3. If you know that the AC-to-DC converter puts out 0.05 V for a 1-mV input to the preamplifier, what microphone output voltage is represented by a meter reading of 10 mV?

SECTION C

Using Sound Waves

This section of the module deals with ways to use sound, not by familiar means, such as talking or listening, but by studying and exploiting the behavior of sound as a *wave*. The emphasis here is on *understanding* the basic nature of sound and seeing how that understanding opens the way toward new applications. There will be some experimental work to do, but it will be less detailed and more exploratory than was your previous work. You should focus more on the ideas than on the numbers.

Reminder: Waves Travel

Waves of various kinds are extremely common in nature: there are light waves, water waves, sound waves, etc. Ordinary sound is a wave in air. As a sound wave passes by, you can observe a rapid, repetitive variation of the air pressure–up and down–using a microphone as a sensor.

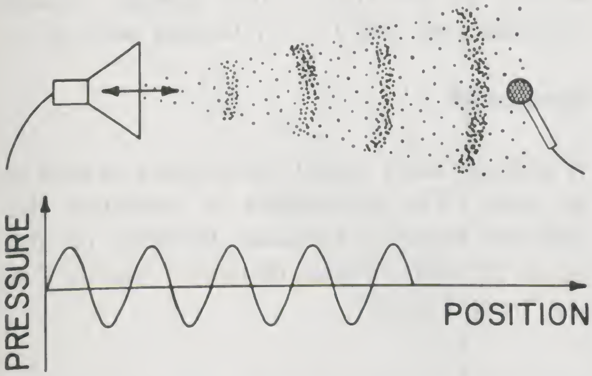


Figure 52. A sound wave in space and its graph. The baseline for the graph is atmospheric pressure.

If you could look closely enough, you would find that, as a sound wave travels through air, the air molecules move back and forth. However, the motion of the molecules is *not* the same as the overall motion of the wave itself. That is, the molecules oscillate back and forth in a given location while the wave travels from one place to another.

Transverse and Longitudinal Waves

Perhaps you have noticed the difference between *particle motion* and *wave motion* for other waves. A wave on a piece of rope involves *transverse* (sideways) motion of the rope itself, whereas the wave travels *along* the rope. The *transverse* wave motion and the particle motion are at *right angles* to one another.

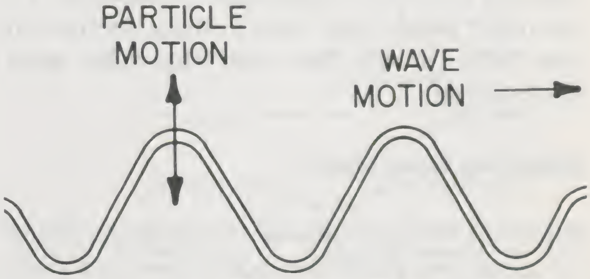


Figure 53. Transverse wave.

By contrast, a compressional wave on a spring involves *longitudinal* motion–back-and-forth vibrations–while the wave travels along the spring. The *longitudinal* wave motion and particle motion are *parallel.*

A sound wave is a *longitudinal* wave in air. The air molecules at a given spot move a short distance back and forth as the wave travels outward from the source. Transverse sound waves are not possible in air.

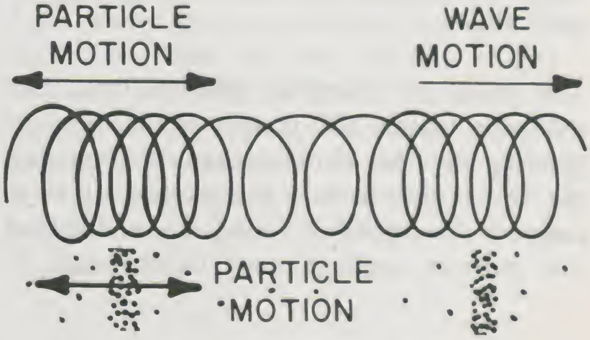


Figure 54. Longitudinal waves.

THE SPEED OF WAVES

One important thing to know about a wave is the *speed* with which it travels along, carrying energy from one place to another. As a wave travels through matter, it moves at a speed which is different from the speed at which the particles of matter are moving.

Usually the speed of the particles is of little interest, while the wave speed matters a great deal. For instance, if an earthquake creates a tidal wave at sea which approaches the shore, the inhabitants waiting for its arrival probably do not care how fast the surface of the water bobs up and down as the wave passes by. On the other hand, they have a major interest in how long it will be until the tidal wave reaches them.

Measuring Wave Speed

When anything moves, its speed can be found by measuring the distance it moves in a given length of time. Then, the definition of speed is:

$$\text{Speed} = \text{distance}/\text{time}$$

The problem of measuring the speed of a wave accurately can be rather tricky, however, and the method chosen depends on the type of wave. This is because of the vast differences between the speeds of various waves. While a wave on a rope travels only a few meters per second, a light wave travels about 300 million meters per second. Thus, to measure the speed of light directly requires the use of either very long distances or very short times.

The speed of sound is between these two extremes, about 300 meters per second, depending on the air temperature, humidity, etc. One of your aims in this section will be to measure the speed of sound. To understand the indirect method used (which can be applied to most waves, including light), one or two other simple, yet general, ideas are needed.

Wave Period

A *periodic wave* is a repetitive phenomenon. Whatever type of wave it is, the action involved (water waving, air vibrating, etc.) repeats again and again. This can happen slowly (geological waves in earthquakes) or rapidly (radio waves). A pure tone of sound is the simplest and most familiar example of a periodic wave.

The *period* of a periodic wave is the length of time between repetitions. For instance, if there are 100 repetitions per second, then a new repetition occurs each 1/100th of a second. The time between repetitions is the wave period T. (T stands for "time.") In other words, $T = 0.01$ s for this example.

Incidentally, this example shows the relation between the period T of a wave and its frequency f. The frequency is just the number of repetitions per second (here, 100 per second, or 100 Hz). A little thought should convince you that $T = 1/f$ for any value of T.

Wavelength

A periodic wave repeats in distance as well as in time. The *wavelength* λ describes the distance between *identical portions* of the wave. It is the "repeat distance," just as T is the "repeat time."

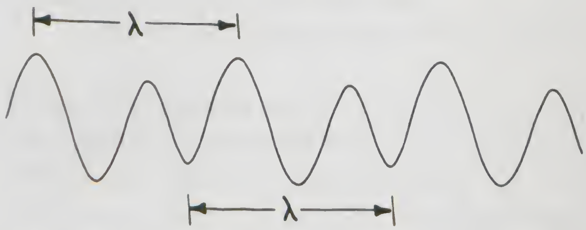


Figure 55. The wavelength of a periodic wave.

A Fundamental Relation

The distance a periodic wave travels in exactly one period is the wavelength. This is because the repetition in time is a result of identical portions of the wave passing a fixed location. Calling the speed s, we can therefore write $s = \lambda/T$.

But dividing by T is the same as multiplying by f. Thus, for *any* periodic wave,

$$s = \lambda f$$

The speed of a periodic wave equals its wavelength times its frequency.

Waves Reflect

One interesting property of all waves is that they are reflected by certain surfaces. You may be familiar with the reflection of water waves from a wall or shore line. You are certainly familiar with the reflection of light from shiny surfaces (although you have to take our word that light is basically wavelike).

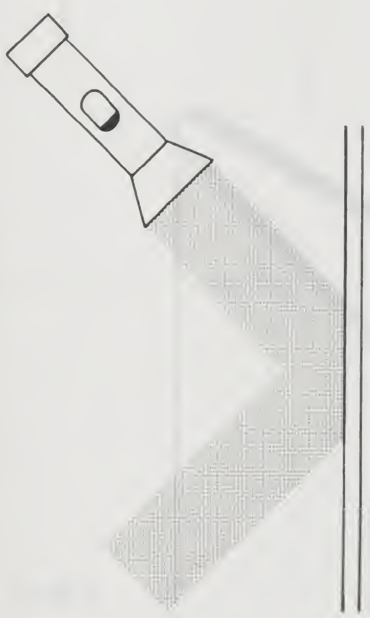


Figure 56. Light waves are reflected by a shiny surface.

If you imagine waves in a stretched spring or in a rope attached to a wall at one end (Figure 57), you can see that reflections also take place there.

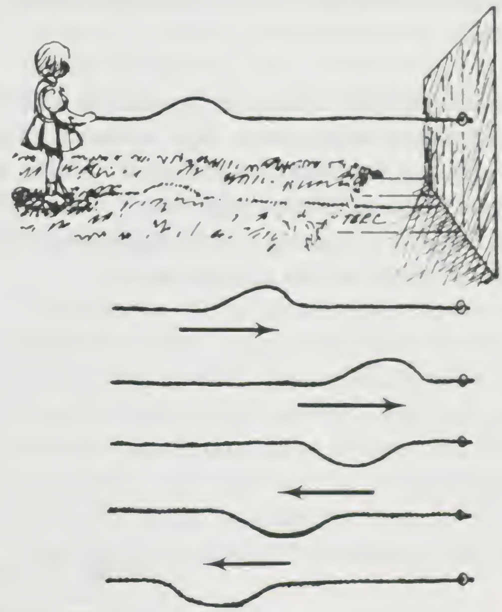


Figure 57. Waves on a rope are reflected by a wall.

Echoes are reflections of sound waves. You will gain a great deal more detailed experience with the reflection of sound waves in the upcoming experiments.

Specular Reflection

Specular reflection is a term referring to "mirror-like" reflections. If a wave is reflected from a smooth surface, we find that its *angle of reflection* is equal to its *angle of incidence.* Light waves provide a familiar example. You can check this, at least roughly, using a flashlight and a flat mirror. You may not have realized that, under certain conditions, the same effect occurs with water waves, sound waves, radio waves, and others. You will search for sound-wave reflections in the next experiment.

EXPERIMENT C-1. Observing Reflections of Sound Waves

The equation $s = \lambda f$, which relates the wave speed, wavelength, and frequency, allows you to find the speed of sound if you know λ and f for a periodic wave. This will be your "indirect" method for finding the speed of sound.

The wavelength λ can be measured with the aid of sound reflections. You will see how to do this in the next experiment. First, you should gain a little experience with these reflections by making a few simple tests. That is the purpose of the present work.

Setup

To do this experiment you will need apparatus for generating sound and for detecting sound. The way to fill both of these requirements is now very familiar to you. Set up the apparatus as indicated in Figure 58. You may use this equipment to search for reflections from any surface you wish. We suggest starting with a large plane surface, such as a 1-m square sheet of plywood or hardboard. Start with the oscillator frequency set at 1000 Hz.

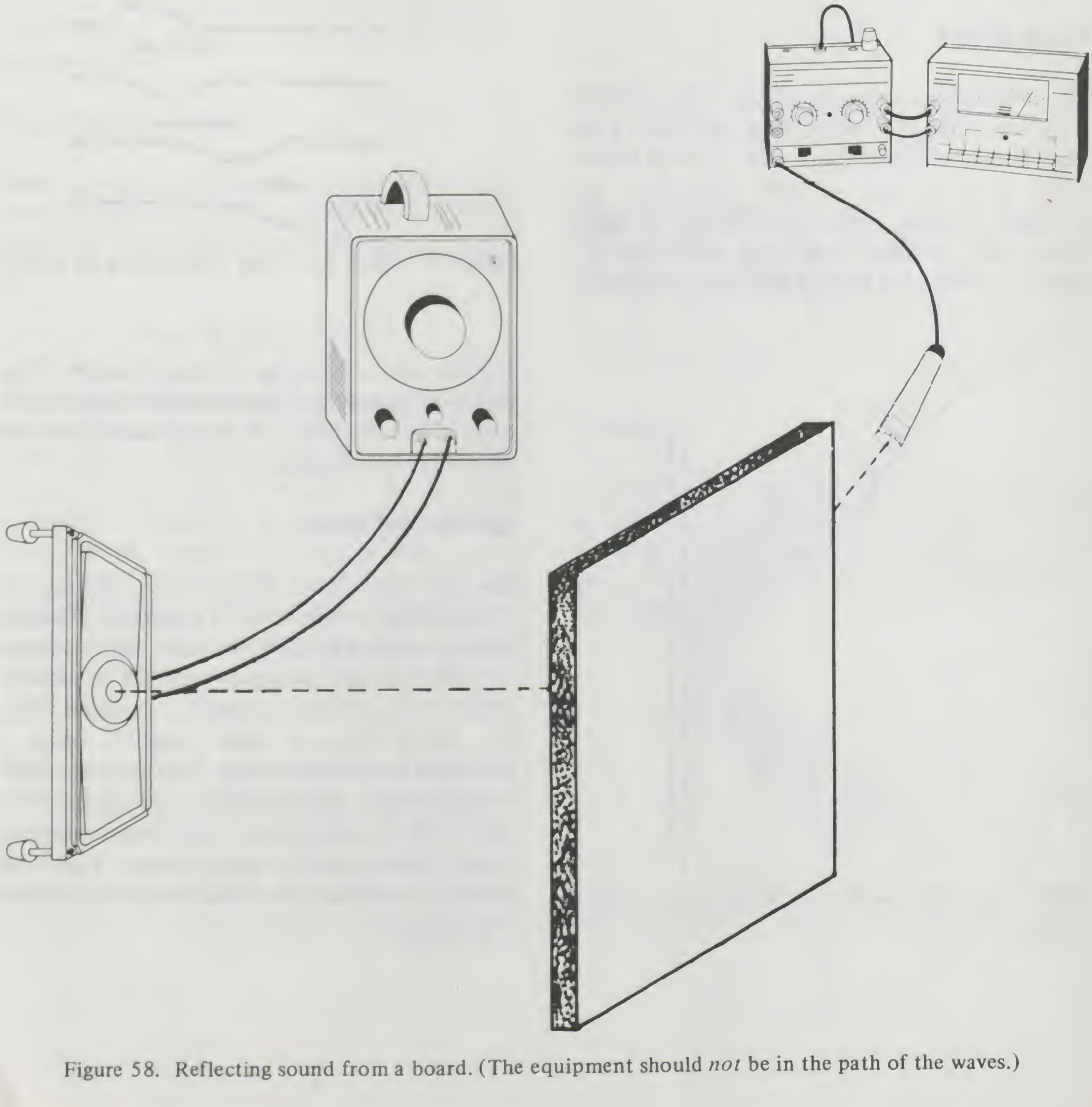


Figure 58. Reflecting sound from a board. (The equipment should *not* be in the path of the waves.)

Procedure

The work here should proceed quite rapidly. However, you may wish to try several things according to the materials and time available. The following steps are offered mainly as suggestions. When you think that you have a reasonable picture in your mind of how sound reflections operate, you should go on to the next experiment.

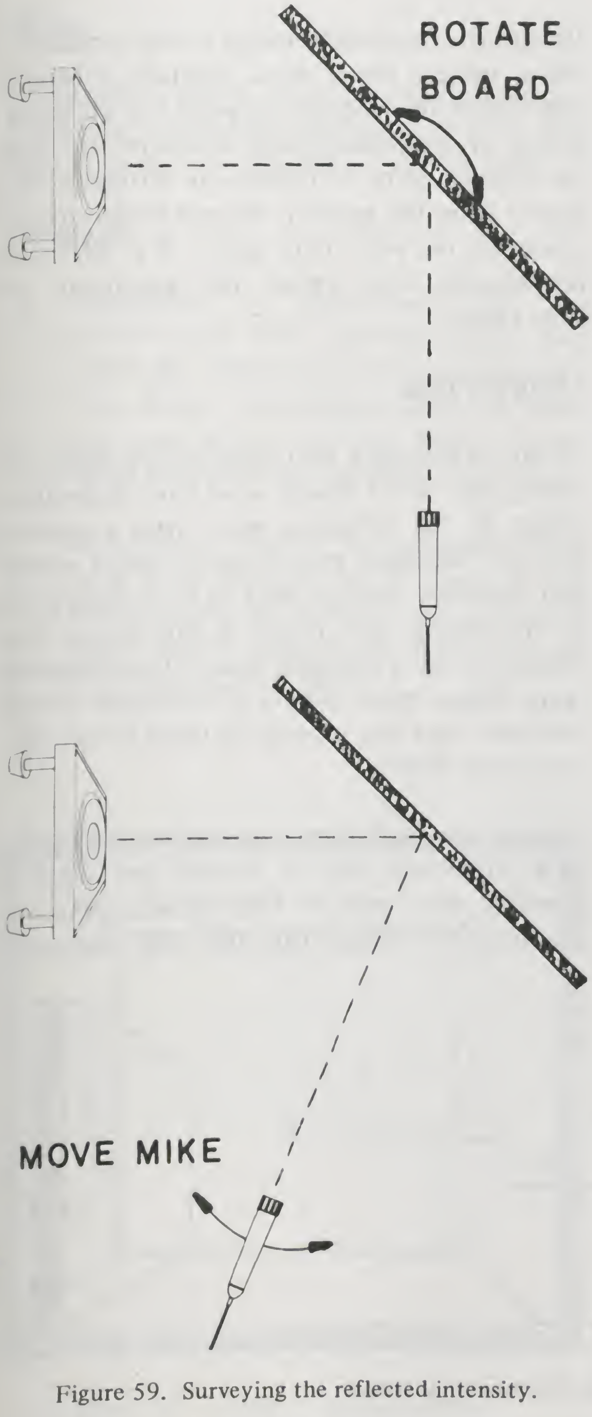


Figure 59. Surveying the reflected intensity.

1. *Specular Reflections.* With the speaker mounted in a fixed location, and the board located a meter or so in front of it, you can vary the angle of the board and the position of the microphone. You should be interested at first in simply surveying the pattern of reflected sound intensity, without recording any numerical data. Choose your angles from what you know of specular reflection with light and what you would expect for sound. The drawings suggest some possibilities. Look for maximum intensity at the microphone.

2. *Absorption.* How would you expect the nature of the board's surface to affect the reflections? Compare the results you obtain using the plywood board with the results you obtain using various materials stretched over the surface of the board. Try fiber glass building insulation, fabric, heavy fabric, or other available materials.

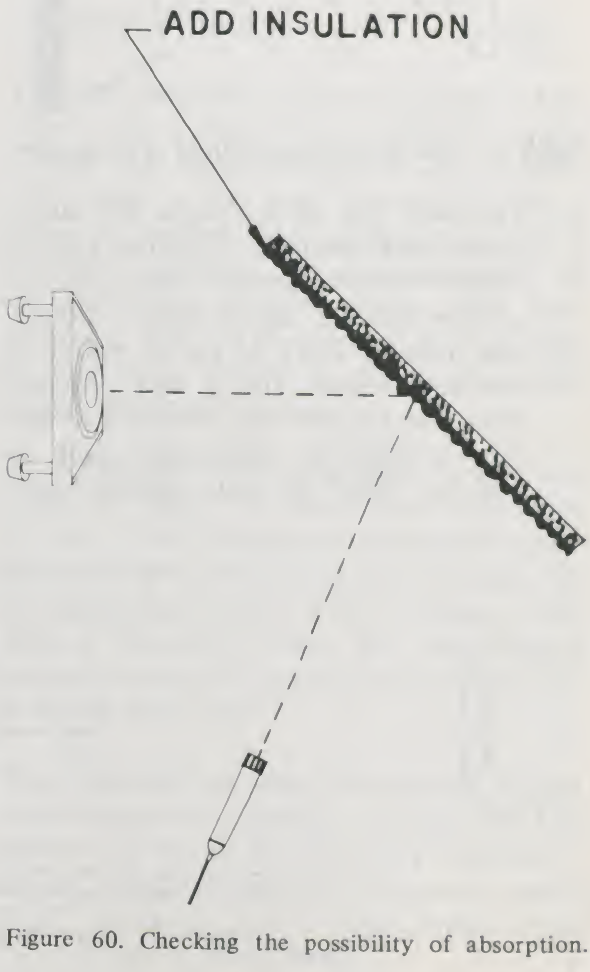


Figure 60. Checking the possibility of absorption.

EXPERIMENT C-2. Using Reflections to Measure the Wavelength

Procedure

1. Without using a reflector, refresh your memory of the behavior of the signal strength as the microphone is moved along the axis of the speaker. The strength should decrease smoothly as the mike is moved away from the sound source, as in Experiment B-2.

2. Add the reflecting board, to reflect the sound back toward the speaker. Use a frequency of 1000 Hz. Between the speaker and the board the microphone will now pick up both the original sound and the reflected sound.

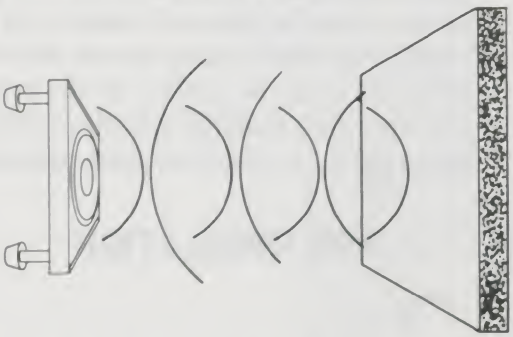


Figure 61. Add the reflector in front of the speaker.

3. Carefully and slowly move the microphone back and forth, as before. Do you observe anything unusual? *See if you can observe increases and decreases of intensity,* roughly every 15 cm of motion of the microphone. This is very different from the behavior you observed earlier. Try a different frequency, such as 500 Hz. What do you observe now? About how far apart are any two positions of maximum intensity?

Taking Data

Choosing convenient frequencies and proceeding as in Step 3, you should now record data from which you will later be able to calculate the speed of sound.

Record the frequency setting of the oscillator. Then, using a meter stick, carefully measure and record the distance between two adjacent points of minimum sound intensity. Do this for several pairs of minima at different distances from the speaker. Record these various examples on your data sheet. Try different frequencies, and repeat the experiment at least twice.

Using the Data

In this experiment, you observed the effect of having the direct sound wave from a speaker added to the reflected wave from a nearby surface. Whenever two or more similar waves add together, there is said to be *interference* of the waves. The result in this case is the formation of a *standing wave.* This standing wave shows many points of decreased sound intensity, and the spacing of those points can be used to find λ.

Shortly you will learn more about the topic of interference, why it occurs, and what a standing wave really is. You will also calculate the speed of sound from the data obtained.

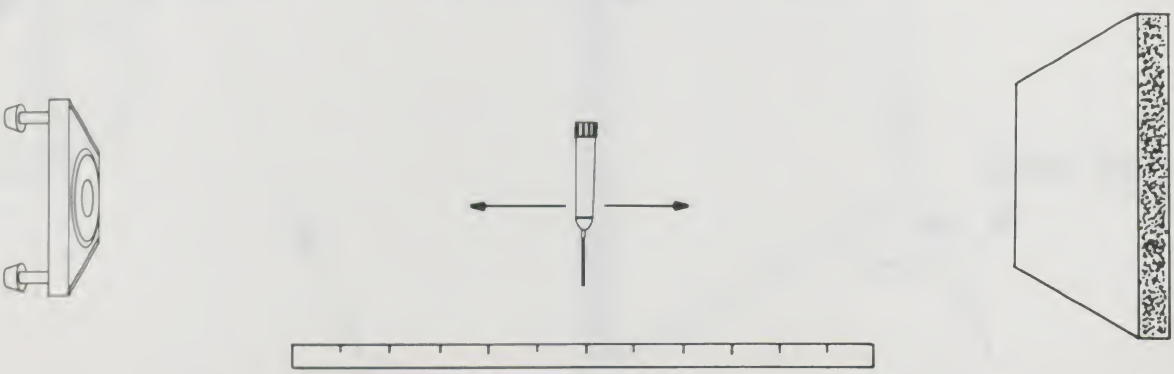


Figure 62. Locate the points of minimum intensity.

Before proceeding, notice that the experimental situation you just studied is not a totally artificial one. Reflections of sound waves occur naturally all around us. The rooms we live in are full of reflecting surfaces. A brief additional experiment will show you one result of this, and how it can be turned to advantage for some purposes.

Setup

Construct a model room as shown in Figure 63, using boards held together with tape. The boards should rest directly on the lab table.

Procedure

1. *Movement of microphone.* Fill the model room with 2000-Hz sound, and move the microphone slowly around. Do you detect resemblances between what you saw before and this more complicated situation with multiple reflections?

2. *Intrusion of objects.* Set the microphone in a fixed position on its stand in the room. Insert and remove your hand and other objects in the room. What happens to the signal intensity?

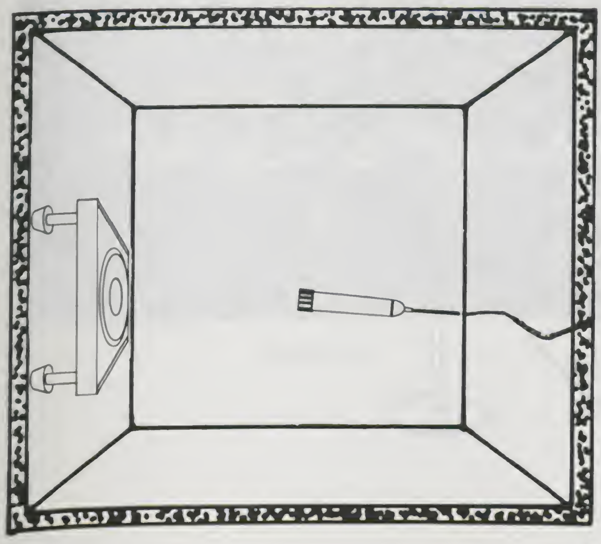


Figure 63. A model room (top view).

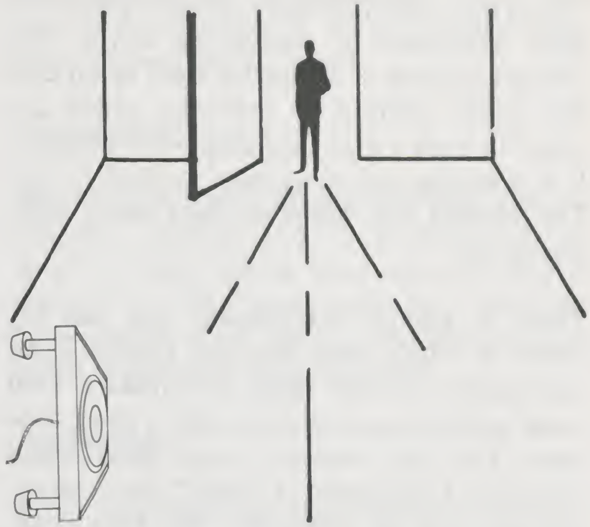


Figure 64. The standing wave pattern is changed by moving objects.

An Intrusion Alarm

You have discovered that the intrusion of an object into a room full of reflected sound changes the sound intensity at any specific point, such as the point where your microphone is located. This phenomenon has a practical application. You could design an intrusion alarm (burglar alarm) using your discovery. All that is necessary is to develop an electronic circuit to detect this signal change and sound an alarm.

A commercial alarm is available which uses reflected waves of ultrasound (frequency 40 kHz). The transmitter and receiver transducers are mounted in a single unit, which can be disguised as a book. While the signal in this case is inaudible (since the frequency is beyond the limit of human hearing) the basic principle is the same.

One question you might ask yourself is what sort of electronic detection circuit should be used in the alarm. Should it be a "threshold" circuit, which is triggered by a certain signal level? Or should it be a circuit which senses only *changes* in signal level?

EXPERIMENT C-4 (OPTIONAL). Echo Ranging with Sound Pulses

Up till now, you have been doing experiments with reflections of continuous waves. The indirect method of measuring wave speed uses the special pattern of intensity which can occur in such a case (*standing wave* pattern). It is a simple and convenient method to use. This method also works for light waves, radio waves, etc.

There is another experiment you can try which is much more like the kind of echo experiment you do when you make a loud noise outdoors and hear an echo a short time later. You can produce a very short sound with your loudspeaker, a "click" for example, and wait for its reflection. The time delay allows you to compute how fast sound travels if you know the distance involved. On the other hand, if you already know the speed of sound, this method can be used to measure the distance to a reflecting object.

This is an optional experiment because it is quite a bit more difficult than the earlier ones. On the other hand, it perhaps gives you some feeling for the difficulties which arise in trying to measure wave speeds directly.

Setup

Any sudden changes of voltage applied to a speaker will produce "clicks." A convenient source of click signals is a *square wave generator.* (Your audio oscillator may have a "square wave" setting.) To get distinct clicks instead of a continuous buzz, the frequency chosen should be below 10 Hz.

An amplifier is again necessary to boost the microphone signal to observe it on the oscilloscope. The overall setup is shown schematically in Figure 65. In practice, the mike should be as close as possible to the speaker at one side. Also, the reflecting board should be about one meter away, directly in front of the speaker. The mike should be pointed toward the board.

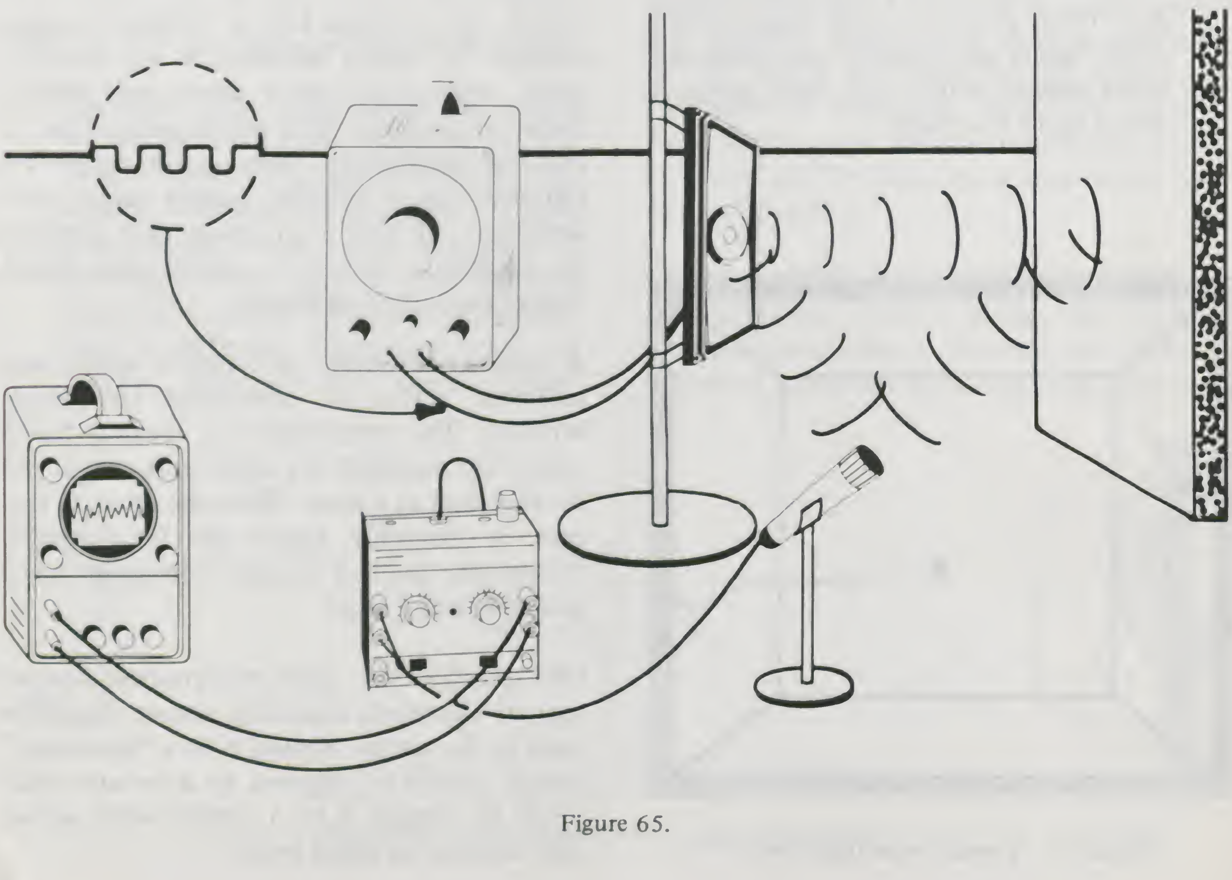


Figure 65.

Procedure

Set your oscilloscope for a sweep rate of approximately 1 ms/cm. The trigger control should be on the "internal" setting (INT), so that a sweep begins each time the scope receives a signal from the mike, caused by the *direct* click from the speaker. The *reflected* click will be received at a later time, depending on the distance to the reflecting board.

With the reflector positioned about one meter from the speaker and oriented carefully to get a maximum signal at the mike, and with the gain of the amplifier set high enough, you should be able to get a trace similar to the example of Figure 66, with two distinct "blips."

The first blip is the signal caused by the click traveling directly from the speaker to the microphone. The second blip is caused by the later arrival of the reflected click. The whole process repeats with each click from the speaker.

The exact shape of each blip is not important. However, the *time delay* between blips depends on the net path length from the speaker to the reflector and back to the microphone. It is the total indirect distance from speaker to mike *minus* the direct speaker to mike distance.

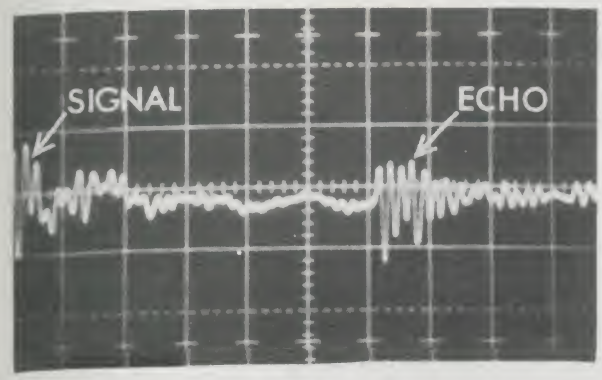


Figure 66.

In the table provided at the back of the module record the time delay for various distances, for use in calculations later. Can you find the speed of sound directly, given the delay and the distance?

An Example

The trace in Figure 66 was obtained with the reflector exactly 114 cm in front of the speaker. The distance from mike to speaker was 15 cm. The trace shows that the delay between the blips was 6.2 ms. What was the speed of sound in this example?

Applications

Sonar represents an application of some of the technology you have studied here. Pulses of sound are transmitted outward through water, and careful sensing detects echoes. The longer the delay before the echo is received, the farther away is the object that caused the echo. The speed of sound in water is known (about three times the speed in air), so the distance can be figured out.

Radar is based on the same principle as sonar but uses high-frequency radio waves. Again, the wave speed is known, so the delay of echoes can be used to locate the reflecting object, such as an airplane or speeding auto.

Bats use a sonar echo-location system to hunt insects for food. They are nearly blind, but hunt mainly at night anyway. They rely on sound to find their food and to avoid flying into obstacles. Their sonar system is very sophisticated and very effective.

A bat can fly very rapidly through a room full of obstacles and catch a single moth "on the fly." The frequency of sound bats use is above the range of human hearing, and is emitted in rapid short bursts when needed.

EXPERIMENT C-5 (OPTIONAL). Phasing Two Speakers

Stereo sound reproduction uses two similar speakers placed a short distance apart. In Section A (page 19) it was mentioned that *phasing* the speakers is important. If driven with the same signal, they should move in the same direction at the same time. Otherwise the sound from the speakers does not add, but tends to cancel by interference.

Set up two planar speakers side by side, and hook them up to a single oscillator. Use fairly long leads, so the speakers can be moved around. Use a low frequency, say 200 Hz.

With a mike right in front of the speakers, check the sound amplitude on the meter. Now reverse the leads to one speaker. What happens? Does the amplitude increase or decrease? Are the speakers "in-phase" or "out-of-phase"?

EXPERIMENT C-6 (OPTIONAL). Standing Waves without Reflections

In Experiment C-2, you saw that when traveling waves of the same frequency, but moving in *opposite* directions, are added together, they produce a pattern of maximum and minimum sound intensities called a "standing" wave. This is most often caused by surfaces or other boundaries which reflect the waves traveling in one direction back on themselves.

To produce standing waves *without* reflections, try to generate opposite traveling waves by using two separate speakers facing each other.

Place the speakers about 1 m apart on the same bench, and connect both at once, in series, to the same audio oscillator. Set the frequency to about 1000 Hz.

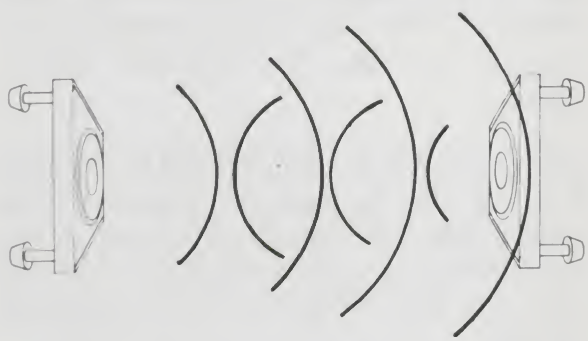


Figure 67. Producing standing waves without reflections.

Hook up a mike to the converter and meter as before, and move it back and forth between the speakers. Can you see evidence of a standing wave pattern? If so, how does the spacing of nodes compare to what you found in Experiment C-2 for the same frequency?

EXPERIMENT C-7 (OPTIONAL). Producing Beats

If possible, use two oscillators to drive the speakers at *slightly different* frequencies. What becomes of the standing waves? What happens when you adjust the frequencies so that they are almost exactly equal, but not quite? This dramatic result can be easily heard, as well as observed on the meter if the frequencies are close enough. Use some effort to see how well you can adjust the two oscillators so that the frequencies become equal. Repeat this in a few frequency ranges, for example, near 200 Hz, 1000 Hz, and 5000 Hz.

When two similar waves of slightly different frequencies are added together, they produce *beats*. With sound waves, the beats produce a "warbling" effect which can give the impression of a new note, the "beat note." The frequency of the beat note is the difference between the two "parent" frequencies.

Beats play an important role in music, because, when two separate tones are sounded together, strong beats give a feeling of "dissonance." Some modern music is characterized by extreme dissonance.

How Can You Describe Sound Waves?

It is important to be able to describe waves accurately. Only in that way can you make accurate predictions of their effects—such as how long they require to travel a certain distance, or what happens when two or more waves traveling in the same region interfere with each other.

To begin, you can get some ideas about how to describe pressure waves in air (sound waves) by considering the behavior of waves on a long stretched rope.

A Traveling Wave

The illustration shows an artist's conception of a series of photographs taken as a wave on a rope travels away from the source (a waving hand). Such a series of pictures make up a "motion picture," and could be projected to re-create the motion of the waves. The pictures should be viewed in order from top to bottom.

Consider for a moment the *shape* of the wave. It is drawn with a very simple shape: that of the mathematical *sine curve.* You may have learned something about the sine curve already in studying trigonometry. It is related in a simple way to the sine of an angle.

However, if you are not already acquainted with the sine curve, don't worry. The important thing for now is just to look at the figure and recognize that it looks fairly reasonable. It turns out that many different waves in nature have almost exactly this shape.

Note especially that the shape of the wave is the same in each snapshot, but the wave is slightly offset to the right in each successive picture. *The whole pattern shifts but does not change shape.* Such a wave is called a *traveling wave.*

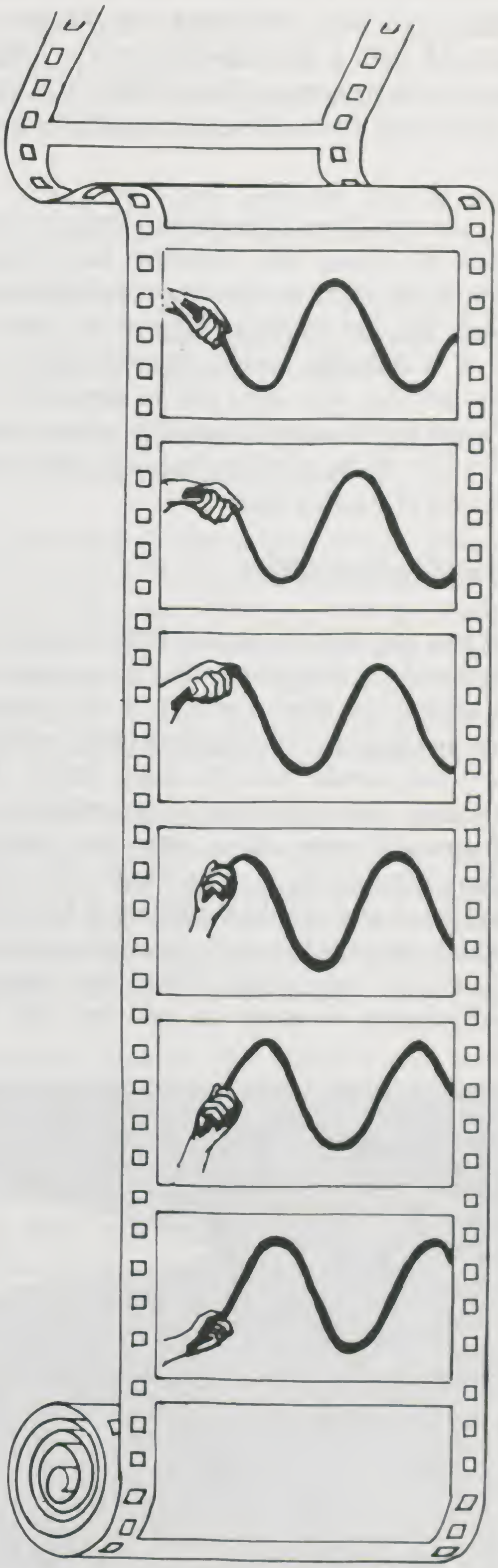


Figure 68.

VISUALIZING WAVE MOTION

In order to study the movement of the wave, imagine you are projecting the images of Figure 68 with a movie projector. The images flash on a screen one after another. In Figure 69 they have been drawn all together, overlaid.

You can see from this figure that, in each instant of time, the traveling wave shape moves to the right by the same small distance. In real life the motion happens so quickly that it is difficult to see. But here you can easily see that the wave can be described by the same mathematical function all the time, but it just shifts a bit in one direction with each unit of passing time.

A Traveling Sound Wave

Now you can take the description of waves on a rope and apply it to sound. For example, at any instant of time a graph of air pressure versus position for the simplest sound wave (a pure tone) would have the basic shape of a sine curve. So it too can be *described* by a mathematical sine curve, this one relating *pressure* and *position* (see Figure 52). The sound wave also would be shifted a bit in the direction of travel at each succeeding instant of time. If you could make the pressure measurements as easily as you can see the rope, you would see that the pressure varies in the same way as does the displacement of the rope.

Standing Waves: A Mini-Experiment

Now we are going to look at a more complicated situation. This is like the situation you studied when you reflected sound waves back on themselves.

Tie a rope or piece of tubing to a support. Hold the rope fairly taut, then snap your wrist sharply upward and back to produce a single wave pulse, as in Figure 57. Observe the behavior of the pulse as it reflects from the fixed end of the rope. If the pulse is strong enough to return to your hand, is it reflected again there?

Now move your hand up and down repeatedly to produce a periodic wave, more or less as shown in Figure 70. You should find, by varying the rate at which you shake the rope, that you can create the wave patterns called *standing waves.* These can be identified by the presence of points on the rope which move very little. They are the *nodes.* This effect is startling. The behavior of the rope now represents the *sum* of the original traveling wave and the reflected traveling wave. How must you change the rate of shaking in order to add another node to the pattern?

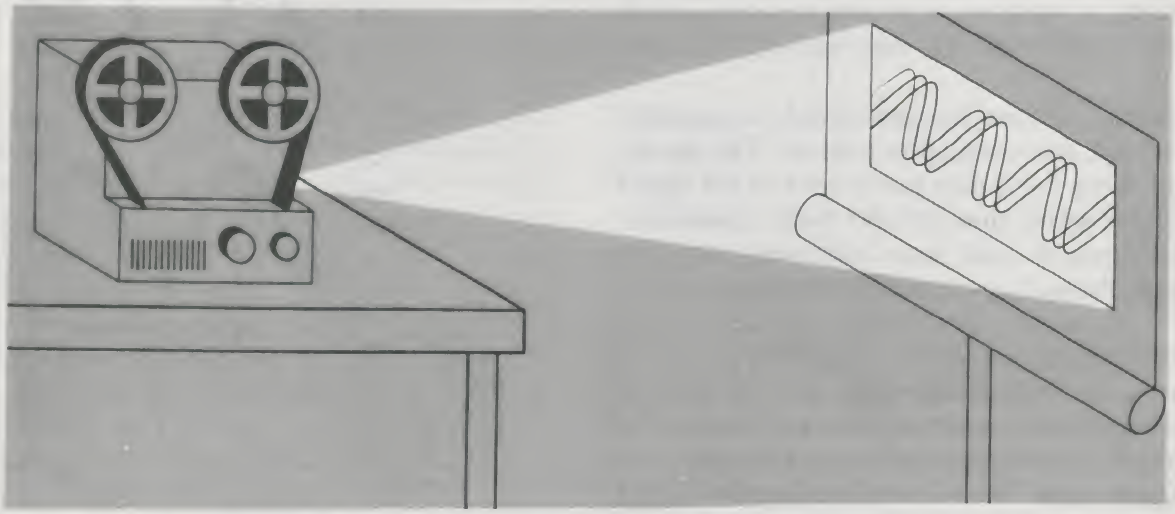


Figure 69. A traveling wave "motion picture," with successive positions of the wave all shown, overlaid.

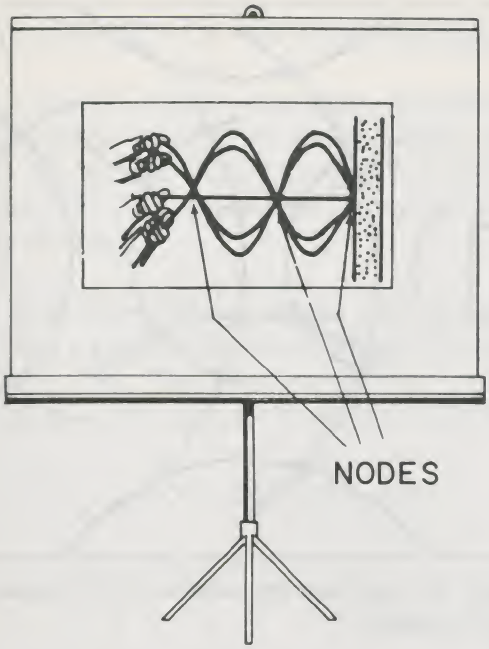


Figure 70. Standing wave.

Notice especially that a standing wave does not appear to move or travel anywhere. It vibrates in place. That is the reason for the name. Standing waves are caused by the addition of identical waves moving in opposite directions to produce a wave which "stands still." (Of course, the rope is not standing still, but the wave is.)

Do not be discouraged if you have difficulty seeing how the two traveling waves add up to form a standing wave. It requires thought and examination before one is comfortable with this idea. In the graphical exercise outlined next, you can work out how this takes place in the example of the rope. Even after doing that, you may have to think some more to understand how a standing wave takes place in air.

Graphical Addition of the Two Sine Waves (OPTIONAL)

The drawing of Figure 71 below shows several cycles of a typical wave on a long rope. You can add this wave *graphically* to a second wave of the same type if you:

1. *Copy the sine wave on tracing paper,* including the baseline shown.

2. *Shift the paper to the left slightly and trace the wave again.* The copied baseline must remain over its original.

3. *Sketch the sum wave* by adding the first two waves point by point.

If you measure the height of one wave over the baseline at a point and find that it is 7 mm, while over the *same point* the height of the other wave is 4 mm, then the height of the sum wave over that point is 11 mm. (The baseline is "zero" for each wave. Where a wave is *below* the baseline, it is *negative.*)

The sum you obtain is another sine wave with a different amplitude. If you repeat the construction a few times, using various shifts, you will find that while the *amplitude* of the sum is always changing, its zeroes (nodes) remain midway between zeroes of the added waves.

You can think of the added waves as two waves moving left and right on the same rope. If their zeroes move in opposite directions at the same speed, the nodes of the sum remain fixed: i.e., the two waves add up in such a way that some points in space are nodes, where no motion of the particles occurs.

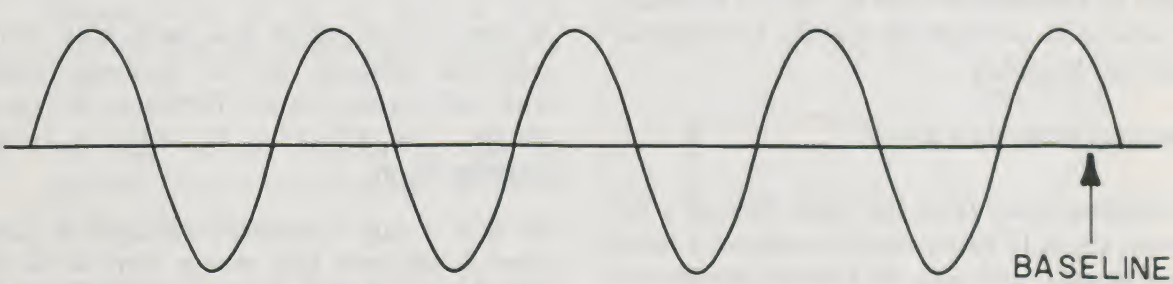


Figure 71. Sine wave for the use in the graphical exercise.

INTERFERENCE

Sound Added to Sound

You now have a better idea how cancellation can occur when two waves occupy the same location. Whether they are sound waves or waves on a rope, the basic point is that one wave may be going "up" while the other is going "down." The result is called *destructive interference.* The total wave is reduced at points of destructive interference.

The opposite effect is called *constructive interference.* Here, both waves are going up or down together. The total wave size is increased at points of constructive interference.

As you know from earlier work, what goes up or down in a sound wave is the pressure. Thus if two sound waves interfere destructively at some location, the pressure variations are smaller there. This happens in a standing wave. A microphone shows the effect, as you found in Experiment C-2.

Calculating the Speed of Sound

By taking your standing wave data from Experiment C-2, and knowing that the distance between nodes is one-half wavelength, $\lambda/2$, you can calculate the speed of sound in air. A worksheet is provided for that purpose at the back of the module.

Calculate the speed of sound from your data. How does your average value compare with the accepted value of 344 m/s? This is the value in relatively dry air at 20°C. The speed of sound in air depends slightly on temperature and humidity.

Standing Waves in a Room

Reflecting sound from the many walls of your model room (Experiment C-3) creates a more complicated situation. In a room you have a three-dimensional situation. The nodes are no longer half a wavelength apart, but the concept is the same.

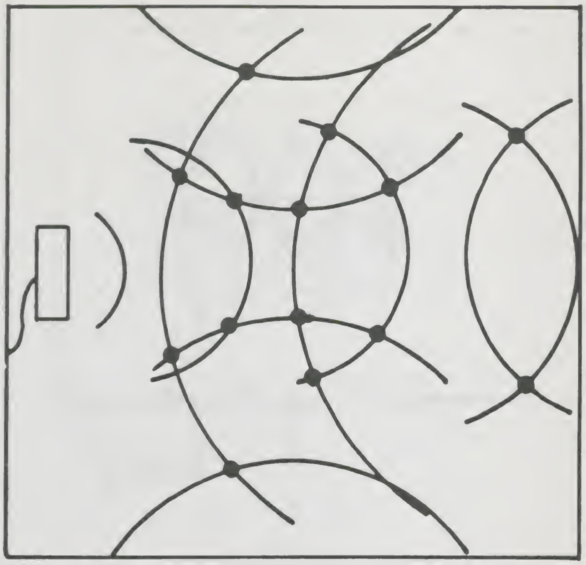


Figure 72. Standing wave patterns can be complex in three dimensions.

Actually, the "nodes" of standing waves in a room are not points at all, but rather whole surfaces called *nodal surfaces,** where the sound is relatively weak.

Live Rooms and Dead Rooms

The presence of standing waves in a room can be very annoying. Your appreciation of stereo music may depend on where you are in the room. If you are at a node for certain tones, then you will not hear those tones well.

This situation is greatly improved by reducing the amount of reflection of the sound waves. As you probably discovered, soft materials reflect less (and absorb more).

A "live" room, which is a room with many reflecting surfaces, can be deadened somewhat by drapes, carpet, furniture, and even people. The effect is to make a better listening room.

*If there is only incomplete cancellation of partial waves in the total wave pattern, then we do not observe any true nodes. Properly speaking, there are only *standing wave minima,* where the pressure changes are less than at nearby locations (as you actually observed in Experiment C-2).

MATHEMATICS OF WAVES (OPTIONAL)

Why are the simple waves being discussed in this section called "sine waves"? What have they to do with the numbers called sines of angles, which you may have learned about in trigonometry?

To help answer this question, look at Figure 73. It is a graph of the sines of angles between 0° and 360°. The values of the sine can be found in a sine table or by using a calculator.

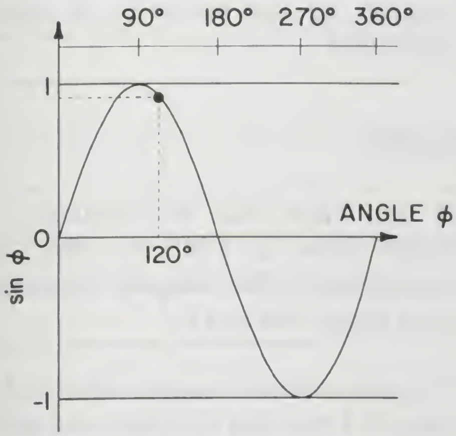


Figure 73. Graph of the sine.

To find the sine of any particular angle on the graph, simply look on the graph at the appropriate place, and estimate the value of sin ϕ from the vertical scale. For example, sin 120° is about 0.9. Sin 270° is −1.0, etc.

There obviously is a close similarity between this curve and the wave shapes you saw earlier. However, there are two apparent difficulties:

1. The sine of an angle is just a number, ranging from +1 to −1, instead of a quantity like air pressure or height of a rope.

2. An angle is not a position, such as distance from a sound source, distance along a rope, etc.

It is easy to cure the first difficulty. Just multiply the vertical axis in the graph by a factor A_0 which correctly gives the maximum size of the variable quantity in the actual wave. A_0 is called the *amplitude* of the variation. Thus, write:

$$A = A_0 \sin \phi$$

where A might represent deflection (for a wave on a rope) or pressure change (for a sound wave in air).

To cure the second difficulty, define an angle ϕ which is proportional to the distance. When the distance increases by one wavelength, this angle must increase by 360°. Then sin ϕ returns to its original value, and the wave repeats. A suitable *phase angle* for this purpose is:

$$\phi = 360° \, (x/\lambda)$$

where x is the distance and λ is the wavelength. At $x = 0$, $\phi = 0°$, while at $x = \lambda$, $\phi = 360°$.

A graph of

$$A = A_0 \sin [360° \, (x/\lambda)]$$

versus the distance x is shown in Figure 74. Because of the way in which the amplitude and phase angle are defined, this figure is identical to the original graph of the sine curve, Figure 73. The only difference is in the scales of the two axes. The shape is that of a pure sine wave.

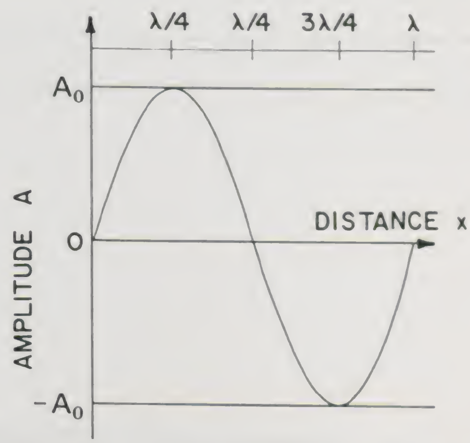


Figure 74. Graph of a sine wave. The cycle repeats each time x increases by an amount λ.

SUMMARY OF SECTION C

The experiments in Section C dealt with the properties of sound *waves* and their motion. Waves travel at characteristic speeds. Ordinary sound waves are *longitudinal, vibrational* waves in air, although sound can also travel in other gases as well as in liquids and solids.

Sound waves are reflected from surfaces. Echoes are a familiar example. A large flat surface can cause *specular reflection,* that is, "mirror-like" reflections.

Traveling waves *interfere* with each other. Under certain conditions *standing* waves are formed. In the simplest cases, the nodes of standing waves are spaced one-half wavelength apart. This fact can be used to measure the wavelength, and thus the speed, of sound, using the fundamental relation:

$$s = \lambda f$$

You found that such standing waves, which are present to some extent in any closed room, can be an annoyance in listening to music, or they can be useful in developing an intrusion alarm.

You went on to measure directly the speed of sound by an *echo-ranging* experiment. The physics of this experiment and the others you have done apply to a wide range of familiar objects, including musical instruments, communication devices, and navigation equipment.

QUESTIONS

1. If a standing wave is formed by sound waves whose wavelength is 30 cm, what is the distance between *nodes* of the standing wave?

2. Explain why the properties of a room affect the apparent performance of a hi-fi speaker.

3. Can one use a large, curved reflecting surface to focus sound waves, like light is focused by the mirror in a reflecting telescope?

PROBLEMS

1. If you know that the frequency of a sound wave is 1545 Hz, what is its wavelength? (The velocity of sound in air is about 344 m/s.)

2. If an echo returns from a reflector 15 cm away 0.5 ms after the pulse was emitted, what is the speed of sound in the medium?

3. A guitar string is 1 m long and is stretched tightly enough to produce a tone of 400 Hz when plucked. What is the speed of a traveling transverse vibrational wave on that string? (Hint: Both ends of the string must be nodes of the standing wave pattern.)

EXPERIMENT A-2. Measuring Motion of the Diaphragm

Data Table

Trial #	Current (Amperes)	Position (Arbitrary Units)	Comments

(Distance from mirror to screen: __________.)

EXPERIMENT A-3. Measuring the Force of the Voice Coil

Data Table

WORKSHEET: Calculating the Magnetic Field

Make three separate calculations, using different forces from your data, one at a high current, one at low current, and one in between. Do you get about the same result each time? Which do you think should be most accurate?

MEASURING QUANTITIES FOR THREE DIFFERENT VALUES OF CURRENT:

Use the data from Experiment A-3:

Weight (grams)	Weight (kilograms)	Weight (newtons)	Current (amperes)
________	________	________	________
________	________	________	________
________	________	________	________

THE FORMULA:

($1\text{ g} = 10^{-3}\text{ kg}$; 1 kg weighs 9.8 N)

$$F(\mathrm{N}) = B(\mathrm{T}) \times L(\mathrm{m}) \times i(\mathrm{A})$$

Therefore:

$$B = \frac{F}{3.2\text{ m} \times i} \quad \text{(in teslas)}$$

CALCULATED FIELD VALUES:

Field Value #1:

Field Value #2:

Field Value #3:

EXPERIMENT B-1. Measuring the Pattern of Radiation

Data Table

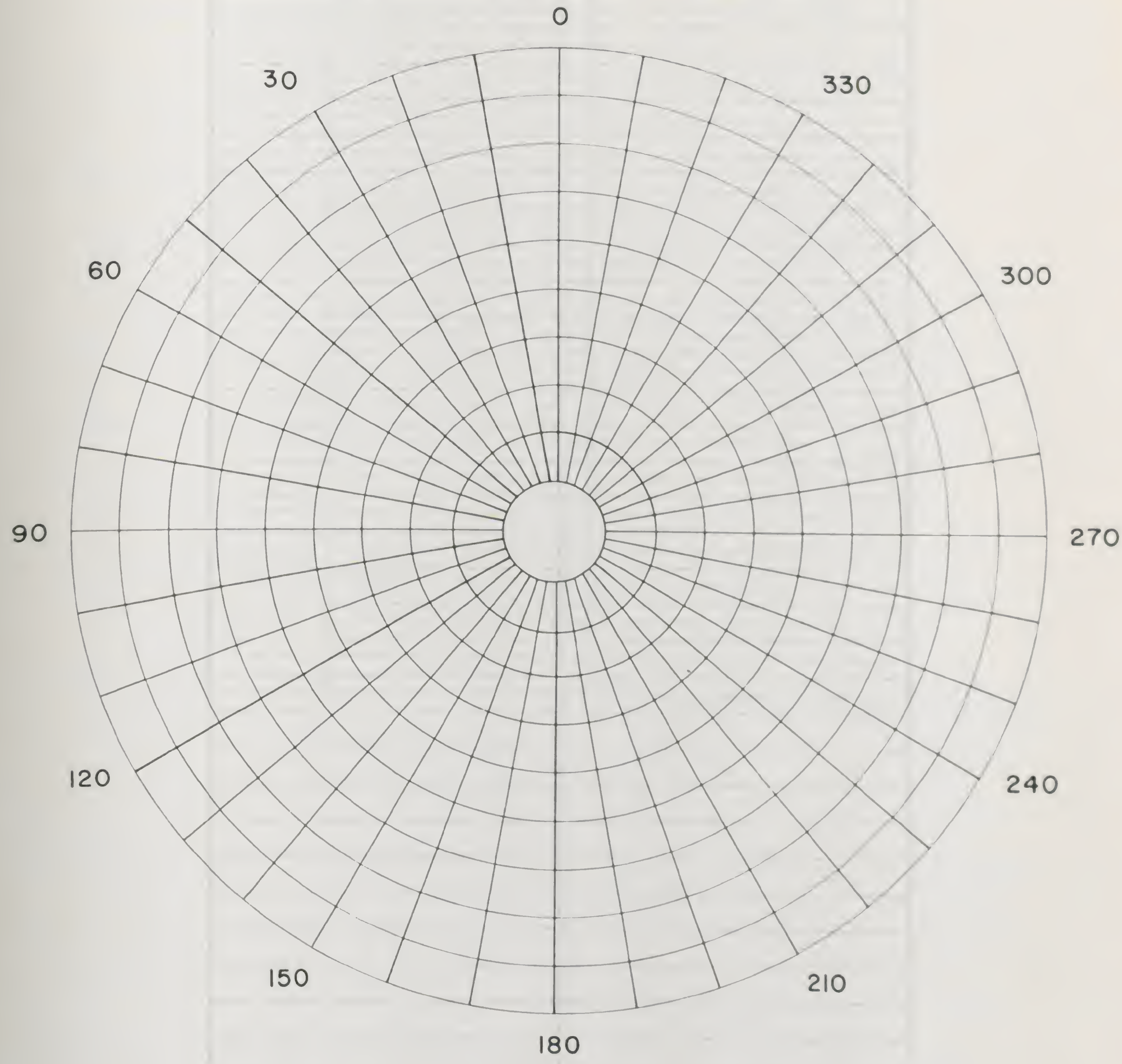
Microphone output: ______________ .

Oscillator frequency: ______________ (about 400 Hz)

Angle	Distance		
0°			

EXPERIMENT B-1. Measuring the Pattern of Radiation

POLAR GRAPH PAPER



Microphone output: ________________.

EXPERIMENT B-2. Measuring the Effect of Distance

Data Table

EXPERIMENT B-4. Directional Character of the Mike

Angle	Voltage	

EXPERIMENT C-2. Using Reflections to Measure the Wavelength

Data Table

FREQUENCY

1. ______________________________

FREQUENCY

2. ______________________________

FREQUENCY

3. ______________________________

DISTANCE BETWEEN MINIMA

____
____
____
____
____
____
____

DISTANCE BETWEEN MINIMA

____
____
____
____
____
____
____

DISTANCE BETWEEN MINIMA

____
____
____
____
____
____
____

EXPERIMENT C-4. Echo Ranging with Sound Pulses

Data Table

Distance (m)	Time Delay (ms)

WORKSHEET. Calculating the Speed of Sound

Transcribe your data from the data sheet for Experiment C-2 into the spaces below and perform the computations.

	Distance between Nodes				Wavelength	Wavelength		Frequency		Velocity
1.	______	X	2	=	________	________	X	________	=	______
	______	X	2	=	________	________	X	________	=	______
	______	X	2	=	________	________	X	________	=	______
2.	______	X	2	=	________	________	X	________	=	______
	______	X	2	=	________	________	X	________	=	______
	______	X	2	=	________	________	X	________	=	______

You can also calculate the speed of sound from the optional experiment with pulse echoes. That experiment represents a direct measurement of velocity. The sound travels out and back with a *delay time t.* The distance from the speaker to the reflector is *D*, so the total distance traveled is 2*D*. From this you must *subtract* the direct distance, *d*, from the speaker to the mike.

Thus:

$$s = \frac{\text{net distance}}{\text{delay time}} = \frac{2D - d}{t}$$

Transcribe your data from the data sheet into the spaces given and perform the computation:

Net Distance		Delay		Velocity
______	÷	____	=	______
______	÷	____	=	______
______	÷	____	=	______

How does this result compare with the result above?

APPENDIX: The Oscilloscope

The oscilloscope is nothing more than a very fast "graphic machine." It is able to present graphs of voltages (usually on the vertical axis) against time or other voltages (on the horizontal axis). Thus the scope displays the size, shape, and frequency of an input signal. How the oscilloscope circuits are designed is not important for this module. Instead, we shall examine the functions of the external controls which are common to most general-purpose laboratory oscilloscopes. Because most oscilloscopes have additional features and because the naming of controls varies among oscilloscopes, the operating instructions for a specific oscilloscope should always be consulted before use. (The terms scope and oscilloscope are used interchangeably here.)

Some of the basic controls found on all oscilloscopes are listed below, along with a brief explanation of their functions. As you read the description of each control, you should refer to the one on your scope.

I. Screen Section

A. The AC ON/OFF control turns the oscilloscope on. It is usually combined with either the INTENSITY control or the SCALE ILLUMINATION control. Most scopes have a panel light to indicate when they are on.

B. The INTENSITY control changes the brightness of the spot (or trace). If a bright spot is left in one place on the screen for too long, it can "burn" away the phosphor on the screen, leaving a permanent dark spot. For this reason, the intensity is usually kept as low as is consistent with good viewing.

C. The FOCUS control adjusts the sharpness of the trace. It should be adjusted to give the narrowest line possible.

D. The VERTICAL and HORIZONTAL POSITION controls change the up-down and left-right positions of the beam. On some scopes, you may find a BEAM FINDER button, which allows immediate location of a beam which is "off the screen." After finding the beam with the beam finder, you can then use the positioning controls to move the beam to the center of the screen.

II. Vertical Amplifier Section

E. The voltage to be analyzed is usually connected to the VERTICAL INPUT. If there is a *probe* provided with the scope, be sure to check its *attenuation factor.* Most probes are designed to reduce the voltage input to the scope by a factor of ten or more. Any calculation of voltage should include this factor.

F. The VOLTS/CM (also called VERTICAL SENSITIVITY) control allows measurement of the size of a voltage signal. The face of most oscilloscope screens is covered with a grid which is ruled in centimeters. This grid is used to read the voltage of a signal. For example, suppose the trace is 2.5 cm high and the VOLT/DIV is set on 10 VOLTS/CM. Then the voltage input to the scope is 2.5 cm X 10 VOLTS/CM or 25 V. (On many scopes, a concentric control knob allows the vertical sensitivity to vary. For calibrated readings this knob must be properly set.)

III. Time Base Section

G. The TIME/CM control (also called the SWEEP RANGE SELECTOR) varies the rate at which the trace "sweeps" across the oscilloscope screen. This feature is necessary to allow measurement of the duration of a signal or of the frequency of the signal. For example, if a voltage pulse is 4 cm long and the TIME/CM switch is set on 20 ms/cm, the time required for the pulse is 4 cm × 20 ms/cm or 80 ms. (On many scopes a *multiplier* switch allows the calibration time to be changed by some factor.)

H. A SYNC (synchronization) control is part of the time base section on many inexpensive scopes. Its purpose is to cause the sweep voltage to always begin at the same point of a recurrent signal so the signal appears to "stand still." The SYNC control is turned until the pattern on the screen appears stationary. If you cannot "stop" the signal, you may need to choose a different sweep rate.

I. The SYNC SELECTOR control selects one of three possible *modes of synchronization.* On INTERNAL SYNC, the input is synchronized with the internal sweep of the scope. On EXTERNAL SYNC, the input is synchronized with an external signal; on LINE SYNC, the input is synchronized with the AC-line voltage. This control is usually set to INTERNAL.

J. On more sophisticated scopes, the TRIGGER controls achieve synchronization. Triggering means that the horizontal sweep is caused to begin (triggered) whenever the input signal reaches a predetermined value.

The following exercise is to acquaint you with the controls of the oscilloscope you will use in the *Loudspeaker* module. You will need to consult the operating manual for your scope. If you have problems, consult your instructor.

LABORATORY EXERCISE

1. With the power off, examine the scope and locate the following controls or their equivalents.

   a. INTENSITY

   b. FOCUS

   c. VERTICAL INPUT TERMINALS

   d. VOLT/CM

   e. TIME/CM

   f. SYNC controls or

   g. TRIGGER controls

2. a. Find out, from the operating manual, how to turn off the internal sweep. This may be done by setting the TRIGGER LEVEL to a high position, or by turning the sweep selector to external (with no external input signal).

   b. Turn the FOCUS and INTENSITY controls fully counterclockwise.

   c. Set the VERTICAL and HORIZONTAL POSITION controls to the midpoints of their positions.

   d. Turn on the scope and allow it to warm up for at least one minute.

   e. Turn up the intensity until you see a spot on the screen. Do not turn the intensity higher than you need to. A glow around the spot indicates a too-high intensity. If no spot is visible, it is probably "off-screen." Use the positioning controls to move the spot onto the screen.

   f. Rotate the FOCUS control to see the effect, and adjust to get a very small bright dot on the screen.

g. Observe the effects of the HORIZONTAL and VERTICAL position controls and center the dot.

h. Set the SWEEP selector to an internal sweep and turn on a recurrent sweep. You will need to consult the operating instructions.

i. Observe the effect of changing the TIME/CM setting. Expand the dot into a line by increasing the sweep rate.

3. You are now ready to observe an input signal.

   a. Connect the voltage output of a sine- and square-wave generator to the vertical input of the scope. Connect the ground terminal of the generator to the ground input of the scope.

   b. If you have a triggered scope, set the TRIGGER to "auto." If the scope is not triggered, set the SYNC selector to INTERNAL.

   c. Set the TIME/CM switch to the slowest sweep rate (on the order of 1 s/cm). (Make sure the vertical is set for *calibrated* measurements.)

   d. Set the generator to give a sine wave with a frequency of 1000 Hz. Turn on the generator.

   e. Adjust the generator output (and/or VOLTS/CM control) to obtain a signal which is about 5 cm high.

   f. Observe the effect of increasing the sweep rate.

   g. Observe the effect of varying the frequency of the input signal.

   h. Change the output of the generator to "square wave" and observe the pattern.

   i. If you have a triggered scope, set the TRIGGER to INTERNAL. Adjust the TRIGGER LEVEL control until you get a stable trace which "stands still." Center the trace.

   j. Measure the length of a single pattern on the screen and use the TIME/CM setting to compute the duration of the pattern. How does your measured value compare with the value computed from the known input frequency? (Be sure the *multiplier* is set to 1, or that you include the proper factor.)

   k. Change the input frequency and repeat (i) and (j).

0-07-001732-8